GPGN 404
2nd Midterm Exam
November 7, 2008 **Name:** ______________________

Question:	1	2	Total
Points:	25	25	50
Score:			

Question 1 . . . . (25 points)

A digital signal $x[n]$ with a sampling interval $T = 2$ ms is contaminated with noise at *two* frequencies: 0 (DC) and 250 Hz. Your task is to design and implement a filter to attenuate the noise, without significantly altering the amplitude and phase for other frequencies in the output sequence $y[n]$.

(a) [4 points] Determine the Nyquist frequency F_N (in Hz) for the input and output sequences $x[n]$ and $y[n]$.

(b) [4 points] Sketch the locations in the complex z-plane of two poles and two zeros for a causal and stable filter that will eliminate the noise at the two frequencies 0 and 250 Hz, while mostly preserving the signal for other frequencies.

(c) [4 points] Specify the system response $H(z)$ for your filter, including the region of convergence (ROC). Include in your $H(z)$ a scale factor so that the the amplitude response at 125 Hz is *exactly* one.

(d) [5 points] Sketch the amplitude and phase responses $A(F)$ and $\phi(F)$ of your filter for frequencies $-F_N \le F \le F_N$. Label all axes.

(e) [4 points] Write a linear constant-coefficient difference equation for your filter in terms of the input and output sequences $x[n]$ and $y[n]$. Specify numerical values for the coefficients in your equation.

(f) [4 points] Write a computer program fragment that implements your filter. Include in your fragment proper handling of the beginnings and ends of the input and output sequences $x[n]$ and $y[n]$.

Question 2 . . . . (25 points)

The Nyquist frequency for sampled music on a CD is approximately 22 kHz.

(a) [3 points] What is the CD sampling frequency, in kHz?

(b) [3 points] The Nyquist frequency 22 kHz was chosen because:

- typical humans cannot hear anything above 22 kHz?
- frequencies of acoustic waves in air are less than 22 kHz?
- both of the above?

(c) [3 points] Depending on your answer to the previous question, an analog low-pass filter may (or may not) be required *before* the musical recording is sampled. Why is (or is not) such a filter necessary?

(d) [4 points] Sketch the amplitude spectrum (with labeled axes) of sampled CD music, for frequencies between -44 and 44 kHz.

(e) [4 points] In his misspent youth Professor Hale listened too often to loud music, so he can no longer hear anything above 11 kHz. Describe a digital system that will convert *without aliasing* input sequences $x[n]$ in his music collection into output sequences $y[n]$ requiring only half the samples.

(f) [4 points] Audiophile Sam wants his music more finely sampled than on standard CDs. Write a computer program fragment that will resample his CDs so that output sequences $y[n]$ with length nty have half the sampling interval of input sequences $x[n]$ with length $ntx = nty/2$. (Hint: your program need not include the output/input sampling intervals, just their ratio, which is 1/2.)

(g) [4 points] Sketch the amplitude spectrum (with labeled axes) of Sam's CD music *after resampling* with your computer program above, for frequencies between -44 and 44 kHz.