Câu 1: Explain the difference between connectionless unacknowledged service and connectionless acknowledged service. How do the protocols that provide these services differ?

Connectionless service is a network communication service that does not establish a dedicated communication path between the sender and receiver. In connectionless unacknowledged service, also known as "fire and forget", the sender sends a message without ensuring that it has been received by the receiver. The receiver may or may not receive the message, and there is no confirmation sent back to the sender.

In contrast, in connectionless acknowledged service, the sender transmits a message and waits for a confirmation of receipt from the receiver. If the receiver receives the message, it sends an acknowledgement back to the sender. If the sender does not receive an acknowledgement, it can assume that the message was not received.

The protocols that provide these services differ in their handling of packet loss and protocol overhead. Connectionless unacknowledged service protocols, such as UDP (User Datagram Protocol), are lightweight and have minimal overhead but provide no guarantee of message delivery. Connectionless acknowledged service protocols, such as ICMP (Internet Control Message Protocol) and ARP (Address Resolution Protocol), have more overhead but provide greater reliability by sending acknowledgements of successful message delivery.

Câu 2. Explain the difference between connection-oriented acknowledged service and connectionless acknowledged service. How do the protocols that provide these services differ?

Connection-oriented acknowledged service is a network communication service that establishes a dedicated connection between the sender and receiver before any data transmission takes place. In this service, the receiver sends an acknowledgement to the sender after receiving each packet of data.

In contrast, connectionless acknowledged service does not establish a dedicated connection before data transmission. The sender simply sends packets to the receiver, and the receiver sends acknowledgements back to the sender for each received packet.

The protocols that provide these services differ in their handling of packet loss and

protocol overhead. Connection-oriented acknowledged service protocols, such as TCP (Transmission Control Protocol), use a three-way handshake to establish a reliable connection between the sender and receiver. This protocol also implements flow control and congestion control mechanisms to ensure efficient data transfer and minimize packet loss. Once the data transmission is complete, the connection is terminated.

Connectionless acknowledged service protocols, such as UDP (User Datagram Protocol), do not establish a dedicated connection with the receiver. Instead, they simply send the packets to the receiver without guaranteeing their delivery or order. These protocols have lower overhead and are useful in applications where speed is more important than accuracy, such as online gaming and live streaming.

In summary, connection-oriented acknowledged service is reliable but has higher overhead and may be slower, while connectionless acknowledged service is faster but less reliable and does not guarantee the order of packet delivery.

Câu 3. Explain the differences between PPP and HDLC.

	PPP	HDLC
Full form	Point-to-Point Protocol	High-level Data Link Control
Flexibility	PPP is a more flexible protocol than HDLC. It can be used to carry multiple network layer protocols, including IP, IPX, and AppleTalk.	HDLC primarily designed for carrying only one protocol.
Error detection	PPP has a better error detection mechanism. PPP uses a cyclic redundancy check (CRC) for detecting errors. The CRC is considered more effective in detecting errors in data transmission.	HDLC uses a frame check sequence (FCS).
Configuration	PPP is easier to config. PPP uses a configuration protocol called LCP (Link Control Protocol), which	HDLC requires manual configuration of parameters, such as the

	automates the configuration process.	address field and control field.
Authentication	PPP supports authentication mechanisms such as PAP (Password Authentication Protocol) and CHAP (Challenge Handshake Authentication Protocol), which authenticate the identity of the sender	HDLC does not support any authentication mechanism.
Sliding Window Protocol	PPP does not support any silding window protocol.	HDLC uses a sliding window protocol for flow control. The sliding window protocol is used to optimize the flow of data between the sender and receiver, ensuring that the receiver is not overwhelmed with too much data at once.

In summary, PPP is a more flexible, easier to configure and better error detection mechanism than HDLC. Additionally, PPP offers mechanisms like authentication, which HDLC lacks. However, HDLC uses a sliding window protocol for flow control, which PPP does not provide.

Câu 4. A 1.5 Mbps communications link is to use HDLC to transmit information to the moon. What is the smallest possible frame size that allows continuous transmission? The distance between earth and the moon is approximately 375,000 km, and the speed of light is 3×10^8 meters/second.

To determine the smallest possible frame size that allows continuous transmission, we need to calculate the round-trip time (RTT) for a signal to travel from Earth to the Moon and back.

$$RTT = \frac{2 \times distance}{speed\ of\ light}$$

Converting the distance between Earth and the Moon to meters:

375,000 km × 1000 m/km = 375,000,000 meters

Plugging in the values:

$$RTT = \frac{2 \times 375{,}000{,}000}{3 \times 10^8} = 5\ (second)$$

To achieve continuous transmission on a 1.5 Mbps link with HDLC, we need to calculate the minimum frame size that can be transmitted within this RTT.

$$Minimum\ Frame\ Size = \frac{link\ speed \times RTT}{efficiency}$$

The efficiency factor takes into account the protocol overhead, such as header and trailer bits.

Assuming an efficiency factor of 80% (efficiency = 0.8), we get:

Minimum frame size = $\frac{1.5\ \mathrm{Mbps} \times 5\ \mathrm{sec}}{0.8} = 11.25\ megabits$

To convert to bytes:

$$\frac{11.25\ \mathrm{megabits}}{8\ bits/byte} = 1.41\ \mathrm{megabytes}$$

Therefore, the smallest possible frame size that allows continuous transmission on a 1.5 Mbps HDLC link to the Moon is approximately 1.41 megabytes.

Câu 5. Suppose HDLC is used over a 1.5 Mbps geostationary satellite link. Suppose that 250-byte frames are used in the data link control. What is the maximum rate at which information can be transmitted over the link?

To determine the maximum rate at which information can be transmitted over a 1.5 Mbps geostationary satellite link using HDLC with 250-byte frames, we need to take into account the overhead of the HDLC protocol.

The HDLC frame consists of several fields, including flag characters, address, control, data, and CRC (Cyclic Redundancy Check). The flag characters mark the beginning and end of the frame and are each one byte long. The address and control fields are each one byte long. The CRC field is two bytes long.

Therefore, the total size of an HDLC frame is:

$$\begin{aligned} Frame\ Size &= Data + Address\ (1\ byte) + Control\ (1\ byte) \\ &+ CRC\ (2\ bytes) + Flag\ (1\ byte) = Data + 6\ bytes \end{aligned}$$

For 250-byte frames, the total frame size is:

$$Frame\ Size = 250 + 6 = 256\ bytes$$

To calculate the maximum rate of information transmission, we need to divide the link speed by the time it takes to transmit one frame, including the overhead.

$$\begin{aligned} Time\ to\ transmit\ one\ frame &= \frac{frame\ size}{link\ speed} = \frac{256\ bytes}{1.5\ Mbps} \\ &= 0{,}001707\ seconds \end{aligned}$$

Therefore, the maximum rate of information transmission is:

$$Maximun\ rate = \frac{data\ rate}{efficiency}$$

The efficiency factor takes into account the protocol overhead, such as the header and trailer bits.

Assuming an efficiency factor of 80% (efficiency = 0.8), we get:

$$Maximun\ rate = \frac{(250\text{ bytes} \times 8\text{ bits/byte})/0.001707\text{ seconds}}{0.8} =$$
$$\text{25,154 bits per second}$$

Therefore, the maximum rate at which information can be transmitted over a 1.5 Mbps geostationary satellite link using HDLC with 250-byte frames is approximately 23,154 bits per second.

Câu 6: Suppose that a multiplexer receives constant-length packet from N = 60 data sources. Each data source has a probability p = 0.1 of having a packet in a given T-second period. Suppose that the multiplexer has one line in which it can transmit eight packets every T seconds. It also has a second line where it directs any packets that cannot be transmitted in the first line in a T-second period. Find the average number of packets that are transmitted on the first line and the average number of packets that are transmitted in the second line.

Given:

N = 60 data sources

Probability of having a packet in a given T-second period, p = 0.1

The multiplexer can transmit eight packets every T seconds

Any packets that cannot be transmitted in the first line are directed to the second line

To find:

Average number of packets that are transmitted on the first line

Average number of packets that are transmitted in the second line

We can model this scenario as a binomial distribution problem, where each data source has a probability p of transmitting a packet and there are N such sources.

The probability of k sources transmitting a packet is given by the **binomial distribution formula**:

$$P(k) = \binom{N}{k} p^k (1-p)^{N-k}$$

where $\binom{N}{k}$ is the binomial coefficient, given by:

$$\binom{N}{k} = \frac{N!}{k!\,(N-k)!}$$

We want to find the average number of packets that are transmitted on the first line and the second line. Let X be the total number of packets generated in a T-second period. Then, we can divide X into two parts: Y, the number of packets transmitted on the first line, and Z, the number of packets transmitted on the second line.

Since we can transmit up to eight packets on the first line, we have:

$$Y = \min{(X, 8)}$$

For the remaining packets, we send them on the second line, so we have:

$$Z = \max{(0, X - 8)}$$

The expected value of Y is given by:

$$E(Y) \;=\; \sum_{k=0}^{8} kP(k)$$

where P(k) is the probability of k sources transmitting a packet, as computed using the binomial distribution formula.

Similarly, the expected value of Z is given by:

$$E(Z) \;=\; \sum_{k=9}^{N} kP(k)$$

where P(k) is again the probability of k sources transmitting a packet.

Let's compute these values using the given parameters:

N = 60

p = 0.1

T = 1 second (since we are considering a T-second period)

M = 8 (maximum packets that can be transmitted on the first line)

First, let's calculate the probabilities of k sources transmitting a packet:

$$P(k) = \binom{N}{k} p^k (1 - p)^{N-k}$$

For k = 0 to 8:

P(0) = (60 choose 0) * 0.1^0 * 0.9^60 = 0.026

$$P(0) = \binom{60}{0} 0.1^{0}(1 - 0.1)^{60} = 0.026$$

$$P(1) = \binom{60}{1} 0.1^{1}(1 - 0.1)^{59} = 0.157$$

$$P(2) = \binom{60}{2} 0.1^{2}(1 - 0.1)^{58} = 0.318$$

$$P(3) = \binom{60}{3} 0.1^{3}(1 - 0.1)^{57} = 0.306$$

$$P(4) = \binom{60}{4} 0.1^{4}(1 - 0.1)^{56} = 0.185$$

$$P(5) = \binom{60}{5} 0.1^{5}(1 - 0.1)^{55} = 0.08$$

$$P(6) = \binom{60}{6} 0.1^{6}(1 - 0.1)^{54} = 0.027$$

$$P(7) = \binom{60}{7} 0.1^{7}(1 - 0.1)^{53} = 0.007$$

$$P(8) = \binom{60}{8} 0.1^{8}(1 - 0.1)^{52} = 0.001$$

For k = 9 to 60:

$$P(k) = \binom{N}{k} p^{k}(1 - p)^{N-k}$$

For simplicity, we can use the complement rule and subtract the sum of probabilities from 0 to 8 from 1:

$$\begin{aligned} P(k) = 1 - P(0 \leq X \leq 8) \\ &= 1 \\ &- \big(P(0) - P(1) - P(2) - P(3) - P(4) - P(5) - P(6) - P(7) \\ &- P(8)\big) = 0.996 \end{aligned}$$

Now, let's calculate the expected values of Y and Z:

$$E(Y) = \sum_{k=0}^{8} kP(k) = 00.026 + 10.157 + 20.318 + 30.306 + 40.185 + 50.080 + 6 * 0.027 + 7$$

Câu 7. Consider the transfer of a single real-time telephone voice signal across a packet network. Suppose that each voice sample should not be delayed by more than 20 ms.

a. Discuss which of the following adaptation functions are relevant to meeting the requirements of this transfer: handling of arbitrary message size; reliability and sequencing; pacing and flow control; timing; addressing; and privacy, integrity and authentication.

b. Compare a hop-by-hop approach to an end-to-end approach to meeting the requirements of the voice signal.

a. To meet the requirement of transferring a single real-time telephone voice signal across a packet network with a maximum delay of 20 ms, the following adaptation functions are relevant:

Timing: The timing adaptation function is critical in ensuring that each voice sample is delivered within the required deadline. The network must be able to synchronize its clock with the sender and receiver to maintain the required time intervals between packets.

Reliability and sequencing: To ensure that each voice sample is delivered without loss or misordering, the reliability and sequencing adaptation function is necessary. This requires the use of error detection and correction mechanisms, as well as sequencing and resequencing of packets at the receiver end.

Pacing and flow control: To prevent packet loss due to congestion, pacing and flow control mechanisms are necessary. These mechanisms regulate the rate at which packets are transmitted and received to match the capacity of the network.

Addressing: Addressing is necessary to identify the source and destination of each voice sample. It also enables routing of packets through the network.

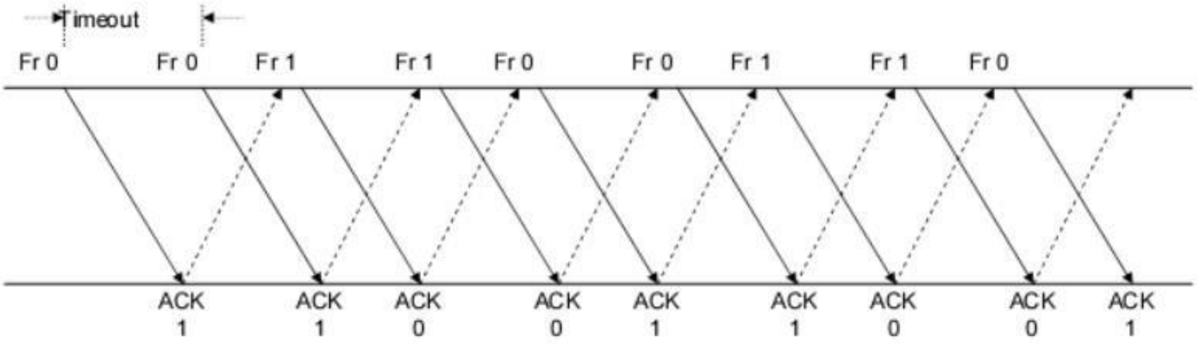
b. There are two approaches for meeting the requirements of a real-time telephone voice signal over a packet network: the hop-by-hop approach and the end-to-end approach.

The hop-by-hop approach involves implementing the required adaptation functions at each intermediate node in the packet network. Each node processes the packets it receives before forwarding them to the next node. This approach can introduce additional delays and overhead due to processing at each node. Furthermore, if a node fails, the entire communication may become compromised.

The end-to-end approach involves implementing the required adaptation functions only at the endpoints of the communication path, i.e., the sender and receiver of the voice signal. The packets are transmitted through the network without modification, and any required processing is performed at the endpoints. This approach minimizes delays and overhead, but it may not be suitable for networks with high packet loss rates or variable delays.

In general, the end-to-end approach is preferred for real-time voice communications over packet networks because it minimizes delays and overhead. However, the hop-by-hop approach may be necessary in some situations, such as when the network has high delay or loss rates, or when additional processing is necessary at intermediate nodes.

Câu 8. Consider the Stop-and-Wait protocol as described. Suppose that the protocol is modified so that each time a frame is found in error at either the sender or receiver, the last transmitted frame is immediately resent.



a. Show that the protocol still operates correctly.

b. Does the state transition diagram need to be modified to describe the new operation?

c. What is the main effect of introducing the immediate-retransmission feature?

a. The modified Stop-and-Wait protocol still operates correctly because it ensures that every frame is received correctly before the next one is sent. If a frame is found in error, the sender immediately resends the last transmitted frame, which guarantees that the receiver will receive a correct copy of the frame.

b. The state transition diagram would need to be modified to reflect the new operation. Specifically, a new transition would need to be added from the "Frame Received, ACK/NAK Lost" state back to the "Frame Sent" state, indicating that the sender should immediately resend the last transmitted frame in response to the error.

c. The main effect of introducing the immediate-retransmission feature is to improve the protocol's error recovery capabilities. With this feature, errors can be quickly corrected by resending the last transmitted frame. This reduces the time required for error recovery and increases the overall efficiency of the protocol. However, it also introduces additional network traffic, which could potentially increase congestion and delay.

Câu 9. Suppose that two peer-to-peer processes provide a service that involves the transfer of discrete messages. Suppose that the peer processes are allowed to exchange PDUs that have a maximum size of M bytes including H bytes of header. Suppose that a PDU is not allowed to carry information from more than one message.

a. Develop an approach that allows the peer processes to exchange messages of arbitrary size.

b. What essential control information needs to be exchanged between the peer processes?

c. Now suppose that the message transfer service provided by the peer processes is shared by several message source-destination pairs. Is additional control information required, and if so, where should it be placed?

a. To allow for the exchange of messages of arbitrary size within the given constraints, the peer processes can use a technique known as segmentation and reassembly. This involves dividing a message into smaller segments, each of which can fit within a single PDU, and then sending these segments over multiple PDUs. The receiver can then reassemble the segments back into the original message.

b. The essential control information that needs to be exchanged between the peer processes includes:

Sequence numbers: These are used to ensure that all segments are received in the correct order and that no segments are missing or duplicated.

Acknowledgment numbers: These are used to confirm that a segment has been successfully received by the receiver.

Window sizes: These are used to allow the sender to adjust the number of unacknowledged segments it can send at any given time based on how much space is available in the receiver's buffer.

c. If the message transfer service is shared by several source-destination pairs, additional control information may be required to differentiate between the different messages being sent. This information could be placed in the header of each PDU and could include the source and destination addresses, session identifiers, or any other information needed to identify the specific message being sent. Additionally, the control information used to manage the flow of PDUs between the sender and receiver may also need to be adjusted to account for multiple concurrent connections. For example, each connection may require its own sequence

and acknowledgment numbers to ensure that segments are properly tracked and acknowledged for each individual message.

Câu 10. A 1 Mbyte file is to be transmitted over a 1 Mbps communication line that has a bit error rate of p = 10^6.

a. What is the probability that the entire file is transmitted without errors? Note for n large and p very small, (1 − p)n ≈ e-np.

b. The file is broken up into N equal-sized blocks that are transmitted separately. What is the probability that all the blocks arrive correctly without error? Does dividing the file into blocks help?

c. Suppose the propagation delay is negligible, explain how Stop-and-Wait ARQ can help deliver the file in error-free form. On the average how long does it take to deliver the file if the ARQ transmits the entire file each time?

a. The probability that the entire 1 Mbyte file is transmitted without errors can be calculated as follows: $P = (1 - p)^n$

where p is the bit error rate and n is the number of bits in the file.

Since the file size is 1 Mbyte, which is equal to 8 million bits, we can calculate the probability as:

$$P \ = (1 - 10^{-6})^{8000000}$$

$$P \approx \ e^{-8}$$

Therefore, the probability that the entire file is transmitted without errors is approximately e^{-8}, which is a very small probability.

b. If the file is broken up into N equal-sized blocks and transmitted separately, the probability that all blocks arrive correctly without error can be calculated as:

$$P \ = (1 - p)^{n \times N}$$

where p is the bit error rate, n is the number of bits in each block, and N is the total number of blocks.

Dividing the file into blocks does help because if an error occurs in one block, only that block needs to be retransmitted instead of the entire file.

Assuming each block is equally sized at 1/N Mbytes or 8/N million bits, the probability can be calculated as:

$P \ = (1 - 10^{-6})^{\frac{8}{N} \times N} = (1 - 10^{-6})^{8}$= 0.99992

Therefore, the probability that all blocks arrive correctly without error is approximately 0.999992.

c. Stop-and-Wait ARQ (Automatic Repeat Request) can help deliver the file in error-free form by ensuring that each block is successfully received before transmitting the next block. In this protocol, the sender transmits one block at a time and waits for an acknowledgment from the receiver before transmitting the next block.

Assuming the propagation delay is negligible, the time required to deliver the file using Stop-and-Wait ARQ can be calculated as follows:

Time required to transmit one block = n/p, where n is the number of bits in each block and p is the bit rate of the communication line.

$$\begin{aligned}&\text{Time required to receive an acknowledgment for one block}\\ &= 2 \times \text{ propagation delay}\end{aligned}$$

$$\begin{aligned}&\text{Total time required to transmit and receive one block}\\ &= \text{ n/p } + \text{ } 2 \times \text{ propagation delay}\end{aligned}$$

Since there are N blocks to be transmitted, the total time required to deliver the file would be N times the time required to transmit and receive one block:

$$Total\ time = N(\frac{n}{p} + 2 \times propagation\ delay)$$

Assuming a negligible propagation delay, the total time required to deliver the entire 1 Mbyte file would be:

$$Total\ time = \frac{8000000}{10^6} + 2 \times 0 = 8\ seconds$$

However, this assumes no errors occur during transmission. If errors occur, additional time will be required for retransmission until all blocks are received correctly.

8000000