Reg. No. : ....

Name : ....

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6658

Fifth Semester B.Tech. Degree Examination, November 2013
(2008 Scheme)
08.502 : DIGITAL SIGNAL PROCESSING (TA)

Time : 3 Hours

Max. Marks : 100

PART – A

Answer **all** questions. **Each** question carries **4** marks.

M1 1. Explain how DFT is useful in linear filtering.

M1 2. What is meant by leakage ? Explain.

3. For a system defined by $y(n) = 0.25\,x(n) + x(n-1) + 0.25\,x(n-2)$, evaluate phase delay and group delay.

M2 4. What properties of the phase factor W_N are exploited in FFT algorithms to reduce the number of computations ?

M3 5. Explain FIR filters using window method.

M3,4,5 6. Compare the characteristics of FIR and IIR filters.

7. Prove that ideal high pass filter is not physically realizable.

M6 8. Discuss round off errors.

9. Explain sub band coding.

10. List the important features of a DSP architecture. **(4×10=40 Marks)**

M5

P.T.O.

PART – B

Answer **any two full** questions from **each** Module.

MODULE – I

11. Let $x_a(t)$ be an analog signal with a bandwidth of 3 KHz. We wish to use N point DFT to compute the spectrum of the signal with a resolution less than or equal to 50 Hz. Determine a) the minimum sampling rate b) the minimum number of required samples, and c) the minimum length of analog signal record. **(3+4+3)**

M1

12. Let $x_1(n)$ and $x_2(n)$ be two finite length sequences of length 5 each. If $y_1(n)$ and $y_2(n)$ denote the linear and 5-point circular convolutions of $x_1(n)$ and $x_2(n)$ respectively, express $y_2(n)$ in terms of $y_1(n)$. **10**

M2

13. Draw 8-pt DIF-IDFT (FFT) algorithm. Use the algorithm to evaluate the inverse of

$$X(K) = \begin{Bmatrix} 20, -5.828 - j\,2.414, 0, -0.171 - j\,0.41 \\ 0, -0.171 + j\,0.41, 0, -5.828 + j\,2.414 \end{Bmatrix}.$$ **10**

MODULE – II

M5

14. Obtain the direct form 2 and transposed direct form 2 of a system described by the difference equation

$$y(n) = \frac{1}{2}y\,(n-1) - \frac{1}{4}y\,(n-2) + x(n) + x(n-1)\,.$$ **(4+6)**

M4

15. Design an IIR filter using bilinear transformation. Specifications are – 3dB at 500 Hz

Monotonic passband and stopband response.

Attenuation of 15 dB at 750 Hz

Sampling rate 2000/sec. **10**

M3

16. Design an FIR filter with the following specifications :

$$H_d(e^{jw}) = e^{-j\omega\alpha}; \quad 0 \le |\omega| \le 1 \;\&\; 2 \le |\omega| \le \pi$$
$$= 0 \quad ; \text{elsewhere}$$

Choose N = 7 and Hann window for the design. **10**

MODULE – III

m5

17. Draw the block diagram of TMS320C6713 DSP processor and explain its architecture. **10**

m6

18. a) Explain the effect of quantization of filter coefficients on a digital filter. **6**

b) Write short note on transmultiplexers. **4**

m6

19. a) Discuss the concept of multirate signal processing.

b) A CD player operates with a sampling rate of 44.1 KHz while DAI drives have a sampling rate of 48 KHz. Draw a block diagram for performing this sampling rate conversion. Obtain the conversion factors and cut-off frequency of the filter employed. **(5+5)**

(10×6=60 Marks)