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

Question Paper Code: 36401

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

Sixth Semester

Electronics and Communication Engineering

01UEC601 - DIGITAL SIGNAL PROCESSING

(Regulation 2013)

Duration: Three hours Maximum: 100 Marks

Answer ALL Questions

PART A - (10 x 2 = 20 Marks)

1. Why Fast Fourier transform is needed?

2. How many multiplications and additions are required to compute 64-point DFT using radix-2 FFT?

3. Sketch the mapping of s-plane to Z-plane in bilinear transformation.

4. Calculate the poles and normalized transfer function of Low pass Butterworth filter for the order N=1.

5. What are the advantages and disadvantages of FIR filters?

6. Define Gibb's phenomenon.

7. Distinguish the fixed point and floating point arithmetic.

8. Define product quantisation error.

9. Draw the block diagram of sub coding.

10. Define interpolation and decimation.

PART - B (5 x 16 = 80 Marks)

11. (a) (i) Compute the Eight point DFT of the sequence

$$x(n) = \{0.5, 0.5, 0.5, 0.5, 0, 0, 0, 0\}$$

using the in-place radix-2 DIT FFT algorithm. (10)

(ii) Compare the DIT and DIF radix-2 FFT. (6)

Or

(b) Perform circular convolution for the sequence $x_1(n)=\{1, 1, 2, 1\}$ and $x_2(n)=\{4, 3, 2, 1\}$ using DFT and IDFT. Justify the result by computing in time domain. (16)

12. (a) The specifications of the desired low pass filter is

$$0.7 \le |H(e^{jw})| \le 1; \quad 0 \le \omega \le \pi/2$$

$$|H(e^{jw})| \le 0.2 \quad ; \quad 3\pi/4 \le \omega \le \pi$$

Design a digital butter worth filter using bilinear transformation. Assume T=1sec. (16)

Or

(b) For the analog transfer function $H(s) = \frac{2}{s^2 + 3s + 2}$. Determine *H(z)* using impulse invariant transformation. Assume *T*=1 second. (16)

13. (a) Design a Low Pass Filter with 11 coefficients for the following Specifications: pass frequency edge is 0.25*kHz* and sampling frequency is 1*kHz* using hanning window. (16)

Or

(b) (i) Show the FIR linear phase realization of the system function $H(z) = (1+ \frac{1}{2} z^{-1} + z^{-2}) (1+ \frac{1}{4} z^{-1} + z^{-2})$. (8)

(ii) Summarize the design procedure for Linear phase FIR system using frequency sampling method. (8)

14. (a) A digital system is characterized by the difference equation $y(n)= 0.95y(n-1) + x(n)$ with $x(n)= 0.875$, $n=0$. Assume b=4 bits. Find out limit cycle of oscillation and estimate the dead band of the system. (16)

Or

(b) For the following system described equation $y(n) = 0.8\, y(n-1) + x(n)$. Solve the output noise power due to input quantization. Assume b=5 bits. (16)

15. (a) Discuss the sub band coding of speech signal with a suitable example. (16)

Or

(b) (i) Describe on sampling rate reduction by an integer factor ‘I’. (8)

(ii) Explain the sub band coding of speech signal. (8)