# IK1203
# Networks and Communication

## Recitation 1 – Introduction and Application Layer

**1.** Answer he following short questions.
   a) What is a communication protocol?
   b) Name the different layers in the Internet protocol stack, and place the following protocols/functions/concepts at the correct layer: IP, TCP, Ethernet, HTTP, bit coding, FTP, IEEE 802.11 WLAN, TP Category 6, Routing, UDP.
   c) What layer in the Internet protocol stack is responsible for the transfer of a data packet over a single link, between two directly connected devices?
   d) A router has two main functions, which can be described by the two terms "routing" and "forwarding". What is the difference between routing and forwarding?
   e) What service does the transport layer describe? Give a short answer.

**2.** Describe shortly how an e-mail message in sent from a sending to a receiving e-mail client (user agent), such as MS Outlook or Mozilla Thunderbird. From the description, it should be clear what application protocols and what systems are involved in the transfer.

**3.** An e-mail message is sent from an e-mail client (MUA) to a server for outgoing e-mail.
   a) Which protocol is used for the transfer?
   b) The communication delay between the client and the server is 2.5 milliseconds (in other words, it takes 2.5 milliseconds from that the sender has started sending the message until the complete message has reached the receiver). How long time does it take before the entire transfer of the e-mail message to the outgoing server has been completed? The time it takes to set up and tear down a TCP connection should not be included in the calculation. The message has one recipient.
   c) Suppose that the message has four recipients. How long time does it take then?

**4.** According to the SMTP protocol, the end of the message is indicated by a line with only a single period '.' Does that mean that it is not possible to send an e-mail message that contains a line with a single period? Explain!

**5.** Suppose that the following request is sent to an HTTP server:

```
GET / HTTP/1.1
Host: www.kth.se
Connection: keep-alive
```

The answer is:

```
HTTP/1.1 200 OK
Date: Mon, 26 Jan 2015 21:42:31 GMT
Server: Apache/2.2.15 (Red Hat)
Set-Cookie: JSESSIONID=00DFBC112EABCE74FAB714A56ABF3282; Path=/; Secure; HttpOnly
Content-Language: sv-SE
Content-Length: 60044
Connection: close
Content-Type: text/html;charset=UTF-8

<html>
<head>
<title>KTH | Välkommen till KTH</title>
```

*A lot of information removed...*

Answer the following questions:

a) What is the complete URL for the requested object?
b) Was the request successful?
c) How large is the requested object?
d) Explain the header field "Connection:" that appears both in the request and in the response. What does the client want, and what is the response from the server?
e) What are the first five characters in the returned object?

## Problems from course book (Kurose and Ross, 7th ed)

### Chapter 1

P2.
Equation 1.1, $d_{\text{end-to-end}} = N*(L/R)$, 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.

P6.
This elementary problem begins to explore propagation delay and transmission delay, two central concepts in data networking. Consider two hosts, A and B, connected by a single link of rate $R$ bps. Suppose that the two hosts are separated by $m$ meters, and suppose the propagation speed along the link is $s$ meters/sec. Host A is to send a packet of size $L$ bits to Host B.

a. Express the propagation delay, $d_{\text{prop}}$, in terms of $m$ and $s$.
b. Determine the transmission time of the packet, $d_{\text{trans}}$, 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 $d_{\text{prop}}$ is greater than $d_{\text{trans}}$. At time $t = d_{\text{trans}}$, where is the first bit of the packet?
f. Suppose $d_{\text{prop}}$ is less than $d_{\text{trans}}$. At time $t = d_{\text{trans}}$, where is the first bit of the packet?
g. Suppose $s = 2.5*10^8$, $L = 120$ bits, and $R = 56$ kbps. Find the distance $m$ so that $d_{\text{prop}}$ equals $d_{\text{trans}}$.

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)?

P10.
Consider the network illustrated in Figure 1.16 (below). 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?

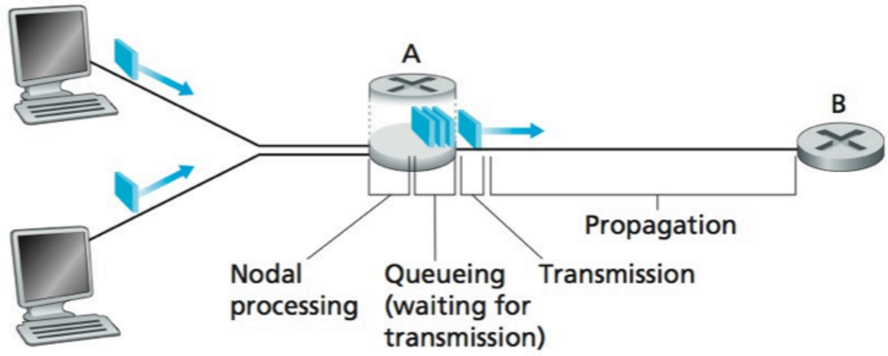


**Figure 1.16** ♦ The nodal delay at router A

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?

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 $NL/R$ seconds. What is the average queuing delay of a packet?

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## Recitation 1 – Introduction and Application Layer

## Solutions

**1.** Short questions:

a) A communication protocol is an agreement between communicating parties defining the rules for communication.

b) Application: HTTP, FTP.

Transport: TCP, UDP.

Network: IP, Routing.

Link: Ethernet, IEEE 802.11 WLAN.

Physical: Bit coding, TP Category 6.

c) The link layer.

d) ”Routing” is about selecting a route (or path) – when the switches (routers) decide which path a packet should take through the network. ”Forwarding” is when a switch (router) redirects a packet from an incoming port to an outgoing port.

e) The transport layer provides the services of transferring data from a process on one computer (host) to a process on another computer (end-to-end delivery).

**2.** In the first step, the sending e-mail client sends the message via SMTP to the local server for outgoing e-mail (Message Transfer Agent). Second, the e-post message is sent from the server for outgoing e-mail to the recipient's server for incoming e-mail. This step also uses SMTP. Finally, in the third step, the recipient uses a protocol for e-mail access, such as POP or IMAP, to collect the e-mail message from the server for incoming e-mail.

**3.**

a) SMTP

b) The communication between the client and the server includes a number of handshakes. In each handshake, the client sends a command to the server and waits for a response. Each handshake takes $2 \times 2.5$ ms = 5 ms. A complete transfer to the outgoing server involves the following handshakes::

- ”HELO”, ”MAIL FROM”, ”RCPT TO”, ”DATA”, the actual e-mail message, ”QUIT”

All in all, the transfer takes 30 ms.

c) RCPT TO adds a recipient for the message. Hence, there is one handshake for each recipient. So 45 ms total.

**4.** Yes, it is possible. According to the SMTP standard, the is solved by replacing any period at the beginning of a line with two periods. The SMTP receiver will change back to a single period.

**5.** HTTP-request/response:

a) http://www.kth.se/

b) Yes, it was successful. The answer is ”200 OK”.

c) Content-Length gives the size of ”entity-body”, in other words, the data portion od the response: 60044 byte.

d) The client asks for a ”persistent connection”, meaning that the TCP connection should remain open so that the client can send more HTTP requests over the same connection. The server does not accept, and wants to close the TCP connection directly after the HTTP response.

e) The five first characters of the returned object are: ”<html”.

## Problems from course book (Kurose and Ross, 7th ed)

P2.

At time $N*(L/R)$ the first packet has reached the destination, the second packet is stored in the last router, the third packet is stored in the next-to-last router, etc. At time $N*(L/R) + L/R$, the second packet has reached the destination, the third packet is stored in the last router, etc. Continuing with this logic, we see that at time $N*(L/R) + (P-1)*(L/R) = (N+P-1)*(L/R)$ all packets have reached the destination.

P6.

a. $d_{prop} = m/s$ seconds.
b. $d_{trans} = L/R$ seconds.
c. $d_{end\text{-}to\text{-}end} = d_{prop} + d_{trans} = (m/s + L/R)$ seconds
d. The bit is just leaving Host A.
e. The first bit is in the link and has not reached Host B.
f. The first bit has reached Host B.
g. Want
   $m = (L/R)*s = (120/56*10^3)*2.5*10^8 = 536$ km

P7.

Consider the first bit in a packet. Before this bit can be transmitted, all of the bits in the packet must be generated. This requires $56*8/(64*10^3) = 7$ ms.
The time required to transmit the packet is $56*8/(2*10^6) = 224$ µs
Propagation delay = 10 ms.
The delay until decoding is
7 ms + 224 µs + 10 ms = 17.224 ms
A similar analysis shows that all bits experience a delay of 17.224 ms.

P10.

The first bit sent by the host with the lowest propagation delay reaches Router A after 2 ms while the last bit of the packet arrives after $2*10^{-3} + 1500*8/(4*10^6)$ s = 5 ms.
The first bit sent by the host with the highest propagation delay reaches Router A after 6 ms. At that time the packet sent by the other host is already fully received by the router, so no queuing delay occurs.

P11.

Assume $d_1<d_2$. No buffering occurs when $d_2>d_1 + L/R_1$.

P13.

a. The queuing delay is 0 for the first transmitted packet, $L/R$, for the second transmitted packet, and generally, $(n-1)*L/R$ for the $n^{th}$ transmitted packet. Thus, the average delay for the $N$ packets is:

$(L/R + 2L/R + \ldots\ldots. + (N\text{-}1)L/R)/N$
$= L/(RN)*(1 + 2 + \ldots.. + (N\text{-}1))$
$= L/(RN) * N(N\text{-}1)2$
$= LN(N\text{-}1)/(2RN)$
$= (N\text{-}1)L/(2R)$
Note that here we used the well-known fact:
$1 + 2 + \ldots.. + N = N(N+1)/2$

b. It takes $NL/R$ seconds to transmit the $N$ packets. Thus, the buffer is empty when each batch of $N$ packets arrive. Thus, the average delay of a packet across all batches is the average delay within one batch, i.e., $(N\text{-}1)L/2R$.

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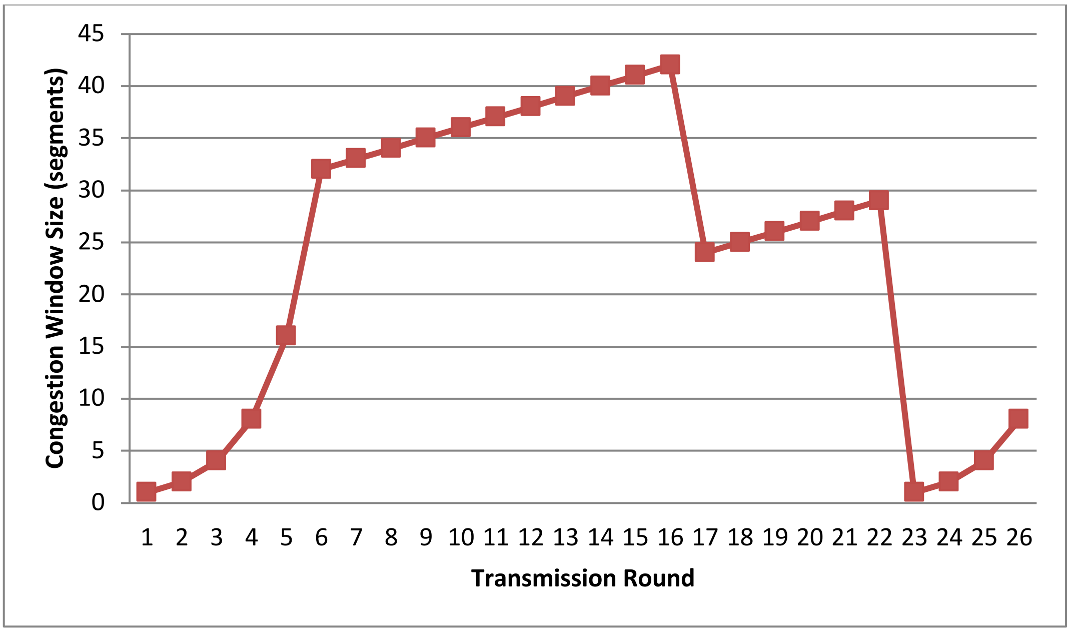
## Recitation 2 – Transport layer

1. TCP uses delayed ACKs instead of sending and ACK directly after a correctly received packet. Answer the two following questions related to delayed ACKs in TCP.
   a) An ACK must not be delayed more than 500 ms. Why?
   b) Assume that a TCP segment arrives with the expected sequence number. The previous segment arrived in correct order and it has not been ACKed yet. What will the receiver do now?
2. TCP uses both flow control and congestion control. Explain the overall difference between these. What do they mean? What are their purposes?
3. An application uses TCP and sends data in full size windows (65 535 bytes) over a 1 Gbps channel having a one-way delay of 10 ms. The transmission time can be neglected.
   a) What is the maximum throughput that can be achieved?
   b) What channel utilization can be achieved, i.e., how large part of the available bandwidth can be used?
4. A client application establishes a TCP connection to a server application to transfer 15 kB of data. The (one-way) delay is 5 ms, RTT (round-trip time) is 10 ms, and the receive window (rwnd) is 24 kB. Assume that the initial congestion window is 2 kB. There is no congestion in the network, the transmission time can be neglected, and the connection establishment phase can be neglected. Calculate the total transfer time.

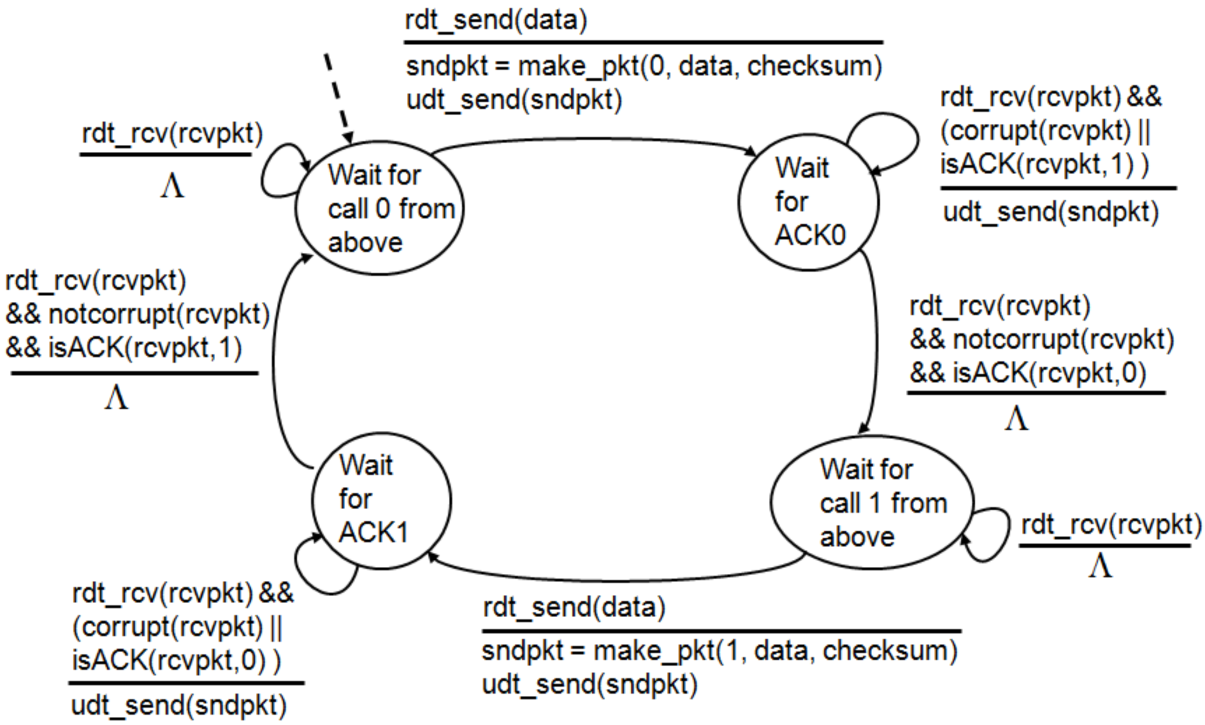
**5.** The figure below shows how the congestion window (CWND) varies in TCP Reno (i.e., with fast retransmit and recovery).

a) Mark the intervals when TCP is in slow start.
b) Mark the intervals when TCP is in congestion avoidance.
c) During the $16^{th}$ transmission round, a packet loss occurs. How is it detected? Is detected through a timeout or through the reception of three duplicate ACKs?
d) During the 22nd transmission round, is segment loss detected by a triple duplicate ACK or by a timeout?
e) What is the initial window 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?

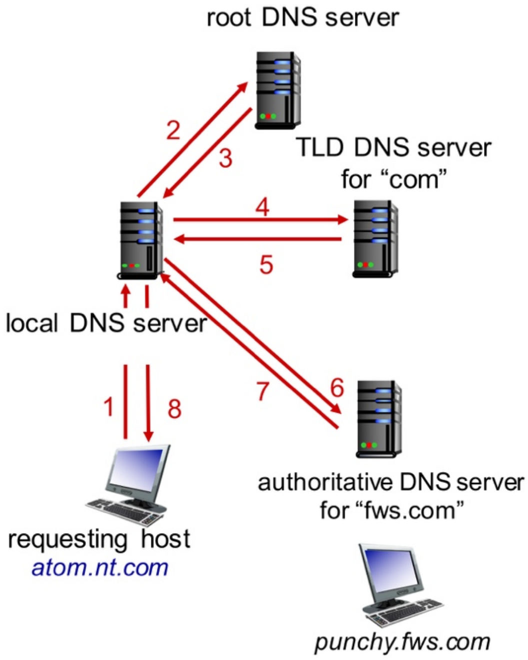


6. The figure below illustrates the finite state machine (FSM) on the sender side of a reliable stop-and-wait transport protocol, which can handle corrupted packets but it cannot handle lost packets. The notation in the FSM is the same as the one used in the course book.

   Complete the FSM with a timer-based retransmission mechanism so that the transport protocol can handle also packet losses. We assume that the functions `start_timer` and `stop_timer` are there to start and stop a timer, and that a `timeout` event occurs when a timer expires.




7. The figure below illustrates what happens when a host computer “atom.nt.com” at the fictive company Newton Technologies (“nt.com”) does a DNS lookup of the domain name of a computer “punchy.fws.com” at the (also fictive) company Future Web Services (“fws.com”). The numbers on the arrows in the figure indicate the order in which the operations are done.



a) Based on the figure, explain what happens during the DNS lookup. From the description, the different types of DNS servers should be clear.
b) There are two different kinds of lookup schemes procedures: recursive and iterative. Which of the DNS queries are recursive, and which are iterative?
c) Suppose that Newton Technologies has a web server with the domain name "www.nt.com". The company does now want the web server to have a second name in the ".biz"-domain, "www.nt.biz". Is this possible? Can a web server have domain names in different top level domains?

**8.** DNS is used for translating host names to IP addresses. But DNS is really more general: it is a distributed database for storing Resource Records. IP addresses represent one type of resource records.

A DNS query has two parts, a name and a type, and the response contains one or more resource records. Which of the entries in the table below describes valid queries that are support by DNS? For the valid queries, write the corresponding query type. (The textbook may not be sufficient to answer this question; you may need to consult the web.)

| Query description | Query type |
|---|---|
| **IP version 4 address for "www.kth.se"** | |
| **IP version 6 address for "www.kth.se"** | |
| **TCP port number for HTTP server at "www.kth.se"** | |
| **Incoming mail server for "kth.se"** | |
| **Outgoing mail server for "kth.se"** | |
| **Authoritative name server for "kth.se"** | |
| **Web server for "kth.se"** | |
| **Main (canonical) name for alias "www.kth.se"** | |
| **Host name with address "130.237.28.40"** | |

## Problems from course book (Kurose and Ross, 7th ed)

P45.
Recall the macroscopic description of TCP throughput. In the period of time from when the connection's rate varies from W/(2 · 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

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

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

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

P46.

Consider that only a single TCP (Reno) connection uses one 10 Mbps 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 ms; 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?

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## Recitation 2 – Transport layer

## Solutions

**1.**

a) Because the retransmission timer can then expire and cause an unnecessary retransmission.
b) The receiver will immediately send an ACK (no more delay). This is what causes the typical "ACK every other segment" pattern in case of bulk transfers over TCP.

**2.** Flow control is a way for the receiver to regulate the sender's transmission pace. The purpose is to prevent the receiver from becoming overwhelmed with more data than it can process. Congestion control deals with the situation when packets are dropped by routers along the path between sender and receiver. The purpose of TCP congestion control is to regulate the sender's transmission pace based on the current network conditions.

**3.**

a) A window of data can be sent every 20th ms. It gives a maximum throughput of:
$(65535 * 8)/(20*10^{-3})$ bps = 26 214 000 bps (≈ 26 Mbps).
b) Channel utilization = 26 214 000/1 000 000 000 ≈ 2,6%

**4.** 35 ms. TCP begins in slow start by first sending 2 kB. After one RTT TCP will send 4kB; after two RTT TCP will send 8 kB; after 3 RTT TCP will send the last 1 kB of data (2+4+8+1 = 15). The last kB will reach the server application 5 ms later. So, in total: 3 * RTT + RTT/2 = 35 ms.

**5.**

a) Slow start: [1, 6] and [23, 26]
b) Congestion avoidance: [6, 16] and [17, 22].
c) In this case, the packet loss is detected through three duplicate ACKs. This can be concluded since TCP goes to fast recovery and reduces the congestion window to half (roughly) the current size. If the packet loss had been detected through a timeout, TCP would go to slow start and reduced the congestion window to 1 MSS.
d) After the 22nd transmission round, segment loss is detected due to timeout, and hence the congestion window size is set to 1.
e) The threshold is initially 32, since it is at this window size that slow start stops and congestion avoidance begins.
f) The threshold is set to half the value of the congestion window when packet loss is detected. When loss is detected during transmission round 16, the congestion windows size is 42. Hence the threshold is 21 during the 18th transmission round.
g) The threshold is set to half the value of the congestion window when packet loss is detected. When loss is detected during transmission round 22, the congestion windows size is 29. Hence the threshold is 14 (taking lower floor of 14.5) during the 24th transmission round.
h) During the 1st transmission round, packet 1 is sent; packet 2-3 are sent in the 2nd transmission round; packets 4-7 are sent in the 3rd transmission round; packets 8-15 are sent in the 4th transmission round; packets 16-31 are sent in the 5th transmission round; packets 32-63 are sent in

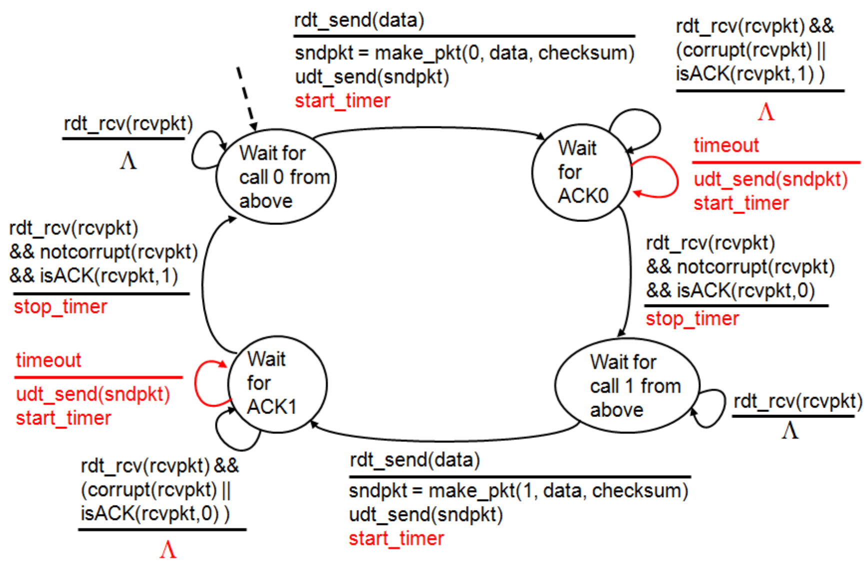
the 6th transmission round; packets 64 – 96 are sent in the 7th transmission round. Thus packet 70 is sent in the 7th transmission round.

i) The threshold will be set to half the current value of the congestion window (8) when the loss occurred and congestion window will be set to the new threshold value + 3 MSS. Thus the new values of the threshold and window will be 4 and 7 respectively.

j) Threshold is 21, and congestion window size is 1.

k) Round 17, 1 packet; round 18, 2 packets; round 19, 4 packets; round 20, 8 packets; round 21, 16 packets; round 22, 21 packets. So, the total number is 52.

**6.** The finite state machine is illustrated below. The receiver's FSM is not changed when the timer-based retransmission mechanism is introduced.




**7.** a)

1. The computer on Newton Technologies, "atom.nt.com," sends a DNS query to the local DNS server.

2. Since the local DNS server has an empty cache, the local DNS server sends the query to a root DNS server.

3. The root DNS server knows the IP address(es) of the the Top-Level Domain (TLD) servers for the top-level domain "com" and responds with the IP addresses of those servers to the local DNS server.

4. The local DNS server sends the query to one of the TLD servers for the "com" top-level domain.

5. The ".com" TLD DNS server has the IP address to the authoritative DNS server for the domain name "fws.com". The TLD DNS server responds with this IP address to the local DNS server.
6. The local DNS server sends the query to the authoritative DNS server for the domain name "fws.com". The authoritative DNS server is responsible for all names in this domain, so (most likely) it can answer the question.
7. The authoritative DNS server responds to the local DNS server with the IP address(es) of the host in question.
8. The local DNS server now has the complete answer to the query, and sends this answer back to "atom.nt.com".

b) DNS query (1) is processed in a recursive way by the local DNS server, while queries (2), (4) are (6) handled iteratively.

c) Yes, there is nothing in DNS that would prevent this.

**8.** "Dig" is a very useful program for doing DNS queries from the command line. The dig commands for the (valid) queries are included in the table below.

| **Query description** | **Query type** | **Dig command** |
|---|---|---|
| **IP version 4 address for "www.kth.se"** | A | dig a www.kth.se |
| **IP version 6 address for "www.kth.se"** | AAAA | dig aaaa www.kth.se |
| **TCP port number for HTTP server at "www.kth.se"** | – | |
| **Incoming mail server for "kth.se"** | MX | dig mx kth.se |
| **Outgoing mail server for "kth.se"** | – | |
| **Authoritative name server for "kth.se"** | NS | dig ns kth.se |
| **Web server for "kth.se"** | – | |
| **Main (canonical) name for alias "www.kth.se"** | CNAME | dig cname www.kth.se |
| **Host name with address "**130.237.28.40**"** | PTR | dig –x 130.237.28.40 *or*<br>dig ptr 40.28.237.130.in-addr.arpa |

## Problems from course book (Kurose and Ross, 7th ed), Solutions

P45.

a) The loss rate, L, is the ratio of the number of packets lost over the number of packets sent. In a cycle, 1 packet is lost. The number of packets sent in a cycle is

$$\frac{W}{2} + \left(\frac{W}{2} + 1\right) + \cdots + W = \sum_{n=0}^{W/2} \left(\frac{W}{2} + n\right)$$

$$= \left(\frac{W}{2} + 1\right)\frac{W}{2} + \sum_{n=0}^{W/2} n$$

$$= \left(\frac{W}{2} + 1\right)\frac{W}{2} + \frac{\frac{W}{2}(\frac{W}{2} + 1)}{2}$$

$$= \frac{W^2}{4} + \frac{W}{2} + \frac{W^2}{8} + \frac{W}{4}$$

$$= \frac{3}{8}W^2 + \frac{3}{4}W$$

Thus the loss rate is

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

b) For W large, $\frac{3}{8}W^2 \gg \frac{3}{4}W$. Thus $L \approx \frac{3}{8}W^2$ or $W \approx \sqrt{\frac{8}{3L}}$. From the text, we therefore have average throughput

$$= \frac{3}{4}\sqrt{\frac{8}{3L}} \times \frac{MSS}{RTT}$$

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

P46.

a) Let W denote the max window size measured in segments. Then, W*MSS/RTT = 10 Mbps, as packets will be dropped if the maximum sending rate exceeds link capacity. Thus, we have W*1500*8/0.15=10*10^6, then W is about 125 segments.
b) As congestion window size varies from W/2 to W, then the average window size is 0.75W=94 (ceiling of 93.75) segments. Average throughput is 94*1500*8/0.15 = 7.52 Mbps.
c) 94/2 *0.15 = 7.05 seconds, as the number of RTTs (that this TCP connections needs in order to increase its window size from W/2 to W) is given by W/2. Recall the window size increases by one in each RTT.