

COMPUTER NETWORKS

Unit1: Computer Networks and the Internet

Topic: Introduction to Computer Networks- What is Internet? A Nuts and Bolts description

- 1. 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?**

Ans:

There is no difference. Throughout this text, the words “host” and “end system” are used interchangeably. End systems include PCs, workstations, Web servers, mail servers, PDAs, Internet-connected game consoles, etc.

- 2. Describe the protocol that might be used by two people having a telephonic conversation to initiate and end the conversation.**

Ans:

From Wikipedia: Diplomatic protocol is commonly described as a set of international courtesy rules. These well-established and time-honored rules have made it easier for nations and people to live and work together. Part of protocol has always been the acknowledgment of the hierarchical standing of all present. Protocol rules are based on the principles of civility.

Topic: What is a Protocol

- 1. 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.

Ans:

There is no single right answer to this question. Many protocols would do the trick. Here's a simple answer below:

Messages from ATM machine to Server

Msg name	Purpose
-----	-----
HELO <userid>	Let server know that there is a card in the ATM machine
ATM card transmits user ID to Server	
PASSWD <passwd>	User enters PIN, which is sent to server
BALANCE	User requests balance
WITHDRAWL <amount>	User asks to withdraw money
BYE	User all done

Messages from Server to ATM machine (display)

Msg name	Purpose
-----	-----
PASSWD	Ask user for PIN (password)
OK	Last requested operation (PASSWD, WITHDRAWL) OK
ERR	Last requested operation (PASSWD, WITHDRAWL) in ERROR
AMOUNT <amt>	Sent in response to BALANCE request
BYE	User done, display welcome screen at ATM

Correct operation:

client		server
HELO (userid)	----->	(check if valid userid)
	<-----	PASSWD
PASSWD <passwd>	----->	(check password)
	<-----	OK (password is OK)

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BALANCE ----->
                <----- AMOUNT <amt>
WITHDRAWL <amt> ----->    Check if enough $ to cover
withdrawl
                <----- OK
ATM dispenses $
                BYE ----->
                <----- BYE

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In situation when there's not enough money:

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HELO (userid) ----->    (check if valid userid)
                <----- PASSWD
PASSWD <passwd> ----->    (check password)
                <----- OK (password is OK)
BALANCE ----->
                <----- AMOUNT <amt>
WITHDRAWL <amt> ----->    Check if enough $ to cover
withdrawl
                <----- ERR (not enough funds)
error msg displayed
no $ given out
                BYE ----->
                <----- BYE

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Topic: Access Networks

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

Ans:

1. Dial-up modem over telephone line: home; 2. DSL over telephone line: home or small office; 3. Cable to HFC: home; 4. 100 Mbps switched Ethernet: enterprise; 5. Wifi (802.11): home and enterprise; 6. 3G and 4G: wide-area wireless.

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

Ans:

HFC bandwidth is shared among the users. On the downstream channel, all packets emanate from a single source, namely, the head end. Thus, there are no collisions in the downstream channel.

3. What access network technologies would be most suitable for providing Internet access in rural areas?

Ans:

In most American cities, the current possibilities include: dial-up; DSL; cable modem; fiber-to-the-home.

4. Dial-up modems and DSL both use the telephone line (a twisted-pair copper cable) as their transmission medium. Why then is DSL much faster than dial-up access?

Ans:

Ethernet LANs have transmission rates of 10 Mbps, 100 Mbps, 1 Gbps and 10 Gbps.

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

Ans:

Today, Ethernet most commonly runs over twisted-pair copper wire. It also can run over fibers optic links.

6. Dial-up modems, 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.

Ans:

Dial up modems: up to 56 Kbps, bandwidth is dedicated; ADSL: up to 24 Mbps downstream and 2.5 Mbps upstream, bandwidth is dedicated; HFC, rates up to 42.8 Mbps and upstream rates of up to 30.7 Mbps, bandwidth is shared. FTTH: 2-10Mbps upload; 10-20 Mbps download; bandwidth is not shared.

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- 7. Describe the different wireless technologies you use during the day and their characteristics. If you have a choice between multiple technologies, why do you prefer one over another?**

Ans:

There are two popular wireless Internet access technologies today:

a) Wifi (802.11) In a wireless LAN, wireless users transmit/receive packets to/from an base station (i.e., wireless access point) within a radius of few tens of meters. The base station is typically connected to the wired Internet and thus serves to connect wireless users to the wired network.

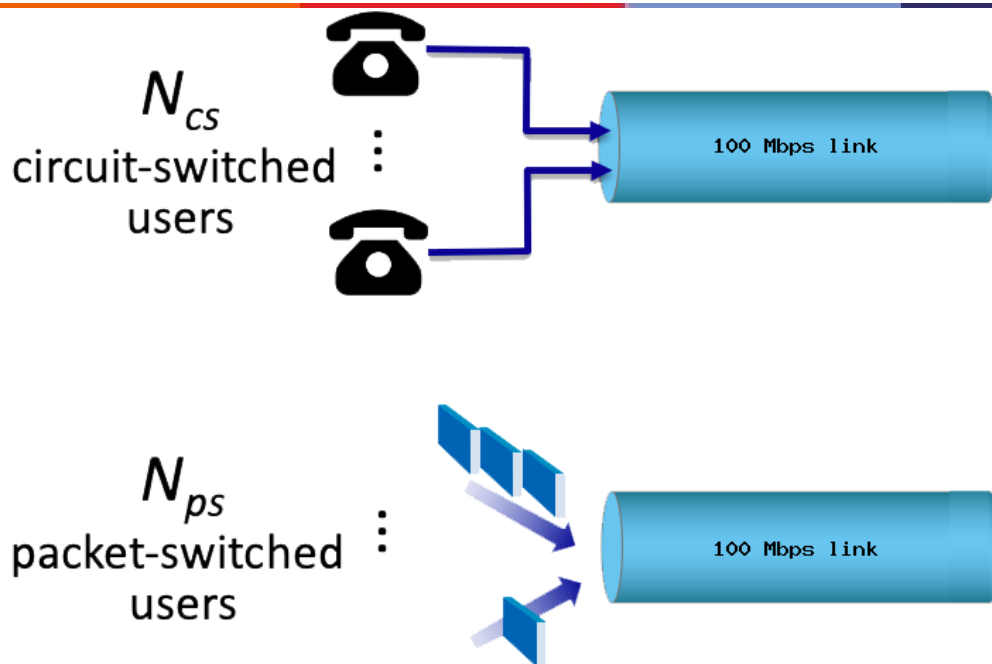
b) 3G and 4G wide-area wireless access networks. In these systems, packets are transmitted over the same wireless infrastructure used for cellular telephony, with the base station thus being managed by a telecommunications provider. This provides wireless access to users within a radius of tens of kilometers of the base station.

Topic: Circuit Switching Vs Packet Switching

- 1. Consider the two scenarios below:**

A circuit-switching scenario in which N_{cs} users, each requiring a bandwidth of 25 Mbps, must share a link of capacity 100 Mbps.

A packet-switching scenario with N_{ps} users sharing a 100 Mbps link, where each user again requires 25 Mbps when transmitting, but only needs to transmit 20 percent of the time.



Round your answer to two decimals after leading zeros.

Questions

- When circuit switching is used, what is the maximum number of users that can be supported?
- Suppose packet switching is used. If there are 7 packet-switching users, can this many users be supported under circuit-switching? Yes or No.
- Suppose packet switching is used. What is the probability that a given (specific) user is transmitting, and the remaining users are not transmitting?
- Suppose packet switching is used. What is the probability that one user (any one among the 7 users) is transmitting, and the remaining users are not transmitting?
- When one user is transmitting, what fraction of the link capacity will be used by this user? Write your answer as a decimal.
- What is the probability that any 3 users (of the total 7 users) are transmitting and the remaining users are not transmitting?
- What is the probability that more than 4 users are transmitting?

Ans:

- When circuit switching is used, at most 4 users can be supported. This is because each circuit-switched user must be allocated its 25 Mbps

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- bandwidth, and there is 100 Mbps of link capacity that can be allocated.
- b. No. Under circuit switching, the 7 users would each need to be allocated 25 Mbps, for an aggregate of 175 Mbps - more than the 100 Mbps of link capacity available.
 - c. The probability that a given (specific) user is busy transmitting, which we'll denote p , is just the fraction of time it is transmitting, i.e. 0.2. The probability that one specific other user is not busy is $(1-p)$, and so the probability that all of the other $N_{ps}-1$ users are not transmitting is $(1-p)^{N_{ps}-1}$. Thus the probability that one specific user is transmitting and the remaining users are not transmitting is $p \cdot (1-p)^{N_{ps}-1}$, which has the numerical value of 0.052.
 - d. The probability that exactly one (any one) of the N_{ps} users is transmitting is N_{ps} times the probability that a given specific user is transmitting and the remaining users are not transmitting. The answer is thus $N_{ps} \cdot p \cdot (1-p)^{N_{ps}-1}$, which has the numerical value of 0.37.
 - e. This user will be transmitting at a rate of 25 Mbps over the 100 Mbps link, using a fraction 0.25 of the link's capacity when busy.
 - f. The probability that 3 specific users of the total 7 users are transmitting and the other 4 users are idle is $p^3(1-p)^4$. Thus the probability that any 4 of the 7 users are busy is $\text{choose}(7, 3) \cdot p^3(1-p)^4$, where $\text{choose}(7, 3)$ is the $(7, 3)$ coefficient of the binomial distribution). The numerical value of this probability is 0.11.

The probability that more than 4 users of the total 7 users are transmitting is $\sum_{i=5,7} \text{choose}(7, i) \cdot p^i(1-p)^{7-i}$. The numerical value of this probability is 0.0047. Note that 4 is the maximum number of users that can be supported using circuit switching. With packet switching, nearly twice as many users (7) are supported with a small probability that more than 4 of these packet-switching users are busy at the same time.

Topic: Internet Structure

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

If the two ISPs do not peer with each other, then when they send traffic to each other they have to send the traffic through a provider ISP (intermediary), to which they have to pay for carrying the traffic. By peering with each other directly, the two ISPs can reduce their payments to their provider ISPs. An Internet Exchange Points (IXP) (typically in a standalone building with its own switches) is a meeting point where multiple ISPs can connect and/or peer together. An ISP earns its money by charging each of the the ISPs that connect to the IXP a relatively small fee, which may depend on the amount of traffic sent to or received from the IXP.

2. Why is a content provider considered a different Internet entity today? How does a content provider connect to other ISPs? Why?

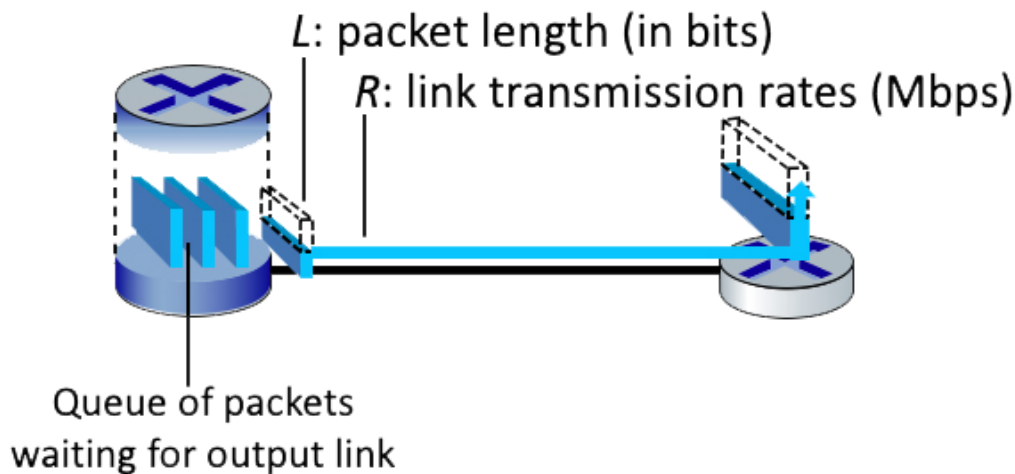
Ans:

Google's private network connects together all its data centers, big and small. Traffic between the Google data centers passes over its private network rather than over the public Internet. Many of these data centers are located in, or close to, lower tier ISPs. Therefore, when Google delivers content to a user, it often can bypass higher tier ISPs. What motivates content providers to create these networks? First, the content provider has more control over the user experience, since it has to use few intermediary ISPs. Second, it can save money by sending less traffic into provider networks. Third, if ISPs decide to charge more money to highly profitable content providers (in countries where net neutrality doesn't apply), the content providers can avoid these extra payments.

Topic: Overview of delay in Packet-switched networks, Queuing delay and Packet

Problems on Delay - 1

- 1. Consider the figure below, in which a single router is transmitting packets, each of length L bits, over a single link with transmission rate R Mbps to another router at the other end of the link.**



Suppose that the packet length is $L = 16000$ bits, and that the link transmission rate along the link to router on the right is $R = 100$ Mbps. Round your answer to two decimals after leading zeros.

Questions

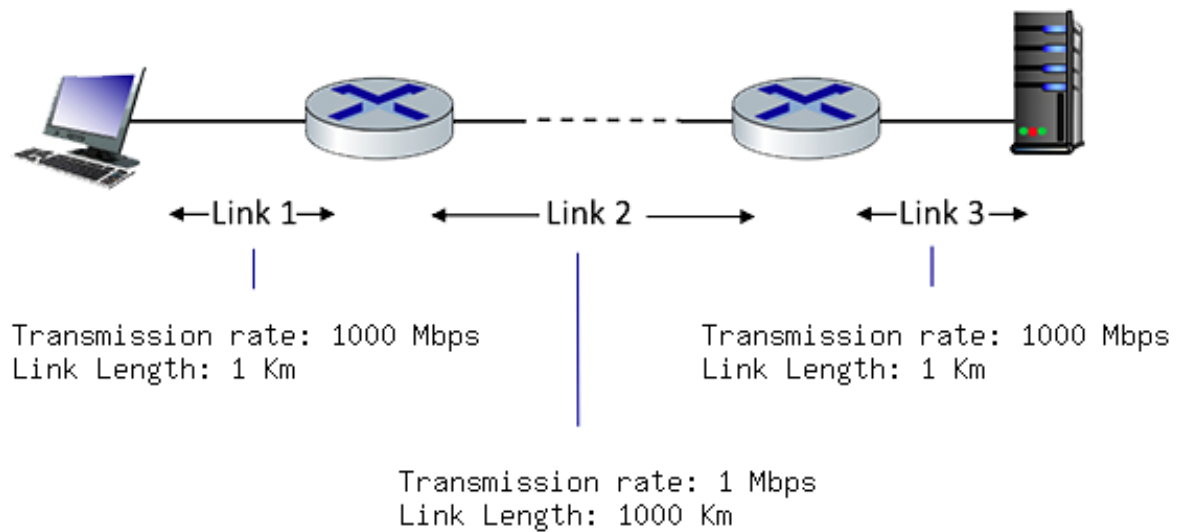
- What is the transmission delay?
- What is the maximum number of packets per second that can be transmitted by this link?

Ans:

- The transmission delay = $L/R = 16000 \text{ bits} / 100000000 \text{ bps} = 0.00016$ seconds
- The number of packets that can be transmitted in a second into the link = $R / L = 100000000 \text{ bps} / 16000 \text{ bits} = 6250$ packets

Problems on Delay - 2

Consider the figure below, with three links, each with the specified transmission rate and link length.



Assume the length of a packet is 8000 bits. The speed of light propagation delay on each link is 3×10^8 m/sec. Round your answer to two decimals after leading zeros.

Questions

- What is the transmission delay of link 1?
- What is the propagation delay of link 1?
- What is the total delay of link 1?
- What is the transmission delay of link 2?
- What is the propagation delay of link 2?
- What is the total delay of link 2?
- What is the transmission delay of link 3?
- What is the propagation delay of link 3?
- What is the total delay of link 3?
- What is the total delay?

Ans:

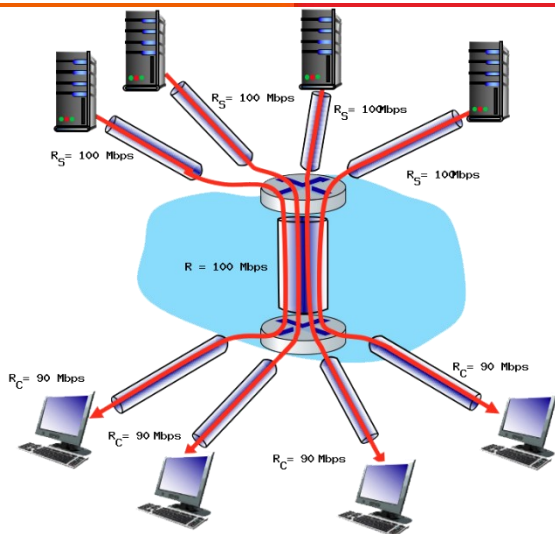
- Link 1 transmission delay = $L/R = 8000 \text{ bits} / 1000 \text{ Mbps} = 8.00 \times 10^{-6}$ seconds
- Link 1 propagation delay = $d/s = (1 \text{ Km}) * 1000 / 3 \times 10^8 \text{ m/sec} = 3.33 \times 10^{-6}$ seconds

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- c. Link 1 total delay = $d_t + d_p = 8.00E-6 \text{ seconds} + 3.33E-6 \text{ seconds} = 1.13E-5 \text{ seconds}$
 - d. Link 2 transmission delay = $L/R = 8000 \text{ bits} / 1 \text{ Mbps} = 0.008 \text{ seconds}$
 - e. Link 2 propagation delay = $d/s = (1000 \text{ Km}) * 1000 / 3 * 10^8 \text{ m/sec} = 0.0033 \text{ seconds}$
 - f. Link 2 total delay = $d_t + d_p = 0.008 \text{ seconds} + 0.0033 \text{ seconds} = 0.011 \text{ seconds}$
 - g. Link 3 transmission delay = $L/R = 8000 \text{ bits} / 1000 \text{ Mbps} = 8.00E-6 \text{ seconds}$
 - h. Link 3 propagation delay = $d/s = (1 \text{ Km}) * 1000 / 3 * 10^8 \text{ m/sec} = 3.33E-6 \text{ seconds}$
 - i. Link 3 total delay = $d_t + d_p = 8.00E-6 \text{ seconds} + 3.33E-6 \text{ seconds} = 1.13E-5 \text{ seconds}$
 - j. The total delay = $d_{L1} + d_{L2} + d_{L3} = 1.13E-5 \text{ seconds} + 0.011 \text{ seconds} + 1.13E-5 \text{ seconds} = 0.011 \text{ seconds}$

Topic: Throughput in computer networks

1. Consider the scenario shown below, with four different servers connected to four different clients over four three-hop paths. The four pairs share a common middle hop with a transmission capacity of $R = 100 \text{ Mbps}$. The four links from the servers to the shared link have a transmission capacity of $RS = 100 \text{ Mbps}$. Each of the four links from the shared middle link to a client has a transmission capacity of $RC = 90 \text{ Mbps}$.

You might want to review Figure 1.20 in the text before answering the following questions.



Questions

- What is the maximum achievable end-end throughput (in Mbps) for each of four client-to-server pairs, assuming that the middle link is fairly shared (divides its transmission rate equally)?
- Which link is the bottleneck link? Format as R_c , R_s , or R .
- Assuming that the servers are sending at the maximum rate possible, what are the link utilizations for the server links (R_s)? Answer as a decimal.
- Assuming that the servers are sending at the maximum rate possible, what are the link utilizations for the client links (R_c)? Answer as a decimal.
- Assuming that the servers are sending at the maximum rate possible, what is the link utilizations for the shared link (R)? Answer as a decimal.

Ans:

- The maximum achievable end-end throughput is the capacity of the link with the minimum capacity, which is 25 Mbps.
- The bottleneck link is the link with the smallest capacity between R_s , R_c , and $R/4$. The bottleneck link is R .
- The server's utilization = $R_{\text{bottleneck}} / R_s = 25 / 100 = 0.25$
- The client's utilization = $R_{\text{bottleneck}} / R_c = 25 / 90 = 0.28$

e. The shared link's utilization = $R_{\text{bottleneck}} / (R / 4) = 25 / (100 / 4) = 1$

Topic: Protocol layers - The OSI model Layered Architecture

- 1. If two end-systems are connected through multiple routers and the data-link level between them ensures reliable data delivery, is a transport protocol offering reliable data delivery between these two end-systems necessary? Why?**

Ans:

Five generic tasks are error control, flow control, segmentation and reassembly, multiplexing, and connection setup. Yes, these tasks can be duplicated at different layers. For example, error control is often provided at more than one layer.

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

Ans:

The five layers in the Internet protocol stack are – from top to bottom – the application layer, the transport layer, the network layer, the link layer, and the physical layer. The principal responsibilities are outlined in Section 1.5.1.

- 3. What do encapsulation and de-encapsulation mean? Why are they needed in a layered protocol stack?**

Ans:

Application-layer message: data which an application wants to send and passed onto the transport layer; transport-layer segment: generated by the transport layer and encapsulates application-layer message with transport layer header; network-layer datagram: encapsulates transport-layer segment with a network-layer header; link-layer frame: encapsulates network-layer datagram with a link-layer header.

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- 4. 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?**

Ans:

Routers process network, link and physical layers (layers 1 through 3). (This is a little bit of a white lie, as modern routers sometimes act as firewalls or caching components, and process Transport layer as well.) Link layer switches process link and physical layers (layers 1 through 2). Hosts process all five layers.

Topic: TCP variants located in the ISO/OSI protocol stack

- 1. Early versions of TCP combined functions for both forwarding and reliable delivery. How are these TCP variants located in the ISO/OSI protocol stack? Why were forwarding functions later separated from TCP? What were the consequences?**

Ans:

The circuit-switched telephone networks and the Internet are connected together at "gateways". When a Skype user (connected to the Internet) calls an ordinary telephone, a circuit is established between a gateway and the telephone user over the circuit switched network. The skype user's voice is sent in packets over the Internet to the gateway. At the gateway, the voice signal is reconstructed and then sent over the circuit. In the other direction, the voice signal is sent over the circuit switched network to the gateway. The gateway packetizes the voice signal and sends the voice packets to the Skype user.