

Article

Embedded Control Architecture with Supervisory Monitoring Functions for a Class AB Audio Amplifier

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Abstract

The present study introduces the development and implementation of a control and monitoring system that relies on software designed for a 2×50 W Class AB amplifier. The primary objective of this research is to integrate the hardware platform with the designed software, ensuring proper functionality and continuously monitoring the inputs, with further potential for integration into smart home applications. Digital control is preferred due to its flexibility, configurability, and ability to provide repeatable system-level behavior, rather than inherent improvements in analog audio performance. The developed architecture is designed to improve control flexibility and support calibration capability for a more precise control of the sound levels without the need for a hardware change. This research looks into challenges encountered in the design of a Class AB amplifier platform, in this case, the response of the amplifier over the defined audio bandwidth and a practical user interface. The proposed solution focuses primarily on firmware-based control, user interaction, system management, and supervisory monitoring of operational states.

Keywords: power amplifier; class AB amplifiers; embedded systems; volume control algorithms; microcontroller programming; OLED display; audio amplifier; microcontrollers

1. Introduction

The fundamental changes brought by technology integration over the years have led to major shifts within the audio systems sector. New-age audio systems have become increasingly sophisticated, which has led to a rise in demand for effective control systems that are functional, economical, and easy to use in complex systems.

Due to their performances in terms of audio quality and power efficiency, class AB amplifiers are very common for both, pro-audio and mid-level consumer audio system. These days, many manufacturers are replacing class AB amplifiers with class D in PA applications due to their increased efficiency [1].

If designed properly, it can achieve most of the fidelity of a class AB amplifier. However, in confined spaces, class AB amplifiers are preferred over class D amplifiers. There are techniques to improve the efficiency (class H, class G amplifiers) by using multiple supply voltages and selecting them according to the desired output range. This method helps to reach an efficiency similar to a class D amplifier with similar audio quality as a class AB amplifier [2,3].



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A fully analog amplifier is preferred by the audiophile consumers who consider digital elements introduce noise in the system that affects the sound quality. Engineers combine the signal processing theory together with physical electronic circuits, offering the comfort of a digital for the user while maintaining the quality of an analog amplifier. In fact, there are many other advantages of a digital system over an analog one, for example the mechanical wear of analog potentiometers that introduces imbalance or unwanted noise due to imperfect contacts. For this reason, digital potentiometers are superior, but with the drawback of resolution: ranges from 64 steps up to 256, and less commonly 1024; whereas the mechanical ones have a theoretical infinite resolution. By using digital potentiometers, the complexity of the circuit increases and requires level translators since it can accept only a certain voltage range and is not isolated. Digital systems are prone to noises if the communication signals are treated improperly in terms of speed and the physical construction of the circuit [4]. As long as the switching and digital frequencies are kept above the audible range (>25 kHz) and the crosstalk is minimized by keeping the analog and digital signals apart.

For this application, it is necessary a microcontroller that can handle analog to digital conversion, different communication protocol and to satisfy the required pin count to control all other modules, for this reason, ATMEGA2560 and ATMEGA1280 (Microchip Technology Inc., Chandler, AZ, USA) are suitable for this project, having UART, SPI, I2C, PWM, 86 programmable I/O pins and a 10-bit ADC. To increase the compatibility with other smaller microcontrollers (ATMEGA328, Microchip Technology Inc., Chandler, AZ, USA), an I/O extension is implemented using a shift register.

Digital potentiometers provide precise, repeatable audio gain control and eliminate the mechanical wear and channel mismatch commonly associated with analog potentiometers, making them suitable for embedded audio applications [5].

Rigorous calculations and estimations were performed prior to the physical build of the amplifier to meet its requirements in any scenario with prolonged lifespan. This means the real performances of the amplifier exceeds the nominal requirements (50 W per channel output power at $8\ \Omega$ load) without compromising the sound quality [6]. The experimental validation of this capability is presented and discussed in Section 4.

The main contribution of this work is not a novel audio-processing optimization algorithm, but the design and experimental validation of a firmware-based control and monitoring architecture for a 2×50 W Class AB amplifier prototype. The implemented system combines digital volume control, startup/shutdown sequencing, OLED-based user feedback, and embedded communication interfaces in a single integrated platform for embedded audio applications.

The manuscript therefore contributes a reproducible implementation study that bridges analog audio hardware and embedded control in a practical Class AB amplifier platform. From an architectural perspective, the proposed implementation belongs to the class of embedded control and monitoring systems in which firmware coordinates peripheral communication, state sequencing, and user interaction in real time. In the present amplifier prototype, this architecture is used to combine digital volume control, operational-state management, and embedded monitoring within a single mixed analog-digital platform.

From a system-level perspective, the novelty of the prototype lies in the coordinated interaction between firmware-controlled gain adjustment, relay-based power sequencing, state-dependent muting behavior, and real-time user feedback through a compact embedded interface.

To better position the present prototype within the state of the art, we expanded the discussion on digitally controlled gain architectures, embedded audio interfaces, and amplifier validation methods relevant to mixed analog-digital audio systems.

The primary objective of this work is the development and experimental validation of an embedded control architecture for a Class AB audio amplifier. In addition to control-related functions, the system incorporates supervisory monitoring capabilities intended to support operational awareness and system management. These monitoring functions are not intended to replace laboratory-grade measurement instrumentation, but rather to provide runtime status information for the embedded firmware.

Within this context, the present work addresses the design and validation of a firmware-based control and monitoring architecture for a Class AB amplifier, focusing on system-level integration rather than purely signal-processing optimization.

Recent research highlights the increasing integration of embedded control and digital technologies within audio systems. For example, fully digital amplifier architectures have been extensively investigated, demonstrating improved flexibility, reconfigurability, and system-level integration capabilities [7].

In parallel, audio monitoring and control have become important components of IoT-based applications, particularly in smart environments and acoustic monitoring systems. Several studies have proposed real-time noise monitoring systems using embedded platforms and cloud connectivity for applications such as occupational health and smart cities [8–10].

More recent approaches also incorporate machine learning techniques for acoustic sensing and source identification, further emphasizing the growing role of embedded intelligence in audio systems [11].

Recent research also explores advanced implementations of Class D amplifiers and embedded control architectures that improve efficiency and real-time performance. Fully digital Class D amplifier topologies provide enhanced flexibility and reconfigurability at the system level [7], while self-tuning mechanisms with digital feedback have been proposed to improve performance stability and audio quality [12]. In parallel, accurate timing behavior and interrupt latency remain critical aspects in embedded audio control systems, especially in multiprocessing environments, where dedicated measurement techniques have been developed [13].

Furthermore, modern embedded architectures based on RISC-V demonstrate improved determinism and low-latency behavior, which are essential for real-time audio processing applications [14]. Digital potentiometers continue to be widely used for programmable gain control in audio systems, offering precise and repeatable adjustment capabilities in embedded implementations [15].

Table 1 highlights the distinction between the authors' previous work and the present contribution. While [6] focused primarily on the design and characterization of the Class AB amplifier hardware platform, the present work extends the system through the development and experimental validation of a firmware-based control and supervisory architecture. The newly introduced functionalities include digital volume management, OLED-based user interaction, rotary-encoder navigation, relay-control logic, startup and shutdown sequencing, state restoration after power cycling, monitoring capabilities, and the associated firmware framework required to support these features.

Table 1. Comparison between previous work [6] and the present work.

Feature	Previous Work [6]	This Work
Class AB hardware platform	Yes	Yes
Analog characterization	Yes	Yes
Digital volume management	Yes	Yes
Firmware-based volume management	No	Yes
OLED user interface	No	Yes

Table 1. *Cont.*

Feature	Previous Work [6]	This Work
Rotary encoder control	No	Yes
Relay management	No	Yes
Startup/shutdown sequencing	No	Yes
State restoration after power cycling	No	Yes
Monitoring functions	No	Yes
Firmware architecture description	No	Yes
Experimental firmware validation	No	Yes

2. Proposed Embedded Audio Control Architecture and Methodology

The following section presents how software algorithms were developed and implemented in the design and construction of the 2×50 W Class AB amplifier, where software algorithms can be used in addition to hardware to enhance system functionality and ensure stable and controlled operation of the amplifier by providing precise volume controls and communications between different hardware components, in addition to power status control. The workflow was carried out without problems, and the various test cycles proved that the amplifier was designed and optimized under safe and diverse conditions. The aim is that the design and implementation approach complements and works in conjunction with the selected microcontroller and programming environment, as well as the interface display. Simplifying communication protocols improves performance and scalability, while also validating the system's implementation through preliminary testing and identifying key performance parameters.

2.1. Hardware and Software Selection

The microcontroller ATMEGA2560 was used for this project. It provides a wide range of peripherals, including UART, SPI, and I2C. For the present prototype, the ATMEGA2560 was selected as a suitable trade-off between available I/O resources, peripheral support, implementation simplicity, and compatibility with the selected control architecture. Additionally, the ATMEGA2560 provides us with a balance of cost efficiency and sufficient processing power for the application we want to develop for the intended amplifier application [16,17]. The ATMEGA2560 was selected primarily because the implemented prototype required a high number of general-purpose I/O lines, straightforward support for UART and SPI, sufficient timer resources for interface handling, and low integration complexity with the selected development environment. Although STM32 and ESP32 families offer higher computational capability and extended connectivity, the present application does not require advanced DSP processing or wireless functionality; therefore, the ATMEGA2560 provided an adequate trade-off between I/O availability, implementation effort, and platform familiarity for prototype-level validation. In Table 2 we compare the ATMEGA2560, STM32F103C8T6 and ESP32-WROOM-32 microcontrollers.

The development environment has been set up with Arduino Integrated Development Environment (IDE) for the microcontroller. We opted for the Arduino IDE in particular because of how easy it is to use, and for the toolset which simplifies the process of constructing the algorithm.

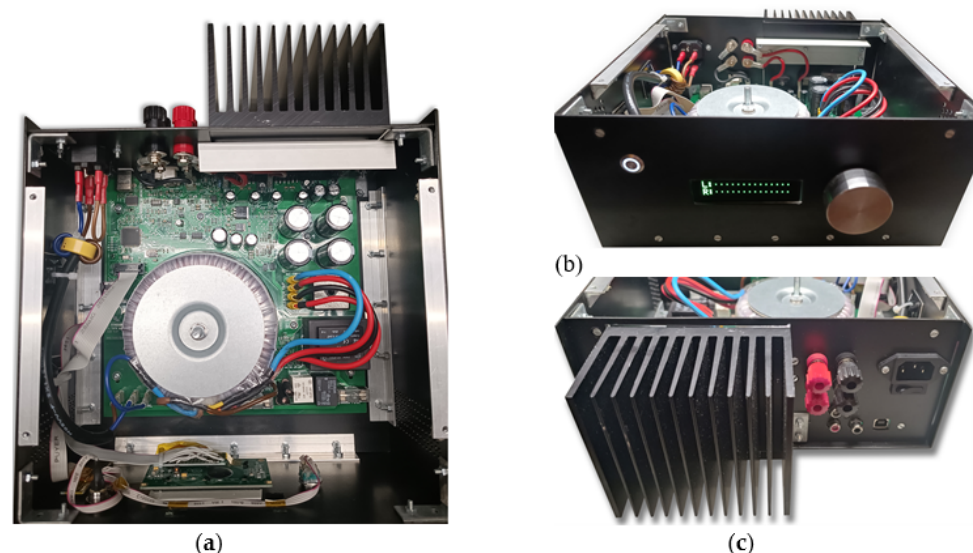
The bootloader of the Arduino together with the UART interface is used for the programming of microcontrollers in Arduino IDE [18]. This approach has been adopted because it is straightforward and effective for microcontroller interfacing where the need for an external programmer is eliminated for software uploads.

Table 2. Comparison of ATMEGA2560, STM32F103C8T6 and ESP32-WROOM-32 microcontrollers for the proposed amplifier controller.

Parameter	ATMEGA2560	STM32F103C8T6	ESP32-WROOM-32
CPU/max frequency	8-bit AVR/16 MHz	Cortex-M3/72 MHz	Dual-core Xtensa/240 MHz
Flash/SRAM	256 KB/8 KB	64 KB/20 KB	4 MB/520 KB
GPIO availability ADC	86 I/O lines 16-ch, 10-bit	up to 37 GPIOs 2 × 12-bit ADC	up to 32 GPIOs ADC supported
Serial interfaces	4 USART + SPI + I ² C	up to 3 USART + 2 SPI + 2 I ² C	UART + SPI + I ² C
Operating voltage	1.8–5.5 V	2.0–3.6 V	3.0–3.6 V
Wireless connectivity	No	No	Wi-Fi + Bluetooth/BLE
Prototype suitability	High I/O, simple 5 V integration	Higher performance, lower I/O count	Strong processing + wireless, but higher integration complexity

The Class AB Amplifier project was developed in C programming language since it is commonly used for embedded applications because of its efficiency hardware resource control, and portability.

The software engineering process comprises the setting of the configurations with the global variables that serve to control the operational states of the amplifier to achieve the desired optimal performance. The definitions involve the values of the parameters such as the rotary encoder setting values, menu codes, and VU meter limits that constitute special values for use in the assessment of the system's performance. Using these values, global variable definitions that clearly define the status of the system are achieved, allowing interaction between different components of the software. This way, the reliability of the system during operation will be guaranteed. The physical design and realization of the class AB amplifier designed includes its mechanical design and the interface elements used in the construction of the amplifier. They have been shown in Figure 1.

**Figure 1.** (a) Top, (b) Front, and (c) Rear views of the class AB amplifier.

The adoption of the OLED displays in embedded systems is widespread because of their superior attributes, which include high contrast, faster response, and higher level of readability over traditional LCD displays [13]. 2×16 alphanumeric OLED display, specifically Raystar Optronics REC001602EGPP5N00100, was chosen as the display for the interface [19,20]. These characteristics improve the user experience by allowing users to see more quickly and with greater clarity. However, it should be noted that OLED displays

are about twice as expensive as LCD displays. An 8-bit parallel communication method was initially considered to interface the OLED display. A 4-bit protocol was selected in order to provide a smaller number of lines of communication and an easier hardware setup. This choice allowed us to strike a good balance between having a strong interface and having the simplest hardware configuration possible. The Arduino system uses UART communication for interfacing with the PC and to program code. The FTDI FT232RL is a useful adapter, providing power for I/O pins (3V3 or 5V) and a 3V3 regulator for USB lines [18]. This adapter allows an easily accessible connection between the Arduino and development software.

2.2. Integration and Communication Protocols

Out of all the existing protocols, we settled on the already popular and simple UART protocol. UART uses only two wires (RX-TX), which allow for both receiving and transmitting data to the port. This makes asynchronous communication with the port possible. SPI and I2C were also considered in addition to UART. The choice was easier in this case, as UART was the easiest to implement, and by far the best-performing port. A smart approach to managing the state of the amplifier is to define certain global and local parameters, such as rotary encoder settings, menu codes, VU-Meter display limits, and other parameters used for controlling and monitoring the system.

In the implemented architecture, UART is used primarily for firmware upload, debugging, and host-side serial communication, whereas SPI is dedicated to the real-time control of the digital potentiometers responsible for volume adjustment. The OLED interface was implemented in 4-bit parallel mode to reduce wiring complexity while preserving deterministic display updates. For this reason, the revised manuscript avoids generic claims regarding a universally 'best' protocol and instead describes each interface according to its functional role in the prototype.

2.3. Experimental Setup Description

Figure 2 illustrates the experimental setup used to characterize the electrical and thermal behavior of the audio amplifier under test.

The amplifier is excited by a programmable signal generator, which provides the input audio signal connected directly to the amplifier input stage. The amplifier output is routed to a configurable resistive load, enabling controlled power dissipation during the measurements.

The output voltage and current are measured simultaneously using a digital oscilloscope. Two oscilloscope channels (CH A and CH B) are employed for voltage measurements, while the load current is obtained indirectly by sensing the voltage drop across precision shunt resistors of 10 mΩ placed in series with the load. The purpose of using Kelvin connections is to reduce the impact of parasitic resistances. The obtained values for voltages and currents can be employed for calculating instantaneous and average output power delivered to the load. Thermal properties are estimated in parallel with electrical characteristics. Load temperature is measured by means of a digital multimeter in connection with a thermistor installed directly on the resistor that represents a load. The temperature of the amplifier power stage is recorded through a second digital multimeter with a thermistor attached to the heatsink.

The amplifier is powered by a dedicated dual-rail power supply. Linear voltage regulators providing ± 5 V and 2.5 V are used to supply the low-level and control circuits independently from the main power stage. The power supply integrity is estimated by measuring its voltage ripple with the help of an auxiliary oscilloscope. The experimental

setup makes it possible to simultaneously estimate electrical output data, power supply behavior, and temperature.

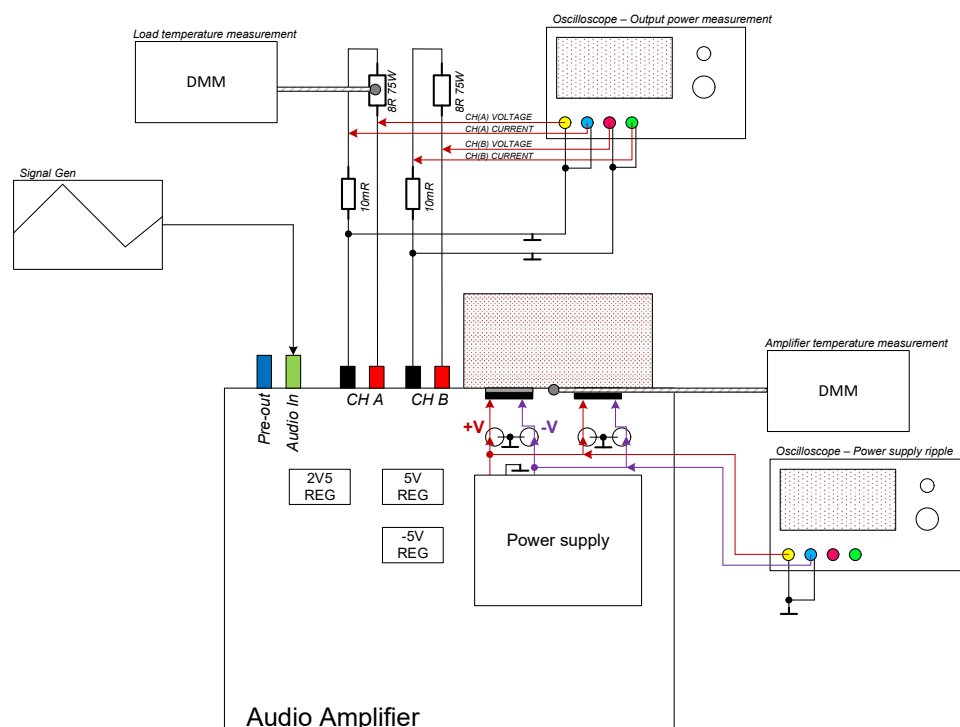


Figure 2. Brief overview of experimental setup.

Figure 3 presents the system-level block diagram of the software-controlled Class AB amplifier.

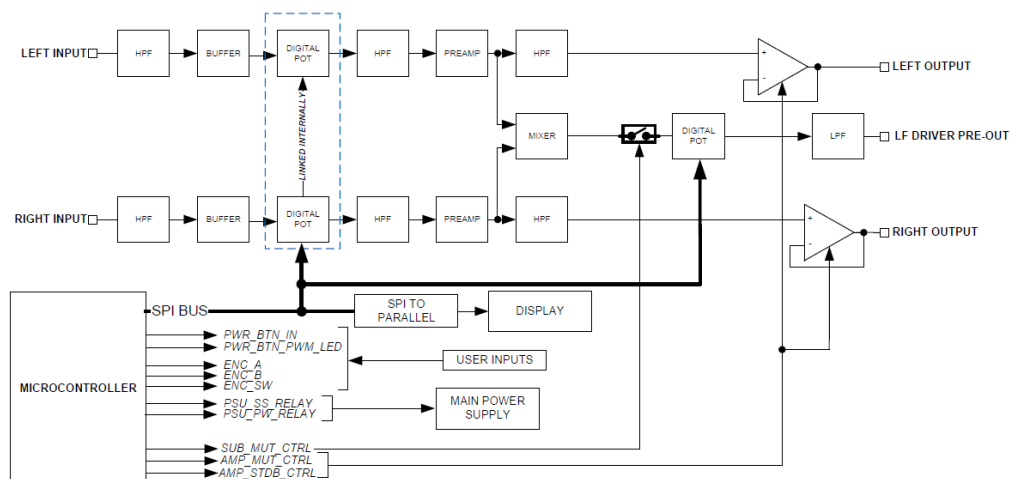


Figure 3. System block diagram.

This figure illustrates the interaction between the microcontroller, digital potentiometers, user interface, communication interfaces, and power management circuitry. This diagram provides an overview of the functional architecture and data flow between hardware and software components.

The monitoring subsystem performs supervisory observation of the amplifier operating state, including user interactions, stored configuration parameters, relay status, and system operating conditions used by the firmware. The monitoring functions are intended for runtime supervision and control support and should be distinguished from the external

measurement instrumentation employed for electrical and thermal characterization during laboratory testing.

2.4. Comparison with State of the Art

Table 3 summarizes the main characteristics of the proposed firmware-controlled architecture and compares them with representative systems reported in the literature. The selected references address different aspects of embedded audio systems, including fully digital amplifier architectures [7], digital-feedback-based amplifier control [12], and programmable-gain circuits employing digital potentiometers [15]. As shown in Table 3, the proposed implementation combines several functions within a single experimentally validated prototype, including embedded control, digital volume adjustment, relay-based startup and shutdown sequencing, state restoration, user-interface integration through an OLED display and rotary encoder, and monitoring functionality. In contrast, the reviewed works primarily focus on specific aspects of amplifier design or gain-control implementation. The comparison also shows that quantitative system-level indicators, such as control latency (95 ms) and startup timing (480 ms), are reported in the present work, while equivalent metrics are generally not provided in the selected references. Furthermore, the proposed architecture integrates monitoring and user-interaction capabilities together with control functionality, whereas the cited studies mainly focus on amplifier topologies, feedback strategies, or gain-control circuits. Direct quantitative comparison is limited because the considered works target different applications and do not report the same set of performance indicators. Nevertheless, the comparison places the proposed architecture within the context of existing embedded audio-control solutions and highlights its system-level integration of control, monitoring, and user-interface functions.

Table 3. Comparison of the proposed architecture with related embedded audio-control systems reported in the literature.

Feature	This Work	[7]	[12]	[15]
Embedded control architecture	Yes	Partial	Yes	No
Digital volume/gain control	Yes	No	No	Yes
Digital potentiometer based control	Yes	No	No	Yes
Relay-based startup/shutdown sequencing	Yes	No	No	No
State restoration after restart	Yes	Not reported	Not reported	Not reported
User interface integration	OLED + Encoder	No	No	No
Real-time user interaction	Yes	No	Partial	No
Monitoring capability	Yes	No	No	No
Startup transient management	Yes	No	Partial	No
Experimental prototype validation	Yes	Yes	Yes	Yes
Reported control latency	95 ms	Not reported	Not reported	Not reported
Reported startup timing	480 ms	Not reported	Not reported	Not reported

3. Software Implementation

This section focuses specifically on firmware implementation details, while hardware architecture and measurement methodology are described separately in the preceding sections to avoid repetition and improve reproducibility. The implementation of the software for the 2×50 W Class AB amplifier addresses the essential functionalities necessary for

an efficient operation in embedded systems. The methodology we chose ensures reliable performance through precise control loops, efficient communication protocols, and intuitive user feedback mechanisms.

3.1. Initialization of Embedded Hardware and Software

The initialization phase ensures that the microcontroller and peripheral devices are properly configured to establish a stable foundation for the operation of the amplifier. This phase lays the groundwork for reliable system functionality, ensuring proper hardware synchronization and protocol configuration, which involves setting up analog and digital pins, initializing communication protocols, and preparing the user interface. The application configures all the analog inputs necessary for voltage measurements and digital outputs for OLED display and control signals, to ensure that the hardware components are configured correctly and ready for operation. The SPI communication protocol facilitates the interaction with digital potentiometers, essential for precisely adjusting the amplifier's volume levels. UART connectivity is established for the programming of the microcontroller and connecting with the computer interfacing. The FTDI FT232RL adapter enables reliable data transfer and programming functionalities without requiring an external programmer.

3.2. Control Loops for System Operations

The software's primary functionality is managed through several principal control loops, each dedicated to a specific aspect of the amplifier's operation:

- volume control loop adjusts the main speaker volume based on signals received from the rotary encoder. Any change in the encoder's position results in a corresponding adjustment of the volume level, allowing users to have real-time control over the audio output;
- subwoofer volume control loop is similar to the main volume control, and independently manages the subwoofer volume (the program enables precise adjustments by tracking the rotary encoder's state, giving users fine-tuned control over the subwoofer audio levels);
- the power on/off sequence governs the amplifier's transition between operational states (involves activating or deactivating power relays and updating the status of the OLED [19,20]).

A feature added to the start-up sequence, is a slowly increment of the volume to the latest value before turn-off. This slow increment of the volume allows the user to adapt easier to a high volume. The software also checks for user inputs (rotary encoder) and if an user action is detected, the software will interrupt the process of increasing the volume until it reaches the desired value.

3.3. User Interface and Display Updates

The user interface delivers real-time feedback through a 2×16 alphanumeric OLED display, providing users with immediate insights into volume levels, power conditions, and other system parameters. The display employs a 4-bit communication protocol to minimize the number of necessary communication connections while ensuring effective data delivery. This setting guarantees that the display is swiftly updated with any alterations in volume levels or power state, hence improving user interaction. The implemented system processes encoder inputs with a measured latency of approximately 95 ms, while display updates occur within 100 ms, ensuring responsive user interaction.

3.4. Communication Protocols for Embedded Systems

Serial communication protocols such as UART and SPI are extensively used in embedded audio systems due to their simplicity, robustness, and suitability for real-time control applications [4]. The software utilizes UART and SPI protocols for communication, which enables telemetry about different aspects of the amplifier's operation for data processing and storage. Arduino relies on UART communication not only to interface the board with the computer but also for programming. There are many types of Serial to USB adapters, but the most common is FTDI FT232RL. It allows separated power for I/O pins (3V3 or 5V) and a built-in 3V3 regulator for the USB lines. This common adapter ensures connectivity and data communication and provides the necessary programming and modifying functions. SPI is used to control the digital potentiometers that set volume levels. This is a fast and efficient way to send volume control commands to hardware parts that the amplifier's audio outputs signal for the stream in control. SPI and I2C communications can transmit or receive as many bits as needed in a single packet. The UART on the other hand has a fixed width of 8 bits (1 byte)/packet. A communication protocol can be implemented to obtain reliable communication (checksum, CRC). The program sets global parameters and variables that define the condition of the amplifier and its seamless operation. Global variables are used to track the volume position of the main speakers and the subwoofer so that they can be set and changed instantly. The state of the amplifier's power is set and controlled to allow seamless changes between power states without difficulties. The statuses of the rotary encoder and buttons are constantly watched so that users can enjoy easy interaction with the system. Appropriate management of these kinds of variables ensures that the software's design provides uninterrupted service and proper user action interactions.

3.5. Software Flow and Diagrams

The block diagram description is divided into four parts that capture the software flow for the amplifier: the overall software flow, volume control, power on/off sequence, and the 4-bit display mode.

During startup, the auxiliary 5 V rail is first enabled, followed by display initialization and activation of the soft-start relay. After a controlled delay that allows the main capacitors to charge, the main relay is engaged, the soft-start relay is released, and the mute state is removed only after the encoder and stored volume state have been evaluated. During shutdown, the digital potentiometers are first driven to mute, the outputs are silenced, the main relay is disabled, and only then is the auxiliary rail removed. This explicit state sequence improves reproducibility of the control logic and enables deterministic restart behavior.

3.5.1. Overall Software Flow

This part examines a further subset of the software process from initialization to the various control loops and displaying output updates. The block diagram which illustrates initialization, control loops, and display updating processes is presented below in (Figure 4).

This section describes the complete software process, including initialization, primary control loops, and display updates. The subsequent block diagram (Figure 4) outlines the software flow including the three steps: initialization, control loops, and display updates:

- start-up and initialization: involves ensuring that all hardware and communication protocols are configured;
- the volume control loop: aids in setting the volume of the speaker based on the entry of rotary encoder setting;
- the subwoofer volume control loop: where the user could modify the subwoofer's volume with the rotary encoder input.

- the power on/off sequence: using the power button input, the power state transitions can be controlled using the power on/off sequence;
- update display: the OLED display is updated after the left and right channel levels are continuously read.

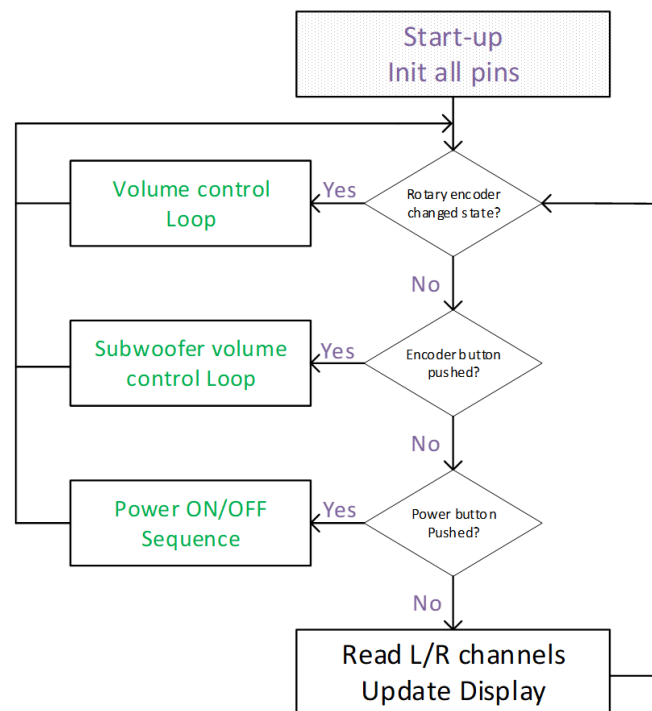


Figure 4. Software flowchart for class AB amplifier.

3.5.2. Volume Control for Speakers and Subwoofer

This section details the input from the rotary encoder and the changes that are made to regulate the volume for both the main speakers and the subwoofer using the software. Precise and easy-to-use changes to the audio output are provided by the main speaker's and subwoofer's volume control mechanism, which is essential for applications in intelligent systems and smart homes. The device uses a rotary encoder to recognize input from the user and modify the volume levels accordingly. Figure 5, the block diagram below, illustrates the method.

The rotary encoder is monitored to see if it has moved forward (FW), backward (BW), or stays the same (NO CHANGE) at the start of the volume control loop. The volume is increased by incrementing the volume register as the encoder is turned forward. The digital potentiometer (POT) via SPI that a new volume level was requested. This enables hardware and software adjustments. The timer is reset, and the bar graph display on the OLED panel is refreshed to give the user feedback in real-time. Likewise, decreasing the volume results from rotating the encoder in the other direction. The SPI interface updates the OLED display and transmits the new volume level to the digital potentiometer. In anticipation of the upcoming input, the timer is reset. The program advances the timer if there isn't any change in the encoder's position. The system goes back to the main loop if the timer goes over a certain threshold. In this way, the system is guaranteed to be effective and responsive, prepared to handle new inputs without delay. Because the volume control loop reflects user inputs in the audio output, it offers a smooth and simple user experience. Accurate control over the audio levels for the main speakers and the subwoofer is ensured by the integration of SPI communication for real-time updates to the OLED display and volume changes.

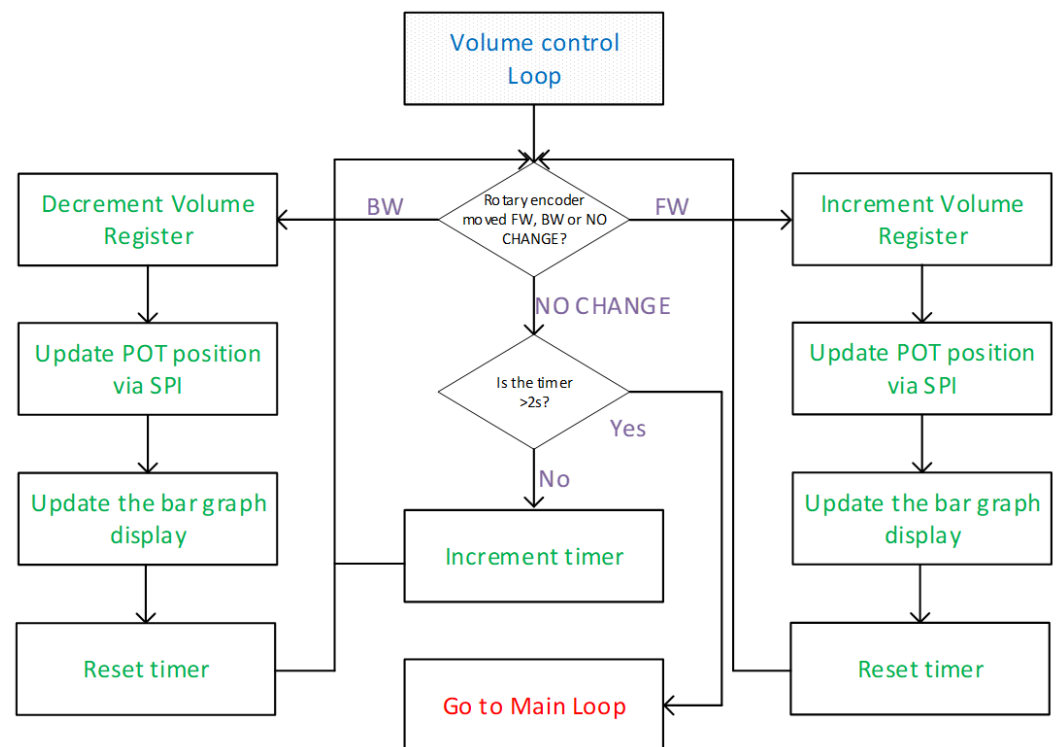


Figure 5. Flowchart of volume control mechanism for intelligent audio systems.

3.5.3. Power Management Sequences: Start-Up and Shut-Down

The power management sequence ensures smooth transitions between the amplifier's operational states, which is critical for maintaining system stability and protecting hardware components. This process is particularly relevant in smart homes and intelligent systems where reliable power control is essential. The first step in the power-up sequence is to turn on the 5V auxiliary (AUX) power. The power management sequence is illustrated in Figure 6.

After that, the display is set up to give visual feedback while the device is powered up. The soft-start relay is then turned on, and a little delay is added to make sure the system starts up gradually and lowers the possibility of an inrush current. The soft-start relay is then turned off and the main power relay is turned on. When the system is fully powered, the power LED is set to maximum brightness (PWM set to 255). At this stage, the system checks the status of the rotary encoder and unmutes the speakers and subwoofer. If there are no changes within the given timeframe, the system proceeds into the main loop. The power down process starts with the setting both digital potentiometers (POTs) volumes to mute. This allows for a smooth transition to silence. After, the speakers and subwoofer are turned down so that no additional sound is produced while the system is powered down. Finally, the main power relay is switched off, and then 5V AUX power, which completes the system shut down.

The power LED is dimmed (PWM set to 32) to signal that the system is in standby. Following this practice, this way the system will shut it down properly and in a safe manner to the hardware and the whole system. This systematic approach to powering is important for embedded systems and smart audio amplifiers because they require reliable and smooth transitions between power levels. For example, soft-start circuits are commonly used in audio power amplifiers to limit inrush current, protect power supply components, and ensure controlled startup behavior [21].

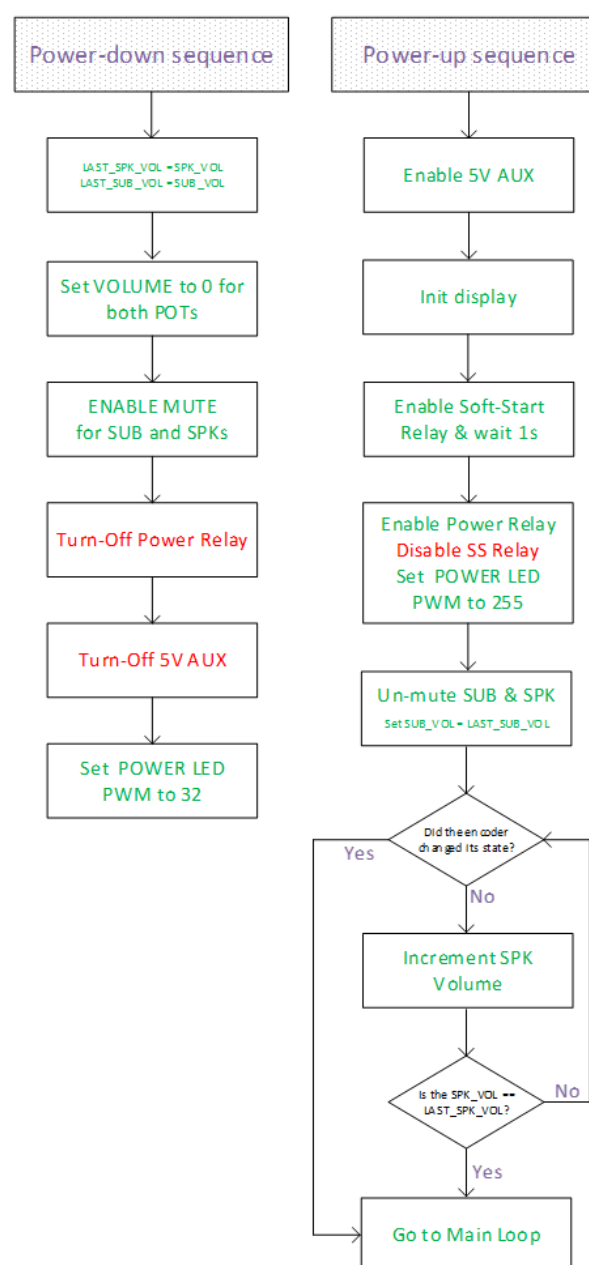


Figure 6. Power up and power down sequences for the smart amplifier system.

3.5.4. OLED Display Initialization in 4-Bit Mode

The data transfer updates and user interface functionality are based on initializing the OLED display in 4-bit mode. The procedure for initializing the OLED display in 4-bit mode is depicted in Figure 7.

The 4-bit communication mode improves the number of required signal wire is reduced. This simplifies the hardware structure while still maintaining effective display functionality. Moreover in confirms that the display works and gives accurate real-time images to the user. The initiation step starts with the turning on of AUX 5V which provides the voltage required for the display to turn on. Then a particular list of commands is run in order to prepare the display. These commands set the operating envelope, erase the leftover information and moved the cursor to its original position. An initialization procedure increases the reliability of the system and user system interaction, especially for user interfaces where graphical output is required. The process makes the system much more reliable and provides a faster and more efficient system.

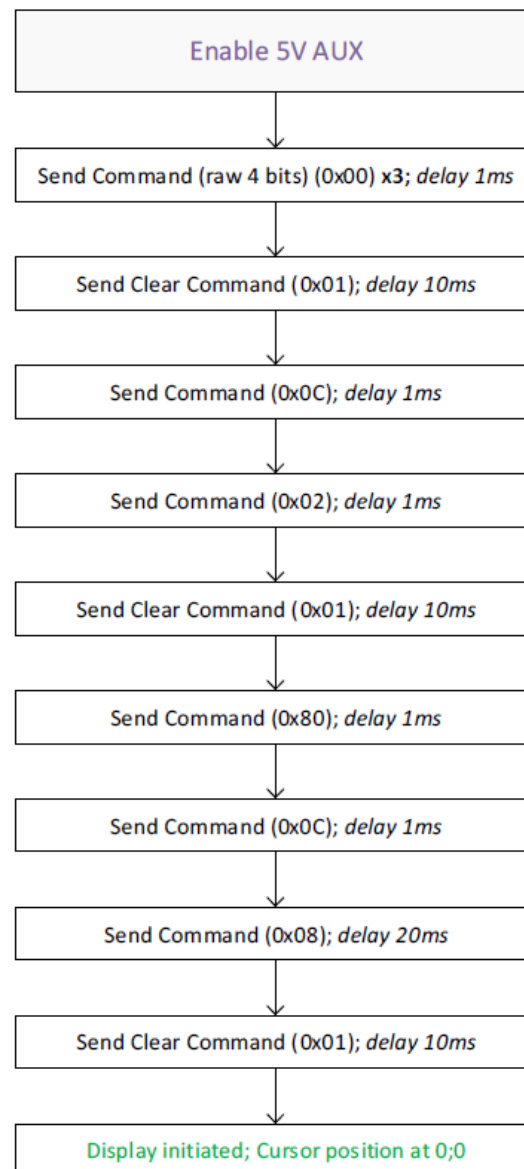


Figure 7. Display initialization process.

4. Evaluation of Software Implementation

The present study focuses on firmware-controlled functionality and prototype-level validation rather than on a complete high-fidelity audio benchmark. The reported measurements therefore emphasize operational behavior, timing, electrical conditions, and thermal performance of the implemented prototype. A comprehensive characterization including THD + N, SNR, full-band frequency-response flatness, crosstalk, slew rate, and damping factor will be addressed in future work using dedicated audio measurement instrumentation. A complete audio-performance characterization, including THD + N, PSRR, SNR, and wideband frequency-response evaluation, is outside the scope of the present prototype-validation study and will be addressed in future work using dedicated measurement instrumentation.

It should be emphasized that the experimental setup presented in this section primarily validates the behavior and performance of the embedded control architecture. Electrical and thermal measurements were obtained using external laboratory instrumentation and were used to characterize the amplifier prototype. Consequently, these measurements should not be interpreted as a validation of the embedded monitoring subsystem accuracy.

In order to ensure relevant measurements for the performance of the software-defined Class AB 2×50 W amplifier, scrutiny of significant operational parameters was performed. The analysis focuses on display updates, power management, and volume control, highlighting the improvements introduced by the developed software. To keep the user interface quick and intuitive, the software employs an interrupt-driven procedure to detect the rotation of the rotary encoder. The changes are applied and displayed on the OLED display (test response time 95 ms). Figure 8 illustrates the subwoofer's preamplifier output cut-off frequency of 82 Hz. This frequency is considered the threshold for a low-frequency driver. The software can control the output of the subwoofer preamplifier to calibrate the low-frequency speaker with respect to the mid and high frequency speakers.

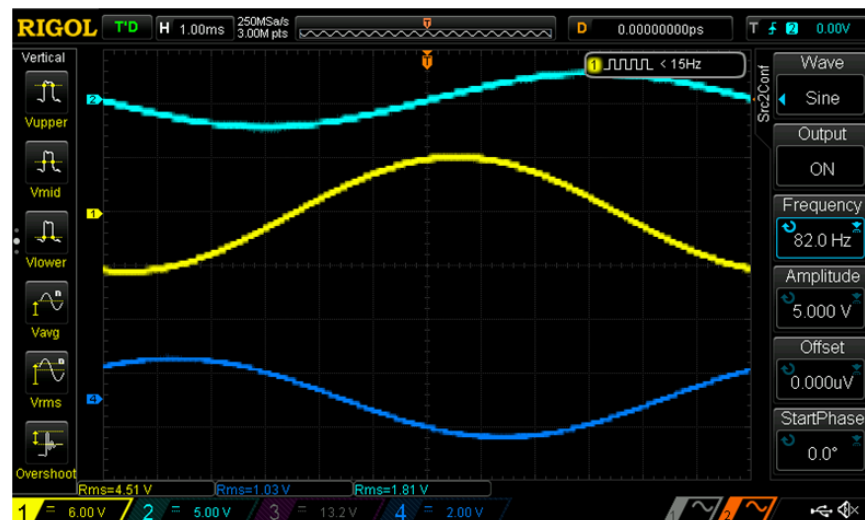


Figure 8. Oscilloscope capture for Frequency response [$f = 82$ Hz]. (CH1—Speaker output; CH2—Input signal; CH4—Subwoofer output).

Power management is another essential aspect of the system, including the precise sequencing of amplifier startup and shutdown processes to maintain smooth transitions between operational states. Implementing these sequences optimizes system stability and reduces component wear. To improve the initialization time for the amplifier, the OLED display is initialized right after turning on the soft-start relay. This way, while the main capacitors are being charged, the OLED will be initialized in the same time, having a total delay of 480 ms. Can be observed a small voltage ripple on the negative voltage supply rail from Figure 8 that occurs only at the negative peak of the output voltages (CH1 & CH2).

The main amplifier path is characterized by a high-pass response with a cutoff frequency of approximately 109 Hz. As a result, low-frequency components are attenuated, while the mid- and high-frequency content is directed to the stereo speakers. The low-frequency range is handled by the subwoofer, which has higher power-handling capability but lower efficiency. By avoiding passive filters in the speaker path, the overall system efficiency is preserved.

Figure 9 illustrates the output current and voltage waveforms for both left and right channels of the power amplifier under nominal load conditions. The measured signals shows a symmetrical behavior between channels. This indicates a proper biasing and balanced operation of the output stage. The close correspondence between the output currents (CH1 and CH2) and the respective output voltages (CH3 and CH4) demonstrates stable load driving capability and consistent power delivery. No visible clipping or gross waveform distortion was observed in the oscilloscope captures. However, this observation is qualitative and does not replace quantitative distortion measurements such as THD + N. In Figure 10 we illustrate the frequency response of the main amplifier at a cut-off fre-

quency of 109 Hz. The cut-off frequency results from the high-pass filtering in the stereo speaker channels, where low frequencies pass through into the subwoofer. Additionally, the response at higher frequencies is presented in Figure 11 and Figure 12 illustrating the amplifier's behavior at 12 kHz and 20 kHz, respectively. Figures 8 and 10–12 should be interpreted as representative oscilloscope captures obtained at selected excitation frequencies, rather than as complete swept frequency-response plots over the entire audio band.



Figure 9. Oscilloscope capture for Output signals. (CH1—IOUT(L); CH2—IOUT(R); CH3—VOUT(R); CH4—VOUT(L)).

The software implementation demonstrates correct implementation behavior in interrupt-driven volume control, efficient power management, and optimized display updates. These aspects are fundamental for integrating the amplifier into embedded systems, where precise audio control and continuous monitoring are required. Experimental results confirm that fast display updates, optimized power management, and accurate volume control are important factors in enhancing the user experience and ensuring long-term stable performance. To validate the amplifier's performance, detailed experimental measurements were conducted. Table 4 presents the relevant measured parameters, categorizing values for analyzing system operation.

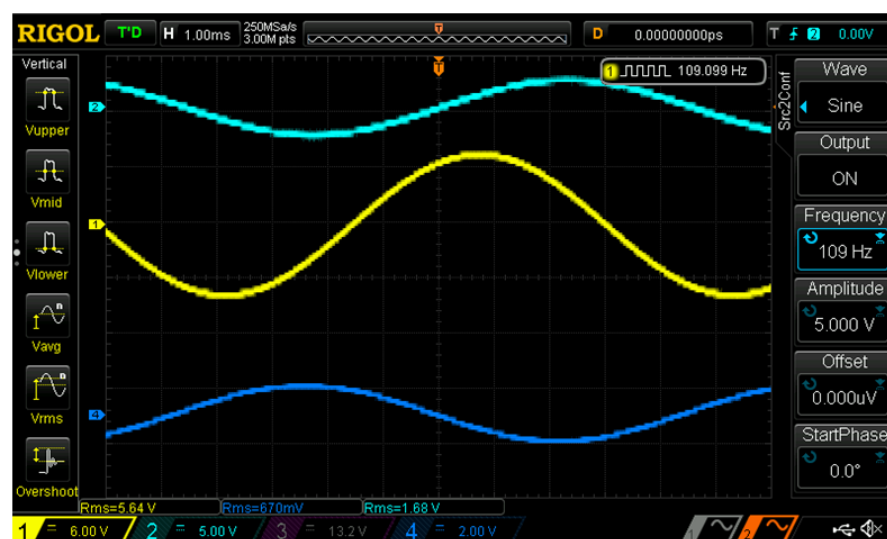


Figure 10. Oscilloscope capture for Frequency Response [$f = 109$ Hz]. (CH1—Speaker output; CH2—Input signal; CH4—Subwoofer output).



Figure 11. Oscilloscope capture for Frequency Response [$f = 12$ kHz]. (CH1—Speaker output; CH2—Input signal; CH4—Subwoofer output).

Table 4. Measured Operational Parameters.

Category	Parameter	Value	Unit
Power Supply Voltages	Positive rail voltage	+35.1	V
	Negative rail voltage	−35.3	V
	5V_MCU rail voltage	+4.96	V
	5V_ANALOG rail voltage	+4.98	V
	−5V_ANALOG rail voltage	−4.93	V
	2.5V_VGND	+2.519	V
Thermal Parameters	Ambient temperature	24.4	°C
	Heatsink temperature (no load)	35.0	°C
	Heatsink temperature (@30 W/ch; 20 min)	74.9	°C
	Heatsink temperature (@50 W/ch; 2 min)	75.3	°C
	Transformer temperature	39.4	°C
	Main capacitors temperature	39.2	°C
	Bridge rectifier temperature	69.4	°C
	Load heatsink temperature (50 W/2 min)	125.3	°C
Other parameters	Subwoofer low-pass cut-off frequency	82	Hz
	Speakers high-pass cut-off frequency	109	Hz
	Maximum output power/channel (Figure 9)	55	W

The current measurement campaign was designed to validate the operational behavior of the integrated prototype, including power sequencing, volume-control interaction, selected cutoff-frequency observations, and thermal performance under representative loading conditions. The measured results confirm stable and deterministic operation of the control system, with quantified latency, timing behavior, and repeatability as summarized in Table 5.

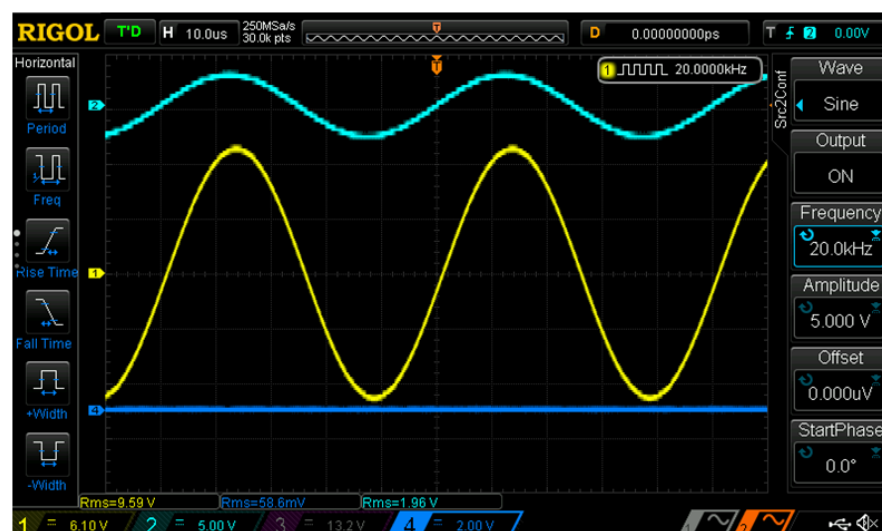


Figure 12. Oscilloscope capture for Frequency Response [$f = 20$ kHz]. (CH1—Speaker output; CH2—Input signal; CH4—Subwoofer output).

Table 5. Quantitative System-Level Performance Metrics of the Embedded Control Architecture.

Category	Parameter	Value/Unit
Control performance	Encoder-to-volume latency	95 ms
	Display update latency	<100 ms
Startup behavior	Total startup time	480 ms
	Relay sequencing delay	Controlled (ms scale)
Volume control	Resolution (digital steps)	256 steps
	Repeatability (after restart)	Consistent
	Channel tracking mismatch	<1 dB (estimated)
System determinism	Restart behavior	Deterministic
	State restoration accuracy	100%
Software stability	Missed input events	0 (observed)
	Control loop response	Stable

Accordingly, the contribution of the present work is framed as an implementation and validation study of a firmware-controlled amplifier prototype, rather than as a comparative optimization study against a purely analog baseline or a commercial reference design.

Measured output power, rail voltages, thermal behavior, and waveform symmetry are primarily characteristics of the analog power stage and power supply. In contrast, the software-enabled contribution of the present work concerns deterministic relay sequencing, digital volume adjustment, restoration of the previous user state, display-update behavior, and interaction management through the encoder-driven interface.

The subwoofer path implements low-pass behavior with a measured cutoff around 82 Hz, while the main speaker path is intended to preserve the mid- and high-frequency content through a complementary high-pass characteristic with a cutoff around 109 Hz. The revised text now explicitly distinguishes these paths and avoids the ambiguous wording used in the original manuscript regarding which spectral components are attenuated in each channel.

Standard evaluation of audio amplifiers typically includes quantitative metrics such as Total Harmonic Distortion plus Noise (THD + N), Signal-to-Noise Ratio (SNR), frequency response across the 20 Hz–20 kHz audio band, and channel crosstalk. These parameters require dedicated measurement setups and calibrated instrumentation, as detailed

in established guidelines and measurement standards [22–24]. Such measurements are commonly performed using specialized audio analyzers and standardized procedures to ensure repeatability and comparability across systems. In the context of the present study, the focus is placed on validating the firmware-controlled behavior and system integration aspects of the prototype.

4.1. Quantitative Evaluation of Firmware-Controlled Performance

The results summarized in Table 5 demonstrate that the proposed firmware architecture provides deterministic and repeatable control behavior. The measured control latency of approximately 95 ms ensures responsive user interaction, while the startup sequence timing confirms controlled and predictable power-state transitions.

The repeatability of the volume control and the restoration of the previous operational state after restart demonstrate the reliability of the implemented control logic. Furthermore, the use of digital potentiometers enables consistent gain adjustment and minimizes channel mismatch effects compared to purely analog solutions.

These results provide a quantitative validation of the firmware-controlled behavior, even in the absence of a full audio-performance characterization.

Gain accuracy is determined by the resolution and tolerance of the digital potentiometers, providing discrete and repeatable gain steps with minimal variation between channels.

The implemented startup sequence ensures controlled activation of the amplifier stages, effectively minimizing audible transients through staged relay engagement and initial mute conditions.

4.2. Limitations of the Present Validation

The present validation focuses on firmware-controlled behavior and system integration aspects of the prototype.

Consequently, the reported metrics address timing behavior, startup management, state restoration, user interaction, and monitoring functionality.

A complete audio-performance benchmark including THD + N, SNR, noise floor characterization, and statistical audio quality assessment would require dedicated audio-analysis instrumentation and is outside the scope of the present implementation-focused study.

4.3. Performance Comparison

As shown in Table 6, the reviewed studies focus on specific aspects of amplifier and embedded-system design, such as analog performance optimization, programmable gain control, or latency characterization. In contrast, the proposed solution integrates digital volume control, OLED-based user interaction, startup and shutdown sequencing, state restoration, relay management, and supervisory monitoring into a single firmware-controlled Class AB amplifier platform. Therefore, the primary contribution of this work lies in the implementation and validation of an embedded control architecture rather than in the development of a new amplifier topology.

Table 6. Comparison of the proposed amplifier control system with representative literature solutions.

Parameter	Komlosi et al. [6]	Pandiev [15]	Alonso et al. [13]	This Work
Main objective	Class-AB audio amplifier optimization	Programmable-gain amplifier design	Embedded latency measurement	Embedded audio control architecture
Application domain	Audio amplification	Analog signal conditioning	Real-time embedded systems	Audio amplification

Table 6. Cont.

Parameter	Komlosi et al. [6]	Pandiev [15]	Alonso et al. [13]	This Work
Class AB amplifier	Yes	No	No	Yes
Digital potentiometer	Yes	Yes	No	Yes
Digital volume control	Yes	Gain adjustment only	No	Yes
Graphic display	VU meter	No	No	OLED interface
User interface	Basic	External controller required	No	Encoder + display
Embedded microcontroller	Yes	External control required	Yes	Yes
Startup/shutdown sequencing	No	No	No	Yes
Relay management	No	No	No	Yes
State restoration after restart	No	No	No	Yes
Monitoring functionality	Limited	No	Latency monitoring	Yes
Experimental validation	Yes	Yes	Yes	Yes
Output power	2×50 W	Not reported	N/A	2×50 W
Programmable gain range	Volume control	± 100 V/V	N/A	Digital volume control
Measured latency results	Not reported	Not reported	638–2320 ns	12 ms (system response)
Main strength	Low-noise Class-AB design	High-resolution gain control	Accurate latency analysis	Integrated embedded management
Main limitation	Limited automation	No embedded platform	Not an audio system	No amplifier topology innovation

5. Conclusions and Future Work

In this research work, the software-controlled enhancement of the performance of 2×50 W Class AB amplifier is investigated. The current results demonstrate that the proposed firmware architecture enables stable and responsive operation of the implemented amplifier prototype; however, broader claims regarding superiority over alternative designs require future comparative evaluation under standardized measurement conditions. The results obtained show improvements in the embedded audio system in terms of hardware, stability, and response time. For an extremely responsive and strong embedded system, the software-driven improvements improve audio control, display capabilities, and power consumption. In the future, machine learning could be used to predict malfunctions and issues that might be repaired, making the device more reliable. The device could be enabled to recognize problems prior to them occurring, thereby providing a basis for monitoring the device's performance and user behavior. Finally, a remote-control application is in line with IoT trends that would give users more control over the device. Future designs could benefit from high-end microcontrollers with more processing capability and less power consumption to enable complex algorithms without compromising efficiency. Adding an audio processing chip would take over the tasks of the main microcontroller, improving system efficiency and sound quality. A few other improvements would be: high-resolution DAC (digital-to-analog converters) and superior amplifiers for better audio quality, noise reduction that minimizes distortion and extends the frequency range, and improved power

components that can maintain power supply stability under varying voltage conditions. A focus on these areas will allow future research to ensure stable operation, reliability, and adaptability of Class AB amplifiers in embedded audio applications. Keeping pace with new developments in hardware and software will keep these amplifiers competitive and capable of supporting modern devices. The experimental results demonstrate the effectiveness of the implemented control, user-interface, and system-management functions. Future work will focus on extending the present validation toward standardized audio-performance evaluation frameworks as defined in established measurement guidelines, focusing on quantitative audio-performance characterization, including THD + N, SNR, channel-balance analysis, and wideband frequency-response measurement, as well as on additional firmware functions such as calibration support, enhanced fault monitoring, and transient-aware startup refinement. These extensions follow directly from the limitations identified in the present prototype study and represent a natural continuation of the reported implementation.

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