B.E.(with Credits)-Regular-Semester 2012 - Electrical Engineering & (E. & P.) /Electronics & Telecommunication / Communication Engineering Sem VI

## EE602/ET604 - Digital Signal Processing

## GUG/W/16/5363 P. Pages: 2 Time : Three Hours Max. Marks: 80 Notes : 1. All questions carry equal marks 2. Due credit will be given to neatness and adequate dimensions. Assume suitable data wherever necessary. 3. 4. Illustrate your answers wherever necessary with the help of neat sketches. Find the DFT of a sequence 1. a) 8 x(n) = 1for $o \le n \le 2$ =0otherwise for N=8 plot |x(k)| and $\langle x(k)$ . Find the IDFT of the sequence b) 8 $x(k) = \{5,0,1-i,0,1,0,1+i,0\}$ OR 2. a) Find the circular convolution of the two sequences $x_1(n) = \{1, 2, 2, 1\}$ and $x_2(n) = \{1, 2, 3, 1\}$ 8 using concentric circle method. Find the 8 point DFT of the sequence $x(n) = \{1,1,1,1,1,0,0,0\}$ using DIT-FFT algorithm. b) 8 3. a) Obtain the cascade realisation of FIR system function. 8 $H(z) = (1 + 2z^{-1} - z^{-2})(1 + z^{-1} - z^{-2})$ b) Draw the general realisation structure in direct form of FIR system with necessary 8 expressions and derivation. OR

- 4. a) Obtain the direct form -I realisation for the IIR system described by difference equation. 8 y(n) = 0.5y(n-1) - 0.25y(n-2) + x(n) + 0.4x(n-1)
  - b) Determine the transposed direct form II for the given system.  $y(n) = \frac{1}{2}y(n-1) - \frac{1}{4}y(n-2) + x(n) + x(n-1)$ 8
- 5. a) Distinguish between FIR and IIR filters. b) If the frequency response of a linear phase FIR filter is given by  $H(e^{j\omega}) = e^{-j2\omega}(0.30 + 0.5\cos\omega + 0.3\cos2\omega)$ determine the filter coefficients. 8

## OR

<i>a)</i>	Explain the procedure for designing FIR filters using windows.	8
b)	Explain Hamming window method for designing of the FIR filters.	8
a)	Explain the steps to design an analog Chebyshev low pass filter.	8
b)	Design a Chebyshev filter with a maximum passband attenuation of 2.5dB, at $\Omega_P = 20$ rad/sec and the stopband attenuation of 30dB at $\Omega_S = 50$ rad/sec.	8
	OR	
a)	For the analog transfer function $H(S) = \frac{2}{(S+1)(S+2)}$ determine H (z) using impulse invariance method Assume T=1sec.	8
b)	Explain the designing of IIR filter using Bilinear transformation.	8
a)	Show that linear interpolation scheme can be realised by basic digital signal processing.	8
b)	Explain the process of reducing the sampling rate by a factor D. ie. decimation by a factor D.	8
	OR	
a)	<ul> <li>Consider the signal x(n) = a<sup>n</sup>u(n),  a &lt;1</li> <li>i) Determine the spectrum x(ω)</li> <li>ii) The signal x(n) is applied to a decimator that reduces the rate by a factor of 2. Determine the output spectrum.</li> </ul>	8
b)	Explain the analysis filter bank in detail.	8
	<ul> <li>a)</li> <li>b)</li> <li>a)</li> <li>b)</li> <li>a)</li> <li>b)</li> <li>a)</li> <li>b)</li> <li>a)</li> <li>b)</li> <li>b)</li> <li>b)</li> </ul>	<ul> <li>a) Explain the proceeder for designing if it much a unit withows:</li> <li>b) Explain Hamming window method for designing of the FIR filters.</li> <li>a) Explain the steps to design an analog Chebyshev low pass filter.</li> <li>b) Design a Chebyshev filter with a maximum passband attenuation of 2.5dB, at Ω<sub>P</sub> = 20 rad/sec and the stopband attenuation of 30dB at Ω s=50rad/sec.</li> <li>OR</li> <li>a) For the analog transfer function H(S) = 2/(S+1)(S+2) determine H (z) using impulse invariance method Assume T=1sec.</li> <li>b) Explain the designing of IIR filter using Bilinear transformation.</li> <li>a) Show that linear interpolation scheme can be realised by basic digital signal processing.</li> <li>b) Explain the process of reducing the sampling rate by a factor D. ie. decimation by a factor D.</li> <li>OR</li> <li>a) Consider the signal x(n) = a<sup>n</sup>u(n),  a &lt;1</li> <li>i) Determine the spectrum x(ω)</li> <li>ii) The signal x(n) is applied to a decimator that reduces the rate by a factor of 2. Determine the output spectrum.</li> <li>b) Explain the analysis filter bank in detail.</li> </ul>