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**Reg. No. :**

**Question Paper Code: 46041**

B.E. / B.Tech. DEGREE EXAMINATION, NOV 2017

Sixth Semester

Electronics and Communication Engineering

14UEC601 – DIGITAL SIGNAL PROCESSING

(Regulation 2014)

Duration: Three hours Maximum: 100 Marks

Answer ALL Questions

PART A - (10 x 1 = 10 Marks)

1. How many numbers of multipliers and adders are required for a 5 point DFT?

 (a) 20,25 (b) 25,20 (c) 11.6,5.8 (d) 5.8,11.6

2. DFT of δ(n) is

 (a) 1 (b) 0 (c) ∞ (d) -1

3. In impulse invariant method , relationship between ω and Ω is given by,

 (a)  (b) 

 (c)  (d) 

4. Magnitude function of Chebyshev filter is given by,

 (a) 

 (b) 

 (c)

 (d) 

5. Substitution of values for names whose values are constant, is done in

 (a) Is a Recursive (b) Use less memory

 (c) Is Unstable (d) Has linear phase response

6. What are the parameters predict the performance of a window in FIR

 (a) Main lobe , Bandwidth (b) Relative side lobe , Main lobe

 (c) Bandwidth , Relative side lobe (d) Bandwidth, Relative side lobe , Main lobe

7. Non-linear distortion can be limited by using

 (a) Rounding (b) Truncation

 (c) 2’s complement (d) Signal scaling

8. Sign magnitude representation of -7/8 is

 (a) 1.001 (b) 1.111 (c) 1.100 (d) 0.111

9. Speech coding is concerned with the development of techniques which exploit the

 \_\_\_\_\_\_\_\_in the speech signal.

 (a) Up sampling (b) Down sampling (c) Redundancy (d) Quantization

10. In subband coding, the input signal is first split into number of non-overlapping

 frequency by

 (a) Low pass filter (b) High pass filter

 (c) Band pass filter (d) Band stop filter

 PART - B (5 x 2 = 10 Marks)

11. Find the IDFT of Y(k)={1,0,1,0}.

12. Mention any two procedures for digitizing the transfer function of an analog filter.

13. What is the reason that FIR filter is always stable?

14. What are the advantages of floating point point arthimatic?

15. List out the applications of Multirate signal processing.

PART - C (5 x 16 = 80 Marks)

16. (a) Consider the finite duration sequence x (n) = {1, 2, 3, 4,-5, 6, 7, 8} Compute the

 eight point DFT using the in-place Radix-2 decimation in time algorithm of the

 sequence. (16)

 Or

 (b) Find the FIR system response for a given input signal

 x(n) ={3,-1,0,1,3,2 ,0,1,2 ,1} and impulse h(n) = {1,1,1) by using fast convolution

 techniques. Also justify the response of the system.

17. (a) Design a digital chebyshev filter that satisfying the following frequency response

 with T=1 sec using impulse Invariant

 Transformation technique (16)

Or

 (b) Design a Digital Butterworth high pass filter with a minimum passband attenuation

 of 2.5dB at Ωp=20rad/sec and the stopband attenuation of 30 dB at Ωs=50rad/sec

 use bilinear transformation with the sampling time of 1 sec. (16)

18. (a) Design a FIR Linear phase, Digital filter approximating the ideal high-pass filter

 with a frequency response 

 1.Determine the co-efficient of 11 tap filter based on the window method Hanning.

 2.Determine and plot the magnitude and phase response of the filter. (16)

Or

(b) Design a Lowpass FIR filter using Frequency sampling technique having cutoff freq

 of π/2 rad / sample. The filter should have linear phase and length of 17. (16)

19. (a) A non-recursive system H (z) is designed such a way that, two Linear phase systems

 and are connected in cascade. Which are given as 

 and. Find the output round off noise power? Assume a1 =0.5

 and a2 =0.6. (16)

 .

Or

 (b) Analyze the characteristics of a limit cycle oscillation with respect to the first order

 recursive system described by the difference equation y (n) = 0.95y (n-1) + x (n).

 Determine the dead band of the filter. Assume 4-bit sign magnitude representation

 (excluding sign bit) and the input is given by, (16) 

20. (a) Explain in detail about two basic operations in Multirate Signal Processing. (16)

Or

 (b) (i) Draw and explain the block diagram of sub band coding system. (8)

 (ii) Discuss about the Musical Sound Processing. (8)