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**MANIPAL INSTITUTE OF TECHNOLOGY**
MANIPAL
*(A constituent unit of MAHE, Manipal)*

**FOURTH SEMESTER B.Tech. (E & C) DEGREE END SEMESTER EXAMINATION**
**APRIL 2018**
**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. Determine the response of LTI system defined by the difference equation y[n] - 0.2y[n-1] - 0.03y[n-2] = x[n] + 0.4x[n-1], for the input x[n] = $0.2^n$ u[n] and with the initial condition y[-2] = 0, y[-1] = 0.5.

1B. Determine all possible impulse responses associated with the system function $H(z) = \frac{5z^{-1}}{(3-z^{-1})(1-2z^{-1})}$. Indicate ROC of H(z) in each case.

1C. Obtain the second order recursive system function (Goertzel algorithm) for efficient calculation of DFT.

(5+3+2)

2A. With mathematical analysis, describe design of linear phase FIR filters using window functions. Highlight the selection of window functions.

2B. Determine N-point DFT of Hamming window given by $w(n) = 0.54 - 0.46\cos\left(\frac{2\pi n}{N-1}\right)$; $0 \le n \le N-1$

2C. Draw the signal flow diagram of basic radix-2 DIT butterfly computation for FFT. How many such computations are required for pipe-line implementation of 16 point FFT?

(5+3+2)

3A. A third order Low-pass Butterworth filter is required to meet the following specifications $\omega_p = \frac{\pi}{3}, \omega_s = 0.5\pi, R_p = -1.5dB, A_s = -10dB.$ Determine the pre-warped analog edge frequencies $\Omega_p$ *and* $\Omega_s$, 3-dB cut-off frequency $\Omega_c$, transfer function H(s) of the analog filter. Obtain the digital filter system function H(z) using bilinear transformation at 1Hz sampling.

3B. Apply matched z transform to convert the analog filter $H(s) = \frac{s+1}{s^2+8s+12}$ to a digital filter H(z). Assume sampling frequency of 1 Hz. Plot the poles and zeroes of H(z) in z-plane.

3C. A linear phase FIR digital filter is described by difference equation $y(n) = \sum_{k=0}^{2} b_k x(n-k)$. Determine the filter coefficient such that it rejects a frequency component at $\omega_0 = \frac{\pi}{3}$ and its frequency response is normalised so that $\left|H(e^{j\omega})\right|_{\omega=0} = 1$.

(5+3+2)

4A. Derive the recursive frequency sampling structure for M length FIR digital filter. Draw the implementation diagram for M=7 with H(k) = 1 for k=0,1,6

=0 for k=2,3,4,5

Highlight the advantage of recursive frequency sampling structure.

4B. Describe the digitisation of analog filter using impulse invariance transformation technique.

4C. Determine the poles and zeroes of IIR notch filter to suppress 50 Hz interference with maximum allowable bandwidth 5Hz. The sampling frequency is 500 Hz.

(5+3+2)

5A. Determine the lattice-ladder parameters for the following digital filter. Sketch lattice-ladder structure and direct form-II structure. $H(z)=\dfrac{1-8z^{-1}+0.15z^{-2}}{1+0.1z^{-1}-0.72z^{-2}}$

5B. Describe Periodogram method of PSD estimation. Illustrate the spectral leakage and resolution problems.

5C. Bring out the salient features of parametric PSD estimation.

(5+3+2)