Individual Assignment 2

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1. Introduction

This assignment presents the design and analysis of a Digital Signal Processing (DSP) system, implemented in MATLAB, to perform audio signal recording, processing, and playback. The system includes three primary functionalities: filter design via pole-zero placement, a 12-band graphical equalizer, and an echo effect with variable delay. The MATLAB code enables visualization of the original and processed signals, showcasing the effects of each processing step.

2. Methods

The proposed DSP system comprises the following key elements:

- **Signals Input/Output**
  Input: Audio signal captured using MATLAB with microphone and digitized at a sampling frequency of 44.1 kHz.
  Output: The processed signal is played back through the speakers for user evaluation.
- **Digital Signal Processing Core**
  Pole-Zero Placement for Filter Design: A simple interactive filter constructed using predefined poles and zeros.
  Equalizer: A 12-band equalizer to adjust specific frequency ranges.
  Echo Effect: A delay-based feature to add depth and ambiance to the audio signal.
- **User Interface (UI)**
  Displays processed signal waveforms and pole-zero plots. This allows parameter adjustments for real-time interaction.

3. Functional Description

- **Pole-Zero Filter Design:** A filter is implemented with a single zero at $z = 0.8$ and no poles. The filter suppresses frequencies near the zero location. **Figure 1** shows the pole-zero plot, where the absence of poles indicates a stable FIR structure. The filtered signal **Figure 2** demonstrates attenuation, making it useful for noise suppression. The simplicity of FIR filters ensures stability, and visualize poles/zeros
- **12-Band Equalizer:** The audio spectrum is segmented into 12 bands (e.g., 50 Hz to 20 kHz). Each band's gain is adjustable, allowing users to amplify or attenuate specific frequencies. For example, boosting lower frequencies enhances bass, while reducing higher frequencies minimizes harshness. Real-time updates ensure the system remains responsive to user changes.
- **Echo Effect:** The echo is achieved by adding a delayed version of the signal,

$$y[n] = x[n] + 0.5 * x[n - D]$$

  where $D = 22050$ samples (corresponding to a 0.5-second delay). The echo signal **Figure 2** exhibits repeated patterns with diminishing amplitudes, simulating reverberation. The FIR structure avoids feedback instability.

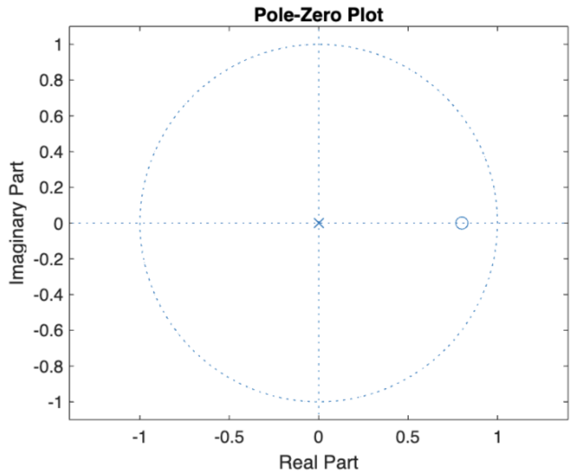

Figure 1. Pole-Zero Plot

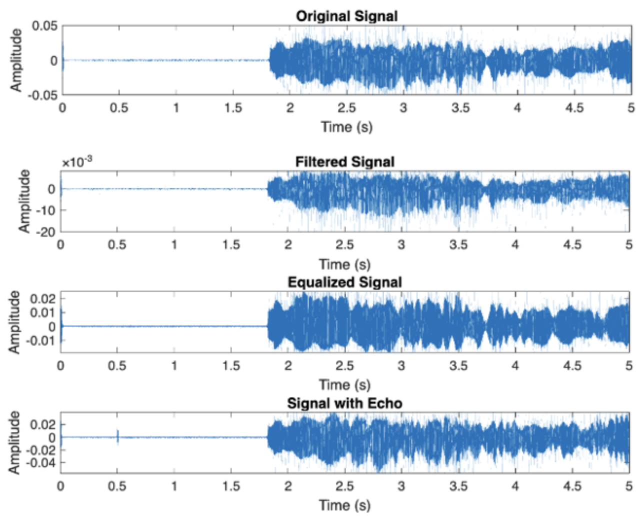

Figure 2. Original, Filtered, Equalized, Echo Signal Plots

4. Results

- **Original Signal:** The original signal in **Figure 2** shows a raw waveform with consistent amplitude. It serves as a baseline for evaluating subsequent processing steps.
- **Filtered Signal:** The filtered signal in **Figure 2** reflects attenuation, as expected from the zero at $z = 0.8$. The reduction in high-frequency components aligns with the filter design.
- **Equalized Signal:** The equalizer introduces distinct amplitude changes across frequency bands, as seen in **Figure 2**.

- **Echoed Signal:** The echoed signal in **Figure 2** demonstrates repeated waveforms, creating depth. The delayed signal blends well with the original, producing a natural echo effect.

5. Conclusion

his DSP system effectively demonstrates key signal processing functionalities, including filter design, equalization, and echo effects. The MATLAB implementation highlights the effects of these features through visualizations and playback. While the system has some limitations, such as computational constraints and user-friendliness, it serves as a robust framework for audio processing applications.