Homework Assignment 5
Interpolation Filters for Upsampling

ECEN 463 - Digital Signal Processing
University of Nebraska-Lincoln
Instructor: Maxx Seminario

Due: November 21, 2024

Assignment Overview

This homework assignment focuses on interpolation filters for upsampling digital audio signals. You will implement and compare various interpolation techniques, analyze their frequency responses, and evaluate their performance through both objective plots and subjective listening tests.

Instructions

- Complete all programming tasks with detailed MATLAB implementation
- Submit your solutions as a **single PDF report** (2-3 pages using IEEE format template) via Canvas
- Attach your **MATLAB code as a separate document**
- Include nicely labeled plots with time-aligned signal segments
- Provide concise descriptions of audio quality and observations
- IEEE formatting templates available at: `https://www.ieee.org/conferences/publishing/templates.html`

Problem Statement

Write a MATLAB program to read and upsample the provided `voice_samp_8k.wav` file (sampled at $F_s = 8000$ samples per second) by an integer factor L (choose a value in the range 3 to 7).

Required Implementations

Implement and compare the following five interpolation approaches:

(i) **No Interpolation Filtering:** Upsample only with zero insertion, no filtering

(ii) **Hold Interpolation:** Implement as a causal FIR filter using the `filter()` function

(iii) **Linear Interpolation:** Implement as a causal FIR filter using the `filter()` function

(iv) **Short FIR Lowpass Filter:** Design a shorter FIR interpolation filter (∼20–200 taps) using `fir1()`

(v) **Long FIR Lowpass Filter:** Design a longer FIR interpolation filter (∼200–2000 taps) using `fir1()`

Analysis Requirements

For each interpolation method:

- Listen to the output using `soundsc()`
- Plot short segments (∼50–100 samples) of the output signals
- Create time-aligned plots comparing all five methods
- Plot frequency responses using `freqz()` where applicable
- Verify correct operation through visual inspection of plots

Report Requirements

Your write-up should include:

- Description of the sounds heard for each interpolation approach
- Analysis of the time-aligned output segment plots
- Discussion of frequency response characteristics
- Any interesting observations about filter performance
- Comparison of computational complexity vs. audio quality

Helpful MATLAB Functions

- `audioread()` – Read the input speech file (12.5 seconds)
- `kron()` – Single-line upsampling implementation
- `fir1()` – FIR filter design
- `filter()` – Apply filtering to upsampled signals
- `freqz()` – Frequency response analysis
- `soundsc()` – Listen to audio outputs
- `plot()` – Create visualizations

Additional Notes

- The value of concision (complete and brief) is important in technical writing
- Bring questions to class for discussion
- Consider: "What do you mean by time-aligned plot?"
- Ensure all plots are properly labeled with axes, titles, and legends

Academic Integrity

This is an individual assignment. While you may discuss general concepts with classmates, all submitted work must be your own. Copying solutions from other students, online sources, or solution manuals constitutes academic dishonesty and will result in a failing grade for the assignment.