

# Tennessee TECH

# **Active Noise Control Within a Classroom**

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#### Overview

Students have a wide variety of distractions in a classroom at Tennessee Tech, with loud noises being a primary factor. Nearby construction makes a loud low-frequency noise that permeates through the walls and windows. The construction that the campus undergoes creates loud and obnoxious sounds that make it more difficult to focus in a classroom. This project is intended to take in the noise from outside a classroom and produce anti-noise to cancel it without being a distracting annoyance itself.

While the project does partially work, many improvements will have to be made to maximize the efficiency of the design. The main improvements include: Updating the DSP hardware platform for ease of use, updating the error PCB to include input and output connections, creating an integrated PCB for input, output and power subsystems, and the optimization of the algorithm used to actively control the noise.

#### Overall design

Distracting noise gets captured by a microphone and sent to a processor to calculate the proper anti-noise to send to the speaker. The overall sound profile in the room is non-linear and an adaptive filtering algorithm known as Filtered-X Least Mean Square (FXLMS) was utilized to adapt to the changing sound profiles in the room and generate the appropriate anti-noise. Considerations were made for the system to be unobtrusive and non-distracting. This led to a simple and hideable design.

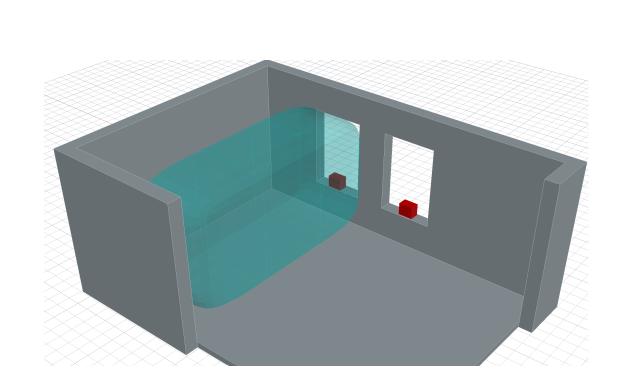


Figure 1. Idealized Area of Effect in a Classroom

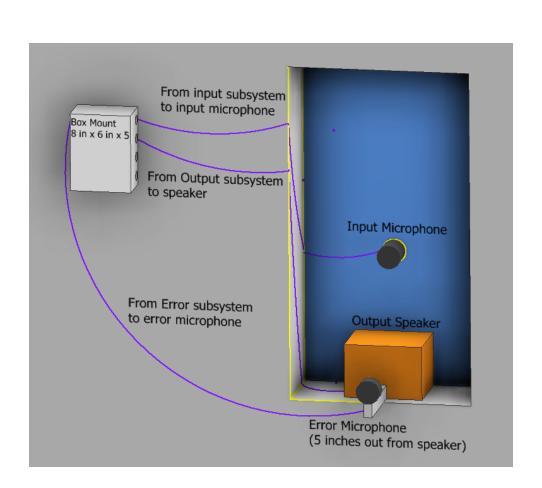


Figure 2. To-scale Mounting by a Classroom Window

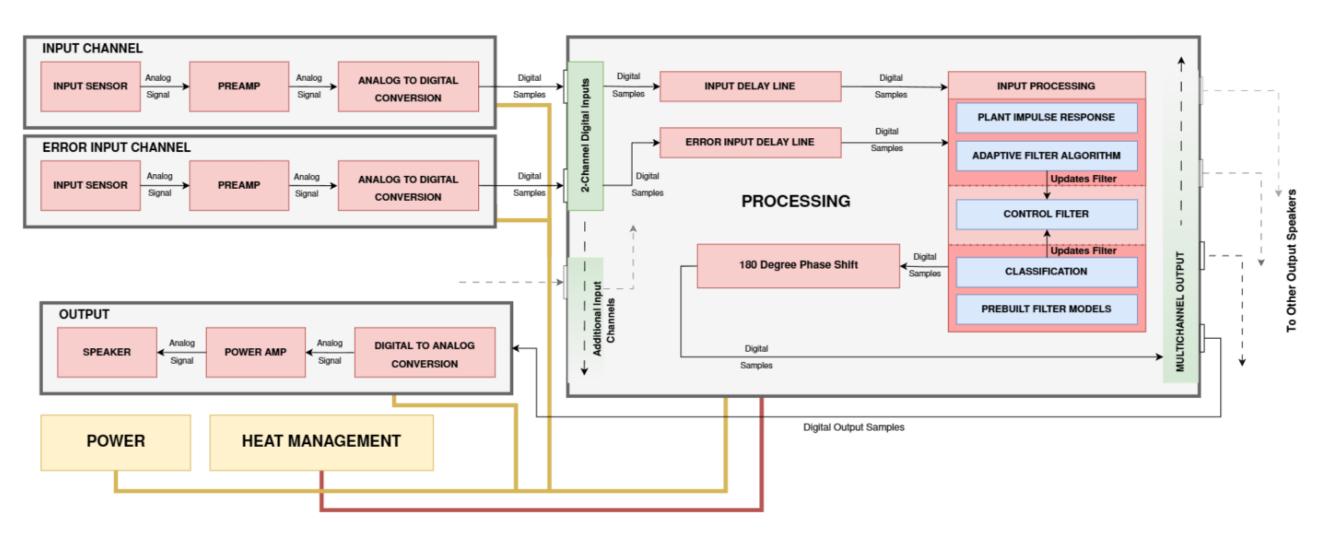


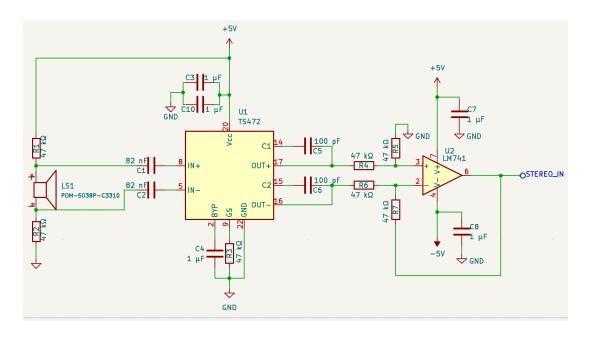
Figure 3. Proposed Area of Effect and Functional Block Diagram

# The Team



Figure 4. From Left to Right: Dylan, Carson, Caleb, Jalene, Jared

# **Working Design**





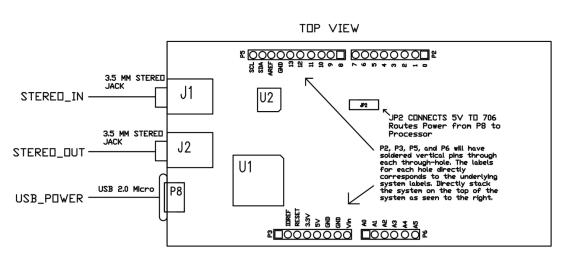


Figure 6. Main Processor Blackfin Schematic

# **Input and Error**

To capture sound vibrations, an electret microphone was employed. The resulting output was amplified before being forwarded to processing. To establish a feedback loop for error minimization, an error microphone was incorporated. The amplified error signal was processed using the same circuit designed for the input subsystem.

#### Output

The system's output configuration is optimized to enhance sound-canceling efficiency. Multiple speakers are strategically positioned to ensure effective redistribution of the processed signal throughout the room.

#### Power

The power subsystem provided 12 V to the output, 5 V to the main processor, and 10 V to the input/error subsystem.

# **Processing**

The system's adaptability is realized through the integration of a **Filtered-X Least Mean Square** algorithm and a 1-dimensional **Convolutional Neural Network** (CNN).

The FXLMS algorithm, widely recognized for its effectiveness in **Active Noise Control** (ANC), employs a Least Mean Squares approach with a filtered input. The filter represents an approximation of the room's impulse response and other transformative sources.

Complementing this, the 1-dimensional Convolutional Neural Network processes an array of samples, aiding in the prediction of the most suitable filter from a pre-existing database for the current scenario.

Bi-directional data exchange between the Digital Signal Processing (DSP) hardware executing FXLMS and the laptop running the CNN model was done through **Bluetooth Low Energy** (BLE). This communication was established using two Arduino devices.

### **Experimental Setup**

- 1. **Setup** a simulation of a classroom with outside noise coming in towards the input microphone and through to the speakers which point to the error microphone
- Record ten-second audio clips using a microphone at the error location with ambient noise three times
  Record ten-second audio clips using a microphone at the error location with generated noise without active noise cancellation three times
- 4. **Record** ten-second audio clips using a microphone at the error location with generated noise with active noise cancellation three times
- 5. **Save** all results, process data with MATLAB

# **Experimental Results**

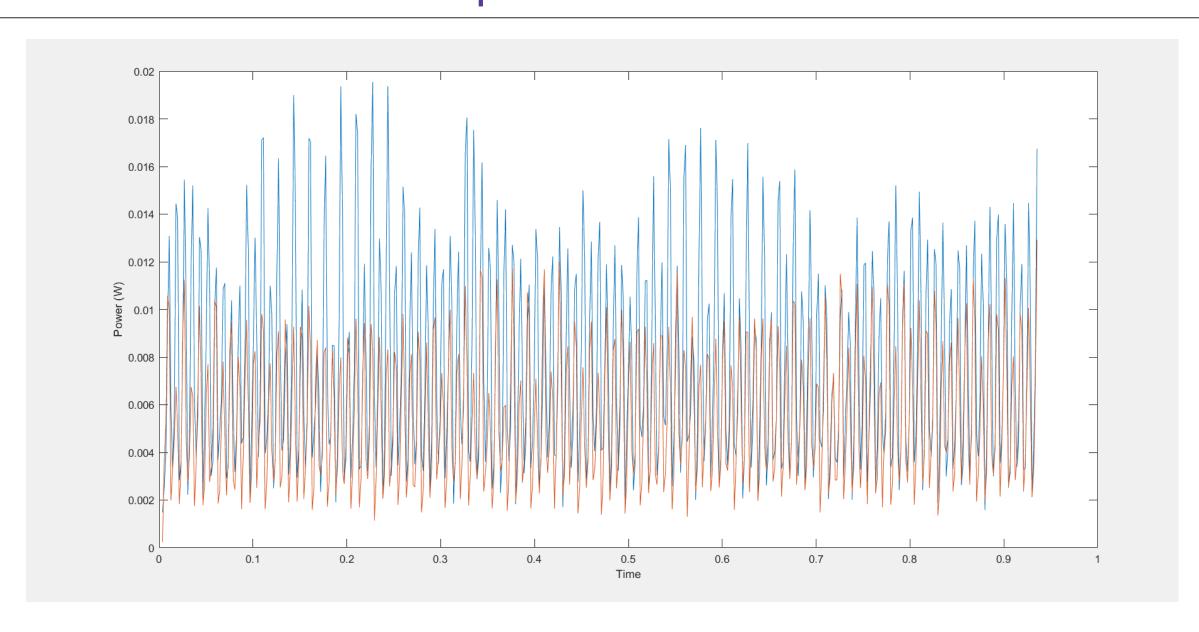


Figure 7. Signal Power Over Time, No ANC (Blue), ANC (Orange)

#### Acknowledgements

We would like to thank Prof. Roberts for his constant support and advice for our design. We would also like to thank Dr. Austen for his assistance in troubleshooting the DSP and for allowing us access to his lab for our experimentation. Dr. Van Neste was also critical for our success in giving our team access to his 3-D printer which made putting together the design much simpler.