ECE 5630: Digital Signal and Image Processing - Multi-Rate Processing

Due: November 14, 2014 at 5:00 pm

Name (Print):

Objectives

This programming assignment has three objectives:

1. Give you practice in using a program to perform digital signal processing.
2. Help to solidify your understanding of multi-rate processing, and in particular the concepts of decimation and interpolation.
3. Determine the specifications required for a digital filter.

Instructions

1. Use a sample rate conversion of $\frac{L}{M} = \frac{3}{4}$ for this assignment.
2. The program should be written in C or C++.
3. Use the `.wav` file `ghostbustersray.wav` to test your program. This file is available on Canvas and is sampled at 11,025 Hz.
4. Use the Matlab tool `fdatool` to design your FIR filter.
5. For all of the frequency response plots, the normalized frequency running along the x-axis in your plot should be scaled to units of cycles/sample and should cover the range from 0 to 1.
6. For all of the magnitude frequency response plots, the y-axis should be scaled to units of dB by computing $(20 \log_{10} |H(f)|)$.
7. Attach your work, answers, plots, and code to this page with a single staple in the top left-hand corner.

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1. Design a linear phase FIR low-pass (prototype) filter to be used in your sample-rate converter. Be sure that the cutoff of the filter is appropriate for the conversion.
   (a) Plot the impulse response $h(n)$ of the filter.
   (b) Plot the magnitude and phase response.
   (c) What is the length of the filter?
   (d) What method did you use to design the filter?
   (e) What are the pass-band and stop-band edge frequencies in the filter you designed?
   (f) What is the size of the ripple in the pass-band (make this small ≤ 0.001 dB)?
   (g) What is the peak side-lobe level (in dB) (make this ≤ 80 dB)?
2. Using the simple and noble identities, derive the signal flow graph for a polyphase filter that can change your sample rate by a factor of $\frac{3}{4}$ using the minimum number of operations/sample. Draw the final signal flow graph that will be implemented in your program. Assume an input sample rate of 11.025 kHz.

(a) What is the final sample rate of the output?

(b) What is the number of operations (multiplies and adds) required per input sample?

3. If the prototype filter had an impulse response $h(n)$ of length K, show how you would deal the coefficients to the $M \times L$ polyphase type filters $R_{l,m}(z), l = 0, \ldots, L, m = 0, \ldots, M$.

4. Using the prototype filter you designed, plot the magnitude and phase responses of the 12 polyphase filters $R_{l,m}(z)$.

5. In C or C++, write code to implement the straightforward realization for decimation ($\frac{L}{M}$) by upsampling by L, filtering with the low-pass filter with response $H(z)$, and downsampling by M. Then do the following.

   (a) Generate a cosine signal with frequency f_0, $x(n) = \cos(2\pi f_0 n)$ for $n = 0, 1, \cdots, N-1$. Let N be chosen so that it is at least 30 times the length of the $h(n)$ filter. If possible, let N be 100 times longer than the length of your prototype filter. Pass $x(n)$ through the decimating low-pass filter. Let $y(n)$ be the output.

   (b) Plot the magnitude response of $x(n)$ and $y(n)$.

   (c) Repeat this for each of the frequencies f_0 in the table below. Record in the table the frequency $\hat{f}_0$ of the decimated signal.

f_0	$\hat{f}_0$
$\frac{1}{16}$	
$\frac{1}{8}$	
$\frac{1}{4}$	

6. In C or C++, write code to implement the polyphase filter bank realization for decimation (by $\frac{L}{M}$) using the low-pass filter with response $H(z)$. Then repeat 5(a)–(c).

f_0	$\hat{f}_0$
$\frac{1}{16}$	
$\frac{1}{8}$	
$\frac{1}{4}$	

7. Use the Matlab function `wavread()` to generate the samples of the file `ghostbustersray.wav`. Decimate the signal using your program. Import the result back into Matlab, and play the original signal (using `sound()`) at a sampling rate of 11,025 Hz, and the processed signal at the decimated sample rate. Do they sound the same? Write out the final results in a .wav file for the instructor to listen to.