

## Discrete Time Signal Processing Laboratory

T.E. Sem-VI (CBSGS) ENTC - DTSP  
(3 Hours)

Q.P. Code : 588402

2011/16

Total Marks : 80

N. B. : (1) Q.1 is compulsory.

(2) Solve any three questions from remaining 6 questions

(3) Assume suitable data if it is required.

Q.1 (a) Explain phase delay and group delay [20]  
(b) What are the advantages of digital filter over analog filter?  
(c) State and prove frequency shifting property of DFT  
(d) Compare FIR filter and IIR filter.

Q.2.(a)(i)  $x(n) = \{1, 2, 3, 4\}$  find DFT  $X(k)$  [10]

(ii) Using results obtained in part (i) and otherwise find DFT of following sequences

$$a(n) = \{4, 1, 2, 3\} \quad b(n) = \{2, 3, 4, 1\} \quad c(n) = \{3, 4, 1, 2\} \quad d(n) = \{4, 6, 4, 6\}$$

(b) A digital filter is described by the following differential equation [10]

$$y(n) = 0.9y(n-1) + bx(n)$$

(i) Determine  $b$  such that  $|H(0)| = 1$

(ii) Determine the frequency at which  $|H(w)| = \frac{1}{\sqrt{2}}$

(iii) Identify the filter type based on the passband.

Q3 (a) If  $x(n) = \{1, 0, 2, 0, 3, 0, 4, 0\}$ , Find  $X(K)$  using DIFFFT. Compare computational complexity of [10]  
above algorithm with DFT.

(b) Explain effect of aliasing in Impulse Invariant Technique

Using this method, determine  $H(Z)$  if  $H(s) = \frac{3}{(s+2)(s+3)}$  if  $T = 0.1$  sec [10]

Q.4 (a) Design a Linear Phase FIR Low Pass filter of Length 7 and cut off frequency 1 rad/sec using [10]  
Hamming window.

(b) if  $x(n) = \{1, 2, 3, 2\}$  and  $h(n) = \{5, 6, 7, 8\}$  [10]

a) Find circular convolution using time domain method.

b) Find circular convolution using DFT / IDFT method.

c) Find linear convolution using circular convolution.

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[TURN OVER]

Q.5(a) Design a digital Butterworth filter for following specifications using Bilinear Transformation Technique

[10]

Attenuation in Pass band = 1.93 dB, Pass band Edge frequency =  $0.2\pi$   
Attenuation in Stop band = 13.97dB Stop band Edge frequency =  $0.6\pi$

(b) With a suitable block diagram describe sub-band coding of speech signals.

[10]

Q.6(a) Determine FIR lattice coefficient of system with transfer function

[10]

$$H(Z) = 1 + \frac{13}{24}Z^{-1} + \frac{5}{8}Z^{-2} + \frac{1}{3}Z^{-3}$$

(b) Write a note on Frequency Sampling realization of FIR Filter

[10]

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