Reg No.:_______________ Name:__________________________

APJ ABDUL KALAM TECHNOLOGICAL UNIVERSITY

FIFTH SEMESTER B.TECH DEGREE EXAMINATION(R&S), DECEMBER 2019

Course Code: EC301

Course Name: DIGITAL SIGNAL PROCESSING

Max. Marks: 100 Duration: 3 Hours

PART A

Answer any two full questions, each carries 15 marks. Marks

1 a) StateParseval's theorem of DFT?Using DFT find the energy of the sequence$x(n) = 0.2^n\ u(n)\,, n < 4$. (7)

b) Compute 8-point DFT of the sequence x(n)= $\{\frac{1}{2},\frac{1}{2},\frac{1}{2},\frac{1}{2},0,0,0,0\}$ using DITFFT algorithm. Follow exactly the corresponding flow graphs and keep track of all the intermediate quantities by putting them on diagram. (8)

2 a) Find linear convolution of the sequences x(n) and z(n) using circular convolution. Given x(n)={1,2,3,1} and z(n)={4,3,2}. (7)

b) Explain how N point DFTs of two real sequences can be found using by computing a single DFT. Illustrate with the sequences $x_1(n)$={4,3,-1,5} and $x_2(n)$={6,-4,2,5}. (8)

3 a) Find the number of real multiplications and additions involved in the computation of 64-point DFT using i) direct computation ii) FFT algorithm.Also comment on the computational advantage of FFT algorithm over the direct method. (7)

b) Using Overlap Add method, find the output of the filter with filter response h(n) when an input x(n)={1,2,2,3,4,2,2,1,1} is given. Take data block size of length L= 3 and h(n)= {2,3,4}. (8)

PART B

Answer any two full questions, each carries 15 marks.

4 a) Design a linear phase FIR low pass filter with cut off frequency of 2 kHz and sampling rate of 8 kHz with a filter length 11 using Hanning window. (10)

b) Find the filter transfer function H(z) from the analog filter with system function H(s) using Impulse Invariance method. (5)

$$H(s) = \frac{s+1}{s^2 + 0.2s + 16.01}$$

5 a) Apply frequency sampling technique to design a linear phase FIR filter of length N=7 with following specification. (10)

$$H_d(e^{j\omega}) = e^{-j\alpha\omega}; \quad 0 \le |\omega| \le 0.55\pi$$
$$= 0 \quad otherwise$$

b) Transform the prototype low pass filter with system function $H(s) = \frac{\Omega c}{s+\Omega c}$ into high pass and band pass filters. (5)

6 a) Design a Butterworth low pass digital IIR filter witha pass band edge frequency of**0.25π** with a ripple not exceeding **0.5 dB** and a minimum stop band attenuation **15dB** with a stop band edge frequency of **0.55π**. Use bilinear transformation. (10)

b) Compare the performance of FIR filter design using rectangular window and Hamming window. (5)

PART C

Answer any two full questions, each carries 20 marks.

7 a) Determine a direct form realization of the FIR filter with the following filter function using minimum number of multipliers. (4)

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

b) Draw the cascade and parallel form realisation of the filter with following transfer function (8)

$$H(z) = \frac{3(5-2z^{-1})}{\left(1+\frac{1}{2}z^{-1}\right)(3-z^{-1})}$$

c) How upsampling and downsampling by a factor of 3 affect the frequency spectrum of a signal x(n) with frequency spectrum $X(e^{j\omega})$? What is the need of low pass filter prior to downsampling? (8)

8 a) For the signal $x(n) = 0.2^n \; u(n), n \le 8$, plot the following signals (4)

(i) x(n) downsampled by 3 (ii) x(n) upsampled by 3

b) With an example illustrate the error introduced by truncation and rounding in fixed point representation of numbers. (8)

c) What is the effect of coefficient quantization in IIR filter structures? (8)

9 a) Obtain the direct form II, cascade and transposed direct form II structures for the (10)

system.

$$y(n) = -0.1y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$$

b) Explain the architecture of TMS320C67xx DSP with block diagram. (10)

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