Homework 3

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1 R4

Since TCP provides reliable data transfer and flow control, it might compromise the data sending rate during congestion. Applications that do not require reliable data transfer but need high rate(for example, video conferencing application) prefers UDP over TCP.

2 R6

Yes, it is. Developer can do this by achieving reliability within the application itself, for example, using ACKs or NACKs and keeping a receiving window.

3 R10

If a packet is transmitted but the ACK for it never gets sent back, we have to retransmit the packet. The timer is used to determine whether retransmission is needed after a certain period of time(how long after a packet is sent without an ACK will be considered packet loss).

4 R14

(a) False. Host B will send acknowledgments to Host A.

(b) False. The size of rwnd changes.

(c) True.

(d) False. Maybe the size of the segment is not 1.

(e) True.

(f) False. TimeoutInterval might be less than Sample RTT.

(g) False. The acknowledgement number might be less than 42.

5 R15

(a) 110 - 90 = 20 bytes

(b) The acknowledgement number will be 90 since first segment hasn't been received.

6 P3

The sum of first two 8-bit bytes are:
01010011 + 01100110 = 10111001
The sum of three 8-bit bytes are:
10111001 + 01110100 = 100101101
Wrap around: 00101110, so the 1s complement of the sum of these 8-bit bytes is 11010001

UDP takes 1s complement of the sum because the 1s complement of the sum is the checksum which the receiver host uses to check if errors occur in the segment.

The receiver detects errors in this way:
All the bytes plus checksum are being added at the receiver's end. If the sum has any 0s in its bits, there are errors. Otherwise there is no error(all 1s in its bits).

1 bit errors will not go undetected since if there's any bit as 0 in the sum, it will be detected.
2 bit errors might be undetected since the sum might appear the same for different bits.

7 P5

No, the receiver cannot be absolutely certain that no bit errors have occurred. This is because for example, if bits in two 16 bit words are flipped, the sum will remain the same and the errors will not be detected using checksum.

8 P14

No. A NAK-only protocol would not be preferable to a protocol that uses ACKs, because a receiver will not know if packets have been lost until it receives following packets(packets with higher sequence number). Especially in this case, the sender sends data infrequently. So in this way, the sender cannot know in time if it has to retransmit any lost packet.

9 P15

dtrans $= L/R = 1500*8bits/10^9 bitspersecond = 0.000012seconds = 0.012milliseconds$
Since $0.98 = windowsize * \frac{L/R}{RTT+L/R}$
$0.98 = windowsize * \frac{0.012}{30+0.012}$
So windowsize = 2451 packets.

10 Problem 10

No, we cannot use TCP with multicast because TCP is connection oriented protocol for communicates between two endpoints. If connections are established with TCP between server and multiple clients, the network will get very congested and delayed.
We can build an alternative reliable protocol by first establish TCP connections between server and some medium. Then let the medium multicast data using UDP protocol. But we achieve reliability of the protocol within applications of receiving hosts(using ACKs), and the medium which keep copies can help server retransmit lost packets.

11 P28

Since host A and host B are connected with a 100Mbps link, host A cannot send data at a rate exceeding 100Mbps. Host B reads and removes data from the receiving buffer at a rate of 50 Mbps. When the receiving buffer is filled up, host B sets receive window to be 0, and sends this information to host A indicating no more data should be sent. Then host A waits until it gets the information that host B's receiving window is greater than 0. Then host A restarts to transmit data.

12 Problem 12

The server will experience a denial of service attack. Its resources will be overflown by the flood of forged requests, thus unresponsive to legitimate traffic. So at some point it will be unable to process incoming connections.

13 P33

TCP avoids measuring the SampleRTT for retransmitted segments because it is hard to track the RTTs for retransmitted segments. Since TCP doesn't know how many segments are retransmitted, so calculating the RTT according to the ACK of retransmitted packets is not possible. It does not know which one that ACK corresponds to.

14 P55

(h) The server will send its response to address Y.

(i) Yes, the server can be certain that the client is Y.
Since in the ACK, the client not only has to send back the ack which is server_isn+1, it also has to send back the correct sequence number. It is not possible for the attacker to send the right sequence number which is randomly initialized by the server.