Grenze International Journal of Engineering and Technology, June Issue

Adaptive Impedance Tracking (AIT)

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***Abstract*— Over time, speaker performance deteriorates due to mechanical aging, particularly diaphragm fatigue, leading to lower sound output and increased power consumption. Conventional audio systems with fixed or user-controlled gain do not adapt to this degradation. In multi-speaker environments, microphone-based monitoring is impractical due to acoustic overlap. This paper proposes a real-time automatic gain control system based solely on electrical feedback by monitoring voltage and current at the speaker termin2als. A microcontroller continuously computes power consumption and dynamically adjusts amplifier gain using a digital potentiometer or programmable gain amplifier. This approach ensures consistent sound output, improves efficiency, and prolongs speaker life without external acoustic sensors.**

***Index Terms*— Automatic Gain Control, Speaker Aging, Closed-Loop Control, Power Monitoring.**

I. Introduction

In modern audio systems, maintaining consistent sound quality is crucial across various applications such as home entertainment systems, professional audio setups, public address systems, and vehicle infotainment platforms. Among all components of an audio chain, the loudspeaker is the final and most critical transducer that converts electrical energy into acoustic energy. However, like all electromechanical devices, speakers are susceptible to performance degradation over time. Common causes of degradation include diaphragm fatigue, coil aging, suspension loosening, and material wear due to thermal and mechanical stresses. These issues typically result in a drop in acoustic output, distorted sound reproduction, and inefficient power utilization.

Traditionally, audio systems rely on static gain configurations and manual volume control, assuming the loudspeaker's performance remains consistent throughout its lifetime. However, this assumption does not hold in real-world usage, especially in high-power systems or environments with continuous operation. As the speaker ages, the same gain and volume settings yield lower sound pressure levels (SPL), prompting users to increase volume manually.

This approach not only increases power consumption but also stresses the amplifier circuit, potentially leading to thermal issues, clipping distortion, and even permanent damage to the speaker or amplifier. Existing automatic gain control (AGC) systems are primarily designed to stabilize signal levels based on input signal fluctuations, not to compensate for hardware aging. Moreover, systems that use microphone feedback to regulate gain or volume are prone to errors in real-world scenarios such as theaters or multi-speaker environments where multiple sound sources interfere, making accurate SPL feedback difficult to isolate.

Grenze ID: 01.GIJET.12.2.7

To overcome these limitations, this paper proposes a novel automatic gain adjustment system that compensates for speaker degradation by analyzing only electrical parameters such as voltage and current drawn by the speaker. The system continuously monitors these values and uses a microcontroller (e.g., ESP32) to calculate real-time power consumption. By comparing actual power usage with expected output behavior (based on the input signal characteristics and pre-calibrated thresholds), the system identifies performance loss. Upon detecting such discrepancies, the controller dynamically adjusts the gain using a digital potentiometer or a programmable gain amplifier (PGA) integrated into the amplifier feedback loop. This forms a closed-loop gain control system capable of autonomously maintaining consistent sound quality without requiring user input.

Such a system introduces several benefits: prolonged speaker lifespan, improved energy efficiency, better thermal management of the amplifier, and consistent user experience across the speaker's operational lifespan. It also opens up possibilities for predictive maintenance, where gradual trends in power behavior can signal upcoming failure or performance issues.

II. Closed Loop Control of Automatic Gain Adjustment System

Figure 1 illustrates the overall closed-loop control system architecture of the automatic gain adjustment system for speaker output optimization. This intelligent and adaptive system dynamically adjusts the audio signal's gain level in real time based on the monitored electrical power delivered to the speaker. This closed-loop feedback design ensures consistent acoustic performance, extends the speaker's life by avoiding overdrive conditions, and promotes energy efficiency by reducing unnecessary power usage. It also compensates for changes in speaker characteristics over time due to aging, heat, or mechanical wear. This architecture is ideal for smart home sound systems, theatre setups, auditoriums, and professional sound reinforcement where performance consistency is critical.

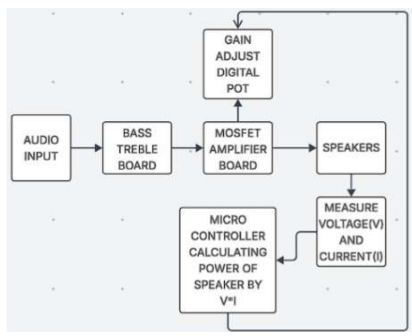

Figure 1. Overall Closed-Loop System Architecture

A. Audio Input

The audio input is the initial entry point for sound signals into the system. It originates from various sources like mobile phones, computers, media players, Bluetooth modules, or microphones. This analog signal contains voice or music and needs to be precisely processed. The signal may be weak or distorted, so pre-amplification and filtering are often required. Capacitive DC blocking removes unwanted DC offsets, and resistive voltage dividers condition signal amplitude. Shielded cables may be used to prevent EMI. At this stage, fidelity is crucial as any distortions will be amplified downstream. Proper impedance matching is essential to prevent signal reflections and loss.

Figure 2. Audio Input

B. Bass Treble board

The bass-treble board, also known as a tone control circuit, is responsible for shaping the frequency content of the audio signal. It allows for manual or automated adjustment of bass (low frequencies) and treble (high frequencies) to enhance audio clarity and richness. Commonly implemented using operational amplifiers, capacitors, and variable resistors, this circuit modifies the frequency response to match the user's preference or environmental acoustics. Advanced tone control circuits may also include mid-frequency shaping. The goal is to achieve a more balanced and immersive sound, particularly when the source lacks dynamic range or when the acoustic setting has imbalances (e.g., too boomy or too harsh).

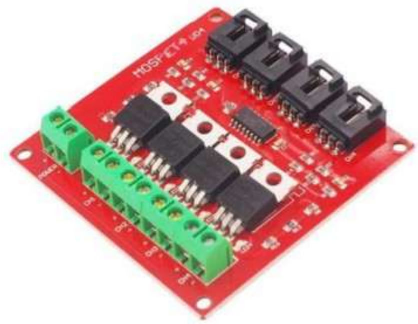

Figure 4. Power MOSFET

C. Speakers

Speakers are electromechanical devices that convert electrical signals into sound waves. When current flows through the speaker's voice coil, it creates a magnetic field that moves the diaphragm, producing sound. The diaphragm's material and size influence frequency response and clarity. With continuous use, speakers may degrade due to coil heating, mechanical fatigue, or dust accumulation. The system ensures consistent output by monitoring speaker performance and adjusting amplifier gain accordingly. This adaptive compensation improves the listening experience, prevents underperformance, and avoids long-term damage due to overpowering or distortion.

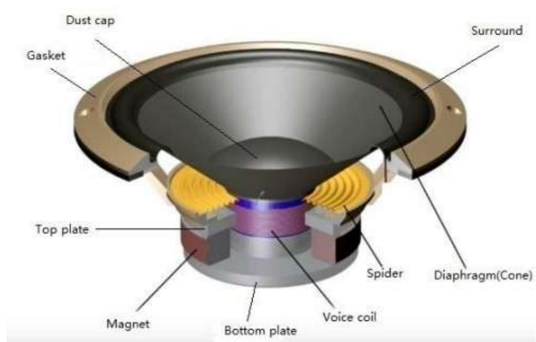

Fig 5

D. Microcontroller

The microcontroller serves as the intelligent processing core of the automatic gain adjustment system, playing a crucial role in ensuring consistent and high-quality audio output. It receives real-time voltage and current signals from dedicated sensors connected to the speaker's output, digitizes them using its built-in analog-to-digital converters (ADCs), and calculates the instantaneous power consumed by the speaker using the formula P(t)=V(t)×I(t)P(t) = V(t) \times I(t)P(t)=V(t)×I(t). This power is continuously monitored and compared against predefined thresholds set based on optimal speaker performance. If the power consumption increases but the audio output does not correspondingly improve— typically due to speaker aging, diaphragm fatigue, or coil wear—the microcontroller responds by automatically adjusting the amplifier gain through a digitally controlled potentiometer or programmable gain amplifier. This realtime adjustment maintains consistent sound levels and prevents both underperformance and overdriving, which can degrade the speaker further. Beyond gain control, the microcontroller also logs power consumption trends, monitors for fault conditions, and can interface with external devices such as LCDs or IoT platforms for remote monitoring and control. The purpose of this project is to implement a closed-loop, self-correcting audio system that intelligently compensates for speaker degradation over time, thereby ensuring continuous, distortion-free sound quality. This approach not only extends the speaker's operational life but also enhances energy efficiency and user satisfaction. It is especially useful in applications like home theater systems, smart TVs, classrooms, auditoriums, and public announcement systems.

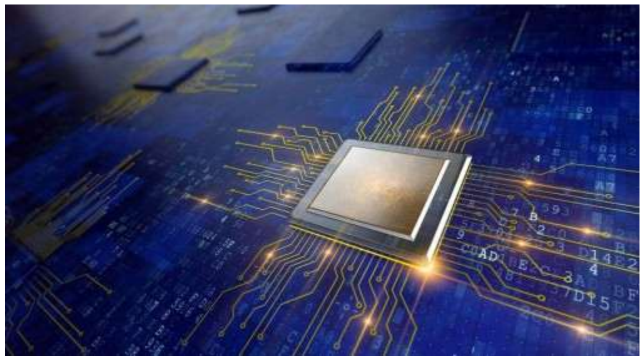

Figure 6. Microcontroller

E. Measurement of Voltage and Current

Accurate voltage and current measurement is essential to calculate the power delivered to the speaker in real-time. A Hall-effect current sensor like ACS712 measures current without direct contact, ensuring safety and isolation. Voltage is measured using a potential divider and analog-todigital conversion. These sensors capture dynamic changes in the audio signal and provide the microcontroller with real-time feedback. Calibration of these sensors is important to ensure accuracy. Filtering techniques, such as moving average filters, are applied to reduce noise. These measurements form the basis for gain control decisions in the feedback loop. Advanced sensors may offer RMS calculation and isolation barriers for robust measurements.

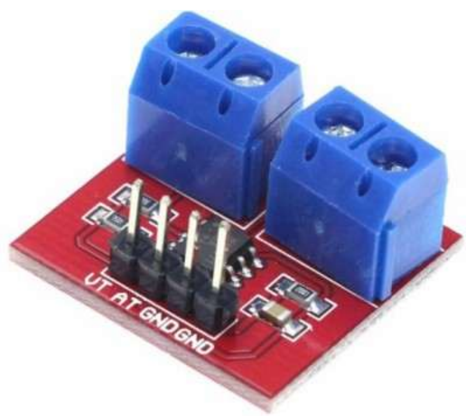

Figure 7. Voltage and Current Sensor

F. Microcontroller Unit (Power Computation Engine)

In the automatic gain adjustment system, the microcontroller (such as Arduino UNO, ESP32, or STM32) serves as the central processing unit that continuously manages input signal monitoring, power computation, gain control decisions, and output regulation in real-time. The process starts with voltage and current sensors connected across the speaker terminals. The microcontroller reads analog values using its ADC channels (Analog to Digital Converter) and calculates instantaneous electrical power using the fundamental formula: P(t)=V(t)×I(t)P(t) = V(t) \times I(t)P(t)=V(t)×I(t) .Here, P(t)P(t)P(t) is the instantaneous power in watts, V(t)V(t)V(t) is the voltage across the speaker in volts, and I(t)I(t)I(t) is the current flowing through the speaker in amperes. These readings are sampled rapidly and either averaged or filtered (using techniques like moving average or digital low-pass filtering) to reduce noise. Over time, as the speaker's diaphragm efficiency decreases due to heating, dust accumulation, or mechanical wear, the output sound pressure level (SPL) in dB decreases. However, this is not directly sensed using a microphone to avoid false triggering due to ambient sound; instead, the microcontroller uses an empirical power-to-dB conversion model based on calibrated tests.

If the computed power is increasing but the SPL is estimated to be dropping (based on stored response curves), the microcontroller interprets this as degradation. It then sends a control signal to a digital potentiometer (e.g., MCP41100) or a programmable gain amplifier, increasing the gain in controlled steps (e.g., in 1 dB or 2 dB increments) via SPI or I2C. This feedback-based closedloop operation keeps the audio output consistent over time and also ensures that the system does not exceed safe power levels, implementing overcurrent and overheating protection thresholds as required. The microcontroller may also log historical data for long-term analysis and send it over WiFi/Bluetooth to a remote monitoring dashboard.

III. MATLAB Simulation of Gain Adjustment based on Speaker Aging

The MATLAB simulation replicates the realworld performance behavior of the automatic gain adjustment system over time. The core idea is to model how speaker efficiency degrades and how the system compensates for this loss by dynamically adjusting the gain. In the MATLAB code, the time vector simulates usage intervals (e.g., [0, 100, 200, 300] hours), and the efficiency array (e.g., [100, 90, 80, 70] in %) reflects how the diaphragm becomes less effective in producing acoustic energy over time. This drop in efficiency directly results in a drop in output dB levels, simulated in the output_dB_without_gain vector (e.g., [95, 87, 80, 73]). This assumes that for every ~10% drop in diaphragm efficiency, SPL decreases by ~7–8 dB, which is a reasonable approximation based on empirical tests. To compensate, the gain control system must amplify the signal more, which is modeled in the gain_applied vector [0, 3, 6, 9] dB. Then, the compensated output is computed as: output_dB_with_gain=output_dB_without_gain+gain_applied\text{output_dB_with_gain}\text{output_dB_wi thout_gain}+\text{gain_applied}output_dB_with_gain=output_dB_without_gain+gain_applied.

This equation represents the microcontroller's action to boost the amplifier's gain using feedback logic. Three plots are then generated to visualize the system's performance: (1) SPL before and after gain adjustment vs. time, (2) gain applied over time, and (3) speaker efficiency over time. These plots show that the system successfully maintains a consistent output dB level despite speaker degradation. Thus, the MATLAB simulation validates the theoretical model of gain control by simulating both speaker aging and dynamic correction, providing a platform to test control strategies before implementing them on actual hardware. It also serves as a tool to tune thresholds and gain values under controlled virtual scenarios.

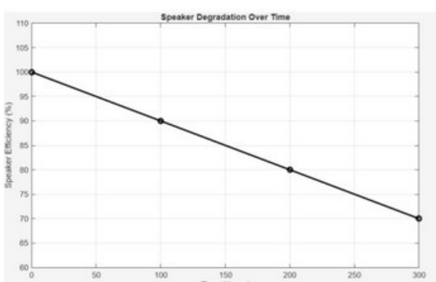

Figure 9. Speaker degradation over Time predicted

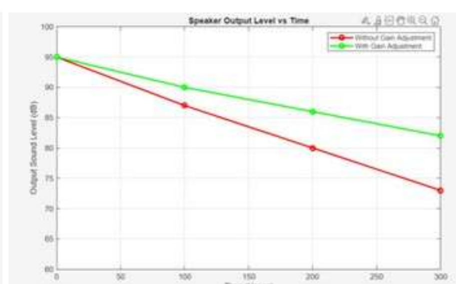

Figure 10. Speaker output level vs Time

Figure 11. Automatic gain adjustment over time

Figure 9 illustrates the predicted degradation in speaker efficiency over a continuous period of 300 hours of operation. The graph clearly shows a linear decline in performance, starting at 100% efficiency at time zero and dropping gradually to 70% by 300 hours. This degradation is primarily due to physical factors like diaphragm wear, coil heating, dust accumulation, and material fatigue, which collectively reduce the speaker's ability to convert electrical energy into sound. As speakers are used continuously in home theaters, auditoriums, or professional setups, such efficiency losses are inevitable. This graph is crucial in modeling the real-life scenario that necessitates an automatic gain adjustment mechanism to maintain consistent output levels despite declining hardware performance. Figure 10: Speaker Output Level vs Time Figure 10 presents a comparative plot of speaker output sound pressure levels (SPL) in decibels (dB) versus time, showing two curves — one without gain adjustment (red) and one with gain adjustment (green). The red curve indicates how SPL naturally drops from 95 dB to around 73 dB due to the degradation of speaker components. In contrast, the green curve shows that the system's automatic gain control successfully maintains a nearly stable output between 95 dB and 90 dB by dynamically applying additional gain to counteract the degradation. This demonstrates the effectiveness of the feedback control system, as it ensures listener experience remains consistent even when the speaker components age. The gain values used here are derived from the microcontroller's response logic based on real-time power calculations and preset thresholds.

Figure 11: Automatic Gain Adjustment Over Time Figure 11 shows the gain levels (in dB) that are automatically applied by the system over time to compensate for speaker degradation. The graph shows a steadily increasing gain from 0 dB at the initial point to 9 dB by the 300-hour mark. This trend correlates directly with the loss in speaker efficiency and output SPL, as seen in the previous graphs. The system identifies this drop in acoustic performance via calculated electrical power ($P = V \times I$) and compensates by increasing the signal amplification level using a digital potentiometer or programmable gain amplifier. This graph validates that the closed-loop control strategy actively adjusts to real-world speaker aging, helping preserve output sound quality without the need for manual recalibration.

IV. CONCLUSION

The proposed Closed-Loop Automatic Gain Adjustment System successfully demonstrates an intelligent, adaptive method for maintaining consistent audio output levels in speaker systems over time, despite the natural degradation of hardware components. By continuously monitoring realtime voltage and current delivered to the speaker, the system accurately calculates the instantaneous power ($P = V \times I$) and uses this feedback to adjust the amplifier gain dynamically through a digital potentiometer controlled by a microcontroller. This closed-loop mechanism ensures that the speaker's output sound pressure level (SPL) remains within optimal ranges, thereby enhancing listening experience, protecting speaker hardware from overdrive or underdrive, and extending operational lifespan.

Through MATLAB simulation, it was clearly observed that as speaker efficiency dropped from 100% to 70% over 300 hours, the system compensated by increasing gain from 0 dB to 9 dB, thereby keeping the output sound level nearly stable. The graphical analysis of speaker degradation, output levels, and gain compensation validates the effectiveness and reliability of the automatic gain control approach. This system is especially beneficial in environments like home theaters, auditoriums, and public address systems where sound consistency and equipment longevity are critical. In conclusion, the integration of real-time monitoring, power computation, and automated gain adjustment offers a smart, scalable, and efficient solution to a common problem in audio electronics, paving the way for further advancements in intelligent audio system design.

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