USN | | | | | | | | | | |

14ECS23

Second Semester M.Tech. Degree Examination, June/July 2018
Modern DSP

Time: 3 hrs. Max. Marks:100

Note: ***Answer any FIVE full questions.***

1 a. Draw the block diagram of Analog to Digital converter and explain each block in detail. **(08 Marks)**

b. Consider the analog signal $X_a(t) = 3\cos 2000\pi t + 5\sin 6000\pi t + 10\cos 12{,}000\pi t$.
   i) What is the Nyquist rate for this signal?
   ii) Assume that this signal is sampled at a rate of 5000 samples/sec. What is the discrete time signal obtained after sampling?
   iii) What is the analog signal $Y_a(t)$ that we can reconstruct from the samples if we use ideal interpolation. **(08 Marks)**

c. The discrete time signal $x(n) = 6.35\cos(\pi/10)^n$ is quantized with a resolution i) $\Delta = 0.1$ or ii) $\Delta = 0.02$. How many bits are required in the A/D converter in each case? **(04 Marks)**

2 a. Consider the following 8 point sequences defined for $0 \le n \le 7$
   i) $x_1(n) = \{1, 1, 1, 0, 0, 0, 1, 1\}$
   ii) $x_2(n) = \{1, 1, 1, 0, 0, 0, -1, -1\}$

Which sequences have a real 8-point DFT? Which sequences have an imaginary valued 8-point DFT? **(05 Marks)**

b. Let X(k) be a 14 point DFT of a real sequence x(n). The first 8 samples of x(k) are given by : $x(0) = 12$, $x(1) = -1 + j3$, $x(2) = 3 + j4$, $X(3) = 1 - j5$, $X(4) = -2 + j2$, $x(5) = 6 + j3$, $x(6) = -2 - j3$, $x(7) = 10$.
Determine the remaining samples of X(k). Also evaluate the following functions without computing the IDFT i) X(0) ii) X(7) iii) $\sum_{n=0}^{13} x(n)$ iv) $\sum_{n=0}^{13} |x(n)|^2$ **(11 Marks)**

c. State and prove circular time shift property. **(04 Marks)**

3 a. Find the output y(n) of a filter whose impulse response is $h(n) = \{1, 2, 3, 4\}$ and the input signal to the filter is $x(n) = \{1, 2, 1, -1, 3, 0, 5, 6, 2\}$ using overlap-add method. [Use 6 point circular convolution]. **(10 Marks)**

b. Determine a sequence y(n) such that $y(k) = x_1(k)\, x_2(k)$
Given $x_1(n) = \{0, 1, 2, 3, 4\}$, $x_2(n) = \{0, 1, 0, 0, 0\}$
Use DFT properties. **(05 Marks)**

c. State and prove Parseval's theorem. **(05 Marks)**

4 a. A filter is to be designed with the following desired frequency response,

$$H_d(w) = \begin{cases} e^{-j3w}, & 0 < w < \frac{\pi}{2} \\ 0, & \frac{\pi}{2} < w < \pi \end{cases}$$

Find the frequency response of FIR filter using Hamming window for N = 7. **(10 Marks)**

b. Compare FIR and IIR filters. **(04 Marks)**

c. Explain the design of FIR differentiators. **(06 Marks)**

5 a. Design a Butterworth filter using Bilinear transformation for the following specifications.

$0.8 \le |H(e^{jw})| \le 1$ for $0 \le w \le 0.2\pi$

$|H(e^{jw})| \le 0.2,$ for $0.6\pi \le w \le \pi$ **(10 Marks)**

b. Explain how an analog filter is mapped on to a digital filter using impulse invariance method. What are the limitations of the method? **(10 Marks)**

6 a. Explain the concept of sampling rate conversion by a factor D and factor I. show the effect of sampling rate conversion on the frequency spectrum of the signal. **(14 Marks)**

b. What is Multirate DSP? Explain the methods of sampling rate conversion. **(06 Marks)**

7 a. Develop polyphase structures for Decimation and Interpolation and explain. **(10 Marks)**

b. Explain the use of Multirate DSP in sub-band coding of speech signals. **(10 Marks)**

8 a. Explain with a block diagram, the application of adaptive filters in channel equalization. **(10 Marks)**

b. Explain the concept of minimum mean square error criterion with relevant equations. **(10 Marks)**

* * * * *