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PES UNIVERSITY

PES University, Bangalore
(Established under Karnataka Act 16 of 2013)

UE14EC302

END SEMESTER ASSESSMENT (ESA) B.TECH. V SEMESTER- DECEMBER. 2016

UE14EC302 - DIGITAL SIGNAL PROCESSING

Time: 3 Hrs | Answer All Questions | Max Marks: 100

1.	a)	A narrowband signal is sampled at 8 kHz and we take the DFT of 16 points as follows. Determine the best estimate of the frequency of the sinusoid, its possible range of values and an estimate of the amplitude X = [0.4889, 4.0267-j24.6698, 2.0054-j5.0782, 1.8607-j2.8478, 1.8170-j1.8421, 1.7983-j1.2136, 1.7893-j0.7471, 1.7849-j0.3576, 1.7837, 1.7849+ j0.3576, 1.7893+j0.7471, 1.7983+j1.2136, 1.8170+j1.842, 1.8607+j2.8478, 2.0054+j5.0782, 4.0267+j24.6698]	5
	b)	Compute 4-point DFT of the following using linear transformation equations: (i) $x(n)=\cos(\pi n)$ (ii) $x(n)=\sin(\pi n/2)$	6
	c)	Determine N-point IDFT of $X(k)=\delta(k)$, k=0....N-1, and hence for k_0=0....N-1, compute the N-point DFT of $x_1(n)=\cos(2\pi k_0 n/N)$.	4
	d)	The impulse response of an LTI system is $h(n)=\{1,0,-1\}$. For an input $x(n)=(n+1)$, where n=0....9, find the response of y(n) of the system using overlap-add method. Perform only 6-point circular convolution.	5
2.	a)	Compute IDFT of the given $X(k)=\{7, -0.707-j0.707, -j, 0.707-j0.707, 1, 0.707+j0.707, j, -0.707+j0.707\}$ using inverse DIT-FFT algorithm.	8
	b)	Develop a radix-2 DIF- FFT algorithm for N=8 and draw the corresponding signal flow graph.	6
	c)	Using DIT- FFT algorithm, compute the output y(n) of a linear filter described by h(n)={1, 2} and x(n)={1,1,1}.	6
3.	a)	In your CD the data is sampled at 44.1kHz (CD quality), and we want to have a good sound quality up to 21kHz. If you had to use an analog Chebyshev type-I filter as a reconstruction filter, what would be the order of your filter? Assume a passband ripple of 1 dB and a stopband attenuation of 40dB.	5
	b)	Derive an expression to determine the poles of a normalized analog lowpass Butterworth filter. Determine the poles for N=2 and sketch the poles of $H_a(s)H_a(-s)$.	6
	c)	Design a second order bandpass Chebyshev type-I filter with the passband from 200Hz to 300Hz, at an allowed ripple of 0.5 dB.	6
	d)	Find the recursive relation to determine the Chebyshev polynomial $T_N(x)=\cos Nt \mid_{x=\cos t}$ for various values of N.	3
4.	a)	Using bilinear transformation design a second order lowpass Butterworth filter with a cut-off frequency of 1kHz and sampling frequency of 10,000 samples/sec.	6

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	b)	Find the digital filter transfer function of the given H(s) using impulse invariant method and realize it employing direct form II method. Assume T=1 sec. $$H(s) = \frac{s+0.2}{(s+0.2)^2+3^2}$$	7
	c)	Obtain the parallel form realization of the given LTI system governed by the difference equation, `y(n)=(5/8)y(n-1)- (1/16)y(n-2)+x(n)- 3x(n-1)+ 3x(n-2)- x(n-3)`	7
5.	a)	Design a band reject FIR filter for N=5, ω_{c1}=0.5 rad and ω_{c2}=0.75 rad. Using hamming window determine the coefficients of the filter.	6
	b)	Using frequency sampling method design a bandpass filter with the following specifications: Sampling frequency: 8kHz Cut-off frequencies F_{c1}=1 kHz and F_{c2}=3kHz Length of the filter M=7	6
	c)	Consider an FIR lattice filter with co-efficients, k_0=0.65, k_1=-0.34, and k_2=0.8 (i) Find the transfer function H(z) of the filter (ii) Draw the equivalent direct form structure	8