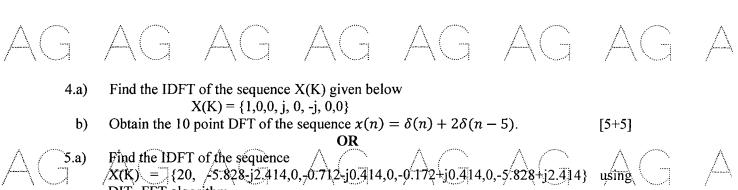
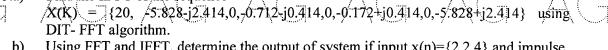
Code No: 126EK JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY HYDERABAD B. Tech III Year II Semester Examinations, April - 2018 DIGITAL SIGNAL PROCESSING (Common to ECE, EIE) Max. Marks: Time: 3 hours Note: This question paper contains two parts A and B. Part A is compulsory which carries 25 marks. Answer all questions in Part A. Part B consists of 5 Units. Answer any one full question from each unit. Each question carries 10 marks and may have a, b, c as sub questions. (25 Marks [2] Show that $\delta(n) = u(n) - u(n-1)$ 1.a) Find the Z-transform $f(n) = n^2 u(n)$ [3] b) State and prove the any three properties of DFT. [2] c) What is the basic operation of DIF algorithm? [3] d) What are the properties of Butterworth Low pass filter? e) Discuss the stability of the impulse invariant mapping technique f) Explain the effects of truncating an infinite Fourier series into a finite series. [2] g) What is the condition for the impulse response of FIR filter to satisfy for constant h) group and phase delay and for constant group delay? [3] [2] What is the need for Multirate Digital Signal Processing i) What do you mean by quantization step size? [3] j) (50 Marks) An LTI system is characterized by an impulse response 2.a) Find the step response of the system Also, evaluate the output of the system at n=±5 Consider a discrete-time system characterized by the following input-output relationship y(n) = x(n-2) - 2x(n-17). Determine whether the system is memory less, time-Invariant, linear, causal and stable. Given the difference equation $y(n) + b^2y(n-2) = 0$ for $n \ge 0$ and |b| < 1. With 3.a) initial conditions yf(-1) = 0 and y(-2) = -1, Show that [5+5]Find the Z-transform of the sequence f(n) defined below: b)





Using FFT and IFFT, determine the output of system if input $x(n) = \{2,2,4\}$ and impulse b) response $h(n) = \{1,1\}.$ [5+5]

Design a digital low pass filter using Chebyshev filter that meets the following 6.a) Specifications: Passband magnitude characteristics that is constant to within AdB for recurrences below $\omega = 0.2\pi$ and stoppand attenuation of atleast 15dB/for frequencies between $\omega = 0.3\pi$ and π . Use bilinear transformation.

b) An analog filter has the following system function. Convert this filter into a digital filter by using the impulse invariant technique: [5+5]

$$H(s) = \frac{1}{(s+0.1)^2 + 9}$$

Using a bilinear transformation, design a Butterworth filter which satisfies the following conditions:

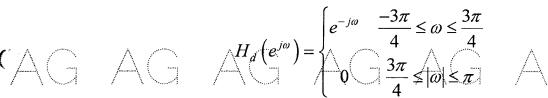
[5+5]

$$0.8 \le |H(e^{j\omega})| \le 1 \quad 0 \le \omega \le 0.2\pi$$

 $|H(e^{j\omega})| \le 0.2 \quad 0.6\pi \le \omega \le \pi$

b) Determine H(z) using impulse invariance method for the following system function:

8.a) The desired frequency response of a low pass filter is given



Find $H(e^{j\omega})$ for M=7 using a rectangular window

Explain the type II frequency sampling method of designing an FIR digital filter. [5+5] b) OR

9.a) Design a band pass filter which approximates the ideal filter with cutoff-frequencies at 0.2 rad/sec and 0.3 rad/sec. The filter order is M=7. Use the Hanning window function. Design an ideal band pass filter with a frequency response.

