

## LGeorge AI Log

### Research:

- YAMNet keeps misclassifying the audio because we don't map enough of the 512 classes to our 7 categories. Even the original classification (before mapping) is often wrong. We want to use a custom ML model. How do you suggest doing this? What audio processing techniques would we need? What kind of model should we use?

### Development:

- Can you write code to calculate the dB level for an audio clip for this file [attached recorder.py].
- I'm realizing that a hardcoded decibel threshold is not feasible because whenever I change rooms I have to change the calibration myself. Can you write code that records the audio and calculates the average dB level during that time and sets a dynamic threshold based on that level.
- Can you add code to [main.py](#) to save everything to an SQLite database and create a UI (website) that records a log of all detected audio events (per event: timestamp, sound type, confidence of classification, dB level). First explain what you are going to do, and once I approve, create and modify all necessary files to do this.

### Debugging/Testing:

- Add logging and exception handling in all files to handle possible errors. And also an endpoint in node2's [main.py](#) for me to conduct a health check on the connection.
- I can't run the server yet because the model is not complete. For now could you write a dummy model script so I can test that the communication protocol works.
- I keep getting dependency issues when I try to run the code. A while ago I downloaded many python dependencies into a custom directory, however I believe that is now messing with my newer projects. How do I fix this? Here is this specific error:
  - × Getting requirements to build wheel did not run successfully.
  - | exit code: 1
  - ↳ [33 lines of output]
  - Traceback (most recent call last):
  - File
  - "/Users/userb/Desktop/quadrupeds/miniforge3/lib/python3.12/site-packages/pip/\_vendor/pyproject\_hooks/\_in\_process/\_in\_process.py", line 353, in <module>
- I want to test that the audio data is being recorded correctly. Can you write a chunk of code that uses os (or another library, if that is preferable), to save the 4 second WAV clips into a folder so I can listen to it myself.