Reg. No. : | | | | | | | | | | |

Question Paper Code: 46401

B.E. / B.Tech. DEGREE EXAMINATION, APRIL 2019

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. The convolution by FFT is called

   (a) linear convolution (b) circular convolution

   (c) fast convolution (d) slow convolution

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) $\Omega = \frac{2}{T_s}\tan(\frac{\omega}{2})$ (b) $\omega = \frac{\Omega}{T_s}$

   (c) $\Omega = \frac{1}{T_s}\tan(\frac{\omega}{2})$ (d) $\omega = \Omega T_s$

4. If N_B and N_C are the orders of the Butterworth and Chebyshev filters respectively to meet the same frequency specifications, then which of the following relation is true?

   (a) $N_C=N_B$ (b) $N_C<N_B$ (c) $N_C>N_B$ (d) Cannot be determined

5. Which region of the frequency specification has to be optimized to reduce side lobes of the FIR filter?

(a) Stop band (b) Pass band
(c) Transition band (d) None of these

6. The values of cutoff frequencies in general depend on

(a) Type of the window (b) Length of the window
(c) Neither (a) nor (b) (d) Both (a) and (b)

7. Calculate the improvement of signal to quantization noise ratio with an increase of 2 bits to existing bits.

(a) 2dB (b) 6dB (c) 4dB (d) 12dB

8. Which of the following is not a quantization error occuring in digital systems?

(a) Input quantization error (b) Product quantization error
(c) Coefficient quantization error (d) Output quantization error

9. Which of the following is the disadvantage of sampling rate conversion by converting the signal into analog signal?

(a) Signal distortion
(b) Quantization effects
(c) New sampling rate can be arbitrarily selected
(d) Both (a) and (b)

10. What value should the bandwidth of x(n) has to be reduced in order to avoid aliasing?

(a) F/D (b) F/2D (c) F/4D (d) none of these

PART - B (5 x 2 = 10 Marks)

11. What are the differences and similarities between DIF and DIT algorithms?

12. What is pre-warping?

13. Write the equation of Hamming and Blackman window functions.

14. Define zero input limit cycle oscillations

15. Give the steps in multistage sampling rate converter design.

PART - C (5 x 16 = 80 Marks)

16. (a) Perform circular convolution of the following sequence. X(n)={-1,1,2,-1,1,2 } and

h(n)={2,1,-2}. (16)

Or

(b) Perform Linear convolution of the following sequence by using overlap save and over lap add method. X(n)={1,1,2,1,2,1,-1,-1} and h(n)={2,1}. (16)

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

$$0.707 \leq |H(e^{j\omega})| \leq 1 \qquad for \quad 0 \leq \omega \leq \frac{\pi}{2}$$

$$|H(e^{j\omega})| \leq 0.2 \qquad for \quad \frac{3\pi}{4} \leq \omega \leq \pi$$

with T=1 sec using impulse Invariant Transformation technique (16)

Or

(b) Design a digital Butterworth filter using impulse invariance method satisfying the constraints. Assume T = 1s.

$$0.8 \leq |H(w)| \leq 1; \quad 0 \leq w \leq 0.2\pi$$
$$|H(w)| \leq 0.2; \quad 0.6\,\pi \leq w \leq \pi$$ (16)

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

with a frequency response $H_d(e^{j\omega}) = \begin{cases} 1 & for\ \frac{\pi}{4} \leq |\omega| \leq \pi \\ 0 & |\omega| < \frac{\pi}{4} \end{cases}$

(i) Determine the co-efficient of 11 tap filter based on the window method Hanning.
(ii) Determine and plot the magnitude and phase response of the filter. (16)

Or

(b) Design a LP FIR filter using Frequency sampling technique having cutoff freq of $\pi/2$ 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 $H_1(z)$ and $H_2(z)$ are connected in cascade. Which are given as $H_1(z) = \frac{1}{1 - a_1 z^{-1}}$ and $H_2(z) = \frac{1}{1 - a_2 z^{-1}}$. Find the output round off noise power? Assume $a_1 = 0.5$ and $a_2 = 0.6$. (16)

Or

(b) (i) What is quantization of analog signals? Derive the expression for the quantization error. (8)

(ii) Summarize the addressing modes of Digital Signal Processor TMS320C5X. (8)

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

Or

(b) (i) Explain the multistage implementation of sampling rate conversion with a block diagram. (8)

(ii) A signal x(n) is given by x(n) = {0, 1, 2, 3, 4, 5, 6, 0, 1, 2, 3…}. Obtain the decimated signal with a factor of 2 and the interpolated signal with a factor of 2. (8)