Question Paper Details					
Course	**Stream**	**Semester**	**Subject**	**Paper Code**	**Chapter**
B.TECH	**ECE**	**VI**	**Digital Siganl Processing**	**EC-602**	**1 Discrete time signals & LTI Systems**

Paper Setter Details			
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MCQ [Type-1] **[Maximum marks: 1]**

1)The system $y(n) = x(n) + x(n-1)$ is
a) linear time-invariant
b) non-linear time-invariant
c) linear time-variant
d) none of these.

2) The digital system in $y(n) = x(n\,2)$ is
a) non-linear and causal b) linear and causal
c) linear and non-causal d) non-linear and non-causal.

3) The system described by $y[n] = nx[n]$ is
a) Linear, time varying and stable
b) Non-linear, time invariant and unstable.
c) Non-linear, time varying and stable.
d) Linear, time varying and unstable.

4) For an analog signal $x(t) = 3\cos(50\pi t) + 10\sin(300\pi t)$.
The Nyquist sampling rate is
a) 150 Hz.
b) 300 Hz.
c) 25 Hz.

5) A discrete time LTI system is known as causal system if its,
a) $h(n) = 0$, $n < 0$ b) $h(n) = 0$, $n > 0$
c) $h(n)$ is positive, $n < 0$ d) none of these.

6) A system having impulse response $h(t)$ will be BIBO stable if

a) $|h(t)|$ dt <1

b) $h(t)$ dt <0

c) $|h(t)|$ dt >0

d) $|h(t)|$ dt = 0

7) A signal is a power signal if

a) $E<\infty$, $P=0$ b) $P<\infty$, $E = 0$

c) $P<\infty$, $E = \infty$ d) $P = \infty$, $E = 0$

8) If $h(n) = \{ 1, 0, 1 \}$ and $y(n) = x(n)*h(n) = \{ 1, -2, 4, -2, 3 \}$, then $x(n)$ is

a) $x(n) = \{ 1, -2, 1 \}$ b) $x(n) = \{ 1, -1, 3 \}$

c) $x(n) = \{ 1, -2, 3 \}$ d) $x(n) = \{ 1, -2, -3 \}$.

9)If $F s$ is the minimum sampling rate, $F max$ is the highest frequency available in the analog signal, then at Nyquist rate

a) $F s = 2 F max$ b) $F s = 0.5 F m x$

c) $F s = F max$ d) $F s < F max$.

10) The discrete –time system y(n)= 3 s(n)+5 is

a) linear, causal system b) Nonlinear ,causal system

c) linear non causal system d) Nonlinear non causal system

Short Question [Type-2] [Maximum marks: 2]

Q1. Find out the fundamental period of the signal s(t)= 2 cos (10t+1)-sin (4t-1).

Q2**.**Check the causality of the following systems

i) y(t)= s(t) cos (t+1) ii) y(n) = k[s(n+1)-s(n)]

Q3**.**Check Weather the signal is periodic or not S(t)=sin20πt+sin 5πt.

Q4**.** Input –output relationship for some discrete time system is given as

y(n)= s(n-2)- 2 s(n-8)check above system for memory less or with memory system.

Q5**.** Verify whether signal s(t)= Ae^{-an} u(n), A>0 is an energy signal or not.

Q6.Test whether the following signal is periodic or not and if periodic then find period of signal: $X[n] = \cos(n\pi/5) + \sin(n\pi/6)$

Q7.Determine the output y(t) of a continuous time LTI system with impulse response $h(t)= e^{-An} u(n)$, A>0 to input $s(t)= e^{-Bn} u(n)$, B >0

Q8.What is Nyquist sampling theorem? How reconstruction of signal is done?

Q9. Specify the Nyquist rate for following signals
i) $x1(t)= \cos(2\pi * 10^{3} n)$
ii) $x2(t)= \cos(2\pi * 10^{3} n) + \cos(6\pi * 10^{3} n)$

Q10. What is convolution sum? Find the convolution of sequences $s_1(n)=\{1,-2,1\}$ and $s_2(n)= \{1,1,1\}$

Subjective question [Type-3] **[Maximum marks: 3]**

Q1.Define signals . Give the classification of signals.

Q2.With suitable examples distinguish a deterministic signal from a random signal.

Q3.What is unit step function and unit impulse function?

Q4. Describe the procedure used to determine whether the sum of two periodic signals is periodic or not.

Q5.Distinguish between systems with memory from memory less systems. Give example of each system.

Q6. What do you mean by differential equations which describe continuous time systems? Give an example.

Q7. What are the building blocks or elements of a system?

Q8.Give the condition of BIBO stability of continuous –time and discrete time LTI systems in terms of impulse response.

Q9. Discuss the following properties of continuous –time and discrete –time LTI systems

a) Commutative property b) Distributive property

c) Associative property

Q10.Express the output y(n) of a linear time invariant system with impulse response h(n) in terms of its step response s(n)= hn)* u(n and the input x(n).

Broad Question [Type-4] **[Maximum marks: 5]**

1. Determine the values of power and energy of the following signals.Find whether the signals are power , energy or neither energy nor power signals.

2.Determine if the system described by the following input – output equations is linear or non-linear.

a) $y(n)= x(n)+ 1/x(n-1)$ b) $y(n) = nx(n)$

3. Determine if the following systems are time –invariant or time variant.

a) $y(n) = x(n)+ x(n-1)$ b) $y(n)= x(-n)$

4. State and explain sampling theorem for continuous –time signals.

5. What is aliasing phenomenon? How can aliasing phenomena eliminated.

6.What do you mean by impulse train sampling of a continuous –time signal?

7. What are interpolation techniques for the reconstruction of a continuous –time signal from its samples.

8 . Specify the Nyquist rate for following signals

i) $x1(t)= \cos(2\pi * 10^3 n)$

ii) $x2(t)= \cos(2\pi * 10^3 n)+ \cos(6\pi * 10^3 n)$

9. Determine the convolution sum of two sequences $x(n)= \{3,2,1,2\}$, $h(n)= \{1,2,1,2\}$.

10.Find the cross – correlation of two finite length sequences $x(n)= \{1,2,1,1\}$ $y(n)= \{1,1,2,1\}$

Question Paper Details					
Course	**Stream**	**Semester**	**Subject**	**Paper Code**	**Chapter**
B.TECH	**ECE**	**VI**	**Digital signal processing**	**EC-602**	**2 Z-Transform Discrete , Fourier Transform, Fast Fourier Transform**

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MCQ [Type-1] **[Maximum marks: 1]**

1) Z-transform of $u[n-1]$ is
a) 1/(1− z −1)
b) 1/(1+ z −1)
c) 1/ (z − z 1)
d) 1 + z –1.

2) Z-transform of the sequence x(n)= a^n u(n)
a) 1/ (1-az)
b) 1/ (1- az-1)
c) –z / (z-a)
d) 1/ (z-a)

3. The relationship between s –plan and z-plane is given by

a) s= e^{zt}
b) z= e^{sT}

c) z= log (sT)
d) s= log (zT)

4. Which of the following is/ are not property of Region of convergence

a) ROC must be connected region

b) ROC contain poles

c) ROC of an LTI system contain the unit circle

d) ROC is a ring or disk in the z-plane centered at the origin.

5)Stability region of *z*-transform is
a) within unit *z*-circle
b) outside the unit *z*-circle

c) on the unit *z*-circle only
d) entire *z*-plane.

6) The Z-transform of δ (*n*) is
a) 0 b) Z –1
c) 1/(1 – Z –1) d) 1.

7. Why 16 point DFT is preferable than 4 point DFT ?
a) Resolution of spectrum is poor for 4 point DFT than 16 point DFT
b) Resolution of spectrum is high but not reliable in 4 point DFT
c) Calculation of 4 point DFT is more complex
d) None of these.

8. If *x* (*n*) = { 1, 0, 0, 1 }, the DFT value *X* (0) is
a) 2 b) 1 + *j*
c) 0 d) 1 – *j*.

9. Overlap save method is used to find
a) circular convolution b) linear convolution
c) DFT d) *Z* transform.

10. The direct evaluation of DFT require
a) *N* 2 multiplications and *N* 2 additions.
b) *N* 2 multiplications and *N* (*N* – 1) additions.
c) *N* (*N* – 1) multiplica ions and *N* 2 additions.
d) *N* (*N* – 1) multiplications and *N* (*N* – 1) additions

11. The 4-point DFT of {1,1,0,0}

a) {2,0,2,0} b) {1,2,-1,1,2,1}
c) {-1, 2, 1,2,1,2} d) {1,2,1,1,2,-1}
12.The N-point DFT of x(n)= δ(n-n_0) is
a) 1 b) e –j2πkn_0 /n
c) e-j2πn_0 /N d) e-jπkn_0/N

13.When the DFT of a sequence x(n)= is imaginary
a) x(n)is real and even b) x(n)is imaginary and odd
c) x(n)is real and odd d) x(n)is real

14. Between circular convolution and linear convolution
a) length of linear convolution is greater
b) length of circular convolution is greater
c) lengths of both are same
d) none of these.

15..FFT is a modification of DFT in terms of

a) computational speed b) noise

c) linearity d) none of these.

16The twiddle factor satisfies

a) $w^k{}_N = w^k{}_{N/2}$ b) $w^{k+N/2}{}_N = w^k{}_N$

c) $w^{k+N}{}_N = - w^k{}_N$ d) $w^{k+ N/2}{}_N = - w^k{}_N$

17. The number of stages in flow graph is

a) $\log_2{}^N$ b) 2^N

c) $\log_{10}{}^N$ d) 10^N

18. The total number of real additional required for direct evolution of the DFT is

a) N(4 N-1) b) 4 N^2

c) N (4N -2) d) N (4 N-3)

19. In DIT –FFT algorithm if M is number of stages and m represent s stage index then the number of sections of butterflies in each stage is

a) 2^{M+m} b) 2^{M-m}

c) $2^{M-m/2}$ d) 2^{M-2m}

20.The number complex multiplications required to perform DFT using DIT algorithm is

a) $N/2 \log_{10}{}^N$ b) $N \log_{10}{}^N$

c) $N/2 \log_2{}^N$ d) $N \log_2{}^N$

Short Question [Type-2] **[Maximum marks: 2]**

1. Find the Z transform of signal s(n)= - A^n u(-n-1)

2. Find the inverse Z transform of
$X(z) = z/(2-3z^{-1}+z^{-2})$; ROC : $|Z| > 1$ by long division method.

3. Find the auto – correlation sequence of the signal s1(n)= A^n u(n), $-1< A < 1$.

4. For the sequence $x(n)$ = { 1, 1, 0, – 1, – 1, 0, 0 }, determine the 8-point DFT.

5. Find the IDFT of the sequence X(k)= {5,0,1,-j,0,1,1+j,0}

6. Perform the circular convolution of the following sequences $x_1(n)$= { 1, 1,2,1}, $x_2\{n\}$= {1,2,3,4}

7.Determine linear convolution and circular convolution of sequences { 1, 2, 3, –1 } and { 3, 4, 2, –3 }.

8. Evaluate 8-point for the sequence using DIT-FFT
algorithm :
$x(n) = 1, -3 \leq n \leq 3$
$= 0$, elsewhere.
9. Compute a DFT of the following sequence using DIT algorithm
X(n)= {0,1,2,3,4,5,6,7}
10.Compute an IDFT of the sequences using DIF algorithim

X (k)= { 1,1+ j , 1-J 2, 1,0, 1+ 2j , 1+ j}

Subjective question [Type-3] **[Maximum marks: 3]**

1. What do you mean by circular convolution?

2.What is the difference between circular convolution and linear convolution?

3. Show that the circular convolution is commutative.

4.What do you mean by filtering long duration sequences.

5.Explain overlap – save method and overlap add method

6. Determine and explain the relationship between *s*-plane and *z*-plane.

7. Describe correlation and multiplication using-transform.

8. Explain aliasing error and overlapping.

9.Explain Twiddle factor. Discuss the relationship of DFT with Z-transform

10. Discuss 8-point Radix-2 decimation-in-time FFT algorithms

Broad Question [Type-4] **[Maximum marks: 5]**

1. Determine the inverse z- transform of S(z)= 5z/ (z-1)(z-2) using the contour integration method.
2.Solve the following difference equation by using z- transform method
s(n+2) + 3 s(n+1) + 2 s(n)=0
3.Given two discrete –time signals $s_1(n)= (1/3)^n\, u(n)$ and $s_2(n)= (1/5)\, u(n)$. Find S(z) by using convolution property of z-transform.
4. Find the system function and impulse response of the system described by the difference equation
$Y(n) = X(n) + 2X(n-1) - 4X(n-2) + X(n-3)$
5. Find the linear convolution using circular convolution
for the two sequences :
$x(n) = \{1, 2, -1, 2, 3, -2, -3, -1, 1, 1, 2, -1\}$
$h(n) = \{1, 2\}$
6.Evaluate 8-point for the sequence using DIT-FFT

algorithm :

$x(n) = 1, -3 \leq n \leq 3$

$\quad = 0$, elsewhere.

7. Compute the circular convolution of the following two sequences :

$$x1(n) = \{2, \underset{\uparrow}{1}, 2, 1\}$$

$$x2(n) = \{1, 2, 3, 4\}$$

8.The impulse respons of one LTI system is $h(n) = \{1, 2, 1, -1\}$. Determine the response of the system to the input signal, $x(n) = \{1, 2, 3, 1\}$

9.Compute the 8 point DFT of the following sequence :
x(n)= {1/2, ½, ½, ½, 0,0,0,0} Use in-place radix-2 decimation in time FFT algorithm.

10 What is a butterfly regarding FFT ? Point out the properties of ROC of Z-transform

11.Find the IDFT of the sequence $X(k)= \{6, -2+j^{2}, -2,-2,-j^{2}\}$

12. What is DFT and FFT ? Describe time decimation of FFT computation

13.Prove that $X_3(k) = X_1(k)\, X_2(k)$

14.Write short notes on any *three* of the following

b) DIF algorithm

c) Utility of FFT and DFT

15. What do you mean by filtering long duration sequences. Explain overlap – save method and overlap add method.

Question Paper Details					
Course	**Stream**	**Semester**	**Subject**	**Paper Code**	**Chapter**
B.TECH	**ECE**	**VI**	**Digital signal processing**	**EC-602**	**3 Filter Design**

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MCQ [Type-1] **[Maximum marks: 1]**

1. For FIR filter the poles be always at

a) Origin b) Outside the unit circle

c) On the unit circle d) At infinity

2. Which of the following windows gives a low pass filter with high transition band

a) Rectangular window b) Hamming window

c) Triangular window d) Blackmann window

3.The frequency response of rectangular window is

a) sin(w N/2) / sin (w/2) b) sin(w N/2) / (w/2)

c) sin(w /2) / sin (wN/2) d) sin(w N/2) / (w/2)

4) The poles of the Chebyshev filter lie
a) on an e lipse b) on a circle
c) on par bola d) on a rectangle.

5. The main to be width of rectangular window is

a) π/N b) 2 π/N

c) 4π /N d) 8 π/N

Short Question [Type-2] **[Maximum marks: 2]**

1.Design alow pass filter using rectangular window by taking samples of w(n)= and with cut –off frequency of 1.2 rad/ sec.

2. Obtain the cascade realization of system function H (z) = $(1+ 2 z^{-1} – z^{-2})()\ 1+ z^{-1} – z^{-2})$

3.Determine the direct form realization of system function H(z)= $1+ 2 z^{-1} – 3 z^{-2} + 5 z^{-4}$

4. Design the filter using Fourier series method. Take N=7 low pass filter H(ejw) = 1 for $0\leq |w| \leq \pi/6$
0 otherwise

5. Determine the direct form of realization of a linear phase FIR filter specified by the impulse response $h (n) = \{ 2, 4, 6, 6, 4, 2 \}$.

Subjective question [Type-3] **[Maximum marks: 3]**

1. Describe windowing. Explain Gibbs oscillation in this context.

2. Describe Butterworth IIR filter using impulse invariant method.

3.What is all pass system? Draw its typical pole-zero plot?

4.What is bit reversal?

5.Differentiate between FIR and IIR filters.What is windowing?

6. Explain the function of rectangular and Hamming windows for filter realization.

7. What are the difference and similarities between DIT and DIF algorithms ?

8.Write short notes on Butterworth filter

9.What do you mean by Bilinear transformation

10. Distinguish between recursive realization and non- recursive realization.

11. Explain the function of rectangular and Hamming windows for filter realization.

12. What are the difference and similarities between DIT and DIF algorithms ?

Broad Question [Type-4] **[Maximum marks: 5]**

1.Consider a causal IIR system with the system function
H(z)= $1+2z^{-1}+3z^{-1}+2\ z^{-3}/1+0.9z^{-1}-0.8z^{-2}+0.5z^{-3}$
Determine the equivalent lattice-ladder structure.

2. Find the transposed direct form II realization of the system described by the difference equation

$y(n) = 0.5y(n-1) – 0.25y(n-2) + x(n) – 2x(n-1) + x(n-2)$

4.How IIR filter Designing can be done by the use of following methods. Discuss each methods-
(i) Approximation of Derivatives Method.
(ii) Impulse Invariance Method.

5.Determine the cascade and parallel realizations for the system described by the system function
$H(z)=10\,(1- \frac{1}{2} z^{-1})(1-2/3z^{-1})(1+2z^{-1}) / (1-3/4z^{-1})(1-1/8z^{-1})\{1-(1/2+j/2)z-1\}\{1-(1/2-j/2)z^{-1}\}$
|
6.Develop cascade & parallel realization structure of following transfer function:
$H(z)=\{z/6+5/24+5/24z +1/24z^2\}/\{1-1/2z +1/4z^2\}$

7. A low pass filter is to be designed with following desired frequency response
$H_d(e^{jw}) = e^{-j2w}$, $\pi /4 \le w \le \pi /4$
$= 0$, $\pi /4 < |w| < \pi$
8.Determine the filter coefficient $h_d[n]$ if the window function is defined as
$W[n] = 1; 0 \le n \le 4$
$= 0$; Otherwise
9.Find the order and cut-off frequency of a digital filter with the following specifications :
$0.89 \le |H (e^{jw}) | \le 1, 0 \le w \le 0.4\pi$,$|H (e^{jw})| \le 0.18, 0.6\pi \le w \le \pi$ Use impulse invariance method.
10.Determine the direct form II and transposed direct form II for the given system y(n)= ½ y (n-1)_ ¼ y(n-2) + x(n)+ x(n-1)

11. Design a Chebyshev lowpass filter with the specification α_p= 1 Db ripple in the pass band $0<w< 0.2$, α_s =15 Db ripple in the stop band $0.3\pi < w < \pi$,using a) bilinear transformation b) Impulse invariance.

12.Given the specification α_p =3Db ,α_s =16 Db , f_p =1 KHz and f_s = 2KHz . Determine the order of the filter using Chebyshev approximation .Find H(s).

13.For the given specificationα_p =3 Db ,α_s = 15 Db , Ω_p =1000rad/sec and Ω_s =500 rad/sec design a high pass filter.

14.Determine the filter coefficient $h_d[n]$ if the window function is defined as
$W[n] = 1; 0 \le n \le 4$
$= 0$; Otherwise
15. Convert the analog filter with the system function $G(s) = s+0.1 / (s+0.1)^2 +16$ into a digital filter using bilinear transformation .The f digital filter should have a resonant frequency $w_r = \pi/4$

Question Paper Details					
Course	**Stream**	**Semester**	**Subject**	**Paper Code**	**Chapter**
B.TECH	**ECE**	**VI**	**Digital Signal Processing**	**EC-602**	*4* Digital Signal Processor, FPGA

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MCQ [Type-1]

Short Question [Type-2]

1. What is pipelining?

2. What is pieline depth?

What are the application of PDSPs?

3. Give some examples for fixed point DSP.

4. What are the factor that influence selection of DSPs.

5. List the various registers used with ARAU.

Subjective question [Type-3]

1.What are the classification of digital signal processor?

2. What are different buses of TMS 320 C 5x and their functions?

3. What are the elements that the control processing unit of ‘C5x ?

4. What is the advantage of Harvard architecture of TMS 320 series?

5. List the onchip peripherals in ‘C5x.

Broad Question [Type-4]

1.Explain the Architecture of digital signal processor.

2.Draw a neat sketch of the functional block diagram of TM 320C50 and mark various blocks.

3.Write the addressing modes of TM320C50and explain briefly each of them.

4. Describe the multiplier and accumulator unit in DSP processors.

5.What are the advantages of DSP processors over conventional microprocessor.