BITS, PILANI – DUBAI CAMPUS

Knowledge Village, Dubai

Year IV – Semester I 2006– 2007

Comprehensive examination (Closed Book)

Course No.: EEE UC 415 Course Title: DSP

Date: December 28, 2006 Time: 3 hours Max. Marks = 40

(Clearly state any assumptions made)

1. Find the effect of coefficient quantisation on pole locations of the given second order IIR system, when it is realized in direct form I and in cascade form. Assume a word length of 4 bits through truncation.
   $H(z) = 1/[(1 - 0.5\,z^{-1})(1 - 0.45\,z^{-1})]$ (5M)

2. Consider an audio band signal with a nominal band width of 4 KHz that has been sampled at a rate of 8 KHz. It is required to down rate the sampling frequency to 200 Hz. The highest frequency of interest after decimation is 75 Hz. Design a suitable optimum two stage decimator which will satisfy the following overall specifications.

   Pass band ripple = 0.01; Stop band ripple = 0.0001

   Filter length $N = \dfrac{-10 \log(\delta_s \delta_p) - 13}{14.6\,\Delta f} + 1$;

   where Δf is the normalized frequency.

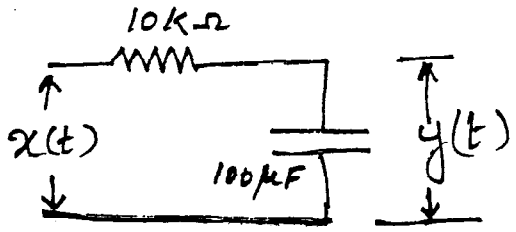
   Draw also the frequency response of the designed decimator stages. (6M)

3. Starting with the equation for the **mean square error**, derive the Wiener – Hopf equation to estimate the optimum weights of the adaptive filter. (5M)

4. A 4-point linear phase FIR filter is characterized by the following frequency samples. $|H(k)| = 1,\ k = 0$
   $= ½,\ k = 1, 3$
   $= 0,\ k = 2$
   a) Determine the transfer function of the Recursive FIR frequency sampling filter and show that it contains four zeroes and three poles.
   b) Sketch the pole -zero diagram
   c) Determine the four coefficients of the filter which could be implemented. (6M)

5. With a neat block diagram, explain the architecture of TMS 320 C5x P-DSP. (5M)

6. Using Bilinear - Z transform method of designing digital filters, determine the transfer function and hence the difference equation for the digital equivalent of the Resistance - Capacitance filter shown in figure below. Assume a sampling frequency of 150 Hz and a cut off frequency of 30 Hz.



(6M)

7. Obtain the coefficients of an FIR low pass digital filter to meet the specifications given below using window method.
   Pass-band edge frequency : 3.4 KHz; Stop band attenuation : 50 dB
   Transition width : 0.6 KHz
   Sampling frequency : 8 KHz
   Give your comments on the window used and the reason for your choice. (You may refer the windows table provided) (7 M)

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BITS, PILANI – DUBAI CAMPUS

Knowledge Village, Dubai
Year IV – Semester I 2006– 2007
Test I (Closed Book)

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

Date: November 02, 2006 Time: 50 Minutes Max. Marks = 30

(Answer all Questions)

1. A discrete time filter transfer function is given by
   $H(z) = (1 - 1.6\, z^{-1} + z^{-2}) / (1- 1.5\, z^{-1} + 0.8\, z^{-2})$

   a) What is its DC amplitude response?
   b) For the system transfer function given above, What are the radii of the poles and zeroes? (2 + 4)

2. Obtain the digital filter transfer function of the analog filter with $H(s) = (s+1) / (2s^2 + 5s + 2)$, using Impulse Invariant method. Assume a sampling frequency of 1kHz. (6)

3. An FIR digital filter has impulse response h(n) defined over the interval $0 \le n \le N-1$. if $N = 5$ and h (n) satisfies the symmetry condition $h(n) = h (N-n-1)$, Show that the filter has a linear phase characteristics. (6)

4. Describe the various addressing modes of Programmable DSP TMS 320 C5X (6)

5. Design a digital butterworth filter satisfying the constraints

   $$0.707 \le | H(e^{j\omega})| \le 1 \quad \text{for } 0 \le \omega \le \pi / 2$$

   $$| H(e^{j\omega})| \le 0.2 \quad \text{for } \tfrac{3}{4}\pi \le \omega \le \pi$$

   with T = 1 sec using bilinear transformation. Realise the filter using most convenient realization form. (6)

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BITS PILANI – DUBAI CAMPUS

Knowledge Village, Dubai
Year IV – Semester I 2006– 2007
Test II (Open Book)

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

Date: 07th December, 2006 Time: 50 min Max. Marks = 30

(State clearly the assumptions made if any)

1. Scale the first order digital filter structure of figure 1 using $\mathcal{L}_2$ – norm scaling rule (8)

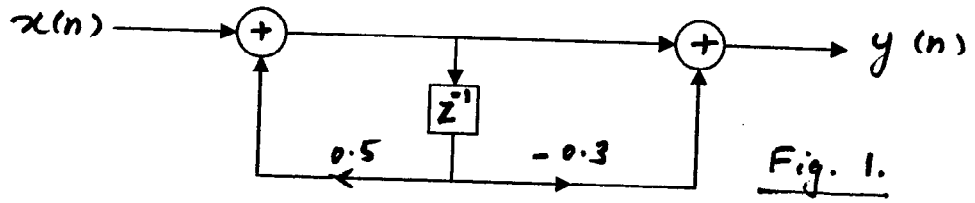

Fig. 1.

2. Design an efficient decimator for reduction in sampling rate of a signal from 12 kHz to 400 Hz. The highest frequency of interest of the signal is 180 Hz. Assume that an optimum linear phase FIR filter is to be used, with an overall pass band ripple of 0.01 and stop band deviation of 0.002. (Consider all possible no. of stages and decimation factors).

   Also draw the block diagram and the frequency response of each stage. (12)

3. Show that the impulse response of an ideal band pass filter is given by (8)

$$h_D(n) = \sin(n\omega_2)/n\pi - \sin(n\omega_1)/n\pi \quad \text{for } n \neq 0$$
$$= 2(f_2 - f_1) \quad \text{for } n = 0$$

x ⟶ x

BITS, PILANI – DUBAI CAMPUS

Knowledge Village, Dubai
Year IV – Semester I 2006– 2007
Quiz I (Open Book)

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

Date: October 16, 2006 Time: 30 Minutes Max. Marks = 40

(Question nos. **1- 20 carries one mark** each and **Question Nos 21- 30 carries 2 marks** each)

PART A

1. In modified Harward Architecture, for fetching the content of program and data memory, a separate bus is used for _____memory and a single bus is used for _____ memory.

2. VLIW architecture differs from conventional P-DSP in which of the following aspect?
a) Instruction Cache b) Number of functional Units
c) Use pipelining d) A single word fetch from memory using many instructions.

3. The addressing mode that is convenient for FFT computation is
a) indirect addressing b) circular mode
c) Bit reversed addressing d) Memory mapped

4. The _____ holds the result of multiplication and is _____ bit wide
a) Preg, 32 b) Treg, 32 c) Acc, 32 d) Preg, 16

5. _____ permits the contents of memory to be shifted by 0-16 bits before they are either fed to ALU or stored to memory.

6. The bit of ST1 used for comparing one register against another register memory is
a) SXM b) OV c) TC d) C

7. The pointers that are contained in the status register0 are _____
a) ARP b) DP c) ARB d) INTM e) IPTR

8. The data memory used with C5X processor is split into _____ pages each of _____ words long.

9. The memory mapped direct addressing mode is used to access data in page_____
a) 1 b) 0 c) 511 d) 512

10. Using RPT #k instruction, the maximum no. of times a single instruction can be repeatedly executed is _____

11. What addressing mode is represented by the syntax *0+

12. What is the syntax for long immediate addressing? Give an example?

13. The two special purpose memory mapped registers in the CPU are _____ and _____

14. Name the memory mapped registers that control the circular buffer operation.

15. The _____ bit _____ register enables and disables the circular buffer operation.

16. In the direct addressing mode of C5X _____ bits of the address is specified in the instruction and _____ bits are taken from the DP

17. With memory – mapped register addressing, the MMRs can be modified without affecting the current data page pointer value. **(True / False)**

18. The starting address of the data memory page pointed by the DP register whose content is 7 will be _____

19. When MACD instruction is executed, the product register is shifted by the count specified by the _____

20. For all the multiplication instructions of C5X, one of the operands is to be kept in _____ and the other can be specified by using one of the addressing modes.

BITS, PILANI – DUBAI CAMPUS

Knowledge Village, Dubai
Year IV – Semester I 2006– 2007
Test I (Closed Book)

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

Date: November 02, 2006 Time: 50 Minutes Max. Marks = 30

(Answer all Questions)

1. A discrete time filter transfer function is given by
   $H(z) = (1 - 1.6 z^{-1} + z^{-2}) / (1- 1.5 z^{-1} + 0.8 z^{-2})$
   a) What is its DC amplitude response?
   b) For the system transfer function given above, What are the radii of the poles and zeroes?
   (2 + 4)

2. Obtain the digital filter transfer function of the analog filter with $H(s) = (s+1) / (2s^2 + 5s + 2)$, using Impulse Invariant method. Assume a sampling frequency of 1kHz.
   (6)

3. An FIR digital filter has impulse response h(n) defined over the interval $0 \leq n \leq N-1$. if $N = 5$ and h (n) satisfies the symmetry condition $h(n) = h (N-n-1)$, Show that the filter has a linear phase characteristics. (6)

4. Describe the various addressing modes of Programmable DSP TMS 320 C5X
   (6)

5. Design a digital butterworth filter satisfying the constraints
   $$0.707 \leq | H(e^{j\omega})| \leq 1 \quad \text{for } 0 \leq \omega \leq \pi / 2$$
   $$| H(e^{j\omega})| \leq 0.2 \quad \text{for } \tfrac{3}{4}\pi \leq \omega \leq \pi$$
   with T = 1 sec using bilinear transformation. Realise the filter using most convenient realization form.
   (6)

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