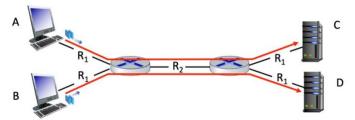
Midterm practice problems – 2

- 1. Which of the following are included in the packet delay? (a, b, c, d)
 - a. Time spent transmitting packets bits into the link.
 - b. Time needed for bits to physically propagate through the transmission medium from one end of a link to the other.
 - c. Time spent waiting in packet buffers for link transmission.
 - d. Time needed to perform an integrity check, lookup packet information in a local table and move the packet from an input link to an output link in a router.
- 2. What transport services are provided by UDP? (e) What transport services are provided by TCP? (b, c, f)
 - a. Throughput guarantee. The socket can be configured to provide a minimum throughput guarantee between sender and receiver.
 - b. Loss-free data transfer. The service will reliably transfer all data to the receiver, recovering from packets dropped in the network due to router buffer overflow.
 - c. Flow Control. The provided service will ensure that the sender does not send so fast as to overflow receiver buffers.
 - d. Real-time delivery. The service will guarantee that data will be delivered to the receiver within a specified time bound.
 - e. Best effort service. The service will make a best effort to deliver data to the destination but makes no guarantees that any particular segment of data will actually get there.
 - f. Congestion control. The service will control senders so that the senders do not collectively send more data than links in the network can handle.
- 3. Which of the following statements about the difference between TCP and UDP socket are correct? (a, c, d)
 - a. socket(AF_INET, SOCK_STREAM) creates a TCP socket while socket(AF_INET, SOCK_DGRAM) creates a UDP socket
 - b. Both TCP and UDP sockets can be fully identified by the destination IP address and destination port number.
 - c. A TCP server socket uses <code>socket.accept()</code> to accept client socket requests, while a UDP client socket does not require acceptance by the UDP server
 - d. "Two segments from different clients can be sent to the same destination IP address with the same destination port number." This statement is true for both UDP segments and TCP segments.
- 4. We learned that in HTTP web browser caching, HTTP local web server caching, and in local DNS caching, that a user benefits (e.g., shorter delays over the case of no caching) from finding a local/nearby copy of a requested item. In which of the following forms of caching does a user benefit not only from its own recent requests (and cached replies) but also from recent requests made from other users? (a, c)
 - a. Local DNS server caching
 - b. HTTP browser caching
 - c. HTTP local web caching

5. Consider the network shown below, with two senders (A,B) on the left, sending packets to two different receivers (C,D) on the right. Each sender is sending packets to its receivers over a separate TCP connection. The links have transmission rates of $R_1 = 100$ Mbps and $R_2 = 150$ Mbps. Suppose each packet is 1 Mbit in size.

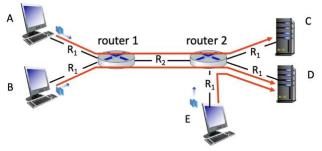


a. How long does it take for a sender to transmit a packet into its link (i.e., the packet transmission delay at sender A or B)?

$$D_{trans} = \frac{1 \, Mbit}{R_1} = 10 \, \text{msec}$$

- b. In the shared middle hop with capacity R_2 , bandwidth is initially distributed equally among connections passing through that link. If any connection does not fully utilize its assigned bandwidth, the unused bandwidth is then redistributed equitably among the remaining connections, ensuring a fair distribution. What is the maximum end-to-end throughput achieved by the A-to-C session, assuming both sessions are sending at the maximum rate possible?

 A-to-C and B-to-D sessions each get a transmission rate of $R_2/2 = 75 Mbps$ in the shared link. The throughput of A-to-C session is $\min(R_1, R_2/2) = 75 Mbps$
- c. Consider now a new sender (E) is added to the network, as shown below. Again, assume that the bandwidth of a shared link is initially distributed equally among connections passing through that link. If any connection does not fully utilize its assigned bandwidth, the unused bandwidth is then redistributed equitably among the remaining connections, ensuring a fair distribution. What is the maximum end-to-end throughput achieved by the new E-to-D session and the A-to-C session, assuming all senders are sending to receivers at the maximum rate possible?



In the initial distribution, A-to-C and B-to-D sessions each get a transmission rate of $R_2/2 = 75Mbps$ in the router1-router2 link.

E-to-D and B-to-D sessions each get a transmission rate of $R_1/2 = 50 \; Mbps$ in the router2-D link.

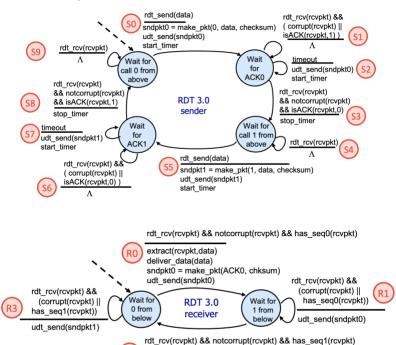
Throughput of E-to-D session: $min(R_1, R_1/2) = 50 Mbps$

Throughput of B-to-D session: $min(R_1, R_2/2, R_1/2) = 50 Mbps$

Since B-to-D session only uses 50 Mbps in the router1-router2 link, the unused 25 Mbps in the router1-router2 link will be re-distributed to A-to-C session.

Throughput of A-to-C session: $min(R_1, \frac{R_2}{2} + 25 \ Mbps, R_1) = 100 \ Mbps$

- 6. Consider the rdt 3.0 sender and receiver below, with FSM transitions labeled in red circle. What is the interleaved sequence of sender and receiver transitions that occur when:
 - (i) the sender sends a message;
 - (ii) the first message is lost;
 - (iii) a retransmission then occurs;
 - (iv) the retransmitted packet is correctly received;
 - (v) an ACK is then correctly sent and received.



extract(rcvpkt,data) deliver_data(data)

sndpkt1 = make_pkt(ACK1, chksum) udt_send(sndpkt1)

The sequence of transitions is: $SO \rightarrow S2 \rightarrow RO \rightarrow S3$