Problems

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.

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.

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?

P4. Consider the circuit-switched network in Figure 1.13. Recall that there are 4 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?

P5. Review the car-caravan analogy in Section 1.4. Assume a propagation speed of 100 km/hour.

a. Suppose the caravan travels 150 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.

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.

VideoNote
Exploring propagation delay and transmission delay

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 $t = d_{\text{trans}}$, where is the last bit of the packet?

e. Suppose d_{prop} is greater than d_{trans}. At time $t = d_{\text{trans}}$, where is the first bit of the packet?

f. Suppose d_{prop} is less than d_{trans}. At time $t = d_{\text{trans}}$, where is the first bit of the packet?

g. Suppose $s = 2.5 \cdot 10^8$, $L = 120$ bits, and $R = 56$ kbps. Find the distance m so that d_{prop} equals d_{trans}.

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 2 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)?

P8. Suppose users share a 3 Mbps link. Also suppose each user requires 150 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 21 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 the network illustrated in Figure 1.16. Assume the two hosts on the left of the figure start transmitting packets of 1500 bytes at the same time towards Router B. Suppose the link rates between the hosts and Router A is 4-Mbps. One link has a 6-ms propagation delay and the other has a 2-ms propagation delay. Will queuing delay occur at Router A?

P11. Consider the scenario in Problem P10 again, but now assume the links between the hosts and Router A have different rates R_1 and R_2 byte/s in addition to different propagation delays d_1 and d_2. Assume the packet lengths for the two hosts are of L bytes. For what values of the propagation delay will no queuing delay occur at Router A?

P12. Consider a client and a server connected through one router. Assume the router can start transmitting an incoming packet after receiving its first h bytes instead of the whole packet. Suppose that the link rates are R byte/s and that the client transmits one packet with a size of L bytes to the server. What is the end-to-end delay? Assume the propagation, processing, and queuing delays are negligible. Generalize the previous result to a scenario where the client and the server are interconnected by N routers.

P13. (a) Suppose N packets arrive simultaneously to a link at which no packets are currently being transmitted or queued. Each packet is of length L and the link has transmission rate R. What is the average queuing delay for the N packets?

(b) Now suppose that N such packets arrive to the link every LN/R seconds. What is the average queuing delay of a packet?

P14. Consider the queuing delay in a router buffer. Let I denote traffic intensity; that is, $I = La/R$. Suppose that the queuing delay takes the form $IL/R\,(1 - I)$ for $I < 1$.

a. Provide a formula for the total delay, that is, the queuing delay plus the transmission delay.

b. Plot the total delay as a function of L/R.

P15. Let a denote the rate of packets arriving at a link in packets/sec, and let μ denote the link's transmission rate in packets/sec. Based on the formula for the total delay (i.e., the queuing delay plus the transmission delay) derived in the previous problem, derive a formula for the total delay in terms of a and μ.

P16. Consider a router buffer preceding an outbound link. In this problem, you will use Little's formula, a famous formula from queuing theory. Let N denote the average number of packets in the buffer plus the packet being transmitted. Let a denote the rate of packets arriving at the link. Let d denote the average total delay (i.e., the queuing delay plus the transmission delay) experienced by a packet. Little's formula is $N = a \cdot d$. Suppose that on average, the buffer contains 10 packets, and the average packet queuing delay is 10 msec. The link's transmission rate is 100 packets/sec. Using Little's formula, what is the average packet arrival rate, assuming there is no packet loss?

P17. Consider the network illustrated in Figure 1.12. Would Equation 1.2 hold in such a scenario? If so, under which conditions? If not, why? (Assume N is the number of links between a source and a destination in the figure.)

P18. Perform a Traceroute between source and destination on the same continent at three different hours of the day.

VideoNote
Using Traceroute to discover network paths and measure network delay

a. Find the average and standard deviation of the round-trip delays at each of the three hours.

b. Find the number of routers in the path at each of the three hours. Did the paths change during any of the hours?

c. Try to identify the number of ISP networks that the Traceroute packets pass through from source to destination. Routers with similar names and/or similar IP addresses should be considered as part of the same ISP. In your experiments, do the largest delays occur at the peering interfaces between adjacent ISPs?

d. Repeat the above for a source and destination on different continents. Compare the intra-continent and inter-continent results.

P19. (a) Visit the site www.traceroute.org and perform traceroutes from two different cities in France to the same destination host in the United States. How many links are the same in the two traceroutes? Is the transatlantic link the same?

(b) Repeat (a) but this time choose one city in France and another city in Germany.

(c) Pick a city in the United States, and perform traceroutes to two hosts, each in a different city in China. How many links are common in the two traceroutes? Do the two traceroutes diverge before reaching China?

P20. Consider the throughput example corresponding to Figure 1.20(b). Now suppose that there are M client-server pairs rather than 10. Denote R_s, R_c, and R for the rates of the server links, client links, and network link. Assume all other links have abundant capacity and that there is no other traffic in the network besides the traffic generated by the M client-server pairs. Derive a general expression for throughput in terms of R_s, R_c, R, and M.

P21. Assume a client and a server can connect through either network (a) or (b) in Figure 1.19. Assume that $R_i = (R_c + R_s) / i$, for $i = 1, 2, ..., N$. In what case will network (a) have a higher throughput than network (b)?

P22. Consider Figure 1.19(b). Suppose that each link between the server and the client has a packet loss probability p, and the packet loss probabilities for these links are independent. What is the probability that a packet (sent by the server) is successfully received by the receiver? If a packet is lost in the path from the server to the client, then the server will re-transmit the packet. On average, how many times will the server re-transmit the packet in order for the client to successfully receive the packet?

P23. Consider Figure 1.19(a). Assume that we know the bottleneck link along the path from the server to the client is the first link with rate R_s bits/sec. Suppose we send a pair of packets back to back from the server to the client, and there is no other traffic on this path. Assume each packet of size L bits, and both links have the same propagation delay d_{prop}.

a. What is the packet inter-arrival time at the destination? That is, how much time elapses from when the last bit of the first packet arrives until the last bit of the second packet arrives?

b. Now assume that the second link is the bottleneck link (i.e., $R_c < R_s$). Is it possible that the second packet queues at the input queue of the second link? Explain. Now suppose that the server sends the second packet T seconds after sending the first packet. How large must T be to ensure no queuing before the second link? Explain.

P24. Consider a user who needs to transmit 1.5 gigabytes of data to a server. The user lives in a small town where only dial-up access is available. A bus visits the small town once a day from the closest city, located 150 km away, and stops in front of the user's house. The bus has a 100-Mbps WiFi connection. It can collect data from users in rural areas and transfer them to the Internet through a 1 Gbps link once it gets back to the city. Suppose the average speed of the bus is 60 km/h. What is the fastest way the user can transfer the data to the server?

P25. Suppose two hosts, A and B, are separated by 20,000 kilometers and are connected by a direct link of $R = 2$ Mbps. Suppose the propagation speed over the link is $2.5 \cdot 10^8$ meters/sec.

a. Calculate the bandwidth-delay product, $R \cdot d_{prop}$.

b. Consider sending a file of 800,000 bits from Host A to Host B. Suppose the file is sent continuously as one large message. What is the maximum number of bits that will be in the link at any given time?

c. Provide an interpretation of the bandwidth-delay product.

d. What is the width (in meters) of a bit in the link? Is it longer than a football field?

e. Derive a general expression for the width of a bit in terms of the propagation speed s, the transmission rate R, and the length of the link m.

P26. Consider problem P25 but now with a link of $R = 1$ Gbps.

a. Calculate the bandwidth-delay product, $R \cdot d_{prop}$.

b. Consider sending a file of 800,000 bits from Host A to Host B. Suppose the file is sent continuously as one big message. What is the maximum number of bits that will be in the link at any given time?

c. What is the width (in meters) of a bit in the link?

P27. Consider the scenario illustrated in Figure 1.19(a). Assume R_s is 20 Mbps, R_c is 10 Mbps, and the server is continuously sending traffic to the client. Also assume the router between the server and the client can buffer at most four messages. After how many messages sent by the server will packet loss starts occurring at the router?

P28. Generalize the result obtained in Problem P27 for the case where the router can buffer m messages.

P29. Suppose there is a 10-Mbps microwave link between a geostationary satellite and its base station on Earth. Every minute the satellite takes a digital photo and sends it to the base station. Assume a propagation speed of $2.4 \cdot 10^8$ meters/sec.

a. What is the propagation delay of the link?

b. What is the bandwidth-delay product, $R \cdot d_{prop}$?

c. Let x denote the size of the photo. What is the minimum value of x for the microwave link to be continuously transmitting?

P30. Consider the airline travel analogy in our discussion of layering in Section 1.5, and the addition of headers to protocol data units as they flow down the protocol stack. Is there an equivalent notion of header information that is added to passengers and baggage as they move down the airline protocol stack?

P31. In modern packet-switched networks, including the Internet, the source host segments long, application-layer messages (for example, an image or a music file) into smaller packets and sends the packets into the network. The receiver then reassembles the packets back into the original message. We refer to this process as *message segmentation*. Figure 1.27 illustrates the end-to-end transport of a message with and without message segmentation. Consider a message that is $8 \cdot 10^6$ bits long that is to be sent from source to destination in Figure 1.27. Suppose each link in the figure is 2 Mbps. Ignore propagation, queuing, and processing delays.

a. Consider sending the message from source to destination *without* message segmentation. How long does it take to move the message from the source host to the first packet switch? Keeping in mind that each switch uses store-and-forward packet switching, what is the total time to move the message from source host to destination host?

b. Now suppose that the message is segmented into 800 packets, with each packet being 10,000 bits long. How long does it take to move the first packet from source host to the first switch? When the first packet is being sent from the first switch to the second switch, the second packet is being sent from the source host to the first switch. At what time will the second packet be fully received at the first switch?

c. How long does it take to move the file from source host to destination host when message segmentation is used? Compare this result with your answer in part (a) and comment.

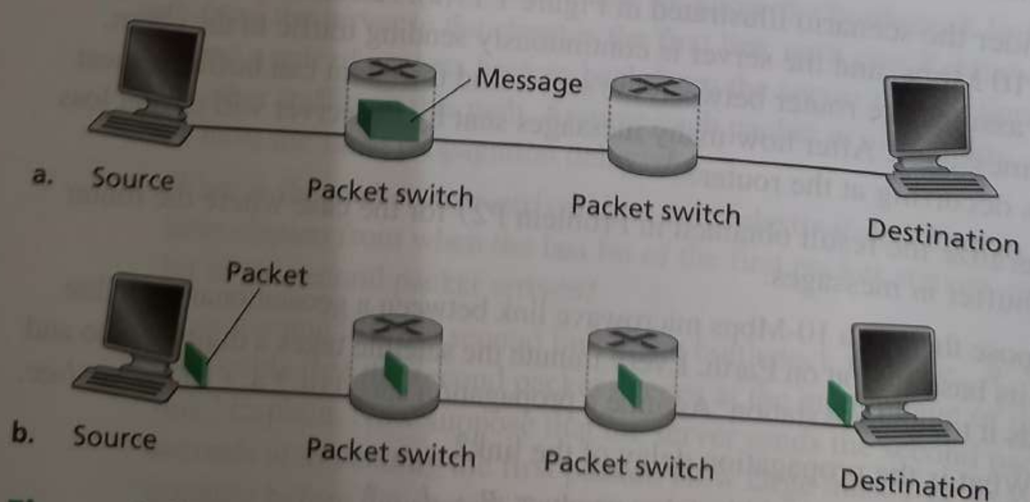


Figure 1.27 ♦ End-to-end message transport: (a) without message segmentation; (b) with message segmentation

d. In addition to reducing delay, what are reasons to use message segmentation?

e. Discuss the drawbacks of message segmentation.

P32. Consider Problem P31 and assume that the propagation delay is 250 ms. Recalculate the total time needed to transfer the source data with and without segmentation. Is segmentation more beneficial or less if there is propagation delay?

P33. Consider sending a large file of F bits from Host A to Host B. There are three links (and two switches) between A and B, and the links are uncongested (that is, no queuing delays). Host A segments the file into segments of S bits each and adds 80 bits of header to each segment, forming packets of $L = 80 + S$ bits. Each link has a transmission rate of R bps. Find the value of S that minimizes the delay of moving the file from Host A to Host B. Disregard propagation delay.

P34. 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?

Wireshark Lab

"Tell me and I forget. Show me and I remember. Involve me and I understand."

Chinese proverb

One's understanding of network protocols can often be greatly deepened by seeing them in action and by playing around with them—observing the sequence of messages exchanged between two protocol entities, delving into the details of protocol operation, causing protocols to perform certain actions, and observing these actions and their consequences. This can be done in simulated scenarios or in a real network environment such as the Internet. The interactive animations at the textbook Web site take the first approach. In the Wireshark labs, we'll take the latter approach. You'll run network applications in various scenarios using a computer on your desk, at home, or in a lab. You'll observe the network protocols in your computer, interacting and exchanging messages with protocol entities executing elsewhere in the Internet. Thus, you and your computer will be an integral part of these live labs. You'll observe—and you'll learn—by doing.

The basic tool for observing the messages exchanged between executing protocol entities is called a **packet sniffer**. As the name suggests, a packet sniffer passively copies (sniffs) messages being sent from and received by your computer; it also displays the contents of the various protocol fields of these captured messages. A screenshot of the Wireshark packet sniffer is shown in Figure 1.28. Wireshark is a free packet sniffer that runs on Windows, Linux/Unix, and Mac computers.

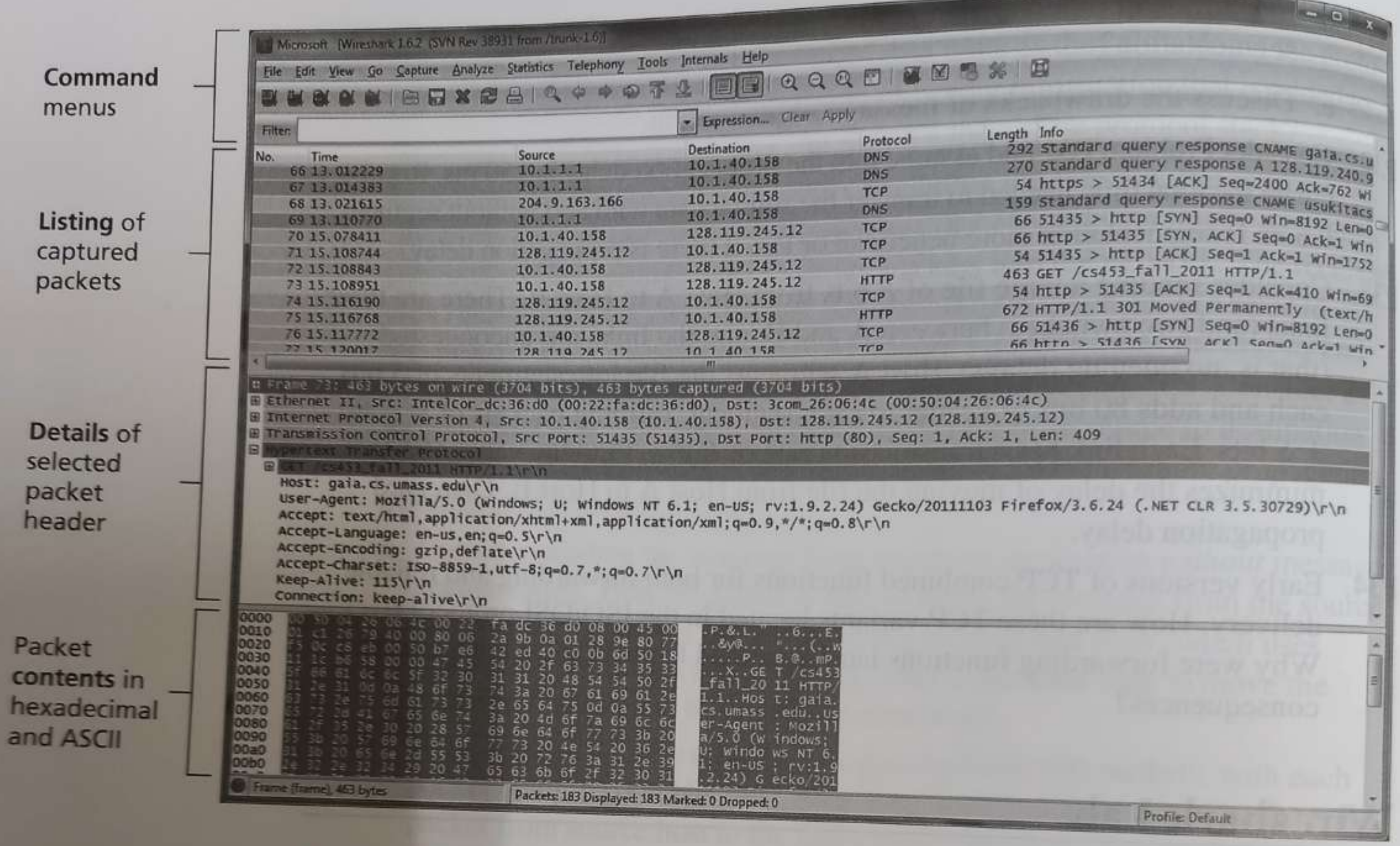


Figure 1.28 ♦ A Wireshark screenshot (Wireshark screenshot reprinted by permission of the Wireshark Foundation.)

Throughout the textbook, you will find Wireshark labs that allow you to explore a number of the protocols studied in the chapter. In this first Wireshark lab, you'll obtain and install a copy of Wireshark, access a Web site, and capture and examine the protocol messages being exchanged between your Web browser and the Web server.

You can find full details about this first Wireshark lab (including instructions about how to obtain and install Wireshark) at the Web site http://www.pearsonglobaleditions.com/kurose.

R27. For the client-server application over TCP described in Section 2.7, why must the server program be executed before the client program? For the client-server application over UDP, why may the client program be executed before the server program?

Problems

P1. True or false?

a. A user requests a Web page that consists of some text and three images. For this page, the client will send one request message and receive four response messages.

b. Two distinct Web pages (for example, `www.mit.edu/research.html` and `www.mit.edu/students.html`) can be sent over the same persistent connection.

c. With nonpersistent connections between browser and origin server, it is possible for a single TCP segment to carry two distinct HTTP request messages.

d. The `Date:` header in the HTTP response message indicates when the object in the response was last modified.

e. HTTP response messages never have an empty message body.

P2. SMS, iMessage, and WhatsApp are all smartphone real-time messaging systems. After doing some research on the Internet, for each of these systems write one paragraph about the protocols they use. Then write a paragraph explaining how they differ.

P3. Assume you open a browser and enter `http://yourbusiness.com/about.html` in the address bar. What happens until the webpage is displayed? Provide details about the protocol(s) used and a high-level description of the messages exchanged.

P4. Consider the following string of ASCII characters that were captured by Wireshark when the browser sent an HTTP GET message (i.e., this is the actual content of an HTTP GET message). The characters *<cr><lf>* are carriage return and line-feed characters (that is, the italized character string *<cr>* in the text below represents the single carriage-return character that was contained at that point in the HTTP header). Answer the following questions, indicating where in the HTTP GET message below you find the answer.

```
GET /cs453/index.html HTTP/1.1<cr><lf>Host: gai
a.cs.umass.edu<cr><lf>User-Agent: Mozilla/5.0 (
Windows;U; Windows NT 5.1; en-US; rv:1.7.2) Gec
ko/20040804 Netscape/7.2 (ax) <cr><lf>Accept:ex
t/xml, application/xml, application/xhtml+xml, text
/html;q=0.9, text/plain;q=0.8,image/png,*/*;q=0.5
```

```
<cr><lf>Accept-Language: en-us,en;q=0.5<cr><lf>Accept-
Encoding: zip,deflate<cr><lf>Accept-Charset: ISO
-8859-1,utf-8;q=0.7,*;q=0.7<cr><lf>Keep-Alive: 300<cr>
<lf>Connection:keep-alive<cr><lf><cr><lf>
```

a. What is the URL of the document requested by the browser?

b. What version of HTTP is the browser running?

c. Does the browser request a non-persistent or a persistent connection?

d. What is the IP address of the host on which the browser is running?

e. What type of browser initiates this message? Why is the browser type needed in an HTTP request message?

P5. The text below shows the reply sent from the server in response to the HTTP GET message in the question above. Answer the following questions, indicating where in the message below you find the answer.

```
HTTP/1.1 200 OK<cr><lf>Date: Tue, 07 Mar 2008
12:39:45GMT<cr><lf>Server: Apache/2.0.52 (Fedora)
<cr><lf>Last-Modified: Sat, 10 Dec2005 18:27:46
GMT<cr><lf>ETag: "526c3-f22-a88a4c80"<cr><lf>Accept-
Ranges: bytes<cr><lf>Content-Length: 3874<cr><lf>
Keep-Alive: timeout=max=100<cr><lf>Connection:
Keep-Alive<cr><lf>Content-Type: text/html; charset=
ISO-8859-1<cr><lf><cr><lf><!doctype html public "-
//w3c//dtd html 4.0transitional//en"><lf><html><lf>
<head><lf> <meta http-equiv="Content-Type"
content="text/html; charset=iso-8859-1"><lf> <meta
name="GENERATOR" content="Mozilla/4.79 [en] (Windows NT
5.0; U) Netscape]"><lf> <title>CMPSCI 453 / 591 /
NTU-ST550ASpring 2005 homepage</title><lf></head><lf>
<much more document text following here (not shown)>
```

a. Was the server able to successfully find the document or not? What time was the document reply provided?

b. When was the document last modified?

c. How many bytes are there in the document being returned?

d. What are the first 5 bytes of the document being returned? Did the server agree to a persistent connection?

P6. Obtain the HTTP/1.1 specification (RFC 2616). Answer the following questions:

a. Explain the mechanism used for signaling between the client and server to indicate that a persistent connection is being closed. Can the client, the server, or both signal the close of a connection?

b. What encryption services are provided by HTTP?

c. Can a client open three or more simultaneous connections with a given server?

d. Either a server or a client may close a transport connection between them if either one detects the connection has been idle for some time. Is it possible that one side starts closing a connection while the other side is transmitting data via this connection? Explain.

P7. Assume that the RTT between a client and the local DNS server is TT_l, while the RTT between the local DNS server and other DNS servers is RTT_r. Assume that no DNS server performs caching.

a. What is the total response time for the scenario illustrated in Figure 2.19?

b. What is the total response time for the scenario illustrated in Figure 2.20?

c. Assume now that the DNS record for the requested name is cached at the local DNS server. What is the total response time for the two scenarios?

P8. Referring to Problem P7, suppose the HTML file references eight very small objects on the same server. Neglecting transmission times, how much time elapses with

a. Non-persistent HTTP with no parallel TCP connections?

b. Non-persistent HTTP with the browser configured for 5 parallel connections?

c. Persistent HTTP?

P9. Consider Figure 2.12, for which there is an institutional network connected to the Internet. Suppose that the average object size is 850,000 bits and that the average request rate from the institution's browsers to the origin servers is 16 requests per second. Also suppose that the amount of time it takes from when the router on the Internet side of the access link forwards an HTTP request until it receives the response is three seconds on average (see Section 2.2.5). Model the total average response time as the sum of the average access delay (that is, the delay from Internet router to institution router) and the average Internet delay. For the average access delay, use $\Delta/(1 - \Delta\beta)$, where Δ is the average time required to send an object over the access link and β is the arrival rate of objects to the access link.

a. Find the total average response time.

b. Now suppose a cache is installed in the institutional LAN. Suppose the miss rate is 0.4. Find the total response time.

P10. Assume you request a webpage consisting of one document and five images. The document size is 1 kbyte, all images have the same size of 50 kbytes, the download rate is 1 Mbps, and the RTT is 100 ms. How long does it take to obtain the whole webpage under the following conditions? (Assume no DNS name query is needed and the impact of the request line and the headers in the HTTP messages is negligible).

a. Nonpersistent HTTP with serial connections.
b. Nonpersistent HTTP with two parallel connections.
c. Nonpersistent HTTP with six parallel connections.
d. Persistent HTTP with one connection.

P11. Generalize the results obtained for the first and the last scenario in the previous problem to a document size of L_d bytes, N images with size of L_i bytes (for $0 \leq i < N$), a rate of R byte/s and an RTT of RTT_{avg}.

P12. Write a simple TCP program for a server that accepts lines of input from a client and prints the lines onto the server's standard output. (You can do this by modifying the TCPServer.py program in the text.) Compile and execute your program. On any other machine that contains a Web browser, set the proxy server in the browser to the host that is running your server program; also configure the port number appropriately. Your browser should now send its GET request messages to your server, and your server should display the messages on its standard output. Use this platform to determine whether your browser generates conditional GET messages for objects that are locally cached.

P13. Describe a few scenarios in which mail access protocols are not needed.

P14. Why does an SMTP server retry to transmit a message even though TCP is used to connect with the destination?

P15. Read RFC 5321 for SMTP. What does MTA stand for? Consider the following received spam e-mail (modified from a real spam e-mail). Assuming only the originator of this spam e-mail is malicious and all other hosts are honest, identify the malacious host that has generated this spam e-mail.

```
From - Fri Nov 07 13:41:30 2008
Return-Path: <tennis5@pp33head.com>
Received: from barmail.cs.umass.edu (barmail.cs.umass.
edu
```

```
[128.119.240.3]) by cs.umass.edu (8.13.1/8.12.6) for
<hg@cs.umass.edu>; Fri, 7 Nov 2008 13:27:10 -0500
Received: from asusus-4b96 (localhost [127.0.0.1]) by
barmail.cs.umass.edu (Spam Firewall) for <hg@cs.umass.
edu>; Fri, 7
Nov 2008 13:27:07 -0500 (EST)
Received: from asusus-4b96 ([58.88.21.177]) by barmail.
cs.umass.edu
for <hg@cs.umass.edu>; Fri, 07 Nov 2008 13:27:07 -0500
(EST)
Received: from [58.88.21.177] by inbnd55.exchangeddd.
com; Sat, 8
Nov 2008 01:27:07 +0700
From: "Jonny" <tennis5@pp33head.com>
To: <hg@cs.umass.edu>

Subject: How to secure your savings
```

P16. Read the DNS SRV RFC, RFC 2782. What is the purpose of the SRV record?

P17. Consider accessing your e-mail with POP3.

a. Suppose you have configured your POP mail client to operate in the download-and-delete mode. Complete the following transaction:

```
C: list
S: 1 498
S: 2 912
S: .
C: retr 1
S: blah blah ...
S: ..........blah
S: .
?
?
```

b. Suppose you have configured your POP mail client to operate in the download-and-keep mode. Complete the following transaction:

```
C: list
S: 1 498
S: 2 912
S: .
C: retr 1
```

```
S: blah blah ...
S: ..........blah
S: .
?
?
```

c. Suppose you have configured your POP mail client to operate in the download-and-keep mode. Using your transcript in part (b), suppose you retrieve messages 1 and 2, exit POP, and then five minutes later you again access POP to retrieve new e-mail. Suppose that in the five-minute interval no new messages have been sent to you. Provide a transcript of this second POP session.

P18. a. What is a *whois* database?

b. Use various whois databases on the Internet to obtain the names of two DNS servers. Indicate which whois databases you used.

c. Use nslookup on your local host to send DNS queries to three DNS servers: your local DNS server and the two DNS servers you found in part (b). Try querying for Type A, NS, and MX reports. Summarize your findings.

d. Use nslookup to find a Web server that has multiple IP addresses. Does the Web server of your institution (school or company) have multiple IP addresses?

e. Use the ARIN whois database to determine the IP address range used by your university.

f. Describe how an attacker can use whois databases and the nslookup tool to perform reconnaissance on an institution before launching an attack.

g. Discuss why whois databases should be publicly available.

P19. In this problem, we use the useful *dig* tool available on Unix and Linux hosts to explore the hierarchy of DNS servers. Recall that in Figure 2.19, a DNS server in the DNS hierarchy delegates a DNS query to a DNS server lower in the hierarchy, by sending back to the DNS client the name of that lower-level DNS server. First read the man page for *dig*, and then answer the following questions.

a. Starting with a root DNS server (from one of the root servers [a-m].root-servers.net), initiate a sequence of queries for the IP address for your department's Web server by using *dig*. Show the list of the names of DNS servers in the delegation chain in answering your query.

b. Repeat part (a) for several popular Web sites, such as google.com, yahoo.com, or amazon.com.

P20. Consider the scenarios illustrated in Figures 2.12 and 2.13. Assume the rate of the institutional network is R_l and that of the bottleneck link is R_b. Suppose there are N clients requesting a file of size L with HTTP at the same time. For what values of R_l would the file transfer takes less time when a proxy is installed at the institutional network? (Assume the RTT between a client and any other host in the institutional network is negligible.)

P21. Suppose that your department has a local DNS server for all computers in the department. You are an ordinary user (i.e., not a network/system administrator). Can you determine if an external Web site was likely accessed from a computer in your department a couple of seconds ago? Explain.

P22. Consider distributing a file of $F = 15$ Gbits to N peers. The server has an upload rate of $u_s = 30$ Mbps, and each peer has a download rate of $d_i = 2$ Mbps and an upload rate of u. For $N = 10$, 100, and 1,000 and $u = 300$ Kbps, 700 Kbps, and 2 Mbps, prepare a chart giving the minimum distribution time for each of the combinations of N and u for both client-server distribution and P2P distribution.

P23. Consider distributing a file of F bits to N peers using a client-server architecture. Assume a fluid model where the server can simultaneously transmit to multiple peers, transmitting to each peer at different rates, as long as the combined rate does not exceed u_s.

a. Suppose that $u_s/N \leq d_{\min}$. Specify a distribution scheme that has a distribution time of NF/u_s.

b. Suppose that $u_s/N \geq d_{\min}$. Specify a distribution scheme that has a distribution time of $F/d_{\min}$.

c. Conclude that the minimum distribution time is in general given by $\max\{NF/u_s, F/d_{\min}\}$.

P24. Consider distributing a file of F bits to N peers using a P2P architecture. Assume a fluid model. For simplicity assume that $d_{\min}$ is very large, so that peer download bandwidth is never a bottleneck.

a. Suppose that $u_s \leq (u_s + u_1 + \ldots + u_N)/N$. Specify a distribution scheme that has a distribution time of F/u_s.

b. Suppose that $u_s \geq (u_s + u_1 + \ldots + u_N)/N$. Specify a distribution scheme that has a distribution time of $NF/(u_s + u_1 + \ldots + u_N)$.

c. Conclude that the minimum distribution time is in general given by $\max\{F/u_s, NF/(u_s + u_1 + \ldots + u_N)\}$.

P25. Suppose Bob joins a BitTorrent torrent, but he does not want to upload any data to any other peers (so called free-riding).

a. Bob claims that he can receive a complete copy of the file that is shared by the swarm. Is Bob's claim possible? Why or why not?

b. Bob further claims that he can further make his "free-riding" more efficient by using a collection of multiple computers (with distinct IP addresses) in the computer lab in his department. How can he do that?

P26. Consider a DASH system for which there are N video versions (at N different rates and qualities) and N audio versions (at N different rates and qualities). Suppose we want to allow the player to choose at any time any of the N video versions and any of the N audio versions.

a. If we create files so that the audio is mixed in with the video, so server sends only one media stream at given time, how many files will the server need to store (each a different URL)?

b. If the server instead sends the audio and video streams separately and has the client synchronize the streams, how many files will the server need to store?

P27. Install and compile the Python programs TCPClient and UDPClient on one host and TCPServer and UDPServer on another host.

a. Suppose you run TCPClient before you run TCPServer. What happens? Why?

b. Suppose you run UDPClient before you run UDPServer. What happens? Why?

c. What happens if you use different port numbers for the client and server sides?

P28. Suppose that in UDPClient.py, after we create the socket, we add the line:

```
clientSocket.bind(('', 5432))
```

Will it become necessary to change UDPServer.py? What are the port numbers for the sockets in UDPClient and UDPServer? What were they before making this change?

P29. Can you configure your browser to open multiple simultaneous connections to a Web site? What are the advantages and disadvantages of having a large number of simultaneous TCP connections?

P30. We have seen that Internet TCP sockets treat the data being sent as a byte stream but UDP sockets recognize message boundaries. What are one advantage and one disadvantage of byte-oriented API versus having the API explicitly recognize and preserve application-defined message boundaries?

P31. What is the server placement strategy adopted by Netflix for its CDN? How is content replicated at the different servers?

Socket Programming Assignments

The Companion Website includes six socket programming assignments. The first four assignments are summarized below. The fifth assignment makes use of the ICMP protocol and is summarized at the end of Chapter 5. The sixth assignment employs multimedia protocols and is summarized at the end of Chapter 9. It is highly recommended that students complete several, if not all, of these assignments. Students can find full details of these assignments, as well as important snippets of the Python code, at the Web site www.pearsonglobaleditions.com/kurose.

Assignment 1: Web Server

In this assignment, you will develop a simple Web server in Python that is capable of processing only one request. Specifically, your Web server will (i) create a connection socket when contacted by a client (browser); (ii) receive the HTTP request from this connection; (iii) parse the request to determine the specific file being requested; (iv) get the requested file from the server's file system; (v) create an HTTP response message consisting of the requested file preceded by header lines; and (vi) send the response over the TCP connection to the requesting browser. If a browser requests a file that is not present in your server, your server should return a "404 Not Found" error message.

In the Companion Website, we provide the skeleton code for your server. Your job is to complete the code, run your server, and then test your server by sending requests from browsers running on different hosts. If you run your server on a host that already has a Web server running on it, then you should use a different port than port 80 for your Web server.

Assignment 2: UDP Pinger

In this programming assignment, you will write a client ping program in Python. Your client will send a simple ping message to a server, receive a corresponding pong message back from the server, and determine the delay between when the client sent the ping message and received the pong message. This delay is called the Round Trip Time (RTT). The functionality provided by the client and server is similar to the functionality provided by standard ping program available in modern operating systems. However, standard ping programs use the Internet Control Message Protocol (ICMP) (which we will study in Chapter 5). Here we will create a nonstandard (but simple!) UDP-based ping program.

Your ping program is to send 10 ping messages to the target server over UDP. For each message, your client is to determine and print the RTT when the corresponding pong message is returned. Because UDP is an unreliable protocol, a packet sent by the client or server may be lost. For this reason, the client cannot wait indefinitely for a reply to a ping message. You should have the client wait up to one second for a reply from the server; if no reply is received, the client should assume that the packet was lost and print a message accordingly.

In this assignment, you will be given the complete code for the server (available in the Companion Website). Your job is to write the client code, which will be very similar to the server code. It is recommended that you first study carefully the server code. You can then write your client code, liberally cutting and pasting lines from the server code.

Assignment 3: Mail Client

The goal of this programming assignment is to create a simple mail client that sends e-mail to any recipient. Your client will need to establish a TCP connection with a mail server (e.g., a Google mail server), dialogue with the mail server using the SMTP protocol, send an e-mail message to a recipient (e.g., your friend) via the mail server, and finally close the TCP connection with the mail server.

For this assignment, the Companion Website provides the skeleton code for your client. Your job is to complete the code and test your client by sending e-mail to different user accounts. You may also try sending through different servers (for example, through a Google mail server and through your university mail server).

Assignment 4: Multi-Threaded Web Proxy

In this assignment, you will develop a Web proxy. When your proxy receives an HTTP request for an object from a browser, it generates a new HTTP request for the same object and sends it to the origin server. When the proxy receives the corresponding HTTP response with the object from the origin server, it creates a new HTTP response, including the object, and sends it to the client. This proxy will be multi-threaded, so that it will be able to handle multiple requests at the same time.

For this assignment, the Companion Website provides the skeleton code for the proxy server. Your job is to complete the code, and then test it by having different browsers request Web objects via your proxy.

Wireshark Lab: HTTP

Having gotten our feet wet with the Wireshark packet sniffer in Lab 1, we're now ready to use Wireshark to investigate protocols in operation. In this lab, we'll explore several aspects of the HTTP protocol: the basic GET/reply interaction, HTTP message formats, retrieving large HTML files, retrieving HTML files with embedded URLs, persistent and non-persistent connections, and HTTP authentication and security.

As is the case with all Wireshark labs, the full description of this lab is available at this book's Web site, www.pearsonglobaleditions.com/kurose.

Wireshark Lab: DNS

In this lab, we take a closer look at the client side of the DNS, the protocol that translates Internet hostnames to IP addresses. Recall from Section 2.5 that the client's role in the DNS is relatively simple—a client sends a query to its local DNS server and receives a response back. Much can go on under the covers, invisible to the DNS clients, as the hierarchical DNS servers communicate with each other to either recursively or iteratively resolve the client's DNS query. From the DNS client's standpoint, however, the protocol is quite simple—a query is formulated to the local DNS server and a response is received from that server. We observe DNS in action in this lab.

As is the case with all Wireshark labs, the full description of this lab is available at this book's Web site, www.pearsonglobaleditions.com/kurose.

and ac... ...e segments? ... the sequence number

SECTION 3.7

R17. Consider two hosts, Host A and Host B, transmitting a large file to Server C over a bottleneck link with a rate of R kbps. To transfer the file, the hosts use TCP with the same parameters (including MSS and RTT) and start their transmissions at the same time. Host A uses a single TCP connection for the entire file, while Host B uses 9 simultaneous TCP connections, each for a portion (i.e., a *chunk*) of the file. What is the overall transmission rate achieved by each host at the beginning of the file transfer? (*Hint:* the overall transmission rate of a host is the sum of the transmission rates of its TCP connections.) Is this situation fair?

R18. True or false? Consider congestion control in TCP. When the timer expires at the sender, the value of `ssthresh` is set to one half of its previous value.

R19. According to the discussion of TCP splitting in the sidebar in Section 3.7, the response time with TCP splitting is approximately $4 \times RTT_{FE} + RTT_{BE} +$ processing time, as opposed to $4 \times RTT +$ processing time when a direct connection is used. Assume that RTT BE is $0.5 \times RTT$. For what values of RTT_{FE} does TCP splitting have a shorter delay than a direct connection?

Problems

P1. Suppose Client A requests a web page from Server S through HTTP and its socket is associated with port 33000.

a. What are the source and destination ports for the segments sent from A to S?

b. What are the source and destination ports for the segments sent from S to A?

c. Can Client A contact to Server S using UDP as the transport protocol?

d. Can Client A request multiple resources in a single TCP connection?

P2. Consider Figure 3.5. What are the source and destination port values in the segments flowing from the server back to the clients' processes? What are the IP addresses in the network-layer datagrams carrying the transport-layer segments?

P3. UDP and TCP use 1s complement for their checksums. Suppose you have the following three 8-bit bytes: 01010011, 01100110, 01110100. What is the 1s complement of the sum of these 8-bit bytes? (Note that although UDP and TCP use 16-bit words in computing the checksum, for this problem you are being asked to consider 8-bit sums.) Show all work. Why is it that UDP takes the 1s complement of the sum; that is, why not just use the sum? With the 1s complement scheme, how does the receiver detect errors? Is it possible that a 1-bit error will go undetected? How about a 2-bit error?

P4. Assume that a host receives a UDP segment with 01011101 11110010 (we separated the values of each byte with a space for clarity) as the checksum. The host adds the 16-bit words over all necessary fields *excluding the checksum* and obtains the value 00110010 00001101. Is the segment considered correctly received or not? What does the receiver do?

P5. Suppose that the UDP receiver computes the Internet checksum for the received UDP segment and finds that it matches the value carried in the checksum field. Can the receiver be absolutely certain that no bit errors have occurred? Explain.

P6. Consider our motivation for correcting protocol `rdt2.1`. Show that the receiver, shown in Figure 3.57, when operating with the sender shown in Figure 3.11, can lead the sender and receiver to enter into a deadlock state, where each is waiting for an event that will never occur.

P7. In protocol `rdt3.0`, the ACK packets flowing from the receiver to the sender do not have sequence numbers (although they do have an ACK field that contains the sequence number of the packet they are acknowledging). Why is it that our ACK packets do not require sequence numbers?

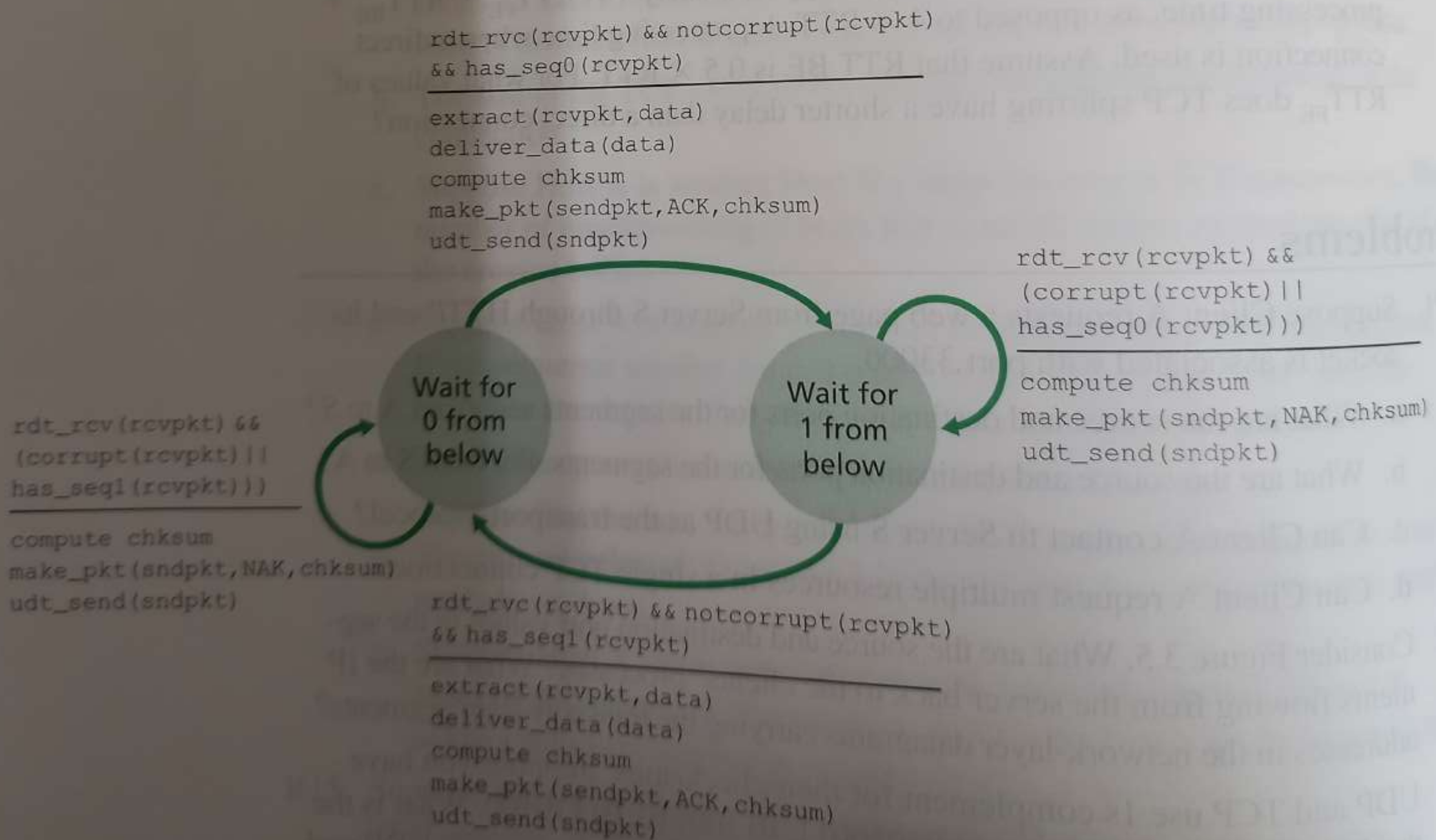


Figure 3.57 ♦ An incorrect receiver for protocol `rdt 2.1`

P8. Draw the FSM for the receiver side of protocol `rdt3.0`.

P9. Give a trace of the operation of protocol `rdt3.0` when data packets and acknowledgment packets are garbled. Your trace should be similar to that used in Figure 3.16.

P10. Consider a channel that can lose packets but has a maximum delay that is known. Modify protocol `rdt2.1` to include sender timeout and retransmit. Informally argue why your protocol can communicate correctly over this channel.

P11. Consider the `rdt2.2` receiver in Figure 3.14, and the creation of a new packet in the self-transition (i.e., the transition from the state back to itself) in the Wait-for-0-from-below and the Wait-for-1-from-below states: `sndpkt=make_pkt(ACK,1,checksum)` and `sndpkt=make_pkt(ACK,0,checksum)`. Would the protocol work correctly if this action were removed from the self-transition in the Wait-for-1-from-below state? Justify your answer. What if this event were removed from the self-transition in the Wait-for-0-from-below state? [*Hint*: In this latter case, consider what would happen if the first sender-to-receiver packet were corrupted.]

P12. The sender side of `rdt3.0` simply ignores (that is, takes no action on) all received packets that are either in error or have the wrong value in the `acknum` field of an acknowledgment packet. Suppose that in such circumstances, `rdt3.0` were simply to retransmit the current data packet. Would the protocol still work? (*Hint*: Consider what would happen if there were only bit errors; there are no packet losses but premature timeouts can occur. Consider how many times the *n*th packet is sent, in the limit as *n* approaches infinity.)

P13. Assume Host A is streaming a video from Server B using UDP. Also assume that the network suddenly becomes very congested while Host A is seeing the video. Is there any way to handle this situation with UDP? What about with TCP? Is there any other option?

P14. Consider a stop-and-wait data-transfer protocol that provides error checking and retransmissions but uses only negative acknowledgments. Assume that negative acknowledgments are never corrupted. Would such a protocol work over a channel with bit errors? What about over a lossy channel with bit errors?

P15. Consider the cross-country example shown in Figure 3.17. How big would the window size have to be for the channel utilization to be greater than 98 percent? Suppose that the size of a packet is 1,500 bytes, including both header fields and data.

P16. Suppose an application uses `rdt 3.0` as its transport layer protocol. As the stop-and-wait protocol has very low channel utilization (shown in the cross-country example), the designers of this application let the receiver keep sending back a number (more than two) of alternating ACK 0 and ACK 1 even if the corresponding data have not arrived at the receiver. Would this application design increase the channel utilization? Why? Are there any potential problems with this approach? Explain.

P17. Consider two network entities, A and B, which are connected by a perfect bi-directional channel (i.e., any message sent will be received correctly; the channel will not corrupt, lose, or re-order packets). A and B are to deliver data messages to each other in an alternating manner: First, A must deliver a message to B, then B must deliver a message to A, then A must deliver a message to B and so on. If an entity is in a state where it should not attempt to deliver a message to the other side, and there is an event like `rdt_send(data)` call from above that attempts to pass data down for transmission to the other side, this call from above can simply be ignored with a call to `rdt_unable_to_send(data)`, which informs the higher layer that it is currently not able to send data. [Note: This simplifying assumption is made so you don't have to worry about buffering data.]

Draw a FSM specification for this protocol (one FSM for A, and one FSM for B!). Note that you do not have to worry about a reliability mechanism here; the main point of this question is to create a FSM specification that reflects the synchronized behavior of the two entities. You should use the following events and actions that have the same meaning as protocol rdt1.0 in Figure 3.9: `rdt_send(data)`, `packet = make_pkt(data)`, `udt_send(packet)`, `rdt_rcv(packet)`, `extract (packet,data)`, `deliver_data(data)`. Make sure your protocol reflects the strict alternation of sending between A and B. Also, make sure to indicate the initial states for A and B in your FSM descriptions.

P18. In the generic SR protocol that we studied in Section 3.4.4, the sender transmits a message as soon as it is available (if it is in the window) without waiting for an acknowledgment. Suppose now that we want an SR protocol that sends messages two at a time. That is, the sender will send a pair of messages and will send the next pair of messages only when it knows that both messages in the first pair have been received correctly.

Suppose that the channel may lose messages but will not corrupt or reorder messages. Design an error-control protocol for the unidirectional reliable transfer of messages. Give an FSM description of the sender and receiver. Describe the format of the packets sent between sender and receiver, and vice versa. If you use any procedure calls other than those in Section 3.4 (for example, `udt_send()`, `start_timer()`, `rdt_rcv()`, and so on), clearly state their actions. Give an example (a timeline trace of sender and receiver) showing how your protocol recovers from a lost packet.

P19. Suppose Host A and Host B use a GBN protocol with window size $N = 3$ and a long-enough range of sequence numbers. Assume Host A sends six application messages to Host B and that all messages are correctly received, except for the first acknowledgment and the fifth data segment. Draw a timing diagram (similar to Figure 3.22), showing the data segments and the acknowledgments sent along with the corresponding sequence and acknowledge numbers, respectively.

P20. Consider a scenario in which Host A and Host B want to send messages to Host C. Hosts A and C are connected by a channel that can lose and corrupt (but not reorder) messages. Hosts B and C are connected by another channel (independent of the channel connecting A and C) with the same properties. The transport layer at Host C should alternate in delivering messages from A and B to the layer above (that is, it should first deliver the data from a packet from A, then the data from a packet from B, and so on). Design a stop-and-wait-like error-control protocol for reliably transferring packets from A and B to C, with alternating delivery at C as described above. Give FSM descriptions of A and C. (*Hint:* The FSM for B should be essentially the same as for A.) Also, give a description of the packet format(s) used.

P21. Suppose we have two network entities, A and B. B has a supply of data messages that will be sent to A according to the following conventions. When A gets a request from the layer above to get the next data (D) message from B, A must send a request (R) message to B on the A-to-B channel. Only when B receives an R message can it send a data (D) message back to A on the B-to-A channel. A should deliver exactly one copy of each D message to the layer above. R messages can be lost (but not corrupted) in the A-to-B channel; D messages, once sent, are always delivered correctly. The delay along both channels is unknown and variable.

Design (give an FSM description of) a protocol that incorporates the appropriate mechanisms to compensate for the loss-prone A-to-B channel and implements message passing to the layer above at entity A, as discussed above. Use only those mechanisms that are absolutely necessary.

P22. Consider the GBN protocol with a sender window size of 4 and a sequence number range of 1,024. Suppose that at time t, the next in-order packet that the receiver is expecting has a sequence number of k. Assume that the medium does not reorder messages. Answer the following questions:

a. What are the possible sets of sequence numbers inside the sender's window at time t? Justify your answer.

b. What are all possible values of the ACK field in all possible messages currently propagating back to the sender at time t? Justify your answer.

P23. Give one example where buffering out-of-order segments would significantly improve the throughput of a GBN protocol.

P24. Consider a scenario where Host A, Host B, and Host C are connected as a ring (i.e., Host A to Host B, Host B to Host C, and Host C to Host A). Assume that Host A and Host C run protocol `rdt3.0`, while Host B simply relays all messages received from Host A to Host C. Does this arrangement enable reliable delivery of messages from Host A to Host C? Can Host B tell if a certain message has been correctly received by Host A?

P25. Consider the Telnet case study in Section 3.5.2. Assume a Telnet session is already active between Host A and Server S. The user at Host A then types the word "Hello."

a. How many TCP segments will be created at the transport layer of Host A?

b. Is there any guarantee that each segment will be sent into the TCP connection as soon as it is created?

c. Does TCP provide any mechanism that can be useful for an interactive Telnet session?

d. Would UDP offer a viable alternative to TCP for Telnet sessions over a reliable channel?

P26. Consider transferring an enormous file of L bytes from Host A to Host B. Assume an MSS of 536 bytes.

a. What is the maximum value of L such that TCP sequence numbers are not exhausted? Recall that the TCP sequence number field has 4 bytes.

b. For the L you obtain in (a), find how long it takes to transmit the file. Assume that a total of 66 bytes of transport, network, and data-link header are added to each segment before the resulting packet is sent out over a 155 Mbps link. Ignore flow control and congestion control so A can pump out the segments back to back and continuously.

P27. Host A and B are communicating over a TCP connection, and Host B has already received from A all bytes up through byte 126. Suppose Host A then sends two segments to Host B back-to-back. The first and second segments contain 80 and 40 bytes of data, respectively. In the first segment, the sequence number is 127, the source port number is 302, and the destination port number is 80. Host B sends an acknowledgment whenever it receives a segment from Host A.

a. In the second segment sent from Host A to B, what are the sequence number, source port number, and destination port number?

b. If the first segment arrives before the second segment, in the acknowledgment of the first arriving segment, what is the acknowledgment number, the source port number, and the destination port number?

c. If the second segment arrives before the first segment, in the acknowledgment of the first arriving segment, what is the acknowledgment number?

d. Suppose the two segments sent by A arrive in order at B. The first acknowledgment is lost and the second acknowledgment arrives after the first timeout interval. Draw a timing diagram, showing these segments and all other segments and acknowledgments sent. (Assume there is no additional packet loss.) For each segment in your figure, provide the sequence number and the number of bytes of data; for each acknowledgment that you add, provide the acknowledgment number.

P28. Host A and B are directly connected with a 100 Mbps link. There is one TCP connection between the two hosts, and Host A is sending to Host B an enormous file over this connection. Host A can send its application data into its TCP socket at a rate as high as 120 Mbps but Host B can read out of its TCP receive buffer at a maximum rate of 50 Mbps. Describe the effect of TCP flow control.

P29. SYN cookies were discussed in Section 3.5.6.

a. Why is it necessary for the server to use a special initial sequence number in the SYNACK?

b. Suppose an attacker knows that a target host uses SYN cookies. Can the attacker create half-open or fully open connections by simply sending an ACK packet to the target? Why or why not?

c. Suppose an attacker collects a large amount of initial sequence numbers sent by the server. Can the attacker cause the server to create many fully open connections by sending ACKs with those initial sequence numbers? Why?

P30. Consider the network shown in Scenario 2 in Section 3.6.1. Suppose both sending hosts A and B have some fixed timeout values.

a. Argue that increasing the size of the finite buffer of the router might possibly decrease the throughput (λ_{out}).

b. Now suppose both hosts dynamically adjust their timeout values (like what TCP does) based on the buffering delay at the router. Would increasing the buffer size help to increase the throughput? Why?

P31. Suppose that the five measured `SampleRTT` values (see Section 3.5.3) are 106 ms, 120 ms, 140 ms, 90 ms, and 115 ms. Compute the `EstimatedRTT` after each of these SampleRTT values is obtained, using a value of $\alpha = 0.125$ and assuming that the value of `EstimatedRTT` was 100 ms just before the first of these five samples were obtained. Compute also the `DevRTT` after each sample is obtained, assuming a value of $\beta = 0.25$ and assuming the value of `DevRTT` was 5 ms just before the first of these five samples was obtained. Last, compute the TCP `TimeoutInterval` after each of these samples is obtained.

P32. Consider the TCP procedure for estimating RTT. Suppose that $\alpha = 0.1$. Let $\texttt{SampleRTT}_1$ be the most recent sample RTT, let $\texttt{SampleRTT}_2$ be the next most recent sample RTT, and so on.

a. For a given TCP connection, suppose four acknowledgments have been returned with corresponding sample RTTs: $\texttt{SampleRTT}_4$, $\texttt{SampleRTT}_3$, $\texttt{SampleRTT}_2$, and $\texttt{SampleRTT}_1$. Express `EstimatedRTT` in terms of the four sample RTTs.

b. Generalize your formula for n sample RTTs.

c. For the formula in part (b) let n approach infinity. Comment on why this averaging procedure is called an exponential moving average.

P33. In Section 3.5.3, we discussed TCP's estimation of RTT. Why do you think TCP avoids measuring the `SampleRTT` for retransmitted segments?

P34. What is the relationship between the variable `SendBase` in Section 3.5.4 and the variable `LastByteRcvd` in Section 3.5.5?

P35. What is the relationship between the variable `LastByteRcvd` in Section 3.5.5 and the variable y in Section 3.5.4?

P36. In Section 3.5.4, we saw that TCP waits until it has received three duplicate ACKs before performing a fast retransmit. Why do you think the TCP designers chose not to perform a fast retransmit after the first duplicate ACK for a segment is received?

P37. Compare GBN, SR, and TCP (no delayed ACK). Assume that the timeout values for all three protocols are sufficiently long such that 5 consecutive data segments and their corresponding ACKs can be received (if not lost in the channel) by the receiving host (Host B) and the sending host (Host A) respectively. Suppose Host A sends 5 data segments to Host B, and the 2nd segment (sent from A) is lost. In the end, all 5 data segments have been correctly received by Host B.

a. How many segments has Host A sent in total and how many ACKs has Host B sent in total? What are their sequence numbers? Answer this question for all three protocols.

b. If the timeout values for all three protocol are much longer than 5 RTT, then which protocol successfully delivers all five data segments in shortest time interval?

P38. In our description of TCP in Figure 3.53, the value of the threshold, `ssthresh`, is set as `ssthresh=cwnd/2` in several places and `ssthresh` value is referred to as being set to half the window size when a loss event occurred. Must the rate at which the sender is sending when the loss event occurred be approximately equal to `cwnd` segments per RTT? Explain your answer. If your answer is no, can you suggest a different manner in which `ssthresh` should be set?

P39. Consider Figure 3.46(b). If λ'_{in} increases beyond $R/2$, can λ_{out} increase beyond $R/3$? Explain. Now consider Figure 3.46(c). If λ'_{in} increases beyond $R/2$, can λ_{out} increase beyond R/4 under the assumption that a packet will be forwarded twice on average from the router to the receiver? Explain.

P40. Consider Figure 3.58. Assuming TCP Reno is the protocol experiencing the behavior shown above, answer the following questions. In all cases, you should provide a short discussion justifying your answer.

VideoNote
Examining the behavior of TCP

a. Identify the intervals of time when TCP slow start is operating.

b. Identify the intervals of time when TCP congestion avoidance is operating.

c. After the 16th transmission round, is segment loss detected by a triple duplicate ACK or by a timeout?

d. After the 22nd transmission round, is segment loss detected by a triple duplicate ACK or by a timeout?

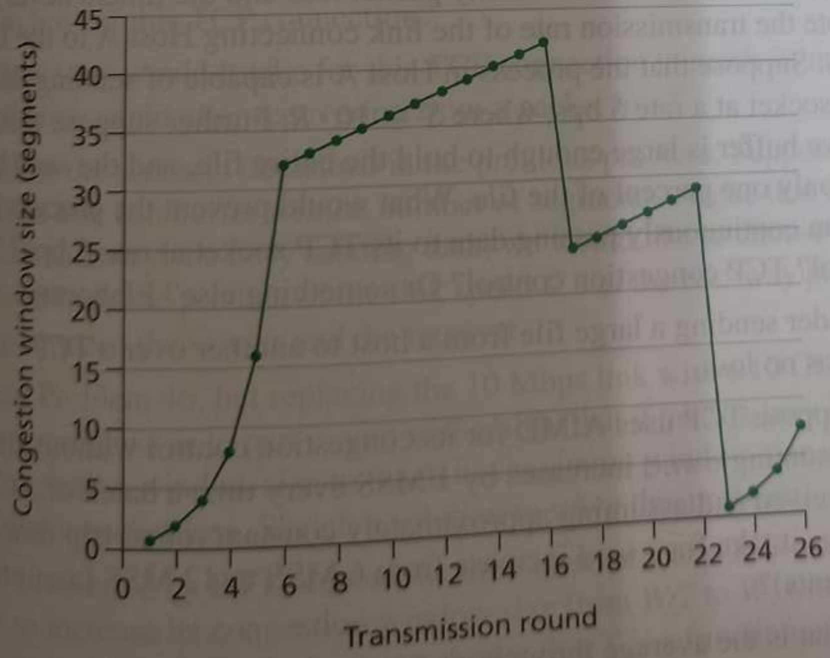


Figure 3.58 ♦ TCP window size as a function of time

e. What is the initial value of `ssthresh` at the first transmission round?
f. What is the value of `ssthresh` at the 18th transmission round?
g. What is the value of `ssthresh` at the 24th transmission round?
h. During what transmission round is the 70th segment sent?
i. Assuming a packet loss is detected after the 26th round by the receipt of a triple duplicate ACK, what will be the values of the congestion window size and of `ssthresh`?
j. Suppose TCP Tahoe is used (instead of TCP Reno), and assume that triple duplicate ACKs are received at the 16th round. What are the `ssthresh` and the congestion window size at the 19th round?
k. Again suppose TCP Tahoe is used, and there is a timeout event at 22nd round. How many packets have been sent out from 17th round till 22nd round, inclusive?

P41. Refer to Figure 3.55, which illustrates the convergence of TCP's AIMD algorithm. Suppose that instead of a multiplicative decrease, TCP decreased the window size by a constant amount. Would the resulting AIAD algorithm converge to an equal share algorithm? Justify your answer using a diagram similar to Figure 3.55.

P42. In Section 3.5.4, we discussed the doubling of the timeout interval after a timeout event. This mechanism is a form of congestion control. Why does TCP need a window-based congestion-control mechanism (as studied in Section 3.7) in addition to this doubling-timeout-interval mechanism?

P43. Host A is sending an enormous file to Host B over a TCP connection. Over this connection there is never any packet loss and the timers never expire. Denote the transmission rate of the link connecting Host A to the Internet by R bps. Suppose that the process in Host A is capable of sending data into its TCP socket at a rate S bps, where $S = 10 \cdot R$. Further suppose that the TCP receive buffer is large enough to hold the entire file, and the send buffer can hold only one percent of the file. What would prevent the process in Host A from continuously passing data to its TCP socket at rate S bps? TCP flow control? TCP congestion control? Or something else? Elaborate.

P44. Consider sending a large file from a host to another over a TCP connection that has no loss.

a. Suppose TCP uses AIMD for its congestion control without slow start. Assuming `cwnd` increases by 1 MSS every time a batch of ACKs is received and assuming approximately constant round-trip times, how long does it take for `cwnd` increase from 6 MSS to 12 MSS (assuming no loss events)?
b. What is the average throughout (in terms of MSS and RTT) for this connection up through time = 6 RTT?

P45. Recall the macroscopic description of TCP throughput. In the period of time from when the connection's rate varies from $W/(2 \cdot RTT)$ to W/RTT, only one packet is lost (at the very end of the period).

a. Show that the loss rate (fraction of packets lost) is equal to

$$L = \text{loss rate} = \frac{1}{\frac{3}{8}W^2 + \frac{3}{4}W}$$

b. Use the result above to show that if a connection has loss rate L, then its average rate is approximately given by

$$\approx \frac{1.22 \cdot MSS}{RTT\sqrt{L}}$$

P46. Consider that only a single TCP (Reno) connection uses one 10Mbps link which does not buffer any data. Suppose that this link is the only congested link between the sending and receiving hosts. Assume that the TCP sender has a huge file to send to the receiver, and the receiver's receive buffer is much larger than the congestion window. We also make the following assumptions: each TCP segment size is 1,500 bytes; the two-way propagation delay of this connection is 150 msec; and this TCP connection is always in congestion avoidance phase, that is, ignore slow start.

a. What is the maximum window size (in segments) that this TCP connection can achieve?

b. What is the average window size (in segments) and average throughput (in bps) of this TCP connection?

c. How long would it take for this TCP connection to reach its maximum window again after recovering from a packet loss?

P47. Consider the scenario described in the previous problem. Suppose that the 10Mbps link can buffer a finite number of segments. Argue that in order for the link to always be busy sending data, we would like to choose a buffer size that is at least the product of the link speed C and the two-way propagation delay between the sender and the receiver.

P48. Repeat Problem 46, but replacing the 10 Mbps link with a 10 Gbps link. Note that in your answer to part c, you will realize that it takes a very long time for the congestion window size to reach its maximum window size after recovering from a packet loss. Sketch a solution to solve this problem.

P49. Let T (measured by RTT) denote the time interval that a TCP connection takes to increase its congestion window size from $W/2$ to W, where W is the maximum congestion window size. Argue that T is a function of TCP's average throughput.

P50. Consider a simplified TCP's AIMD algorithm where the congestion window size is measured in number of segments, not in bytes. In additive increase, the congestion window size increases by one segment in each RTT. In multiplicative decrease, the congestion window size decreases by half (if the result is not an integer, round down to the nearest integer). Suppose that two TCP connections, C_1 and C_2, share a single congested link of speed 30 segments per second. Assume that both C_1 and C_2 are in the congestion avoidance phase. Connection C_1's RTT is 50 msec and connection C_2's RTT is 100 msec. Assume that when the data rate in the link exceeds the link's speed, all TCP connections experience data segment loss.

a. If both C_1 and C_2 at time t_0 have a congestion window of 10 segments, what are their congestion window sizes after 1000 msec?

b. In the long run, will these two connections get the same share of the bandwidth of the congested link? Explain.

P51. Consider the network described in the previous problem. Now suppose that the two TCP connections, C1 and C2, have the same RTT of 100 msec. Suppose that at time t_0, C1's congestion window size is 15 segments but C2's congestion window size is 10 segments.

a. What are their congestion window sizes after 2200 msec?

b. In the long run, will these two connections get about the same share of the bandwidth of the congested link?

c. We say that two connections are synchronized, if both connections reach their maximum window sizes at the same time and reach their minimum window sizes at the same time. In the long run, will these two connections get synchronized eventually? If so, what are their maximum window sizes?

d. Will this synchronization help to improve the utilization of the shared link? Why? Sketch some idea to break this synchronization.

P52. Consider a modification to TCP's congestion control algorithm. Instead of additive increase, we can use multiplicative increase. A TCP sender increases its window size by a small positive constant a ($0 < a < 1$) whenever it receives a valid ACK. Find the functional relationship between loss rate L and maximum congestion window W. Argue that for this modified TCP, regardless of TCP's average throughput, a TCP connection always spends the same amount of time to increase its congestion window size from $W/2$ to W.

P53. In our discussion of TCP futures in Section 3.7, we noted that to achieve a throughput of 10 Gbps, TCP could only tolerate a segment loss probability of $2 \cdot 10^{-10}$ (or equivalently, one loss event for every 5,000,000,000 segments). Show the derivation for the values of $2 \cdot 10^{-10}$ (1 out of 5,000,000) for the RTT and MSS values given in Section 3.7. If TCP needed to support a 100 Gbps connection, what would the tolerable loss be?

P54. In our discussion of TCP congestion control in Section 3.7, we implicitly assumed that the TCP sender always had data to send. Consider now the case that the TCP sender sends a large amount of data and then goes idle (since it has no more data to send) at t_1. TCP remains idle for a relatively long period of time and then wants to send more data at t_2. What are the advantages and disadvantages of having TCP use the `cwnd` and `ssthresh` values from t_1 when starting to send data at t_2? What alternative would you recommend? Why?

P55. In this problem we investigate whether either UDP or TCP provides a degree of end-point authentication.

a. Consider a server that receives a request within a UDP packet and responds to that request within a UDP packet (for example, as done by a DNS server). If a client with IP address X spoofs its address with address Y, where will the server send its response?

b. Suppose a server receives a SYN with IP source address Y, and after responding with a SYNACK, receives an ACK with IP source address Y with the correct acknowledgment number. Assuming the server chooses a random initial sequence number and there is no "man-in-the-middle," can the server be certain that the client is indeed at Y (and not at some other address X that is spoofing Y)?

P56. In this problem, we consider the delay introduced by the TCP slow-start phase. Consider a client and a Web server directly connected by one link of rate R. Suppose the client wants to retrieve an object whose size is exactly equal to 15 S, where S is the maximum segment size (MSS). Denote the round-trip time between client and server as RTT (assumed to be constant). Ignoring protocol headers, determine the time to retrieve the object (including TCP connection establishment) when

a. $4\,S/R > S/R + RTT > 2S/R$

b. $S/R + RTT > 4\,S/R$

c. $S/R > RTT$.

Programming Assignments

Implementing a Reliable Transport Protocol

In this laboratory programming assignment, you will be writing the sending and receiving ... menting a simple reliable data transfer pro- ... version and the

R30. Compare and contrast the IPv4 and the IPv6 header fields. Do they have any fields in common?

R31. It has been said that when IPv6 tunnels through IPv4 routers, IPv6 treats the IPv4 tunnels as link-layer protocols. Do you agree with this statement? Why or why not?

SECTION 4.4

R32. How does generalized forwarding differ from destination-based forwarding?

R33. What is the difference between a forwarding table that we encountered in destination-based forwarding in Section 4.1 and OpenFlow's flow table that we encountered in Section 4.4?

R34. What is meant by the "match plus action" operation of a router or switch? In the case of destination-based forwarding packet switch, what is matched and what is the action taken? In the case of an SDN, name three fields that can be matched, and three actions that can be taken.

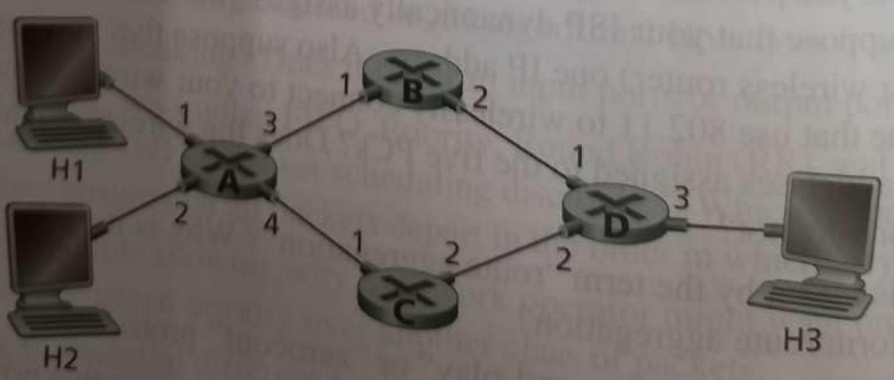
R35. Name three header fields in an IP datagram that can be "matched" in OpenFlow 1.0 generalized forwarding. What are three IP datagram header fields that *cannot* be "matched" in OpenFlow?

Problems

P1. Consider the network below.

a. Show the forwarding table in router A, such that all traffic destined to host H3 is forwarded through interface 3.

b. Can you write down a forwarding table in router A, such that all traffic from H1 destined to host H3 is forwarded through interface 3, while all traffic from H2 destined to host H3 is forwarded through interface 4? (*Hint:* This is a trick question.)



P2. Suppose two packets arrive to two different input ports of a router at exactly the same time. Also suppose there are no other packets anywhere in the router.

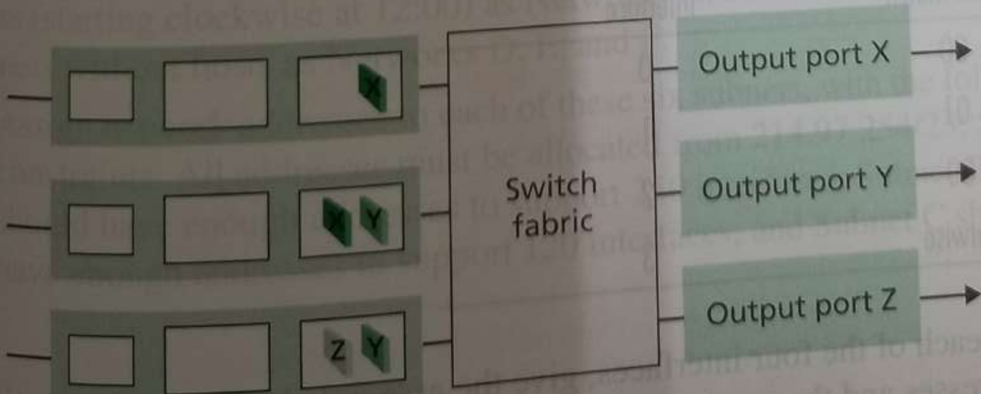
a. Suppose the two packets are to be forwarded to two different output ports. Is it possible to forward the two packets through the switch fabric at the same time when the fabric uses a shared bus?

b. Suppose the two packets are to be forwarded to two different output ports. Is it possible to forward the two packets through the switch fabric at the same time when the fabric uses switching via memory?

c. Suppose the two packets are to be forwarded to the same output port. Is it possible to forward the two packets through the switch fabric at the same time when the fabric uses a crossbar?

P3. In Section 4.2.4 it was said that if *R_switch* is *N* times faster than *R_line*, then only negligible queuing will occur at the input ports, even if all the packets are to be forwarded to the same output port. Now suppose that *R_switch* = *R_line*, but all packets are to be forwarded to different output ports. Let *D* be the time to transmit a packet. As a function of *D*, what is the maximum input queuing delay for a packet for the (a) memory, (b) bus, and (c) crossbar switching fabrics?

P4. Consider the switch shown below. Suppose that all datagrams have the same fixed length, that the switch operates in a slotted, synchronous manner, and that in one time slot a datagram can be transferred from an input port to an output port. The switch fabric is a crossbar so that at most one datagram can be transferred to a given output port in a time slot, but different output ports can receive datagrams from different input ports in a single time slot. What is the minimal number of time slots needed to transfer the packets shown from input ports to their output ports, assuming any input queue scheduling order you want (i.e., it need not have HOL blocking)? What is the largest number of slots needed, assuming the worst-case scheduling order you can devise, assuming that a non-empty input queue is never idle?



P5. Consider a datagram network using 32-bit host addresses. Suppose a router has four links, numbered 0 through 3, and packets are to be forwarded to the link interfaces as follows:

Destination Address Range	Link Interface
11100000 00000000 00000000 00000000 through 11100000 00000000 11111111 11111111	0
11100000 00000001 00000000 00000000 through 11100000 00000001 11111111 11111111	1
11100000 00000010 00000000 00000000 through 11100001 11111111 11111111 11111111	2
otherwise	3

a. Provide a forwarding table that has five entries, uses longest prefix matching, and forwards packets to the correct link interfaces.

b. Describe how your forwarding table determines the appropriate link interface for datagrams with destination addresses:

11111000 10010001 01010001 01010101
11100000 00000000 11000011 00111100
11100001 10000000 00010001 01110111

P6. Consider a datagram network using 8-bit host addresses. Suppose a router uses longest prefix matching and has the following forwarding table:

Prefix Match	Interface
00	0
01	1
100	2
otherwise	3

For each of the four interfaces, give the associated range of destination host addresses and the number of addresses in the range.

P7. Consider a datagram network using 8-bit host addresses. Suppose a router uses longest prefix matching and has the following forwarding table:

Prefix Match	Interface
11	0
101	1
100	2
otherwise	3

For each of the four interfaces, give the associated range of destination host addresses and the number of addresses in the range.

P8. Consider a router that interconnects three subnets: Subnet 1, Subnet 2, and Subnet 3. Suppose all of the interfaces in each of these three subnets are required to have the prefix 223.1.17/24. Also suppose that Subnet 1 is required to support up to 62 interfaces, Subnet 2 is to support up to 106 interfaces, and Subnet 3 is to support up to 15 interfaces. Provide three network addresses (of the form a.b.c.d/x) that satisfy these constraints.

P9. Suppose there are 35 hosts in a subnet. What should the IP address structure look like?

P10. What is the problem of NAT in P2P applications? How can it be avoided? Is there a special name for this solution?

P11. Consider a subnet with prefix 192.168.56.128/26. Give an example of one IP address (of form xxx.xxx.xxx.xxx) that can be assigned to this network. Suppose an ISP owns the block of addresses of the form 192.168.56.32/26. Suppose it wants to create four subnets from this block, with each block having the same number of IP addresses. What are the prefixes (of form a.b.c.d/x) for the four subnets?

P12. Consider the topology shown in Figure 4.20. Denote the three subnets with hosts (starting clockwise at 12:00) as Networks A, B, and C. Denote the subnets without hosts as Networks D, E, and F.

a. Assign network addresses to each of these six subnets, with the following constraints: All addresses must be allocated from 214.97.254/23; Subnet A should have enough addresses to support 250 interfaces; Subnet B should have enough addresses to support 120 interfaces; and Subnet C should have enough addresses to support 120 interfaces. Of course, subnets D, E and F should each be able to support two interfaces. For each subnet, the assignment should take the form a.b.c.d/x or a.b.c.d/x – e.f.g.h/y.

b. Using your answer to part (a), provide the forwarding tables (using longest prefix matching) for each of the three routers.

P13. IPsec has been designed to be backward compatible with IPv4 and IPv6. In particular, in order to reap the benefits of IPsec, we don't need to replace the protocol stacks in all the routers and hosts in the Internet. For example, using the transport mode (one of two IPsec "modes"), if two hosts want to securely communicate, IPsec needs to be available only in those two hosts. Discuss the services provided by an IPsec session.

P14. Consider sending a 1,600-byte datagram into a link that has an MTU of 500 bytes. Suppose the original datagram is stamped with the identification number 291. How many fragments are generated? What are the values in the various fields in the IP datagram(s) generated related to fragmentation?

P15. Suppose datagrams are limited to 1,500 bytes (including header) between source Host A and destination Host B. Assuming a 20-byte IP header, how many datagrams would be required to send an MP3 consisting of 5 million bytes? Explain how you computed your answer.

P16. Consider the network setup in Figure 4.25. Suppose that the ISP instead assigns the router the address 24.34.112.235 and that the network address of the home network is 192.168.1/24.

a. Assign addresses to all interfaces in the home network.

b. Suppose each host has two ongoing TCP connections, all to port 80 at host 128.119.40.86. Provide the six corresponding entries in the NAT translation table.

P17. Suppose you are interested in detecting the number of hosts behind a NAT. You observe that the IP layer stamps an identification number sequentially on each IP packet. The identification number of the first IP packet generated by a host is a random number, and the identification numbers of the subsequent IP packets are sequentially assigned. Assume all IP packets generated by hosts behind the NAT are sent to the outside world.

a. Based on this observation, and assuming you can sniff all packets sent by the NAT to the outside, can you outline a simple technique that detects the number of unique hosts behind a NAT? Justify your answer.

b. If the identification numbers are not sequentially assigned but randomly assigned, would your technique work? Justify your answer.

P18. In this problem we'll explore the impact of NATs on P2P applications. Suppose a peer with username Arnold discovers through querying that a peer with username Bernard has a file it wants to download. Also suppose that Bernard and Arnold are both behind a NAT. Try to devise a technique that will allow Arnold to establish a TCP connection with Bernard without application-specific NAT configuration. If you have difficulty devising such a technique, discuss why.

P19. Consider the SDN OpenFlow network shown in Figure 4.30. Suppose that the desired forwarding behavior for datagrams arriving at s2 is as follows:

- any datagrams arriving on input port 1 from hosts h5 or h6 that are destined to hosts h1 or h2 should be forwarded over output port 2;
- any datagrams arriving on input port 2 from hosts h1 or h2 that are destined to hosts h5 or h6 should be forwarded over output port 1;
- any arriving datagrams on input ports 1 or 2 and destined to hosts h3 or h4 should be delivered to the host specified;
- hosts h3 and h4 should be able to send datagrams to each other.

Specify the flow table entries in s2 that implement this forwarding behavior.

P20. Consider again the SDN OpenFlow network shown in Figure 4.30. Suppose that the desired forwarding behavior for datagrams arriving from hosts h3 or h4 at s2 is as follows:

- any datagrams arriving from host h3 and destined for h1, h2, h5 or h6 should be forwarded in a clockwise direction in the network;
- any datagrams arriving from host h4 and destined for h1, h2, h5 or h6 should be forwarded in a counter-clockwise direction in the network.

Specify the flow table entries in s2 that implement this forwarding behavior.

P21. Consider again the scenario from P19 above. Give the flow tables entries at packet switches s1 and s3, such that any arriving datagrams with a source address of h3 or h4 are routed to the destination hosts specified in the destination address field in the IP datagram. (*Hint:* Your forwarding table rules should include the cases that an arriving datagram is destined for a directly attached host or should be forwarded to a neighboring router for eventual host delivery there.)

P22. Consider again the SDN OpenFlow network shown in Figure 4.30. Suppose we want switch s2 to function as a firewall. Specify the flow table in s2 that implements the following firewall behaviors (specify a different flow table for each of the four firewalling behaviors below) for delivery of datagrams destined to h3 and h4. You do not need to specify the forwarding behavior in s2 that forwards traffic to other routers.

- Only traffic arriving from hosts h1 and h6 should be delivered to hosts h3 or h4 (i.e., that arriving traffic from hosts h2 and h5 is blocked).
- Only TCP traffic is allowed to be delivered to hosts h3 or h4 (i.e., that UDP traffic is blocked).

- Only traffic destined to h3 is to be delivered (i.e., all traffic to h4 is blocked).
- Only UDP traffic from h1 and destined to h3 is to be delivered. All other traffic is blocked.

Wireshark Lab

In the Web site for this textbook, www.pearsonglobaleditions.com/kurose, you'll find a Wireshark lab assignment that examines the operation of the IP protocol, and the IP datagram format in particular.

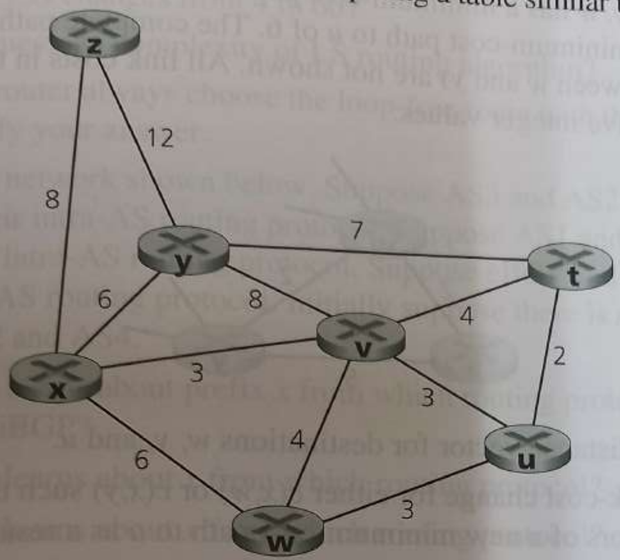
Problems

P1. Looking at Figure 5.3, enumerate the paths from y to u that do not contain any loops.

P2. Repeat Problem P1 for paths from x to z, z to u, and z to w.

P3. Consider the following network. With the indicated link costs, use Dijkstra's shortest-path algorithm to compute the shortest path from x to all network nodes. Show how the algorithm works by computing a table similar to Table 5.1.

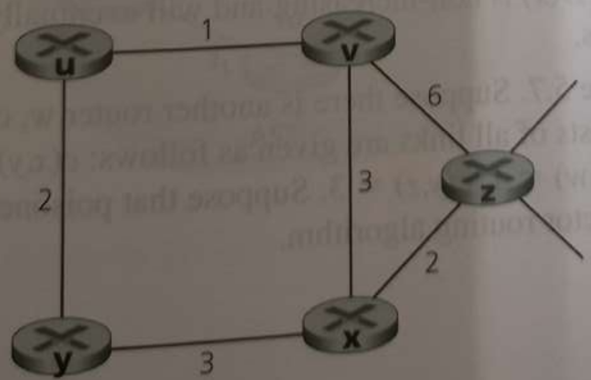
VideoNote
Dijkstra's algorithm: discussion and example



P4. Consider the network shown in Problem P3. Using Dijkstra's algorithm, and showing your work using a table similar to Table 5.1, do the following:

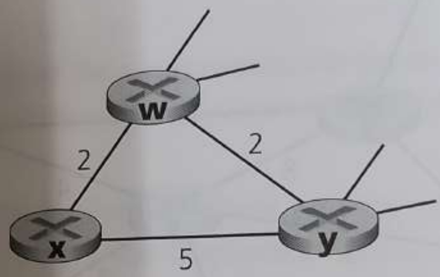
a. Compute the shortest path from t to all network nodes.
b. Compute the shortest path from u to all network nodes.
c. Compute the shortest path from v to all network nodes.
d. Compute the shortest path from w to all network nodes.
e. Compute the shortest path from y to all network nodes.
f. Compute the shortest path from z to all network nodes.

P5. Consider the network shown below, and assume that each node initially knows the costs to each of its neighbors. Consider the distance-vector algorithm and show the distance table entries at node z.



P6. Consider a general topology (that is, not the specific network shown above) and a synchronous version of the distance-vector algorithm. Suppose that at each iteration, a node exchanges its distance vectors with its neighbors and receives their distance vectors. Assuming that the algorithm begins with each node knowing only the costs to its immediate neighbors, what is the maximum number of iterations required before the distributed algorithm converges? Justify your answer.

P7. Consider the network fragment shown below. x has only two attached neighbors, w and y. w has a minimum-cost path to destination u (not shown) of 5, and y has a minimum-cost path to u of 6. The complete paths from w and y to u (and between w and y) are not shown. All link costs in the network have strictly positive integer values.



a. Give x's distance vector for destinations w, y, and u.

b. Give a link-cost change for either $c(x,w)$ or $c(x,y)$ such that x will inform its neighbors of a new minimum-cost path to u as a result of executing the distance-vector algorithm.

c. Give a link-cost change for either $c(x,w)$ or $c(x,y)$ such that x will *not* inform its neighbors of a new minimum-cost path to u as a result of executing the distance-vector algorithm.

P8. Consider the three-node topology shown in Figure 5.6. Rather than having the link costs shown in Figure 5.6, the link costs are $c(x,y) = 3$, $c(y,z) = 6$, $c(z,x) = 4$. Compute the distance tables after the initialization step and after each iteration of a synchronous version of the distance-vector algorithm (as we did in our earlier discussion of Figure 5.6).

P9. Can the *poisoned reverse* solve the general count-to-infinity problem? Justify your answer.

P10. Argue that for the distance-vector algorithm in Figure 5.6, each value in the distance vector $D(x)$ is non-increasing and will eventually stabilize in a finite number of steps.

P11. Consider Figure 5.7. Suppose there is another router w, connected to router y and z. The costs of all links are given as follows: $c(x,y) = 4$, $c(x,z) = 50$, $c(y,w) = 1$, $c(z,w) = 1$, $c(y,z) = 3$. Suppose that poisoned reverse is used in the distance-vector routing algorithm.

a. When the distance vector routing is stabilized, router w, y, and z inform their distances to x to each other. What distance values do they tell each other?

b. Now suppose that the link cost between x and y increases to 60. Will there be a count-to-infinity problem even if poisoned reverse is used? Why or why not? If there is a count-to-infinity problem, then how many iterations are needed for the distance-vector routing to reach a stable state again? Justify your answer.

c. How do you modify $c(y,z)$ such that there is no count-to-infinity problem at all if $c(y,x)$ changes from 4 to 60?

P12. What is the message complexity of LS routing algorithm?

P13. Will a BGP router always choose the loop-free route with the shortest ASpath length? Justify your answer.

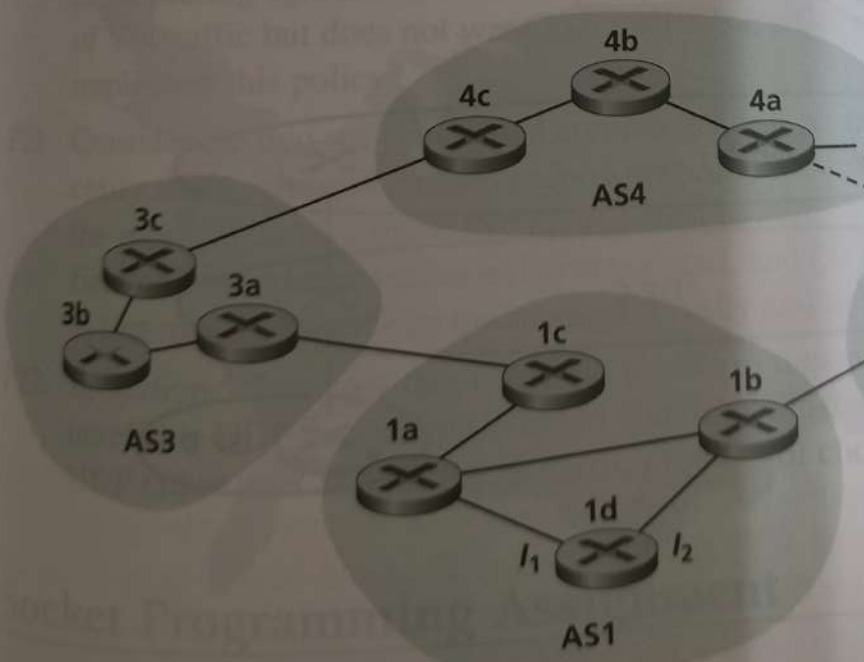
P14. Consider the network shown below. Suppose AS3 and AS2 are running OSPF for their intra-AS routing protocol. Suppose AS1 and AS4 are running RIP for their intra-AS routing protocol. Suppose eBGP and iBGP are used for the inter-AS routing protocol. Initially suppose there is *no* physical link between AS2 and AS4.

a. Router 3c learns about prefix x from which routing protocol: OSPF, RIP, eBGP, or iBGP?

b. Router 3a learns about x from which routing protocol?

c. Router 1c learns about x from which routing protocol?

d. Router 1d learns about x from which routing protocol?



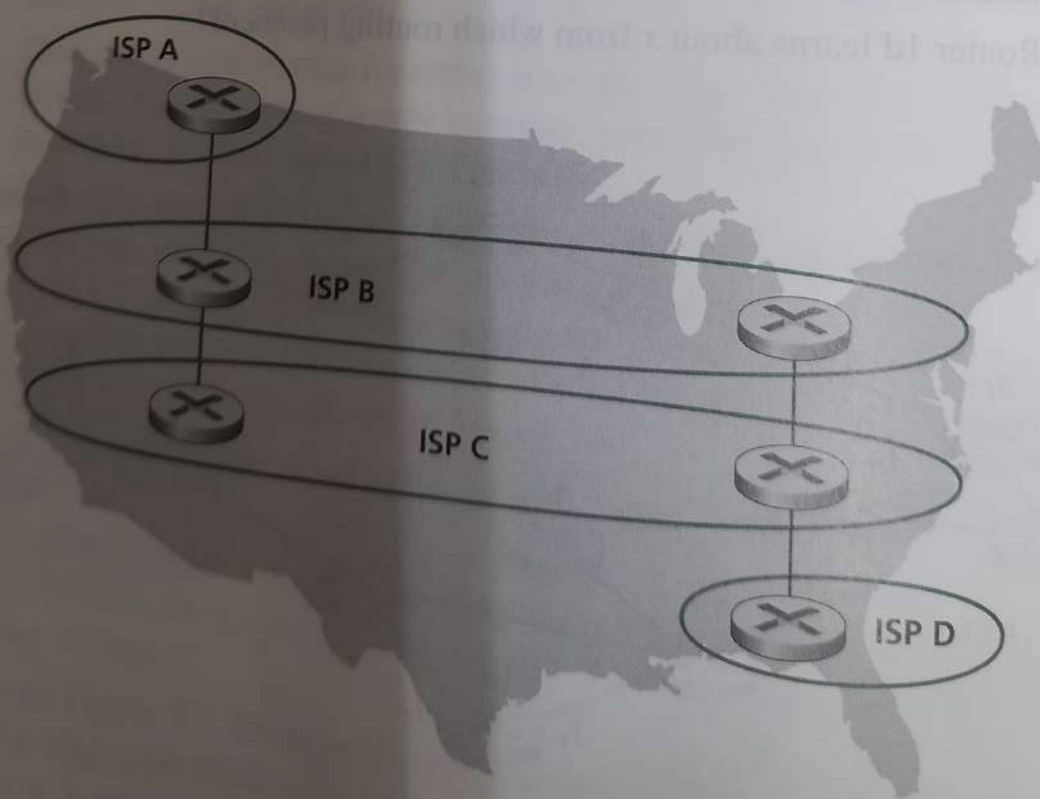
P15. Referring to the previous problem, once router 1d learns about x it will put an entry (x, I) in its forwarding table.

a. Will I be equal to I_1 or I_2 for this entry? Explain why in one sentence.

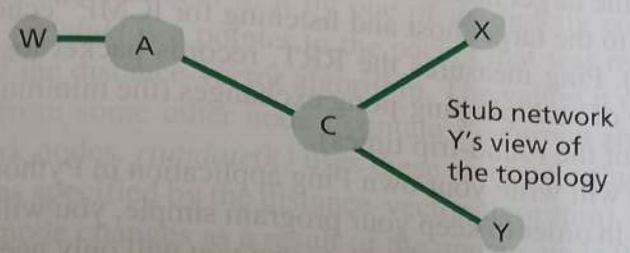
b. Now suppose that there is a physical link between AS2 and AS4, shown by the dotted line. Suppose router 1d learns that x is accessible via AS2 as well as via AS3. Will I be set to I_1 or I_2? Explain why in one sentence.

c. Now suppose there is another AS, called AS5, which lies on the path between AS2 and AS4 (not shown in diagram). Suppose router 1d learns that x is accessible via AS2 AS5 AS4 as well as via AS3 AS4. Will I be set to I_1 or I_2? Explain why in one sentence.

P16. Consider the following network. ISP B provides national backbone service to regional ISP A. ISP C provides national backbone service to regional ISP D. Each ISP consists of one AS. B and C peer with each other in two places using BGP. Consider traffic going from A to D. B would prefer to hand that traffic over to C on the West Coast (so that C would have to absorb the cost of carrying the traffic cross-country), while C would prefer to get the traffic via its East Coast peering point with B (so that B would have carried the traffic across the country). What BGP mechanism might C use, so that B would hand over A-to-D traffic at its East Coast peering point? To answer this question, you will need to dig into the BGP specification.



P17. In Figure 5.13, consider the path information that reaches stub networks W, X, and Y. Based on the information available at W and X, what are their respective views of the network topology? Justify your answer. The topology view at Y is shown below.



P18. Consider Figure 5.13. B would never forward traffic destined to Y via X based on BGP routing. But there are some very popular applications for which data packets go to X first and then flow to Y. Identify one such application, and describe how data packets follow a path not given by BGP routing.

P19. In Figure 5.13, suppose that there is another stub network V that is a customer of ISP A. Suppose that B and C have a peering relationship, and A is a customer of both B and C. Suppose that A would like to have the traffic destined to W to come from B only, and the traffic destined to V from either B or C. How should A advertise its routes to B and C? What AS routes does C receive?

P20. Suppose ASs X and Z are not directly connected but instead are connected by AS Y. Further suppose that X has a peering agreement with Y, and that Y has a peering agreement with Z. Finally, suppose that Z wants to transit all of Y's traffic but does not want to transit X's traffic. Does BGP allow Z to implement this policy?

P21. Consider the two ways in which communication occurs between a managing entity and a managed device: request-response mode and trapping. What are the pros and cons of these two approaches, in terms of (1) overhead, (2) notification time when exceptional events occur, and (3) robustness with respect to lost messages between the managing entity and the device?

P22. In Section 5.7 we saw that it was preferable to transport SNMP messages in unreliable UDP datagrams. Why do you think the designers of SNMP chose UDP rather than TCP as the transport protocol of choice for SNMP?

Socket Programming Assignment

At the end of Chapter 2, there are four socket programming assignments. Below, you will find a fifth assignment which employs ICMP, a protocol discussed in this chapter.

R6. In CSMA/CD, after the fifth collision, what is the probability that a node chooses $K = 4$? The result $K = 4$ corresponds to a delay of how many seconds on a 10 Mbps Ethernet?

R7. While TDM and FDM assign time slots and frequencies, CDMA assigns a different code to each node. Explain the basic principle in which CDMA works.

R8. Why does collision occur in CSMA, if all nodes perform carrier sensing before transmission?

SECTION 6.4

R9. How big is the MAC address space? The IPv4 address space? The IPv6 address space?

R10. Suppose nodes A, B, and C each attach to the same broadcast LAN (through their adapters). If A sends thousands of IP datagrams to B with each encapsulating frame addressed to the MAC address of B, will C's adapter process these frames? If so, will C's adapter pass the IP datagrams in these frames to the network layer C? How would your answers change if A sends frames with the MAC broadcast address?

R11. Why is an ARP query sent within a broadcast frame? Why is an ARP response sent within a frame with a specific destination MAC address?

R12. For the network in Figure 6.19, the router has two ARP modules, each with its own ARP table. Is it possible that the same MAC address appears in both tables?

R13. What is a hub used for?

R14. Consider Figure 6.15. How many subnetworks are there, in the addressing sense of Section 4.3?

R15. Each host and router has an ARP table in its memory. What are the contents of this table?

R16. The Ethernet frame begins with an 8-byte preamble field. The purpose of the first 7 bytes is to "wake up" the receiving adapters and to synchronize their clocks to that of the sender's clock. What are the contents of the 8 bytes? What is the purpose of the last byte?

Problems

P1. Suppose the information content of a packet is the bit pattern 1010 0111 0101 1001 and an even parity scheme is being used. What would the value of the field containing the parity bits be for the case of a two-dimensional parity scheme? Your answer should be such that a minimum-length checksum field is used.

P2. Show (give an example other than the one in Figure 6.5) that two-dimensional parity checks can correct and detect a single bit error. Show (give an example of) a double-bit error that can be detected but not corrected.

P3. Suppose the information portion of a packet contains six bytes consisting of the 8-bit unsigned binary ASCII representation of string "CHKSUM"; compute the Internet checksum for this data.

P4. Compute the Internet checksum for each of the following:

a. the binary representation of the numbers 1 through 6.

b. the ASCII representation of the letters C through H (uppercase).

c. the ASCII representation of the letters c through h (lowercase).

P5. Consider the generator, G = 1001, and suppose that D has the value 11000111010. What is the value of R?

P6. Rework the previous problem, but suppose that D has the value

a. 01101010101.

b. 11111010101.

c. 10001100001.

P7. In this problem, we explore some of the properties of the CRC. For the generator G (= 1001) given in Section 6.2.3, answer the following questions.

a. Why can it detect any single bit error in data D?

b. Can the above G detect any odd number of bit errors? Why?

P8. In Section 6.3, we provided an outline of the derivation of the efficiency of slotted ALOHA. In this problem we'll complete the derivation.

a. Recall that when there are N active nodes, the efficiency of slotted ALOHA is $Np(1 - p)^{N-1}$. Find the value of p that maximizes this expression.

b. Using the value of p found in (a), find the efficiency of slotted ALOHA by letting N approach infinity. *Hint*: $(1 - 1/N)^N$ approaches $1/e$ as N approaches infinity.

P9. Show that the maximum efficiency of pure ALOHA is $1/(2e)$. *Note*: This problem is easy if you have completed the problem above!

P10. Consider two nodes, A and B, that use the slotted ALOHA protocol to contend for a channel. Suppose node A has more data to transmit than node B, and node A's retransmission probability p_A is greater than node B's retransmission probability, p_B.

a. Provide a formula for node A's average throughput. What is the total efficiency of the protocol with these two nodes?

b. If $p_A = 2p_B$, is node A's average throughput twice as large as that of node B? Why or why not? If not, how can you choose p_A and p_B to make that happen?

c. In general, suppose there are N nodes, among which node A has retransmission probability $2p$ and all other nodes have retransmission probability p. Provide expressions to compute the average throughputs of node A and of any other node.

P11. Suppose four active nodes—nodes A, B, C and D—are competing for access to a channel using slotted ALOHA. Assume each node has an infinite number of packets to send. Each node attempts to transmit in each slot with probability p. The first slot is numbered slot 1, the second slot is numbered slot 2, and so on.

a. What is the probability that node A succeeds for the first time in slot 5?

b. What is the probability that some node (either A, B, C or D) succeeds in slot 4?

c. What is the probability that the first success occurs in slot 3?

d. What is the efficiency of this four-node system?

P12. Graph the efficiency of slotted ALOHA and pure ALOHA as a function of p for the following values of N:

a. $N = 15$.

b. $N = 25$.

c. $N = 35$.

P13. Consider a broadcast channel with N nodes and a transmission rate of R bps. Suppose the broadcast channel uses polling (with an additional polling node) for multiple access. Suppose the amount of time from when a node completes transmission until the subsequent node is permitted to transmit (that is, the polling delay) is d_{poll}. Suppose that within a polling round, a given node is allowed to transmit at most Q bits. What is the maximum throughput of the broadcast channel?

P14. Consider three LANs interconnected by two routers, as shown in Figure 6.33.

a. Assign IP addresses to all of the interfaces. For Subnet 1 use addresses of the form 192.168.1.xxx; for Subnet 2 uses addresses of the form 192.168.2.xxx; and for Subnet 3 use addresses of the form 192.168.3.xxx.

b. Assign MAC addresses to all of the adapters.

c. Consider sending an IP datagram from Host E to Host B. Suppose all of the ARP tables are up to date. Enumerate all the steps, as done for the single-router example in Section 6.4.1.

d. Repeat (c), now assuming that the ARP table in the sending host is empty (and the other tables are up to date).

P15. Consider Figure 6.33. Now we replace the router between subnets 1 and 2 with a switch S1, and label the router between subnets 2 and 3 as R1.

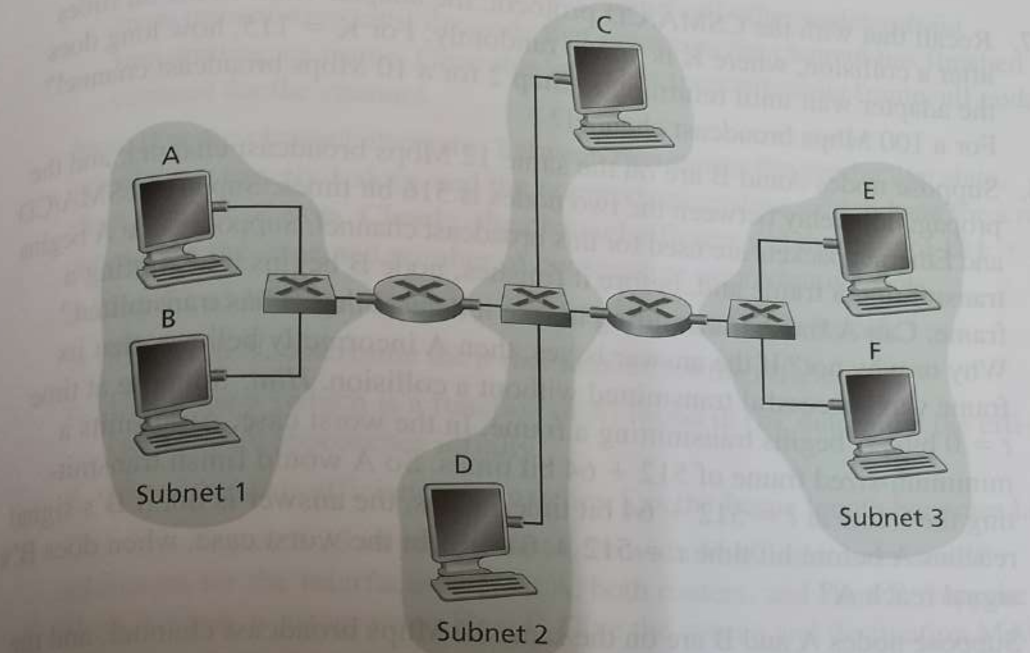


Figure 6.33 ♦ Three subnets, interconnected by routers

a. Consider sending an IP datagram from Host E to Host F. Will Host E ask router R1 to help forward the datagram? Why? In the Ethernet frame containing the IP datagram, what are the source and destination IP and MAC addresses?

b. Suppose E would like to send an IP datagram to B, and assume that E's ARP cache does not contain B's MAC address. Will E perform an ARP query to find B's MAC address? Why? In the Ethernet frame (containing the IP datagram destined to B) that is delivered to router R1, what are the source and destination IP and MAC addresses?

c. Suppose Host A would like to send an IP datagram to Host B, and neither A's ARP cache contains B's MAC address nor does B's ARP cache contain A's MAC address. Further suppose that the switch S1's forwarding table contains entries for Host B and router R1 only. Thus, A will broadcast an ARP request message. What actions will switch S1 perform once it receives the ARP request message? Will router R1 also receive this ARP request message? If so, will R1 forward the message to Subnet 3? Once Host B receives this ARP request message, it will send back to Host A an ARP response message. But will it send an ARP query message to ask for A's MAC address? Why? What will switch S1 do once it receives an ARP response message from Host B?

P16. Consider the previous problem, but suppose now that the router between subnets 2 and 3 is replaced by a switch. Answer questions (a)–(c) in the previous problem in this new context.

P17. Recall that with the CSMA/CD protocol, the adapter waits 536K bit times after a collision, where K is drawn randomly. For K = 115, how long does the adapter wait until returning to Step 2 for a 10 Mbps broadcast channel? For a 100 Mbps broadcast channel?

P18. Suppose nodes Aand B are on the same 12 Mbps broadcast channel, and the propagation delay between the two nodes is 316 bit times. Suppose CSMA/CD and Ethernet packets are used for this broadcast channel. Suppose node A begins transmitting a frame and, before it finishes, node B begins transmitting a frame. Can A finish transmitting before it detects that B has transmitted? Why or why not? If the answer is yes, then A incorrectly believes that its frame was successful transmitted without a collision. *Hint:* Suppose at time $t = 0$ bits, A begins transmitting a frame. In the worst case, Atransmits a minimum-sized frame of 512 + 64 bit times. So A would finish transmitting the frame at $t = 512 + 64$ bit times. Thus, the answer is no, if B's signal reaches A before bit time $t = 512 + 64$ bits. In the worst case, when does B's signal reach A?

P19. Suppose nodes A and B are on the same 10 Mbps broadcast channel, and the propagation delay between the two nodes is 245 bit times. Suppose A and B send Ethernet frames at the same time, the frames collide, and then A and B choose different values of K in the CSMA/CD algorithm. Assuming no other nodes are active, can the retransmissions from A and B collide? For our purposes, it suffices to work out the following example. Suppose A and B begin transmission at $t = 0$ bit times. They both detect collisions at $t = 245$ t bit times. Suppose $K_A = 0$ and $K_B = 1$. At what time does B schedule its retransmission? At what time does A begin transmission? (*Note*: The nodes must wait for an idle channel after returning to Step 2—see protocol.) At what time does A's signal reach B? Does B refrain from transmitting at its scheduled time?

P20. In this problem, you will derive the efficiency of a CSMA/CD-like multiple access protocol. In this protocol, time is slotted and all adapters are synchronized to the slots. Unlike slotted ALOHA, however, the length of a slot (in seconds) is much less than a frame time (the time to transmit a frame). Let S be the length of a slot. Suppose all frames are of constant length $L = kRS$, where R is the transmission rate of the channel and k is a large integer. Suppose there are N nodes, each with an infinite number of frames to send. We also assume that $d_{prop} < S$, so that all nodes can detect a collision before the end of a slot time. The protocol is as follows:

- If, for a given slot, no node has possession of the channel, all nodes contend for the channel; in particular, each node transmits in the slot with probability p. If exactly one node transmits in the slot, that node takes possession of the channel for the subsequent $k - 1$ slots and transmits its entire frame.

- If some node has possession of the channel, all other nodes refrain from transmitting until the node that possesses the channel has finished transmitting its frame. Once this node has transmitted its frame, all nodes contend for the channel.

Note that the channel alternates between two states: the productive state, which lasts exactly k slots, and the nonproductive state, which lasts for a random number of slots. Clearly, the channel efficiency is the ratio of $k/(k + x)$, where x is the expected number of consecutive unproductive slots.

a. For fixed N and p, determine the efficiency of this protocol.

b. For fixed N, determine the p that maximizes the efficiency.

c. Using the p (which is a function of N) found in (b), determine the efficiency as N approaches infinity.

d. Show that this efficiency approaches 1 as the frame length becomes large.

P21. Consider Figure 6.33 in problem P14. Provide MAC addresses and IP addresses for the interfaces at Host A, both routers, and Host F. Suppose Host A sends a datagram to Host F. Give the source and destination MAC addresses in the frame encapsulating this IP datagram as the frame is transmitted *(i)* from A to the left router, *(ii)* from the left router to the right router, *(iii)* from the right router to F. Also give the source and destination IP addresses in the IP datagram encapsulated within the frame at each of these points in time.

P22. Suppose now that the leftmost router in Figure 6.33 is replaced by a switch. Hosts A, B, C, and D and the right router are all star-connected into this switch. Give the source and destination MAC addresses in the frame encapsulating this IP datagram as the frame is transmitted *(i)* from A to the switch, *(ii)* from the switch to the right router, *(iii)* from the right router to F. Also give the source and destination IP addresses in the IP datagram encapsulated within the frame at each of these points in time.

P23. Consider Figure 5.15. Suppose that all links are 120 Mbps. What is the maximum total aggregate throughput that can be achieved among 12 hosts (4 in each department) and 2 servers in this network? You can assume that any host or server can send to any other host or server. Why?

P24. Suppose the three departmental switches in Figure 5.15 are replaced by hubs. All links are 120 Mbps. Now answer the questions posed in Problem P23.

P25. Suppose that *all* the switches in Figure 5.15 are replaced by hubs. All links are 120 Mbps. Now answer the questions posed in Problem P23

P26. Let's consider the operation of a learning switch in the context of a network in which 6 nodes labeled A through F are star connected into an Ethernet switch. Suppose that *(i)* B sends a frame to E, *(ii)* E replies with a frame to B, *(iii)* A sends a frame to B, *(iv)* B replies with a frame to A. The switch table is initially empty. Show the state of the switch table before and after each of these events. For each of these events, identify the link(s) on which the transmitted frame will be forwarded, and briefly justify your answers.

P27. In this problem, we explore the use of small packets for Voice-over-IP applications. One of the drawbacks of a small packet size is that a large fraction of link bandwidth is consumed by overhead bytes. To this end, suppose that the packet consists of P bytes and 5 bytes of header.

a. Consider sending a digitally encoded voice source directly. Suppose the source is encoded at a constant rate of 128 kbps. Assume each packet is entirely filled before the source sends the packet into the network. The time required to fill a packet is the **packetization delay**. In terms of L, determine the packetization delay in milliseconds.

b. Packetization delays greater than 20 msec can cause a noticeable and unpleasant echo. Determine the packetization delay for $L = 1{,}500$ bytes (roughly corresponding to a maximum-sized Ethernet packet) and for $L = 50$ (corresponding to an ATM packet).

c. Calculate the store-and-forward delay at a single switch for a link rate of $R = 622$ Mbps for $L = 1{,}500$ bytes, and for $L = 50$ bytes.

d. Comment on the advantages of using a small packet size.

P28. Consider the single switch VLAN in Figure 6.25, and assume an external router is connected to switch port 1. Assign IP addresses to the EE and CS hosts and router interface. Trace the steps taken at both the network layer and the link layer to transfer an IP datagram from an EE host to a CS host (*Hint:* Reread the discussion of Figure 6.19 in the text).

P29. Consider the MPLS network shown in Figure 6.29, and suppose that routers R5 and R6 are now MPLS enabled. Suppose that we want to perform traffic engineering so that packets from R6 destined for A are switched to A via R6-R4-R3-R1, and packets from R5 destined for A are switched via R5-R4-R2-R1. Show the MPLS tables in R5 and R6, as well as the modified table in R4, that would make this possible.

P30. Consider again the same scenario as in the previous problem, but suppose that packets from R6 destined for D are switched via R6-R4-R3, while packets from R5 destined to D are switched via R4-R2-R1-R3. Show the MPLS tables in all routers that would make this possible.

P31. In this problem, you will put together much of what you have learned about Internet protocols. Suppose you walk into a room, connect to Ethernet, and want to download a Web page. What are all the protocol steps that take place, starting from powering on your PC to getting the Web page? Assume there is nothing in our DNS or browser caches when you power on your PC. (*Hint:* The steps include the use of Ethernet, DHCP, ARP, DNS, TCP, and

HTTP protocols.) Explicitly indicate in your steps how you obtain the IP and MAC addresses of a gateway router.

P32. Consider the data center network with hierarchical topology in Figure 6.30. Suppose now there are 80 pairs of flows, with ten flows between the first and ninth rack, ten flows between the second and tenth rack, and so on. Further suppose that all links in the network are 10 Gbps, except for the links between hosts and TOR switches, which are 1 Gbps.

a. Each flow has the same data rate; determine the maximum rate of a flow.

b. For the same traffic pattern, determine the maximum rate of a flow for the highly interconnected topology in Figure 6.31.

c. Now suppose there is a similar traffic pattern, but involving 20 hosts on each rack and 160 pairs of flows. Determine the maximum flow rates for the two topologies.

P33. Consider the hierarchical network in Figure 6.30 and suppose that the data center needs to support e-mail and video distribution among other applications. Suppose four racks of servers are reserved for e-mail and four racks are reserved for video. For each of the applications, all four racks must lie below a single tier-2 switch since the tier-2 to tier-1 links do not have sufficient bandwidth to support the intra-application traffic. For the e-mail application, suppose that for 99.9 percent of the time only three racks are used, and that the video application has identical usage patterns.

a. For what fraction of time does the e-mail application need to use a fourth rack? How about for the video application?

b. Assuming e-mail usage and video usage are independent, for what fraction of time do (equivalently, what is the probability that) both applications need their fourth rack?

c. Suppose that it is acceptable for an application to have a shortage of servers for 0.001 percent of time or less (causing rare periods of performance degradation for users). Discuss how the topology in Figure 6.31 can be used so that only seven racks are collectively assigned to the two applications (assuming that the topology can support all the traffic).

Wireshark Labs

At the Companion website for this textbook, http://www.pearsonglobaleditions.com/kurose, you'll find a Wireshark lab that examines the operation of the IEEE 802.3 protocol and the Wireshark frame format. A second Wireshark lab examines packet traces taken in a home network scenario.

SECTION 7.8

R23. What are three approaches that can be taken to avoid having a single wireless link degrade the performance of an end-to-end transport-layer TCP connection?

Problems

P1. Consider the single-sender CDMA example in Figure 7.5. What would be the sender's output (for the 2 data bits shown) if the sender's CDMA code were (1, 1, −1, 1, 1, −1, −1, 1)?

P2. Consider sender 2 in Figure 7.6. Assume that both the first two bits sent by sender 2 are −1. What are the sender's outputs to the channel (before being added to the signal from sender 1)?

P3. After selecting the AP with which to associate, a wireless host sends an association request frame to the AP, and the AP responds with an association response frame. Once associated with an AP, the host will want to join the subnet (in the IP addressing sense of Section 4.4.2) to which the AP belongs. What does the host do next?

P4. If two CDMA senders have codes (1, 1, 1, −1, 1, −1, −1, −1) and (1, −1, 1, 1, 1, 1, 1, 1), would the corresponding receivers be able to decode the data correctly? Justify.

P5. Suppose there are two ISPs providing WiFi access in a particular café, with each ISP operating its own AP and having its own IP address block.

a. Further suppose that by accident, each ISP has configured its AP to operate over channel 11. Will the 802.11 protocol completely break down in this situation? Discuss what happens when two stations, each associated with a different ISP, attempt to transmit at the same time.

b. Now suppose that one AP operates over channel 1 and the other over channel 11. How do your answers change?

P6. In step 4 of the CSMA/CA protocol, a station that successfully transmits a frame begins the CSMA/CA protocol for a second frame at step 2, rather than at step 1. What rationale might the designers of CSMA/CA have had in mind by having such a station not transmit the second frame immediately (if the channel is sensed idle)?

P7. Suppose an 802.11b station is configured to always reserve the channel with the RTS/CTS sequence. Suppose this station suddenly wants to transmit 1,000 bytes of data, and all other stations are idle at this time. Assume a transmission rate of 12 Mbps. As a function of SIFS and DIFS, and ignoring propagation delay and assuming no bit errors, calculate the time required to transmit the frame and receive the acknowledgment.

P8. Consider the scenario shown in Figure 7.34, in which there are four wireless nodes, A, B, C, and D. The radio coverage of the four nodes is shown via the shaded ovals; all nodes share the same frequency. When A transmits, it

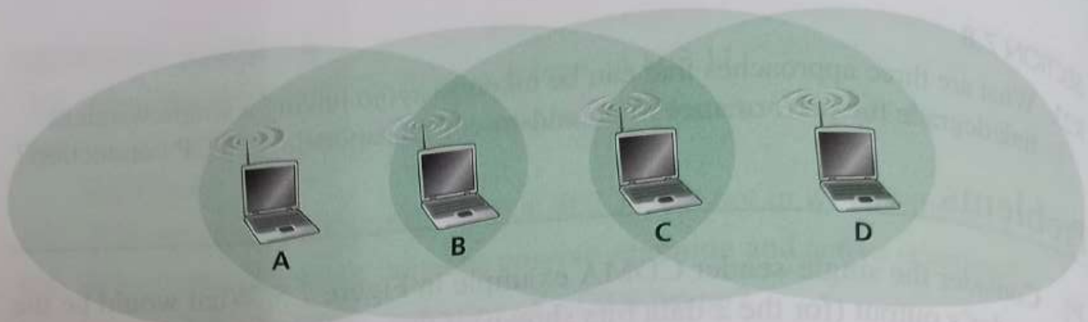


Figure 7.34 ♦ Scenario for problem P8

can only be heard/received by B; when B transmits, both A and C can hear/receive from B; when C transmits, both B and D can hear/receive from C; when D transmits, only C can hear/receive from D.

Suppose now that each node has an infinite supply of messages that it wants to send to each of the other nodes. If a message's destination is not an immediate neighbor, then the message must be relayed. For example, if A wants to send to D, a message from A must first be sent to B, which then sends the message to C, which then sends the message to D. Time is slotted, with a message transmission time taking exactly one time slot, e.g., as in slotted Aloha. During a slot, a node can do one of the following: (*i*) send a message, (*ii*) receive a message (if exactly one message is being sent to it), (*iii*) remain silent. As always, if a node hears two or more simultaneous transmissions, a collision occurs and none of the transmitted messages are received successfully. You can assume here that there are no bit-level errors, and thus if exactly one message is sent, it will be received correctly by those within the transmission radius of the sender.

a. Suppose now that an omniscient controller (i.e., a controller that knows the state of every node in the network) can command each node to do whatever it (the omniscient controller) wishes, i.e., to send a message, to receive a message, or to remain silent. Given this omniscient controller, what is the maximum rate at which a data message can be transferred from C to A, given that there are no other messages between any other source/destination pairs?
b. Suppose now that A sends messages to B, and D sends messages to C. What is the combined maximum rate at which data messages can flow from A to B and from D to C?
c. Suppose now that A sends messages to B, and C sends messages to D. What is the combined maximum rate at which data messages can flow from A to B and from C to D?
d. Suppose now that the wireless links are replaced by wired links. Repeat questions (a) through (c) again in this wired scenario.

e. Now suppose we are again in the wireless scenario, and that for every data message sent from source to destination, the destination will send an ACK message back to the source (e.g., as in TCP). Also suppose that each ACK message takes up one slot. Repeat questions (a)–(c) above for this scenario.

P9. Power is a precious resource in mobile devices, and thus the 802.11 standard provides power-management capabilities that allow 802.11 nodes to minimize the amount of time that their sense, transmit, and receive functions and other circuitry need to be "on." In 802.11, a node is able to explicitly alternate between sleep and wake states. Explain in brief how a node communicates with the AP to perform power management.

P10. Consider the following idealized LTE scenario. The downstream channel (see Figure 7.21) is slotted in time, across F frequencies. There are four nodes, A, B, C, and D, reachable from the base station at rates of 10 Mbps, 5 Mbps, 2.5 Mbps, and 1 Mbps, respectively, on the downstream channel. These rates assume that the base station utilizes all time slots available on all F frequencies to send to just one station. The base station has an infinite amount of data to send to each of the nodes, and can send to any one of these four nodes using any of the F frequencies during any time slot in the downstream sub-frame.

a. What is the maximum rate at which the base station can send to the nodes, assuming it can send to any node it chooses during each time slot? Is your solution fair? Explain and define what you mean by "fair."

b. If there is a fairness requirement that each node must receive an equal amount of data during each one second interval, what is the average transmission rate by the base station (to all nodes) during the downstream sub-frame? Explain how you arrived at your answer.

c. Suppose that the fairness criterion is that any node can receive at most twice as much data as any other node during the sub-frame. What is the average transmission rate by the base station (to all nodes) during the sub-frame? Explain how you arrived at your answer.

P11. In Section 7.5, one proposed solution that allowed mobile users to maintain their IP addresses as they moved among foreign networks was to have a foreign network advertise a highly specific route to the mobile user and use the existing routing infrastructure to propagate this information throughout the network. We identified scalability as one concern. Suppose that when a mobile user moves from one network to another, the new foreign network advertises a specific route to the mobile user, and the old foreign network withdraws its route. Consider how routing information propagates in a distance-vector algorithm (particularly for the case of interdomain routing among networks that span the globe).

a. Will other routers be able to route datagrams immediately to the new foreign network as soon as the foreign network begins advertising its route?

b. Is it possible for different routers to believe that different foreign networks contain the mobile user?

c. Discuss the timescale over which other routers in the network will eventually learn the path to the mobile users.

P12. Suppose the correspondent in Figure 7.23 were mobile. Sketch the additional network-layer infrastructure that would be needed to route the datagram from the original mobile user to the (now mobile) correspondent. Show the structure of the datagram(s) between the original mobile user and the (now mobile) correspondent, as in Figure 7.24.

P13. In mobile IP, what effect will mobility have on end-to-end delays of datagrams between the source and destination?

P14. Consider the chaining example discussed at the end of Section 7.7.2. Suppose a mobile user visits foreign networks A, B, and C, and that a correspondent begins a connection to the mobile user when it is resident in foreign network A. List the sequence of messages between foreign agents, and between foreign agents and the home agent as the mobile user moves from network A to network B to network C. Next, suppose chaining is not performed, and the correspondent (as well as the home agent) must be explicitly notified of the changes in the mobile user's care-of address. List the sequence of messages that would need to be exchanged in this second scenario.

P15. Consider two mobile nodes in a foreign network having a foreign agent. Is it possible for the two mobile nodes to use the same care-of address in mobile IP? Explain your answer.

P16. In our discussion of how the VLR updated the HLR with information about the mobile's current location, what are the advantages and disadvantages of providing the MSRN as opposed to the address of the VLR to the HLR?

Wireshark Lab

At the Web site for this textbook, www.pearsonglobaleditions.com/kurose, you'll find a Wireshark lab for this chapter that captures and studies the 802.11 frames exchanged between a wireless laptop and an access point.