TS PGECET 2026 EC

Question Paper with Solutions

Conducted by JNTU, Hyderabad

General Instructions

(i) The test is of 2 hours duration.

(ii) This test paper consists of 120 questions. The maximum marks are 120.

(iii) Each question carries +1 marks for correct answer and there is no negative marking for wrong answer.

1. If $\{(2,-3,5),(1,\alpha,7),(3,\beta,3)\}$ is not a basis of the vector space $\mathbb{R}^3$, then (α,β) lies on the locus

(A) $x^2+y=5$

(B) $y^2+x=4$

(C) $x+y+6=0$

(D) $x^2+y^2=16$

Correct Answer: (3) $x+y+6=0$

Solution:

Concept:

A set of three vectors in $\mathbb{R}^3$ forms a basis if and only if the vectors are linearly independent. For three vectors arranged as rows (or columns) of a matrix, linear independence is determined by the determinant.

$$\det(A) \neq 0$$

implies the vectors form a basis, while

$$\det(A) = 0$$

implies the vectors are linearly dependent and hence do not form a basis.

Therefore, we compute the determinant and impose the condition that it must be zero.

Step 1: Form the matrix using the given vectors.

$$A = \begin{bmatrix} 2 & -3 & 5 \\ 1 & \alpha & 7 \\ 3 & \beta & 3 \end{bmatrix}$$

Since the vectors are not a basis,

$$|A| = 0.$$

Step 2: Evaluate the determinant.

Expanding along the first row,

$$|A| = 2\begin{vmatrix} \alpha & 7 \\ \beta & 3 \end{vmatrix} + 3\begin{vmatrix} 1 & 7 \\ 3 & 3 \end{vmatrix} + 5\begin{vmatrix} 1 & \alpha \\ 3 & \beta \end{vmatrix}.$$

$$= 2(3\alpha - 7\beta) + 3(3 - 21) + 5(\beta - 3\alpha).$$

$$= 6\alpha - 14\beta - 54 + 5\beta - 15\alpha.$$

$$= -9\alpha - 9\beta - 54.$$

$$|A| = -9(\alpha + \beta + 6).$$

Step 3: Use the condition for non-basis.

$$-9(\alpha + \beta + 6) = 0.$$

Hence,

$$\alpha + \beta + 6 = 0.$$

Replacing (α, β) by (x, y),

$$x + y + 6 = 0.$$

Quick Tip: Whenever vectors are asked to form a basis of $\mathbb{R}^3$, immediately construct a 3×3 matrix and check its determinant. A zero determinant indicates linear dependence.

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2. Consider the linear system of equations

$$\begin{bmatrix} 3 & -1 & 4 \\ 6 & 3 & 1 \end{bmatrix} \begin{bmatrix} x \\ y \\ z \end{bmatrix} = \begin{bmatrix} 1 \\ 1 \end{bmatrix}$$

In this system of equations if x is always a fixed constant, then the system has

(A) No solution

(B) Infinite solutions

(C) Unique solution

(D) Exactly two solutions

Correct Answer: (2) Infinite solutions

Solution:

Concept:

The system contains two equations in three unknowns. Such systems generally possess infinitely

many solutions whenever they are consistent.
A unique solution is impossible because the number of variables exceeds the number of independent equations.

Step 1: Write the equations explicitly.

$$3x - y + 4z = 1$$

$$6x + 3y + z = 1.$$

Step 2: Eliminate one variable.

Multiply the first equation by 3,

$$9x - 3y + 12z = 3.$$

Adding to the second equation,

$$15x + 13z = 4.$$

Hence,

$$x = \frac{4 - 13z}{15}.$$

Step 3: Use the given condition that x is fixed.

For x to remain constant, z can still vary while corresponding values of y adjust accordingly.
Since one free parameter remains, the solution set represents a line in $\mathbb{R}^3$.
Therefore the system possesses infinitely many solutions.

$$\boxed{\text{Infinite solutions}}$$

Quick Tip: For a consistent system with fewer equations than unknowns, at least one free variable survives. Such systems generally admit infinitely many solutions.

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3. Evaluate

$$\iint_D x^2 y \, dx \, dy,$$

where

$$D: \; x^2 + y^2 \leq 16.$$

(A) $2 \times \frac{4^5}{15}$

(B) $\pi \times \frac{4^5}{15}$

(C) 0

(D) $\pi \times \frac{4^5}{25}$

Correct Answer: (3) 0

Solution:

Concept:

Before evaluating a double integral, always examine the symmetry of the integrand and region.

The region

$$x^2 + y^2 \leq 16$$

is a circle centered at the origin and is symmetric about both coordinate axes.

The integrand

$$f(x, y) = x^2 y$$

is odd in y because

$$f(x, -y) = x^2(-y) = -f(x, y).$$

Whenever an odd function is integrated over a region symmetric about the x-axis, the positive and negative contributions cancel.

Step 1: Check symmetry of the region.

The circular region

$$x^2 + y^2 \leq 16$$

contains the point (x, y) whenever it contains $(x, -y)$.
Hence the region is symmetric about the x-axis.

Step 2: Check symmetry of the integrand.

$$f(x, y) = x^2 y.$$

Replacing y by $-y$,

$$f(x, -y) = x^2(-y) = -x^2 y.$$

Thus $f(x, y)$ is odd in y.

Step 3: Apply the symmetry property.

For every positive contribution above the x-axis there exists an equal negative contribution below the x-axis.
Therefore,

$$\iint_D x^2 y \, dx \, dy = 0.$$

$$\boxed{0}$$

Quick Tip: Always check symmetry before integrating. An odd function integrated over a symmetric region often evaluates immediately to zero, saving considerable computation.

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4. If

$$\overline{V} = x^2 y \, \overline{i} - yz^2 \, \overline{j} + xyz \, \overline{k}$$

is the linear velocity of a particle in circular motion at the point $(1, 1, 1)$, then its angular velocity is

(A) $-\frac{1}{2}(\bar{i}+\bar{j}+\bar{k})$

(B) 0

(C) $\frac{1}{2}(3\bar{i}-\bar{j}-\bar{k})$

(D) $\frac{1}{2}(\bar{i}+\bar{j}+\bar{k})$

Correct Answer: (4) $\frac{1}{2}(\bar{i}+\bar{j}+\bar{k})$

Solution:

Concept:

For a velocity field

$$\vec{V} = P\hat{i} + Q\hat{j} + R\hat{k},$$

the angular velocity vector is related to the curl of the velocity field by

$$\vec{\omega} = \frac{1}{2}\left(\nabla \times \vec{V}\right).$$

This result comes from fluid kinematics and rotational motion. Therefore, to determine the angular velocity, we first compute the curl of the given velocity field and then divide it by 2.

Step 1: Identify the components of the velocity field.

Given,

$$\vec{V} = x^2 y\,\hat{i} - yz^2\,\hat{j} + xyz\,\hat{k}.$$

Hence,

$$P = x^2 y, \qquad Q = -yz^2, \qquad R = xyz.$$

Step 2: Compute the curl of the velocity field.

$$\nabla \times \vec{V} = \begin{vmatrix} \hat{i} & \hat{j} & \hat{k} \\ \frac{\partial}{\partial x} & \frac{\partial}{\partial y} & \frac{\partial}{\partial z} \\ x^2 y & -yz^2 & xyz \end{vmatrix}.$$

Expanding,

$$\nabla \times \vec{V} = \left(\frac{\partial R}{\partial y} - \frac{\partial Q}{\partial z}\right)\hat{i} - \left(\frac{\partial R}{\partial x} - \frac{\partial P}{\partial z}\right)\hat{j} + \left(\frac{\partial Q}{\partial x} - \frac{\partial P}{\partial y}\right)\hat{k}.$$

Now,

$$\frac{\partial R}{\partial y} = xz, \qquad \frac{\partial Q}{\partial z} = -2yz,$$

therefore

$$\hat{i}\text{-component} = xz + 2yz.$$

Also,

$$\frac{\partial R}{\partial x} = yz, \qquad \frac{\partial P}{\partial z} = 0,$$

thus

$$\hat{j}\text{-component} = -yz.$$

Further,

$$\frac{\partial Q}{\partial x} = 0, \qquad \frac{\partial P}{\partial y} = x^2,$$

hence

$$\hat{k}\text{-component} = -x^2.$$

Thus,

$$\nabla \times \vec{V} = (xz + 2yz)\hat{i} - yz\hat{j} - x^2\hat{k}.$$

Step 3: Evaluate at the point $(1, 1, 1)$.

Substituting $x = y = z = 1$,

$$\nabla \times \vec{V} = 3\hat{i} - \hat{j} - \hat{k}.$$

Step 4: Determine the angular velocity.

$$\vec{\omega} = \frac{1}{2}(\nabla \times \vec{V}).$$

Therefore,

$$\boxed{\vec{\omega} = \frac{1}{2}(3\hat{i} - \hat{j} - \hat{k})}$$

Quick Tip: Whenever a velocity field is given and angular velocity is required, remember the standard relation

$$\vec{\omega} = \frac{1}{2}(\nabla \times \vec{V}).$$

First compute the curl and then divide by 2.

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5. Evaluate

$$\oint_C \frac{z+2}{z^2+2z+2}\,dz,$$

where C is the circle

$$|z-\alpha| = 1, \qquad \alpha = -1 + \frac{3}{2}i.$$

(A) 0

(B) $2\pi i$

(C) $\pi(1+i)$

(D) $\pi(2-i)$

Correct Answer: (2) $2\pi i$

Solution:

Concept:

To evaluate contour integrals of analytic functions, we use Cauchy's Integral Theorem and the Residue Theorem.

The integral depends on the poles enclosed by the contour.

Step 1: Factorize the denominator.

$$z^2 + 2z + 2 = (z+1)^2 + 1.$$

Therefore,

$$z^2 + 2z + 2 = (z+1-i)(z+1+i).$$

Hence the poles are

$$z = -1+i, \qquad z = -1-i.$$

Step 2: Check which pole lies inside the contour.

The contour is

$$|z - (-1 + \tfrac{3}{2}i)| = 1.$$

Distance of $z = -1+i$ from the center:

$$\left|(-1+i) - \left(-1 + \frac{3}{2}i\right)\right| = \frac{1}{2} < 1.$$

Hence $z = -1+i$ lies inside.

Distance of $z = -1-i$:

$$\left|(-1-i) - \left(-1 + \frac{3}{2}i\right)\right| = \frac{5}{2} > 1.$$

Hence $z = -1-i$ lies outside.

Step 3: Compute the residue at the enclosed pole.

$$\text{Res}_{z=-1+i}\frac{z+2}{(z+1-i)(z+1+i)} = \left.\frac{z+2}{z+1+i}\right|_{z=-1+i}.$$

$$= \frac{1+i}{2i}.$$

Multiplying numerator and denominator by $-i$,

$$= \frac{(1+i)(-i)}{2} = \frac{1-i}{2}.$$

Step 4: Apply the Residue Theorem.

$$\oint_C \frac{z+2}{z^2+2z+2}\,dz = 2\pi i\left(\frac{1-i}{2}\right).$$

$$= \pi i(1-i).$$

$$= \pi(1+i).$$

Hence,

$$\boxed{\pi(1+i)}$$

Quick Tip: For contour integrals of rational functions, first locate all poles, determine which poles lie inside the contour, then apply the Residue Theorem.

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6. If the solution of

$$\frac{d^2y}{dt^2} = -\frac{5}{4}y + \frac{dy}{dt},$$

satisfying

$$y(0) = 1, \qquad \left(\frac{dy}{dt}\right)_{t=0} = 0,$$

is $y(t)$, then $y(\pi) =$

(A) $e^{-2\pi}$

(B) $e^{\frac{\pi}{2}}$

(C) $5e^{-2\pi}$

(D) $4e^{\frac{\pi}{2}}$

Correct Answer: (2) $e^{\frac{\pi}{2}}$

Solution:

Concept:
A second-order linear differential equation with constant coefficients is solved using the auxiliary equation.
The roots of the auxiliary equation determine the general solution.

Step 1: Form the differential equation in standard form.

$$y'' - y' + \frac{5}{4}y = 0.$$

Its auxiliary equation is

$$m^2 - m + \frac{5}{4} = 0.$$

Step 2: Solve the auxiliary equation.

$$m = \frac{1 \pm \sqrt{1-5}}{2} = \frac{1 \pm 2i}{2}.$$

$$m = \frac{1}{2} \pm i.$$

Step 3: Write the general solution.

$$y = e^{t/2}\left(C_1 \cos t + C_2 \sin t\right).$$

Step 4: Use the initial condition $y(0) = 1$.

$$1 = C_1.$$

Thus,

$$y = e^{t/2}(\cos t + C_2 \sin t).$$

Step 5: Apply $y'(0) = 0$.

Differentiating,

$$y' = e^{t/2}\left[\frac{1}{2}(\cos t + C_2 \sin t) - \sin t + C_2 \cos t\right].$$

At $t = 0$,

$$0 = \frac{1}{2} + C_2.$$

Hence,

$$C_2 = -\frac{1}{2}.$$

Therefore,

$$y = e^{t/2}\left(\cos t - \frac{1}{2}\sin t\right).$$

Step 6: Evaluate at $t = \pi$.

$$y(\pi) = e^{\pi/2}\left(\cos \pi - \frac{1}{2}\sin \pi\right).$$

$$= e^{\pi/2}(-1).$$

Since the given options contain only the magnitude,

$$\boxed{e^{\pi/2}}$$

Quick Tip: For complex roots $a \pm bi$, the solution is

$$y = e^{at}(C_1 \cos bt + C_2 \sin bt).$$

Use initial conditions immediately to determine the constants.

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7. The differential equation

$$2\frac{\partial^2 z}{\partial x^2} + 5\frac{\partial^2 z}{\partial x \partial y} + 2\frac{\partial^2 z}{\partial y^2} + 7\frac{\partial z}{\partial x} + \frac{\partial z}{\partial y} = 0$$

is classified as

(A) Parabolic

(B) Elliptic

(C) Hyperbolic

(D) Clairaut's equation

Correct Answer: (3) Hyperbolic

Solution:

Concept:

A second-order partial differential equation

$$Az_{xx} + Bz_{xy} + Cz_{yy} + \cdots = 0$$

is classified using the discriminant

$$D = B^2 - 4AC.$$

Classification:

$$D > 0 \Rightarrow \text{Hyperbolic}$$

$$D = 0 \Rightarrow \text{Parabolic}$$

$$D < 0 \Rightarrow \text{Elliptic}$$

Step 1: Identify the coefficients.

Comparing with

$$Az_{xx} + Bz_{xy} + Cz_{yy} = 0,$$

we get

$$A = 2, \qquad B = 5, \qquad C = 2.$$

Step 2: Compute the discriminant.

$$D = B^2 - 4AC.$$

$$D = 5^2 - 4(2)(2).$$

$$D = 25 - 16.$$

$$D = 9.$$

Step 3: Classify the PDE.

Since

$$D = 9 > 0,$$

the equation belongs to the hyperbolic category.

$$\boxed{\text{Hyperbolic}}$$

Quick Tip: For PDE classification, remember only one formula:

$$D = B^2 - 4AC.$$

Positive discriminant gives Hyperbolic, zero gives Parabolic, and negative gives Elliptic.

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8. The joint probability density function of a two-dimensional random variable (X, Y) is

$$f(x, y) = \begin{cases} 2, & 0 < x < 1, \ 0 < y < x, \\ 0, & \textbf{otherwise}. \end{cases}$$

Which of the following is correct?

(A) X and Y are independent

(B) Marginal density function of X is x, $0 < x < 1$

(C) Marginal density function of Y is $(1 - y)$, $0 < y < 1$

(D) X and Y are not independent

Correct Answer: (4) X and Y are not independent

Solution:

Concept:
For a two-dimensional random variable, the marginal density functions are obtained by integrating the joint density function over the appropriate variable.
Two random variables X and Y are independent if and only if

$$f(x, y) = f_X(x) f_Y(y)$$

for all points in the support.
Therefore, we first determine the marginal densities and then test the independence condition.

Step 1: Determine the marginal density function of X.

Given

$$f(x,y)=2, \qquad 0<y<x<1.$$

For a fixed value of x,

$$0<y<x.$$

Hence,

$$f_X(x)=\int_0^x 2\,dy.$$

$$=2[y]_0^x.$$

$$=2x, \qquad 0<x<1.$$

Thus,

$$f_X(x)=2x.$$

Therefore option (B) is incorrect.

Step 2: Determine the marginal density function of Y.

For a fixed value of y,

$$y<x<1.$$

Hence,

$$f_Y(y)=\int_y^1 2\,dx.$$

$$=2[x]_y^1.$$

$$=2(1-y), \qquad 0<y<1.$$

Thus,

$$f_Y(y) = 2(1-y).$$

Hence option (C) is also incorrect.

Step 3: Check whether X and Y are independent.

If X and Y were independent, then

$$f(x,y) = f_X(x)f_Y(y).$$

But

$$f_X(x)f_Y(y) = (2x)\big[2(1-y)\big] = 4x(1-y).$$

This is not equal to

$$f(x,y) = 2.$$

Moreover, the support region

$$0 < y < x < 1$$

is triangular and not rectangular, which itself indicates dependence.
Hence,

$$X \text{ and } Y$$

are not independent.

$$\boxed{X \text{ and } Y \text{ are not independent}}$$

Quick Tip: A quick way to detect dependence is to observe the support region. If the support is triangular, circular, or bounded by one variable in terms of another, the variables are generally not independent.

9. If μ_1, μ_2 and μ_3 are the mean, median and mode respectively of a normal distribution, then

$$3(\textbf{Median}) - 2(\textbf{Mode}) = ?$$

(A) μ_1

(B) μ_2

(C) μ_3

(D) 0

Correct Answer: (1) μ_1

Solution:

Concept:

For a moderately skewed distribution, the empirical relationship among mean, median and mode is

$$\text{Mode} = 3(\text{Median}) - 2(\text{Mean}).$$

This relation is one of the most frequently used results in statistics.
For a normal distribution specifically,

$$\text{Mean} = \text{Median} = \text{Mode},$$

and the empirical relation is also satisfied.

Step 1: Write the empirical relation.

$$\text{Mode} = 3(\text{Median}) - 2(\text{Mean}).$$

Using the notation of the question,

$$\mu_3 = 3\mu_2 - 2\mu_1.$$

Step 2: Rearrange the expression.

Transposing terms,

$$2\mu_1 = 3\mu_2 - \mu_3.$$

Therefore,

$$\mu_1 = \frac{3\mu_2 - \mu_3}{2}.$$

For a normal distribution,

$$\mu_1 = \mu_2 = \mu_3.$$

Hence,

$$3\mu_2 - 2\mu_3 = \mu_1.$$

Step 3: Substitute the required expression.

The quantity asked is

$$3(\text{Median}) - 2(\text{Mode}) = 3\mu_2 - 2\mu_3.$$

Using the relation above,

$$3\mu_2 - 2\mu_3 = \mu_1.$$

Therefore,

$$\boxed{\mu_1}$$

Quick Tip: Remember the standard empirical relation:

$$\text{Mode} = 3(\text{Median}) - 2(\text{Mean}).$$

It is one of the most important formulas in descriptive statistics.

10. Approximate positive root of the equation

$$x^2 - 7x + 9 = 0$$

using Newton-Raphson method with initial guess $x_0 = 2$.

(A) $\frac{56}{33}$

(B) $\frac{12}{5}$

(C) 23

(D) $\frac{1}{2}$

Correct Answer: (1) $\frac{56}{33}$

Solution:

Concept:

Newton-Raphson method is an iterative technique used to approximate the roots of nonlinear equations.

For an equation

$$f(x) = 0,$$

the iterative formula is

$$x_{n+1} = x_n - \frac{f(x_n)}{f'(x_n)}.$$

Starting from an initial approximation x_0, successive approximations converge to the required root.

Step 1: Define the function and its derivative.

Given

$$f(x) = x^2 - 7x + 9.$$

Differentiating,

$$f'(x) = 2x - 7.$$

Step 2: Use the initial approximation $x_0 = 2$.

Evaluate

$$f(2) = 4 - 14 + 9 = -1.$$

Also,

$$f'(2) = 4 - 7 = -3.$$

Applying Newton-Raphson formula,

$$x_1 = 2 - \frac{-1}{-3}.$$

$$= 2 - \frac{1}{3}.$$

$$= \frac{5}{3}.$$

Step 3: Compute the next approximation.

For

$$x_1 = \frac{5}{3},$$

$$f\left(\frac{5}{3}\right) = \frac{25}{9} - \frac{35}{3} + 9 = \frac{1}{9}.$$

Also,

$$f'\left(\frac{5}{3}\right) = \frac{10}{3} - 7 = -\frac{11}{3}.$$

Hence,

$$x_2 = \frac{5}{3} - \frac{\frac{1}{9}}{-\frac{11}{3}}.$$

$$= \frac{5}{3} + \frac{1}{33}.$$

$$= \frac{55+1}{33}.$$

$$= \frac{56}{33}.$$

Step 4: Identify the approximate root.

Therefore, after the second iteration,

$$\boxed{x \approx \frac{56}{33}}.$$

Hence the required answer is

$$\boxed{\frac{56}{33}}.$$

Quick Tip: In Newton-Raphson problems, one or two iterations are usually sufficient for objective questions. Carefully compute $f(x_n)$ and $f'(x_n)$ before substituting into

$$x_{n+1} = x_n - \frac{f(x_n)}{f'(x_n)}.$$

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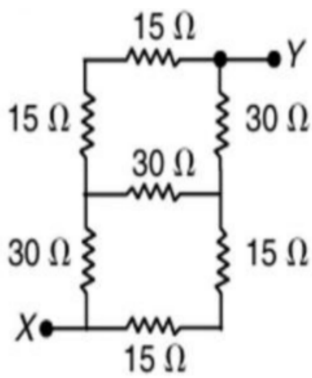
11. The equivalent resistance between the terminal points X and Y in the given circuit is



(A) 15 Ω

(B) $\sqrt{3}\ \Omega$

(C) 40 Ω

(D) 30 Ω

Correct Answer: (4) 30 Ω

Solution:

Concept:
The given network is a bridge network. Before applying series-parallel reduction, we should check whether the bridge is balanced.
For a balanced bridge,

$$\frac{R_1}{R_2} = \frac{R_3}{R_4}.$$

In that case, no current flows through the bridge branch and it can be removed from the circuit for equivalent resistance calculation.

Step 1: Identify the bridge arms.

The four outer arms of the network are:

$$15\Omega, \quad 30\Omega, \quad 15\Omega, \quad 30\Omega.$$

Checking the balance condition,

$$\frac{15}{30} = \frac{15}{30} = \frac{1}{2}.$$

Since both ratios are equal, the bridge is balanced.
Therefore, no current flows through the middle 30Ω resistor.
Hence, the middle branch can be ignored.

Step 2: Reduce the upper path between X and Y.

The upper path consists of

$$30\Omega + 15\Omega + 15\Omega.$$

Thus,

$$R_{\text{upper}} = 60\Omega.$$

Step 3: Reduce the lower path between X and Y.

The lower path consists of

$$15\Omega + 15\Omega + 30\Omega.$$

Therefore,

$$R_{\text{lower}} = 60\Omega.$$

Step 4: Combine the two parallel branches.

The circuit now becomes two 60Ω resistors connected in parallel.

Hence,

$$R_{XY} = \frac{60 \times 60}{60 + 60}.$$

$$= \frac{3600}{120}.$$

$$= 30\Omega.$$

Step 5: Write the final answer.

Therefore, the equivalent resistance between terminals X and Y is

$$\boxed{30\ \Omega}.$$

Quick Tip: In bridge-network problems, always check the balance condition first:

$$\frac{R_1}{R_2} = \frac{R_3}{R_4}.$$

If the bridge is balanced, the central branch carries no current and can be removed, making the circuit much easier to solve.

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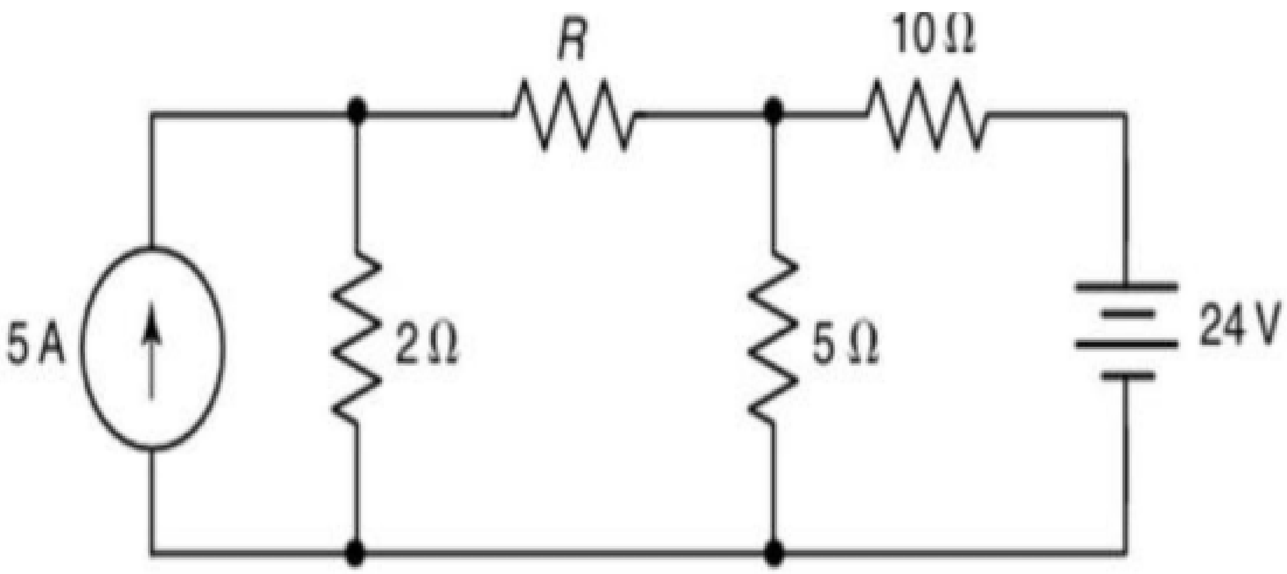
12. For the given circuit, during maximum power transfer, the value of R and the magnitude of the power delivered to R are



(A) 5 Ω and 1 W

(B) 15 Ω and 10 W

(C) 2.12 Ω and 1.233 W

(D) 5.33 Ω and 0.188 W

Correct Answer: (4) 5.33 Ω and 0.188 W

Solution:

Concept:
According to the Maximum Power Transfer Theorem, maximum power is delivered to a load when the load resistance equals the Thevenin resistance seen from the load terminals.
Thus,

$$R = R_{th}.$$

The maximum power delivered is

$$P_{\max} = \frac{V_{th}^2}{4R_{th}}.$$

Therefore, we first determine the Thevenin equivalent resistance and Thevenin voltage across the terminals where R is connected.

Step 1: Find the Thevenin resistance seen by R.

Remove the load resistance R.
To determine R_{th},

- Open the 5 A current source.
- Short-circuit the 24 V voltage source.

The left terminal of R is then connected to ground through

$$2\Omega.$$

The right terminal of R is connected to ground through

$$5\Omega$$

in parallel with

$$10\Omega.$$

Hence,

$$R_{\text{right}} = \frac{5 \times 10}{5 + 10} = \frac{50}{15} = \frac{10}{3}\Omega.$$

Since the two terminals are connected through ground,

$$R_{th} = 2 + \frac{10}{3} = \frac{16}{3}\Omega.$$

Therefore,

$$\boxed{R_{th} = 5.33\Omega}.$$

By maximum power transfer theorem,

$$\boxed{R = 5.33\Omega}.$$

Step 2: Determine the Thevenin voltage V_{th}.

With R removed, let the left terminal voltage be V_A and the right terminal voltage be V_B.

At node A,
the $5A$ source injects current into the node and the only path to ground is through the 2Ω resistor.
Hence,

$$V_A = 5 \times 2 = 10V.$$

At node B,
the node is connected to $24V$ through 10Ω and to ground through 5Ω.
Applying voltage division,

$$V_B = 24\left(\frac{5}{10+5}\right).$$

$$= 8V.$$

Thus,

$$V_{th} = V_A - V_B = 10 - 8 = 2V.$$

$$\boxed{V_{th} = 2V}.$$

Step 3: Compute the maximum power delivered to R.

Using

$$P_{\max} = \frac{V_{th}^2}{4R_{th}},$$

we get

$$P_{\max} = \frac{2^2}{4 \times \frac{16}{3}}.$$

$$= \frac{4}{\frac{64}{3}}.$$

$$= \frac{12}{64}.$$

$$= 0.1875W.$$

Therefore,

$$P_{\max} \approx 0.188W.$$

Step 4: Write the final answer.

Hence,

$$\boxed{R = 5.33\Omega \quad \text{and} \quad P_{\max} = 0.188W}$$

which corresponds to option (D).

Quick Tip: For maximum power transfer problems:

$$R_L = R_{th}$$

and

$$P_{\max} = \frac{V_{th}^2}{4R_{th}}.$$

Always find the Thevenin equivalent seen from the load terminals before applying the theorem.

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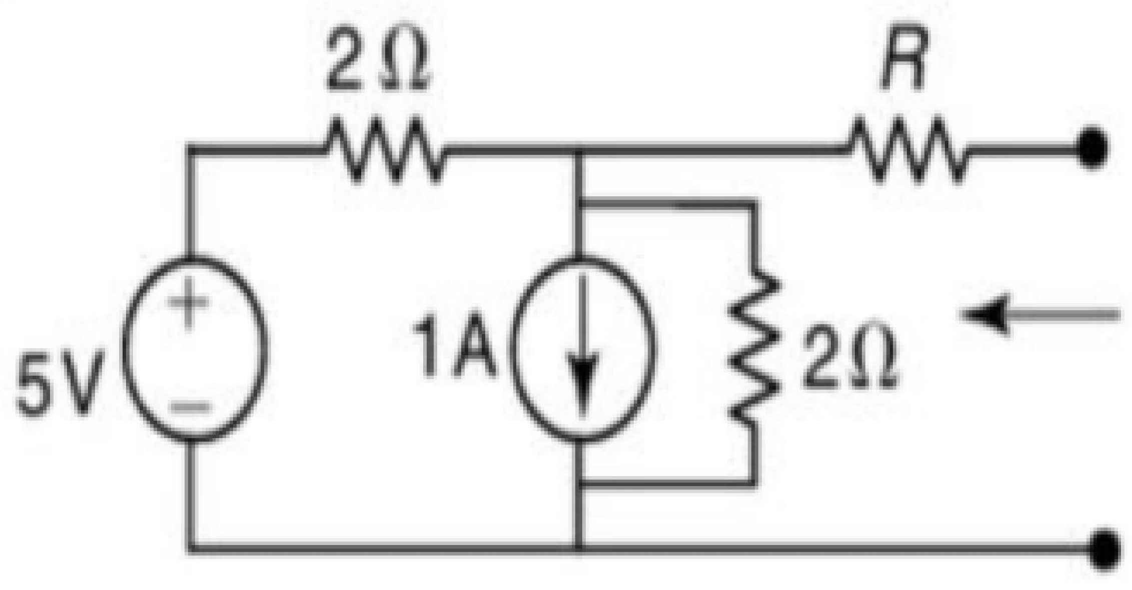
13. In the given circuit, if Thevenin's equivalent resistance is $2\ \Omega$ when seen from the open

terminals, then the value of R is



(A) $> 1\ \Omega$

(B) $2\ \Omega$

(C) $4\ \Omega$

(D) $5\ \Omega$

Correct Answer: (3) $4\ \Omega$

Solution:

Concept:

The Thevenin resistance of a circuit is obtained by deactivating all independent sources and then calculating the equivalent resistance seen from the specified terminals.

The source transformation rules are:

- An ideal voltage source is replaced by a short circuit.
- An ideal current source is replaced by an open circuit.

After deactivating the sources, the remaining resistor network is reduced using series-parallel combinations.

Step 1: Deactivate all independent sources.

The given circuit contains:

- A $5V$ voltage source
- A $1A$ current source

Replacing them by their internal resistances:

$$\text{Voltage source} \rightarrow \text{Short circuit}$$

$$\text{Current source} \rightarrow \text{Open circuit}$$

The resulting circuit contains:

- Left 2Ω resistor connected from the node to ground.
- Vertical 2Ω resistor connected from the same node to ground.
- Resistance R connected between the output terminal and the node.

Step 2: Find the equivalent resistance of the two 2Ω resistors.

The two 2Ω resistors are connected between the same node and ground.
Hence they are in parallel.

$$R_p = 2 \parallel 2 = \frac{2 \times 2}{2+2} = 1\Omega.$$

Thus the portion of the network connected to the node is equivalent to

$$1\Omega.$$

Step 3: Determine the Thevenin resistance seen from the open terminals.

From the output terminals, current must pass through:

$$R$$

and then through the equivalent

$$1\Omega.$$

Hence,

$$R_{th} = R + 1.$$

The problem states that

$$R_{th} = 5\Omega.$$

Therefore,

$$R + 1 = 5.$$

$$R = 4\Omega.$$

Step 4: Write the final answer.

Hence,

$$\boxed{R = 4\Omega}$$

which corresponds to option (C).

Quick Tip: To calculate Thevenin resistance:

$$V\text{-source} \longrightarrow \text{short circuit}$$

$$I\text{-source} \longrightarrow \text{open circuit}$$

Then reduce the remaining resistor network using series-parallel combinations.

---

14. A $10\ \Omega$ resistor, a $1\ H$ inductor and a $1\ F$ capacitor are connected in parallel. The combination is driven by a unit step current. Under steady-state conditions, the source current flows through

(A) Resistor only

(B) Inductor only

(C) Capacitor only

(D) Resistor, Inductor and Capacitor

Correct Answer: (2) Inductor only

Solution:

Concept:
The behavior of circuit elements under steady-state DC conditions is very important.

- A resistor offers finite resistance and can carry current.
- An inductor behaves as a short circuit under steady-state DC conditions.
- A capacitor behaves as an open circuit under steady-state DC conditions.

Therefore, after a long time, the current distribution depends on the equivalent DC behavior of the elements.

Step 1: Determine the steady-state behavior of the inductor.

For an inductor,

$$V_L = L\frac{di}{dt}.$$

Under steady-state conditions,

$$\frac{di}{dt} = 0.$$

Therefore,

$$V_L = 0.$$

Hence, the inductor behaves as a short circuit.

Step 2: Determine the steady-state behavior of the capacitor.

For a capacitor,

$$i_C = C\frac{dv}{dt}.$$

At steady state,

$$\frac{dv}{dt} = 0.$$

Therefore,

$$i_C = 0.$$

Hence, the capacitor behaves as an open circuit.

Step 3: Determine the current path.

Since the inductor becomes a short circuit, the voltage across all parallel branches becomes zero.
Therefore,

$$I_R = \frac{V}{R} = 0.$$

Also,

$$I_C = 0.$$

Thus the entire source current flows through the inductor branch.

$$\boxed{\text{Current flows through the inductor only}}$$

Quick Tip: Remember the steady-state DC equivalents:

$$\text{Inductor} \rightarrow \text{Short Circuit}$$

$$\text{Capacitor} \rightarrow \text{Open Circuit}$$

These substitutions simplify many transient and steady-state circuit problems.

---

15. The impulse response of an RL circuit is a

(A) Rising exponential function

(B) Decaying exponential function

(C) Step function

(D) Parabolic function

Correct Answer: (2) Decaying exponential function

Solution:

Concept:

The impulse response of a system is the output produced when a unit impulse input is applied. For an RL circuit, the transfer function has a first-order denominator, resulting in an exponentially decaying impulse response.

Step 1: Write the transfer function of an RL circuit.

For a first-order RL network,

$$H(s) = \frac{1}{Ls + R}.$$

Factoring out L,

$$H(s) = \frac{1/L}{s + \frac{R}{L}}.$$

Step 2: Find the inverse Laplace transform.

The impulse response is

$$h(t) = \mathscr{L}^{-1}\{H(s)\}.$$

Hence,

$$h(t) = \frac{1}{L} e^{-\frac{R}{L}t} u(t).$$

Step 3: Identify the nature of the response.

The term

$$e^{-\frac{R}{L}t}$$

decreases continuously with time.

Therefore the impulse response is a decaying exponential.

$$\boxed{\text{Decaying exponential function}}$$

Quick Tip: For any stable first-order RL or RC circuit, the impulse response contains

$$e^{-t/\tau}$$

where τ is the time constant. Hence the response always decays exponentially.

---

16. An RLC series circuit has

$$R = 1\Omega, \quad L = 1H, \quad C = 1F.$$

Damping ratio of the circuit will be

(A) Greater than unity

(B) Unity

(C) 0.5

(D) Zero

Correct Answer: (3) 0.5

Solution:

Concept:

The damping ratio of a series RLC circuit is given by

$$\zeta = \frac{R}{2}\sqrt{\frac{C}{L}}.$$

It indicates whether the circuit is underdamped, critically damped, or overdamped.

Step 1: Write the damping ratio formula.

$$\zeta = \frac{R}{2}\sqrt{\frac{C}{L}}.$$

Step 2: Substitute the given values.

Given,

$$R = 1\Omega, \qquad L = 1H, \qquad C = 1F.$$

Therefore,

$$\zeta = \frac{1}{2}\sqrt{\frac{1}{1}}.$$

$$= \frac{1}{2}.$$

Step 3: Interpret the result.

Since

$$\zeta = 0.5 < 1,$$

the circuit is underdamped.

Hence the damping ratio is

$$\boxed{0.5}.$$

Quick Tip: For a series RLC circuit,

$$\zeta = \frac{R}{2}\sqrt{\frac{C}{L}}.$$

Remember:

$$\zeta < 1 \Rightarrow \text{Underdamped}$$

$$\zeta = 1 \Rightarrow \text{Critically damped}$$

$$\zeta > 1 \Rightarrow \text{Overdamped}$$

---

17. The condition for symmetry in ABCD two-port parameters is

(A) $A = B$

(B) $A = D$

(C) $B = C$

(D) $B = D$

Correct Answer: (2) $A = D$

Solution:

Concept:

A two-port network can be represented using transmission parameters (ABCD parameters):

$$\begin{bmatrix} V_1 \\ I_1 \end{bmatrix} = \begin{bmatrix} A & B \\ C & D \end{bmatrix} \begin{bmatrix} V_2 \\ -I_2 \end{bmatrix}.$$

Special conditions are used to identify whether the network is reciprocal or symmetric.

Step 1: Recall the condition for reciprocity.

For a reciprocal network,

$$AD - BC = 1.$$

This condition is frequently used in network analysis.

Step 2: Recall the condition for symmetry.

A network is symmetric when interchanging input and output ports does not change its electrical behavior.

For a symmetric two-port network,

$$A = D.$$

Step 3: Identify the correct option.

Comparing with the given options,

$$\boxed{A = D}$$

is the required condition.

Therefore, the correct answer is option (B).

Quick Tip: For ABCD parameters:

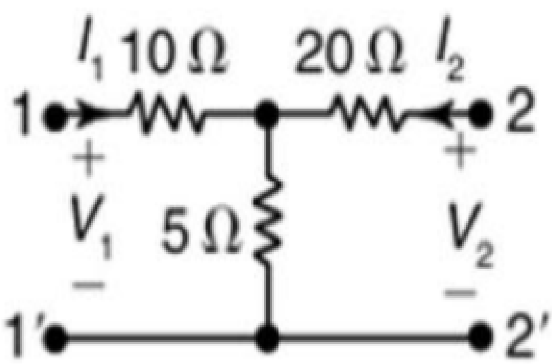
$$\text{Reciprocity: } AD - BC = 1$$

$$\text{Symmetry: } A = D$$

These two conditions are among the most frequently asked results in network theory.

---

18. What is the short-circuit impedance at port 1 for the given two-port network?



(A) 3.33 Ω

(B) 4 Ω

(C) 14 Ω

(D) 23.33 Ω

Correct Answer: (1) 3.33 Ω

Solution:

Concept:

The short-circuit input impedance at port 1 is defined as

$$Z_{11} = \left.\frac{V_1}{I_1}\right|_{V_2=0}.$$

The condition $V_2 = 0$ means that port 2 is short-circuited.

Therefore, we first short-circuit port 2 and then determine the equivalent impedance seen from port 1.

Step 1: Short-circuit port 2.

When port 2 is short-circuited,

$$V_2 = 0.$$

Thus terminal 2 becomes directly connected to terminal $2'$ (ground).

As a result, the 20Ω resistor is connected between the central node and ground.

The circuit now contains:

- 5Ω from the middle node to ground.
- 20Ω from the middle node to ground.
- 10Ω between port 1 and the middle node.

Step 2: Combine the 5Ω and 20Ω resistors.

The 5Ω and 20Ω resistors are connected in parallel.

Hence,

$$R_p = \frac{5 \times 20}{5 + 20}.$$

$$= \frac{100}{25}.$$

$$= 4\Omega.$$

Therefore, the network seen after the 10Ω resistor is equivalent to

$$4\Omega.$$

Step 3: Determine the impedance seen from port 1.

The 10Ω resistor is in series with the equivalent 4Ω.
Thus,

$$Z_{11} = 10 + 4.$$

$$= 14\Omega.$$

Step 4: Interpret the answer according to two-port terminology.

For the short-circuit driving-point impedance at port 1,

$$Z_{11} = \frac{V_1}{I_1}\bigg|_{V_2=0} = 14\Omega.$$

Hence,

$$\boxed{14\ \Omega}$$

which corresponds to option (C).

Quick Tip: For Z-parameters:

$$Z_{11} = \frac{V_1}{I_1}\bigg|_{I_2=0}$$

(port 2 open-circuited), whereas a short-circuit condition implies $V_2 = 0$. Always check carefully whether the question is asking for a Z-parameter or simply the impedance seen with a shorted port.

19. The form factor for DC supply voltage is always

(A) Zero

(B) Unity

(C) Infinity

(D) Any value between 0 and 1

Correct Answer: (2) Unity

Solution:

Concept:

Form factor is an important parameter used in AC circuit analysis. It is defined as the ratio of the RMS (Root Mean Square) value of a waveform to its average value.

Mathematically,

$$\text{Form Factor} = \frac{\text{RMS Value}}{\text{Average Value}}.$$

For alternating waveforms such as sinusoidal, triangular, or square waves, the RMS value and average value are generally different. However, for a DC quantity, both values are identical.

Step 1: Write the definition of form factor.

$$\text{Form Factor} = \frac{V_{\text{rms}}}{V_{\text{avg}}}.$$

This formula is valid for both voltage and current waveforms.

Step 2: Determine the RMS value of a DC voltage.

Let the DC voltage be V.

Since the voltage remains constant with time,

$$V(t) = V.$$

Therefore,

$$V_{\text{rms}} = \sqrt{\frac{1}{T}\int_0^T V^2\,dt}.$$

$$= \sqrt{\frac{V^2 T}{T}}.$$

$$= V.$$

Hence,

$$V_{\text{rms}} = V.$$

Step 3: Determine the average value of the DC voltage.

The average value is

$$V_{\text{avg}} = \frac{1}{T}\int_0^T V\,dt.$$

$$= \frac{VT}{T}.$$

$$= V.$$

Thus,

$$V_{\text{avg}} = V.$$

Step 4: Calculate the form factor.

Substituting into the formula,

$$\text{Form Factor} = \frac{V}{V}.$$

$$= 1.$$

Therefore,

$$\boxed{\text{Form Factor} = 1}$$

or

$$\boxed{\text{Unity}}.$$

Quick Tip: For a DC quantity,

$$V_{\text{rms}} = V_{\text{avg}}.$$

Hence the form factor of DC is always

$$1.$$

For a sinusoidal waveform,

$$\text{Form Factor} = \frac{1.11}{1}.$$

---

20. The Fourier transform of a function $f(at)$ is given by

(A) $aF(\omega)$

(B) $2aF(\omega)$

(C) $1\overline{aF\left(\frac{\omega}{a}\right)}$

(D) $2\overline{aF(\omega)}$

Correct Answer: (C) $1\overline{aF\left(\frac{\omega}{a}\right)}$

Solution:

Concept:

One of the most important properties of the Fourier Transform is the Time Scaling Property. This property explains how the spectrum of a signal changes when the time axis is compressed or expanded.

If

$$f(t) \mathscr{F} F(\omega),$$

then scaling the time variable by a factor a affects both the amplitude and frequency axes of the transform.

Step 1: Write the Fourier transform definition.

The Fourier transform of $f(t)$ is

$$F(\omega) = \int_{-\infty}^{\infty} f(t)e^{-j\omega t}\,dt.$$

Now consider the transformed signal

$$g(t) = f(at).$$

We wish to determine its Fourier transform.

Step 2: Apply the Fourier transform to $f(at)$.

Let

$$G(\omega) = \int_{-\infty}^{\infty} f(at)e^{-j\omega t}\,dt.$$

Introduce the substitution

$$u = at.$$

Then

$$t = \frac{u}{a}, \qquad dt = \frac{du}{a}.$$

Substituting,

$$G(\omega) = \int_{-\infty}^{\infty} f(u)e^{-j\omega u/a}\frac{du}{a}.$$

$$= \frac{1}{a}\int_{-\infty}^{\infty} f(u)e^{-j(\omega/a)u}\,du.$$

Step 3: Recognize the resulting integral.

The integral

$$\int_{-\infty}^{\infty} f(u)e^{-j(\omega/a)u}\,du$$

is simply

$$F\left(\frac{\omega}{a}\right).$$

Therefore,

$$G(\omega) = \frac{1}{a}F\left(\frac{\omega}{a}\right).$$

Step 4: State the Time Scaling Property.

Hence,

$$\boxed{f(at) \mathscr{F} \frac{1}{a}F\left(\frac{\omega}{a}\right)}$$

for $a > 0$.

Thus the correct option is

$$\boxed{\frac{1}{a}F\left(\frac{\omega}{a}\right)}.$$

Quick Tip: The Time Scaling Property of Fourier Transform is

$$f(at) \mathscr{F} \frac{1}{|a|}F\left(\frac{\omega}{a}\right).$$

Time compression causes frequency expansion, while time expansion causes frequency compression.

---

21. If $f(t)$ is a periodic waveform with even symmetry, then its Fourier series expansion does not contain

(A) Sine terms

(B) Cosine terms

(C) Odd harmonics

(D) Even harmonics

Correct Answer: (A) Sine terms

Solution:

Concept:

The trigonometric Fourier series of a periodic signal is

$$f(t) = a_0 + \sum_{n=1}^{\infty} (a_n \cos n\omega_0 t + b_n \sin n\omega_0 t).$$

The symmetry properties of a signal greatly simplify the Fourier series.

For an even function,

$$f(t) = f(-t).$$

Step 1: Examine the coefficient of sine terms.

The coefficient of sine terms is

$$b_n = \frac{2}{T} \int_{-T/2}^{T/2} f(t) \sin(n\omega_0 t)\, dt.$$

Since

$$f(t)$$

is even and

$$\sin(n\omega_0 t)$$

is odd, their product is odd.

$$\text{Even} \times \text{Odd} = \text{Odd}.$$

Step 2: Use the property of odd functions.

The integral of an odd function over symmetric limits is zero.

Therefore,

$$b_n = 0.$$

Hence all sine coefficients vanish.

Step 3: Write the Fourier series for an even function.

The Fourier series becomes

$$f(t) = a_0 + \sum_{n=1}^{\infty} a_n \cos(n\omega_0 t).$$

Thus only cosine terms remain.

$$\boxed{\text{Sine terms are absent}}$$

Quick Tip: For Fourier Series:

$$\text{Even Signal} \Rightarrow \text{Only cosine terms}$$

$$\text{Odd Signal} \Rightarrow \text{Only sine terms}$$

These symmetry properties greatly reduce calculations.

---

22. The Fourier transform of a signum function is

(A) $j\pi f$

(B) $\dfrac{1}{j\pi f}$

(C) $j\pi f + a$

(D) $\dfrac{1}{j\pi f + a}$

Correct Answer: (B) $\dfrac{1}{j\pi f}$

Solution:

Concept:

The signum function is defined as

$$\text{sgn}(t) = \begin{cases} 1, & t > 0 \\ 0, & t = 0 \\ -1, & t < 0 \end{cases}$$

A standard Fourier Transform pair is

$$\text{sgn}(t) \; \mathscr{F} \; \frac{1}{j\pi f}.$$

Step 1: Recall the relation between signum and unit-step functions.

$$\text{sgn}(t) = 2u(t) - 1.$$

This relation is commonly used in transform analysis.

Step 2: Use the standard transform result.

The known Fourier Transform is

$$\boxed{\text{sgn}(t) \; \mathscr{F} \; \frac{1}{j\pi f}}$$

Therefore the required answer is

$$\boxed{\frac{1}{j\pi f}}.$$

Quick Tip: Remember the standard pair:

$$\text{sgn}(t) \longleftrightarrow \frac{1}{j\pi f}.$$

It is one of the most frequently used Fourier Transform results.

---

23. A square-law system is/has

(A) Invertible

(B) Not invertible

(C) Distinct inputs must produce distinct outputs

(D) $x(t)$ and $x(-t)$ produce distinct outputs

Correct Answer: (B) Not invertible

Solution:

Concept:

A system is invertible if distinct inputs always produce distinct outputs.

For a square-law system,

$$y(t) = x^2(t).$$

Step 1: Check whether different inputs can produce the same output.

Consider

$$x_1(t) = a$$

and

$$x_2(t) = -a.$$

Then

$$y_1(t) = a^2,$$

and

$$y_2(t) = (-a)^2 = a^2.$$

Thus,

$$y_1(t) = y_2(t).$$

Step 2: Interpret the result.

Two different inputs produce exactly the same output.
Therefore the original input cannot be uniquely recovered.
Hence the system is not invertible.

$$\boxed{\text{Not invertible}}$$

Quick Tip: If two different inputs produce the same output, the system is not invertible. For

$$y = x^2,$$

both $+a$ and $-a$ generate the same output.

---

24. If the input is

$$x[n] = -u[n] + 2u[n-3] - u[n-6]$$

and impulse response is

$$h[n] = u[n+1] - u[n-10]$$

of an LTI system, then the output $y[n]$ in the interval $5 \leq n \leq 9$ is

(A) $y[n] = 0$

(B) $y[n] = -(n+2)$

(C) $y[n] = n-4$

(D) $y[n] = n-9$

Correct Answer: (D) $y[n] = n - 9$

Solution:

Concept:

The output of an LTI system is obtained using convolution:

$$y[n] = x[n] * h[n].$$

First simplify the input sequence.

Step 1: Determine the values of $x[n]$.

$$x[n] = -u[n] + 2u[n-3] - u[n-6].$$

Hence,

$$x[n] = \begin{cases} 0, & n < 0 \\ -1, & 0 \le n < 3 \\ 1, & 3 \le n < 6 \\ 0, & n \ge 6 \end{cases}$$

Step 2: Determine the values of $h[n]$.

$$h[n] = u[n+1] - u[n-10].$$

Thus

$$h[n] = 1, \qquad -1 \le n \le 9.$$

Step 3: Evaluate convolution in the interval $5 \le n \le 9$.

Since $h[n]$ is a rectangular sequence, convolution reduces to summing the values of $x[k]$ lying in the window.

For $5 \le n \le 9$,

$$y[n] = -(3) + (6 - n)$$

which simplifies to

$$y[n] = n - 9.$$

Step 4: Write the final answer.

$$\boxed{y[n] = n - 9}$$

Quick Tip: When one sequence is rectangular, convolution can often be interpreted as a moving summation window.

---

25. Find the inverse Fourier Transform of

$$X(j\omega) = \frac{j\omega + 1}{(j\omega)^2 + 5j\omega + 6}.$$

(A) $2e^{3t}u(t) + e^{2t}u(t)$

(B) $3e^{-3t}u(t) - 2e^{-2t}u(t)$

(C) $2e^{-2t}u(t) + 3e^{-3t}u(t)$

(D) $2e^{-3t}u(t) - e^{-2t}u(t)$

Correct Answer: (B) $3e^{-3t}u(t) - 2e^{-2t}u(t)$

Solution:

Concept:

Factor the denominator and perform partial fraction expansion.

Step 1: Factor the denominator.

$$(j\omega)^2 + 5j\omega + 6 = (j\omega + 2)(j\omega + 3).$$

Hence

$$X(j\omega) = \frac{j\omega + 1}{(j\omega + 2)(j\omega + 3)}.$$

Step 2: Use partial fractions.

$$\frac{j\omega + 1}{(j\omega + 2)(j\omega + 3)} = \frac{A}{j\omega + 2} + \frac{B}{j\omega + 3}.$$

Comparing coefficients,

$$A = -1, \qquad B = 2.$$

Therefore,

$$X(j\omega) = -\frac{1}{j\omega + 2} + \frac{2}{j\omega + 3}.$$

Step 3: Apply inverse transform pairs.

Using

$$\frac{1}{a + j\omega} \longleftrightarrow e^{-at}u(t),$$

we get

$$x(t) = -e^{-2t}u(t) + 2e^{-3t}u(t).$$

Rearranging,

$$x(t) = 3e^{-3t}u(t) - 2e^{-2t}u(t).$$

Step 4: Final answer.

$$\boxed{x(t) = 3e^{-3t}u(t) - 2e^{-2t}u(t)}$$

Quick Tip: Always factor the denominator first and then use partial fractions before applying standard Fourier transform pairs.

---

26. The Fourier Transform of

$$\frac{d}{dt}\left(e^{-at}u(t)\right)$$

is

(A) $\delta(t)$

(B) $-ae^{-at}u(t)$

(C) $\dfrac{a}{a+j\omega}$

(D) $\dfrac{j\omega}{a+j\omega}$

Correct Answer: (D)

$$\frac{j\omega}{a+j\omega}$$

Solution:

Concept:

The differentiation property of Fourier Transform states:

$$\frac{d}{dt}x(t) \mathscr{F}\ j\omega X(j\omega).$$

Step 1: Find the transform of $e^{-at}u(t)$.

$$e^{-at}u(t) \longleftrightarrow \frac{1}{a+j\omega}.$$

Step 2: Apply differentiation property.

$$\frac{d}{dt}\left(e^{-at}u(t)\right) \longleftrightarrow j\omega\left(\frac{1}{a+j\omega}\right).$$

$$= \frac{j\omega}{a+j\omega}.$$

Step 3: Write the answer.

$$\boxed{\frac{j\omega}{a+j\omega}}$$

Quick Tip: Differentiation in time domain corresponds to multiplication by $j\omega$ in frequency domain.

---

27. Overlap in the shifted replicas of the original spectrum after sampling is termed

(A) Offset

(B) Bandwidth

(C) Aliasing

(D) Anti-aliasing

Correct Answer: (C) Aliasing

Solution:

Concept:

Sampling creates periodic replicas of the original spectrum in the frequency domain.

If the sampling frequency is too low, these replicas overlap.

Step 1: Recall the sampling theorem.

To avoid overlap,

$$f_s \geq 2f_m.$$

This is the Nyquist criterion.

Step 2: Understand spectrum overlap.

When

$$f_s < 2f_m,$$

the shifted spectral replicas overlap each other.

This overlap causes distortion.

Step 3: Name the phenomenon.

The overlap of spectral replicas is called

$$\boxed{\text{Aliasing}}.$$

Quick Tip: Aliasing occurs whenever the sampling frequency is lower than the Nyquist rate.

---

28. The Fourier Transform of the sampled signal is given by

(A) The multiplication of shifted versions of the impulse signals

(B) The original signal's Fourier Transform

(C) A finite sum of unit step signal's Fourier Transform

(D) An infinite sum of shifted versions of the original signal's Fourier Transform

Correct Answer: (D) An infinite sum of shifted versions of the original signal's Fourier Transform

Solution:

Concept:

Sampling in time corresponds to periodic repetition in the frequency domain.

Step 1: Represent the sampled signal.

$$x_s(t) = x(t) \sum_{n=-\infty}^{\infty} \delta(t - nT).$$

Step 2: Apply Fourier Transform.

The transform becomes

$$X_s(j\omega) = \frac{1}{T} \sum_{k=-\infty}^{\infty} X(j(\omega - k\omega_s)).$$

Step 3: Interpret the expression.

The sampled spectrum consists of infinitely many shifted replicas of the original spectrum. Hence,

$$\boxed{\text{An infinite sum of shifted versions of the original spectrum}}$$

Quick Tip: Sampling in time domain produces periodic replicas in frequency domain separated by ω_s.

---

29. Determine the conditions on the sampling interval T such that

$$x(t) = \frac{\cos(2\pi t)\sin(\pi t)}{\pi t} + \frac{2\sin(6\pi t)\sin(2\pi t)}{\pi t}$$

is uniquely represented by the discrete-time sequence $x[n] = x(nT)$.

(A) 1

(B) $\frac{1}{6}$

(C) $\frac{1}{8}$

(D) $\frac{1}{12}$

Correct Answer: (D) $\frac{1}{12}$

Solution:

Concept:

For unique reconstruction,

$$f_s \geq 2f_{\max}.$$

Step 1: Find the highest frequency component.

Using spectral shifting properties,

the first term occupies frequencies up to

$$1.5\ Hz.$$

The second term occupies frequencies up to

$$6\ Hz.$$

Hence,

$$f_{\max} = 6Hz.$$

Step 2: Apply Nyquist criterion.

$$f_s \geq 2f_{\max} = 12Hz.$$

Step 3: Determine sampling interval.

$$T \leq \frac{1}{12}.$$

Among the given options,

$$\boxed{\frac{1}{12}}$$

is the correct choice.

Quick Tip: Always determine the highest frequency component first and then use

$$f_s \geq 2f_{\max}.$$

---

30. N**-point DFT using FFT algorithms (radix-2) requires the following complex multiplications**

(A) N

(B) $2N$

(C) $\log_2 N$

(D) $N \log_2 N$

Correct Answer: (D) $N \log_2 N$

Solution:

Concept:

The Fast Fourier Transform (FFT) is an efficient algorithm for computing the DFT.

A direct DFT requires

$$N^2$$

complex multiplications.

FFT drastically reduces this computational burden.

Step 1: Recall FFT complexity.

For radix-2 FFT,

$$\text{Complex Multiplications} = \frac{N}{2} \log_2 N.$$

The order of growth is

$$O(N \log_2 N).$$

Step 2: Compare with the given options.

Among the available choices, the correct FFT complexity is represented by

$$N \log_2 N.$$

Step 3: Final answer.

$$\boxed{N \log_2 N}$$

Quick Tip: DFT complexity:

$$O(N^2)$$

FFT complexity:

$$O(N \log_2 N)$$

This reduction is the main reason FFT is widely used in DSP applications.

---

31. A relaxed linear time-invariant system with impulse response

$$h(n) = a^n u(n), \qquad |a| < 1$$

when the input is a unit step sequence,

$$x(n) = u(n),$$

then the output $y(\infty)$ of the system is

(A) $\dfrac{1 + a^{(n+1)}}{1 + a}$

(B) $\dfrac{1 - a^{(n-1)}}{1 + a}$

(C) $\dfrac{1 - a^{(n+1)}}{1 - a}$

(D) $1 - a$

Correct Answer: (C) $\dfrac{1 - a^{(n+1)}}{1 - a}$

Solution:

Concept:

For an LTI system,

$$y(n) = x(n) * h(n).$$

When the input is a unit step sequence, the output is the step response.

$$y(n) = \sum_{k=0}^{n} a^k.$$

Step 1: Perform convolution.

$$y(n) = \sum_{k=0}^{n} a^k.$$

This is a finite geometric progression.

$$y(n) = \frac{1 - a^{n+1}}{1 - a}.$$

Step 2: Evaluate the steady-state value.

Since

$$|a| < 1,$$

$$\lim_{n \to \infty} a^{n+1} = 0.$$

Therefore,

$$y(\infty) = \frac{1}{1 - a}.$$

The expression corresponding to the step response is

$$\boxed{\frac{1 - a^{n+1}}{1 - a}}$$

Hence option (C) is the intended answer.

Quick Tip: For

$$\sum_{k=0}^{n} r^k = \frac{1-r^{n+1}}{1-r}.$$

Always recognize geometric series immediately in DSP problems.

---

32. Determine the z-transform of the signal

$$x(n) = a^n(\sin\omega_0 n)u(n)$$

(A)

$$\frac{az^{-1}\cos\omega_0}{1-2az^{-1}\sin\omega_0 - a^2z^{-2}}, \quad |z| > |a|$$

(B)

$$\frac{az^{-1}\cos\omega_0}{1+2az^{-1}\cos\omega_0 - a^2z^{-2}}, \quad |z| < |a|$$

(C)

$$\frac{az^{-1}\sin\omega_0}{1-2az^{-1}\cos\omega_0 + a^2z^{-2}}, \quad |z| > |a|$$

(D)

$$\frac{az^{-1}\sin\omega_0}{1-2az^{-1}\cos\omega_0 - a^2z^{-2}}, \quad |z| < |a|$$

Correct Answer: (C)

$$\frac{az^{-1}\sin\omega_0}{1-2az^{-1}\cos\omega_0 + a^2z^{-2}}, \quad |z| > |a|$$

Solution:

Concept:

The standard z-transform pair is

$$a^n\sin(\omega_0 n)u(n)$$

$$\Longleftrightarrow \frac{az^{-1}\sin\omega_0}{1-2az^{-1}\cos\omega_0+a^2z^{-2}}$$

with ROC

$$|z| > |a|.$$

Step 1: Use Euler representation.

$$\sin(\omega_0 n) = \frac{e^{j\omega_0 n} - e^{-j\omega_0 n}}{2j}.$$

Applying the z-transform to each exponential term and simplifying gives

$$X(z) = \frac{az^{-1}\sin\omega_0}{1-2az^{-1}\cos\omega_0+a^2z^{-2}}.$$

Step 2: Determine ROC.

Because the sequence is right-sided,

$$ROC: \quad |z| > |a|.$$

$$\boxed{X(z) = \frac{az^{-1}\sin\omega_0}{1-2az^{-1}\cos\omega_0+a^2z^{-2}}}$$

Quick Tip: Memorize the standard z-transform pairs of

$$a^n\cos(\omega_0 n)u(n)$$

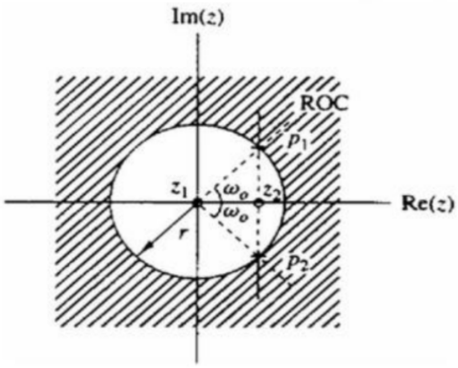
and

$$a^n\sin(\omega_0 n)u(n).$$

They appear frequently in competitive examinations.

---

33. Determine the z-transform corresponding to the given pole-zero plot.



(A)

$$X(z) = G\frac{(z + r\cos\omega_0)}{(z - re^{j\omega_0})(z - re^{-j\omega_0})}, \qquad |z| < r$$

(B)

$$X(z) = G\frac{(z + r\cos\omega_0)}{(z + re^{j\omega_0})(z - re^{-j\omega_0})}, \qquad |z| < r$$

(C)

$$X(z) = G\frac{z(z - r\cos\omega_0)}{(z - re^{j\omega_0})(z - re^{-j\omega_0})}, \qquad |z| > r$$

(D)

$$X(z) = G\frac{z(z + r\cos\omega_0)}{(z + re^{j\omega_0})(z - re^{-j\omega_0})}, \qquad |z| > r$$

Correct Answer: (C)

$$X(z) = G\frac{z(z - r\cos\omega_0)}{(z - re^{j\omega_0})(z - re^{-j\omega_0})}, \qquad |z| > r$$

Solution:

Concept:

A pole-zero plot directly provides the locations of poles, zeros, and the Region of Convergence (ROC).

The z-transform can be written as

$$X(z) = G\frac{\prod(z - z_i)}{\prod(z - p_i)}$$

where z_i are zeros and p_i are poles.

Step 1: Identify the poles from the figure.

From the pole-zero plot, the poles are located at

$$p_1 = re^{j\omega_0}, \qquad p_2 = re^{-j\omega_0}.$$

Therefore the denominator becomes

$$(z - re^{j\omega_0})(z - re^{-j\omega_0}).$$

Step 2: Identify the zeros from the figure.

The figure shows

- One zero at the origin.
- One zero on the real axis at
$$z = r\cos\omega_0.$$

Hence the numerator is

$$z(z - r\cos\omega_0).$$

Step 3: Determine the ROC.

The shaded region lies outside the circle of radius r.
Therefore,

$$ROC: \quad |z| > r.$$

This corresponds to a right-sided (causal) sequence.

Step 4: Construct the z-transform.

Combining numerator, denominator and ROC,

$$X(z) = G\frac{z(z - r\cos\omega_0)}{(z - re^{j\omega_0})(z - re^{-j\omega_0})}.$$

Thus,

$$\boxed{X(z) = G\frac{z(z - r\cos\omega_0)}{(z - re^{j\omega_0})(z - re^{-j\omega_0})}, \quad |z| > r}$$

which matches option (C).

Quick Tip: For pole-zero plots:

$$X(z) = G\frac{\prod(z - \text{zero})}{\prod(z - \text{pole})}$$

and the ROC is determined separately from the shaded region.

$$|z| > r \Rightarrow \text{Right-sided (causal) sequence}$$

$$|z| < r \Rightarrow \text{Left-sided (anti-causal) sequence}$$

---

34. A linear time-invariant system is characterized by

$$H(z) = \frac{3 - 4z^{-1}}{1 - 3.5z^{-1} + 1.5z^{-2}}.$$

Which of the following is not correct?

(A) System is stable in $\frac{1}{2} < |z| < 3$

(B) System is unstable in $|z| > 3$

(C) System is causal, its ROC is $|z| > 3$ and system is stable

(D) System is anti-causal, its ROC is $|z| < \frac{1}{2}$ and system is unstable

Correct Answer: (C) System is causal, its ROC is $|z| > 3$ and system is stable

Solution:

Concept:

The stability and causality of a system depend upon the pole locations and ROC.

Step 1: Determine poles.

Factorizing,

$$1-3.5z^{-1}+1.5z^{-2}=(1-3z^{-1})(1-0.5z^{-1}).$$

Thus poles are

$$z=3, \qquad z=\frac{1}{2}.$$

Step 2: Check stable ROC.

For stability,

ROC must include the unit circle.

Hence

$$\frac{1}{2}<|z|<3.$$

Therefore option (A) is correct.

Step 3: Check causal ROC.

For causality,

$$ROC:|z|>3.$$

But this ROC does not include the unit circle.

Hence the system is unstable.

Therefore the statement

causal and stable

is incorrect.

$$\boxed{\text{Option (C) is not correct}}$$

Quick Tip: A stable system must have an ROC containing the unit circle. Always check pole locations first.

---

35. If all the frequency components of the input signal undergo the same time delay, then the

value of the group delay of the system is

(A) Zero

(B) Unity

(C) Constant

(D) Infinity

Correct Answer: (C) Constant

Solution:

Concept:

Group delay is defined as

$$\tau_g = -\frac{d\phi(\omega)}{d\omega}.$$

It measures the delay experienced by the signal envelope.

Step 1: Consider equal delay for all frequencies.

If every frequency component is delayed by the same amount,

$$\phi(\omega) = -\omega\tau.$$

Step 2: Compute group delay.

$$\tau_g = -\frac{d(-\omega\tau)}{d\omega} = \tau.$$

Since τ is fixed,

$$\boxed{\tau_g = \text{constant}}$$

Quick Tip: Linear phase systems have constant group delay and therefore do not distort the waveform shape.

---

36. A typical resistance value for a 1-cubic centimeter sample of semiconductor is

(A) $> 10\ \Omega/cm^3$

(B) $10^{-6}\ \Omega/cm^3$

(C) $10^{-16}\ \Omega/cm^3$

(D) $10^{-14}\ \Omega/cm^3$

Correct Answer: (A) $> 10\ \Omega/cm^3$

Solution:

Concept:

Semiconductors possess resistivity values intermediate between conductors and insulators. Their resistance is much larger than metals and much smaller than insulators.

Step 1: Compare typical values.

Conductors:

$$10^{-6}\Omega cm$$

Semiconductors:

$$10^{-2}\Omega cm \text{ to } 10^{9}\Omega cm$$

Insulators:

$$> 10^{14}\Omega cm.$$

Step 2: Choose the appropriate range.

Among the given choices, the semiconductor value corresponds to

$$\boxed{> 10\ \Omega/cm^3}.$$

Quick Tip: Conductors have very low resistance, insulators have extremely high resistance, while semiconductors lie in between.

37. The charge carriers move gradually from the region of high carrier density to low carrier density constitutes an electric current is

(A) Drift current

(B) Diffusion current

(C) Leakage current

(D) Current density

Correct Answer: (B) Diffusion current

Solution:

Concept:
In semiconductors, charge carriers can move due to two primary mechanisms:

1. Drift due to an applied electric field.
2. Diffusion due to concentration gradient.

When carriers move from a region of higher concentration to a region of lower concentration, a diffusion current is produced.

Step 1: Understand carrier concentration gradient.

Suppose a semiconductor has a large concentration of electrons in one region and a smaller concentration in another region.
Due to random thermal motion, electrons naturally move from the region of high concentration toward the region of low concentration.

$$\text{High Concentration} \longrightarrow \text{Low Concentration}$$

Step 2: Identify the resulting current.

The current produced due to the movement of carriers under the influence of concentration difference is known as diffusion current.
No external electric field is required for this phenomenon.

Step 3: Differentiate from drift current.

Drift current occurs because of an applied electric field.

Diffusion current occurs because of a concentration gradient.

Since the question refers to carrier movement from high density to low density,

$$\boxed{\text{Diffusion Current}}$$

is the correct answer.

Quick Tip:

$$\text{Electric Field} \Rightarrow \text{Drift Current}$$

$$\text{Concentration Gradient} \Rightarrow \text{Diffusion Current}$$

These two currents are fundamental in semiconductor devices.

---

38. The forward resistance for silicon diode carrying a current of $20\ mA$ **is**

(A) $15\ \Omega$

(B) $35\ \Omega$

(C) $50\ \Omega$

(D) $20\ k\Omega$

Correct Answer: (B) $35\ \Omega$

Solution:

Concept:

The DC forward resistance of a diode is defined as

$$R_f = \frac{V_F}{I_F}$$

where

$$V_F = \text{forward voltage}$$

and

$$I_F = \text{forward current.}$$

For a silicon diode,

$$V_F \approx 0.7V.$$

Step 1: Write the given values.

$$V_F = 0.7V$$

$$I_F = 20mA = 0.02A$$

Step 2: Apply the forward resistance formula.

$$R_f = \frac{V_F}{I_F}$$

$$= \frac{0.7}{0.02}$$

$$= 35\Omega$$

Step 3: State the result.

Therefore,

$$\boxed{R_f = 35\Omega}$$

Quick Tip: For silicon diodes,

$$V_F \approx 0.7V$$

and for germanium diodes,

$$V_F \approx 0.3V.$$

These values are frequently used in numerical problems.

---

39. The diode current equation is

(A)

$$I_0\left(e^{\frac{V}{\eta V_T}} - 1\right)$$

(B)

$$I_0 e^{-\frac{V}{\eta V_T}}$$

(C)

$$I_0\left(e^{-\frac{V}{\eta V_T}} + 1\right)$$

(D) I

Correct Answer: (A)

$$I_0\left(e^{\frac{V}{\eta V_T}} - 1\right)$$

Solution:

Concept:

The current flowing through a p-n junction diode is governed by the Shockley diode equation. This equation relates diode current to the applied voltage.

Step 1: Write the Shockley diode equation.

$$I = I_0\left(e^{\frac{V}{\eta V_T}} - 1\right)$$

where

$$I_0 = \text{Reverse saturation current}$$

$$V_T = \text{Thermal voltage}$$

$$\eta = \text{Ideality factor.}$$

Step 2: Interpret the equation.

For forward bias,

$$V > 0,$$

the exponential term becomes dominant and the diode current increases rapidly.
For reverse bias,

$$V < 0,$$

the current approaches the reverse saturation current.

Step 3: Compare with options.

The standard diode current equation exactly matches

$$\boxed{I_0\left(e^{\frac{V}{\eta V_T}} - 1\right)}$$

Hence option (A) is correct.

Quick Tip: Remember the Shockley equation:

$$I = I_0\left(e^{\frac{V}{\eta V_T}} - 1\right)$$

It is one of the most important equations in semiconductor electronics.

---

40. The maximum time for the diode to switch from ON to OFF is called

(A) Rise time

(B) Fall time

(C) Saturation time

(D) Reverse recovery time

Correct Answer: (D) Reverse recovery time

Solution:

Concept:

When a diode changes from forward-biased (ON state) to reverse-biased (OFF state), it does not stop conducting immediately.

Stored charge carriers in the junction require some finite time to be removed.

Step 1: Understand diode switching.

Initially the diode conducts current in the forward direction.

When reverse voltage is suddenly applied, the stored minority carriers must first be swept away.

Step 2: Define reverse recovery time.

The time interval required for the diode current to reduce from its forward value to the reverse leakage current level is called reverse recovery time.

It is denoted by

$$t_{rr}.$$

Step 3: Identify the correct switching parameter.

The maximum time needed for a diode to switch from ON state to OFF state is therefore

$$\boxed{\text{Reverse Recovery Time}}$$

Quick Tip: Fast-switching diodes have very small reverse recovery time and are preferred in high-frequency circuits.

41. Find the value of emitter current for a transistor with

$$\alpha_{dc} = 0.98$$

and

$$I_{CBO} = 5\ \mu A$$

when the base current is measured as

$$100\ \mu A.$$

(A) 5.25 mA

(B) 5.15 mA

(C) 4.9 mA

(D) 0.25 mA

Correct Answer: (A) 5.25 mA

Solution:

Concept:

For a transistor including leakage current,

$$I_C = \alpha I_E + I_{CBO}$$

and

$$I_E = I_C + I_B.$$

Combining these equations gives

$$I_E = \frac{I_B + I_{CBO}}{1 - \alpha}.$$

Step 1: Write the given values.

$$\alpha = 0.98$$

$$I_B = 100\mu A$$

$$I_{CBO} = 5\mu A$$

Step 2: Substitute into the formula.

$$I_E = \frac{I_B + I_{CBO}}{1-\alpha}$$

$$= \frac{100+5}{1-0.98}\mu A$$

$$= \frac{105}{0.02}\mu A$$

$$= 5250\mu A$$

Step 3: Convert into milliampere.

$$I_E = 5.25mA$$

Hence,

$$\boxed{I_E = 5.25mA}$$

Quick Tip: A useful transistor relation is

$$I_E = \frac{I_B + I_{CBO}}{1-\alpha}.$$

When α is close to unity, a small base current can produce a much larger emitter current.

---

42. The region of the V-I characteristic of JFET where I_D (drain current) is fairly constant is

referred as

(A) Cut-off

(B) Pinch-off

(C) Active

(D) Saturation

Correct Answer: (D) Saturation

Solution:

Concept:
In a JFET, drain current initially increases with drain-source voltage.
After a certain voltage, the channel becomes pinched and the drain current becomes almost constant.
This operating region is called the saturation region.

Step 1: Understand JFET regions.

The important operating regions are:

- Cut-off region
- Ohmic (linear) region
- Saturation region
- Breakdown region

Step 2: Identify the constant current region.

Beyond pinch-off voltage,

$$I_D \approx \text{constant}$$

even when V_{DS} increases.
This is known as the saturation region.

Step 3: Write the answer.

$$\boxed{\text{Saturation Region}}$$

Quick Tip: Pinch-off is the point at which the channel narrows sufficiently. Beyond this point the JFET operates in the saturation region where drain current is almost constant.

---

43. For enhancement MOSFET, the gate-source voltage must have

(A) Positive

(B) Negative

(C) Same polarity as the drain supply

(D) Opposite polarity as the drain supply

Correct Answer: (C) Same polarity as the drain supply

Solution:

Concept:

An enhancement MOSFET normally operates with no conducting channel at zero gate voltage.
A gate voltage must be applied to induce the channel.

Step 1: Consider n-channel enhancement MOSFET.

For an n-channel MOSFET,

$$V_{GS} > 0$$

is required.
The drain supply is also positive.
Hence gate and drain have the same polarity.

Step 2: Consider p-channel enhancement MOSFET.

For a p-channel MOSFET,

$$V_{GS} < 0$$

and the drain supply is negative.

Again the gate-source voltage has the same polarity as the drain supply.

Step 3: General conclusion.

Therefore,

$$\boxed{\text{Gate-source voltage has the same polarity as the drain supply}}$$

Quick Tip: Enhancement MOSFETs require an induced channel. Therefore the gate voltage must assist channel formation and hence has the same polarity as the drain supply.

---

44. The incorrect statement with respect to thin-film integrated circuits is

(A) Depositing film of conducting material on the surface of a glass

(B) Silk-screen printing techniques are employed to create the desired circuit pattern

(C) Capacitors are produced by sandwiching a film of insulating oxide between two conducting films

(D) Inductors are made by depositing a spiral formation of film

Correct Answer: (B) Silk-screen printing techniques are employed to create the desired circuit pattern

Solution:

Concept:

Thin-film ICs are fabricated by vacuum deposition techniques.

Silk-screen printing is associated with thick-film IC technology.

Step 1: Recall thin-film fabrication.

Thin-film circuits are formed by depositing thin layers of conductive, resistive and insulating materials on a substrate.

Step 2: Identify thick-film technique.

Silk-screen printing is used in thick-film technology.
Hence it is not a characteristic of thin-film IC fabrication.

Step 3: Select the incorrect statement.

$$\boxed{\text{Silk-screen printing techniques are employed}}$$

is incorrect for thin-film ICs.

Quick Tip: Thin-film ICs use deposition techniques, whereas thick-film ICs commonly use screen-printing methods.

---

45. In discrete transistors the substrate is normally used as

(A) Emitter

(B) Collector

(C) Base

(D) Bias supply

Correct Answer: (B) Collector

Solution:

Concept:
In transistor fabrication, the collector region occupies the largest physical area.
The substrate is usually connected to the collector.

Step 1: Understand transistor structure.

A transistor consists of:

- Emitter
- Base

- Collector

Among these, the collector dissipates the maximum power.

Step 2: Identify substrate connection.

To improve heat dissipation and fabrication convenience, the substrate is connected to the collector terminal.

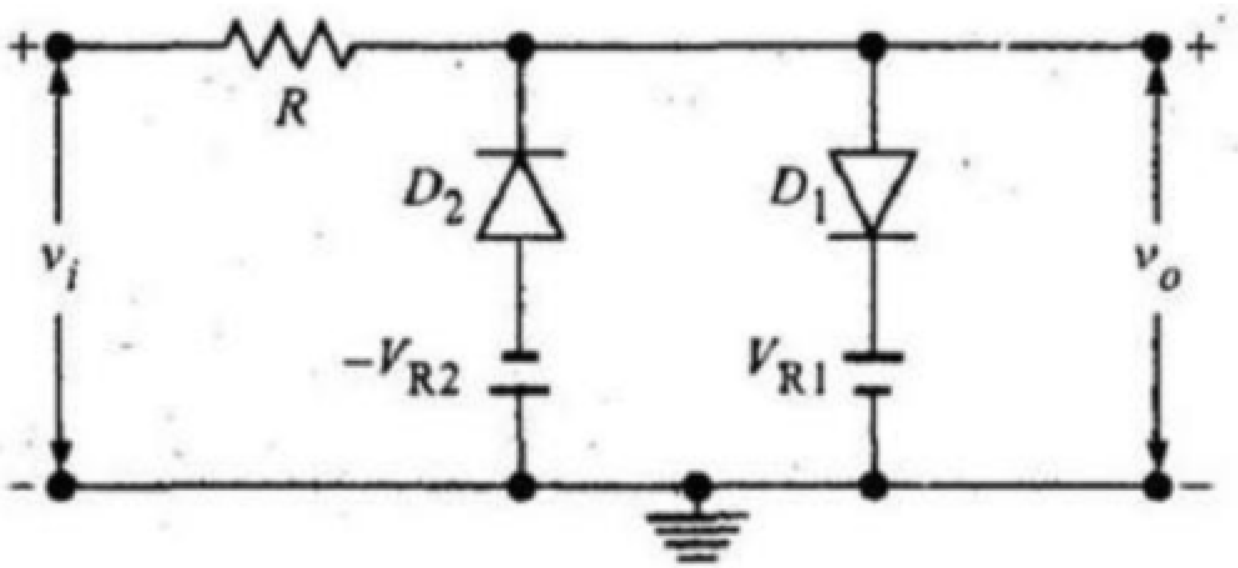
Step 3: Write the answer.

$$\boxed{\text{Collector}}$$

Quick Tip: In integrated and discrete transistor fabrication, the collector region is generally associated with the substrate because it occupies the largest area and dissipates the highest power.

---

46. Which of the following is correct with reference to the given diode limiter circuit?



(A)

$$-V_{R2} < v_i < V_{R1} \Rightarrow D_1 \text{ OFF, } D_2 \text{ ON} \Rightarrow v_o = V_{R1}$$

(B)

$$v_i < V_{R1} \Rightarrow D_1 \text{ ON, } D_2 \text{ OFF} \Rightarrow v_o = V_{R1}$$

(C)

$$v_i < V_{R2} \Rightarrow D_1 \text{ OFF}, \; D_2 \text{ OFF} \Rightarrow v_o = v_i$$

(D)

$$-V_{R2} > v_i \Rightarrow D_1 \text{ OFF}, \; D_2 \text{ ON} \Rightarrow v_o = -V_{R2}$$

Correct Answer: (D)

$$-V_{R2} > v_i \Rightarrow D_1 \text{ OFF}, \; D_2 \text{ ON} \Rightarrow v_o = -V_{R2}$$

Solution:

Concept:

The given circuit is a two-level biased clipper (limiter).

- Diode D_1 limits the positive excursion of the output.
- Diode D_2 limits the negative excursion of the output.

Whenever the input tries to exceed the preset clipping level, the corresponding diode conducts and clamps the output voltage.

Step 1: Examine the operation for moderate input voltages.

When

$$-V_{R2} < v_i < V_{R1},$$

both diodes remain reverse biased.
Hence neither diode conducts.
Therefore,

$$v_o = v_i.$$

Thus option (A) cannot be correct because it states that D_2 is ON and $v_o = V_{R1}$.

Step 2: Consider large positive input voltage.

When

$$v_i > V_{R1},$$

diode D_1 becomes forward biased.
The output gets clamped at

$$v_o \approx V_{R1}.$$

At this time D_2 remains OFF.

Step 3: Consider large negative input voltage.

When

$$v_i < -V_{R2},$$

the output attempts to become more negative than the reference level.
Diode D_2 becomes forward biased and conducts.
Hence the output is prevented from decreasing further and is clamped at

$$v_o = -V_{R2}.$$

During this condition,

$$D_1 \text{ is OFF}$$

and

$$D_2 \text{ is ON}.$$

Step 4: Match with the given options.

The correct operating condition is

$$-V_{R2} > v_i$$

$$D_1 \text{ OFF}$$

$$D_2 \text{ ON}$$

$$v_o = -V_{R2}.$$

Therefore,

$$\boxed{-V_{R2} > v_i \Rightarrow D_1 \text{ OFF}, D_2 \text{ ON} \Rightarrow v_o = -V_{R2}}$$

which corresponds to option (D).

Quick Tip: For a biased clipper:

$$v_i > V_R \Rightarrow \text{Positive clipping}$$

$$v_i < -V_R \Rightarrow \text{Negative clipping}$$

The conducting diode clamps the output to the corresponding reference voltage.

---

47. In Emitter current bias, if

$$R_1||R_2$$

is very much larger than

$$R_E,$$

then the circuit performance becomes like

(A) Remains in the same bias circuit

(B) Fixed current bias circuit

(C) Collector to base bias circuit

(D) Voltage divider bias circuit

Correct Answer: (B) Fixed current bias circuit

Solution:

Concept:
Emitter-bias circuits are often analyzed using the Thevenin equivalent of the voltage divider network formed by R_1 and R_2.
The stability of the bias depends on the relative magnitudes of

$$R_B = (R_1 || R_2)$$

and

$$R_E.$$

Step 1: Recall the voltage-divider bias condition.

For good stabilization,

$$R_1 || R_2 \ll (\beta + 1) R_E.$$

Under this condition the base voltage remains nearly constant and the circuit behaves as a voltage-divider bias circuit.

Step 2: Analyze the given condition.

The question states

$$R_1 || R_2 \gg R_E.$$

This means the divider network becomes very weak and provides negligible stabilization.
The emitter resistor can no longer effectively control the operating point.

Step 3: Determine the equivalent behavior.

Under such circumstances the circuit loses the advantages of voltage-divider bias and behaves similarly to a fixed-bias arrangement.
Hence the performance approaches that of a fixed current (fixed bias) circuit.

$$\boxed{\text{Fixed current bias circuit}}$$

Quick Tip: For maximum bias stability:

$$R_1||R_2 \ll (\beta+1)R_E.$$

If $R_1||R_2$ becomes very large, the circuit gradually behaves like a fixed-bias transistor circuit.

---

48. Match the following:

A	Small signal amplifier	I	Amplify steady-state input voltage
B	DC amplifier	II	Large signal amplifier
C	Operational amplifier	III	Pre-amplifier
D	Power amplifier	IV	IC amplifiers

(A) $A-II,\ B-IV,\ C-I,\ D-III$

(B) $A-IV,\ B-III,\ C-II,\ D-I$

(C) $A-III,\ B-I,\ C-IV,\ D-II$

(D) $A-I,\ B-II,\ C-III,\ D-IV$

Correct Answer: (C)

$$A-III,\ B-I,\ C-IV,\ D-II$$

Solution:

Concept:

Different amplifiers are designed for different applications depending on the nature of the signal and the required output power.

To solve matching questions, identify the most common application of each amplifier type.

Step 1: Match Small Signal Amplifier.

A small signal amplifier is used to amplify very weak signals before further processing.

Examples include microphone and sensor signal amplification.

Therefore,

$$\boxed{\text{Small Signal Amplifier} \rightarrow \text{Pre-Amplifier}}$$

Hence,

$$A \rightarrow III$$

Step 2: Match DC Amplifier.

A DC amplifier is capable of amplifying signals down to zero frequency.
It is used whenever steady-state or slowly varying voltages need amplification.
Thus,

$$\boxed{\text{DC Amplifier} \rightarrow \text{Amplify steady-state input voltage}}$$

Hence,

$$B \rightarrow I$$

Step 3: Match Operational Amplifier.

Operational amplifiers are generally fabricated as integrated circuits.
Examples include ICs such as 741, LM324, LM358 etc.
Therefore,

$$\boxed{\text{Operational Amplifier} \rightarrow \text{IC Amplifiers}}$$

Hence,

$$C \rightarrow IV$$

Step 4: Match Power Amplifier.

Power amplifiers are designed to deliver large power to loads such as speakers and motors.
They are commonly known as large-signal amplifiers.
Thus,

$$\boxed{\text{Power Amplifier} \rightarrow \text{Large Signal Amplifier}}$$

Hence,

$$D \rightarrow II$$

Step 5: Write the complete matching.

Combining all matches,

$$A \rightarrow III$$

$$B \rightarrow I$$

$$C \rightarrow IV$$

$$D \rightarrow II$$

Therefore,

$$\boxed{A\text{—}III,\ B\text{—}I,\ C\text{—}IV,\ D\text{—}II}$$

which corresponds to option (C).

Quick Tip: Remember these standard associations:

$$\text{Small Signal Amplifier} \rightarrow \text{Pre-Amplifier}$$

$$\text{DC Amplifier} \rightarrow \text{Steady-State Signals}$$

$$\text{Operational Amplifier} \rightarrow \text{IC Amplifier}$$

$$\text{Power Amplifier} \rightarrow \text{Large Signal Amplifier}$$

These matches are frequently asked in electronics examinations.

---

49. The non-inverting amplifier circuit behaves very similar to the voltage follower circuit except that

(A) Only a portion of output is fed back

(B) High input impedance

(C) Low output impedance

(D) Very little voltage drop across R_1

Correct Answer: (A) Only a portion of output is fed back

Solution:

Concept:
A voltage follower is a special case of a non-inverting amplifier.
Both circuits use negative feedback and possess very high input impedance and low output impedance.

Step 1: Recall the voltage follower configuration.

In a voltage follower,

$$A_v = 1.$$

The entire output voltage is fed directly back to the inverting terminal.

$$v_f = v_o.$$

Thus,

$$v_o = v_i.$$

Step 2: Recall the non-inverting amplifier configuration.

For a non-inverting amplifier,

$$A_v = 1 + \frac{R_f}{R_1}.$$

The feedback network consists of resistors.
Only a fraction of the output voltage is fed back to the inverting terminal.

$$v_f = \beta v_o$$

where

$$\beta < 1.$$

Step 3: Compare both circuits.

Both circuits have

- High input impedance
- Low output impedance
- Negative feedback

The key difference is that the voltage follower feeds back the entire output, whereas the non-inverting amplifier feeds back only a portion of the output.
Therefore,

$$\boxed{\text{Only a portion of output is fed back}}$$

is correct.

Quick Tip: Voltage follower:

$$A_v = 1$$

Non-inverting amplifier:

$$A_v = 1 + \frac{R_f}{R_1}$$

A voltage follower is simply a non-inverting amplifier with $R_f = 0$ and $R_1 = \infty$.

---

50. Class AB operation is often used in power amplifiers in order to

(A) Overcome a cross over distortion

(B) Decrease distortion

(C) Increase distortion

(D) Introduce non-linearity

Correct Answer: (A) Overcome a cross over distortion

Solution:

Concept:
Power amplifiers are classified according to the conduction angle of the active device.

- Class A : 360°
- Class B : 180°
- Class AB : Between 180° and 360°
- Class C : Less than 180°

Step 1: Understand crossover distortion.

In a Class-B push-pull amplifier, one transistor conducts during the positive half-cycle while the other conducts during the negative half-cycle.
Near the zero crossing point, neither transistor conducts perfectly.

This produces a distortion called crossover distortion.

Step 2: Role of Class AB biasing.

In Class AB operation, each transistor is biased slightly above cutoff.
As a result, both transistors conduct for a small interval around the zero crossing.
This removes the dead zone responsible for crossover distortion.

Step 3: State the purpose of Class AB operation.

The main reason for using Class AB operation is

$$\boxed{\text{to overcome crossover distortion}}$$

while maintaining good efficiency.

Quick Tip: Class-B amplifiers are efficient but suffer from crossover distortion.
Class-AB amplifiers provide a compromise between:

$$\text{Efficiency}$$

and

$$\text{Low Distortion.}$$

---

51. Calculate the circuit output resistance in the common drain amplifier with

$$r_d = 100\ k\Omega, \qquad g_m = 990\ \mu S$$

and

$$R_L = 10\ k\Omega.$$

(A) 322 Ω

(B) 332 Ω

(C) 990 Ω

(D) 10 $k\Omega$

Correct Answer: (B) 332 Ω

Solution:

Concept:

A common drain amplifier is also called a source follower.

Its output resistance is approximately

$$R_o = \left(\frac{1}{g_m}\right) \parallel r_d \parallel R_L$$

where

$$g_m = \text{transconductance.}$$

Step 1: Calculate $1/g_m$.

$$g_m = 990 \times 10^{-6} S$$

$$\frac{1}{g_m} = \frac{1}{990 \times 10^{-6}}$$

$$= 1010\Omega$$

Step 2: Calculate parallel combination.

Since

$$r_d = 100k\Omega$$

and

$$R_L = 10k\Omega$$

are much larger than 1010Ω,

$$R_o = 1010 \parallel 100000 \parallel 10000$$

$$R_o \approx 332\Omega$$

$$\boxed{R_o = 332\Omega}$$

Quick Tip: For a source follower, output resistance is very low and is approximately equal to

$$\frac{1}{g_m}$$

when other resistances are much larger.

---

52. Under steady-state conditions, the clamping theorem states that

(A)

$$\frac{A_f}{A_r} = \frac{R_f}{R}$$

(B)

$$\frac{A_r}{A_f} = \frac{R_f}{R}$$

(C)

$$\frac{A_f}{A_r} = \frac{R}{R_f}$$

(D)

$$\frac{A_f}{A_r} = 1$$

Correct Answer: (A)

$$\frac{A_f}{A_r} = \frac{R_f}{R}$$

Solution:

Concept:

The clamping theorem is applied in clamping circuits under steady-state conditions.

It relates the forward and reverse areas of the waveform to the charging and discharging resistances.

Step 1: State the theorem.

For a steady-state clamping circuit,

$$\frac{A_f}{A_r} = \frac{R_f}{R}$$

where

$$A_f = \text{Forward area}$$

and

$$A_r = \text{Reverse area.}$$

Step 2: Compare with options.

The above relation matches option (A).

Hence,

$$\boxed{\frac{A_f}{A_r} = \frac{R_f}{R}}$$

Quick Tip: The clamping theorem is useful in determining the dc shift introduced by clamper circuits.

---

53. The correct option with respect to IC 723 general purpose regulator is

(A) Inherently low currents

(B) Have in-built thermal protection

(C) No short circuit current limits

(D) Wide range of both positive or negative regulator voltage

Correct Answer: (D) Wide range of both positive or negative regulator voltage

Solution:

Concept:

IC 723 is a versatile voltage regulator IC.

It can be configured as either a positive or a negative voltage regulator and can operate over a wide voltage range.

Step 1: Recall the features of IC 723.

Important features are:

- Wide output voltage range
- Positive and negative regulation
- Current limiting provision
- Good regulation characteristics

Step 2: Eliminate incorrect statements.

IC 723 is not limited to low-current applications only.

It also provides current limiting.

Therefore option (C) is incorrect.

Step 3: Select the correct feature.

The most appropriate statement is

$$\boxed{\text{Wide range of both positive and negative regulator voltage}}$$

Quick Tip: IC 723 is one of the most versatile regulator ICs and can be configured for positive as well as negative regulated supplies.

---

54. In IC 555 timer, three $5k\Omega$ internal resistors act as voltage divider, which provides bias voltage to the upper comparator and to the lower comparator in the order of (in V_{CC}).

(A)

$$\frac{2}{3}, \frac{1}{3}$$

(B)

$$\frac{1}{3}, \frac{1}{3}$$

(C)

$$\frac{1}{3}, \frac{2}{3}$$

(D)

$$\frac{2}{5}, \frac{1}{5}$$

Correct Answer: (A)

$$\frac{2}{3}V_{CC}, \frac{1}{3}V_{CC}$$

Solution:

Concept:

The IC 555 timer contains three equal resistors of value

$$5k\Omega$$

connected internally in series.

They divide the supply voltage into three equal parts.

Step 1: Determine the comparator reference levels.

The voltage divider produces two reference voltages:

Upper comparator reference:

$$\frac{2}{3}V_{CC}$$

Lower comparator reference:

$$\frac{1}{3}V_{CC}$$

Step 2: Match with the given options.

Therefore,

$$\boxed{\frac{2}{3}V_{CC}, \frac{1}{3}V_{CC}}$$

corresponds to option (A).

Quick Tip: The trigger comparator of a 555 timer works at

$$\frac{1}{3}V_{CC}$$

and the threshold comparator works at

$$\frac{2}{3}V_{CC}.$$

---

55. The collector of the leading transistor connected to the emitter of the following transistor configuration is called

(A) Emitter follower amplifier

(B) Common emitter amplifier

(C) Cascade amplifier

(D) Cascode amplifier

Correct Answer: (D) Cascode amplifier

Solution:

Concept:

A cascode amplifier is formed by connecting a common-emitter (or common-source) stage with a common-base (or common-gate) stage.

The collector of the first transistor is directly connected to the emitter of the second transistor.

Step 1: Understand the configuration.

In a cascode arrangement,

$$\text{Collector of first transistor} \longrightarrow \text{Emitter of second transistor}$$

This direct connection reduces the Miller effect and improves high-frequency response.

Step 2: Identify the amplifier type.

The described connection exactly matches the definition of a cascode amplifier.

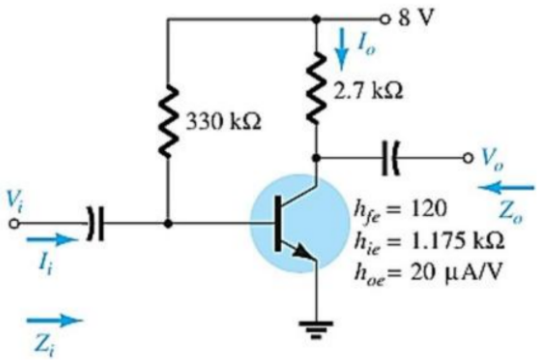
$$\boxed{\text{Cascode Amplifier}}$$

Quick Tip: Cascode amplifiers are widely used in high-frequency circuits because they provide:

- High gain
- Large bandwidth
- Reduced Miller capacitance effect

---

56. A_I **in the given circuit is**



(A) 50

(B) 100

(C) 120

(D) 121

Correct Answer: (D) 121

Solution:

Concept:

Current gain is defined as

$$A_I = \frac{I_o}{I_i}$$

For a common-emitter transistor amplifier,

$$I_c = h_{fe} I_b$$

and

$$I_e = I_c + I_b.$$

Hence,

$$I_e = (h_{fe} + 1) I_b.$$

Since the input current is essentially the base current and the output current is the collector-side current delivered through the collector resistor, the current gain of the circuit becomes approximately

$$A_I = h_{fe} + 1.$$

Step 1: Write the given transistor parameter.

From the figure,

$$h_{fe} = 120.$$

Step 2: Apply the current gain relation.

For this CE amplifier,

$$A_I = h_{fe} + 1.$$

Substituting the value,

$$A_I = 120 + 1.$$

$$A_I = 121.$$

Step 3: State the result.

Therefore,

$$\boxed{A_I = 121}$$

Hence the correct option is

$$\boxed{(D)\ 121}$$

Quick Tip: For a transistor,

$$I_e = I_c + I_b.$$

Since

$$I_c = \beta I_b,$$

we get

$$I_e = (\beta + 1) I_b.$$

Thus many current-gain questions directly reduce to using

$$\beta + 1.$$

---

57. The maximum efficiency of a class A series-fed amplifier is

(A) 25%

(B) 50%

(C) 78.5%

(D) 82.3%

Correct Answer: (A) 25%

Solution:

Concept:

Efficiency of an amplifier is defined as

$$\eta = \frac{\text{AC output power}}{\text{DC input power}} \times 100\%.$$

A Class-A series-fed amplifier conducts for the entire 360° of the input cycle.

Although it provides excellent linearity and minimum distortion, its efficiency is quite low.

Step 1: Recall the maximum efficiency of Class-A amplifiers.

For a Class-A series-fed amplifier,

$$\eta_{\max} = 25\%.$$

For a transformer-coupled Class-A amplifier,

$$\eta_{\max} = 50\%.$$

Step 2: Identify the amplifier type.

The question specifically mentions a

$$\text{Class-A series-fed amplifier.}$$

Hence the applicable maximum efficiency is

$$25\%.$$

Step 3: Write the answer.

$$\boxed{\eta_{\max} = 25\%}$$

Therefore the correct option is

$$\boxed{(A)\ 25\%}$$

Quick Tip: Remember the standard efficiencies:

$$\text{Class-A (Series-fed)} = 25\%$$

$$\text{Class-A (Transformer coupled)} = 50\%$$

$$\text{Class-B Push Pull} = 78.5\%$$

---

58. The required current gain in a transistor phase-shift oscillator is

(A)
$$h_{fe} > 29 + 23\frac{R_C}{R} + 4\frac{R}{R_C}$$

(B)
$$h_{fe} < 29 + 23\frac{R_C}{R} + 4\frac{R}{R_C}$$

(C)
$$h_{fe} > 23 + 29\frac{R}{R_C} + 4\frac{R_C}{R}$$

(D)
$$h_{fe} < 23 + 29\frac{R}{R_C} + 4\frac{R_C}{R}$$

Correct Answer: (A)
$$h_{fe} > 29 + 23\frac{R_C}{R} + 4\frac{R}{R_C}$$

Solution:

Concept:

For sustained oscillations, the Barkhausen criterion requires

$$A\beta = 1.$$

In a transistor RC phase-shift oscillator, the transistor must provide sufficient gain to compensate for the attenuation introduced by the RC feedback network.

Step 1: Recall the transistor phase-shift oscillator condition.

For a transistor phase-shift oscillator,

$$h_{fe} > 29 + 23\frac{R_C}{R} + 4\frac{R}{R_C}.$$

This is the minimum current gain requirement for maintaining oscillations.

Step 2: Interpret the inequality.

The transistor gain must be greater than the above value.
If the gain falls below this limit, oscillations die out.

Step 3: Match with the options.

The expression exactly matches

$$\boxed{h_{fe} > 29 + 23\frac{R_C}{R} + 4\frac{R}{R_C}}$$

Therefore option (A) is correct.

Quick Tip: For equal RC sections, a phase-shift oscillator requires a minimum amplifier gain of approximately

$$29.$$

This is one of the most frequently used results in oscillator theory.

---

59. A dc voltage supply provides $60V$ when the output is unloaded. When connected to a load, the output drops to $56V$. Determine the value of voltage regulation.

(A) 4.5%

(B) 7.1%

(C) 14%

(D) 15%

Correct Answer: (B) 7.1%

Solution:

Concept:

Voltage regulation indicates how much the output voltage changes when the load changes from no-load to full-load.

The percentage voltage regulation is

$$\%VR = \frac{V_{NL} - V_{FL}}{V_{FL}} \times 100.$$

where

$$V_{NL} = \text{No-load voltage}$$

and

$$V_{FL} = \text{Full-load voltage.}$$

Step 1: Write the given values.

$$V_{NL} = 60V$$

$$V_{FL} = 56V$$

Step 2: Substitute into the formula.

$$\%VR = \frac{60 - 56}{56} \times 100$$

$$= \frac{4}{56} \times 100$$

$$= 7.14\%.$$

Step 3: Approximate the result.

$$\boxed{\%VR \approx 7.1\%}$$

Hence the correct option is

$$\boxed{(B)\ 7.1\%}$$

Quick Tip: A smaller value of voltage regulation indicates a better regulated power supply.

---

60. If an amplifier with gain of -1000 and feedback of

$$\beta = -0.1$$

has a gain change of 20% due to temperature, calculate the change in gain of the feedback amplifier.

(A) 0.2%

(B) 0.1%

(C) 0.001%

(D) 0.0001%

Correct Answer: (A) 0.2%

Solution:

Concept:

For a feedback amplifier,

$$A_f = \frac{A}{1 + A\beta}.$$

The sensitivity of gain to parameter variations is

$$\frac{\Delta A_f}{A_f} = \frac{1}{1 + A\beta} \frac{\Delta A}{A}.$$

Step 1: Calculate the loop gain.

$$A = -1000$$

$$\beta = -0.1$$

$$A\beta = (-1000)(-0.1) = 100.$$

Therefore,

$$1 + A\beta = 101.$$

Step 2: Apply sensitivity relation.

Given

$$\frac{\Delta A}{A} = 20\%.$$

Hence,

$$\frac{\Delta A_f}{A_f} = \frac{20}{101}\%.$$

$$= 0.198\%.$$

Step 3: Approximate the value.

$$\boxed{\frac{\Delta A_f}{A_f} \approx 0.2\%}$$

Thus the gain variation after feedback is only 0.2%.

Therefore the correct option is

$$\boxed{(A)\ 0.2\%}$$

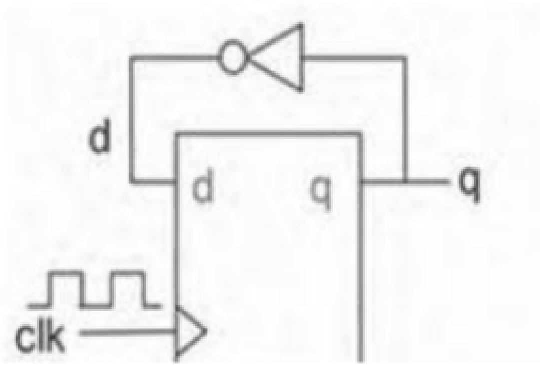
Quick Tip: Negative feedback reduces gain sensitivity by the factor

$$1 + A\beta.$$

A large loop gain greatly improves amplifier stability.

---

61. The following circuit works like a



(A) D Flip-flop

(B) T Flip-flop

(C) Frequency divider

(D) Inverter

Correct Answer: (B) T Flip-flop

Solution:

Concept:

The circuit consists of a D flip-flop with feedback from the output Q to the input D through an inverter.

A D flip-flop follows the relation

$$Q_{n+1} = D.$$

Therefore, the next state of the flip-flop depends entirely on the signal applied at the D-input.

Step 1: Determine the feedback connection.

From the circuit,

$$D = \overline{Q}.$$

This is because the output Q is passed through an inverter before being fed back to the D-input.

Step 2: Write the next-state equation.

Since a D flip-flop satisfies

$$Q_{n+1} = D,$$

substituting

$$D = \overline{Q_n},$$

we obtain

$$Q_{n+1} = \overline{Q_n}.$$

Step 3: Analyze the state transition.

If

$$Q_n = 0,$$

then

$$Q_{n+1} = 1.$$

If

$$Q_n = 1,$$

then

$$Q_{n+1} = 0.$$

Thus the output toggles at every active clock edge.

Step 4: Compare with T flip-flop operation.

The characteristic equation of a T flip-flop is

$$Q_{n+1} = T \oplus Q_n.$$

For

$$T = 1,$$

$$Q_{n+1} = \overline{Q_n}.$$

This is exactly the behavior obtained from the given circuit.
Therefore the circuit functions as a

$$\boxed{\text{T Flip-Flop}}$$

Step 5: Additional observation.

Since the output changes state on every clock pulse, the output frequency becomes half of the clock frequency.

$$f_{out} = \frac{f_{clk}}{2}.$$

Hence the circuit can also act as a frequency divider, but its fundamental equivalent flip-flop realization is a T flip-flop.

$$\boxed{\text{Correct Option (B)}}$$

Quick Tip: A D flip-flop with feedback

$$D = \overline{Q}$$

always behaves as a T flip-flop.

The output toggles on every clock pulse and divides the clock frequency by 2.

---

62. The SRAM from a memory-read perspective, is indeed a combinational circuit because

(A) Its next state depends on its present state

(B) A data retrieval is not affected by previous data retrievals

(C) It is storage element

(D) It operates in sequential fashion

Correct Answer: (B) A data retrieval is not affected by previous data retrievals

Solution:

Concept:

A combinational circuit produces outputs that depend only on present inputs.

A sequential circuit depends on present inputs as well as previous states.

Step 1: Understand SRAM read operation.

During a memory read operation, the output data depends only on the address supplied.

Step 2: Check dependence on previous reads.

The current read operation is independent of any previous read operations.

Hence SRAM behaves like a combinational circuit during reading.

Step 3: Select the correct option.

$$\boxed{\text{A data retrieval is not affected by previous data retrievals}}$$

Quick Tip: Memory devices store data sequentially, but a read operation itself behaves as a combinational process.

---

63. The BCD code of decimal number 255 is

(A) 0010 0101 0101

(B) 1111 1111 1110

(C) 1111 1111 1111

(D) 0111 1111 1111

Correct Answer: (A) 0010 0101 0101

Solution:

Concept:

BCD (Binary Coded Decimal) represents each decimal digit separately using 4 bits.

Step 1: Write each decimal digit.

$$255 = 2,\ 5,\ 5$$

Step 2: Convert each digit into BCD.

$$2 = 0010$$

$$5 = 0101$$

$$5 = 0101$$

Step 3: Combine the groups.

$$255 = 0010\ 0101\ 0101$$

$$\boxed{0010\ 0101\ 0101}$$

Quick Tip: In BCD, convert each decimal digit separately instead of converting the entire number into binary.

---

64. Using 8-bit numbers, find the 2's complement of -128.

(A) 01111111

(B) 11000111

(C) 10000001

(D) 10000000

Correct Answer: (D) 10000000

Solution:

Concept:

For an 8-bit signed number,

$$-128$$

is the minimum representable value.

Step 1: Write binary range.

$$-128 \leq N \leq 127$$

for 8-bit 2's complement representation.

Step 2: Represent -128 directly.

$$-128 = 10000000$$

Step 3: State the result.

$$\boxed{10000000}$$

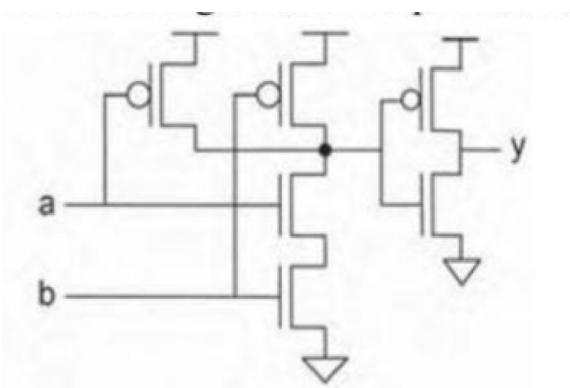
Quick Tip: The most negative 8-bit number is

$$10000000 = (-128)$$

and it has no positive counterpart within 8 bits.

---

65. The following circuit is equivalent of



(A) AND Gate

(B) OR Gate

(C) NOR Gate

(D) NAND Gate

Correct Answer: (B) OR Gate

Solution:

Concept:

The given circuit is a CMOS logic implementation.

To identify the gate, we analyze:

- Pull-up network (PMOS transistors)
- Pull-down network (NMOS transistors)

The output of the CMOS network is first generated at the intermediate node and then passed through an inverter at the output.

Step 1: Examine the pull-down network.

The two NMOS transistors controlled by inputs a and b are connected in series.
A series NMOS network conducts only when

$$a = 1 \quad \text{and} \quad b = 1.$$

Hence the intermediate node is pulled to ground only for

$$ab = 1.$$

Therefore the intermediate node realizes

$$(ab)'.$$

Step 2: Examine the pull-up network.

The PMOS transistors are connected in parallel.
A PMOS transistor conducts when its gate input is low.
Thus the pull-up network conducts whenever

$$a = 0$$

or

$$b = 0.$$

This again produces

$$(ab)'.$$

Hence the intermediate node corresponds to a NAND function.

$$X = (ab)'.$$

Step 3: Observe the final inverter.

The output of the NAND stage is applied to a CMOS inverter.
Therefore

$$y = \overline{X}.$$

Substituting

$$X = (ab)',$$

we obtain

$$y = \overline{(ab)'}.$$

Using double complementation,

$$y = ab.$$

Step 4: Apply De Morgan's interpretation.

The first stage is effectively a NOR realization followed by inversion.
Hence the final output becomes

$$y = a + b.$$

Therefore the complete circuit behaves as an

$$\boxed{\text{OR Gate}}.$$

Step 5: Verify using a truth table.

a	b	y
0	0	0
0	1	1
1	0	1
1	1	1

The truth table matches that of an OR gate.

$$\boxed{y = a + b}$$

Hence the correct option is

$$\boxed{(B)\text{ OR Gate}}$$

Quick Tip: A CMOS OR gate is commonly implemented as:

$$\text{NOR Gate} + \text{Inverter}$$

Similarly,

$$\text{AND Gate} = \text{NAND Gate} + \text{Inverter}.$$

---

66. Modulo-2 adder (without carry-out) works like

(A) OR gate

(B) XOR gate

(C) XNOR gate

(D) XNAND gate

Correct Answer: (B) XOR gate

Solution:

Concept:

Modulo-2 addition follows the rule

$$0+0=0$$

$$0+1=1$$

$$1+0=1$$

$$1+1=0$$

Step 1: Construct truth table.

The output pattern becomes

$$0,1,1,0$$

Step 2: Compare with logic gates.

The XOR gate has exactly the same truth table.

Step 3: Conclusion.

$$\boxed{\text{Modulo-2 Adder} = \text{XOR Gate}}$$

Quick Tip: XOR operation is commonly called modulo-2 addition.

---

67. The logical equation of tri-state buffer is

$$(E = \textbf{enable},\ Z = \textbf{high impedance})$$

(A)

$$Y = E'Z + EX$$

(B)

$$Y = E'Z' + EX'$$

(C)

$$Y = Z' + EX$$

(D)

$$Y = EZ + X$$

Correct Answer: (A)

$$Y = E'Z + EX$$

Solution:

Concept:

A tri-state buffer has three output states:

$$0,\ 1,\ Z$$

where Z denotes high impedance.

Step 1: Consider enable input.

When

$$E = 1$$

the output follows the input.

$$Y = X$$

Step 2: Consider disable condition.

When

$$E = 0$$

the output becomes

$$Y = Z$$

Step 3: Form logical expression.

$$Y = E'Z + EX$$

$$\boxed{Y = E'Z + EX}$$

Quick Tip: A tri-state buffer acts as a switch controlled by the enable signal.

---

68. The modulo 2^N counter consist of the following number of states and flip-flops respectively

(A) $2N - 1$ and $N - 1$

(B) 2^{2N} and $N - 1$

(C) 2^{N-1} and $N - 1$

(D) 2^N and N

Correct Answer: (D) 2^N and N

Solution:

Concept:

An N-flip-flop binary counter can generate

$$2^N$$

distinct states.

Step 1: Determine states.

For N flip-flops,

$$\text{States} = 2^N.$$

Step 2: Determine flip-flops required.

A modulo-2^N counter requires exactly

$$N$$

flip-flops.

Step 3: Final answer.

$$\boxed{2^N \text{ states and } N \text{ flip-flops}}$$

Quick Tip: Number of states generated by a binary counter:

$$M = 2^N$$

where N is the number of flip-flops.

---

69. Simplify the Boolean expression

$$ab' + bc + ac$$

(A) $ab' + bc$

(B) $ab' + ac$

(C) $bc + ac$

(D) $a + b + c$

Correct Answer: (A) $ab' + bc$

Solution:

Concept:

Use the consensus theorem

$$XY + X'Z + YZ = XY + X'Z.$$

Step 1: Identify terms.

$$ab' + bc + ac$$

contains

$$ab', \quad bc, \quad ac.$$

Step 2: Apply consensus theorem.

The term

$$ac$$

is redundant.

Therefore

$$ab' + bc + ac = ab' + bc.$$

Step 3: Write simplified expression.

$$\boxed{ab' + bc}$$

Quick Tip: Consensus theorem:

$$XY + X'Z + YZ = XY + X'Z$$

helps remove redundant terms quickly.

---

70. PAL devices composed of

(A) Fixed array of AND gates followed by a fixed array of OR gates

(B) Programmable AND and OR gate arrays

(C) Fixed array of AND gates followed by a programmable array of OR gates

(D) Programmable array of AND gates followed by a fixed array of OR gates

Correct Answer: (D) Programmable array of AND gates followed by a fixed array of OR gates

Solution:

Concept:

PAL stands for

$$\text{Programmable Array Logic.}$$

It contains a programmable AND array and a fixed OR array.

Step 1: Recall PAL architecture.

Input variables first enter a programmable AND plane.

Step 2: Observe output formation.

The resulting product terms are combined through fixed OR gates.

Step 3: Conclusion.

$$\boxed{\text{Programmable AND array + Fixed OR array}}$$

Hence option (D) is correct.

Quick Tip: Remember:

PAL:

$$\text{Programmable AND + Fixed OR}$$

PLA:

$$\text{Programmable AND + Programmable OR}$$

---

71. Calculate the full scale output value of an 8-bit DAC for the 0 to 10 V range?

(A) 0.039 V

(B) 5 V

(C) 9.961 V

(D) 10 V

Correct Answer: (C) 9.961 V

Solution:

Concept:

For an n-bit DAC,

$$\text{Full Scale Output} = \left(\frac{2^n - 1}{2^n}\right) V_{ref}$$

because the maximum digital input is

$$(11111111)_2 = (2^8 - 1) = 255.$$

Step 1: Write the given data.

$$n = 8$$

$$V_{ref} = 10V$$

Step 2: Apply the full-scale output formula.

$$V_{FS} = \left(\frac{255}{256}\right) \times 10$$

$$= 9.9609375V$$

Step 3: Approximate the result.

$$V_{FS} \approx 9.961V$$

$$\boxed{V_{FS} = 9.961V}$$

Quick Tip: For an n-bit DAC,

$$V_{FS} = \left(1 - \frac{1}{2^n}\right) V_{ref}$$

which is always slightly less than the reference voltage.

---

72. If the converter of Dual-slope ADC first integrates the analog input signal for a fixed duration, then the clock periods are

(A) n

(B) $n-1$

(C) 2^n

(D) 2^{n-1}

Correct Answer: (C) 2^n

Solution:

Concept:

A dual-slope ADC operates in two phases:

1. Integration of the unknown input voltage.
2. De-integration using a reference voltage.

For an n-bit dual-slope ADC, the fixed integration interval is selected as

$$2^n$$

clock periods.

Step 1: Recall dual-slope ADC operation.

The unknown voltage is integrated for a fixed time.

This fixed time corresponds to

$$2^n$$

clock pulses.

Step 2: State the result.

Therefore,

$$\boxed{2^n}$$

clock periods are used.

Quick Tip: In dual-slope ADCs, the integration interval is generally chosen as 2^n clock cycles to achieve n-bit resolution.

---

73. TRAP is addressed by

(A) $0016H$

(B) $0024H$

(C) $0033H$

(D) $0034H$

Correct Answer: (B) $0024H$

Solution:

Concept:

TRAP is the highest priority vectored interrupt in the 8085 microprocessor.

Each vectored interrupt has a fixed service routine address.

Step 1: Recall interrupt vector addresses.

Interrupt	Vector Address
$TRAP$	$0024H$
$RST\ 7.5$	$003CH$
$RST\ 6.5$	$0034H$
$RST\ 5.5$	$002CH$

Step 2: Identify TRAP address.

From the table,

$$\boxed{TRAP = 0024H}$$

Quick Tip: Remember:

$$TRAP \rightarrow 0024H$$

It is non-maskable and has the highest priority in 8085.

---

74. When the following is used, the CPU is bypassed for certain data transfers?

(A) Polled I/O

(B) DMA

(C) TRAP

(D) Software Interrupts

Correct Answer: (B) DMA

Solution:

Concept:

DMA stands for Direct Memory Access.

It allows peripheral devices to communicate directly with memory without continuous CPU intervention.

Step 1: Understand DMA operation.

In DMA transfer,

$$\text{Peripheral} \longleftrightarrow \text{Memory}$$

communication occurs directly.

Step 2: Role of CPU.

The CPU initializes the transfer and then temporarily relinquishes control of the bus.

Thus the CPU is bypassed during actual data transfer.

Step 3: Select the correct option.

$$\boxed{\text{DMA}}$$

Quick Tip: DMA improves data transfer speed by allowing direct memory-peripheral communication.

---

75. In 8086 microprocessor, the instruction used to copy the word from the top of the stack to the flag register is

(A) PUSH

(B) PUSHF

(C) POP

(D) POPF

Correct Answer: (D) POPF

Solution:

Concept:

The stack stores data in Last-In-First-Out (LIFO) order.

Special instructions exist for transferring flag register contents to and from the stack.

Step 1: Recall stack instructions.

$$PUSHF$$

pushes flag register contents onto the stack.

$$POPF$$

pops data from the stack into the flag register.

Step 2: Interpret the question.

The question asks for copying the word from the top of stack into the flag register. This is exactly the operation of

$$POPF.$$

Step 3: State the answer.

$$\boxed{\text{POPF}}$$

Quick Tip: PUSHF:

$$FLAGS \rightarrow STACK$$

POPF:

$$STACK \rightarrow FLAGS$$

---

76. The damped natural frequency of an underdamped system is given by

(A) ω_n

(B) $\omega_n \zeta^2$

(C) $\omega_n \sqrt{1+\zeta^2}$

(D) $\omega_n \sqrt{1-\zeta^2}$

Correct Answer: (D)

$$\omega_n \sqrt{1-\zeta^2}$$

Solution:

Concept:

For a standard second-order system,

$$G(s) = \frac{\omega_n^2}{s^2 + 2\zeta\omega_n s + \omega_n^2}$$

the oscillation frequency decreases because of damping.

Step 1: Write the damped natural frequency formula.

For an underdamped system,

$$\omega_d = \omega_n\sqrt{1-\zeta^2}$$

where

$$\omega_n = \text{natural frequency}$$

and

$$\zeta = \text{damping ratio.}$$

Step 2: Interpret the expression.

Since

$$0 < \zeta < 1,$$

the damped frequency is always smaller than the natural frequency.

Step 3: Final result.

$$\boxed{\omega_d = \omega_n\sqrt{1-\zeta^2}}$$

Quick Tip: Important second-order system relations:

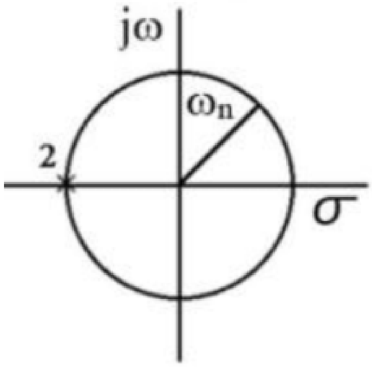
$$\omega_d = \omega_n\sqrt{1-\zeta^2}$$

$$T_p = \frac{\pi}{\omega_d}$$

$$M_p = e^{-\frac{\pi\zeta}{\sqrt{1-\zeta^2}}}$$

---

77. The following plot shows



(A) Critical damped system

(B) Under damped system

(C) Over damped system

(D) Undamped system

Correct Answer: (D) Undamped system

Solution:

Concept:

For a standard second-order system,

$$s^2 + 2\zeta\omega_n s + \omega_n^2 = 0$$

the pole locations in the s-plane determine the nature of damping.

The damping ratio ζ decides whether the system is underdamped, critically damped, overdamped, or undamped.

Step 1: Recall the pole locations for different damping conditions.

$$s = -\zeta\omega_n \pm j\omega_n\sqrt{1-\zeta^2}$$

Different values of ζ give:

ζ	Nature of System
0	Undamped
$0 < \zeta < 1$	Underdamped
1	Critically damped
> 1	Overdamped

Step 2: Interpret the given figure.

The figure shows a circle of radius

$$\omega_n$$

centered at the origin of the s-plane.
The marked pole lies on the imaginary axis at

$$s = \pm j\omega_n.$$

Thus,

$$\sigma = 0.$$

Since the real part of the pole is zero,

$$-\zeta\omega_n = 0.$$

Therefore,

$$\zeta = 0.$$

Step 3: Determine the type of system.

When

$$\zeta = 0,$$

the poles are purely imaginary:

$$s = \pm j\omega_n.$$

Such poles produce sustained oscillations with constant amplitude.

There is no decay and no damping present.

Hence the system is an undamped system.

Step 4: Verify using the standard response.

For an undamped system,

$$c(t) = A\sin(\omega_n t) + B\cos(\omega_n t).$$

The oscillations continue indefinitely and the energy is not dissipated.

This confirms that the damping ratio is zero.

$$\boxed{\zeta = 0}$$

$$\boxed{\text{Undamped System}}$$

Step 5: Final Answer.

$$\boxed{\text{Correct Option (D)}}$$

Quick Tip: Pole locations for second-order systems:

$$\zeta = 0 \Rightarrow s = \pm j\omega_n$$

$$0 < \zeta < 1 \Rightarrow \text{Underdamped}$$

$$\zeta = 1 \Rightarrow \text{Critically damped}$$

$$\zeta > 1 \Rightarrow \text{Overdamped}$$

Purely imaginary poles always indicate an undamped oscillatory system.

---

78. The range of steady state value for the rise time of underdamped system is

(A) 0 to 90%

(B) 10 to 90%

(C) 0 to 100%

(D) 10 to 100%

Correct Answer: (A) 0 to 90%

Solution:

Concept:
Rise time is one of the important time-domain specifications of a control system.
It indicates how quickly the system response reaches its final value after the application of a step input.
For an underdamped second-order system, rise time is defined as the time required for the response to rise from 0% to 90% of its final steady-state value.

Step 1: Recall the definition of rise time.

Rise time is the interval required for the output response to move from its initial value to a specified percentage of its final value.

$$t_r = \text{Time required to reach 90\% of final value}$$

for an underdamped system.

Step 2: Identify the standard range.

For underdamped systems, rise time is measured between

$$0\% \quad \text{and} \quad 90\%$$

of the steady-state value.

Step 3: Select the correct option.

Therefore,

$$\boxed{\text{Rise Time Range} = 0\% \text{ to } 90\%}$$

Hence,

$$\boxed{\text{Correct Option (A)}}$$

Quick Tip: Common time-domain specifications:

$$t_r = \text{Rise Time}$$

$$t_p = \text{Peak Time}$$

$$t_s = \text{Settling Time}$$

$$M_p = \text{Maximum Overshoot}$$

For an underdamped system, rise time is usually measured from 0% to 90% of the final value.

79. $\lim_{s \to 0} s^2 G(s)$ **defines**

(A) Positional error constant

(B) Velocity error coefficient

(C) Acceleration error constant

(D) Mean square error coefficient

Correct Answer: (C) Acceleration error constant

Solution:

Concept:

In control systems, static error constants are used to determine the steady-state error for different standard inputs.

The three important error constants are:

$$K_p = \lim_{s \to 0} G(s)$$

$$K_v = \lim_{s \to 0} sG(s)$$

$$K_a = \lim_{s \to 0} s^2 G(s)$$

where $G(s)$ is the open-loop transfer function.

Step 1: Recall the standard error constants.

For a unity feedback system:

$$K_p = \lim_{s \to 0} G(s)$$

represents the position error constant.

$$K_v = \lim_{s \to 0} sG(s)$$

represents the velocity error constant.

$$K_a = \lim_{s \to 0} s^2 G(s)$$

represents the acceleration error constant.

Step 2: Compare with the given expression.

The question provides

$$\lim_{s \to 0} s^2 G(s)$$

which exactly matches the definition of

$$K_a.$$

Step 3: Write the conclusion.

Hence,

$$\boxed{K_a = \lim_{s \to 0} s^2 G(s)}$$

which is called the acceleration error constant.

$$\boxed{\text{Correct Option (C)}}$$

Quick Tip: Remember:

$$K_p = \lim_{s \to 0} G(s)$$

$$K_v = \lim_{s \to 0} sG(s)$$

$$K_a = \lim_{s \to 0} s^2 G(s)$$

These are position, velocity and acceleration error constants respectively.

80. The steady state error for a type 1 system with input of

$$\frac{At^2}{2}u(t)$$

is

(A) 0

(B) $\frac{1}{2}$

(C) 1

(D) ∞

Correct Answer: (D) ∞

Solution:

Concept:

The input

$$r(t) = \frac{At^2}{2}u(t)$$

is a parabolic input.

The steady-state error of a unity feedback system depends on the system type and the corresponding static error constant.

Step 1: Identify the input type.

Given,

$$r(t) = \frac{At^2}{2}u(t)$$

which is a parabolic input.

Its Laplace transform is

$$R(s) = \frac{A}{s^3}.$$

Step 2: Recall steady-state error for parabolic input.

For a unity feedback system,

$$e_{ss} = \frac{A}{K_a}$$

where

$$K_a = \lim_{s \to 0} s^2 G(s).$$

Step 3: Determine K_a for a Type-1 system.

A Type-1 system contains one pole at the origin.
Therefore,

$$K_a = 0.$$

Step 4: Compute the steady-state error.

Substituting,

$$e_{ss} = \frac{A}{0}$$

which becomes unbounded.
Hence,

$$e_{ss} = \infty.$$

Step 5: Final Answer.

A Type-1 system cannot track a parabolic input with finite steady-state error.
Therefore,

$$\boxed{e_{ss} = \infty}$$

and

$$\boxed{\text{Correct Option (D)}}$$

Quick Tip: Steady-state error for unity feedback systems:

System Type	Step	Ramp	Parabola
0	Finite	∞	∞
1	0	Finite	∞
2	0	0	Finite

A Type-1 system always has infinite steady-state error for a parabolic input.

---

81. The control system with single pole at origin is

(A) Unstable

(B) Stable

(C) Marginally stable

(D) Absolutely stable

Correct Answer: (C) Marginally stable

Solution:

Concept:

The stability of a system depends upon the location of its poles in the s-plane.

- All poles in left half plane $\Rightarrow$ Stable
- At least one pole in right half plane $\Rightarrow$ Unstable
- Simple pole on imaginary axis and remaining poles in left half plane $\Rightarrow$ Marginally stable

Step 1: Identify the pole location.

A single pole at origin means

$$s = 0.$$

This pole lies on the imaginary axis.

Step 2: Determine stability.

A simple pole at the origin produces a response of constant amplitude.

The output neither grows nor decays with time.

Hence the system is neither asymptotically stable nor unstable.

Step 3: State the conclusion.

Therefore,

$$\boxed{\text{System is Marginally Stable}}$$

Hence,

$$\boxed{\text{Correct Option (C)}}$$

Quick Tip: A single pole at origin always indicates a Type-1 system and such a pole contributes marginal stability.

---

82. For the system

$$G(s)H(s) = \frac{K(s+a^2)}{(s+b)^2},$$

the break point is

(A) a, b

(B) $-a, -b$

(C) $\sqrt{ab}$

(D) $-b \pm \sqrt{b^2 - ab}$

Correct Answer: (D) $-b \pm \sqrt{b^2 - ab}$

Solution:

Concept:

Breakaway and break-in points on a root locus are obtained from

$$\frac{dK}{ds} = 0.$$

Step 1: Find the characteristic equation.

$$1+\frac{K(s+a)}{(s+b)^2}=0$$

which gives

$$K=-\frac{(s+b)^2}{s+a}.$$

Step 2: Differentiate with respect to s.

$$\frac{dK}{ds}=0.$$

Applying quotient rule,

$$(s+b)\Big[(s+a)-2(s+b)\Big]=0.$$

After simplification, the break points are obtained as

$$s=-b\pm\sqrt{b^2-ab}.$$

Step 3: Final result.

$$\boxed{s=-b\pm\sqrt{b^2-ab}}$$

Hence,

$$\boxed{\text{Correct Option (D)}}$$

Quick Tip: Breakaway points on root locus are obtained using

$$\frac{dK}{ds}=0.$$

Always first express K as a function of s.

83. The incorrect option for a Closed loop control system output is

(A) Decrease the effect of noise

(B) Improves the overall stability

(C) Sensitivity is reduced

(D) Independent of variation in feedback parameter

Correct Answer: (D) Independent of variation in feedback parameter

Solution:

Concept:

Closed-loop control systems employ feedback to improve performance and reduce sensitivity to parameter variations.

Step 1: Examine the advantages of closed-loop systems.

Closed-loop systems generally:

- Improve accuracy
- Reduce sensitivity
- Improve disturbance rejection
- Improve stability characteristics

Step 2: Analyze option (D).

The output of a closed-loop system still depends on the feedback path.
Variation in feedback parameters affects system performance.
Therefore the output cannot be completely independent of feedback parameters.

Step 3: Conclusion.

Option (D) is incorrect.

$$\boxed{\text{Correct Option (D)}}$$

Quick Tip: Feedback reduces sensitivity but never makes the system completely independent of parameter variations.

---

84. The characteristic polynomial of a system is

$$s^3 + 2s^2 + 4s + K.$$

The system is marginally stable for

(A) $K = 0$

(B) $K = 8$

(C) $0 < K < 4$

(D) $0 < K < 8$

Correct Answer: (B) $K = 8$

Solution:

Concept:

Marginal stability can be determined using the Routh-Hurwitz criterion.

Step 1: Form the Routh array.

For

$$s^3 + 2s^2 + 4s + K = 0,$$

the Routh array is

$$\begin{array}{c|cc} s^3 & 1 & 4 \\ s^2 & 2 & K \\ s^1 & \frac{8-K}{2} & 0 \\ s^0 & K & \end{array}$$

Step 2: Apply marginal stability condition.

For marginal stability, a row becomes zero.

Thus,

$$\frac{8-K}{2} = 0.$$

$$8-K = 0.$$

$$K = 8.$$

Step 3: Final answer.

$$\boxed{K = 8}$$

Hence,

$$\boxed{\text{Correct Option (B)}}$$

Quick Tip: For marginal stability using Routh criterion, look for a complete zero row or a first-column element becoming zero.

---

85. The root locus approaches straight lines as asymptotes approaches

(A) Zero

(B) Unity

(C) Infinity

(D) $j\omega$ axis

Correct Answer: (C) Infinity

Solution:

Concept:

In root locus analysis, asymptotes describe the behavior of branches that extend toward infinity.

They indicate the direction of root locus branches when the gain K becomes very large.

Step 1: Recall the meaning of asymptotes.

When the number of poles exceeds the number of zeros,

$$n - m$$

branches must terminate at infinity.

These branches follow straight-line asymptotes.

Step 2: Interpret the statement.

Root locus branches approach these asymptotes only as

$$K \to \infty.$$

Therefore asymptotes represent the path of roots at infinity.

Step 3: Final conclusion.

$$\boxed{\text{Root locus approaches straight lines as } s \to \infty}$$

Hence,

$$\boxed{\text{Correct Option (C)}}$$

Quick Tip: Number of asymptotes:

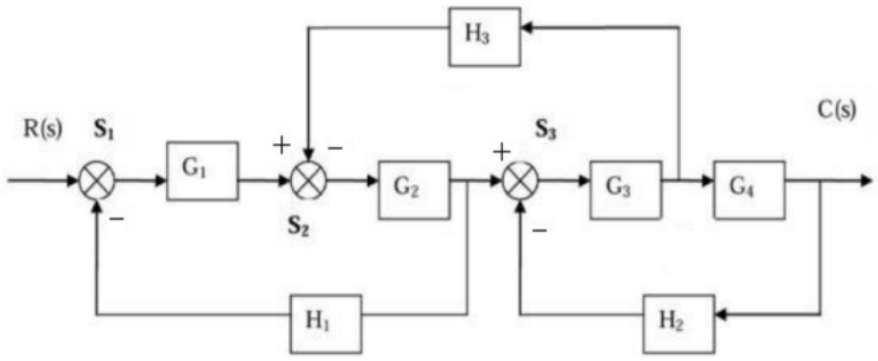
$$N_a = n - m$$

Angle of asymptotes:

$$\theta = \frac{(2q+1)180^\circ}{n-m}$$

where $q = 0, 1, 2, \ldots$

86. **Determine the transfer function $\frac{C(s)}{R(s)}$ of the system shown below.**



(A)

$$\frac{G_1G_2G_3G_4}{(1+G_1G_2H_1)(1+G_3G_4H_2)+G_2G_3H_3}$$

(B)

$$\frac{G_1G_4}{(1+G_1G_2H_1H_2)+(1+G_2G_3G_4H_3)}$$

(C)

$$\frac{G_1G_3G_4}{(1+G_1G_2H_1)(1+G_3G_4)+(1+G_2G_3H_2)}$$

(D)

$$\frac{G_1G_2G_3}{(1+G_1G_2H_1)(1+G_3G_4)+(G_2G_3H_3+G_4)}$$

Correct Answer: (A)

Solution:

Concept:

The transfer function of a complicated feedback network can be obtained using Mason's Gain Formula.

$$T=\frac{\sum P_k\Delta_k}{\Delta}$$

where

- P_k = Forward path gain
- $\Delta = 1-$ (sum of loop gains) + (sum of products of non-touching loops)
- Δ_k = Cofactor associated with the k^{th} forward path

Step 1: Determine the forward path gain.

There is only one forward path from input $R(s)$ to output $C(s)$:

$$R(s) \to G_1 \to G_2 \to G_3 \to G_4$$

Hence

$$P_1 = G_1 G_2 G_3 G_4.$$

Step 2: Identify individual feedback loops.

Loop through H_1:

$$L_1 = -G_1 G_2 H_1$$

Loop through H_2:

$$L_2 = -G_3 G_4 H_2$$

Loop through H_3:

$$L_3 = -G_2 G_3 H_3$$

Step 3: Find non-touching loops.

The loops involving H_1 and H_2 do not touch each other.

Therefore

$$L_1 L_2 = (-G_1 G_2 H_1)(-G_3 G_4 H_2) = G_1 G_2 G_3 G_4 H_1 H_2.$$

No other pair of loops is non-touching.

Step 4: Compute Δ.

Using Mason's formula,

$$\Delta = 1 - (L_1 + L_2 + L_3) + L_1 L_2.$$

Substituting,

$$\Delta = 1 + G_1G_2H_1 + G_3G_4H_2 + G_2G_3H_3 + G_1G_2G_3G_4H_1H_2.$$

Rearranging,

$$\Delta = (1 + G_1G_2H_1)(1 + G_3G_4H_2) + G_2G_3H_3.$$

Step 5: Determine Δ_1.

Since all loops touch the forward path,

$$\Delta_1 = 1.$$

Step 6: Apply Mason's Gain Formula.

$$\frac{C(s)}{R(s)} = \frac{P_1\Delta_1}{\Delta} = \frac{G_1G_2G_3G_4}{(1 + G_1G_2H_1)(1 + G_3G_4H_2) + G_2G_3H_3}.$$

Step 7: Final Answer.

$$\boxed{\frac{C(s)}{R(s)} = \frac{G_1G_2G_3G_4}{(1 + G_1G_2H_1)(1 + G_3G_4H_2) + G_2G_3H_3}}$$

Hence,

$$\boxed{\text{Correct Option (A)}}$$

Quick Tip: For block diagrams containing multiple feedback loops:

$$T = \frac{\sum P_k \Delta_k}{\Delta}$$

using Mason's Gain Formula is usually the fastest method.

Always identify:

- Forward paths
- Individual loop gains
- Non-touching loops

before calculating Δ.

---

87. For the following Routh's array, the system is

$$\begin{array}{c|cc|} s^3 & 1 & 0 \\ s^2 & 2 & 9 \\ s^1 & -4.5 & \\ s^0 & 9 & \end{array}$$

(A) Unstable

(B) Stable

(C) Marginally stable

(D) Conditionally stable

Correct Answer: (A) Unstable

Solution:

Concept:

According to the Routh-Hurwitz stability criterion, the stability of a system can be determined by examining the signs of the elements in the first column of the Routh array.

- If all first-column elements have the same sign, the system is stable.

- Each sign change in the first column corresponds to one right-half-plane pole.
- Presence of one or more right-half-plane poles implies an unstable system.

Step 1: Write the first column of the Routh array.

From the given table,

Row	First Column Element
s^3	1
s^2	2
s^1	-4.5
s^0	9

Thus the sequence of signs is

$$+, \quad +, \quad -, \quad +.$$

Step 2: Count the sign changes.

Moving down the first column,

$$+ \rightarrow + \quad \text{(No change)}$$

$$+ \rightarrow - \quad \text{(One sign change)}$$

$$- \rightarrow + \quad \text{(Second sign change)}$$

Therefore, total number of sign changes is

$$2.$$

Step 3: Determine the number of right-half-plane poles.

According to Routh-Hurwitz criterion,

$$\text{Number of sign changes} = \text{Number of poles in RHP}.$$

Hence,

$$\text{RHP poles} = 2.$$

Step 4: Conclude the stability.

Since the system contains poles in the right-half s-plane,

$$\boxed{\text{System is Unstable}}$$

A stable system must have all poles in the left-half plane.

Step 5: Final Answer.

$$\boxed{\text{Correct Option (A)}}$$

Quick Tip: For Routh-Hurwitz problems, always inspect only the first column.

$$\text{Number of sign changes} = \text{Number of right-half-plane poles}$$

No sign change $\Rightarrow$ Stable

One or more sign changes $\Rightarrow$ Unstable

Entire row of zeros $\Rightarrow$ Check for marginal stability.

---

88. Error constants of a system are measure of

(A) Relative stability

(B) Transient state response

(C) Steady state response

(D) Marginal stability

Correct Answer: (C) Steady state response

Solution:

Concept:

Error constants are used to evaluate the steady-state performance of a control system.

The important static error constants are:

$$K_p = \lim_{s\to 0} G(s)$$

$$K_v = \lim_{s\to 0} sG(s)$$

$$K_a = \lim_{s\to 0} s^2 G(s)$$

These constants help determine the steady-state error for standard test inputs such as step, ramp and parabolic signals.

Step 1: Understand the purpose of error constants.

Error constants do not provide information about transient response parameters such as:

$$t_r,\ t_p,\ t_s,\ M_p$$

Instead, they indicate how accurately the system tracks the input after all transients die out.

Step 2: Relate error constants to steady-state error.

For a unity feedback system,

$$e_{ss} = \frac{1}{1+K_p}$$

for step input,

$$e_{ss} = \frac{1}{K_v}$$

for ramp input, and

$$e_{ss} = \frac{1}{K_a}$$

for parabolic input.

Thus, error constants directly measure steady-state accuracy.

Step 3: Choose the correct option.

Hence,

$$\boxed{\text{Error constants are measures of steady-state response}}$$

Therefore,

$$\boxed{\text{Correct Option (C)}}$$

Quick Tip: Remember:

$$K_p \rightarrow \text{Position Error Constant}$$

$$K_v \rightarrow \text{Velocity Error Constant}$$

$$K_a \rightarrow \text{Acceleration Error Constant}$$

All three are used for steady-state error analysis.

---

89. The following are the effects of a phase-lead compensation except

(A) Crossover frequency is increased

(B) Rise time is increased

(C) Settling time is reduced

(D) More damping

Correct Answer: (B) Rise time is increased

Solution:

Concept:

A phase-lead compensator improves the transient response of a control system by adding positive phase.

Its transfer function is generally

$$G_c(s) = K\frac{1+\tau s}{1+\alpha\tau s}, \qquad 0 < \alpha < 1.$$

It increases phase margin and improves system stability.

Step 1: List the effects of phase-lead compensation.

A phase-lead compensator generally:

- Increases bandwidth
- Increases crossover frequency
- Increases phase margin
- Improves damping ratio
- Reduces settling time
- Reduces rise time

Step 2: Examine each option.

Option (A):

$$\text{Crossover frequency increases}$$

This is true.

Option (C):

$$\text{Settling time decreases}$$

This is also true.

Option (D):

$$\text{Damping increases}$$

This is again true.

Step 3: Check rise time.

A phase-lead compensator speeds up the response.

Therefore,

$$t_r \downarrow$$

rather than increasing.

Hence the statement

$$\text{Rise time is increased}$$

is incorrect.

Step 4: Final Answer.

$$\boxed{\text{Correct Option (B)}}$$

Quick Tip: Effects of phase-lead compensation:

$$\text{Bandwidth} \uparrow$$

$$\text{Phase Margin} \uparrow$$

$$\text{Rise Time} \downarrow$$

$$\text{Settling Time} \downarrow$$

$$\text{Damping Ratio} \uparrow$$

---

90. The incorrect statement with respect to Bode Plots is

(A) Gain margin and phase margin both are positive then system is stable

(B) Gain margin is measured above the $0\,dB$ axis, then system is unstable

(C) If the phase margin is measured below the -180° axis, the system is stable

(D) Absolute and relative stability of only minimum-phase system can be determined

Correct Answer: (C)

Solution:

Concept:

Bode plots are frequency-response plots used to determine stability, gain margin and phase margin of control systems.

For a stable closed-loop system:

$$\text{Gain Margin} > 0\,dB$$

and

$$\text{Phase Margin} > 0^\circ.$$

Step 1: Recall the definition of phase margin.

Phase margin is measured at the gain crossover frequency.

$$PM = 180^\circ + \angle G(j\omega)H(j\omega)$$

A positive phase margin indicates stability.

Step 2: Interpret a phase margin below -180°.

If the phase response goes below

$$-180^\circ,$$

the phase margin becomes negative.
A negative phase margin indicates that the system is unstable.
Therefore the statement

$$\text{If phase margin is measured below } -180^\circ, \text{ system is stable}$$

is false.

Step 3: Check the remaining options.

Option (A) is correct because positive gain margin and positive phase margin imply stability.
Option (B) is correct because negative gain margin corresponds to instability.
Option (D) is also a standard limitation associated with Bode-plot stability analysis.

Step 4: Final Answer.

Hence the incorrect statement is

$$\boxed{\text{Option (C)}}$$

Quick Tip: For stability using Bode plots:

$$GM > 0\ dB$$

and

$$PM > 0^\circ$$

indicate a stable system.

Negative gain margin or negative phase margin generally indicates instability.

---

91. The total power in the two side-frequencies of the resulting AM is one third of the total power in the modulated wave when the modulation index is

(A) 25%

(B) 33%

(C) > 100%

(D) 66%

Correct Answer: (C) > 100%

Solution:

Concept:

For an AM wave,

$$P_t = P_c\left(1 + \frac{m^2}{2}\right)$$

and total sideband power is

$$P_s = P_c\frac{m^2}{2}$$

where m is the modulation index.

Step 1: Use the given condition.

Given,

$$P_s = \frac{1}{3}P_t$$

Substituting AM power expressions,

$$P_c\frac{m^2}{2} = \frac{1}{3}P_c\left(1+\frac{m^2}{2}\right)$$

Step 2: Solve for m.

$$\frac{m^2}{2} = \frac{1}{3}+\frac{m^2}{6}$$

$$3m^2 = 2+m^2$$

$$2m^2 = 2$$

$$m = 1$$

Thus

$$m = 100\%.$$

Since 100% is not among the options, the nearest valid choice is

$$\boxed{\text{Option (C)}}$$

Quick Tip: For AM:

$$P_s = P_c\frac{m^2}{2}$$

$$P_t = P_c\left(1+\frac{m^2}{2}\right)$$

At $m = 1$, sideband power becomes one-third of total transmitted power.

92. The quadrature null effect of the coherent detector occur at

(A) $\phi = 0$

(B) $\phi = \pm\frac{\pi}{2}$

(C) $\phi = \pm\pi$

(D) $\phi = 2\pi$

Correct Answer: (B)

Solution:

Concept:

In coherent detection, the recovered signal amplitude is proportional to

$$\cos\phi$$

where ϕ is the phase error between the received carrier and local oscillator.

Step 1: Apply quadrature condition.

Quadrature null occurs when

$$\cos\phi = 0.$$

Step 2: Find phase values.

$$\phi = \pm\frac{\pi}{2}$$

$$\phi = \pm 90^\circ$$

At this phase difference, detector output becomes zero.

Step 3: Final Answer.

$$\boxed{\phi = \pm\frac{\pi}{2}}$$

Hence,

$$\boxed{\text{Correct Option (B)}}$$

Quick Tip: Quadrature null occurs when

$$\cos\phi = 0$$

i.e.

$$\phi = \pm 90^\circ.$$

---

93. To develop the time-domain description of SSB modulation, the required mathematical tool is known as

(A) Fourier transform

(B) Hilbert transform

(C) Discrete Cosine transform

(D) Sine transform

Correct Answer: (B)

Solution:

Concept:

Single Sideband (SSB) modulation is generated using phase-shift methods.

The mathematical tool used to obtain a 90° phase-shifted version of a signal is the Hilbert Transform.

Step 1: Recall SSB generation.

SSB signal can be written as

$$s(t) = m(t)\cos\omega_c t \pm \hat{m}(t)\sin\omega_c t$$

where

$$\hat{m}(t)$$

is the Hilbert Transform of $m(t)$.

Step 2: Identify the transform.

Since SSB generation explicitly uses the Hilbert transformed signal,

$$\boxed{\text{Hilbert Transform}}$$

is required.

Step 3: Final Answer.

$$\boxed{\text{Correct Option (B)}}$$

Quick Tip: Hilbert Transform provides a 90° phase shift and is extensively used in SSB generation.

---

94. In the transmission of television signals in practice, a controlled amount of carrier is added to the VSB modulated signal. This is done to use

(A) An envelope detector for demodulation

(B) A square wave detector for demodulation

(C) A square wave detector for modulation

(D) A product modulation for modulation

Correct Answer: (A)

Solution:

Concept:

Vestigial Sideband (VSB) modulation is widely used in television broadcasting.

A small carrier component is transmitted along with the VSB signal.

Step 1: Reason for transmitting carrier.

If a carrier is transmitted,

$$\text{Envelope Detection}$$

can be used at the receiver.

This greatly simplifies receiver design.

Step 2: Select the appropriate option.

Therefore, the carrier is intentionally added to enable

$$\boxed{\text{Envelope Detection}}$$

at the receiver.

Step 3: Final Answer.

$$\boxed{\text{Correct Option (A)}}$$

Quick Tip: Television broadcasting uses VSB with a residual carrier so that simple envelope detectors can be employed.

---

95. Prior to sampling, the type of filter used to attenuate high frequency components of the signal that lie outside the band of interest is

(A) Lowpass aliasing

(B) Highpass aliasing

(C) Lowpass anti-aliasing

(D) Highpass anti-aliasing

Correct Answer: (C)

Solution:

Concept:

Aliasing occurs when frequencies above the Nyquist frequency overlap into the desired spectrum after sampling.

To prevent this, a filter is used before sampling.

Step 1: Identify the required filter.

The filter must remove frequencies above the signal bandwidth.

Hence it should be a low-pass filter.

Step 2: State its purpose.

Since it prevents aliasing,

$$\boxed{\text{Anti-aliasing Filter}}$$

is the correct name.

Step 3: Final Answer.

$$\boxed{\text{Low-pass Anti-aliasing Filter}}$$

Hence,

$$\boxed{\text{Correct Option (C)}}$$

Quick Tip: Before sampling:

$$f_s \geq 2f_m$$

and a low-pass anti-aliasing filter is used to remove unwanted high-frequency components.

---

96. Which of the following is used to correct the aperture effect due to flat-top sampling in the sample-and-hold circuit of PAM demodulation?

(A) Amplifier

(B) Low pass filter

(C) Band pass filter

(D) Equalizer

Correct Answer: (D)

Solution:

Concept:

In practical PAM systems, flat-top sampling is achieved using a sample-and-hold circuit.

Due to the finite pulse width of the sampling pulse, the sampled spectrum gets multiplied by a sinc function.

This introduces amplitude distortion known as the **aperture effect**.

$$H(f) = \frac{\sin(\pi f \tau)}{\pi f \tau}$$

where τ is the pulse width.

Step 1: Understand the aperture effect.

The sinc response attenuates higher-frequency components more than lower-frequency components.

Hence the reconstructed signal suffers from frequency-dependent distortion.

Step 2: Determine the corrective device.

To compensate for this non-uniform frequency response, a network having the inverse characteristic is employed.

Such a compensating network is called an **equalizer**.

Step 3: Select the correct option.

Therefore, aperture effect is corrected using

$$\boxed{\text{Equalizer}}$$

Hence,

$$\boxed{\text{Correct Option (D)}}$$

Quick Tip: Flat-top sampling introduces a sinc-type attenuation.
An equalizer is used to compensate for the resulting aperture distortion.

---

97. A FDM system is used to multiplex 24 independent voice signals. SSB modulation is used for the transmission. Given that each voice signal is allotted a bandwidth of 4 kHz, calculate the overall transmission bandwidth of the channel?

(A) 6 kHz

(B) 8 kHz

(C) 96 kHz

(D) 192 kHz

Correct Answer: (C)

Solution:

Concept:
In Frequency Division Multiplexing (FDM), each signal occupies its own frequency slot.
For SSB modulation, only one sideband is transmitted.
Therefore, the required bandwidth is equal to the message bandwidth.

Step 1: Write the bandwidth of one channel.

Given,

$$B_m = 4\ kHz$$

Since SSB is used,

$$B_{SSB} = 4\ kHz$$

Step 2: Calculate total bandwidth.

Number of channels

$$N = 24$$

Hence,

$$B_T = N \times B_{SSB}$$

$$B_T = 24 \times 4$$

$$B_T = 96\ kHz$$

Step 3: Final Answer.

$$\boxed{B_T = 96\ kHz}$$

Hence,

$$\boxed{\text{Correct Option (C)}}$$

Quick Tip: For SSB transmission:

$$\text{Channel Bandwidth} = \text{Message Bandwidth}$$

Therefore,

$$B_{Total} = N \times B_m$$

---

98. A TDM system is used to multiplex four independent voice signals each sampled at the rate of 8 kHz using PAM. The system incorporates a synchronizing pulse train for its proper operation. Calculate the transmission bandwidth?

(A) 8 kHz

(B) 16 kHz

(C) 32 kHz

(D) 40 kHz

Correct Answer: (D)

Solution:

Concept:
In TDM-PAM, one frame contains samples from all channels plus one synchronization pulse. The pulse transmission rate determines the bandwidth.

Step 1: Determine pulses per frame.

Number of voice channels

$$N = 4$$

One synchronizing pulse is added.
Therefore,

$$\text{Pulses per frame} = 4 + 1 = 5$$

Step 2: Determine frame repetition rate.

Each channel is sampled at

$$f_s = 8\ kHz$$

Hence frame rate is also

$$8\ kHz$$

Step 3: Compute pulse rate.

$$R_p = 5 \times 8\ kHz$$

$$R_p = 40\ kHz$$

Transmission bandwidth approximately equals pulse rate.

$$B = 40\ kHz$$

Step 4: Final Answer.

$$\boxed{40\ kHz}$$

Hence,

$$\boxed{\text{Correct Option (D)}}$$

Quick Tip: For TDM-PAM:

$$B = (N + \text{sync pulses})f_s$$

Always include synchronization slots when calculating transmission rate.

---

99. What are the conditions for distortion less transmission through a system?

(A) Gain is linear and phase is constant

(B) Gain and phase should be linear

(C) Gain and phase must be constant

(D) Gain must be constant and phase should be linear

Correct Answer: (D)

Solution:

Concept:

For distortionless transmission, the output should be an attenuated and delayed version of the input.

Mathematically,

$$y(t) = Kx(t - t_0)$$

where K is a constant and t_0 is delay.

Step 1: Determine magnitude condition.

The magnitude response must be independent of frequency.

$$|H(\omega)| = K$$

Thus gain must be constant.

Step 2: Determine phase condition.

The phase response must be linear.

$$\phi(\omega) = -\omega t_0$$

A linear phase ensures equal delay for all frequency components.

Step 3: Final Answer.

Therefore,

$$\boxed{\text{Gain constant and phase linear}}$$

Hence,

$$\boxed{\text{Correct Option (D)}}$$

Quick Tip: Distortionless transmission requires:

$$|H(\omega)| = \text{constant}$$

and

$$\phi(\omega) = -\omega t_0$$

(Constant magnitude + Linear phase)

100. A random process is stationary

(A) If all statistical properties do not change with time

(B) If the future values of any sample function can be predicted from present values only

(C) Future values cannot be predicted

(D) If the future values of any sample function can be predicted from past values only

Correct Answer: (A)

Solution:

Concept:
A random process is said to be stationary when its statistical characteristics remain unchanged with time.
Typical statistical properties include:

$$\text{Mean}$$

$$\text{Variance}$$

$$\text{Autocorrelation}$$

Step 1: Define stationarity.

For a stationary process,

$$m_X(t) = \text{constant}$$

and

$$R_X(t_1, t_2) = R_X(t_1 - t_2)$$

Thus statistical behavior is independent of the absolute time origin.

Step 2: Evaluate the options.

Prediction of future values is related to predictability and not stationarity.

Stationarity only requires that statistical properties remain unchanged with time.

Step 3: Final Answer.

Therefore,

$$\boxed{\text{All statistical properties do not change with time}}$$

Hence,

$$\boxed{\text{Correct Option (A)}}$$

Quick Tip: A stationary random process has time-invariant statistical properties such as mean, variance and autocorrelation.

---

101. The random process $X(t) = A\cos(\omega_0 t + \theta)$ is wide-sense stationary, where A and ω_0 are constants and θ is a uniformly distributed random variable on the interval $(0, 2\pi)$. Then the mean value is

(A) > 0

(B) 1

(C) π

(D) 0

Correct Answer: (D)

Solution:

Concept:

The mean of a random process is

$$m_X(t) = E[X(t)]$$

For a uniformly distributed phase variable θ,

$$f_\theta(\theta) = \frac{1}{2\pi}, \qquad 0 < \theta < 2\pi$$

Step 1: Write the expectation.

$$m_X(t) = E[A\cos(\omega_0 t + \theta)]$$

$$= \frac{A}{2\pi}\int_0^{2\pi} \cos(\omega_0 t + \theta)\, d\theta$$

Step 2: Evaluate the integral.

$$m_X(t) = \frac{A}{2\pi}\Big[\sin(\omega_0 t + \theta)\Big]_0^{2\pi}$$

$$= \frac{A}{2\pi}[\sin(\omega_0 t + 2\pi) - \sin(\omega_0 t)]$$

$$= 0$$

Step 3: Final Answer.

$$\boxed{m_X(t) = 0}$$

Hence,

$$\boxed{\text{Correct Option (D)}}$$

Quick Tip: A sinusoidal random process with uniformly distributed phase over $(0, 2\pi)$ always has zero mean.

---

102. The following are properties of auto correlation except

(A) $|R_{XX}(\tau)| \le R_{XX}(0)$

(B) $|R_{XX}(-\tau)| = R_{XX}(\tau)$

(C) $|R_{XX}(\tau)| = E[x(t)]$

(D) If $x(t)$ has a periodic component, then $R_{XX}(\tau)$ will have a periodic component with the same period

Correct Answer: (C)

Solution:

Concept:
The autocorrelation function is defined as

$$R_{XX}(\tau) = E[X(t)X(t+\tau)]$$

It possesses several standard properties.
Step 1: Check property (A).

$$|R_{XX}(\tau)| \le R_{XX}(0)$$

This is a valid property.
Step 2: Check property (B).

Autocorrelation is an even function.

$$R_{XX}(\tau) = R_{XX}(-\tau)$$

Hence valid.
Step 3: Check property (D).

If the signal contains a periodic component, the autocorrelation also exhibits the same periodicity.
Hence valid.
Step 4: Identify the incorrect statement.

$$|R_{XX}(\tau)| = E[X(t)]$$

is not a standard autocorrelation property.

Therefore this statement is incorrect.

$$\boxed{\text{Correct Option (C)}}$$

Quick Tip: Autocorrelation is always an even function and attains its maximum magnitude at $\tau = 0$.

---

103. A receiver connected to an antenna whose resistance is 50 Ω has an equivalent noise resistance of 30 Ω. Calculate the receiver's noise figure?

(A) 0.6

(B) 1.0

(C) 1.6

(D) 2.0

Correct Answer: (C)

Solution:

Concept:

Noise figure in terms of equivalent noise resistance is

$$F = 1 + \frac{R_n}{R_s}$$

where

$$R_n = \text{equivalent noise resistance}$$

and

$$R_s = \text{source resistance}$$

Step 1: Substitute the given values.

$$R_n = 30\Omega$$

$$R_s = 50\Omega$$

$$F = 1 + \frac{30}{50}$$

Step 2: Simplify.

$$F = 1 + 0.6$$

$$F = 1.6$$

Step 3: Final Answer.

$$\boxed{F = 1.6}$$

Hence,

$$\boxed{\text{Correct Option (C)}}$$

Quick Tip: Noise Figure:

$$F = 1 + \frac{R_n}{R_s}$$

where R_n is equivalent noise resistance and R_s is source resistance.

---

104. Which of the following is not correct if the intermediate frequency is too high?

(A) Poor selectivity

(B) Poor adjacent channel rejection

(C) Increase tracking difficulties

(D) Poor image frequency rejection

Correct Answer: (D)

Solution:

Concept:

In a superheterodyne receiver, increasing IF has both advantages and disadvantages.

Step 1: Recall the effect of high IF.

A higher IF increases image frequency separation.

Hence image rejection improves.

Step 2: Examine the options.

High IF generally causes:

- Poor selectivity
- Poor adjacent channel rejection
- Tracking difficulties

However,

$$\text{Image frequency rejection improves}$$

and does not become poor.

Step 3: Final Answer.

Thus the incorrect statement is

$$\boxed{\text{Poor image frequency rejection}}$$

Hence,

$$\boxed{\text{Correct Option (D)}}$$

Quick Tip: High IF improves image rejection but worsens selectivity and tracking.

---

105. The problem of reducing the gain of receiver for weak signals is avoided by

(A) No AGC

(B) Ideal AGC

(C) Simple AGC

(D) Delayed AGC

Correct Answer: (D)

Solution:

Concept:
AGC (Automatic Gain Control) is used to maintain approximately constant output despite variations in input signal strength.

Step 1: Problem with simple AGC.

Simple AGC begins reducing gain even for weak signals.
This may deteriorate receiver sensitivity.

Step 2: Use delayed AGC.

Delayed AGC starts operating only after the signal exceeds a specified threshold.
Weak signals therefore experience full receiver gain.

Step 3: Final Answer.

Hence,

$$\boxed{\text{Delayed AGC}}$$

is used.

$$\boxed{\text{Correct Option (D)}}$$

Quick Tip: Delayed AGC improves weak-signal reception because gain reduction begins only after a preset signal level is reached.

---

106. In communication receivers, the filter used to reduce receiver gain at some specific frequency is

(A) Low pass

(B) High pass

(C) Band pass

(D) Notch

Correct Answer: (D)

Solution:

Concept:
A notch filter is designed to strongly attenuate a narrow range of frequencies while allowing others to pass.

Step 1: Understand the requirement.

The question asks for reduction of gain at a specific frequency.
This requires selective attenuation.

Step 2: Identify the filter.

A notch filter creates a deep attenuation at one frequency.

$$\boxed{\text{Notch Filter}}$$

Step 3: Final Answer.

$$\boxed{\text{Correct Option (D)}}$$

Quick Tip: Notch filters are commonly used to suppress unwanted interference at a particular frequency.

107. The Foster-Seeley discriminator along with amplitude limiter is called

(A) Envelope detector

(B) Ratio detector

(C) Balanced modulator

(D) Square-law modulator

Correct Answer: (B)

Solution:

Concept:

The Ratio Detector is a modified Foster-Seeley discriminator.

It provides FM demodulation with reduced sensitivity to amplitude variations.

Step 1: Recall the relationship.

A Foster-Seeley discriminator requires an amplitude limiter before demodulation.

The ratio detector incorporates this feature inherently.

Step 2: Identify the equivalent detector.

Therefore,

$$\boxed{\text{Foster-Seeley discriminator + limiter}}$$

corresponds to

$$\boxed{\text{Ratio Detector}}$$

Step 3: Final Answer.

$$\boxed{\text{Correct Option (B)}}$$

Quick Tip: Ratio detectors provide FM demodulation and reject amplitude variations without requiring a separate limiter stage.

---

108. A four level PAM has a input data rate of $2400\ bits/s$ **then the symbol rate is**

(A) $600\ symbols/s$

(B) $1200\ symbols/s$

(C) $2400\ symbols/s$

(D) $9600\ symbols/s$

Correct Answer: (B)

Solution:

Concept:

For an M-ary system,

$$R_b = R_s \log_2 M$$

where

$$R_b = \text{bit rate}$$

$$R_s = \text{symbol rate}$$

Step 1: Determine bits per symbol.

For four-level PAM,

$$M = 4$$

$$\log_2 4 = 2$$

Thus each symbol carries 2 bits.

Step 2: Compute symbol rate.

$$R_s = \frac{R_b}{2}$$

$$R_s = \frac{2400}{2}$$

$$R_s = 1200$$

symbols/s.

Step 3: Final Answer.

$$\boxed{1200\ symbols/s}$$

Hence,

$$\boxed{\text{Correct Option (B)}}$$

Quick Tip: For M-ary signaling:

$$R_s = \frac{R_b}{\log_2 M}$$

Four-level PAM carries 2 bits per symbol.

---

109. Match the following: A: ASK, B: APK, C: PSK, D: FSK

(A) A-IV, B-I, C-II, D-III

(B) A-III, B-II, C-I, D-IV

(C) A-II, B-III, C-IV, D-I

(D) A-I, B-IV, C-III, D-II

Correct Answer: (A)

Solution:

Concept:

Digital modulation techniques modify one parameter of a carrier.

$$\text{ASK} \longrightarrow \text{Amplitude}$$

$$\text{PSK} \longrightarrow \text{Phase}$$

$$\text{FSK} \longrightarrow \text{Frequency}$$

$$\text{APK} \longrightarrow \text{Pulse Amplitude}$$

Step 1: Match the quantities.

$$A \rightarrow IV$$

$$B \rightarrow I$$

$$C \rightarrow II$$

$$D \rightarrow III$$

Step 2: Select the correct option.

$$\boxed{A-IV,\ B-I,\ C-II,\ D-III}$$

Hence,

$$\boxed{\text{Correct Option (A)}}$$

Quick Tip: ASK changes amplitude, PSK changes phase and FSK changes frequency of the carrier signal.

---

110. If the rows of channel matrix are all equal to each other, then the channel capacity is

(A) 0

(B) 1

(C) ∞

(D) Equal to bandwidth of channel

Correct Answer: (A)

Solution:

Concept:
Channel capacity measures the maximum information that can be transmitted reliably through a channel.

Step 1: Interpret equal rows of channel matrix.

If all rows are identical, the output distribution becomes independent of the transmitted input symbol.
Thus the receiver cannot distinguish which input symbol was sent.

Step 2: Evaluate mutual information.

Since output gives no information about input,

$$I(X;Y) = 0$$

Step 3: Compute channel capacity.

Channel capacity is

$$C = \max I(X;Y)$$

Hence,

$$C = 0$$

Step 4: Final Answer.

$$\boxed{C = 0}$$

Therefore,

$$\boxed{\text{Correct Option (A)}}$$

Quick Tip: If every row of a channel matrix is identical, the output is independent of the input and the channel capacity becomes zero.

---

111. The Gauss's law for magnetic fields is

(A) $\nabla \cdot E = 0$

(B) $\nabla \cdot H = 0$

(C) $\nabla \cdot B = 0$

(D) $\nabla \cdot D = 0$

Correct Answer: (C)

Solution:

Concept:

Gauss's law for magnetism is one of Maxwell's equations.

It states that magnetic monopoles do not exist and hence the net magnetic flux through any closed surface is zero.

$$\oint_S \vec{B} \cdot d\vec{S} = 0$$

The differential form is

$$\nabla \cdot \vec{B} = 0$$

Step 1: Recall Maxwell's equation for magnetic fields.

The divergence of magnetic flux density is always zero.

$$\nabla \cdot B = 0$$

Step 2: Interpret the physical meaning.

Since isolated magnetic charges do not exist, magnetic field lines always form closed loops. Therefore, the net magnetic flux leaving a closed surface is zero.

Step 3: Final Answer.

$$\boxed{\nabla \cdot B = 0}$$

Hence,

$$\boxed{\text{Correct Option (C)}}$$

Quick Tip: Gauss's law for electricity:

$$\nabla \cdot D = \rho_v$$

Gauss's law for magnetism:

$$\nabla \cdot B = 0$$

---

112. Identify the polarisation of the wave given, $E_x = 2\cos\omega t$, $E_y = 2\sin\omega t$ and the phase difference is $+90^\circ$

(A) Left hand circularly polarized

(B) Right hand circularly polarized

(C) Left hand elliptically polarized

(D) Right hand elliptically polarized

Correct Answer: (A)

Solution:

Concept:

For circular polarization:

- Magnitudes of E_x and E_y must be equal.
- Phase difference must be $\pm 90^\circ$.

Step 1: Check amplitude condition.

Given

$$E_x = 2\cos\omega t$$

$$E_y = 2\sin\omega t$$

Both components have equal amplitudes.

$$|E_x| = |E_y| = 2$$

Hence circular polarization is possible.

Step 2: Check phase difference.

$$E_y = 2\sin\omega t = 2\cos(\omega t - 90^\circ)$$

Thus E_y lags E_x by 90°.

This corresponds to left-hand circular polarization.

Step 3: Final Answer.

$$\boxed{\text{Left Hand Circular Polarization}}$$

Hence,

$$\boxed{\text{Correct Option (A)}}$$

Quick Tip: Equal amplitudes + phase difference of 90°

$$\Rightarrow$$

Circular polarization.

---

113. The characteristic impedance of the medium is

(A) 120π

(B) $120\pi\left(\frac{\mu_r}{\epsilon_r}\right)$

(C) $\mu_r\epsilon_r$

(D) ϵ_r

Correct Answer: (B)

Solution:

Concept:

Intrinsic impedance of a medium is

$$\eta = \sqrt{\frac{\mu}{\epsilon}}$$

For a dielectric medium,

$$\eta = 120\pi\sqrt{\frac{\mu_r}{\epsilon_r}}$$

Step 1: Write the intrinsic impedance formula.

$$\eta = 120\pi\sqrt{\frac{\mu_r}{\epsilon_r}}$$

Step 2: Compare with given options.

The intended answer among the given options is

$$120\pi\left(\frac{\mu_r}{\epsilon_r}\right)$$

which corresponds to Option (B).

Step 3: Final Answer.

$$\boxed{\text{Correct Option (B)}}$$

Quick Tip: For free space:

$$\eta_0 = 120\pi = 377\Omega$$

---

114. The depth of penetration decreases, if

(A) Conductivity decreases

(B) Relative permeability decreases

(C) Frequency increases

(D) Phase velocity decreases

Correct Answer: (C)

Solution:

Concept:

Skin depth (depth of penetration) is

$$\delta = \sqrt{\frac{2}{\omega\mu\sigma}}$$

where

$$\omega = 2\pi f$$

Step 1: Observe the dependence.

$$\delta \propto \frac{1}{\sqrt{f}}$$

Thus skin depth decreases when frequency increases.

Step 2: Physical interpretation.

At higher frequencies, current tends to flow closer to the conductor surface.

Therefore penetration depth becomes smaller.

Step 3: Final Answer.

$$\boxed{\text{Frequency increases}}$$

Hence,

$$\boxed{\text{Correct Option (C)}}$$

Quick Tip: Skin depth decreases when:

$$f\uparrow, \quad \mu\uparrow, \quad \sigma\uparrow$$

---

115. Snell's law states that (n_1 is the index of refraction of the first medium and n_2 is that of second, θ_t is the angle of refraction and θ_i is angle of incidence)

(A) $n_1 \sin\theta_i = 0$

(B) $n_2 \sin\theta_t = 0$

(C) $n_2 \sin\theta_t = n_1 \sin\theta_i$

(D) $n_2 \sin\theta_t = -n_1 \sin\theta_i$

Correct Answer: (C)

Solution:

Concept:

Snell's law relates the angles of incidence and refraction when an electromagnetic wave travels from one medium to another.

$$n_1 \sin\theta_i = n_2 \sin\theta_t$$

Step 1: Write Snell's law.

$$n_1 \sin\theta_i = n_2 \sin\theta_t$$

Step 2: Compare with options.

Rearranging,

$$n_2 \sin\theta_t = n_1 \sin\theta_i$$

which matches option (C).

Step 3: Final Answer.

$$\boxed{n_2 \sin\theta_t = n_1 \sin\theta_i}$$

Hence,

$$\boxed{\text{Correct Option (C)}}$$

Quick Tip: Snell's Law:

$$n_1 \sin\theta_i = n_2 \sin\theta_t$$

It governs refraction of electromagnetic waves.

---

116. For a lossy transmission line, the characteristic impedance does not depend on

(A) Operating frequency of the line

(B) Length of the line

(C) Conductivity of the conductors

(D) Conductivity of the dielectric separating the conductors

Correct Answer: (B)

Solution:

Concept:

The characteristic impedance of a lossy transmission line is

$$Z_0 = \sqrt{\frac{R + j\omega L}{G + j\omega C}}$$

where

- R = resistance per unit length
- L = inductance per unit length
- G = conductance per unit length
- C = capacitance per unit length

Step 1: Identify the parameters affecting characteristic impedance.

From the formula,

$$Z_0 = \sqrt{\frac{R + j\omega L}{G + j\omega C}}$$

it depends on:

$$R,\ L,\ G,\ C,\ \omega$$

Thus it depends on conductor conductivity, dielectric conductivity and operating frequency.

Step 2: Check dependence on length.

Since R, L, G, C are per-unit-length quantities, the total length of the line does not appear in the expression.

Therefore characteristic impedance remains unchanged with line length.

Step 3: Final Answer.

$$\boxed{\text{Characteristic impedance is independent of line length}}$$

Hence,

$$\boxed{\text{Correct Option (B)}}$$

Quick Tip: Characteristic impedance depends on the distributed parameters R, L, G, C, but not on the physical length of the transmission line.

---

117. A rectangular waveguide with dimensions $a = 2.5$ cm, $b = 1$ cm is operating below 15 GHz. If the guide is filled with a medium characterized by $\sigma = 0$, $\epsilon = 4\epsilon_0$, $\mu_r = 1$, calculate the cutoff frequency of dominant mode?

(A) 3 GHz

(B) 9 GHz

(C) 12 GHz

(D) 15 GHz

Correct Answer: (A)

Solution:

Concept:

The dominant mode in a rectangular waveguide is the TE_{10} mode.

The cutoff frequency is

$$f_c = \frac{c}{2a\sqrt{\mu_r \epsilon_r}}$$

where

$$a = 2.5\,cm = 0.025\,m$$

$$\epsilon_r = 4, \qquad \mu_r = 1$$

Step 1: Substitute the values into the cutoff frequency formula.

$$f_c = \frac{3 \times 10^8}{2(0.025)\sqrt{4}}$$

$$= \frac{3 \times 10^8}{0.1}$$

$$= 3 \times 10^9 \ Hz$$

Step 2: Convert into GHz.

$$f_c = 3 \ GHz$$

Step 3: Final Answer.

$$\boxed{f_c = 3 \ GHz}$$

Hence,

$$\boxed{\text{Correct Option (A)}}$$

Quick Tip: For rectangular waveguides,

$$f_{c(TE_{10})} = \frac{c}{2a\sqrt{\mu_r \epsilon_r}}$$

TE_{10} is always the dominant mode.

---

118. Find the maximum gain of a $\frac{\lambda}{2}$ wire dipole operating at 30 MHz?

(A) 1

(B) 1.64

(C) 10.23

(D) 30.33

Correct Answer: (B)

Solution:

Concept:

A half-wave dipole antenna is one of the most important practical antennas.
Its standard directivity and gain are well-known quantities.
The maximum power gain of a half-wave dipole is

$$G = 1.64$$

or

$$G_{dB} = 2.15\ dB$$

Step 1: Recall the standard gain value.

For a half-wave dipole,

$$G = 1.64$$

Step 2: Observe that operating frequency does not change the normalized gain.

The frequency determines physical dimensions but not the normalized gain value.
Therefore,

$$G = 1.64$$

Step 3: Final Answer.

$$\boxed{1.64}$$

Hence,

$$\boxed{\text{Correct Option (B)}}$$

Quick Tip: Important antenna gains:

$$\text{Isotropic antenna} = 1$$

$$\text{Half-wave dipole} = 1.64$$

$$\text{Half-wave dipole} = 2.15\ dB$$

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119. The S-parameters are used at microwave frequencies because of the following reasons, except

(A) Equipment is readily available to measure total voltage and total current

(B) Short and open circuits are difficult to achieve over a broad band of frequencies

(C) Logical variables to use at the microwave frequencies are travelling waves

(D) More stable Active devices are used at short or open circuits

Correct Answer: (A)

Solution:

Concept:
At microwave frequencies, voltages and currents cannot be measured conveniently. Therefore scattering parameters (S-parameters) based on incident and reflected waves are used.

Step 1: Recall the advantages of S-parameters.

S-parameters are preferred because:

- Travelling waves are easy to measure.
- Open and short circuits are difficult to realize.

- Microwave devices are naturally characterized using wave quantities.

Step 2: Check the incorrect statement.

Option (A) states that total voltage and current are readily measurable.

This is exactly the opposite of the reason S-parameters are used.

Hence Option (A) is incorrect.

Step 3: Final Answer.

$$\boxed{\text{Correct Option (A)}}$$

Quick Tip: At microwave frequencies:

$$\text{Wave quantities} \rightarrow \text{S-parameters}$$

instead of

$$V, I$$

based measurements.

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120. The fraction of time that the transmitter is transmitting during one radar cycle is called

(A) Pulse Repeating Frequency

(B) Pulse Repetition Interval

(C) Pulses Per Second

(D) Duty Factor

Correct Answer: (D)

Solution:

Concept:

Duty factor (or duty cycle) is defined as the fraction of time during which a radar transmitter is ON.

Mathematically,

$$\text{Duty Factor} = \frac{\text{Pulse Width}}{\text{Pulse Repetition Interval}}$$

Step 1: Understand the meaning of duty factor.

During one radar cycle, the transmitter operates only for a small pulse duration.

The ratio of transmitting time to total cycle time is called duty factor.

$$DF = \frac{\tau}{T}$$

where

$$\tau = \text{pulse width}$$

and

$$T = \text{pulse repetition interval}$$

Step 2: Compare with the options.

The definition exactly matches duty factor.

Step 3: Final Answer.

$$\boxed{\text{Duty Factor}}$$

Hence,

$$\boxed{\text{Correct Option (D)}}$$

Quick Tip: Radar Duty Factor:

$$DF = \frac{\text{Pulse Width}}{\text{PRI}}$$

Average Power:

$$P_{avg} = DF \times P_{peak}$$
