Register Number.........................

B.TECH. DEGREE EXAMINATIONS: NOVEMBER 2009

Fifth Semester

INFORMATIOIN TECHNOLOGY

U07IT501: Digital Signal Processing

Time: Three Hours **Maximum Marks: 100**

Answer ALL the Questions:-

PART A (10 × 1 = 10 Marks)

1. The energy of the signal x(n) = u(n) is
   a. 1 b.0 c. Infinity d. Constant

2. Aliasing is caused when a signal is sampled
   a. =2fm b.>2fm c.<2fm d.>=2fm

3. The value of W_8^{8} is equal to
   a. 8 b.1 c.64 d.16

4. In radix – 2 FFT, the number of complex addition required for calculating 1024 – point DFT is
   a. 2560 b. 5120 c. 10240 d. 1024

5. The poles of Chebychev filter lies on the
   a. A Circle in the s-plane. b.An ellipse in the s-plane
   c. A circle in the t- plane d.An ellipse in the t – plane.

6. The stop band frequency of a digital butterworth filter is 3pi / 4. Assume T = 1s. What is the stop band frequency after prewarping?
   a. 3.14 b.4.828 c.8.424 d. 0.75

7. The phase response of the FIR filter is
   a. Linear. b. Non-linear c. Constant d. Irregular

8. The type of Ideal digital Hilbert transformer is
   a. Low pass filter. b. All reject filter
   c. All pass filter d. High pass filter.

9. The quantization errors due to finite wordlength registers in digital filters are
   i. Quantisation error in A / D conversion
   ii. Sampling error.
   iii. Co-efficient error
   a. A, B & C b. A only c. B only d. A & C only

10. Decimator is used to
    a. Increase the sampling rate. b. Decrease the sampling rate.
    b. Multiply the sampling rate. d. Divide the sampling rate.

PART B (10 x 2 = 20 Marks)

11. State the sampling theorem.
12. Calculate the average power of $x(n) = u(n)$
13. Draw the basic structure of DIT & DIF – FFT of radix – 2.
14. Differentiate circular and linear convolution.
15. Compare Butterworth and Chebychev filters.
16. Write bilinear transformation?
17. State Gibb's phenomenon?
18. Define phase delay.
19. Define truncation error for sign magnitude representation.
20. Outline limit cycle oscillation?

PART C (5 x 14 = 70 Marks)

21(a) i) Determine the Z- transform and compute the ROC of the following signal

ii) State and explain the scaling and time delay properties of Z – transform.

(OR)

(b) Determine the Zero – input response of the system described by the second difference equation $y(n) - 5y(n-1) + 4y(n-2) = 0$ with the initial conditions y(9-

and $y(-2) = 0$.

22 (a) Compute the DFT using DIT – FFT and DIF – FFT algorith

$x(n) = \{1, 2, 1, 2, 1, 2, 1, 2\}$

(OR)

(b) Using radix – 2 FFT DIT algorithm determine the 8 – point ID

$x(n) = \{7,1,-1,1,-1,1,-1,7\}$.

23 (a) Design an analog Chebychev filter that has a 2 db pass band ripple of 1 db at the frequency 0.628 rad / s and a stop band attenuation of 15 db at 1.256 rad / s.

(OR)

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(b) Design a digital IIR filter from the analog filter transfer function H(s) = 1 / (s+1) (s+2) by using bilinear transformation. Assume T = 1s.

24 (a) Design a LPF with a frequency response

$$Hd(\omega) = \begin{cases} 1, & \dfrac{-\pi}{4} \leq \omega \geq \dfrac{\pi}{4} \\ 0, & \dfrac{\pi}{4} \leq \omega \geq \pi \end{cases}$$

(OR)

(b) Design a 9 tap Hilbert transformer using rectangular window. Also find the magnitude frequency response.

25 (a) (i) Give the round – off error for fixed point representation. (2)

(ii) Propose the round – off effects in digital filters. (12)

(OR)

(b) (i) Generalize channel vocoder. (2)

(ii) Develop the sub – band coding of speech signals. (12)

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