Total No. of Questions : 10]

SEAT No. :

P3503

[Total No. of Pages : 4

[5560]-152
T.E. (E & TC)
DIGITAL SIGNAL PROCESSING
(2012 Course) (304182)

Time : 2½ Hours] *[Max. Marks : 70*

Instructions to the candidates:

1) *Answer all questions.*
2) *Figures to the right indicate full marks.*

Q1) a) Draw the spectrum for three Nyquist cases of sampling as

i) $f_s > 2\ f_{max}$

ii) $f_{max} < f_s < 2\ f_{max}$

iii) $f_s < f_{max}$

with respect of frequency axis. **[6]**

b) i) Write analysis and synthesis equations for DTFT. Write down its basis function.

ii) What is orthogonality? Write its application.

[4]

OR

Q2) a) Calculate 4-point DFT using DIT-FFT algorithm for $x(n)=2^{(2n)}$ **[6]**

b) Find $X(5), X(6), \& X(7)$ for given 8-point DFT,

X(k) = {20, –5.82 –2.41j, 0, –0.17 –0.41j, 0, ___, ___, ___,}

Which property did you use for writing remaining three values? **[4]**

Q3) a) Draw the ROC for

i) Stable & causal

ii) Stable & non-causal

iii) Unstable & causal

IIR systems. **[6]**

b) Write any two properties of DFT along with their mathematical equations. **[4]**

OR

P.T.O.

Q4) a) Determine the system function H(Z) of

$$y(n)+\frac{3}{4}y(n-1)+\frac{1}{8}y(n-2)=x(n)+x(n-1).$$

Show poles & zeros in Z-plane. **[6]**

b) Calculate Z-transform of

$$x(n)=\left(\frac{1}{4}\right)^{(n-1)}.$$ Draw ROC. **[4]**

Q5) a) i) Convert the analog filter with system function,

$$\mathrm{Ha}(\mathrm{S})=\frac{\mathrm{S}+0.1}{(\mathrm{S}+0.1)^2+16}$$

into a digital IIR filter by means of the Bilinear Transformation. The digital filter is to have a resonant frequency of $\mathrm{Wr} = \pi/2$.

ii) Implement this filter using Direct form - II structure. **[8]**

b) Draw the labelled magnitude response for

i) Butterworth LPF

ii) Chebyshev Type I & Type II LPF

Show f_p, f_c, f_s in the diagram. **[8]**

OR

Q6) a) i) Write the substitutions for 'S' for

1) Approximation of derivatives

2) Impulse Invariance

3) Bilinear Transformations

to convert the analog TF to digital TF (transfer function)

ii) State one advantage & one limitation of Impulse Invariance Method.

[8]

b) Realize the

i) Cascade &

ii) Parallel form

structure for the given TF :

$$H(Z)=\frac{\left(1-\frac{1}{2}Z^{-1}\right)}{\left(1-\frac{1}{4}Z^{-1}\right)\left(1+\frac{1}{4}Z^{-1}\right)}$$ **[8]**

Q7) a) Determine a Direct form realization for the following linear phase filters

i) h(n) = {1, 2, 3, 4, 3, 2, 1}

ii) h(n) = {1, 2, 3, 3, 2, 1} **[8]**

b) Write expressions for

i) Phase delay

ii) Group delay

iii) Linearity condition for symmetrical & antisymmetrical FIR systems.

[8]

Draw

i) Symmetric and

ii) Asymmetric impulse responses

OR

Q8) a) i) What are the possible types of impulse response for linear phase FIR filters?

ii) The frequency response of a digital filter is

$$H\left(e^{jw}\right)=\left(0.4+0.7\cos 2w-0.5\cos 4w\right).e^{-j(0.3\pi+4w)}$$

Determine the phase delay and group delay. **[8]**

b) Design a linear phase FIR low pass filter using rectangular window by taking 7 samples of window sequence, and with a cut-off frequency, $W_c = 0.2\pi$ rad/sample.

Implement the above designed FIR LPF using linear phase structure.

[8]

Q9) a) i) Draw block schematic for

1) Decimation

2) Interpolation

ii) Consider the discrete time signal,

$x(n) = \{1,2,3,4,5,6,7,8,9,10\}$

Determine the result of the signal when

1) D = 2 &

2) I = 3

3) I = 2

[9]

b) Discuss:

i) DMA

ii) MAC and

iii) VLIW architecture

[9]

OR

Q10) Write short notes on: **[18]**

i) Music signal processing

ii) Image processing

iii) Radar signal processing

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