## Fifth Semester B.E. Degree Examination, June / July 08

# Digital Signal Processing

Time: 3 hrs. Max. Marks:100

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

1 a. Find 8 point DFT of the signal x[n] = [1, 1, 1, 1]. **(08 Marks)**

b. Find the DFT of $x[n] = a^n u[n]$ for $0 \le n \le N-1$. **(04 Marks)**

c. Find the 6 point inverse DFT of

$X(K) = 3 \quad K = 0$

$= 1 \quad 1 \le K \le 5$. **(08 Marks)**

2 a. Consider the sequence: $x[n] = 4\delta[n] + 3\delta[n-1] + 2\delta[n-2] + \delta[n-3]$.

i) Find the finite length sequence y[n] that has a six point DFT $Y(K) = W_6^{4K} X(K)$

ii) Find the finite length sequence w[n] that has a six point DFT which is equal to real part of X(K). **(08 Marks)**

b. By means of DFT and IDFT determine the circular convolution of the sequences: x[n] = [1, 2, -1, -1] and h[n] = [-3, -2, -1, 0]. **(12 Marks)**

3 a. A long sequence is filtered through a filter of impulse response h[n] to give the output y[n] for the input x[n]. Given h[n] and x[n] as follows, compute y[n] using overlap and add method.

x[n] = [1, 1, 1, 1,1, 3, 1, 1, 4, 2, 1, 1, 3, 1, 1, 1]; h[n] = [1, -1].

Use only five point circular convolution in your approach. **(10 Marks)**

b. Explain in detail, the overlap and save method of filtering a long sequence through an FIR filter. **(10 Marks)**

4 a. Use the 8 point DIT radix-2 FFT algorithm to find the DFT of the sequence x[n] = [0.707, 1, 0.707, 0, -0.707, -1, -0.707, 0]. What is the number of the computations in this computation? **(10 Marks)**

b. Develop DIF FFT algorithm with all the necessary steps and signal flow graph, used in computing the N point DFT of a signal x[n]. Using the same compute the four point DFT of the signal x[n] = [44, 22, 33, 22]. **(10 Marks)**

5 a. The Z plane pole-zero plot for a certain digital filter is shown in the Fig.Q5(a) below. The filter has unity gain at DC. Determine the system transfer function in the form:

$$H(z) = A\left[\frac{(1 + a_1 z^{-1})(1 + b_1 z^{-1} + b_2 z^{-2})}{(1 + c_1 z^{-1})(1 + d_1 z^{-1} + d_2 z^{-2})}\right]$$

and draw the block diagram in i) Direct form II and ii) Cascade form. **(12 Marks)**

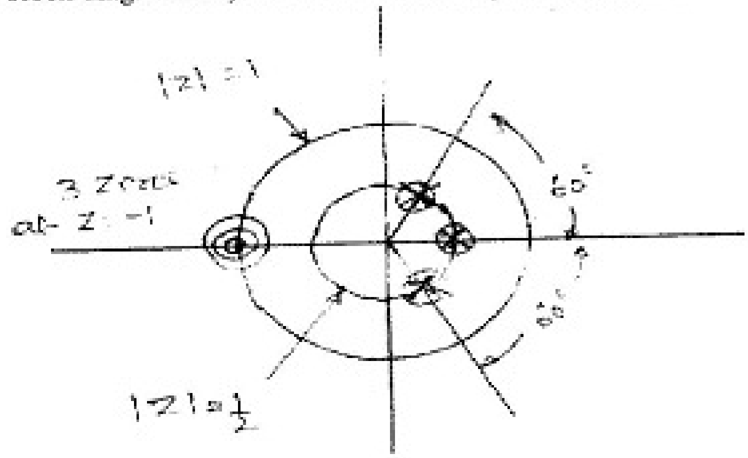


Fig.Q5(a)

b. Realize the following system function by linear phase FIR structure:
$H(z)=1+\frac{1}{2}z^{-1}+\frac{1}{3}z^{-2}+\frac{1}{7}z^{-3}+\frac{1}{3}z^{-4}+\frac{1}{2}z^{-5}+z^{-6}$. **(08 Marks)**

6 a. Obtain the difference equation using the impulse invariance transformation for the analog system function: $H(s)=\frac{s+1}{s^2+5s+6}$. **(08 Marks)**

b. Show that the Impulse Invariance Transformation Maps:
   i) The j Ω axis in the S-plane on the unit circle in Z-plane
   ii) The left half S-plane on the area inside the unit circle in the Z-plane
   iii) The frequency transformation is many to one. **(12 Marks)**

7 a. Design an IIR filter that, when used in the pre filter A/D – H(z) – D/A structure, will satisfy the following equivalent analog specifications:
   i) Pass band with – 1 dB cut off at 50 Hz
   ii) Stop band attenuation of 35 dB at 500 Hz
   iii) Monotonic pass band and stop band
   iv) Sampling rate of 2000 samples / sec.
   Use bilinear transformation. **(10 Marks)**

b. Design the Chebychev filter using bilinear transformation to meet the following specifications:
$0.707 \le |H(\omega)| \le 1; \quad 0 \le \omega \le 0.2\pi$
$|H(\omega)| \le 0.1; \quad 0.5\pi \le \omega \le \pi$ **(10 Marks)**

8 a. Describe various types of windows used in the design of FIR filters. **(06 Marks)**

b. Design a low pass filter with pass band gain of unity. Cut off frequency of 1000 Hz and working at a sampling frequency of 5 kHz. The length of impulse response is 7. Use rectangular window. **(10 Marks)**

c. Explain the Harvard Architecture of the DSP processor. **(04 Marks)**

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