CN Final 16 Questions and Answers

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Qno=1). For each of the following four networks, discuss the consequences if a connection fails.

Answer:

- *If a connection fails in a star topology network, the device that the failed connection is connected to will be isolated and unable to communicate with the rest of the network. However, the rest of the devices will still be able to communicate with each other.*
- *If a connection fails in a bus topology network, the segment of the network that the failed connection is on will be isolated and unable to communicate with the rest of the network. However, the rest of the segments will still be able to communicate with each other.*
- *If a connection fails in a ring topology network, the entire network will be affected as the connection failure will break the ring and disrupt the flow of data.*
- *If a connection fails in a mesh topology network, the network will still function because there are multiple paths for data to travel between devices. The failed connection can be bypassed and the network will continue to operate.*

Qno=2) Performance is inversely related to delay. When you use the Internet, which of the following applications are more sensitive to delay?

Answer:

When you use the Internet, the applications that are more sensitive to delay are typically real-time applications such as:

Voice over IP (VoIP) and video conferencing, as even a small amount of delay can make the conversation difficult to understand

Online gaming, as even a small amount of delay can cause lag and make the game less responsive

Remote control and telemetry, as even a small amount of delay can make the control less responsive.

Streaming of live events such as sports, concerts and so on, as even a small amount of delay can make the experience less enjoyable.

On the other hand, applications that are less sensitive to delay are those that can tolerate some delay in the transmission of data, such as:

E-mail - A delay in the transmission of e-mails would not significantly impact the user experience.

File transfer - A delay in the transmission of files would not significantly impact the user experience.

Web browsing - A delay in the transmission of web pages would not significantly impact the user experience.

In computer networks, delay is an important factor in determining the performance of a network and its ability to support different types of applications. The network design and the protocols used must be able to minimize delay in order to ensure that real-time applications perform well.

Sending an e-mail

Sending an email is a non-real-time application that is typically less sensitive to delay. Email is a store-and-forward application, which means that messages are stored at various points along the way and then forwarded to their final destination. This process can take some time, but the delay is generally not noticeable to the user. Even if there is a small amount of delay in sending an email, it will not affect the overall experience or the integrity of the message.

Copying a file

Copying a file is a non-real-time application that is typically less sensitive to delay. When copying a file, the data is broken down into smaller packets and sent to the destination, where it is reassembled. The delay in sending each packet, does not affect the overall experience, as long as the packets arrive in the correct order and the file is reassembled correctly at the destination.

Surfing the Internet

Surfing the Internet refers to the activity of navigating and exploring the World Wide Web using a web browser. This can include browsing websites, searching for information, and engaging with online content such as videos, social media, and e-commerce platforms.

Qno=3). A local telephone call made between two parties is a point-to-point connection.

Answer:

A point-to-point connection is a direct communication link between two devices or networks, where the communication is only between the two endpoints and no other devices or networks are involved.

In the case of a local telephone call, the call is made from one telephone to another, and the connection is established directly between the two phones without any other devices or networks involved.

Qno=4). Compare the telephone network and the Internet. What are the similarities? What are the differences?

Answer:

The telephone network and the Internet are both communication networks, but they have some distinct differences.

Similarities:

Both networks use the same technology of converting sound into electrical signals which are then transmitted over wires or wireless mediums.

B*oth networks use routing to direct the signals to their intended destination.*

Both networks have a set of protocols that govern how data is transmitted.

Differences

The telephone network is designed for voice communication, while the Internet is designed for data communication.

The telephone network is optimized for voice transmission and has a higher quality of service for voice communication.

The telephone network uses circuit-switched connections, which means that a dedicated connection is established between two parties for the duration of the call.

Qno=5). Suppose a computer sends a frame to another computer on a bus topology LAN. The physical destination address of the frame is corrupted during the transmission. What happens to the frame? How can the sender be informed about the situation?

Answer:

If the physical destination address of the frame is corrupted during transmission on a bus topology LAN, the receiving computer will not be able to properly identify the frame and will discard it. The sender will not be directly informed about this situation, but it can be determined through network *monitoring tools or by observing a lack of a response from the intended receiver. Additionally, the sender can use error detection and correction techniques, such as checksums, to detect and correct any errors that may occur during transmission, preventing frames from being discarded due to corrupted addresses.*

Qno=6). Suppose a computer sends a packet at the network layer to another computer somewhere in the Internet. The logical destination address of the packet is corrupted. What happens to the packet? How can the source computer be informed of the situation?

Answer:

If the logical destination address of the packet is corrupted at the network layer, when it is in transit through the Internet, the packet will likely be discarded by the routers along the way. The source computer will not be directly informed of this situation, but it can be determined through network monitoring tools or by observing a lack of a response from the intended receiver. Additionally, the source computer can use error detection and correction techniques, such as checksums, to detect and correct any errors that may occur during transmission, preventing packets from being discarded due to corrupted addresses.

Qno=7). Suppose a computer sends a packet at the transport layer to another computer somewhere in the Internet. There is no process with the destination port address running at the destination computer. What will happen?

Answer:

If a computer sends a packet at the transport layer to another computer on the Internet and there is no process running on the destination computer with the specified destination port address, the destination computer will send back a "destination port unreachable" message to the source computer. This message is typically a type of Internet Control Message Protocol (ICMP) message indicating that the destination port is closed and no process is listening on that port. The packet will not be delivered to the intended process and the sending device will be notified of failure to deliver the packet.

Qno=8). If the data link layer can detect errors between hops, why do you think we need another checking mechanism at the transport layer?

Answer:

The data link layer can detect errors that occur within a single hop, such as a corrupted packet due to noise on the physical link. However, errors can also occur between hops, such as packets being dropped or delivered out of order. Additionally, the transport layer provides additional functionality, such as ensuring reliable delivery of data and flow control, which the data link layer does not provide. Therefore, it is necessary to have error detection and correction mechanisms at the transport layer to ensure the end-to-end reliability of the data being transmitted.

Qno=9). 1) Express a period of 100 ms in microseconds

Answer:

100 milliseconds (ms) is equal to 100,000 microseconds (μs).

Qno=9). b) The period of a signal is 100 ms. what is its frequency in kilohertz?

Answer:

The frequency of a signal is the reciprocal of its period. So, if the period of a signal is 100 ms, its frequency can be calculated as:

Frequency (kHz) = 1 / Period (s)

So the frequency in kilohertz (kHz) = 1 / (100 x 10^-3) = 10000 Hz = 10 KHz

Note that 1kHz = 1000 Hz

The period of the signal is 100 ms = 100 * 0.001 = 0.1 s

*Yes, that's correct. To convert milliseconds (ms) to seconds (s), you can simply multiply by 0.001. Therefore, 100 ms = 100 * 0.001 = 0.1 s.*

The frequency of a signal is the reciprocal of its period. So, if the period of a signal is 0.1 s, its frequency can be calculated as:

Frequency (Hz) = 1 / Period (s)

So the frequency in hertz (Hz) = 1 / 0.1 = 10 Hz

Qno=9). c) A sine wave is offset cycle with respect to time 0. What is its phase in degrees and radians?

Answer:

A sine wave that is offset with respect to time "O" is said to be in a phase-shifted state. The amount of phase shift is dependent on the amount of time offset.

The phase shift can be measured in both degrees and radians.

In degrees, the phase shift is given by the formula: Phase Shift (degrees) = (360 x Time offset (s)) / Period (s) of the sine wave

In radians, the phase shift is given by the formula: Phase Shift (rad) = (2π x Time offset (s)) / Period (s) of the sine wave.

Qno=9). d) If a periodic signal is decomposed into five sine waves with frequencies of 100, 300, 500, 700, and 900 Hz, what is its bandwidth? Draw the spectrum, assuming all components have a maximum amplitude of 10V?

Answer:

The bandwidth of a periodic signal is the difference between the highest and lowest frequencies of the signal. In this case, the highest frequency is 900 Hz and the lowest frequency is 100 Hz, so the bandwidth of the signal is 900 - 100 = 800 Hz.

A spectrum is a graphical representation of the frequency components of a signal, with the horizontal axis showing frequency and the vertical axis showing amplitude. The spectrum of a signal that is decomposed into five sine waves with frequencies of 100, 300, 500, 700, and 900 Hz would be shown as five vertical lines, each representing one of the sine wave components.

Assuming all components have a maximum amplitude of 10 V, the vertical axis would be in volts and the horizontal axis in Hz. The vertical lines for the 100, 300, 500, 700, and 900 Hz components would be 10 V, 10 V, 10 V, 10 V and 10 V respectively.

It would be like a bar graph with 5 bars, each with the amplitude of 10 Volts and the horizontal axis representing the frequencies (100, 300, 500, 700 and 900 Hz) respectively.

Qno=9). e) A periodic signal has a bandwidth of 20 Hz. The highest frequency is 60 Hz. What is the lowest frequency? Draw the spectrum if the signal contains all frequencies of the same amplitude.

Answer:

The bandwidth of a periodic signal is the difference between the highest and lowest frequencies present in the signal. If the highest frequency present in the signal is 60 Hz and the bandwidth is 20 Hz, the lowest frequency present in the signal is given by:

Lowest frequency = Highest frequency - Bandwidth

Lowest frequency = 60 Hz - 20 Hz = 40 Hz

A signal with a bandwidth of 20 Hz, and highest frequency of 60 Hz, and lowest frequency of 40 Hz, contains all frequencies of the same amplitude. The frequency spectrum of this signal will be represented as a continuous band of frequencies between 40 Hz and 60 Hz, with each frequency having the same amplitude. The amplitude of the signal is the height of the signal and it's the same for all the frequencies.

It would look something like this:

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Qno=10). a) What is the required bandwidth of a low-pass channel if we need to send 1 Mbps by using baseband transmission?

Answer:

Baseband transmission is a method of transmitting digital data over a single channel where the data is encoded into a digital signal that is transmitted over the channel without modulation. In baseband transmission, the data rate and the bandwidth are equal.

Therefore, if we need to send 1 Mbps (megabits per second) of data by using baseband transmission, the required bandwidth of the channel is also 1 Mbps.

Qno=10). b) We have a low-pass channel with bandwidth 100 kHz. What is the maximum bit rate of this channel?

Answer:

*The maximum bit rate can be achieved if we use the first harmonic. The bit rate is 2 times the available bandwidth, or **200 kbps**.*

Qno=11). a) What is the relationship between period and frequency?

Answer:

Frequency and period are the inverse of each other. $T = 1/f$ and $f = 1/T$..

Qno=11). b)What does the amplitude of a signal measure?

Answer:

*The amplitude (peak) of a wave is **the measure of its displacement from a zero value**. More plainly, it's how tall the "high points" are throughout an analog wave signal, usually measured in volts.*

Qno=11). c)What does the frequency of a signal measure?.

Answer:

The frequency of a signal measures the number of oscillations or cycles of the signal in one second, typically measured in hertz (Hz). It represents how fast the signal oscillates or varies over time.

Qno=11). d)What does the phase of a signal measure?

Answer:

In electronic signaling, a phase is the position of a wave at a point in time (instant) on a waveform cycle. It provides a measurement of exactly where the wave is positioned within its cycle, using either degrees (0-360) or radians (0-2π). One radian of a phase is equal to about 57.3 degrees.

Qno=11) e)How can a composite signal be decomposed into its individual frequencies?

Answer:

A composite signal can be decomposed into its individual frequencies using Fourier analysis, specifically the Fast Fourier Transform (FFT) algorithm which is an efficient way to compute the DFT. This algorithm breaks down a signal into its frequency components allowing for the analysis and manipulation of the individual frequencies.

Qno=12. (a) Name three types of transmission impairment.

Answer:

Noise

Attenuation

Distortion

Qno=12 (b) Distinguish between baseband transmission and broadband transmission

Answer:

Baseband transmission and broadband transmission are two different types of data transmission methods.

Baseband transmission is a method in which a single channel is used to transmit a single digital or analog signal. The signal is transmitted at its original frequency, known as the baseband frequency, without any modulation. This type of transmission is commonly used for digital data transmission over short distances, such as within a LAN.

On the other hand, broadband transmission is a method in which multiple channels are used to transmit multiple signals simultaneously. The signals are modulated to different frequencies, known as the broadband frequencies, and are transmitted together over a single channel. This type of transmission is commonly used for analog and digital data transmission over long distances, such as over a WAN or the internet.

In summary, baseband transmission uses a single channel to transmit a single signal at its original frequency, whereas broadband transmission uses multiple channels to transmit multiple signals at different frequencies simultaneously.

Qno=12 (c) Distinguish between a low-pass channel and a band-pass channel.

Answer:

Low pass channel or medium with the bandwidth that starts from zero.

Band pass channel has the bandwidth that does not start from zero.
A low-pass channel has an upper limit to the frequencies
band-pass channel has both an upper limit and a lower limit to the frequencies

Qno=12 (d) What does the Nyquist theorem have to do with communications?

Answer:

The Nyquist theorem defines the maximum bit rate of a noiseless channel.

The Shannon capacity determines the theoretical maximum bit rate of a noisy Channel.

Qno=12 (e) What does the Shannon capacity have to do with communications

Answer:

The Shannon capacity determines the theoretical maximum bit rate of a noisy channel.

Qno=13) Two channels, one with a bit rate of 100 kbps and another with a bit rate of 200 kbps, are to be multiplexed. How this can be achieved? What is the frame rate? What is the frame duration? What is the bit rate of the link?

Answer:

Multiplexing is the process of combining multiple channels into a single channel to increase the overall capacity of the link. There are several methods to achieve multiplexing, including time-division multiplexing (TDM) and frequency-division multiplexing (FDM).

One way to achieve multiplexing of two channels with bit rates of 100 kbps and 200 kbps is through time-division multiplexing (TDM). In TDM, each channel is given a time slot in which to transmit its data. For example, the 100 kbps channel could transmit its data in the first time slot, and the 200 kbps channel could transmit its data in the second time slot. The time slots are repeated continuously, creating a "frame" that contains the data from both channels.

For Example: *if the frame rate is set at 10 frames per second, the frame duration would be 0.1 seconds (1/10 seconds). The time slot for the 100 kbps channel would be 1/10 of the frame duration, and the time slot for the 200 kbps channel would be 2/10 of the frame duration.*

The bit rate of the link is the sum of the bit rates of the two channels being multiplexed, in this case it would be 100 kbps + 200 kbps = 300 kbps.

It's important to note that, the information from the channels are sent in the frames, the frames are sent in a sequence, and the channels are synchronized with each other for the correct information to be received.

Qno=14) <u>Define FHSS and DSSS and explain how it achieves bandwidth spreading?</u>

FHSS (Frequency Hopping Spread Spectrum) and DSSS (Direct Sequence Spread Spectrum)

FHSS uses a narrowband carrier that changes frequency in a pattern known to both transmitter and receiver. By rapidly switching the carrier frequency among many frequency channels, FHSS spreads the signal's energy over a wide bandwidth.

DSSS uses a direct sequence to spread the signal's energy over a wide bandwidth by modulating the data with a high-rate bit sequence. DSSS uses a spreading code to spread the narrowband data signal over a much wider bandwidth, allowing for more resistance to interference and a greater number of users.

Both FHSS and DSSS achieve bandwidth spreading by spreading the signal energy over a wide frequency band, making it harder to detect and intercept the signal. This technique is known as "frequency spreading" and it allows for more efficient use of the available spectrum and increased resistance to interference

Qno=15) <u>Distinguish between multilevel TDM, multiple slot TDM, and pulse-stuffed TDM?</u>

In multilevel time-division multiplexing (TDM), several signals are transmitted simultaneously in a time-division fashion on multiple levels. Each level carries a group of lower-rate channels, and these groups are then multiplexed together to form a higher-rate channel. This technique allows for a higher capacity than traditional TDM by using multiple levels of multiplexing.

In multiple-slot TDM, instead of dividing the time slot into different channels, the time slot is divided into multiple smaller slots, each of which is assigned to a different channel. This allows for a higher number of channels to be multiplexed together on the same time slot.

In pulse-stuffed TDM, a signal is transmitted by inserting extra pulses between the original pulses of the signal. This allows for a higher number of channels to be multiplexed together by reducing the time between pulses, but at the cost of adding extra overhead to the signal.

Qno=16) <u>Which of the three multiplexing techniques is common for fiber optic links? Explain the reason</u>

Frequency-division multiplexing (FDM) is the most common multiplexing technique used for fiber optic links.

The reason for this is that fiber optic links use light to transmit data, and different wavelengths of light can be used to transmit different channels of data simultaneously. FDM works by dividing the available bandwidth into different frequency ranges (also known as channels) and allocating each channel to a specific data stream.

Fiber optic links have a wide bandwidth compared to traditional copper wire links, so they are able to transmit multiple channels of data simultaneously using FDM. Additionally, FDM is a more robust technique for transmitting data over long distances, as the different channels can be amplified separately, and are less prone to interference.

The other methods, TDM and CDM, are not commonly used in fiber optic links as they are less efficient in terms of bandwidth utilization and they are more susceptible to noise and other forms of interference.