BITS PILANI DUBAI CAMPUS

Dubai International Academic City, Dubai

Year IV – Semester I 2011 – 2012

Comprehensive Examination (Closed Book)

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

Date: January 11, 2012 Time: 3Hrs. Max. Marks = 40

(Any assumptions made should be indicated clearly)

1. Find the effect of coefficient quantisation in 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 6 bits through truncation.

   $\mathbf{H(z) = 1/ (1 - 0.7\, z^{-1} + 0.2\, z^{-2})}$ (6)

2. An FIR digital filter has impulse response h(n) defined over the interval $\mathbf{0 \le n \le N-1}$. Show that if $\mathbf{N=9}$ and h (n) satisfies the symmetry condition $\mathbf{h\,(n) = h\,(N-n-1)}$, the filter has a linear phase characteristics. (4)

3. a) Distinguish between the frequency response of Chebyshev type I filter for N is odd and even (2)

   b) 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: 50 Hz; 3 dB width of notch : +/- 5 Hz;

   Sampling frequency: 400 Hz (5)

4. A digital filter is characterized by the following transfer function.

   $H(z) = (1 + 1.5947\, z^{-1} + z^{-2}) / (1 - 0.6152\, z^{-1} + 0.2581\, z^{-2})$

   The filter is to be implemented using an 8bit system. Assuming that a second order canonic section is used to realize the filter, and each product has to be quantized to 8 bit,

a) Sketch the realization diagram showing the round of errors within the filter

b) Determine the $\mathcal{L}_2$- norm scaling factor for the system. (4+3)

5. Starting with steepest descent algorithm, $W_{k+1} = W_k - \mu \nabla_k$, derive the Widrow – Hopf LMS algorithm for adapting noise cancellation, stating any reasonable assumptions made. (3)

6. Write the MATLAB program to compute the coefficients and plot the frequency response of a band pass, linear phase FIR filter with the following characteristics. Pass band 1000-1500 Hz, transition band 500 Hz, Filter length 41, f_s = 10kHz (4)

7. With proper examples, explain the different modes of addressing available in **TMS 320 C 5X** Processors. (4)

8. Following are the register contents before execution of the instructions given below **(Assume the data missing if any)**

Write the content of the relevant registers which will get changed after execution of each of the instructions given below.

DP = 8   PM=1   CNF= 1   [410h] = 64h   [08F00H] = 04h

TREG0 = 22h   PREG = 01234567h   ACC = 76543210h   ARCR = 2530h

ARP = 2   AR2 = 2350h;   INDX = 10h;   [2350h] = 132h;   [50h] = 4680h

a. SAMM * 0 - , AR0

b. XOR # 05DB2 h, 2

c. MAC 08F00h,10h

d. ADD 10h, 2

e. CMPR 2 (1 x 5)

****

BITS PILANI DUBAI CAMPUS

DIAC, Dubai

Year IV – Semester I 2011– 2012

Test II (Open Book)

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

Date: December 18, 2011 Time: 50 Minutes Max. Marks = 30

(Clearly mention the assumptions made if any)

1. Design an ideal high pass FIR filter with a frequency response

$$H_d(e^{j\omega}) = 1 \quad \text{for} \quad \pi/4 \le |\omega| \le \pi$$

$$= 0 \quad \text{for} \quad |\omega| \le \pi/4$$

using hanning window.

a) Find the values of causal filter coefficient h(n) for N=11.

b) Write H(z)

c) Find $H(e^{j\omega})$ **(6+3+3)**

2. Consider an audio band signal with a nominal band width of 4 KHz that has been sampled at a rate of 10 KHz. It is required to down rate the sampling frequency to 250 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 = 0.001

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. **(10)**

3. The transfer function of a digital filter is given by $H(z) = \dfrac{5z\,(3z-2)}{(z+0.5)(2z-1)}$

Determine the values of the multiplier coefficients of the realization structure in direct form II. Also draw the realization. **(8)**

****

BITS PILANI DUBAI CAMPUS

DIAC, Dubai

Year IV – Semester I 2011– 2012

Test I (Closed Book)

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

Date: October 23, 2011 Time: 50 Minutes Max. Marks = 30

---

1. The frequency response specification for a band-pass discrete-time filter in normalized form is as follows.

   Pass-band : $0.44\pi - 0.65\pi$;

   stop-bands: $0 - 0.32\pi$ and $0.75\pi - \pi$.

   Sampling interval T = 100us (2x4)

   a) Express the specifications in rad/s (de-normalised)
   b) Convert the specs from rad/s to standard units of Hz
   c) Convert the specs to normalized frequency form
   d) Sketch the frequency response for (b) in the interval from 0 to sampling frequency.

2. The transfer function of a DT system has poles at z = 0.45, z = 0.12 $\pm$ j 0.25 and zeroes at z = -1 and at z = 1.
   a) Derive the system transfer function H(z).
   b) Write the difference equation for y(n)
   c) Draw the canonic form of realization of the system. (3x3)

3. Design a digital Butterworth filter satisfying the constraints

   $$0.707 \leq |H(e^{j\omega})| \leq 1 \quad \text{for } 0 \leq \omega \leq \pi/2$$ (7)

   $|H(e^{j\omega})| \leq 0.25$ for $\frac{3}{4}\pi \leq \omega \leq \pi$ with T = 1 sec using bilinear transformation.

4. Determine the frequency response of an IIR filter characterized by

   $$y(n) = 0.3x(n) - x(n-1) + 0.5x(n-2) - y(n-1)$$

   Compute the phase delay and group delay of the filter (6)

****

BITS PILANI DUBAI CAMPUS

DIAC, Dubai

Year IV – Semester I 2011– 2012

Quiz II

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

Date: December 05, 2011 Time: 20 Minutes Max. Marks = 10

(Question nos. **1- 4 carries one mark** each and **Question Nos 5- 7 carries 2 marks** each)

PART A

1. The data memory used with C5X processors is split into ____________ pages each of ________ words long.

   a) 512, 128 b) 256,256 c) 128,512 d) 1024,64

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 status register bit that determines whether multiplier's 32-bit product is left shifted by 0,1,4 or right shifted by 6 with sign extension before it is transferred / added to the ACC is ____________

   a) CNF b) PM c) HM d) XF e) INTM

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

PART B

(Assume the data missing if any)

Following are the register contents before execution of the instructions given below.
Write the content of the relevant registers which will get changed after execution for each of the instructions given below.

DP = 6 [300h] = 04h; [310h] = 24h; [08F00H] = 03h

TREG0 = 22h ACC = 76543210h ARCR = 2530h

ARP = 7 AR7 = 2350h INDX = 10h [2350h] = 456h [50h] = 4680h

[2360] = 123h AR0 = 2900h

5. SAMM * 0 + , AR0

6. XOR # 05DB2 h, 2

7. ADD 10h, 2

********** *GOOD LUCK* ***********

Master Copy

BITS PILANI DUBAI CAMPUS

Dubai International Academic City
Year IV – Semester I 2011– 2012
Quiz I (Closed Book)

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

Date: September 26, 2011 Time: 20 Minutes Max. Marks = 10

(Question nos. **1- 8 carries** ~~**one**~~ half **mark** each
Question Nos **9- 11 carries two marks** each)

1. An Analog band pass signal is to be sampled in accordance with the band pass sampling theorem. Assuming the signal has the frequency band of 6 kHz – 26kHz , what should be the minimum theoretical sampling frequency to avoid aliasing?

2. An analog signal is given by $f(t) = 1.75 + 2\cos 400\Pi t - \cos 2000\Pi t$. What is its fundamental frequency and dc amplitude?

3. The probable speech signal frequency range is between
a) 20Hz – 3400Hz b) 400Hz- 500Hz c) 20Hz – 20 KHz d) 3400 Hz – 3500Hz

4. Which analog filter is suitable for Audio applications?
a) Butterworth type b) Chebyshev Type I c) Bessel d) Elliptic

5. Which filter has ripples in stop band?
a) Butterworth b) Chebyshev (Type-I) c) Chebyshev (Type-II) d) Bessel

6. Which analog filter has maximum order for a given set of specification?
a) Butterworth type b) Chebyshev Type I c) Bessel d) Elliptic

7. Draw the pole zero diagram of the transfer function $\mathbf{H(s) = (s+2) / (2s^2 + 3s + 2)}$

8. A digital Filter is said to be FIR if
   a.) All its pole lie outside the unit circle
   b.) One or more denominator coefficients is non zero
   c.) Current output depends on previous inputs
   d.) Current output depends on previous inputs and outputs

9. Determine the impulse response of an analog low pass filter whose transfer function is given by $H(s) = 1/\,\mathbf{(s^2 + s - 2)}$
10. Determine whether the system whose transfer function is given as $\mathbf{H(s) = (s+.11) / s(s^2 + s + 2)}$ is stable
11. Determine the group delay of the function $1/\,(1+2s)$ at 1rad/sec