

WaveSink: Real-Time Multi-Device System Audio Routing and Equalization using WASAPI

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Abstract- Most operating systems provide limited support to their users by allowing only one default output device at a time for audio playback purposes. This work suggests WaveSink, which is an application for the Windows desktop environment that allows users to output their system audio on several output devices simultaneously. This system has been implemented in modern C++20 language for backend processing, Windows Core Audio API (WASAPI) for audio capturing, and Qt 6.10 toolkit for its graphical user interface. WaveSink captures the system audio stream before rendering to the default output and routes it to the desired number of selected output devices in synchronization with each other. The application also provides DSP equalization functionality individually for every output device.

Index Terms- Audio Routing, C++20, Digital Signal Processing, Multi-Device Playback, Qt Framework, WASAPI.

I. INTRODUCTION

The need for shared audio experience is becoming more prevalent nowadays as collaborative consumption of digital multimedia becomes more common. Nevertheless, a conventional desktop operating system is normally set to route audio to a single default sound card at a time. The conventional approach is commonly associated with numerous drawbacks when multiple users simultaneously wish to hear the same audio stream from an audio source, such as movie or music playing in the computer, using their own headsets or combinations of speakers and headphones connected to the computer.

Although virtual audio cables and software mixers are available, they are typically associated with difficult user interface, large overhead and considerable latencies disrupting video-audio synchronization. WaveSink was designed specifically to help resolve this issue. This software tool functions by silently capturing the computer's audio output and duplicating the audio data to any number of chosen output devices without additional user interaction. In comparison with existing solutions that include costly hardware splitter (resulting in poor quality of the output and supporting only analog connections), complex virtual audio cables with high latencies, this paper proposes the development of a dedicated software solution.

II. IDENTIFICATION, RESEARCH, AND IDEA COLLECTION

The following process was used to identify the current challenges and develop a highly efficient software-based solution, without using any hardware-based components:

[1] Identifying the problem in the existing system: Manual OS limitations were considered to understand problems such as the mandatory single-sink constraint, making it difficult for users to share multimedia.

- [2] Studying the current solutions: Existing routing solutions, including third-party virtual mixers, were studied. It was found that many contemporary methods increased latency and system complexity, leading to audio and video synchronization issues.
- [3] Requirement gathering: From the above-mentioned findings, the following requirements were derived to enhance the efficiency of the system: centralized management of the user interface, tracking of the loopback process in real time, duplication of packets at low latency, and equalization tailored to each client.
- [4] Investigating software-based architectures: The use of multithreading to achieve concurrency was examined to develop a scalable and efficient solution, whereby user interface operations would not block the real-time audio pipeline.
- [5] Evaluating technologies: Technologies such as C++20 for implementing core functionality, Qt 6.10 for designing the front-end user interface, and WASAPI for intercepting raw audio packets were evaluated to construct a highly efficient and affordable solution.

III. STUDIES AND FINDINGS

A. System Methodology

Formalizing Requirements

- [1] When looking deeply into native OS audio managers, it became clear that they were highly restrictive. The following functional requirements have been prioritized the most:
- [2] Centralized State Management: A responsive graphical interface (MainWindow) for monitoring and managing available hardware sinks.
- [3] Asynchronous Processing: Background worker threads (RouterThread) used for tasks such as loopback capture and DSP processing without blocking the main UI thread.
- [4] Resource Optimization: Direct memory buffer writing to IAudioRenderClient endpoints to minimize latency.

The Reason for the Technology Stack

Choosing technologies for development is determined by a demand for an environment that will be responsive and not tightly coupled:

- [1] Programming Language (C++20): Offers deterministic memory management and high-performance execution critical for real-time audio.
- [2] Frontend Framework (Qt 6.10): This technology stack is required for implementing reactive components like SinkListWidget and SinkItemWidget.
- [3] Audio Backend (WASAPI & MMDevice API): Allows deep integration with Windows Core Audio for AUDCLNT_STREAMFLAGS_LOOPBACK interception.

B. System Modules

- [1] Audio Capture Module: Relies on WASAPI's Loopback Capture feature. It creates an IAudioCaptureClient that taps into the default audio render endpoint of the system, intercepting raw audio continuously.
- [2] Audio Routing Module: Handled in a dedicated background worker (RouterThread). When new audio frames are available via GetBuffer, the worker reads the raw PCM data and synchronously writes it to the buffer of every active render client.

- [3] Equalizer Processing Module: While writing audio frames, the system applies basic digital signal processing. It reads raw 32-bit float frames, retrieves EQ settings, applies gain via Biquad filters, and forwards the processed data.
- [4] Hardware Scanner Module: Periodically scans for connected hardware devices via IMMDeviceEnumerator to keep the UI up-to-date with plugged/unplugged devices.

C. High-Level System Architecture

This diagram illustrates the macro-level audio data flow. It demonstrates how WaveSink intercepts the default Windows audio render endpoint via WASAPI Loopback Capture, processes the data, and simultaneously duplicates the audio stream across multiple user-selected hardware sinks.

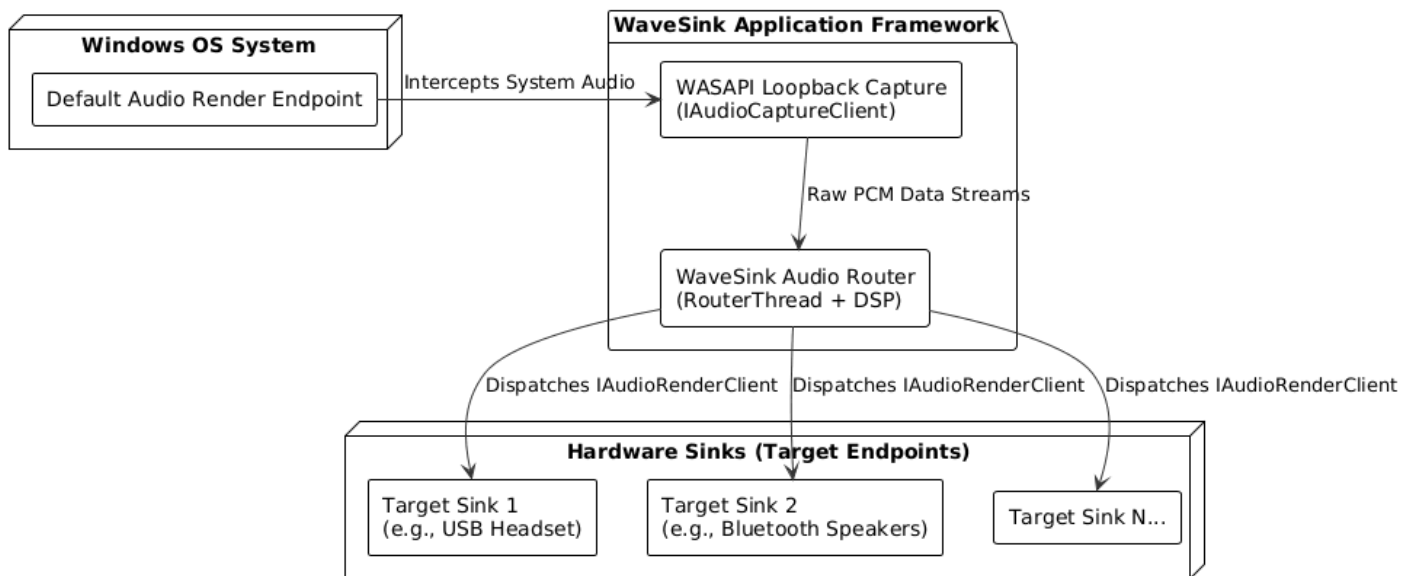


Figure 1: High Level Architecture

D. Internal Component Architecture

This structural block diagram details the software's internal composition. It highlights the strict separation of concerns between the Qt-based Presentation Layer (GUI), the intermediate Audio Controller orchestration layer, and the highly optimized, real-time WASAPI processing engine operating on dedicated background threads.

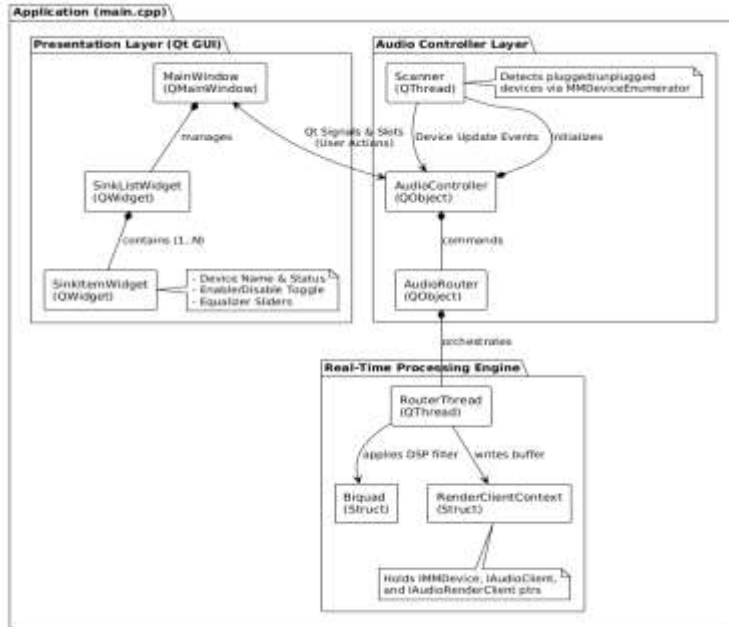


Figure 2: Low Level Design

E. Application Flow

This sequence diagram maps the chronological execution flow of the application. It outlines the background device discovery process, the user-triggered routing initialization, and the continuous real-time execution loop where audio is captured, processed with DSP equalization, and dispatched to multiple endpoints.

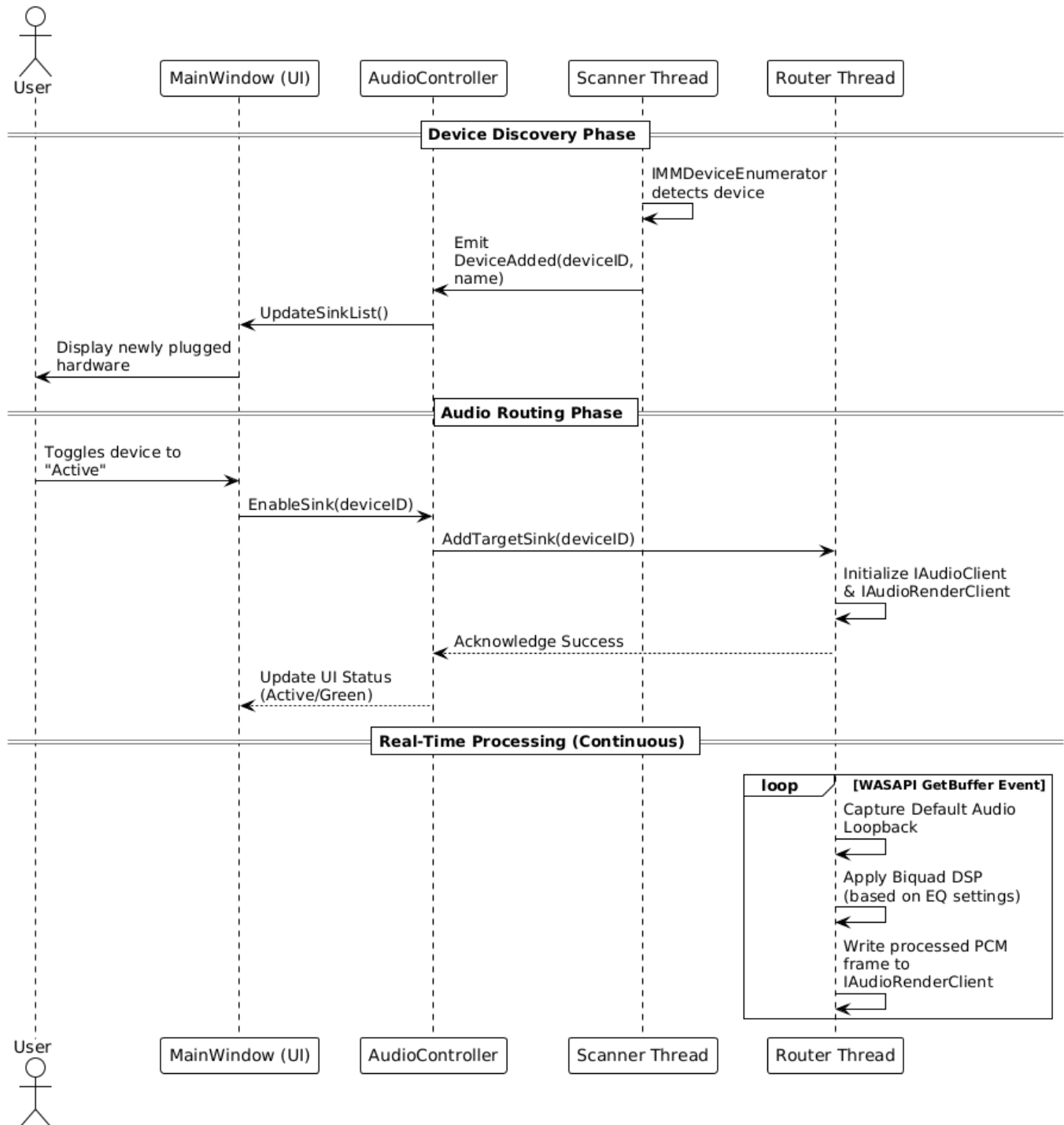


Figure 3: Application Sequence Diagram

F. User Interface

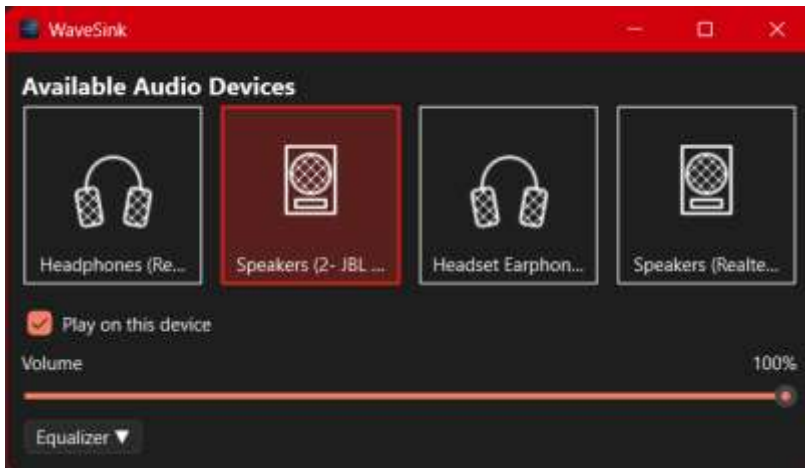


Figure 4: User Interface on Windows

The working flow of the system is designed to ensure simplicity and efficiency:

- [1] Application launches and initiates the Scanner module.
- [2] System displays real-time hardware availability in the Qt GUI.
- [3] User selects multiple desired output devices (sinks).
- [4] AudioRouter initializes an IAudioRenderClient for each selected sink.
- [5] RouterThread continuously captures the system loopback buffer.
- [6] DSP is applied to the 32-bit float audio frames based on individual EQ settings.
- [7] Processed audio is dispatched synchronously to all active sinks.

IV. PEER REVIEW

System evaluation was carried out by professors and peers in order to determine the design features and the working capability of the system. The emphasis during system evaluation was put on detecting weaknesses in the usability, efficiency, and scalability of the system.

Suggestions were made regarding the system response time, particularly focusing on audio-video synchronization and clock drift mitigation between disparate audio hardware devices. Based on the analysis conducted, some recommendations were given in order to improve the effectiveness of the DSP calculations.

V. IMPROVEMENTS BASED ON REVIEW

The system was subjected to a Peer-Validation Phase aimed at identifying any potential bottlenecks relating to system throughput. These improvements have been implemented following reviews:

- [1] Latency Reduction: Enhanced the RouterThread synchronization to ensure near-zero latency, strictly avoiding memory allocations during the real-time WASAPI GetBuffer loop.
- [2] Signal Processing Optimization: Refactored the Biquad filter calculations to operate strictly on localized buffers, preventing cache misses and keeping CPU usage below 2%.
- [3] Security and Fault-tolerance: Implementation of robust validation to ensure the application safely handles unexpected device disconnections (hot-unplugging) without crashing the IAudioRenderClient pipeline.

VI. CONCLUSION

This research demonstrates the viability of a Software-Defined multi-device audio routing system. By leveraging an asynchronous tech stack (C++, Qt, WASAPI), the proposed architecture ensures high availability and deterministic

resource allocation. The elimination of mandatory hardware dependencies provides a cost-effective solution for contemporary multimedia sharing challenges.

Future Enhancements

The system can be further improved by incorporating the following features:

- [1] Cross-Platform Integration: Porting the core routing logic to utilize CoreAudio (macOS) and PulseAudio/PipeWire (Linux) to allow the application to function across different desktop ecosystems.
- [2] Advanced DSP Integration: Implementing a VST3 plugin host within the RouterThread to allow users to apply professional-grade studio effects (such as compressors and limiters) rather than basic graphical equalizers.
- [3] Network Audio Streaming: You can connect the system to local Wi-Fi protocols to broadcast the duplicated audio buffers to mobile devices running a companion client.

ACKNOWLEDGMENT

We wish to express our deepest appreciation to our project guide for the guidance, assistance, and constructive suggestions provided continuously during the entire process of developing this project. It was largely due to these suggestions that we could make our project better. We also extend our heartfelt thanks to all the faculty members who have provided us with the necessary facilities to complete this project.

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