Reg. No. :

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B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2012.

Fifth Semester

Electronics and Communication Engineering

EC 2302/EC 52 – DIGITAL SIGNAL PROCESSING

(Regulation 2008)

(Common to PTEC 2302 – Digital Signal Processing for B.E. (Part – Time) Fourth Semester Electronics and Communication Engineering – Regulation 2009)

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. What is twiddle factor?

2. How many stages of decimations are required in the case of a 64 point radix 2 DIT FFT algorithm?

3. Why is the butterworth response called a maximally flat response?

4. What is frequency warping?

5. What are the features of FIR filter design using the Kaiser's approach?

6. Draw the direct form implementation of the FIR system having difference equation.

   $y(n) = x(n) - 2x(n-1) + 3x(n-2) - 10x(n-6)$.

7. What are limit cycle oscillations?

8. What is dead - band of a filter?

9. What is decimation?

10. Find the expression for the following multirate system.

$$\longrightarrow \boxed{\downarrow 4} \longrightarrow \boxed{\uparrow 24} \longrightarrow \boxed{\downarrow 6} \longrightarrow$$

PART B — ($5 \times 16 = 80$ marks)

11. (a) (i) Differentiate DFT from DTFT. (4)

(ii) Compute an 8 point DFT of the sequence $x(n) = (1,0,1,-1,1,-1,0,1)$. (12)

Or

(b) (i) Prove that FFT algorithms help in reducing the number of computations involved in DFT computation. (6)

(ii) Compute a 8 point DFT of the sequence using DIT-FFT algorithm $x(n) = (1,2,3,2,1,0)$. (10)

12. (a) (i) Explain the procedure for designing analog filters using the chebysher approximation. (6)

(ii) Convert the following analog transfer function into digital using impulse invariant mapping with T = 1 sec.

$$H(s) = \frac{3}{(s+3)(s+5)}. \quad (10)$$

Or

(b) (i) Design a digital second order low - pass butterworth filter with cut - off frequency 2200 Hz using bilinear transformation. Sampling rate is 8000 Hz. (8)

(ii) Determine the cascade form and parallel form implementation of the system governed by the transfer function

$$H(Z) = \frac{(1+Z^{-1})(1-5Z^{-1}-Z^{-2})}{(1+2Z^{-1}+Z^{-2})(1+Z^{-1}+Z^{-2})}. \quad (8)$$

13. (a) Design an FIR low pass digital filter by using the frequency sampling method for the following specifications

Cutoff frequency = 1500Hz

Sampling frequency = 15000 Hz

Order of the filter : N=10

Filter length required $L = N + 1 = 11$.

Or

(b) (i) Explain with neat sketches the implementation of FIR filters in the (6)

(1) direct form

(2) lattice form

(ii) Design a digital FIR band - pass filter with lower cut - off frequency 2000 Hz and upper cut off frequency 3200 Hz using Hamming window of length N = 7. Sampling rate is 10000 Hz. (10)

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

(ii) Explain coefficient quantization in IIR filter.

Or

(b) (i) How to prevent limit cycle oscillations? Explain.

(ii) What is meant by signal scaling? Explain.

15. (a) (i) Explain sampling rate conversion by a rational factor and derive input and output relation in both time and frequency domain. (10)

(ii) Explain the multistage implementation of sampling rate conversion. (6)

Or

(b) (i) Explain the design of a narrow band filter using sampling rate conversion. (8)

(ii) Explain the application of sampling rate conversion in sub - band coding. (8)

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