R1. What is the difference between a host and an end system? List several different types of end systems. Is a Web server an end system?

Host vs. End System:

- A **host** is any device connected to a computer network. This includes all computers, servers, smartphones, tablets, and other devices that participate in a network.
- An **end system** is a type of host that sits at the edge of the network and interacts directly with users. It is the endpoint in a communication network, where data is either generated or consumed.

Types of End Systems:

- Personal computers (PCs)
- Laptops
- Smartphones
- Tablets
- Servers
- Smart TVs
- IoT devices (e.g., smart thermostats, smart speakers)

Web Server as an End System:

- Yes, a Web server is considered an end system. It interacts directly with users (clients) by serving web pages, handling requests, and providing data.

R2. The word protocol is often used to describe diplomatic relations. How does Wikipedia describe diplomatic protocol?

Diplomatic protocol refers to the set of international courtesy rules. These rules govern the conduct and procedures that are followed in the diplomatic interactions and ceremonies. It includes the guidelines for behavior, communication, and the formalities observed during diplomatic events and negotiations to ensure mutual respect and smooth interaction between diplomats and representatives of different countries.

R3. Why are standards important for protocols?

Standards are crucial for protocols because they ensure:

- **Interoperability:** Devices and systems from different manufacturers can communicate and work together effectively.
- **Reliability:** Standards provide a consistent and predictable framework, which enhances the reliability of communication.
- **Scalability:** Standards enable systems to grow and integrate new technologies without compatibility issues.
- **Efficiency:** Well-defined standards optimize resource use and streamline processes, reducing errors and redundancies.
- **Security:** Standards incorporate security measures that protect data and communication integrity.

R4. List four access technologies. Classify each one as home access, enterprise access, or wide-area wireless access.

1. **Digital Subscriber Line (DSL):** Home access
2. **Cable Internet (HFC):** Home access
3. **Ethernet:** Enterprise access
4. **4G/5G Mobile Networks:** Wide-area wireless access

R5. Is HFC transmission rate dedicated or shared among users? Are collisions possible in a downstream HFC channel? Why or why not?

- **HFC Transmission Rate:** Shared among users.
- **Collisions in Downstream HFC Channel:** No, collisions are not possible in a downstream HFC channel. This is because the data flow is managed by the cable modem termination system (CMTS) at the headend, which ensures that data is sent downstream in an orderly and controlled manner, avoiding collisions.

R6. List the available residential access technologies in your city. For each type of access, provide the advertised downstream rate, upstream rate, and monthly price.

(Example for a generic city, as specific details can vary)

1. **DSL:**
    - Downstream rate: 10-100 Mbps
    - Upstream rate: 1-10 Mbps
    - Monthly price: $30-$70
2. **Cable Internet (HFC):**
    - Downstream rate: 100-1000 Mbps
    - Upstream rate: 10-50 Mbps
    - Monthly price: $40-$100
3. **Fiber to the Home (FTTH):**
    - Downstream rate: 100-1000 Mbps
    - Upstream rate: 100-1000 Mbps
    - Monthly price: $50-$120
4. **4G/5G Mobile Networks:**
    - Downstream rate: 10-1000 Mbps
    - Upstream rate: 5-100 Mbps
    - Monthly price: $50-$90 (data plans)

R7. What is the transmission rate of Ethernet LANs?

The transmission rate of Ethernet LANs can vary depending on the specific Ethernet standard:

- **10 Mbps**: Ethernet (10BASE-T)
- **100 Mbps**: Fast Ethernet (100BASE-TX)
- **1 Gbps**: Gigabit Ethernet (1000BASE-T)
- **10 Gbps**: 10 Gigabit Ethernet (10GBASE-T)
- **40 Gbps and 100 Gbps**: Available for higher performance needs (40GBASE and 100GBASE standards)

R8. What are some of the physical media that Ethernet can run over?

- **Twisted pair copper wires (UTP)**
- **Fiber optic cables**
- **Coaxial cables**
- **Wireless (Wi-Fi) for certain standards (though traditionally Ethernet refers to wired connections)**

R9. HFC, DSL, and FTTH are all used for residential access. For each of these access technologies, provide a range of transmission rates and comment on whether the transmission rate is shared or dedicated.

- **HFC (Hybrid Fiber-Coaxial):**
  - Transmission Rates: 100 Mbps to 1 Gbps downstream, 10 Mbps to 50 Mbps upstream.
  - Shared among users in the same neighborhood, leading to potential congestion during peak times.
- **DSL (Digital Subscriber Line):**
  - Transmission Rates: 1 Mbps to 100 Mbps downstream, 0.5 Mbps to 10 Mbps upstream.
  - Dedicated line to each subscriber, but performance can degrade with distance from the central office.
- **FTTH (Fiber to the Home):**
  - Transmission Rates: 100 Mbps to 1 Gbps (and higher) downstream and upstream.
  - Can be either dedicated or shared, depending on the service provider's infrastructure. Generally offers higher performance and reliability compared to HFC and DSL.

R10. Describe the most popular wireless Internet access technologies today. Compare and contrast them.

- **Wi-Fi (Wireless Fidelity):**
  - Commonly used for local area networking (LAN) in homes, offices, and public spaces.
  - Offers high speeds (up to several Gbps with Wi-Fi 6) and low latency.

    - Limited by range and interference from other devices and physical obstructions.
- **4G LTE (Fourth Generation Long-Term Evolution):**
    - Widely used for mobile internet access.
    - Provides high-speed data (up to 100 Mbps), suitable for streaming, browsing, and other mobile applications.
    - Coverage is extensive, but speed can vary based on network congestion and signal strength.
- **5G (Fifth Generation):**
    - The latest generation of mobile networks.
    - Promises significantly higher speeds (up to 10 Gbps), lower latency, and greater capacity compared to 4G.
    - Still in the process of global rollout, with varying availability.
    - Requires new infrastructure and devices compatible with 5G.
- **Satellite Internet:**
    - Used in remote and rural areas where other forms of internet access are unavailable.
    - Provides coverage virtually anywhere.
    - Speeds can range from 25 Mbps to over 100 Mbps with newer technologies like SpaceX's Starlink.
    - Higher latency compared to terrestrial technologies due to the distance signals must travel to and from satellites.

R11. Suppose there is exactly one packet switch between a sending host and a receiving host. The transmission rates between the sending host and the switch and between the switch and the receiving host are R1 and R2, respectively. Assuming that the switch uses store-and-forward packet switching, what is the total end-to-end delay to send a packet of length L? (Ignore queuing, propagation delay, and processing delay.)

To find the total end-to-end delay, we need to consider the time it takes to transmit the packet from the sending host to the switch, the time it takes for the switch to store and forward the packet, and the time it takes to transmit the packet from the switch to the receiving host.

- Transmission delay from sending host to switch:
- Transmission delay from switch to receiving host:

Since the switch must receive the entire packet before it can start forwarding it, we add these two delays:
Total end-to-end delay= +

R12. What advantage does a circuit-switched network have over a packet-switched network? What advantages does TDM have over FDM in a circuit-switched network?

Advantages of Circuit-Switched Network:

- **Dedicated Path:** Provides a dedicated communication path between two endpoints, ensuring a constant and guaranteed bandwidth.
- **Predictable Performance:** Once a circuit is established, the performance is predictable, with no delays due to other traffic.
- **No Congestion:** As the path is reserved for the duration of the connection, there is no risk of congestion from other users' data.

Advantages of TDM over FDM in a Circuit-Switched Network:

- **Flexibility:** TDM can dynamically allocate time slots based on demand, whereas FDM is fixed and less flexible.
- **Simpler Implementation:** TDM does not require complex filtering to separate channels, unlike FDM, which requires precise frequency management.
- **Efficiency:** TDM can be more efficient in environments with bursty traffic patterns, as time slots can be reassigned as needed.

R13. Suppose users share a 2 Mbps link. Also suppose each user transmits continuously at 1 Mbps when transmitting, but each user transmits only 20 percent of the time.

a. When circuit switching is used, how many users can be supported?

In circuit switching, each user requires a dedicated 1 Mbps channel when transmitting.

Number of users=

=

=2

So, 2 users can be supported.

b. For the remainder of this problem, suppose packet switching is used. Why will there be essentially no queuing delay before the link if two or fewer users transmit at the same time? Why will there be a queuing delay if three users transmit at the same time?

With packet switching, the 2 Mbps link is shared among all users. If two or fewer users transmit simultaneously, the total transmission rate does not exceed the link capacity (2 Mbps), hence no queuing delay.

If three users transmit at the same time, the combined rate (3 x 1 Mbps = 3 Mbps) exceeds the link capacity, causing packets to queue and wait for transmission.

c. Find the probability that a given user is transmitting.

Each user transmits 20% of the time.

P(user transmitting)=0.2

d. Suppose now there are three users. Find the probability that at any given time, all three users are transmitting simultaneously. Find the fraction of time during which the queue grows.

The probability that all three users are transmitting simultaneously is the product of their individual probabilities:
$P(\text{all three transmitting})=0.2\times0.2\times0.2=0.008$
The fraction of time during which the queue grows is 0.8%.

R14. Why will two ISPs at the same level of the hierarchy often peer with each other? How does an IXP earn money?

Why ISPs Peer:

- **Reduce Costs:** By peering, ISPs can exchange traffic directly without paying a third-party transit provider, reducing costs.
- **Improve Performance:** Peering can reduce latency and improve speed for end-users by providing more direct routing paths.
- **Increase Redundancy:** Peering adds redundancy to network paths, enhancing reliability.

IXP Earnings:

- **Port Fees:** Charging ISPs and other networks for connecting to the IXP.
- **Cross-Connect Fees:** Fees for the physical cables connecting the IXP's ports to the ISPs' equipment.
- **Service Fees:** Offering additional services such as managed peering, data analytics, and security services.

R15. Some content providers have created their own networks. Describe Google's network. What motivates content providers to create these networks?

Google's Network:

- **Global Backbone:** Google operates a global network backbone, with data centers and network infrastructure across continents.
- **Private Fiber:** Google invests in private fiber optic cables, including undersea cables, to control its data routing and bandwidth.

- **Edge Caching:** Deploys edge servers close to end-users to cache content and reduce latency.

Motivations for Creating Networks:

- **Performance:** Ensuring high-speed, low-latency access to their services and content.
- **Control:** Greater control over data routing, security, and redundancy.
- **Cost Savings:** Reducing dependency on ISPs and transit providers, potentially lowering costs.
- **User Experience:** Improving the end-user experience by reducing buffering and load times.

R16. Consider sending a packet from a source host to a destination host over a fixed route. List the delay components in the end-to-end delay. Which of these delays are constant and which are variable?

Delay Components:

1. **Processing Delay:** Time to process the packet header at each router. (Variable)
2. **Queuing Delay:** Time the packet spends in routing queues. (Variable)
3. **Transmission Delay:** Time to push the packet's bits onto the link. (Constant)
4. **Propagation Delay:** Time for the packet to travel through the physical medium. (Constant)

R17. Visit the Transmission Versus Propagation Delay interactive animation at the companion Web site. Among the rates, propagation delay, and packet sizes available, find a combination for which the sender finishes transmitting before the first bit of the packet reaches the receiver. Find another combination for which the first bit of the packet reaches the receiver before the sender finishes transmitting.

(Since I cannot visit the website, here's the theoretical explanation.)

- **Sender finishes transmitting before the first bit reaches the receiver:**
  - High transmission rate and long propagation distance.
  - Example: Transmission rate = 1 Gbps, distance = 5000 km, propagation speed = 2.5×10^8m/s, packet size = 10 KB.
- **First bit reaches the receiver before the sender finishes transmitting:**
  - Low transmission rate and short propagation distance.
  - Example: Transmission rate = 1 Mbps, distance = 1 km, propagation speed = 2.5×10^8m/s, packet size = 1 KB.

R18. How long does it take a packet of length 1,000 bytes to propagate over a link of distance 2,500 km, propagation speed 2.5×1082.5×108 m/s, and transmission rate 2 Mbps? More generally, how long does it take a packet of length L to propagate over a link of distance d, propagation speed s, and transmission rate R bps? Does this delay depend on packet length? Does this delay depend on transmission rate?

- **Propagation Delay:**
  Propagation Delay == = 0.01s = 10ms
- **General Formula:**
  Propagation Delay =
  This delay depends only on the distance *d* and the propagation speed *s*, not on the packet length *L* or the transmission rate *R*.

R19. Suppose Host A wants to send a large file to Host B. The path from Host A to Host B has three links, of rates R_1=500 kbps, R_2=2 Mbps, and R_3=1 Mbps.

a. **Assuming no other traffic in the network, what is the throughput for the file transfer?**

The throughput is determined by the slowest link in the path. Throughput=min(R1,R2,R3)=min(500 kbps,2 Mbps,1 Mbps)=500 kbps

b. **Suppose the file is 4 million bytes. Dividing the file size by the throughput, roughly how long will it take to transfer the file to Host B?**

First, convert the file size to bits:
4 million bytes=$4\times10^6\times8$ bits=32×10^6 bits
Time to transfer:
Time = = = 54seconds

c. **Repeat (a) and (b), but now with R_2 reduced to 100 kbps.**

- **New throughput:** Throughput=mib(R_1,R_2,R_3)=min(500 kbps,100 kbps,1 Mbps)= 100 kbps
- **New transfer time:** Time = (32×10^6 bits) / (100×10^3 bits/s) = 320 seconds

R20. Suppose end system A wants to send a large file to end system B. At a very high level, describe how end system A creates packets from the file. When one of these packets arrives at a router, what information in the packet does the router use to determine the link onto which the packet is forwarded? Why is packet switching in the Internet analogous to driving from one city to another and asking directions along the way?

Creating Packets:

- End system A divides the large file into smaller chunks, encapsulating each chunk in a packet.
- Each packet is given a header containing necessary information, including source and destination addresses.

Routing Decision:

- When a packet arrives at a router, the router examines the destination IP address in the packet header.
- The router uses its forwarding table to determine the next hop for the packet based on the destination address.

Analogy to Driving:

- Packet switching is like driving from one city to another and asking for directions at each intersection. Each router along the way directs the packet to the next hop based on current routing information, similar to how a driver receives guidance at each decision point on the road.

R21. Visit the Queuing and Loss interactive animation at the companion Web site. What is the maximum emission rate and the minimum transmission rate? With those rates, what is the traffic intensity? Run the interactive animation with these rates and determine how long it takes for packet loss to occur. Then repeat the experiment a second time and determine again how long it takes for packet loss to occur. Are the values different? Why or why not?

Here is a theoretical explanation.

Maximum Emission Rate: The highest rate at which packets are sent. **Minimum Transmission Rate:** The lowest rate at which packets are processed and sent out.

Traffic Intensity:

Traffic Intensity=Emission RateTransmission RateTraffic Intensi ty=Transmission RateEmission Rate

When traffic intensity exceeds 1, the queue will start to grow, leading to packet loss.

Running the Experiment:

- Determine the emission and transmission rates.
- Calculate the traffic intensity.

- Run the animation and observe the time to packet loss.

Repeating the Experiment:

- Due to the stochastic nature of traffic and random variations, the time to packet loss may vary slightly between runs.
- Differences occur due to the probabilistic behavior of packet arrivals and processing times in the simulation.

R22. List five tasks that a layer can perform. Is it possible that one (or more) of these tasks could be performed by two (or more) layers?

Tasks a Layer Can Perform:

1. **Error Detection and Correction:** Identifying and correcting errors in data transmission.
2. **Flow Control:** Managing the rate of data transmission between two nodes to prevent a fast sender from overwhelming a slow receiver.
3. **Segmentation and Reassembly:** Breaking down large data units into smaller segments for transmission and reassembling them at the destination.
4. **Multiplexing and Demultiplexing:** Combining multiple data streams into one stream for transmission and separating them at the destination.
5. **Addressing and Routing:** Assigning addresses to data packets and determining the best path for them to travel through the network.

Overlap Between Layers: Yes, it is possible for tasks to be performed by multiple layers. For example, both the Transport layer and the Network layer can perform error detection and correction. The Transport layer ensures reliable delivery of data between two hosts, while the Network layer ensures data integrity during routing.

R23. What are the five layers in the Internet protocol stack? What are the principal responsibilities of each of these layers?

1. **Application Layer:**
   - **Responsibility:** Provides network services to end-user applications.
   - **Examples:** HTTP, FTP, SMTP.
   - **Function:** Facilitates communication between software applications and the underlying network.
2. **Transport Layer:**
   - **Responsibility:** Ensures end-to-end communication between devices.
   - **Examples:** TCP, UDP.
   - **Function:** Provides reliable data transfer, error detection, and flow control (TCP); provides connectionless data transfer (UDP).
3. **Network Layer:**
   - **Responsibility:** Handles packet forwarding including routing through intermediate routers.
   - **Examples:** IP.
   - **Function:** Determines the best path for data packets to travel from source to destination.
4. **Link Layer:**
   - **Responsibility:** Manages data transfer between adjacent network nodes.
   - **Examples:** Ethernet, Wi-Fi.
   - **Function:** Provides framing, error detection, and MAC addressing.
5. **Physical Layer:**
   - **Responsibility:** Transmits raw bitstreams over a physical medium.
   - **Examples:** Cables, radio frequencies.
   - **Function:** Converts data into electrical, optical, or radio signals for transmission.

R24. What is an application-layer message? A transport-layer segment? A network-layer datagram? A link-layer frame?

- **Application-Layer Message:** The data generated by applications, such as emails, files, or web pages, which are sent to the transport layer for transmission.
- **Transport-Layer Segment:** The data unit created by the transport layer, which encapsulates the application-layer message with transport layer headers (e.g., TCP or UDP headers).
- **Network-Layer Datagram:** The data unit created by the network layer, which encapsulates the transport-layer segment with network layer headers (e.g., IP headers).
- **Link-Layer Frame:** The data unit created by the link layer, which encapsulates the network-layer datagram with link layer headers and trailers (e.g., Ethernet frames).

R25. Which layers in the Internet protocol stack does a router process? Which layers does a link-layer switch process? Which layers does a host process?

- **Router:**
  - **Layers Processed:** Physical layer, Link layer, and Network layer. Routers determine the best path for forwarding packets based on the network layer information.
- **Link-Layer Switch:**
  - **Layers Processed:** Physical layer and Link layer. Switches use MAC addresses to forward frames within a local area network (LAN).
- **Host:**
  - **Layers Processed:** All layers (Physical, Link, Network, Transport, and Application). Hosts generate and consume data and require full stack processing for communication.

R26. What is self-replicating malware?

Self-Replicating Malware: Malware that can make copies of itself and spread to other systems without user intervention. Examples include viruses and worms. Viruses attach themselves to host files, while worms exploit vulnerabilities to spread across networks.

R27. Describe how a botnet can be created and how it can be used for a DDoS attack.

Creating a Botnet:

1. **Infection:** Malware infects multiple computers (bots) through various methods such as phishing emails, malicious downloads, or exploiting vulnerabilities.
2. **Control:** The infected bots are connected to a command-and-control (C&C) server, allowing the attacker to manage and coordinate the bots remotely.
3. **Recruitment:** The botnet grows as more devices are infected and added to the network.

Using a Botnet for a DDoS Attack:

1. **Command:** The attacker instructs the bots to send a massive volume of requests or data packets to a target server or network.
2. **Overload:** The target's resources are overwhelmed by the sheer volume of incoming traffic, causing legitimate requests to be delayed or denied.
3. **Disruption:** The target experiences service disruption or complete downtime, achieving the attacker's goal of denial of service.

R28. Suppose Alice and Bob are sending packets to each other over a computer network. Suppose Trudy positions herself in the network so that she can capture all the packets sent by Alice and send whatever she wants to Bob; she can also capture all the packets sent by Bob and send whatever she wants to Alice. List some of the malicious things Trudy can do from this position.

Malicious Activities Trudy Can Perform:

1. **Eavesdropping:** Trudy can intercept and read all the data being sent between Alice and Bob, gaining access to sensitive information.
2. **Data Modification:** She can alter the contents of the packets, potentially changing the meaning of the messages or corrupting data.
3. **Impersonation:** By sending packets with forged addresses, Trudy can pretend to be either Alice or Bob, deceiving the other party.
4. **Replay Attacks:** She can capture packets and resend them later, causing confusion or unauthorized actions.
5. **Man-in-the-Middle Attack:** Trudy can relay messages between Alice and Bob, making them believe they are communicating directly while she intercepts and potentially alters the communication.
6. **Denial of Service:** By overwhelming either Alice or Bob with a flood of packets, Trudy can disrupt their communication.

P1. Design and describe an application-level protocol to be used between an automatic teller machine and a bank's centralized computer. Your protocol should allow a user's card and password to be verified, the account balance (which is maintained at the centralized computer) to be queried, and an account withdrawal to be made (that is, money disbursed to the user). Your protocol entities should be able to handle the all-too-common case in which there is not enough money in the account to cover the withdrawal. Specify your protocol by listing the messages exchanged and the action taken by the automatic teller machine or the bank's centralized computer on transmission and receipt of messages. Sketch the operation of your protocol for the case of a simple withdrawal with no errors, using a diagram similar to that in Figure 1.2. Explicitly state the assumptions made by your protocol about the underlying end-to-end transport service.

Assumptions:

- Reliable and secure end-to-end transport service (e.g., TCP/IP).
- The ATM and the bank's centralized computer communicate over an encrypted channel to ensure data security.
- The ATM and the bank's centralized computer can handle a predefined set of commands and responses.

Messages Exchanged:

1. **Card Verification:**
    - **ATM to Bank:**
        - Message: **VERIFY_CARD**
        - Content: **Card_Number**
    - **Bank to ATM:**
        - Message: **CARD_VERIFIED** or **CARD_INVALID**
        - Content: **Verification_Status**

2. **Password Verification:**
    - **ATM to Bank:**
        - Message: **VERIFY_PASSWORD**
        - Content: **Card_Number, Password**
    - **Bank to ATM:**
        - Message: **PASSWORD_VERIFIED** or **PASSWORD_INVALID**
        - Content: **Verification_Status**

3. **Balance Query:**
    - **ATM to Bank:**
        - Message: **QUERY_BALANCE**
        - Content: **Card_Number**
    - **Bank to ATM:**
        - Message: **BALANCE**
        - Content: **Account_Balance**

4. **Withdrawal Request:**
    - **ATM to Bank:**
        - Message: **REQUEST_WITHDRAWAL**
        - Content: **Card_Number, Amount**
    - **Bank to ATM:**
        - Message: **WITHDRAWAL_APPROVED** or **INSUFFICIENT_FUNDS**
        - Content: **Approval_Status, Dispensed_Amount**

Actions Taken:

- **ATM on Receiving CARD_VERIFIED:**
    - Prompt user to enter the password.
- **ATM on Receiving CARD_INVALID:**
    - Display error message and eject card.
- **ATM on Receiving PASSWORD_VERIFIED:**
    - Prompt user for action (e.g., balance inquiry, withdrawal).
- **ATM on Receiving PASSWORD_INVALID:**
    - Display error message and allow another attempt.
- **ATM on Receiving BALANCE:**
    - Display the account balance to the user.
- **ATM on Receiving WITHDRAWAL_APPROVED:**
    - Dispense the specified amount of cash.
- **ATM on Receiving INSUFFICIENT_FUNDS:**
    - Display error message indicating insufficient funds.

Operation of the Protocol (Simple Withdrawal, No Errors):

1. User inserts card into ATM.
2. ATM sends **VERIFY_CARD(Card_Number)** to Bank.
3. Bank responds with **CARD_VERIFIED**.
4. ATM prompts user to enter password.
5. User enters password.
6. ATM sends **VERIFY_PASSWORD(Card_Number, Password)** to Bank.
7. Bank responds with **PASSWORD_VERIFIED**.
8. ATM prompts user to choose action.

9. User selects withdrawal and enters amount.
10. ATM sends **REQUEST_WITHDRAWAL(Card_Number, Amount)** to Bank.
11. Bank responds with **WITHDRAWAL_APPROVED**.
12. ATM dispenses cash and displays the transaction summary.

```
User                          ATM                        Bank
 |                      Insert Card                          |
 |  ---------------------------------------------------------->|
 |                                                           |
 |                   VERIFY_CARD(Card_Number)
      |
 |  ---------------------------------------------------------->|
 |                                                           |
 |                    CARD_VERIFIED
      |
 |<----------------------------------------------------------|
 |                      Enter Password                       |
 |---------------------------------------------------------->|
 |                                                           |
 |          VERIFY_PASSWORD(Card_Number, Password)  |
 |  ---------------------------------------------------------->|
 |                                                           |
 |                    PASSWORD_VERIFIED
      |
 |<----------------------------------------------------------|
 |                Select Withdrawal and Enter Amount         |
 |---------------------------------------------------------->|
 |                                                           |
 |     REQUEST_WITHDRAWAL(Card_Number, Amount) -
|
 |  -------------------------------------------------------- -->|
 |
|
```

```
|                    WITHDRAWAL_APPROVED
|
|<------------------------------------------------------------------------ -|
|                    Dispense Cash
|
|------------------------------------------------------------------------> |
|                                                                          |
```

P2. Equation 1.1 gives a formula for the end-to-end delay of sending one packet of length L over N links of transmission rate R. Generalize this formula for sending P such packets back-to-back over the N links.

Given Formula:

Total Delay=$N\cdot(d_{proc}+d_{trans}+d_{prop}+d_{queue})$ where $d_{trans}=R/L$.

For P packets sent back-to-back over N links, the generalized formula for end-to-end delay is:

Total Delay=$P\cdot d_{trans}+N\cdot(d_{proc}+d_{prop}+d_{queue})$ where:

- $d_{trans}=RL$
- d_{proc} is the processing delay per packet.
- d_{prop} is the propagation delay per link.
- d_{queue} is the queuing delay per packet per link.

Explanation:

- The first packet incurs the full delay through each of the$N$ links.
- Each subsequent packet incurs the transmission delay d_{trans} back-to-back without additional propagation or queuing delays once in the pipeline.

P3. Consider an application that transmits data at a steady rate (for example, the sender generates an N-bit unit of data every k time units, where k is small and fixed). Also, when such an application starts, it will continue running for a relatively long period of time. Answer the following questions, briefly justifying your answer: a. Would a packet-switched network or a circuit-switched network be more appropriate for this application? Why? b. Suppose

that a packet-switched network is used and the only traffic in this network comes from such applications as described above. Furthermore, assume that the sum of the application data rates is less than the capacities of each and every link. Is some form of congestion control needed? Why?

a. Packet-Switched Network vs. Circuit-Switched Network:

- **Circuit-Switched Network:**
  - More appropriate because it can provide a dedicated path with guaranteed bandwidth for the duration of the session.
  - This ensures a steady transmission rate without interruptions.
- **Justification:**
  - Circuit switching eliminates the overhead of packet headers and the possibility of packet loss or delay due to network congestion.

b. Need for Congestion Control in Packet-Switched Network:

- **Scenario:**
  - The only traffic in the network is from applications with a steady transmission rate.
  - The sum of application data rates is less than the capacity of each link.
- **Congestion Control:**
  - Not needed in this specific scenario because:
    - The steady transmission rates ensure predictable load.
    - The sum of the data rates being less than the link capacities implies no bottlenecks, thus no congestion.
- **Justification:**
  - Congestion control mechanisms are typically required to handle variable and bursty traffic patterns that can overload network resources. In this

controlled scenario, the traffic is predictable and within the network's capacity.

P4. Consider the circuit-switched network in Figure 1.13. Recall that there are four circuits on each link. Label the four switches A, B, C, and D, going in the clockwise direction. a. What is the maximum number of simultaneous connections that can be in progress at any one time in this network? b. Suppose that all connections are between switches A and C. What is the maximum number of simultaneous connections that can be in progress? c. Suppose we want to make four connections between switches A and C, and another four connections between switches B and D. Can we route these calls through the four links to accommodate all eight connections?

Network Diagram:

```
A------B
|      |
D–-----C
```

a. Maximum Simultaneous Connections:

- Each link has 4 circuits.
- Total links: 4 (AB, BC, CD, DA).
- Maximum simultaneous connections: 4×4=16.

b. Connections Between A and C:

- Possible routes: A-B-C and A-D-C.
- Each route uses 2 links.
- Each link has 4 circuits.
- Maximum connections per route: 4.
- Maximum connections between A and C: 4+4=8.

c. Routing 4 Connections Each for A-C and B-D:

- A-C connections can use routes A-B-C and A-D-C.
- B-D connections can use routes B-C-D and B-A-D.

- Each route uses different links, ensuring no overlap.
- Total connections: 4 (A-C via A-B-C) + 4 (A-C via A-D-C) + 4 (B-D via B-C-D) + 4 (B-D via B-A-D) = 16.
- All 8 connections can be accommodated.

P5. Review the car-caravan analogy in Section 1.4. Assume a propagation speed of 100 km/hour. a. Suppose the caravan travels 175 km, beginning in front of one tollbooth, passing through a second tollbooth, and finishing just after a third tollbooth. What is the end-to-end delay? b. Repeat (a), now assuming that there are eight cars in the caravan instead of ten.

Given:

- Propagation speed: 100 km/hour100km/hour

a. Caravan with Ten Cars:

- Distance traveled by caravan: 175 km175km
- End-to-end delay:
  - Propagation delay: 175 km100 km/hour=1.75 hours100km/hour175km=1.75 hours

b. Caravan with Eight Cars:

- End-to-end delay:
  - Propagation delay: 175 km100 km/hour=1.75 hours100km/hour175km=1.75 hours

P6. This elementary problem begins to explore propagation delay and transmission delay, two central concepts in data networking. Consider two hosts, A and B, connected by a single link of rate R bps. Suppose that the two hosts are separated by m meters, and suppose the propagation speed along the link is s meters/sec. Host A is to send a packet of size L bits to Host B. a. Express the propagation delay, d_{prop}, in terms of m and s. b. Determine the transmission time of the packet, d_{trans}, in terms of L and R. c. Ignoring processing and queuing delays, obtain an

expression for the end-to-end delay. d. Suppose Host A begins to transmit the packet at time t = 0. At time t = d_{trans}, where is the last bit of the packet? e. Suppose d_{prop} is greater than d_{trans}. At time t = d_{trans}, where is the first bit of the packet? f. Suppose d_{prop} is less than d_{trans}. At time t = d_{trans}, where is the first bit of the packet? g. Suppose s = 2.5 *(10^8) , L = 1500 bytes, and R = 10 Mbps. Find the distance m so that d_{prop} equals d_{trans}.

Given:

- Hosts A and B separated by *m* meters
- Propagation speed: *s* meters/sec
- Rate of transmission: *R* bps
- Packet size: *L* bits

a. Propagation Delay (d_{prop}): $d_{prop}=m/s$

b. Transmission Time (d_{trans}): $d_{trans}=L/R$

c. End-to-End Delay: End-to-End Delay=$d_{prop}+d_{trans}$

d. Last Bit of Packet at $t=d_{trans}$:

- The last bit of the packet is transmitted by host A at time $t=d_{trans}$.
- It will take additional time d_{prop} to reach host B.

e. First Bit of Packet at $t=d_{trans}$ (if $d_{prop}>d_{trans}$):

- The first bit of the packet is transmitted by host A at t=0.
- At $t=d_{trans}$, some bits have already reached host B, but not the entire packet.

f. First Bit of Packet at $t=d_{trans}$ (if $d_{prop}<d_{trans}$):

- All bits of the packet have reached host B before $t=d_{trans}$.

g. Finding Distance *m* where $d_{prop}=d_{trans}$: $d_{prop}=d_{trans}$

m/s=*L/R*

Substitute the given values: m=((1500bytes * (2.5 * (10^8)))m/s) / (10*(10^6)bps) =3750meters

P7. In this problem, we consider sending real-time voice from Host A to Host B over a packet-switched network (VoIP). Host A converts analog voice to a digital 64 kbps bit stream on the fly. Host A then groups the bits into 56-byte packets. There is one link between Hosts A and B; its

transmission rate is 10 Mbps and its propagation delay is 10 msec. As soon as Host A gathers a packet, it sends it to Host B. As soon as Host B receives an entire packet, it converts the packet's bits to an analog signal. How much time elapses from the time a bit is created (from the original analog signal at Host A) until the bit is decoded (as part of the analog signal at Host B)?

Given:

- Digital bit stream rate: 64 kbps
- Packet size: 56 bytes
- Transmission rate: 10 Mbps
- Propagation delay: 10 ms

Time Elapsed from Creation to Decoding:

1. Convert 64 kbps to bps: 64×10^3 bps
2. Convert packet size to bits: 56×8bits
3. Transmission time per packet: 56×8 bits/10×10^6 bps
4. Total transmission time for one packet: 56×8 bits/10×10^6 bps+10ms

P8. Suppose users share a 10 Mbps link. Also suppose each user requires 200 kbps when transmitting, but each user transmits only 10 percent of the time. (See the discussion of packet switching versus circuit switching in Section 1.3.) a. When circuit switching is used, how many users can be supported? b. For the remainder of this problem, suppose packet switching is used. Find the probability that a given user is transmitting. c. Suppose there are 120 users. Find the probability that at any given time, exactly n users are transmitting simultaneously. (Hint: Use the binomial distribution.) d. Find the probability that there are 51 or more users transmitting simultaneously.

P9. Consider the discussion in Section 1.3 of packet switching versus circuit switching in which an example is provided with a 1 Mbps link. Users are generating data at a

rate of 100 kbps when busy, but are busy generating data only with probability p = 0.1. Suppose that the 1 Mbps link is replaced by a 1 Gbps link. a. What is N, the maximum number of users that can be supported simultaneously under circuit switching? b. Now consider packet switching and a user population of M users. Give a formula (in terms of p, M, N) for the probability that more than N users are sending data.

P10. Consider a packet of length L that begins at end system A and travels over three links to a destination end system. These three links are connected by two packet switches. Let d_i, s_i, and R_i denote the length, propagation speed, and the transmission rate of link i, for i = 1, 2, 3. The packet switch delays each packet by d_{proc}. Assuming no queuing delays, in terms of d_i, s_i, R_i, (i = 1, 2, 3), and L, what is the total end-to-end delay for the packet? Suppose now the packet is 1,500 bytes, the propagation speed on all three links is 2.5*(10^8)m/s, the transmission rates of all three links are 2.5 Mbps, the packet switch processing delay is 3 msec, the length of the first link is 5,000 km, the length of the second link is 4,000 km, and the length of the last link is 1,000km. For these values, what is the end-to-end delay?