Marks Scored:

KATHMANDU UNIVERSITY
End Semester Examination
June/July, 2023

Level : B.E.
Year : III

Course : ETEG 305
Semester : II

Exam Roll No. :

Time: 30 mins.

F. M. : 10

Registration No.:

Date :

SECTION "A"
[20Q. × 0.5 = 10 marks]

Encircle the most appropriate option.

1. If the sampling frequency while sampling an analog signal is F_s, the highest frequency component that can be recovered from the digital signal is….
   a. F_s b. $2F_s$ c. $\frac{F_s}{2}$ d. $\frac{F_s}{4}$

2. Which operation is essential before sampling an analog signal?
   a. Noise removal b. Band limiting
   c. Modulation d. Signal mixing

3. The Fourier transform of the signal x[n] exists only if……..
   a. no poles lie inside ROC of its z transform
   b. ROC is outside of some circle
   c. ROC includes origin
   d. ROC includes the unit circle

4. Which of the following is **NOT** frequency domain transforms for a DT signal?
   a. Laplace transform
   b. Discrete Time Fourier Series (DTFS)
   c. Discrete Time Fourier Transform (DTFT)
   d. Discrete Fourier Transform (DFT)

5. The z-transform of the signal $x[n] = \delta[n+2]$ is…………………
   a. $e^{-j2\omega}$ b. $e^{j2\omega}$
   c. $\cos 2\omega - j\sin 2\omega$ d. $\sin 2\omega - j\cos 2\omega$

6. ROC of the Z-transform of the signal $x[n] = 2u[n]$ is…………………
   a. $|z| > 1$ b. $|z| < 1$ c. $|z|>2$ d. $|z|<2$

7. Pole perturbation due to finite word length effects is large if……
   a. the order of the system is smaller
   b. the number representation follows floating point representation
   c. if the poles are nearly located to each other.
   d. if poles are located relatively far from each other.

8. FIR systems are also known as……….
   a. all pole systems b. pole zero systems c. no zero systems d. all zero systems

9. If W_4^k denote the phase factor used in computation of 4- point DFT, the symmetry property says that W_4^4 equals to
   a. W_4^2 b. W_4^{-2} c. $-W_4^2$ d. $2W_4^2$

10. The divide and conquer approach can be applied to DFT of point N if......

a. N is even
b. N is composite
c. N is prime
d. N can be expressed as power of 2

11. If $x[n] = \{1, 1, 0, 0\}$, have DFT $X[k]$ then, $x[n] = \{0, 1, 1, 0\}$ will have DFT given as.....

a. $X[n] = e^{j\frac{\pi}{2}k} X[k]$
b. $X[n] = e^{-j\frac{\pi}{2}k} X[k]$
c. $X[n] = X[(k-1)_4]$
d. $X[n] = X[(k+1)_4]$

12. If discrete samples of DTFT are to be used as frequency domain representation of a finite duration signal of length N in time domain, we must sample the DTFT with......

a. 2 N samples in one period
b. N samples in one period
c. N or less than N samples in one period
d. N or more than N samples in one period

13. Radix two FFT algorithm cannot be used to compute DFT of following points:

a. 16
b. 64
c. 140
d. 256

14. A standard second order section used in parallel implementation of IIR filters involve .... coefficients.

a. 4
b. 2
c. 3
d. 5

15. Cascade and parallel both realizations are possible for ....

a. IIR filters only
b. FIR filters only
c. IIR and FIR filters
d. LPF only

16. Which of the following is not the transformation used in converting analog filter to digital filter?

a. Bilinear transformation
b. Impulse invariance transformation
c. Matched z-transform transformation
d. Hilbert transformation

17. The z-transform of $x[n] = \left({}^{1}/_{3}\right)^{n} u[n]$ is equal to.......

a. $\frac{z}{z-3}$; $|z| < \frac{1}{3}$
b. $\frac{z}{z+3}$; $|z| > 3$
c. $\frac{z}{z-{}^{1}/_{3}}$; $|z| > \frac{1}{3}$
d. $\frac{z}{z-{}^{1}/_{3}}$; $|z| > 3$

18. To compute linear convolution between the signals of length 10 and 12, the circular convolution has to be computed with ....zero padding.

a. 0 and 2 respectively
b. 11 and 9 respectively
c. 21
d. 2 and 0 respectively

19. Which of the following IIR filter design transformation suffers from frequency domain aliasing?

a. Bilinear transformation
b. Impulse invariance transformation
c. Approximation of derivative
d. All of these

20. As compared to a rectangular window, Blackman window can significantly reduce ripple in FIR filter, however the tradeoff is .....

a. Non linearity in phase
b. Possible instability
c. Reduced gain in passband
d. a large transition bandwidth

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SECTION "B"
[5Q. × 8 = 40 marks]

Attempt *ANY FIVE* questions. Symbols and abbreviations have usual meanings. Assume suitable values for missing data.

1. a. Explain advantages and limitations of digital signal processing over analog signal processing. [3]

   b. It is known that, $u[n] \overset{Z}{\longleftrightarrow} \frac{1}{1-z^{-1}}$ with ROC: $|z| > 1$. Using this Z-transform pair and differentiation of z-transform property, obtain the Z-transform and ROC of the following signal: $x[n] = n^2u[n]$. [3]

   c. Define z-transform and explain its uses in digital signals and systems. [2]

2. a. The Z-transform of a causal signal $x[n]$ is given as: $(z) = \frac{5z^3}{(z+1)^2(z-1)}$. Using partial fraction expansion and look up table method, find the expression for the signal $x[n]$. [3]

   b. Consider the signals: $p[n] = [2, -0.5, 1]$ and $q[n] = [2, -0.5, 1]$
   If $P[k]$ and $Q[k]$ are three-point DFTs of $p[n]$ and $q[n]$, determine the sequence $r[n]$ such that its three-point DFT $R[k] = P[k]Q[k]$. [2]

   c. How linear shifting is related to circular shifting? State and prove circular convolution property of DFT. [1+2]

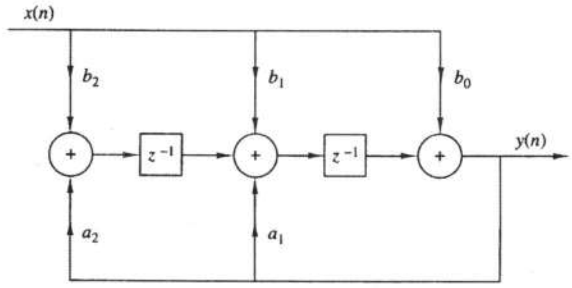
3. a. Derive the expression for the radix 2 DIT FFT algorithm and hence explain the basic butterfly for DIT algorithm. [3]

   b. Find 4 point DFT of the following signal using radix-2 DIF algorithm. Use butterfly diagram for the computation. [3]
   $x[n] = \cos\frac{\pi}{2}n \qquad 0 \le n \le 3$

   c. Explain the idea on using DFT for filtering long data sequences. [2]

4. a. Realize the following IIR system function using direct form-I and direct form-II and parallel structures. [3]
$$y[n] = y[n-1] - \frac{1}{2}y[n-2] + x[n] - x[n-1] + x[n-2]$$

b. Find the system function of the system shown below: [3]



c. What do you understand by FFT? Why are FFT very important in DSP? [2]

5. a. Explain the phrase '*optimum equiripple filter design*'. State 'Remez exchange algorithm' followed in optimum equiripple filter design. [3]

b. A FIR low pass filter is required to be designed with the desired frequency response $H_d(\omega)$ given below. Obtain filter coefficients if the window function is defined as w[n]. [5]

$$H_d(\omega) = \begin{cases} 1 & \text{for,} \quad |\omega| \le \frac{\pi}{3} \\ 0 \text{ for,} & \frac{\pi}{3} \le |\omega| \le \pi \end{cases}$$

$$w[n] = \begin{cases} 0.54 - 0.46\cos\frac{2\pi n}{M-1}, & 0 \le n \le 7 \\ 0, & otherwise \end{cases}$$

Does the filter have a linear phase? Justify your answer.

6. a. Derive the transformation equation for impulse invariance method of IIR filter design. Discuss the nature of mapping from analog to digital frequency. [4]

b. A digital IIR Butterworth filter is to be designed for following digital specifications:
$\alpha_p = 3\,dB, \alpha_s = 30\,dB, \Omega_p = 0.162\,rad, \Omega_s = 1.63\,rad$
Determine the analog specifications and hence the order and transfer function of the normalized analog filter. Also compute the cut off frequency and apply the bilinear transformation to find the system function of the de-normalized digital filter. Assume the sampling frequency to be 8 KHz. [4]

Some useful relationships:
Quotient rule:

$$\frac{dy}{dx} = \frac{v\frac{du}{dx} - u\frac{dv}{dx}}{v^2}$$

Partial fraction expansion for roots of multiplicity:

$$\frac{A_{1k}}{z-p_k} + \frac{A_{2k}}{(z-p_k)^2} + \cdots + \frac{A_{mk}}{(z-p_k)^m}$$

$$A_{mk} = \frac{X(z)}{z}(z-p_k)^m\Big|_{z=p_k} \qquad A_{m-1k} = \frac{d}{dz}\left[\frac{X(z)}{z}(z-p_k)^m\right]_{z=p_k}$$

For bilinear transformation digital frequency Ω is related to analog frequency ω as:

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$\omega = \frac{2}{T} tan \frac{\Omega}{2}$ And the transformation is: $s = \frac{2}{T} \frac{1-z^{-1}}{1+z^{-1}}$

Design equations for analog Butterworth filters:

Minimum order required: $N = \frac{1}{2} \frac{log\left[\left\{\frac{1}{A_2{}^2}-1\right\}/\left\{\frac{1}{A_1{}^2}-1\right\}\right]}{log\frac{\omega_2}{\omega_1}}$

Where A_1 and ω_1 are minimum pass band gain and pass band edge frequency.

A_2andω_2 are maximum stop band gain and stop band edge frequency.

OR,

$$N = \frac{\log_{10}\left(\frac{10^{\alpha_s/10}-1}{10^{\alpha_p/10}-1}\right)}{2\log_{10}\left(\frac{\omega_s}{\omega_p}\right)}$$

Cutoff frequency:

$$\omega_c = \frac{\omega_1}{\left[\frac{1}{A_1{}^2}\right]^{1/2N}}$$

OR:

$$\omega_c = \frac{\omega_s}{\left[10^{\alpha_s/10}-1\right]^{1/2N}}$$

Normalized BW denominator polynomials:

n (order)	Normalized Denominator Polynomials in Factored Form
1	$(1+s)$
2	$(1+1.414s+s^2)$
3	$(1+s)(1+s+s^2)$
4	$(1+0.765s+s^2)(1+1.848s+s^2)$
5	$(1+s)(1+0.618s+s^2)(1+1.618s+s^2)$
6	$(1+0.518s+s^2)(1+1.414s+s^2)(1+1.932s+s^2)$
7	$(1+s)(1+0.445s+s^2)(1+1.247s+s^2)(1+1.802s+s^2)$
8	$(1+0.390s+s^2)(1+1.111s+s^2)(1+1.663s+s^2)(1+1.962s+s^2)$
9	$(1+s)(1+0.347s+s^2)(1+s+s^2)(1+1.532s+s^2)(1+1.879s+s^2)$
10	$(1+0.313s+s^2)(1+0.908s+s^2)(1+1.414s+s^2)(1+1.782s+s^2)(1+1.975s+s^2)$

Analog Butterworth Low pass transfer functions:

Order	Transfer function
2	$H_a(s) = \left(\frac{\omega_c{}^2}{s^2 + 0.707\omega_c s + \omega_c{}^2}\right)$
3	$H_a(s) = \left(\frac{\omega_c}{s + \omega_c}\right)\left(\frac{\omega_c{}^2}{s^2 + \omega_c s + \omega_c{}^2}\right)$
4	$H_a(s) = \left(\frac{\omega_c{}^2}{s^2 + 0.765\,\omega_c s + \omega_c{}^2}\right)\left(\frac{\omega_c{}^2}{s^2 + 1.848\,\omega_c s + \omega_c{}^2}\right)$