Reg. No.										

MANIPAL INSTITUTE OF TECHNOLOGY
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FOURTH SEMESTER B.TECH (E & C) DEGREE END SEMESTER EXAMINATION
MAY/JUNE 2016
SUBJECT: DIGITAL SIGNAL PROCESSING (ECE - 2203)

TIME: 3 HOURS **MAX. MARKS: 50**

Instructions to candidates

- Answer **ALL** questions.
- Missing data may be suitably assumed.

1A. Consider a system described by the difference equation $y[n] = y[n\text{-}1] - y[n\text{-}2] + 0.5x[n] + 0.5x[n\text{-}1]$. Find the response of this system to the input $x[n] = 0.5^n u[n]$ with initial conditions $y[-1] = 0.75$ and $y[-2] = 0.25$.

1B. A signal $x(t)$ bandlimited to 10 kHz is sampled with a sampling frequency of 20 kHz. From the samples obtained, a 1000 point DFT $X[k]$ is computed. To what analog frequency does the index k = 150 correspond? What is the spectral resolution?

1C. Consider a system function of a causal system $H(z) = (z^2+1)/(z^2-0.8z)$. Plot its poles and zeros and describe the filtering characteristics of this system.

(5+3+2)

2A. Starting from the DFT expression, develop the radix 2 decimation in time FFT algorithm and show the complete butterfly diagram of computation of an 8 point DFT.

2B. Consider a finite length sequence $x[n] = \delta[n] + 2\delta[n\text{-}5]$. Find the 10 point DFT $X[k]$ of this sequence and find the sequence $y[n]$ which has a 10 point DFT $Y[k] = \exp(j2k\pi/5)X[k]$.

2C. Let the time taken for a complex multiplication be $1\mu s$. What is maximum time needed for multiplications if a 1024 point DFT is computed directly? How much it would be if radix 2 decimation in frequency FFT algorithm is used.

(5+3+2)

3A. An IIR Butterworth low-pass filter has the following specifications

$$-2dB \le 20\log\left(\left|H(e^{j\omega})\right|\right) \le 0dB\ ,\ 0 \le \omega \le 0.25\,radians$$

$$20\log\left(\left|H(e^{j\omega})\right|\right) \le -20\,dB\ ,\ \omega \ge 1.319\,radians$$

If this digital filter was realized using impulse invariance technique with a sampling frequency of 10 kHz, determine the order, 3 dB cut-off frequency and poles of the equivalent analog filter.

3B. For the above Q.3A determine the transfer function $H(s)$ of the analog filter and convert into digital filter $H(z)$ using impulse invariance technique.

3C. Write the relation between analog frequency Ω, and digital frequency ω in bilinear transformation. What is its impact on the design?

(5+3+2)

4A. A LPF has the desired frequency response $H_d(e^{jw}) = \begin{cases} e^{-j5w} & ,0 \le w \le \frac{\pi}{2} \\ 0, & elsewhere \end{cases}$

Determine the system function using recursive frequency sampling method. Is this a perfectly stable system?

4B. Design a linear phase FIR filter of length M = 5 using rectangular window to approximate the ideal frequency response magnitude $|H_d(e^{jw})| = \begin{cases} 1 & , w \le 0.25\pi \\ 0, & elsewhere \end{cases}$ in the interval of $(-\pi, \pi)$. Also find the frequency response of the resulting FIR filter and mention its group delay.

4C. If two zeros of a linear phase, real coefficient, length 5 FIR filter is 0.5 and -0.25, show all the zeros of this filter in the z-plane and write its system function.

(5+3+2)

5A. Consider a causal IIR system with system function $H(z) = \frac{1+2z^{-1}+3z^{-2}+2z^{-3}}{1+0.9z^{-1}-0.8z^{-2}+0.5z^{-3}}$.

Obtain and draw the lattice-ladder structure for this filter. Is this filter structure stable?

5B. Highlight the main limitations of periodogram in PSD estimation. Discuss how other nonparametric PSD estimation techniques address these limitations.

5C. Write the standard form of system functions obtained through AR and MA modelling.

(5+3+2)