



# Manipal Institute of Technology, Manipal

(A Constituent Institute of Manipal University)



(05)

# V SEMESTER B.TECH (ELECTRICAL & ELECTRONICS ENGINEERING)

## END SEMESTER EXAMINATIONS, NOV/DEC 2015

## SUBJECT: DIGITAL SIGNAL PROCESSING [ELE 303]

### **REVISED CREDIT SYSTEM**

Time: 3 Hours

27 November 2015

MAX. MARKS: 50

#### Instructions to Candidates:

- ✤ Answer ANY FIVE FULL questions.
- Missing data may be suitably assumed.
- DSP Quick reference table may be used.
- **1A.** Let x[n] be a finite length sequence with X(k)={0, 1+j, 1, 1-j}. Using properties find  $DFT\left\{\cos\left(\frac{\pi n}{2}\right)x[n]\right\}$ . (02)
- 1B. Signal x[n]=10sin(20πt) +20cos(40πt) is sampled at 80Hz and 64 samples are collected. 64 point DFT, X(k) is then computed. At what k values, would you expect to see peaks in the DFT magnitude spectrum? (02)
- **1C.** Explain the principles of radix-2 FFT algorithm. Develop N-point radix-2 DIT<br/>FFT algorithm. Illustrate with signal flow diagrams. (assume N=8)(06)
- **2A.** Determine the output y(n) of a filter whose impulse response is h(n)={1 0 1} and input signal is  $x(n)={1 1 0 -2 3 4 -3 8 0 6 7}$  using Overlap- Save 1

method. Take sub frame length of 4.

- **2B.** Obtain the lattice coefficients for the given FIR filter and draw the corresponding lattice structure.  $H(z) = 8+16z^{-1}+24z^{-2}+16z^{-3}$ . Is the system (05) stable?
- **3A.** A digital notch filter is required to remove an undesirable 50Hz hum associated with a power supply in an ECG recording application. The sampling frequency used  $F_s = 250 \text{ samples}/$ . Design a second order FIR notch filter. Choose  $b_0$  so that |H(w)| = 1 for w = 0. Draw the magnitude spectrum. (03)
- **3B.** Show that M length FIR filter with impulse response satisfying the condition h(n) = h (M-1-n) has linear phase response. (04)
- **3C.** The first five coefficients of 8 point DFT of real valued sequence x(n)=r(n)+jv(n) are  $X(k)=\{15, -4+j9.66, -4+j5, -4+j1.66, -4\}$ . Find the 8 point DFT of the real part and imaginary part of x(n). ELE 202 Base 1 of 2

-7.	Design a linear phase FIR bandpass filter to pass frequencies in the range $0.4\pi$ to $0.65\pi$ rad/sample by taking 7 samples using hanning window sequence. Also realize the linear phase structure.	(05)
4B.	A digital low pass Butterworth Filter is required to meet the following specifications.	
	0.8≤ H(ω) ≤1 for 0≤ω≤0.2π	
	H(ω) ≤0.2 for 0.6π≤ω≤π	(05)
	Design the filter using BLT technique.	(05)
5A.	Design an analog Chebyshev type-I filter to meet the following specifications:	
	Pass band ripple $\leq 2 \text{ dB}$	
	Pass band edge frequency =1 rad/s	
	Stop band attenuation ≥ 20 dB	(02)
	Stop band edge frequency = 1.3 rad/s	(03)
5B.	Explain bilinear transformation technique for digitizing analog filter. What is frequency warping?	(03)
5C.	An analog signal $x_a(t) = sin(480\pi t) + 3sin(720\pi t)$ is sampled 600 times per second.	
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5C. 6A. 6B. 6C.	An analog signal $x_a(t) = sin(480\pi t) + 3sin(720\pi t)$ is sampled 600 times per second. (a) Determine the Nyquist sampling rate. (b) Determine the folding frequency. (c) What are the frequencies in the resulting discrete time signal $x(n)$ ? (d) If $x[n]$ is passed through an ideal D/A converter, what is the reconstructed signal $y_a(t)$ ? Consider an FIR system having impulse response $h(n)=[1, 0.5, 0.5, 1]$ . Realize the system using frequency sampling structure. Compare IIR and FIR filters. Consider the filter $y[n]= -0.9y[n-1]+0.1x[n]$ . Determine the frequency at which $ H(\omega)  = \frac{1}{\sqrt{2}}$ . Is this filter low pass, high	(04) (05) (02)