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Question Paper Code: 71774

25/05/17 FN

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

Fourth Semester

Electrical and Electronics Engineering

EE 6403 — DISCRETE TIME SYSTEMS AND SIGNAL PROCESSING

(Common to Electronics and Instrumentation Engineering, Instrumentation and Control Engineering)

(Regulations 2013)

Time: Three hours

Maximum: 100 marks

Answer ALL questions.

PART A — $(10 \times 2 = 20 \text{ marks})$

- 1. What are the energy and power of discrete signal?
- 2. State sampling theorem.
- 3. Write the properties of region of convergence?
- 4. Find the convolution of the input signal {1, 2, 1 } and its impulse response {1, 1, 1 } using Z transform.
- 5. Define twiddle factor. Write its magnitude and phase angle.
- 6. Compute the number of multiplications and additions for 32 point DFT and FFT.
- 7. Write the advantages and disadvantages of digital filters.
- 8. Define prewarping effect.
- 9. What is pipelining and how do define its depth?
- 10. Write some commercial DSP processors.

PART B — $(5 \times 13 = 65 \text{ marks})$

11. (a) Determine the following systems are linear, stability and time invariance of the system (i) y(n) = x(2n) (ii) $y(n) = \cos x(n)$ (iii) y(n) = x(n) + nx(n+1). (13)

Or

- (b) (i) Explain the process of quantization and its error types (10)
 - (ii) Compute the Nyquist sampling frequency of the signal $x(t) = 4 \sin c (3t/\pi)$.
- 12. (a) (i) State and prove convolution and Parseval's theorem using Z transform. (6)
 - (ii) Find the Z transform of the system $x(n) = \cos(n\theta) u(n)$ (7)

Or

- (b) Find the inverse Z transform of X(z) = (x+1)/(x+0.2)(x-1), |z| > 1 using residue method. (13)
- 13. (a) Determine the 8 point DFT of the sequence $x(n) = \{1, 1, 1, 1, 1, 1, 0, 0\}$.

Or

- (b) Compute 8 point DFT of the given sequence using DIT algorithm $x(n) = \begin{cases} n & n \le 7 \\ 0 & \text{otherwise} \end{cases}$ (13)
- 14. (a) Design a 15 tap linear phase filter using frequency sampling method to the following discrete frequency response $H\left(\frac{2\Pi k}{15}\right) = \begin{cases} 1 & 0 \le k \le 3\\ 0.4 & k = 4 \\ 0 & k = 5, 6, 7 \end{cases}$

Or

- (b) Using bilinear transformation, design a high pass filter, monotonic in passband with cutoff frequency of 1000 Hz and down 10 dB at 330 Hz. The sampling frequency is 5000 Hz. (13)
- 15. (a) Discuss the features and architecture of TMS 320C50 processor. (13)

Or .

(b) Explain the addressing modes and registers of DSP processors. (13)

PART C — $(1 \times 15 = 15 \text{ marks})$

16. (a) The analog signal has a bandwidth of 4KHz. If we use N point DFT with N = 2^m (m is an integer) to compute the spectrum of the signal with resolution less than or equal to 25 Hz. Determine the minimum sampling rate, minimum number of required samples and minimum length of the analog signal. What is the step size required for quantize this signal. (15)

Or

(b) Convert the single pole low pass filter with system function $H(z) = \frac{0.5 \left(1 + Z^{-1}\right)}{1 - 0.302 \ Z^{-3}} \text{ into band pass filter with upper and lower cutoff}$ frequencies ωu and ωl respectively. The lowpass filter has 3dB bandwidth and $\omega p = \pi/6$ and $\omega u = 3\pi/4$, $\omega l = \pi/4$ and draw its realization in direct form II.