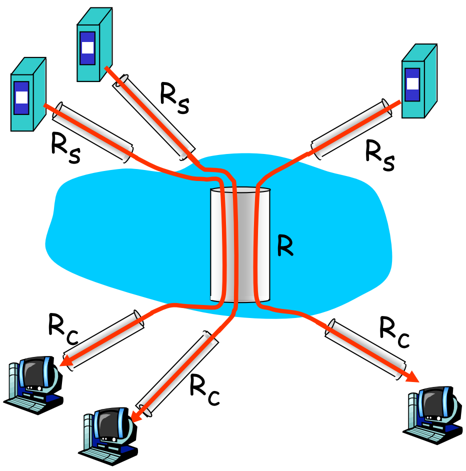
Introduction to Computer Networks
Midterm

1. (3pt) Circuit switching. There are several ways for circuit switching to divide link capacity into "*pieces*", each piece is exclusive for one user. This is also known as multiplexing. Identify **one** of such multiplexing techniques by writing down the term of terminology. ***(Points will only be awarded if the answer contains a correct term of terminology.)***
2. (6pt) Packet switching vs. circuit switching.
   (a) Identify one advantage of packet switching over circuit switching.
   (b) Identify one advantage of circuit switching over packet switching.
3. (6pt) Packet switching vs. circuit switching. Suppose users share a 3Mbps link. Each user requires 1Mbps when transmitting, but each user transmits only 50 percent of the time.
   (a) When circuit switching is used, calculate the maximum number of users that can be served simultaneously?
   (b) Suppose packet switching is used and there are 5 users. Calculate the probability that the link capacity is not big enough for the 5 users.
4. (6pt) Suppose two hosts, A and B, are separated by 10,000 kilometers and are connected by a direct link of R = 1 Mbps. Suppose the propagation speed over the link is 2.5×10^8 meters/sec.
   (a) Calculate the bandwidth-delay product, $R \bullet t_{prop}$. (t_{prop} denotes the propagation delay).
   (b) Consider sending a file of 400,000 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 time?
5. (3pt) Suppose there are M client-server pairs as shown in the figure below. Denote R_s, R_c and R for the rates of the server links, client links, and network link. Assume that all other links have abundant capacity and there is no other traffic in the network besides the traffic generated by the M client-server pairs. Derive a general expression for per-link throughput in terms of R_s, R_c, R, and M.



6. (2pt each) Answer **True/False** on each of these questions, respectively. You must answer these questions in order. Reasoning is ***not*** needed.
   (a) Compared to UDP, TCP is able to send packets at higher rate and enables finer application-level control over what data is sent, and when.
   (b) TCP provides congestion control and guarantees minimum throughput.
   (c) UDP segments for an application are delivered (to the application) in order.
   (d) A TCP socket is uniquely identified by two tuple—destination IP address and destination port

number.

(e) UDP adds no function to the IP protocol, except the multiplexing/demultiplexing function.

(f) UDP/TCP checksum at the sender side performs the 1s complement of the sum of all the 16-bit words in the segment, with any overflow encountered during the sum being wrapped around. Consider the case when exactly 1 bit of the received UDP/TCP segment is corrupted. UDP/TCP can guarantee to detect all 1-bit errors.

(g) Consider the case when exactly 2 bits of the received UDP/TCP segment is corrupted. UDP/TCP can guarantee to detect all 2-bit errors.

(h) UDP/TCP checksum can be used to correct bit errors.

(i) Cookie can be used to track users' states.

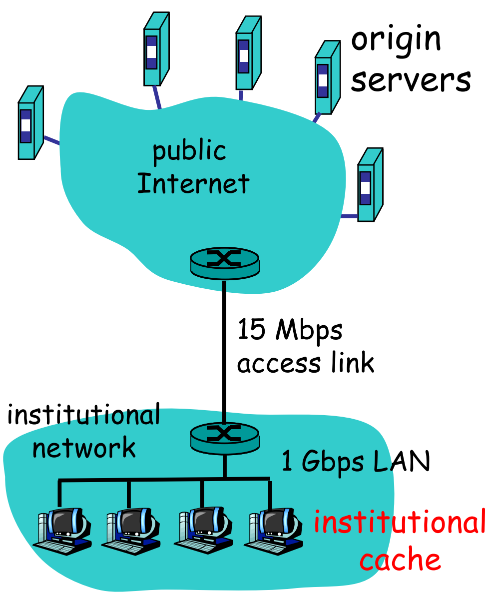
(j) HTTP records and maintains users' states.

(k) For Selective Repeat, an acknowledgement for a packet with sequence number *n* with be taken to be a cumulative acknowledgement.

(l) After receiving the HTTP request message given below, a Web server will use a persistent HTTP connection to send objects to the client.

```
GET /somedir/page.html HTTP/1.1
Host: www.someschool.edu
Connection: close
```

7. (6pt) Web proxy (cache). Consider the figure below, for which there is an institutional network connected to the Internet. Suppose that the average object size is 900,000 bits and that the average request rate from the institution's browsers to the origin servers is 15 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 200 milliseconds on average (that is, the average Internet delay is 200ms.). The access link transmits data at a rate of 15 Mbps. 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 $\Delta / (1 - \lambda \Delta)$, where Δ is the average time required to send an object over the access link and λ is the arrival rate of objects to the access link.

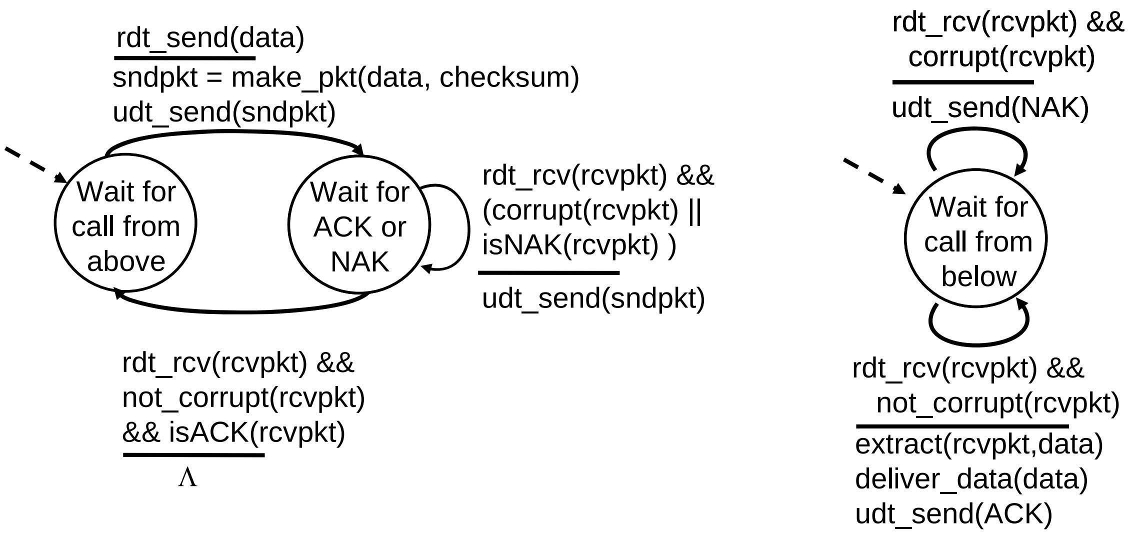


(a) Find the total average response time.

(b) Now suppose a cache is installed in the institutional LAN. Suppose the hit rate is 0.4. Find the total average response time.

8. (9pt) Non-persistent HTTP vs. persistent HTTP. Suppose a very small base HTML file references three very small objects on the same server. Let *RTT* denote the route-trip time between the local host and the server containing the objects. Ignoring transmission times and DNS lookup times, how much time elapses to receive the base HTML file and three other objects with:
   (a) Non-persistent HTTP with no parallel TCP connections.
   (b) Non-persistent HTTP with parallel TCP connections.
   (c) Persistent HTTP with pipelining.
9. (6pt) Go-Back-N protocol. Consider the Go-Back-N protocol with a sender window size of *n* and a large sequence number range ~~of *R* (that is, the sequence number is 0, 1, …, *R*-1)~~. Suppose that at time *t*, the next in-order packet that the receiver is expecting has a sequence number of *k*. Assume *i)* the medium does not reorder packets, and *ii)* the timeout interval is so long that data packets and ACKs (if not lost) have plenty of time to deliver by timeout. 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.
10. (6pt) Assume that all packets deliver (that is, there is no packet loss) but any packet may or may not corrupt. A rdt (reliable data transfer) protocol specified using FSM is shown below.

Sender side ***Receiver side***

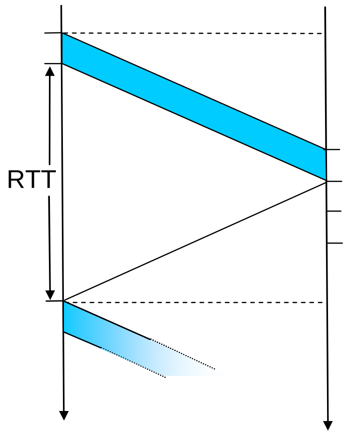


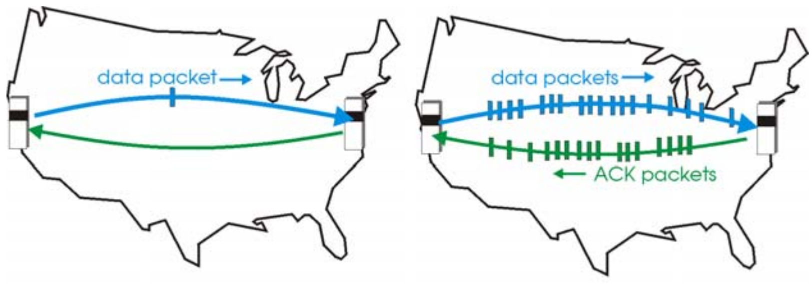
   (a) Identify one fatal flaw of the rdt protocol. (Under what condition, what bad thing will happen?)
   (b) Find a simple solution to the fatal flaw you point out in the 10(a). (**Briefly** write down and explain your solution using some keyword. No need to draw a modified FSM.)
11. (3pt) Sliding window size when the set of sequence numbers is of finite size. Suppose the sequence number space is of size *k*. Assume *i)* the medium does not reorder packets, and *ii)* the timeout interval is so long that all data packets (if not lost) and all ACKs (if not lost) deliver by timeout expiration.
   (a) With Selective Repeat, what is the largest allowable sender window that will avoid the occurrence of the problem that different packets with the same sequence number sent from the sender arrives in the receiver but the receiver cannot tell? Why?
12. (3pt) TCP SEQ# and ACK#. Host A and B are communicating over a TCP connection, and Host B

has already received from A all bytes up through byte 248. Suppose Host A then sends two segments to Host B back-to-back. The first and second segments contain 40 and 60 bytes of data, respectively. In the fist segment, the sequence number is 249, the source port number is 503, and the destination port number is 80. Host B sends an acknowledgement whenever it receives a segment from Host A.

(a) If the second segment sent from Host A to B arrives before the first segment, in the acknowledgement of the first arriving segment, what is the acknowledgement number?

13. (3pt) Pipelined protocols. After analyzing the utilization of a stop-and-wait protocol in class using the figure below, we can know that a stop-and-wait protocol doesn't perform well when the round-trip time is much larger than the transmission time per packet. Therefore, pipelined protocols are proposed to improve channel utilization. Assume the round-trip time (*RTT*) is 10 ms and the transmission time per packet is 1 μs. How big would the window size (in a unit of packets) of a pipelined protocol have to be for the channel utilization to be greater than 90%?





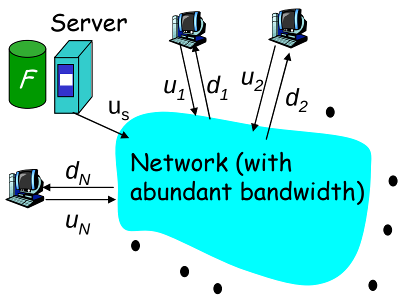
(a) a stop-and-wait protocol in operation

(b) a pipelined protocol in operation

14. (8pt) Consider distributing a file of F bits into N hosts, using a peer-to-peer architecture. Assume a fluid model. For simplicity, assume that $d_{\min} = \min\{d_1, d_2, \ldots, d_N\}$ is so large that peer download bandwidth is never a bottleneck.

(a) Suppose that $u_s \leq (u_s + u_1 + u_2 + \ldots + u_N) / N$. Specify a distribution scheme that has a distribution time of F / u_s.

(b) Suppose that $u_s \geq (u_s + u_1 + u_2 + \ldots + u_N) / N$. Specify a distribution scheme that has a distribution time of $NF / (u_s + u_1 + u_2 + \ldots + u_N)$.



15. (8pt) Consider the packet-switching scenario where packets may be buffered in a router buffer preceding an outbound link, before being sent out over the outbound link. Assume that when any packet enters the router, the probability that the there are n packets ahead in the router buffer and the outbound link is $p_n = (1 - \lambda / \mu)(\lambda / \mu)^n$. Assume that each of these $n + 1$ packets still needs an average transmission time of $1 / \mu$ seconds. Compute the average sojourn time. (Sojourn time is defined as the sum of queueing delay and transmission time.)