Development Progress Report 1 - Negar Ghasemi

In the attached document, I have delivered the 2 main cores of the project:
The code for analyzer, and another code to handle the file that the user shares. The analyzer includes implementing the audio from upload.js, running it through biquad filters, calculating the RMS, and normalizing it by dB. What the project currently lack are listed below:

1) Comparison with the ideal ratio
    a) I achieved this using ChatGPT but only to be able to determine whether the analyzed values match the ideal ratios within the original audio or not.
2) The ideal ratio model itself needs to be achieved by running an RMS calculator in Pro Tools on the three frequency bands for a selected audio.
3) On-page feedback to the user including how much the amplitude of each frequency band needs to be adjusted.
4) Better UI design

The biggest challenge with this step was that despite being aware of the tools that I needed to use, I was not aware of the logic or method through which the elements needed to be used. I knew the basic logic of the smallest sections, but not how to make them interact. I need to be honest that the process of developing software was to sit alongside AI, asking for the big picture, letting it code, and then asking it for further details so I can learn the logic through which I can code. Sometimes the answers I got were so different from what I thought I had learned, and so I asked for different coding approaches to explore how I can do this project in the simplest format. I developed the software step by step starting from the upload function, then deciding whether to use a class or a separate module for the analysis process. Then I developed the filtering process and implemented the analyzer tool. The calculation for RMS and normalization by dB was something that I had no clue how to do other than a simple formula. More specifically, I did not know how to use the data that were acquired from the analyzer tool, and encountered errors several times with the “audioanalyzer” which I ultimately debugged to be a scoping issue and had to declare it globally. The other problems I encountered was insufficient use of the buffer object which only played the audio back, but the code did not get passed the buffer object to do the analysis properly.
Currently, I feel happy that the core functions of the project are developed, yet a little upset that I had to depend a lot on an external source, not knowing how to code from scratch myself. I am planning to implement the mentioned details.