EECS 452 Practice Midterm Exam

Fall 2014

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:

#	Points
Section I	/40
Section II	/30
Section III	/30
Total	**/100**

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.
- Unless otherwise specified 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]

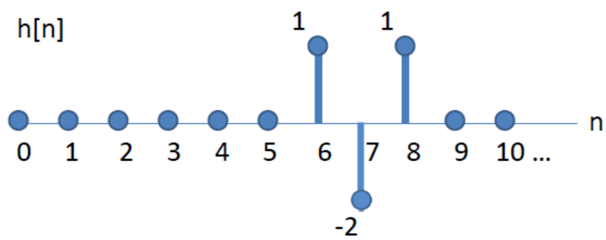
   a) An unsigned binary number: ___________

   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]

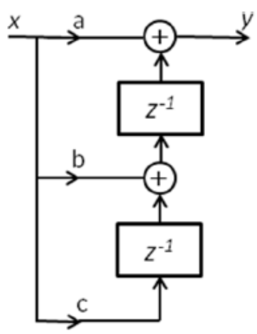
3) If you add together the following Q7 two's complement numbers (without saturation) from left to right will there ever be overflow? If so when does it occur and will it affect the final result? Here is the addition: 0x81+0xFD+0x10. [3]

4) A waveform is sampled using a sample rate of 48 kHz. What is minimum size of the FFT required (minimum number of points in the FFT) to achieve a frequency spacing of less than 100Hz in the DFT spectrum of the samples? [6]

5) A compensated quantizer has an input range from -8V to +8V. Assuming the signal amplitudes are distributed uniformly over any quantization interval, what is the minimum number of bits required to guarantee that the maximum quantization error is less than 0.01V? What are the fewest bits required to guarantee that the root mean squared quantization error is less than 0.01V? [6]

6) In Lab 6 we needed to perform an FFT on collected audio data and collect audio data at the same time. How can we use interrupts and two buffers to help us perform FFTs on data without missing data samples? [5]

7) Write the transfer function of the following filter. [5]



8) A filter has impulse response of the form form $h[n]=(0.9)^n$ for n=0,2,4… and h[n]=0 for n=1,3,5… Draw the pole zero constellation and classify the filter as to FIR vs IIR, LP vs HP vs BP vs BS (band stop). [5]

Section II – longer answers 30 points

1) Consider the discrete time signal $x[n]= \sin(2\pi f_1 n+\phi)+\cos(2\pi f_2 n- \phi)$, $n=1,\ldots,8$. [10]

a) What is the DFT of x[n] if ϕ=60 degrees, $f_1=1/8$ and $f_2=1/4$? Carefully plot the magnitude and phase being sure to label and quantify the ranges of the axes. [5]

b) If x[n] corresponded to a continuous-time signal x(t) sampled at 20kHz what frequencies (Hz) are present in the original signal x(t)? [3]

c) Is there spectral leakage in this example? Why or why not? [2]

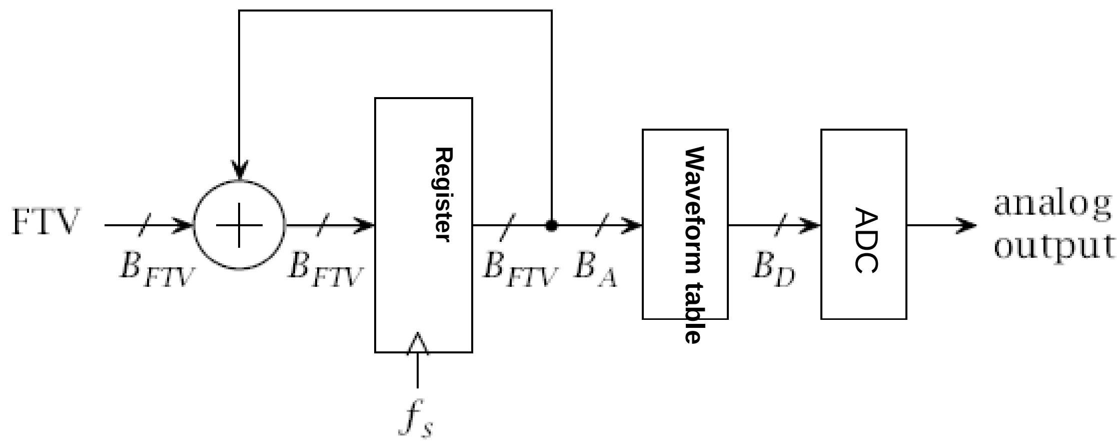
2) A 32 bit PIPO register is implemented using 32 D flip flops. Assuming that the average number of bit transitions per second are equal to half the clock frequency for all bits, what is the total power dissipation of this register and how does it depend on the logic voltage and the clock frequency? **[10]**

3) Using one complex FFT to compute two real FFTs

a) Let {x[n]} and {y[n]} be sets of N samples, i.e. n=0, …,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, …,N_x-1 and m=0, …, 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]**

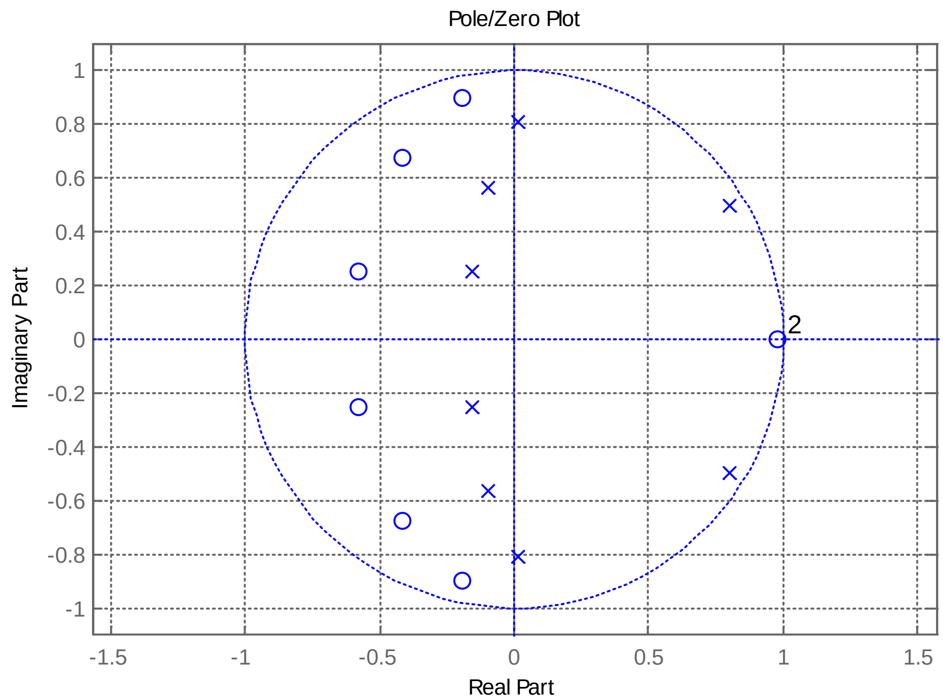
Section III Longer answers 30 points



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]=n 2^{-B} in the following.[10]
   a) Assume a clock rate fs = 10MHz, B_A = BFTV =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. [5]

   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. [5]

2) A pole zero constellation of a filter is shown below. [10]



a) It is desired to implement this filter using a cascade of biquad's as you did in Labs 5 and 6. How many biquads are required to implement it? [3]

b) Circle the pole-zero combinations that you would use for for each biquad in order to minimize the likelihood of overflow in your finite precision filter implementation. Label each combination 1,2,… corresponding to the order (from left to right) in which you would cascade the biquads. [4]

c) Would you need to normalize any of the denominator coefficients of any of the biquads in order to implement this filter in 16 bit Q15?. If so which biquad(s) would you need to normalize in this manner? [3]