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

Lily Li, Rajit Mukhopadhyay, Rahul Nair, Pruthav Patel, Chelsea Reyes, Perfect Sare, Ronoy Sarkar, Bhargav Turnkur, Samuel Waters, Kevin Zhou

Team Mentor: Dr. Sahil Shah
Team Librarian: Shaunda Vasudev

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?

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

Collection
- Environment Selection
- Audio Data Collection

Analysis
- Performance Optimization
- Noise Floor
- Dynamic Range
- Crest Factor

Implementation
- Analytical Environment Selection
- Survey Design
- Survey Data Collection
- Environmental Classification
- Filter Scripting
- Model Training

Deployment Strategy

Figure 3: Tyman 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

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

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.

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