NEW SCHEME

EE55

Reg. No.

Fifth Semester B.E. Degree Examination, January/February 2006

Electrical & Electronics Engineering

Digital Signal Processing

Time: 3 hrs.) (Max.Marks : 100

Note: *1. Answer any FIVE full questions.*
2. Missing data may be suitably assumed but a mention of it is a must.
3. Do no use filter tables.

1. (a) Give f$x(n) = u(n) - u(n-6)$ determine $x(w)$ and $x(k)$ if $x(w)$ is sampled at $w_k = \frac{2\pi}{4}k$ for $k = 0, 1, 2, 3$. Determine an expression to reconstruct $x(n)$ from $x(k)$ and also draw the reconstructed $x(n)$. **(10 Marks)**

   (b) Given $x(n) = (\frac{1}{2})^n \, [u(n) - u(n-4)]$ determine the following without computing $4-pt\ DFT\ x(k)$

   i) if $G(k) = W_4^{2k} x(k)$ find $g(n)$

   ii) $\sum_{n=0}^{3} x(k)x^*(k)$

   iii) $x(0) + x(2)$ **(10 Marks)**

2. (a) A discrete time signal $x(n) = \{1, 4, 1, 4, 1\}$ is passed through a LSI system with impulse response $h(n) = \{\frac{1}{2}, \frac{1}{2}\}$ to generate $y(n)$. Determine $y(n)$ if DFT based convolution scheme is employed. **(10 Marks)**

   (b) Derive and draw the complete decimation-in-frequency flow chart to compute DFT of a 8-pt sequence. Mark all inter mediate outputs. **(10 Marks)**

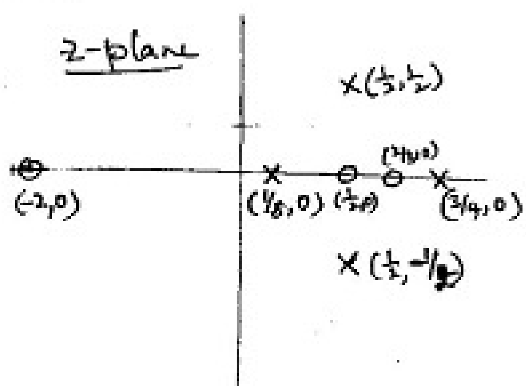
3. (a) Compute 8-pt real sequence given following DFT samples

   $$X(k) = \{2, 0.5 - j1.207, 0, 0.5 - j0.207, 0\}$$

   using a FFt algorithm which accepts input in a bit-reversed order. **(10 Marks)**

   (b) Show that symmetry and antisymmetry in impulse response of a FIR filter leads to linear phase characteristics. **(10 Marks)**

4. (a) A pole zero plot of a discrete time system is as shown in fig. 4a. Obtain transfer function of the system and realize the same in cascade and parallel form. Use only 2nd order structures. **(14 Marks)**



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(b) Check whether the following system has linear phase characteristics and further draw the linear phase realization of the same

$$H(Z) = 1 + \frac{1}{3}Z^{-1} + \frac{1}{4}Z^{-2} + \frac{1}{4}Z^{-3} + \frac{1}{3}Z^{-4} + Z^{-5}.$$ **(6 Marks)**

5. (a) Use the technique of impulse invariance to derive a low pass IIR digital filter from a second order Butter worth analog filter with 3-dB cutoff frequency of 3kHz. The sampling rate used is 30kHz. **(10 Marks)**

(b) Derive the mapping function used in transforming analog filter to digital filter by bilinear transformation. Show that this transformation preserves the frequency selectivity and stability properties of analog filter. **(10 Marks)**

6. (a) Determine the order and the poles of a type I low pass Chebyshev filter that has a 1dB ripple in the pass band and pass band frequency of $\Omega p = 1000\pi$ rad/sec, a stopband of $2000\pi rad/sec$ and an attenuation of 40dB or more. **(10 Marks)**

(b) Using bilinear transform, design a highpass filter, monotonic in pass band with cutoff frequency of 1000 Hz and down to 10dB at 350Hz. The sampling frequency is 5000 Hz. **(10 Marks)**

7. Design an ideal high pass filter with the following frequency response using

i) Rectangular window ii) Hanning window

$$H_d(e^{jw}) = \begin{cases} 1 & \frac{\pi}{4} \le |\omega| \le \pi \\ 0 & |\omega| < \frac{\pi}{4} \end{cases}$$

The filter must have a length = 7. Compare the two designs for their magnitude response and comment on them.

(Note : Hanning window $W_h(n) = 0.5 + 0.5\ cos\ \frac{2\pi n}{N-1}$ for $-(N-1)/2$ to $(\frac{N-1}{2})$ otherwise it is zero) **(20 Marks)**

8. Write an explanatory notes on :

a) DSP's in DTMF signal detection

b) Inplace computation & Buttinfly operation

c) DSP architecture

d) Effects of window characteristics on filter response. **(4x5=20 Marks)**

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