

# NADAR SARASWATHI COLLEGE OF ENGINEERING AND TECHNOLOGY, THENI.

<b>Course/Branch</b> : B.E / ECE	<b>Year / Semester</b> : III / V	Format No.	NAC/TLP-07a.13
<b>Subject Code</b> : EC8553	<b>Subject Name</b> : Discrete Time Signal Processing	Rev. No.	02
<b>Unit No</b> : 1	<b>Unit Name</b> : Discrete Fourier Transform	Date	30.09.2020

## OBJECTIVE TYPE QUESTION BANK

S. No.	Objective Questions (MCQ / True or False / Fill up with Choices )	BTL
1.	<p>If <math>x(n)</math> is a discrete-time signal, then the value of <math>x(n)</math> at non integer value of 'n' is:</p> <p>a) Zero b) Positive c) Negative d) Not defined</p> <p><b>ANSWER: d</b> Explanation: For a discrete time signal, the value of <math>x(n)</math> exists only at integral values of n. So, for a non- integer value of 'n' the value of <math>x(n)</math> does not exist.</p>	L2
2.	<p>The interface between an analog signal and a digital processor is</p> <p>a. D/A converter b. A/D converter c. Modulator d. Demodulator</p> <p><b>ANSWER: (b) A/D converter</b></p>	L2
3.	<p>Which of the following signal is the example for deterministic signal</p> <p>a. Step signal b. Ramp signal c. Exponential d. All of the above</p> <p><b>ANSWER: D</b></p>	L4
4.	<p>For energy signals the energy will be finite and the average power will be a</p> <p>A infinite B finite C 0 D cannot be defined</p> <p><b>ANSWER: C</b></p>	L5
5.	<p>The zero input response or natural response is mainly due to</p> <p>a. initial stored energy in the system b. initial conditions in the system c. specific input signal d. both a and b</p> <p><b>ANSWER: A</b></p>	L1
6.	<p>The evaluation of correlation involves</p> <p>A shifting, rotating and summation</p>	L1

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	<p>B shifting multiplication and the summation</p> <p>C change of Index, folding and the summation</p> <p>D change of Index, folding shifting and multiplication</p> <p><b>ANSWER: B</b></p>	
7.	<p>Sectioned convolution is performed if one of the sequence is very much larger than the other in order to overcome,</p> <p>A long delay in getting output be larger memory space requirement</p>	L2
8.	<p>The Nyquist theorem for sampling</p> <ol style="list-style-type: none"> <li>1) Relates the conditions in time domain and frequency domain</li> <li>2) Helps in quantization</li> <li>3) Limits the bandwidth requirement</li> <li>4) Gives the spectrum of the signal</li> </ol> <p>a. 1, 2 and 3 are correct</p> <p>b. 1 and 2 are correct</p> <p>c. 1 and 3 are correct</p> <p>d. All the four are correct</p> <p><b>ANSWER: (c) 1 and 3 are correct</b></p>	L1
9.	<p>Roll-off factor is</p> <ol style="list-style-type: none"> <li>a. The bandwidth occupied beyond the Nyquist Bandwidth of the filter</li> <li>b. The performance of the filter or device</li> <li>c. Aliasing effect</li> <li>d. None of the above</li> </ol> <p><b>ANSWER: (a) The bandwidth occupied beyond the Nyquist Bandwidth of the filter</b></p>	L1
10.	<p>A discrete time signal may be</p> <ol style="list-style-type: none"> <li>1) Samples of a continuous signal</li> <li>2) A time series which is a domain of integers</li> <li>3) Time series of sequence of quantities</li> <li>4) Amplitude modulated wave</li> </ol> <p>a. 1, 2 and 3 are correct</p> <p>b. 1 and 2 are correct</p>	L3

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	<p>c. 1 and 3 are correct d. All the four are correct</p> <p><b>ANSWER: (a) 1, 2 and 3 are correct</b></p>	
11.	<p>The discrete time function defined as <math>u(n)=n</math> for <math>n \geq 0</math>; <math>=0</math> for <math>n &lt; 0</math> is an:</p> <p>a) Unit sample signal b) Unit step signal c) Unit ramp signal d) None of the mentioned</p> <p><b>ANSWER: C</b></p> <p>Explanation: When we plot the graph for the given function, we get a straight line passing through origin with a unit positive slope. So, the function is called as unit ramp signal.</p>	L2
12.	<p>A real valued signal <math>x(n)</math> is called as anti-symmetric if:</p> <p>a) <math>x(n)=x(-n)</math> b) <math>x(n)=-x(-n)</math> c) <math>x(n)=-x(n)</math> d) None of the mentioned</p> <p><b>ANSWER: B</b></p> <p>Explanation: The definition of anti-symmetric signal, the signal <math>x(n)</math> should be symmetric over origin. So, for the signal <math>x(n)</math> to be symmetric, it should satisfy the condition <math>x(n)=-x(-n)</math>.</p>	L3
13.	<p>The odd part of a signal <math>x(t)</math> is:</p> <p>a) <math>x(t)+x(-t)</math> b) <math>x(t)-x(-t)</math> c) <math>(1/2)*(x(t)+x(-t))</math> d) <math>(1/2)*(x(t)-x(-t))</math></p> <p><b>ANSWER: D</b></p> <p>Explanation: Let <math>x(t)=x_e(t)+x_o(t) \Rightarrow x(-t)=x_e(-t)-x_o(-t)</math> By subtracting the above two equations, we get <math>x_o(t)=(1/2)*(x(t)-x(-t))</math></p>	L1
14.	<p>The function given by the equation <math>x(n)=1</math>, for <math>n=0</math>; <math>=0</math>, for <math>n</math> is not equal to 0 is a:</p> <p>a) Step function b) Ramp function c) Triangular function d) Impulse function</p>	L1

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	<p><b>ANSWER: D</b></p> <p>Explanation: According to the definition of the impulse function, it is defined only at <math>n=0</math> and is not defined elsewhere which is as per the signal given.</p>	
15.	<p>The process of converting discrete-time continuous valued signal into discrete-time discrete valued(digital) signal is known as:</p> <p>a) Sampling b) Quantization c) Coding d) None of the mentioned</p> <p><b>ANSWER: B</b></p> <p>Explanation: In this process, the value of each signal sample is represented by a value selected from a finite set of possible values. Hence this process is known as 'quantization'</p>	L2
16.	<p>The difference between the unquantized <math>x(n)</math> and quantized <math>x_q(n)</math> is known as:</p> <p>a) Quantization coefficient b) Quantization ratio c) Quantization factor d) Quantization error</p> <p><b>ANSWER: D</b></p> <p>Explanation: Quantization error is the difference in the signal obtained after sampling i.e., <math>x(n)</math> and the signal obtained after quantization i.e., <math>x_q(n)</math> at any instant of time.</p>	L3
17.	<p>What is output signal when a signal <math>x(t)=\cos(2*\pi*40*t)</math> is sampled with a sampling frequency of 20Hz?</p> <p>a) <math>\cos(\pi*n)</math> b) <math>\cos(2*\pi*n)</math> c) <math>\cos(4*\pi*n)</math> d) <math>\cos(8*\pi*n)</math></p> <p><b>ANSWER: C</b></p>	L1

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	<p>Explanation: From the question <math>F=40\text{Hz}</math>, <math>F_s=20\text{Hz}</math></p> <p><math>\Rightarrow f=F/F_s \Rightarrow f=40/20 \Rightarrow f=2\text{Hz} \Rightarrow x(n)=\cos(4*\pi*n)</math></p>	
18.	<p>Which of the following is the odd component of the signal <math>x(t)=e(jt)</math>?</p> <p>a) <math>\cos t</math></p> <p>b) <math>j*\sin t</math></p> <p>c) <math>j*\cos t</math></p> <p>d) <math>\sin t</math></p> <p><b>ANSWER: B</b></p>	L5
19.	<p>Let <math>x_1(t)</math> and <math>x_2(t)</math> be periodic signals with fundamental periods <math>T_1</math> and <math>T_2</math> respectively. Which of the following must be a rational number for <math>x(t)=x_1(t)+x_2(t)</math> to be periodic?</p> <p>a) <math>T_1+T_2</math></p> <p>b) <math>T_1-T_2</math></p> <p>c) <math>T_1/T_2</math></p> <p>d) <math>T_1*T_2</math></p> <p><b>ANSWER: C</b></p> <p>Explanation: Let <math>T</math> be the period of the signal <math>x(t) \Rightarrow x(t+T)=x_1(t+mT_1)+x_2(t+nT_2)</math></p> <p>Thus, we must have <math>mT_1=nT_2=T \Rightarrow (T_1/T_2)=(k/m)=</math> a rational number</p>	L1
20.	<p>Let <math>x_1(t)</math> and <math>x_2(t)</math> be periodic signals with fundamental periods <math>T_1</math> and <math>T_2</math> respectively. Then the fundamental period of <math>x(t)=x_1(t)+x_2(t)</math> is:</p> <p>a) LCM of <math>T_1</math> and <math>T_2</math></p> <p>b) HCF of <math>T_1</math> and <math>T_2</math></p> <p>c) Product of <math>T_1</math> and <math>T_2</math></p> <p>d) Ratio of <math>T_1</math> to <math>T_2</math></p> <p><b>ANSWER: A</b></p>	L1

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21.	<p>Which of the following justifies the linearity property of z-transform?<math>[x(n) \rightarrow X(z)]</math> a) <math>x(n)+y(n) \rightarrow X(z)+Y(z)</math></p> <p>b) <math>x(n)y(n) \rightarrow X(z)Y(z)</math></p> <p>c) <math>x(n)+y(n) \rightarrow X(z)Y(z)</math></p> <p>d) <math>x(n)y(n) \rightarrow X(z)+Y(z)</math></p> <p><b>ANSWER: B</b></p> <p>Explanation: According to the linearity property of z-transform, if <math>X(z)</math> and <math>Y(z)</math> are the z-transforms of <math>x(n)</math> and <math>y(n)</math> respectively then, the z-transform of <math>x(n)+y(n)</math> is <math>X(z)+Y(z)</math>.</p>	L2
22.	<p>What is the Fourier transform of the signal <math>x(n)=a^n n </math>, <math> a &lt;1</math>?</p> <p>a) <math>(1+a^2)/(1-2a\cos\theta+a^2)</math></p> <p>b) <math>(1-a^2)/(1-2a\cos\theta+a^2)</math></p> <p>c) <math>2a/(1-2a\cos\theta+a^2)</math></p> <p>d) None of the mentioned</p> <p><b>ANSWER: B</b></p> <p>Explanation: First we observe <math>x(n)</math> can be expressed as <math>x(n)=x_1(n)+x_2(n)</math> where <math>x_1(n)=a^n, n&gt;0</math>  <math>=0, \text{ elsewhere}</math>  <math>x_2(n)=a^{-n}, n&lt;0 =0, \text{ elsewhere}</math> Now applying Fourier transform for the above two signals, we get <math>X_1(\theta)=1/(1-ae^{-j\theta})</math> and <math>X_2(\theta)=(ae^{j\theta})/(1-ae^{j\theta})</math>          Now, <math>X(\theta)=X_1(\theta)+X_2(\theta)=1/(1-ae^{-j\theta})+(ae^{j\theta})/(1-ae^{j\theta})=(1-a^2)/(1-2a\cos\theta+a^2)</math></p>	L3
23.	<p>If <math>X(\theta)</math> is the Fourier transform of the signal <math>x(n)</math>, then what is the Fourier transform of the signal <math>x(n-k)</math>?</p> <p>a) <math>e^{-j\theta k} X(\theta)</math></p> <p>b) <math>e^{j\theta k} X(\theta)</math></p>	L1

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	<p>c) <math>e^{-j\omega k}</math>. <math>X(-\omega)</math></p> <p>d) <math>e^{-j\omega k}</math>. <math>X(\omega)</math></p> <p><b>ANSWER: D</b></p>	
24.	<p>What is the convolution of the sequences of <math>x_1(n)=x_2(n)=\{1,1,1\}</math>?</p> <p>a) <math>\{1,2,3,2,1\}</math></p> <p>b) <math>\{1,2,3,2,1\}</math></p> <p>c) <math>\{1,1,1,1,1\}</math></p> <p>d) <math>\{1,1,1,1,1\}</math></p> <p><b>ANSWER: A</b></p> <p>Explanation: Given <math>x_1(n)=x_2(n)=\{1,1,1\}</math></p> <p>By calculating the Fourier transform of the above two signals, we get</p> $X_1(\omega) = X_2(\omega) = 1 + e^{j\omega} + e^{-j\omega} = 1 + 2\cos\omega$ <p>From the convolution property of Fourier transform we have,</p> $X(\omega) = X_1(\omega) \cdot X_2(\omega) = (1 + 2\cos\omega)^2 = 3 + 4\cos\omega + 2\cos^2\omega$ <p>By applying the inverse Fourier transform of the above signal, we get</p> $x_1(n) * x_2(n) = \{1, 2, 3, 2, 1\}$	L5
25.	<p>What is the energy density spectrum of the signal <math>x(n)=a^n u(n)</math>, <math> a  &lt; 1</math>?</p> <p>a) <math>1/(1+2a\cos\omega+a^2)</math></p> <p>b) <math>1/(1-2a\cos\omega+a^2)</math></p> <p>c) <math>1/(1-2a\cos\omega-a^2)</math></p> <p>d) <math>1/(1+2a\cos\omega-a^2)</math></p> <p><b>ANSWER: B</b></p> <p>Explanation: Given <math>x(n)=a^n u(n)</math>, <math> a  &lt; 1</math> The auto correlation of the above signal is <math>r_{xx}(l) = 1/(1-a^{2 l })</math>, <math>-8 &lt; l &lt; 8</math> According to Wiener-Khintchine Theorem, <math>S_{xx}(\omega) = F\{r_{xx}(l)\} = [1/(1-a^2)] \cdot F\{ a ^{ l }\} = 1/(1-2a\cos\omega+a^2)</math></p>	L1

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26.	<p>DTFT is the representation of</p> <ol style="list-style-type: none"> <li>a. Periodic Discrete time signals</li> <li>b. Aperiodic Discrete time signals</li> <li>c. Aperiodic continuous signals</li> <li>d. Periodic continuous signals</li> </ol> <p><b>ANSWER:(B)</b> Aperiodic Discrete time signals</p>	L1
27.	<p>The transforming relations performed by DTFT are</p> <ol style="list-style-type: none"> <li>1) Linearity</li> <li>2) Modulation</li> <li>3) Shifting</li> <li>4) Convolution</li> </ol> <ol style="list-style-type: none"> <li>a. 1, 2 and 3 are correct</li> <li>b. 1 and 2 are correct</li> <li>c. 1 and 3 are correct</li> <li>d. All the four are correct</li> </ol> <p><b>ANSWER: (D)</b> All the four are correct</p>	L2
28.	<p>The DFT is preferred for</p> <ol style="list-style-type: none"> <li>1) Its ability to determine the frequency component of the signal</li> <li>2) Removal of noise</li> <li>3) Filter design</li> <li>4) Quantization of signal</li> </ol> <ol style="list-style-type: none"> <li>a. 1, 2 and 3 are correct</li> <li>b. 1 and 2 are correct</li> <li>c. 1 and 3 are correct</li> <li>d. All the four are correct</li> </ol> <p><b>ANSWER: (c)</b> 1 and 3 are correct</p>	L3
29.	<p>The Cooley–Tukey algorithm of FFT is a</p> <ol style="list-style-type: none"> <li>a. Divide and conquer algorithm</li> <li>b. Divide and rule algorithm</li> <li>c. Split and rule algorithm</li> <li>d. Split and combine algorithm</li> </ol> <p><b>ANSWER: (a)</b> Divide and conquer algorithm</p>	L1
30.	<p>DIT algorithm divides the sequence into</p>	L2



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	<p>a. Positive and negative values                  b. Even and odd samples                  c. Upper higher and lower spectrum                  d. Small and large samples</p> <p><b>ANSWER: (b)</b> Even and odd samples</p>	
31.	<p>FFT may be used to calculate</p> <p>1) DFT                  2) IDFT                  3) Direct Z transform                  4) In direct Z transform</p> <p>a. 1, 2 and 3 are correct                  b. 1 and 2 are correct                  c. 1 and 3 are correct                  d. All the four are correct</p> <p><b>ANSWER: (b)</b> 1 and 2 are correct</p>	L2
32.	<p><b>The computational procedure for Decimation in frequency algorithm takes</b></p> <p>a. Log<sub>2</sub> N stages                  b. 2Log<sub>2</sub> N stages                  c. Log<sub>2</sub> N<sup>2</sup> stages                  d. Log<sub>2</sub> N/2 stages</p> <p><b>ANSWER:(a)</b> Log<sub>2</sub> N stages</p>	L3
33.	<p><b>The transformations are required for</b></p> <p>1) Analysis in time or frequency domain                  2) Quantization                  3) Easier operations                  4) Modulation</p> <p>a. 1, 2 and 3 are correct                  b. 1 and 2 are correct                  c. 1 and 3 are correct                  d. All the four are correct</p> <p><b>ANSWER: (c)</b> 1 and 3 are correct</p>	L1
34.	<p><b>The s plane and z plane are related as</b></p> <p>a. <math>z = e^{sT}</math>                  b. <math>z = e^{2sT}</math>                  c. <math>z = 2e^{sT}</math>                  d. <math>z = e^{sT}/2</math></p> <p><b>ANSWER: (a)</b> <math>z = e^{sT}</math></p>	L2
35.	<p><b>The similarity between the Fourier transform and the z transform is that</b></p> <p>a. Both convert frequency spectrum domain to discrete time domain</p>	L1

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	<p>b. Both convert discrete time domain to frequency spectrum domain  c. Both convert analog signal to digital signal  d. Both convert digital signal to analog signal</p> <p><b>ANSWER: (b) Both convert discrete time domain to frequency spectrum domain</b></p>	
36.	<p><b>The anti causal sequences have _____ components in the left hand sequences.</b>  a. Positive  b. Negative  c. Both a and b  d. None of the above</p> <p><b>ANSWER: (a) Positive</b></p>	L2
37.	<p><b>FFT algorithm is designed to perform complex operations.</b>  a) True  b) False</p> <p><b>ANSWER:A</b>  Explanation: The FFT algorithm is designed to perform complex multiplications and additions, even though the input data may be real valued. The basic reason for this is that the phase factors are complex and hence, after the first stage of the algorithm, all variables are basically complex valued.</p>	L1
38.	<p><b>Padding of zeros increases the frequency resolution.</b>  a. True  b. False</p> <p><b>ANSWER: False</b></p>	L2
39.	<p><b>Circular shift of an N point is equivalent to</b>  a. Circular shift of its periodic extension and its vice versa  b. Linear shift of its periodic extension and its vice versa  c. Circular shift of its aperiodic extension and its vice versa  d. Linear shift of its aperiodic extension and its vice versa</p> <p><b>ANSWER: B</b></p>	L1
40.	<p><b>The circular convolution of two sequences in time domain is equivalent to</b>  a. Multiplication of DFTs of two sequences  b. Summation of DFTs of two sequences  c. Difference of DFTs of two sequences  d. Square of multiplication of DFTs of two sequences</p> <p><b>ANSWER: A</b></p>	L3