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

R 3301

B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2007.

Fifth Semester

(Regulation 2004)

Electronics and Communication Engineering

EC 1302 — DIGITAL SIGNAL PROCESSING

(Common to B.E. (Part-Time) Fourth Semester Regulation 2005)

Time : Three hours

Maximum : 100 marks

Answer ALL questions.

PART A — ($10 \times 2 = 20$ marks)

1. State and prove Parseval's relation for DFT.

2. What do you mean by the term "bit reversal" as applied to FFT?

3. In the design of FIR digital filters, how is Kaiser window different from other windows?

4. Find the digital transfer function $H(z)$ by using impulse invariant method for the analog transfer function $H(s) = \frac{1}{s+2}$. Assume $T = 0.1$ sec.

5. Express the fraction (-7/32) in signed magnitude and two's complement notations using 6 bits.

6. What do you mean by limit cycle oscillations in digital filters?

7. Define unbiased estimate and consistent estimate.

8. Define the terms : autocorrelation sequence and power spectral density.

9. What are the factors that may be considered when selecting a DSP processor for an application?

10. What is pipelining?

PART B — ($5 \times 16 = 80$ marks)

11. (a) Two finite duration sequences are given by

$x[n] = \sin(n\pi/2)$ for $n = 0, 1, 2, 3$

$h[n] = 2^n$ for $n = 0, 1, 2, 3$

(i) Calculate the 4 point DFT $X[k]$. (5)

(ii) Calculate the 4 point DFT $H[k]$. (5)

(iii) If $Y[k] = X[k]H[k]$, determine the inverse DFT $y[n]$ of $Y[k]$ and sketch it. (6)

Or

(b) (i) Obtain an 8-point decimation-in-frequency FFT flow graph from first principles. (8)

(ii) Using the above flow graph compute DFT of

$x[n] = \cos(n\pi/4), 0 \le n \le 7$. (8)

12. (a) (i) Describe the design of FIR filters using frequency sampling technique. (8)

(ii) The desired frequency response of a low pass filter is given by

$$H_D(e^{j\omega}) = \begin{cases} e^{(-j2\omega)}, & -\pi/4 \le \omega \le \pi/4 \\ 0, & \pi/4 \le |\omega| \le \pi \end{cases}$$

Determine the filter coefficients $h_D[n]$. Obtain the coefficients $h[n]$ of FIR filter using a rectangular window defined by

$$w[n] = \begin{cases} 1, & 0 \le n \le 4 \\ 0, & \text{otherwise} \end{cases}.$$ (8)

Or

(b) Design a digital Butterworth filter satisfying the following specifications

$0.7 \le |H(e^{j\omega})| \le 1, \quad 0 \le \omega \le 0.2\pi$

$|H(e^{j\omega})| \le 0.004, \quad 0.6\pi \le \omega \le \pi$

Assume $T = 1$ sec. Apply impulse-invariant transformation. (16)

13. (a) (i) Consider the truncation of negative fraction numbers represented in $(\beta+1)$- bit fixed point binary form including sign bit. Let $(\beta-b)$ bits be truncated. Obtain the range of truncation errors for signed magnitude, 2's complement and 1's complement representations of the negative numbers. (8)

(ii) An 8-bit ADC feeds a DSP system characterized by the following transfer function

$$H(z)=\frac{1}{z+0.5}$$

Estimate the steady state quantization noise power at the output of the system. (8)

Or

(b) (i) The coefficients of a system defined by

$H(z)=\dfrac{1}{(1-0.4z^{-1})(1-0.55z^{-1})}$ are represented in a number system with a sign bit and 3 data bits using signed magnitude representation and truncation. Determine the new pole locations for direct realization and for cascade realization of first order systems. (8)

(ii) An IIR causal filter has the system function $H(z)=\dfrac{z}{z-0.97}$. Assume that the input signal is zero-valued and the computed output signal values are rounded to one decimal place. Show that under these stated conditions, the filter output exhibits dead band effect. What is the dead band range? (8)

14. (a) (i) With suitable relations, explain briefly the periodogram method of power spectral estimation. Examine the consistency and bias of periodogram. (10)

(ii) Explain power spectrum estimation using the Bartlett method. (6)

Or

(b) (i) Explain how the Blackman and Tukey method is used in smoothing the periodogram? Derive the mean and variance of the power spectral estimate of the Blackman and Tukey method. (10)

(ii) Determine the frequency resolution of the Bartlett, Welch and Blackman-Tukey methods of power spectral estimation for a quality factor Q = 15. Assume that overlap in Welch's method is 50% and the length of the sample is 1500. (6)

15. (a) (i) Explain how Harvard architecture as used by the TMS 320 family differs from the strict Harvard architecture. Compare this with the architecture of a standard von Neumann processor. (8)

(ii) A multiplier - accumulator, with three pipe stages, is required for a digital signal processor. Sketch a block diagram of a suitable configuration for the MAC. With the aid of a timing diagram, explain how the MAC works. (8)

Or

(b) (i) In relation to DSP processor, explain the following techniques :

SIMD, VLIW.

In each case, clearly point out the advantages and disadvantages of the technique in signal processing. (8)

(ii) Explain the operation of CSSU of TMS32OC 54X and explain its use considering the Viterbi operator. (8)
