EECS 452 Midterm Exam
Winter 2011

Name: _______________________________  unique name: _______________

Sign the honor code:

I have neither given nor received aid on this exam nor observed anyone else doing so.

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Scores:

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<tr>
<td>Section I</td>
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NOTES:

- Open book, open notes.
- There are 8 pages including this one.
- Calculators are allowed, but no PDAs, Portables, Cell phones, etc.
- You have 120 minutes for the exam.
- Be sure to show work and explain what you’ve done when asked to do so. You will not receive partial credit without showing work.
- In the context of the exam you are to assume all signed numbers are two’s complement numbers.
Section I -- Short answer 40 points

1) Convert the Hexadecimal value 0xC3 into base 10 assuming it is: [4]

   a) An unsigned binary number: __________
   b) A 2’s complement number: __________
   c) An unsigned Q6 number: __________
   d) A signed (two’s complement) Q6 number __________

2) A FIR filter has the impulse response shown below [6]

   a) Find and plot the magnitude and phase of this filter’s transfer function? [3]

   b) Is this filter linear phase? If so what is its group delay? [3]
3) Show all the steps involved in multiplying the Q7 two’s complement numbers 0x88 and 0x0A together. Express the result of the multiplication in both Q14 and in Q7. [4]

4) If you add together the following Q7 two’s complement numbers (without saturation) from left to right will there ever be overflow? If so when does it occur and will it affect the final result? Here are is the addition: 0x81+0xFD+0x10. [3]

5) A frequency resolution of at least 50Hz is required to perform spectral analysis using the FFT of a waveform sampled at 48kHz. What is the minimum number of points N that are required if the FFT is radix 2 N-point? [3]

6) Let’s say you applied a uniform A to D converter to convert values in the range 5V to -5V to a B-bit two’s complement value.
   a) If the ADC is uncompensated but with no other distortions what is the minimum value of B required for the maximum quantization error to be less than 1mV? [2]

   b) If the ADC is compensated what is the minimum value of B required for the maximum quantization error to be less than 1mV? [2]
c) What is the mean squared quantization error for the compensated ADC of part b? [3]

7) Say you took a 16-point DFT of the following signals sampled at fs=12kHz starting at time t=0. In each case state the number and location of non-zero values of the DFT. [3 each]

   a) \(4 \sin(2 \pi \times 3000t)\)

   b) \(\cos(2 \pi \times 2000t) + 1\)

8) Write the transfer function of the following filter. [3]

9) A filter has impulse response of the form \(h[n] = (0.9)^n\) for \(n=0,2,4\ldots\) and \(h[n]=0\) for \(n=1,3,5\ldots\) Draw the pole zero constellation and classify the filter as to FIR vs IIR, LP vs HP vs BP vs BS (band stop). [3]
1) DFT vs FFT
   a) Let \( \{x[n]\} \) and \( \{y[n]\} \) be sets of \( N \) samples, i.e. \( n=0, \ldots, N-1 \), of real signals \( x(t) \) and \( y(t) \) sampled at identical sampling rates \( f_s \). Explain how you can compute the DFTs \( X[k] \) and \( Y[k] \) simultaneously with a single FFT. [5]

   b) Now let \( \{x[n]\} \) and \( \{y[m]\} \) be sets of samples of different lengths, i.e., \( n=0, \ldots, N_x-1 \) and \( m=0, \ldots, N_y-1 \). Show how you can compute their DFTs with a single FFT by using zero padding. How do you interpret the DFT frequencies for \( X[k] \) and \( Y[k] \) in terms of Herzian frequency if the sampling rates on the two signals are not identical? [5]
2) A 32 bit PIPO register is implemented using 32 D flip flops. Assuming that the average number of bit transitions per second are equal to half the clock frequency for all bits, what is the total power dissipation of this register and how does it depend on the logic voltage and the clock frequency? [10]

3) You want to use a 512-point DFT to detect the presence of a sinusoid at one of two frequencies 200kHz and 212kHz. What is the minimum sampling frequency required to do this without leakage and without aliasing? [10]
1) An arbitrary waveform synthesizer uses the DDS system above to generate a periodic waveform using a waveform table containing samples of a single period of a waveform $s[0],...,s[2^B-1]$ where $B=B_A$. Assume that $s[n]$ is a ramp, $s[n]=n2^B$ in the following.

a) Assume a clock rate $f_s=10$MHz, $B_A=B_{FTV}=16$, and $FTV=1$. Also assume an ideal ADC (e.g., cardinal series reconstruction) and plot three or four periods of the analog output over time. Be careful to label your axes and indicate the amplitude and period of the waveform on your plot. Repeat for $FTV=2$. [7]

b) Assuming the same parameters as above explain what would happen if $FTV=-1$? Plot three or four periods of the waveform. Repeat for $FTV=-2$. [8]
2) A transfer function having the pole-zero plot below is to be implemented in Q15 arithmetic according to the method discussed in class and performed in Lab 5. [15]

(i) How many biquads are necessary to implement this filter? [5]

(ii) By drawing on the pole/zero plot above, show how you would pair poles and zeros to form each biquad. Assign a label “1”, “2”, etc to each to show the ordering of the biquad cascade filter from left (input X) to right (output Y). [5]

(iii) To implement this filter in Q15 would you need to scale the filter coefficients? If so how would you do it? [5]

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