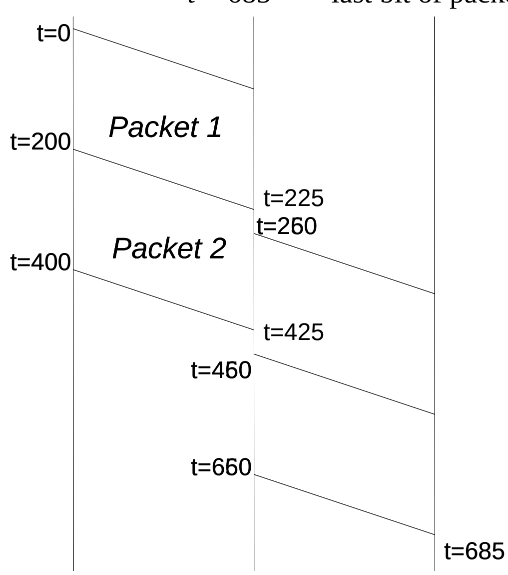
ECE 4380/6380: Computer Communications **Spring 2018**
Problem Set 1 Solutions

Each problem is worth 10 points.

1. Making sure we correctly convert units, y KB is $8 \times 2^{10} \times y$ bits and x Mbps is $x \times 10^6$ bits per second. The transmission time is $(8 \times 1{,}024 \times y) / (x \times 1000000)$ sec = $0.008192\, y / x$ sec = $8.192\, y / x$ msec.

2. **Store-and-forward delays**
   (a) Per-link transmit delay is 40×10^3 bits / 100×10^6 bits/sec = 400 μs. Total transmission time = $2 \times 400 + 2 \times 25 + 35 = 885$ μs.
   (b) When sending as two packets, here is a table of times for various events (in μs)

$t = 0$	start
$t = 200$	A finishes sending packet 1, starts packet 2
$t = 225$	packet 1 finishes arriving at S
$t = 260$	packet 1 departs for B
$t = 400$	A finishes sending packet 2
$t = 460$	packet 2 departs for B
$t = 485$	bit 1 of packet 2 arrives at B
$t = 685$	last bit of packet 2 arrives at B



Note that by splitting the packet into smaller frames, the delay is reduced by approximately 23% in this example. Switch S is a store-and-forward devise and the increased delay is due primarily to the requirement to receive the full frame before it can be forwarded.

3. Consider the total RTT (i.e., the time when A starts to transmit a packet and when the acknowledgment is received). The RTT is the sum of the following delays

Component of delay	value
Packet transmission time A to S	(10,150 bits)/15,000,000 bits/sec = 676.7 μs
Packet transmission time S to B	(10,150 bits)/60,000,000 bits/sec = 169.2 μs
Ack transmission time B to S	(150 bits)/60,000,000 bits/sec = 2.5 μs
Ack transmission time S to A	(150 bits)/15,000,000 bits/sec = 10 μs
Propagation delay	$2\times25+2\times50 = 150$ μs
Switching time	$2\times25 = 50$ μs

(a) The total delay is 1058 μs, and A can transmit 10,000 bits during this time. The throughput is 10,000 bits / 1058 μs = 9.45 Mbps, considerably less than the maximum possible.
(b) The RTT is calculated above as 1058 μs, so the pipe is 1058 μs × 15 Mbps = 15,870 bits
(c) When considering the operation of the connection in steady state (i.e., the average as determined over a long time), the throughput in the reverse direction is the same as part (a). Even though host B can transmit at 60 Mbps, if it transmits continuously at this rate it will cause the packet buffer at the switch to eventually overflow because the switch can only transmit at 15 Mbps to host A. However, if you consider the most amount of data that B could transmit before it receives the first acknowledgement, then you can argue that the pipe is four times larger than found in part (b). During this course we will need to consider when it is appropriate to evaluate performance for a fixed period of time (for example, during one round-trip time) or to evaluate average performance over infinite time (or, practically, over a very long period of time).

4. *optimal packet size for a single packet retransmission*. The number of packets that are required, N, depends on the size of the packet, S, and is equal to

$$N = \left\lceil \frac{1\times 10^6}{S-100} \right\rceil .$$

The total number of bytes that is sent is one million, plus the overhead included in each packet, $N \times 100$, plus the number of bytes in the packet that is retransmitted, S. For this problem assume that each packet has the maximum size expect for the last packet, and that a packet other than the last packet was lost. You could make other assumptions, such as all packets have equal size, or dummy bits are added to the last packet to make all packets have the same size; if you made a different assumption be sure to clearly indicate what you did.

S	N	overhead bytes,	total bytes
1000	1112	112,200	1,112,200
5000	205	25,500	1,025,500
10000	102	20,200	1,020,200
20000	51	25,100	1,025,100

The packet size that minimizes the number of bytes is 10,000. This balances the overhead included in each packet with the overhead when a retransmission attempt is required.

As an alternative solution, suppose we consider a system in which each packet must be *exactly* the specified size, and the last packet must be padded so all packets are the same size. Then, the total number of bytes that are sent is simply $(N + 1)S$, or

S	N	total bytes
1000	1112	1,113,000
5000	205	1,030,000
10000	102	1,030,000
20000	51	1,040,000

Under this assumption, the total bytes are minimized with either the 5,000 or 10,000 byte packet size.

5. The time to send one 300-byte packet is 2400 bits/100Mbps = 24 μs. The length of cable needed to exactly contain such a packet is 24 μs × 2×10^8 m/sec = 4,800 meters. 300 bytes in 4800 meters is 0.5 bits per meter (or 50 bits per 100 m). With an extra 10 bits/100m, we have a total of 60 bits/100m. A 2400-bit packet now fills 2400/(.6 bits/m) = 4000 meters.

6. (a) In STDM each host is assigned one of the slots in a frame. The slot can only be used by the assigned host, and if there are no packets from the host, the slot is wasted. The round-robin access mechanism attempts to find a packet to transmit in each slot. That is, if a host does not have a packet to transmit its turn is skipped and a packet from another host can be sent in the current time slot. If there are any packets waiting, a packet will be sent. Statistical multiplexing is very similar to round-robin forwarding, in that a time slot will be used if there are packets waiting to be sent. The difference between round-robin and statistical multiplexing is only in the order that packets are sent. Statistical multiplexing sends the packet in a first-come-first-serve order, while round-robin guarantees that each host with packets to send gets to send at least one packet in a frame. (Later in the semester we will study weighted fair queueing, which is a generalization of round-robin as described here).
   (b) The utilization of the outgoing link should be the same for round-robin and statistical multiplexing as they both will send a packet in a slot if there is a packet available to send (the only difference is the order in which the packets are sent). Both will allow a greater utilization that STDM because time slots will not be wasted if there is a packet waiting to be forwarded.

7. Transfer times.

We define the transfer to be completed when the destination returns a final acknowledgement to the source to signal the end of the transfer. An alternative interpretation is to measure the time until the last data bit arrives at its destination, and the transfer time is from the start of the initial handshake until the last bit is received at the destination. In this case the total transfer time is decreased by an additional one-way propagation delay (50 ms).

   (a) The total delay is 2 initial RTT's (300ms) + 1500KB/2.5 Mbps (transmit) + 2×propagation delay (150 ms) =0.3 sec plus the transmission time. The total transmission time is 12,288,000 bits/2,500,000 bits/sec = 4.915 sec, for a total delay of 5.365 sec.
   (b) To the above we add the time for 1499 additional RTTs (the number of RTTs between when packet 1 is sent and packet 1500 can be sent), for a total of 5.365 + 224.85 = 230.215 sec.
   (c) There are 50 groups and we need to wait one RTT for each. This is 50 RTTs plus the initial 2, for 7.8 seconds.
   (d) Right after the handshaking is done we send one packet. One RTT after the handshaking we send two packets. At n RTTs past the initial handshaking we have sent $1 + 2 + 4 + \ldots + 2^{n-1} = 2^n - 1$ packets. At n=10 we have thus been able to send 1023 packets. In the next batch we could send 1024 packets but there are only 477 left. Total time is 2 + 11 RTTs, or 1.95 sec.
8. Networking utility ping
   (a) The answer depends somewhat on the time of day that you use ping and the location of your network connection. Using ping on a windows machine, on campus, and a wired network connection, if found the average time to ucsd.edu to be 61 ms, google, 80 ms, and bellsouth 10 ms. One source of difference is because Clemson has two connections to the Internet. One is a connection to the "commodity" network, and the second is a connection with C-Light to "Internet-2" and other high-speed experimental networks. The experimental networks connect large research universities and certain government agencies. You can use ping on campus but the latency is so small that the time is usually reported as 0. If you try to ping from a laptop with only a wireless connection, you will probably find that the ping packets are blocked by a firewall.
   (b) For the first test to www.clemson.edu, on a unix machine in EIB I get

```
floyd% traceroute www.clemson.edu
traceroute to clemson.edu (130.127.69.75), 30 hops max, 40 byte packets
 1  130.127.198.2 (130.127.198.2)  0.662 ms  0.529 ms  0.510 ms
 2  130-127-1-241.generic.clemson.edu (130.127.1.241)  0.371 ms  0.317 ms  0.313 ms
 3  130.127.3.21 (130.127.3.21)  0.408 ms  0.368 ms  0.364 ms
 4  * * *
```

We see that the route goes through three switches that respond to tracert requests, after which a firewall protected network domain is reached and all subsequent packets are dropped.

Next, to ucsd.edu, there are no firewalls:

```
floyd% traceroute ucsd.edu
traceroute to ucsd.edu (132.239.180.101), 30 hops max, 40 byte packets
 1  130.127.198.2 (130.127.198.2)  0.675 ms  0.521 ms  0.506 ms
 2  130-127-1-241.generic.clemson.edu (130.127.1.241)  0.368 ms  0.313 ms  0.307 ms
 3  130.127.3.53 (130.127.3.53)  0.586 ms  0.545 ms  0.525 ms
 4  poole-border-01.clemson.edu (130.127.3.1)  2.981 ms  1.162 ms  1.313 ms
 5  130.127.3.214 (130.127.3.214)  3.915 ms  3.832 ms  3.834 ms
 6  143.215.193.1 (143.215.193.1)  4.909 ms  4.710 ms  4.397 ms
 7  hous-atla-70.layer3.nlr.net (216.24.186.8)  29.930 ms  29.145 ms  29.058 ms
 8  losa-hous-87.layer3.nlr.net (216.24.186.30)  59.956 ms  59.781 ms  59.752 ms
 9  hpr-lax-hpr--nlr-pn.cenic.net (137.164.26.149)  59.835 ms  60.568 ms  65.663 ms
10  hpr-ucsd-10ge--lax-hpr.cenic.net (137.164.26.26)  235.472 ms  224.090 ms  224.164
ms
11  nodem-720--ucsd-t320-gw-10ge.ucsd.edu (132.239.255.134)  62.085 ms  62.446 ms
62.053 ms
12  adcom1-nodem-720-p2p-10ge.ucsd.edu (132.239.254.82)  62.726 ms  62.161 ms  62.249
ms
13  ucsd.edu (132.239.180.101)  63.746 ms  63.365 ms  62.816 ms
```

Notice line 5, this is Clemson's connection to the National Lambda Rail (nlr.net) network, which is our entrance to the Internet-2. Lines 7 and 8 are routers operated by nrl.net (check their web page for a map of their network. Finally, lines 9 and 10 are for a network in California.

If you do a traceroute to google or bellsouth, you will notice a different link from the poole border router to off campus. Also, some routes do not respond to the traceroute packets.

(c) Of course the values that you see depend on the time of day. A few years ago when we tried this test, the web site reported an average delay of around 200 ms, and min and max can be highly variable. Sampling a few of the papers from two years ago showed that the min can be less than 20 ms and the max is occasionally above 800 ms. Last year we found the average delay a bit lower, around 90 ms or so. Looking at papers from this year, the average delay is a bit lower again, around 40 ms or so. Perhaps there continue to be improvements to the core network, or perhaps the web site has changed the set of routes it tests.