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

Question Paper Code: 31461

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

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. Differentiate DTFT and DFT.
2. Define DFT for a sequence *x (n).*
3. Compare direct form I and direct form II realizations of IIR system.
4. Define pre warping. Why it is needed?
5. Collect the properties of Chebyshev filter?
6. Define Gibb's phenomenon.
7. What is truncation?
8. Give some Recommendation to prevent overflow.
9. Give the advantages of multi-rate signal processing.
10. Define interpolation and decimation.

PART - B (5 x 16 = 80 Marks)

11. (a) Given DFT $\{x(n)\} = X(k) = \{7, -0.707-j0.707, -j, 0.707-j0.707, 1, 0.707 + j0.707, j, -0.707 + j0.707\}$. Find IDFT using DIT-FFT algorithm. (16)

Or

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

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

$0.9 \le |H(e^{jw})| \le 1; \quad 0 \le \omega \le \pi/2$

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

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 $H(z) = (1+ ½\ z^{-1} + z^{-2})\ (1+ ¼\ z^{-1} + z^{-2})$. (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) (i) 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. (8)

(ii) Describe the different addressing modes of TMS320C5X with examples. (8)

15. (a) (i) Discuss the multistage implementation of sampling rate conversion. (8)

(ii) Discuss the sub band coding of speech signal with a suitable example. (8)

Or

(b) (i) Illustrate about multi rate signal processing. (8)

(ii) Explain about poly phase realization of filter banks. (8)