N* log 2 (N)

International Islamic University Chittagong
Department of Electronic & Telecommunication Engineering
B. Sc. in ETE Semester Final Examination, Spring 2018
Course Code: ETE-4723/4703, Course Title: Digital Signal Processing
Time: 2:30 Hour Full Marks: 50

[Answer any two questions from Part-A and any three questions from Part-B. The figures in the right margin indicate full marks]

Part A

.(a) Define the following transforms and explain how they are related to each other: 6

(i) Discrete time Fourier transform (DTFT)
(ii) Discrete Fourier transform (DFT)
(iii) Fast Fourier transforms (FFT).

(b) Determine the zero-input response of the system described by the homogenous second-order difference equation 4

$$y(n)-3y(n-1)-4y(n-2)=0$$

2. How does the FFT work? Are FFTs limited to sizes that are powers of 2? Can FFTs be done on *prime* sizes? 10

3. Compute circular convolution of the following two sequences using DFT 10

$$X_1 = \{\underset{\uparrow}{2},1,2,1\}$$
$$X_2 = \{\underset{\uparrow}{1}2,3,4\}$$

PART B

4.(a) Point out the criteria to design the best filter. 3
(b) Describe the filter response characteristics for the following filters 3
i) Chebyshev filter
ii) Butterworth filter
(c) Evaluate and discuss the mathematical expression to determine the pass band ripple and stop band ripple of the filter. 4

5. Design a digital high pass Chebyshev filter with the stop band edge placed at $\omega_c = 0.1\pi$ and the pass band edge placed at $\omega_p = 0.9\pi$. The stop band attenuation should be at least -15dB and the ripple in the pass band should be located between 0 dB and -0.5dB. 10

(i) Sketch the filter specification and calculate δ_2 and ε.
(ii) Perform a pre-warping of the frequencies given in the specification in order to obtain the corresponding analogue high pass frequencies $\hat{\Omega}_p$ and $\hat{\Omega}_s$.
(iii) Use an analogue frequency transform to convert the high pass filter into a low pass filter. I.e. Calculate the low pass filter frequency Ω_s and Ω_p using the high pass

filter frequencies $\hat{\Omega}_p$ and $\hat{\Omega}_s$.

(iv) Calculate the lowest order N required, and the calculate the analogue low pass filter $H_{LP}(s)$.

(v) Use an analogue frequency transformation to convert the filter from low pass to high pass, i.e. convert $H_{LP}(s)$ into $H_{LP}(\hat{s})$

6.a) Briefly state the advantages and disadvantages of infinite impulse-response (IIR) digital filters as compared with finite impulse-response (FIR) types

b) Design a Butterworth filter with 1 dB cutoff at 3 kHz and a minimum attenuation of 50 dB at 7kHz

c) There are two main methods to transform a continuous-time IIR filter into a discrete-time IIR filter. Name the two methods.

7.(a) In Fig. 1 the frequency response of four different low pass FIR filters designed using the window method are shown. The cut off frequency is the same for all filters but they have been constructed using different windows. Clearly state the names of the windows that have been used during the design of the filters a, b, c and d. Motive your answer briefly.

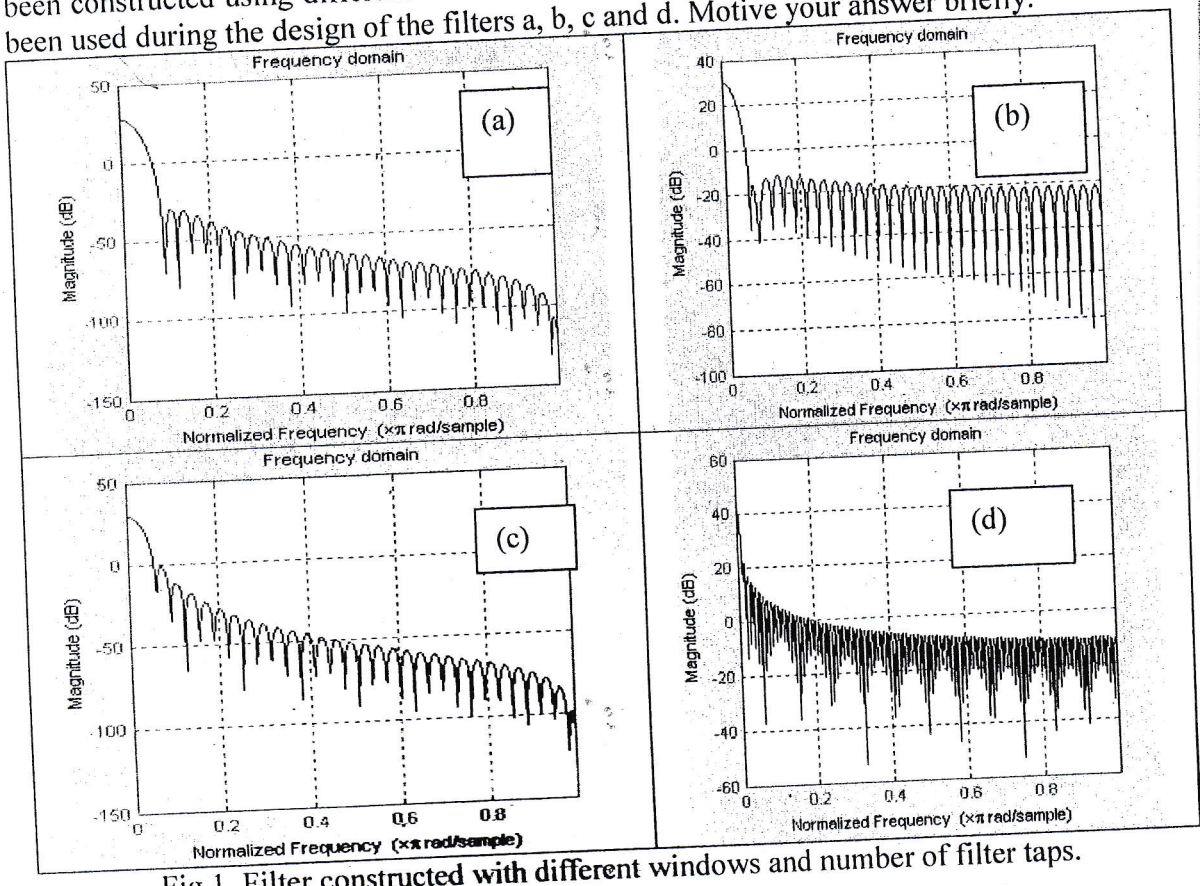

Fig.1. Filter constructed with different windows and number of filter taps.

(b) Briefly define and discuss the principle of multi rate sampling conversion. Give one example of a practical application. Analysis and discuss in detail the down-sampling (decimation) method.