Team ECHO: Optimized Audio Processing for Complex Environments and Hearing Aid User Input

UNIVERSITY OF MARYLAND
HONORS COLLEGE

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Motivation

15% Of people in the US are projected to experience **hearing loss** by the time they reach adulthood

80% Of the current eligible user population are not hearing aid users for a variety of reasons ranging from **inaccessible costs** to poor performance in **complex environments**.

Research Goal

Traditional hearing aid audio processing models are trained to handle **individual sounds** such as **speech**.

Team ECHO aims to improve the hearing aid experience by focusing on processing sounds composing a **full environment**. This can be done through the evaluation of **user desires** and exploration of environment-specific classifiers trained with different **audio filtering** techniques.

Research Questions

1. How can we improve the **sound quality** of traditional hearing aids for **users**?
2. How can we **optimize existing signal processing techniques** to better detect desired noises?

TEAM ECHO

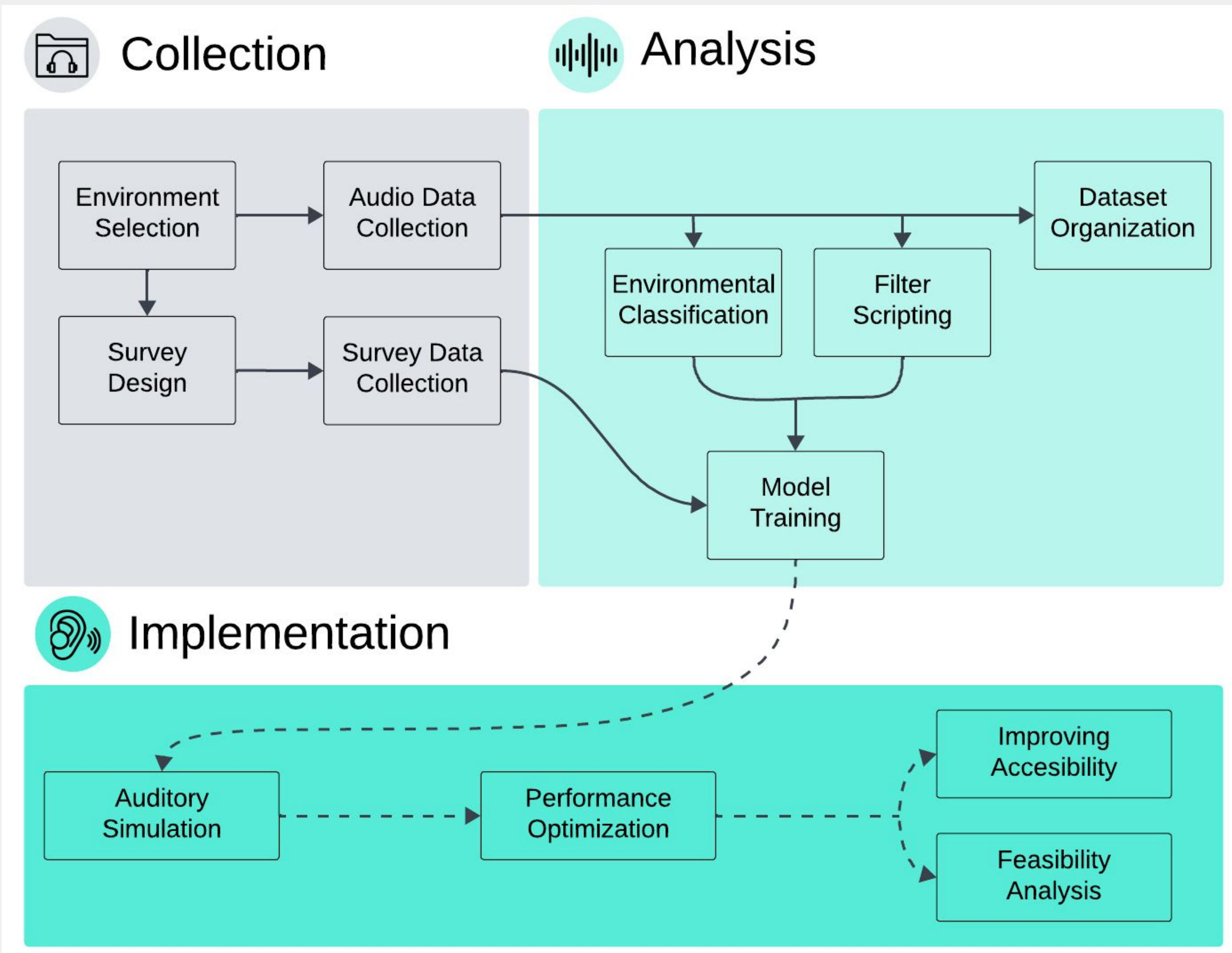
Methodology

Collection

- Set specific audio environment for model creation and begin recording collection
- Divide environmental factors into binary classifications
- Reference **survey** to determine core audio performance indicators

Surrounding vs. Desired Noise Survey

IRB# 2051857-1



Analysis

- Organize and store unique environmental sound data for future research accessibility
- Correlate survey data to filter weights for training the audio processing model
- **Next Steps:** Continue model training for separate environmental factors and finalize evaluation pipeline

Implementation

- **Next Steps:** Simulate audio model
  - Assess technology implementation strengths and feasibility

Performance Metrics:

- Signal-Noise-Ratio (SNR)
- Total Harmonic Distortion (THD)
- Noise Floor
- Dynamic Range
- Crest Factor

Current Results

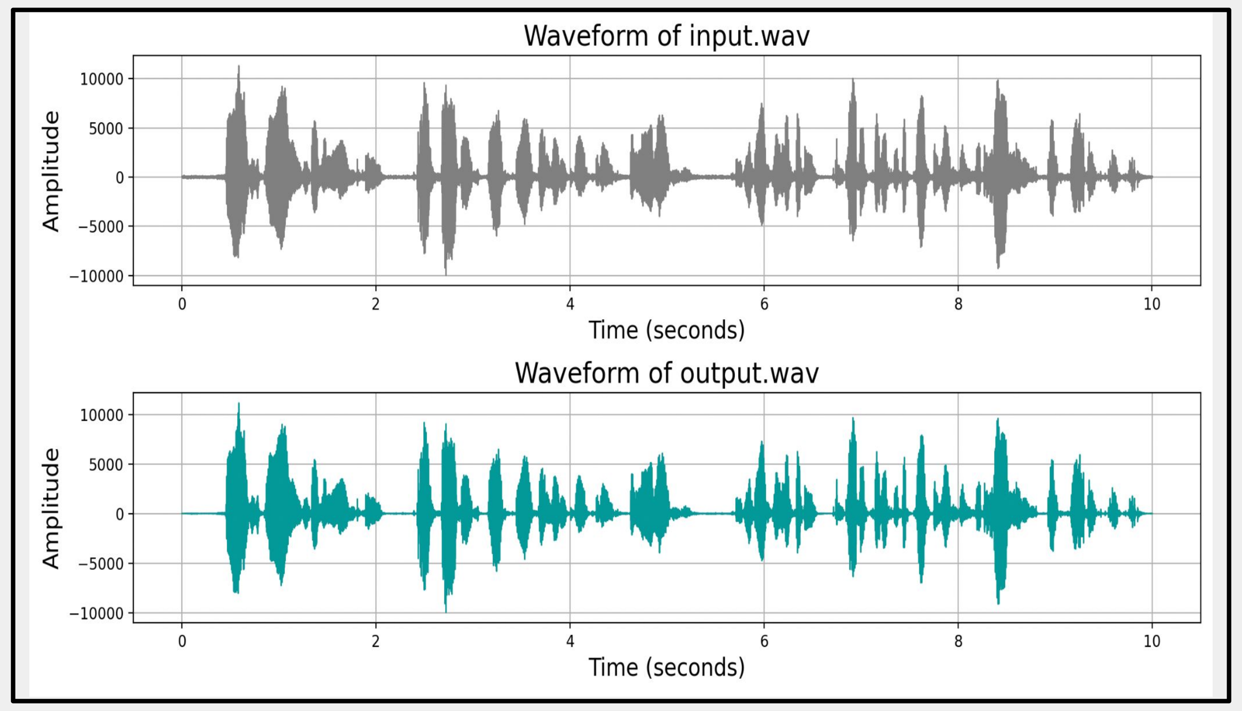


Figure 1: Pre-processed and filtered environmental audio waveforms for indoor, stationary environments using an LMS protocol to reduce background noise.

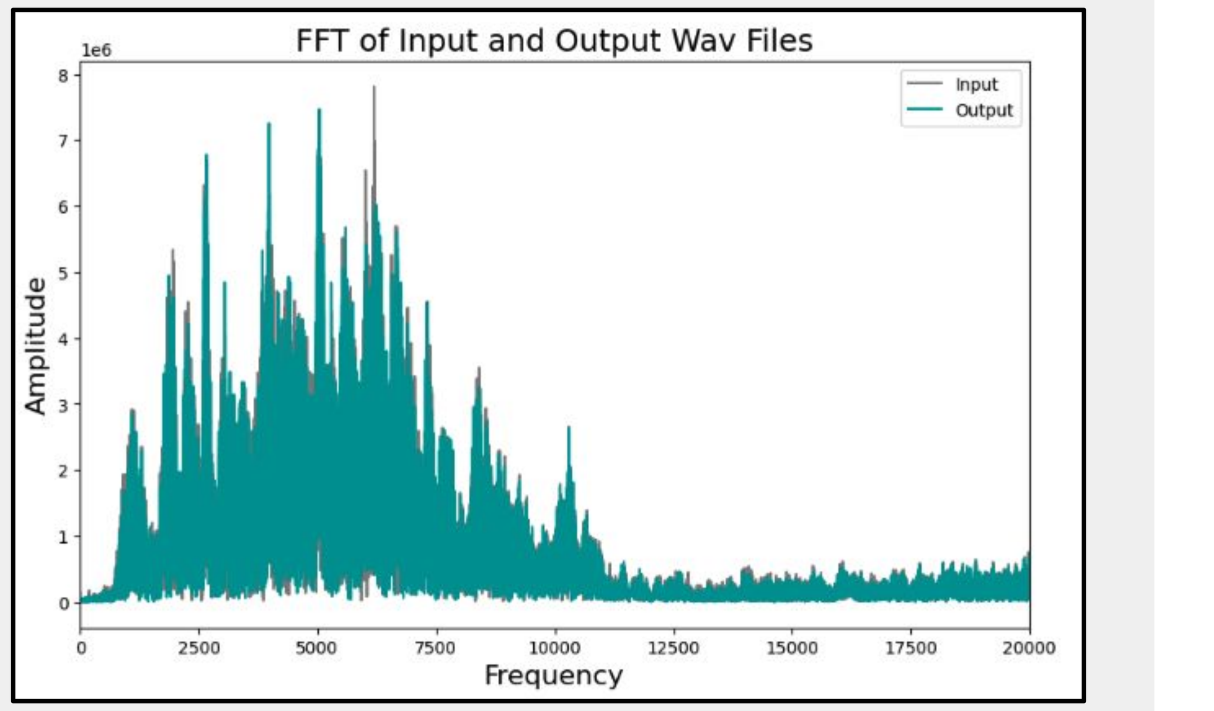


Figure 2: Fourier transform of the indoor stationary environment filter model and outputs demonstrating targeted noise reduction

Deployment Strategy

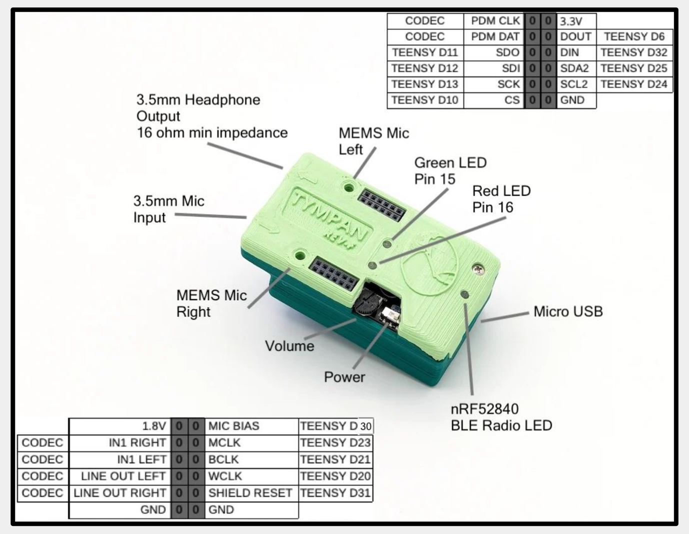


Figure 3: Tympan Rev-F open source hardware to be used for active audio filtering model testing. Compatible with arduino and wireless interfacing.

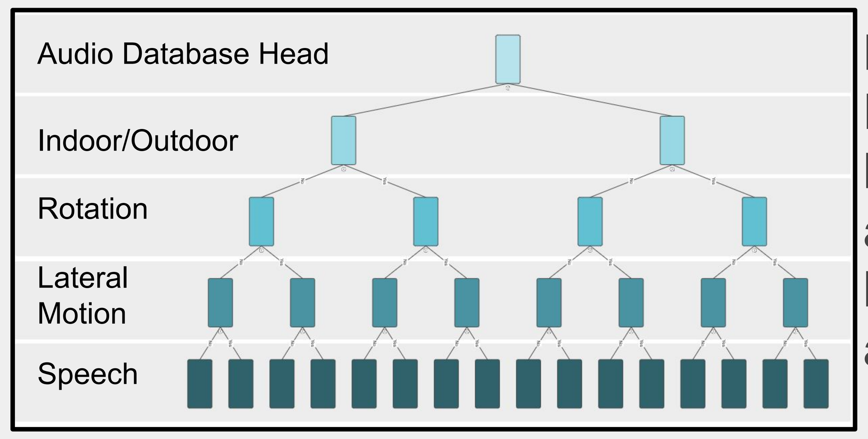


Figure 4: Data collection hierarchy with a four layered binary feature architecture.

Future Research

Integration with wearable **OTC hearing aid** sensors

Investigate Deep Neural Networks (DNNs) to optimize audio model performance and efficiency

Develop dynamic model adjustment for **real-time** environmental conditions

Sources/Acknowledgements

We would like to thank Dr. David Lovell, Dr. Allison Lansverk, our mentor Dr. Sahil Shah, Leslie Lizama, our librarian, Shaunda Vasudev, and the Gemstone Honors Program for supporting this research. We would also like to thank our generous LaunchUMD donors for their financial contributions to our research.

References