Homework 6

Preamble

Please submit to Blackboard exactly two files:

1) A single ZIP file with all your MATLAB (or Python) programs, along with any data files necessary to run your programs.
2) A single report document in PDF. Your report should include procedures and solutions to analytical problems, screen shots of results and graphs from programs, and discussions. Please do not include copies of your code in the report.

Submit your solutions to Blackboard by the deadline.

This homework may be done in groups of two students. You are welcome to discuss your approach with others, but your answers need to be unique. If your answers are found to be highly similar to others' or to some material published, you may be subject to a review of academic integrity.

Part 1 (Both ECE 4364 & 5364 students)

Problem 1 (15 points)

We wish to create a solution to decimate audio files from 48000 samples/second to 8000 samples/second, without causing too much distortion to the frequencies that are kept.

a) Design an anti-aliasing FIR filter that you consider appropriate for this purpose.
   - Explain your design choice: passband ripple, stopband attenuation, fs, fc, design method
   - Plot the frequency response of your filter over frequency in Hz
b) Apply your filter to audio from toneramp.wav (still at 48000 samples/second)
   - Plot the resulting amplitude over time
   - Explain why this result makes sense
c) Implement the rest of the decimation process in MATLAB
   - Save the decimated signal into a new wav-file, which should have 8kHz sampling frequency. Submit this file as part of your solution.
   - Plot the amplitude of the new wav-file and compare it to your graph from part b). Explain what you observe, and why it makes sense.
d) Apply your decimation to the audio file equinox-48kHz.wav.
   - Save your decimated wav-file and submit it as part of your solution.
   - Listen to the audio file and compare it with the original 48 kHz version. Comment on what you hear, and why it makes sense.

Problem 2 (15 points)

We wish to create a solution to interpolate audio files from 8000 samples/second to 48000 samples/second, without causing too much distortion to the signal present.

a) Implement the sample rate increase in MATLAB (by zero insertion)
   - Apply this process to the audio file from problem 1, part c).
   - Plot the spectrum of the signal before and after zero-insertion. Explain what you observe and why it makes sense.

b) Design an FIR image-rejection filter that you consider appropriate for this purpose.
   - Explain your design choice: passband ripple, stopband attenuation, fs, fc, design method
   - Plot the frequency response of your filter
   - Apply it to the result of part a)
   - Plot the spectrum of the signal over frequency in Hz after filtering. Explain what you observe and why it makes sense.
   - Plot the amplitude over time of the filtered signal. Explain what you observe and why it makes sense.

c) Apply your interpolation solution to the audio file from problem 1, part d)
   - Save your interpolated wav-file and submit it as part of your solution. This should be a 48 kHz file.
   - Listen to the audio file and compare it with the original 48 kHz version. Comment on what you hear, and why it makes sense.

- We want to minimize the number of coefficients in the filter.

Problem 3 (15 points)

We wish to create a solution that can split audio files at 48000 samples/second, into two audio files, each at 24000 samples/second. The first file should contain the lower half of the representable frequencies, and the second file the upper half of the frequencies.

a) Design a solution to extract the lower-frequency band (up to 12 kHz), decimate to 24000 samples/second, and save it to a wav-file
   - Show the frequency response of your anti-aliasing filter
   - Apply it to the audio file equinox-48kHz.wav
   - Show the spectrum of the signal before and after conversion. Explain why this makes sense.
   - Listen to the resulting audio file. Explain what you hear and why it makes sense.

b) Design a solution to extract the upper-frequency band (from 12kHz onward), decimate to 24000 samples/second, and save it to a wav-file
   - Show the frequency response of your image-rejection filter
   - Apply it to the audio file equinox-48kHz.wav
   - Show the spectrum of the signal before and after conversion. Explain why this makes sense.
   - Listen to the resulting audio file. Explain what you hear and why it makes sense.

Part 2 (Only ECE 5364 students)

Problem 4 (10 points)
For the filter you designed in problem 1, create a MATLAB program that implements it in an efficient way using polyphase filters. Verify the correctness of your implementation by comparing the output signal to the one from problem 1.

Problem 5 (15 points)
We wish to invert the process done in problem 3. We want to combine the audio contained in two separate 24 kHz audio files, each one corresponding to a separate band, into a single 48 kHz audio file.

a) Create a MATLAB program that can do this. Here are some hints:
   - Read the lower-band file, upsample it to 48 kHz, and remove any images that may result in the process
   - Read the upper-band file, upsample it to 48 kHz, and shift the audio to the upper part of the spectrum.
     - You can do this by creating spectral images (by zero-insertion), and then filtering the lower-band
   - Combine the two bands at 48 kHz into a single waveform (by addition)
   - Store the result into a 48 kHz wav-file
b) Apply your solution to the two 24 kHz files you created in problem 3
c) Compare your result with your original file (equinox-48kHz.wav)
   - Did you lose any significant part of your spectrum?
   - Can you perceive any degradation in audio quality?
   - Measure the distortion as the mean-absolute-error of each sample. Note that the filtering process introduces a delay, so you may have to re-align the signals before making the comparison.

