



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 $0 \leq n \leq 11$,
 $x[n] = [1, 2, 3, 1, 2, 3, 1, 2, 3, 1, 2, 3]$
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 $0.5 \mu s$ 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 $x[(n - 2) \bmod 8]$.
b) If the 8 point DFT of $x[n]$ is $X(k)$, what is the 8 point DFT of $x[(n - 2) \bmod 8]$?

P.T.O.



5. What is Gibb's phenomenon ? How can its effect be reduced ?
6. Is impulse invariant method inappropriate for designing high pass filter ? Why or why not ?
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.

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 ?

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)

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.

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

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- b) Determine the DFT of the signal $x[n] = [1, 1, -1, -1, 1]$.

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

$$x[n] = [1, 2, 1, 2] \quad g[n] = [1, 0, 0, 1]$$

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

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- c) Define Discrete Cosine Transform.

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MODULE – II

(10×2=20 Marks)

14. The specification of a filter is given below
- | | |
|---------------------------------|-------------------------------|
| $0.8 \leq H(\omega) \leq 1.0$ | $0 \leq \omega \leq 0.2\pi$ |
| $ H(\omega) \leq 0.2$ | $0.6\pi \leq \omega \leq \pi$ |

Design a Butterworth digital filter satisfying specifications.

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15. a) What is the condition to be satisfied for linear phase characteristics of a FIR filter ?
 b) Why ideal filter is not physically realizable ?
 c) Compare FIR filter and IIR filter.
 d) Write the frequency response of any two window functions used in FIR filter design.

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

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

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

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

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MODULE – III

(10×2=20 Marks)

17. a) What are limit cycles ?
 b) What is the effect of quantization of filter coefficients in the performance of a digital filter ?
 c) What are the errors generated by A/D process ?
 18. a) Derive the frequency domain representation of an upsampled signal.
 b) For the system shown, find the expression for $y[n]$ in terms of $x[n]$.

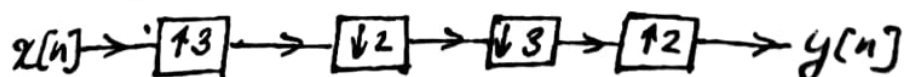
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19. a) Discuss the basic features of a digital signal processing chip.
 b) What are the programming tools required for DSP processors.

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(6×10=60 Marks)