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User-Defined Noise-Cancelling Headphones for Personalized Acoustic Experience

Yutong Chen[1]

[1] Tianjin Farragut School, Tianjin 300042, China

Correspondence: Yutong Chen, Tianjin Farragut School, Tianjin 300042, China.

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Abstract

The user-defined noise-cancelling headphone allows users to personalize the noise-cancelling ability of the headphone. This can help them be aware of both the sound and the surroundings. Prototype is still in progress, so is testing.

Keywords: noise cancellation, headphone

1. Introduction

The user-defined noise-cancelling headphone offers a unique advantage over conventional noise-cancelling headphones by allowing users to blend a specific amount of ambient noise with the audio in a personalized way. Utilizing adaptive noise cancellation technology, this device prevents users from feeling overwhelmed by the headphone's sound output or constantly disrupted by external noises. It strikes a balance between isolating the user from unwanted noise and keeping them aware of their surroundings, all while delivering clear and recognizable audio.

This device is particularly appealing to individuals who need to listen to music or watch videos while commuting on noisy public transportation. With these headphones, users can focus on the audio content without constantly adjusting the volume to accommodate fluctuating noise levels in the environment. They can enjoy even the softest details in the audio without distraction, while still remaining aware of important external sounds — such as announcements, approaching stations, or potential hazards. Once the desired noise level is set before playback, the headphone continuously monitors and adjusts the ambient noise mixed with the audio to maintain this level.

In this context, the user-defined noise-cancelling headphone provides a significant improvement over both traditional active noise-cancelling headphones and standard headphones without noise-cancelling capabilities. By offering the ability to personalize both the audio signal and ambient noise levels, and seamlessly blend them, this device ensures a more tailored and versatile listening experience.

2. Use-Case Requirements

The primary function of this device is to enable users to access and enjoy high-quality audio from various sources, including standard audio files and the soundtrack portions of video files, in real-time. This capability is crucial for scenarios where immediate feedback or on-demand audio access is needed, such as during video streaming, gaming, or music playback on public transportation or in noisy environments.

One key feature of the device is the ability to allow users to adjust the level of noise cancellation dynamically. Users can select a noise cancellation setting ranging from no noise cancellation, allowing full ambient sound to pass through, to complete noise cancellation, where the device blocks out almost all external noise. This range of control provides users with the flexibility to adapt to different environments, whether they need full immersion in a quiet space or a more balanced awareness of their surroundings, such as when walking in busy streets or commuting on public transport.

In addition to controlling noise levels, the device must ensure that the final audio output is delivered at a safe volume for human ears. To prevent hearing damage, the device will include a built-in volume limiter that can be set by the user within a recommended range based on healthy auditory practices. This ensures that even when users increase the volume to hear soft details in the audio or compensate for noisy environments, the sound will not exceed levels that could be harmful over extended periods of use.

3. Architecture and/or Principle of Operation

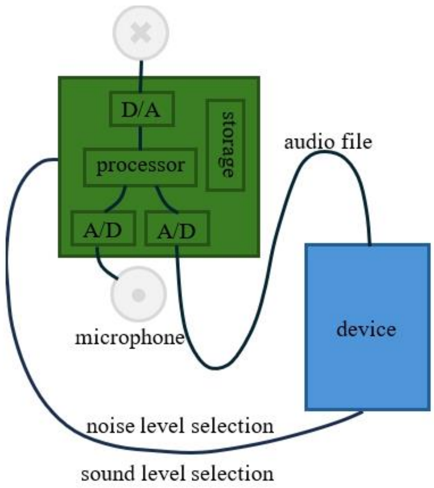

Figure 1. System drawing

4. Design Requirements

To achieve real-time playback, the system must maintain a delay of less than 30 milliseconds, as any noticeable lag can disrupt the listening experience, especially in time-sensitive applications like gaming or video conferencing (Lester, Michael & Jon Boley, 2007). The microphone will capture environmental sounds in the audible frequency range of 20 Hz to 20,000 Hz (Purves D, Augustine GJ, Fitzpatrick D, et al., 2001), which is sufficient for most real-world ambient noises.

Since audio files typically span the same frequency range, the system's analog-to-digital (A/D) and digital-to-analog (D/A) converters will operate at the consumer-standard sampling rate of 44,100 Hz (AES5, 2008), which ensures accurate sound reproduction according to the Nyquist theorem. This rate effectively captures all audible frequencies without introducing distortion.

A processor speed of at least 1 GHz is required to ensure smooth performance when running the noise-cancellation algorithms. The processor will subtract ambient noise from the audio signal based on the user-selected noise level, dynamically adjusting the signal-to-noise ratio (SNR) to match user preferences. These operations, including real-time filtering and signal adjustment, must be completed within milliseconds to maintain audio synchronization and prevent latency issues.

In summary, the system will meet the requirements for real-time playback with a delay of under 30 milliseconds, 44,100 Hz sampling rates for audio conversion, and a processor capable of handling adaptive noise cancellation at a speed of 1 GHz.

5. System Implementation

5.1 Adaptive Noise Cancellation

Adaptive noise cancellation is the core algorithm behind this product. Its fundamental principle is to subtract an estimated noise signal from the combined mixture of desired audio and unwanted noise, resulting in an enhanced version of the original audio signal. This process relies on continuously monitoring the ambient noise and adjusting the cancellation in real-time. The algorithm's effectiveness depends on accurately estimating the noise and applying the appropriate subtraction to minimize interference without distorting the desired signal.

To allow for customization, the algorithm multiplies the estimated noise by specific factors set by the user. These factors control how much noise is reduced, enabling users to adjust the level of external sound they want mixed with their audio. This dynamic adjustment gives users more flexibility in adapting to different environments while maintaining optimal sound quality.

5.2 Processor

The processor should be equipped with at least four single-channel input ports to handle two A/D converters, the microphone, and the audio input from the device. Additionally, it requires a single output channel for delivering the processed audio to the headphones. Given these requirements, the processor must also be compact and lightweight, as it will need to fit within the limited space of the headphone design without adding excessive weight that could impact comfort.

Taking these factors into consideration, the Arduino Nano ESP32 presents a suitable solution. It features 14 digital I/O pins, providing ample connectivity for the necessary input and output channels, and comes with 16 MB of external flash memory for handling data storage and processing tasks efficiently. Its compact size and low power consumption make it ideal for integration into a headphone without compromising performance or portability.

6. Test, Verification and Validation

This aspect of the research will be addressed in future work through the following proposed testing plan.

6.1 Tests for Noise-Cancelling Personalization

To evaluate the performance of the noise-cancelling feature, we will measure whether the output signal achieves the signal-to-noise ratio (SNR) that matches the user's selected level. This can be done by analyzing the processed audio output and comparing the SNR to the desired target. In addition, subjective user feedback will be gathered to complement the technical measurements.

A questionnaire will be distributed to users after each test session to assess their perception of the noise reduction. Participants will be asked to report how much ambient noise they could still hear while using the device, and whether it aligns with their pre-selected noise-cancelling settings. This combination of quantitative data (SNR measurements) and qualitative data (user feedback) will provide a comprehensive evaluation of how well the device personalizes noise cancellation to individual preferences. Further adjustments can be made based on these results to fine-tune the algorithm for improved performance.

In future testing phases, larger sample sizes and various environmental conditions will be introduced to ensure the robustness of the noise-cancelling feature across different real-world scenarios.

Glossary of Acronyms

A/D — analog-to-digital converter

D/A — digital-to-analog converter

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