EE 311 **Name__________________**
Hour Exam 3 **April 6, 2018**

1. The year is 2039 and you have been out of school for 20 years. A transporter portal broke down and you are stranded in Tobongobongo and the only place open is a small bar near the portal. You enter and you notice a very good looking robot tending bar so you sit down and place your dog eared copy of your DSP book on the bar and order a grape Nehi hoping to make a good impression. While serving your grape Nehi, the bartender takes notice of your book and mentions that *it* too took an intro course in DSP as an elective at bartender college. Further conversation reveals that it never really understood why IIR filters are more efficient than FIR filters and neither did it understand what filter efficiency was all about. You want to make a really good impression.

A) What is your explanation?

B) Where is Tobongobongo?

2. Suppose I have an ideal low pass filter which has a cutoff frequency of 1200 Hz and a high pass filter which has a cutoff frequency of 2000 Hz. In both cases the sample frequency is 44,100 Hz. If I put the two filters in series what will the output look like. Draw a rough sketch (not too rough) of the frequency vs. magnitude plot from 0 to fs/2.

3. A 10^{th} order Butterworth filter has a magnitude response of .98 at a frequency of 492 Hz. What is the cutoff frequency?

4. The code below is the main loop for a digital filter. Answer the questions related to this code.

```
while(1)
  {GPIOA_ODR |= (1 << 7);  //Set bit 7 to 1
   ADC1_CR2 |= 0x40000000; //Bit 30 does software start of A/D
   while((ADC1_SR & 0x2) == 0); //Bit 1 is End of Conversion
   uInt = ADC1_DR;
   u = ((float)(uInt & 0xFFF))/(float)4095.0;
   w = k1*u + k2*u1 + k3*u2 + k4*u3 + k5*u4 + k6*u5 + k7*u6;
   v = w – k8*v1 – k9*v2 – k10*v3 – k11*v4 – k12*v5 – k13*v6;
   vInt = (int)(2048*v);
   DAC_DHR12R1 = vInt & 0xFFF; //Converted number to D/A
   DAC_SWTRIGR |= 0x1;         //Start the D/A conversion
   u6 = u5;u5 = u4;u4 = u3;u3 = u2;u2 = u1;u1 = u;
   v6 = v5;v5 = v4;v4 = v3;v3 = v2;v2 = v1;v1 = v;
   //Wait for interrupt bit on TMR 0 = sample period
   //GPIOA_ODR &= ~(1 << 7);     //toggle bit 7
   while((TIM6_CR1 & 1) != 0); //Wait here until timer runs out
   TIM6_CR1 |= 1;              //Restart timer
  }
```

A) What is the filter order?

B) Write the transfer function in terms of the constants k1 to k13 and powers of z.

5. Suppose that h(k) = {1, 0.25, 0.125, 0.0625, 0.03125, …} = $(1/4)^k$. Using Pade's method what are the equations for the coefficients if M = N = 1. It's not necessary to solve the equations. Show all of your work.

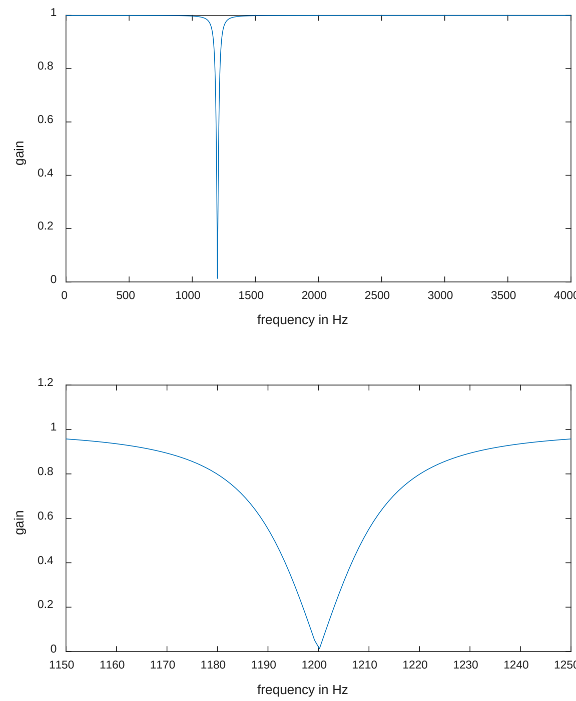
6. Suppose you have a second order low pass filter and you replace z by z^4. Explain what will happen to the frequency response.

7. An elliptic low pass filter has the following specifications:

sample frequency	44100 Hz
pass band edge	10000 Hz
stop band edge	13000 Hz
pass band ripple	0.005
stop band ripple	0.002

What is the minimum number of bits needed in the A/D and D/A converters? Show your work.

8. The diagram below is the magnitude plot of a digital filter which has a sample frequency of 8000Hz. The top figure shows the entire frequency band and the bottom figure shows a blowup of the frequencies from 1150 Hz to 1250 Hz. Find the transfer function.



9. If H(s) is: $H(s)=\dfrac{.002(s+1)}{(s+10)(s+50)}$ Answer the questions below:

A) Is this filter band limited. Justify your answer.

B) What is the gain of this filter at very low frequencies (d.c.). Show how you got your result.

C) If the BLT were used on this filter would the resulting digital filter be stable. Why or why not?

D) If the BLT is used on this filter how many zeros would there be at z = -1.

E) If the BLT is used on this filter what will be the order of the numerator and denominator? Justify your answer.