BITS, PILANI – DUBAI

International Academic City, Dubai

Year IV – Semester II 2007 – 2008

Comprehensive examination

Course No.: **EEE UC 415**

Course Title: **DSP**

Date: May 27, 2008 Time: 3hrs Max. Marks = 80

Any assumptions made should be indicated clearly)

A DSP system is described by the linear difference equation

$$y(n) = 0.2\,x(n) - 0.5\,x(n-2) + 0.4\,x(n-3).$$

Given that the digital input sequence {-1, 1, 0, -1} is applied to this DSP system, determine the corresponding digital output sequence

2. Determine the frequency response, magnitude response and phase response for the system given by $y(n) - (3/4)\,y(n-1) + (1/8)\,y(n-2) = x(n) - x(n-1)$ for T =1msec and $f = 0$ Hz, 10 Hz, 100 Hz and 1 kHz. Also plot the frequency response

3. Design a digital Chebyshev filter using Bilinear transformation and assuming T = 1 to satisfy the constraints

$$0.707 \le |H(\omega)| \le 1 \quad \text{for} \quad 0 \le \omega \le 0.2\pi$$

$$H(\omega)| \le 0.1 \quad \text{for} \quad 0.5\pi \le \omega \le \pi$$

4. A low pass FIR filter has the desired response as given below,

$$H_d(\omega) = e^{-j3\omega} \quad \text{for } 0 \le \omega \le \pi/2$$

$$H_d(\omega) = 0 \quad \text{for } \pi/2 \le \omega \le \pi$$

Determine the Type –I filter coefficients h(n) using frequency sampling techniques for N = 7

(Hint : $\omega_k = 2\pi k/N$) (10)

5. With a relevant example explain how multi rate digital signal processing is used in sub band coding processes (8)

6. What are the basic requirements of Adaptive Digital Filtering. Derive an expression for finding the optimum adaptive filter coefficients. What are the practical limitations of basic Weiner filter design? (10)

7. With a neat block diagram, explain the internal architecture of TMS 320 C5X PDSP. Also explain how does pipelining help to improve the performance of the processor. (10)

(P.T.O)

8. Given below the MATLAB codes program. Write the probable simulation result the program. Explain your results. What the function the program. (10)

Ass m [1 2] and h [2 4]

```
lc
lear ll
close ll
x= npu ( enter the 1st sequence
  nput( enter the 2nd sequenc )
y=conv(x h)
figure subplot(3 1 1)
stem x) ylabel( Ampl tude )
xlabel ( n )
subplot 1 2)
stem(h) ylabel( Ampl tude )
xlabel( (b) )
subplot( 1 3
stem(y) ylabel Ampl tude
xlabel( ( n )
disp( The esultant gnal y
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BITS PILANI – DUBAI

International Academic City, Dubai
Year IV – Semester II 2007– 2008
Test II (Open Book)

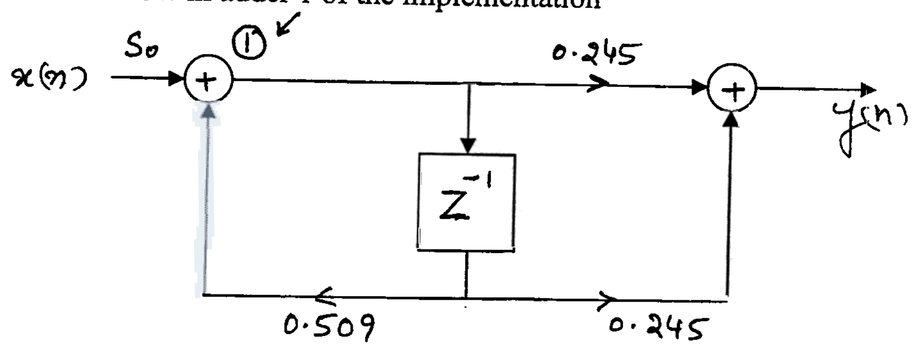
Course No.: **EEE UC 415** Course Title: **DSP**

Date: 04th May, 2008 Time: 50 Minutes Max. Marks = 30

(State clearly the assumptions made if any)

Consider the second order IIR system given by the transfer function $H(z) = 1/[(1-0.5z^{-1})(1-0.4z^{-1})]$. If the coefficient uses one sign bit and 3bit data representation, which of the following form of realisation will be less sensitive to the quantization error – Direct form I or cascade form? Prove your comment. (10)

2. The transfer ~~function of~~ of a low pass discrete time Butterworth filter whose cut off frequency is 1kHz and the sampling frequency is 10kHz is shown below. Find the scaling factor so as to avoid the overflow in adder 1 of the implementation (10)



3. Implement an "optimum" two stage decimator with an overall decimation factor of 100 and the following specifications
Pass band : 0 – 40 Hz; transition width : 10Hz Pass band ripple : 0.01
Stop band ripple: 0.002
Input sampling frequency of the filter is 20kHz. Also the draw the frequency response of the filters. (10)

The filter length is determined by $N = \dfrac{-20\log_{10}\sqrt{\delta_p\delta_s} - 13}{14.6\,\Delta f}$.

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BITS, PILANI – DUBAI

International Academic City, Dubai

Year IV – Semester II 2007 – 2008

Test I (Closed Book)

Course No.: **EEE UC 415** Course Title: **DSP**

Date: March 23, 2008 Time: 50min. Max. Marks = 30

(Any assumptions made should be indicated clearly)

1. Determine the attenuation at 10kHz achieved by Butterworth filters of first order when 3 dB cut-off frequencies are assumed to be 2kHz. (3)

2. Determine the transfer function H(z) of the digital filter whose difference equation is given below and hence draw the pole - zero diagram of the filter.
   $y(n) + (3/4)\, y(n-1) + (1/8)\, y(n-2) = x(n) + x(n-1)$ (6)

3. Convert the analog filter with system function $H(s) = (s+0.1)/[(s+0.1)^2 + 9]$ into a digital IIR filter using bilinear transformation. The digital filter should have a resonant frequency of $\omega_r = \pi/4$ { Hint: $\omega_c = (2/T)\tan(\omega/2)$ }
   Also draw the canonic form of realization of the resultant filter. (7)

4. Design a linear phase digital FIR filter with the following frequency response specifications and a stop band attenuation >= 38 dB, using window method, with N=7
   $H_d(\omega) = e^{-j3\omega}$ for $-\pi/4 \le \omega \le \pi/4$
   $H_d(\omega) = 0$ for $\pi/4 \le \omega \le \pi$
   { $h_d(n) = 1/2\pi \int_{-\pi}^{\pi} H_d(\omega) e^{jn\omega}\, d\omega$ } (7)

5. By pole-zero placement method, obtain the transfer function and the difference equation of a simple digital notch filter that meets the following specifications.
   Notch frequency : 30 Hz ; 3 dB width of notch : +/- 6 Hz
   Sampling frequency : 360 Hz (7)

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