EECS 452 *Midterm* Closed book part

Fall 2010

Name: ___________________________________ unique name: _______________

Sign the honor code:

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

____

Scores:

#	Points
Closed book	
Page 2	/12
Page 3	/12
Open book	**/76**
Total	**/100**

NOTES:

- There are **3** pages including this one.
- On the *closed book* section you may not have books, notes, or calculator. Nothing but a writing utensil.
- Don't spend too much time on any one problem.
- You have about 120 minutes for the exam. You probably shouldn't spend more than 15-20 minutes on this section.

Closed book portion

1. Say you took a 16-point DFT of the following signals with f_s=800Hz starting at time t=0. What would be what the non-zero DFT outputs would be (magnitude and phase)? **[3 points each]**
    a. 2*sin(2 π *200t+ π/2)

    b. 2*cos(2 π *150t- 1) - 2

2. Draw a block diagram which corresponds to the following transfer function. **[6 points]**

$$H(z) = \frac{\quad}{1+ \quad +}$$

3. _Fill in the blank or circle the **best** answer_. Provide all numbers in decimal. **[12 points, -1 per wrong or blank answer, minimum 0]**
   a. If we multiply an 8-bit *Q*7 value by an 8-bit *Q*7 value the result would be a ________-bit Q________ number.

   b. __________ is the smallest number (closest to negative infinity) that can be exactly represented by a signed 12-bit Q9 number. ____________________ is the largest number (closest to positive infinity) that can be exactly represented by a signed 6-bit Q2 number.

   c. In signed 5-bit two's complement Q3, what is the value of 11110? ____________________.

   d. Say you took an N-point DFT of a signal with a sample frequency of f_s. The frequency spacing of the DFT outputs would be __________________

   e. We say a filter is has *linear phase* if the group delay is ***a constant / proportional to the input frequency / zero / inside the unit circle***.

   f. All FIR filters with real-valued coefficients ***have linear phase / are stable / are low-pass***.

   g. A 10Hz cos wave has a period of ____________________ ms.

   h. Say you have an ideal A to D converter which converts values from 8V to -8V to a 4-bit value. In the ideal case, the worst-case quantization error is ±_________ Volts.

   i. FIR filters tend to require ***more / fewer / the same number of*** stages to implement a given filter when compared to an IIR filter.

   j. FIR filters ***can easily / cannot / can with great difficulty*** implement a typical RC analog filter.

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Scores:

#	Points
Open book	
Page 2	/8
Page 3	/7
Page 4	/11
Page 5	/12
Page 6	/9
Page 7	/12
Page 8	/6
Page 9	/11
Total	**/76**

NOTES:

- There are **9** pages including this one.
- On the *open book* section you may use calculators, books and notes, but no PDAs, Portables, Cell phones, etc.
- Don't spend too much time on any one problem.
- You have about 120 minutes for the entire exam.
- Some questions are much harder than others…

1) DFT/FFT

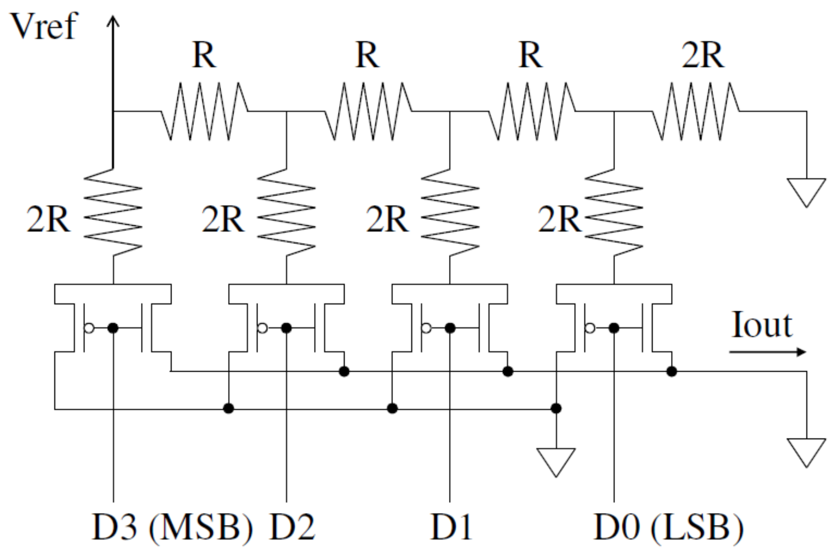
a. Say you have a 60-point DFT which is processing data sampled at 600Hz.

i. What frequency is associated with X(2)? ____________ **[1]**

ii. Why would it be difficult to use the FFT algorithm we discussed in class? Your answer must be 20 words or less. **[2]**

iii. Say the data passed into the DFT were from a 15Hz sinusoid and x(0)=0 while x(5)=1. Would X(0) be positive, negative or zero? You must *clearly* justify your answer to get any points. A picture might be useful… **[5]**

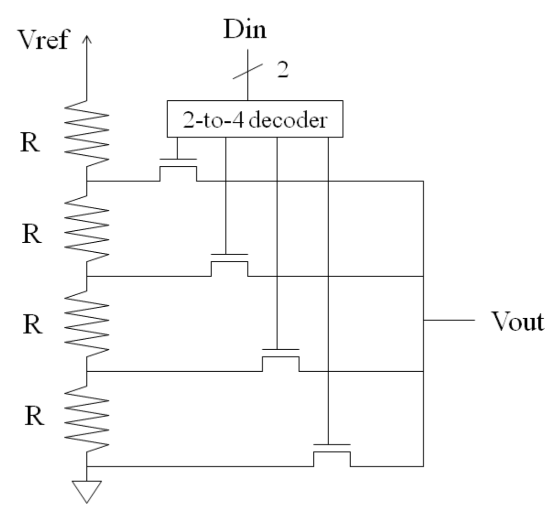
2) Answer the following questions about the ideal D2A converter shown



a. Assuming Vref=2V and R=4kΩ, if D[3:0]=0111, what is Iout? **[2]**

b. If we were going to make an 8-bit converter using this R/2R ladder scheme, how many transistors would we need? **[2]**

3) Consider the traditional voltage divider DAC (a version of which is seen on the right)



a. How many transistors would you need for an 8-bit converter using this scheme? **[1]**

b. For a 4-bit version of this DAC, what value would you expect to have at Vout if Vref=4V, R=4kΩ, and Din=0110? **[2]**

4. Consider the following DDS code. You may assume it is intended for a machine where an int is a 32-bit number.

```
#include<stdio.h>
int array[50]={
          0,      4106,      8148,     12062,     15785,
      19259,     22430,     25247,     27666,     29648,
      31163,     32186,     32702,     32702,     32186,
      31163,     29648,     27666,     25247,     22430,
      19259,     15785,     12062,      8148,      4106,
          0,     -4106,     -8148,    -12062,    -15785,
     -19259,    -22430,    -25247,    -27666,    -29648,
     -31163,    -32186,    -32702,    -32702,    -32186,
     -31163,    -29648,    -27666,    -25247,    -22430,
     -19259,    -15785,    -12062,     -8148,     -4106};

void main()
{
    int counter1=20,counter2=0;
    int total;
    while(1)
    {
        total=array[counter1/10]+array[counter2/10];
        counter1=(counter1+7)%500;
        counter2=(counter2+14)%500;
        printf("%d\n",total);
    }
}
```

a. Assuming “array” is intended to be an approximation of a unit amplitude sine function, what Q value is being used by “array”? **[2]**

b. What would be the *second* number printed by this program? What value does it correspond to given your answer to part “a”? Be sure to pay attention to how counter1 and counter2 are initialized. **[2]**

c. Assuming f_s=1MHz, what are the frequency components of the output? Provide your answer in terms of amplitude, phase and frequency for each component. **[7]**

5. Short answer.
    a. Explain, *<u>though words and a simple illustration</u>*, why having ‘linear phase’ is a desirable property for filters. **[3]**

    b. Provide an example of a filter application where a digital filter cannot be used. Explain why. **[3]**

    c. What would be the result of sampling an 8 kHz sine-wave at
        i. 4kHz? **[2]**

        ii. 6kHz? **[2]**

        iii. 10kHz? **[2]**

d. Draw a block diagram which corresponds to the following difference-equation. **[3]**

$$y[n]=2*x[n] + x[n-2] - y[n-3]$$

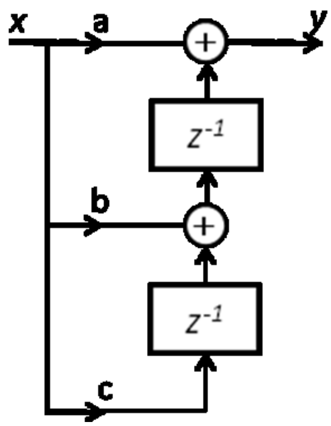
e. Consider the following difference equation

$$y[n] = x[n] + x[n-2] + 0.8\, y[n-1]$$

i. Write the transfer function that corresponds to that difference equation. **[3]**

ii. Is the corresponding filter stable? Justify your answer **[3]**

6. Write a function, `short trans (short x, short *w)` which takes a single input x, and returns the y value. Assume a=1.5, b=-.5 and c=1. You may also assume that the array w is of any fixed size you find useful. All inputs are in 16-bit Q14 and the output is also to be in 16-bit Q14. If the output is determined to be outside of the range of a 16-bit Q14 you should saturate. You are to write this for a 16-bit machine (that is, ints are 16 bits). **[12]**



7. It is known that the group delay of a given linear-phase filter is 0.02 seconds. Assuming the following frequencies are in the pass-band region, what would be the expected phase of each signal? Provide your answers in degrees in the range of 0 to -360 and show your work. **[6]**

   a) 10Hz

   b) 100Hz

   c) 75Hz

8. IIR biquad filters
    a. Why do we generally break down an IIR filter into biquad sections (2 zeros and 2 poles per section) rather than some other size? Specifically why not 1 zero and 1 pole per section? How about 4? **[4]**

    b. Explain how we group poles and zeros into biquad sections. Why do we group them in the way that we do? **[3]**

    c. Explain how we order the biquad sections. Why do we pick that ordering? **[2]**

    d. Why do we normalize the filter coefficients? **[2]**