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Reg. No. : ....

Name : ....

Fifth Semester B.Tech. Degree Examination, September 2014
(2008 Scheme)
(Special Supplementary)
08.502 : DIGITAL SIGNAL PROCESSING (TA)

Time : 3 Hours Max. Marks : 100

PART – A

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

M2 1. Determine the 4 point DFT of the following pair of sequences in the most efficient method x[n] = [1, 0, 1, 1] y[n] = [1, 2, 1, 2].

M1 2. Consider the length 12 sequence defined for $0 \le n \le 11$,

x[n] = [1, 2, 3, 1, 2, 3, 1, 2, 3, 1, 2, 3]

Evaluate the following functions without computing DFT.

i) X(0)

ii) $\sum_{k=0}^{11} X(k)$

3. If the complex multiplication time of a particular computer is 0.5 μs and the addition time is much, much smaller, roughly how long will it take to compute a 1024 point DFT using FFT. Assume multiplications are done sequentially.

M1 4. Suppose x[n] is a sequence defined as x[n] = [0, 1, 2, 3, 5, 6, 7].

a) Illustrate x[(n – 2) mod 8].

b) If the 8 point DFT of x[n] is X(k), what is the 8 point DFT of x[(n – 2)mod 8] ?

P.T.O.

5. What is Gibb's phenomenon ? How can its effect be reduced ?

M4

6. Is impulse invariant method inappropriate for designing high pass filter ? Why or why not ?

M5

7. A filter is specified by its impulse response h[n] given by

$$h[n] = -\frac{1}{3}8[n+1] + \frac{1}{2}8(n) - \frac{1}{3}8[n-1]$$

Is the filter IIR or FIR ? Is it physically realizable ? Justify your answer.

M6

8. Draw the block diagram of the system to convert sampling rate from 1 kHz to 1.5 kHz.

9. What is sub band coding ?

M6

10. Compare floating point arithmetic with fixed point arithmetic. **(10×4=40 Marks)**

PART – B

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

MODULE – I **(10×2=20 Marks)**

M1

11. The impulse response of a filter is given as h[n] = [1, 2, – 1]. Find the output if the input sequence is x[n] = [2, 6], using DFT. Check the answer by finding linear convoluter. 10

M1,2

12. a) Draw the signal flow graph of 8 point radix-2 DIT FFT algorithm. 6

b) Determine the DFT of the signal x[n] = [1, 1, – 1, –1, 1]. 4

M1

13. a) Determine the 4 point DFT of the following pair of sequences in the most efficient method.

x[n] = [1, 2, 1, 2] g[n] = [1, 0, 0, 1]. 3

b) How do we perform linear filtering of long sequences using DFTs ? 4

c) Define Discrete Cosine Transform. 3

MODULE – II

(10×2=20 Marks)

M4 14. The specification of a filter is given below $0.8 \le |H(\omega)| \le 1.0 \quad 0 \le \omega \le 0.2\pi$

$|H(\omega)| \le 0.2 \quad 0.6\pi \le \omega \le \pi$

Design a Butterworth digital filter satisfying specifications. **10**

M5 15. a) What is the condition to be satisfied for linear phase characteristics of a FIR filter ? **2**

b) Why ideal filter is not physically realizable ? **2**

c) Compare FIR filter and IIR filter. **3**

d) Write the frequency response of any two window functions used in FIR filter design. **3**

16. a) Realize the following transfer function with minimum number of multipliers.

$$H(z) = \frac{1}{4} + \frac{1}{2}z^{-1} + \frac{3}{4}z^{-2} + \frac{1}{2}z^{-3} + \frac{1}{4}z^{-4}.$$ **5**

M5 b) Realize an FIR filter with impulse response h[n] given by

$$h[n] = \left(\frac{1}{2}\right)^{n}[u[n] - u[n-4]].$$ **5**

MODULE – III

(10×2=20 Marks)

M6 17. a) What are limit cycles ? **2**

b) What is the effect of quantization of filter coefficients in the performance of a digital filter ? **4**

c) What are the errors generated by A/D process ? **4**

18. a) Derive the frequency domain representation of an upsampled signal. **5**

M6 b) For the system shown, find the expression for y[n] in terms of x[n]. **5**

$$x[n] \rightarrow \boxed{\uparrow 3} \rightarrow \boxed{\downarrow 2} \rightarrow \boxed{\downarrow 3} \rightarrow \boxed{\uparrow 2} \rightarrow y[n]$$

M5 19. a) Discuss the basic features of a digital signal processing chip. **5**

b) What are the programming tools required for DSP processors. **5**

(6×10=60 Marks)
