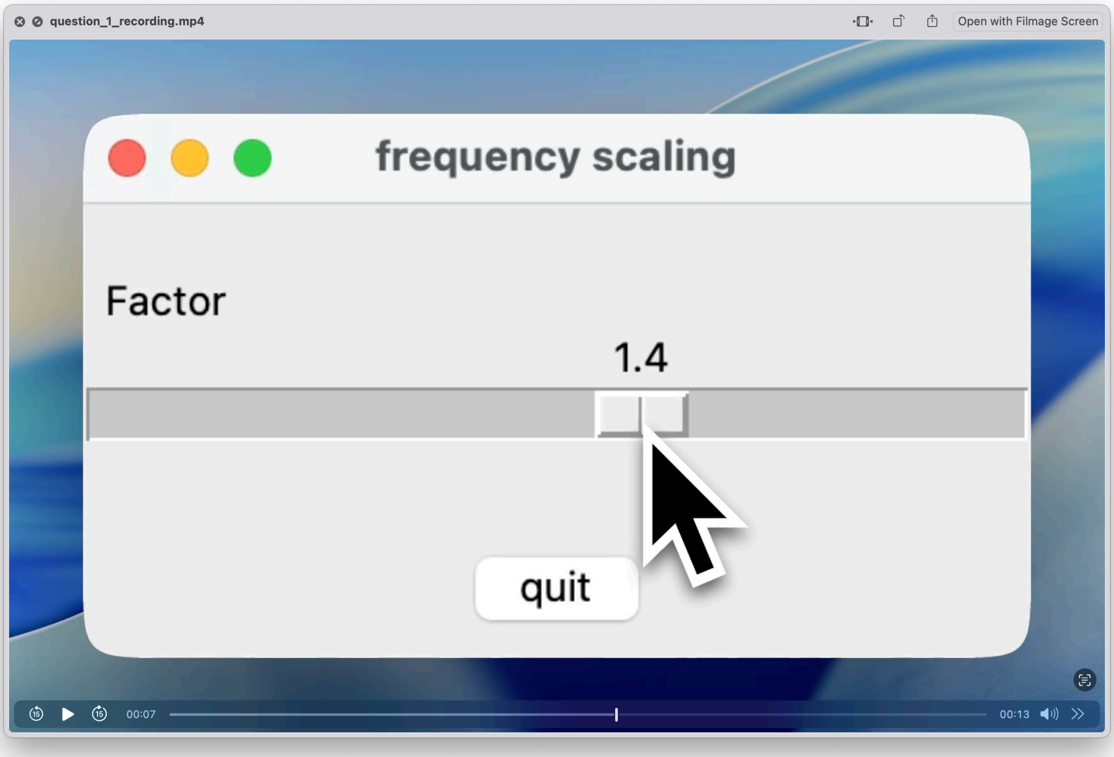
DSP Lab Exam Question 1



Overview

Implements a real-time frequency scaling of audio signals using FFT interpolation, users can modify pitch while preserving harmonic structure.

Core Methods

`process_block()`

- For each output frequency bin k, calculates source position: `old_index = k / scaling_factor`
- Interpolates between adjacent FFT bins to get scaled spectrum
- Returns time-domain signal via inverse FFT

`audio_callback()`

- Retrieves current audio chunk from file buffer
- Processes block with current scaling factor from slider
- Writes processed audio to PyAudio stream

Technical Details

Frequency Scaling Algorithm

```
For each output frequency bin k:
 source_position = k / α
 Interpolate between floor(source_position) and ceil(source_position)
 scaled_spectrum[k] = weighted_average(spectrum[floor], spectrum[ceil])
```