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VIVEKANANDHA COLLEGE OF ENGINEERING FOR WOMEN
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**Question Paper Code: 7013**

B.E. / B.Tech. DEGREE SUPPLEMENTARY EXAMINATIONS – FEB. / MAR. 2020

Fifth Semester

Electronics and Communication Engineering

U15EC515 – DIGITAL SIGNAL PROCESSING

(Regulation 2015)

Time : Three Hours Maximum : 100 Marks

Answer ALL the questions

PART – A

(10 x 2 = 20 Marks)

1. Compute the 4 Point DFT of the sequence x(n) = {2,4,6,8}
2. How many number of multiplications and additions required to compute 64 point DFT using FFT Algorithm.
3. Mention the methods to design FIR filter.
4. Give the equation for hamming window function.
5. What is frequency warping?
6. List the methods to convert analog filter into digital filter.
7. What is the range of error in rounding?
8. List some of the finite word length effects in digital filter.
9. What is pipelining in DSP?
10. What are the addressing modes of TMS320C54x processors?

PART – B

(5 x 13 = 65 Marks)

11. a) Consider the finite duration sequence

$$x(n) = \{1, 2, 3, 4, -5, 6, 7, 8\}$$

Compute the eight point DFT using the Radix-2 decimation in time algorithm of the sequence.

(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

i. Overlap Save Method

ii. Overlap Add Method

12. a) Design a FIR Linear phase, Digital filter approximating the ideal high-pass filter with a frequency response

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

i. Determine the co-efficient of 11 tap filter based on the Hanning window method. (9)

ii. Determine and plot the magnitude and phase response of the filter. (4)

(OR)

b) Design a LP FIR filter using Frequency sampling technique having cutoff frequency of π/2 rad / sample. The filter should have linear phase and length of 17.

13. a) i. Explain about the analog transformation of prototype LPF to other filters. (8)

ii. Compare any 2 types of IIR filter digital realization. (5)

(OR)

b) Design a Digital Butterworth high pass filter with a minimum passband attenuation of 2.5 dB at Ωp = 20 rad/sec and the stopband attenuation of 30 dB at Ωs = 50 rad/sec. Use bilinear transformation with the sampling time of 1 sec.

14. 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 $a_2 = 0.6$.

(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,

$$x(n) = 0.875 \quad \text{for } n = 0$$
$$= 0 \quad \text{otherwise}$$

15. a) Explain TMS320C5X DSP processor's building blocks with necessary schematic.

(OR)

b) Explain the following with neat schematic.

i. Harvard architecture and its advantages (7)

ii. Dedicated MAC unit and its advantages (6)

## PART – C

(1 x 15 = 15 Marks)

16. a) i. Write an assembly language program to perform linear convolution of two sequences for TMS320C5X DSP processor. (10)

ii. Compare DIT-FFT with DIF-FFT. (5)

(OR)

b) i. Write an assembly language program to generate sine wave using TMS320C5X DSP processor. (10)

ii. Explain the process of sampling rate conversion by a rational filter. (5)