COMS W4119: Computer Networks
Homework 3

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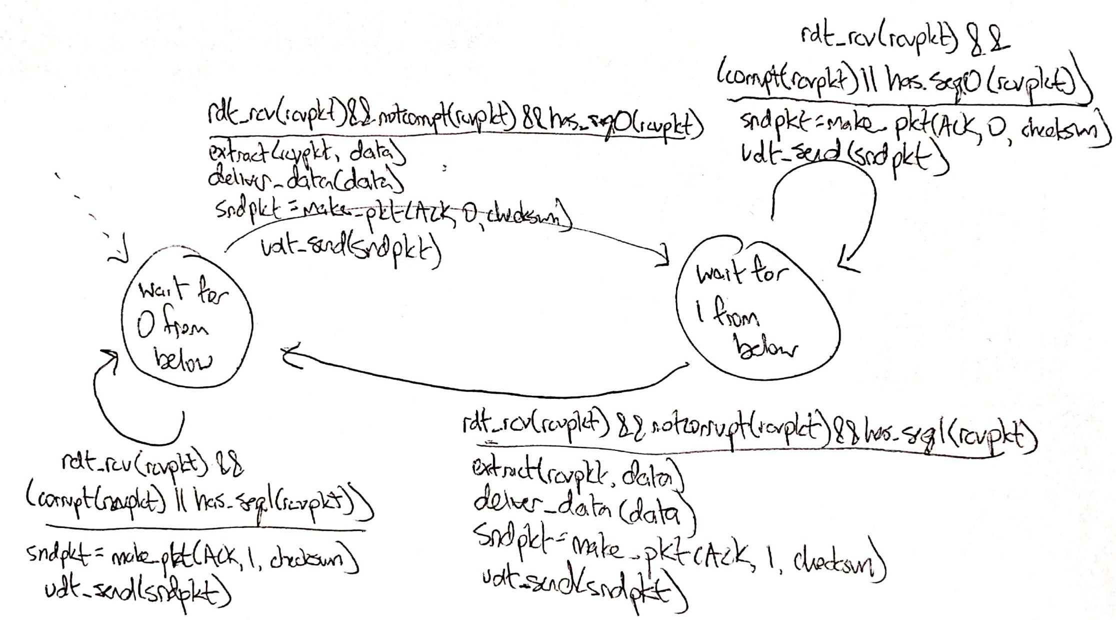
Principles of Transport and Reliable Data Transfer

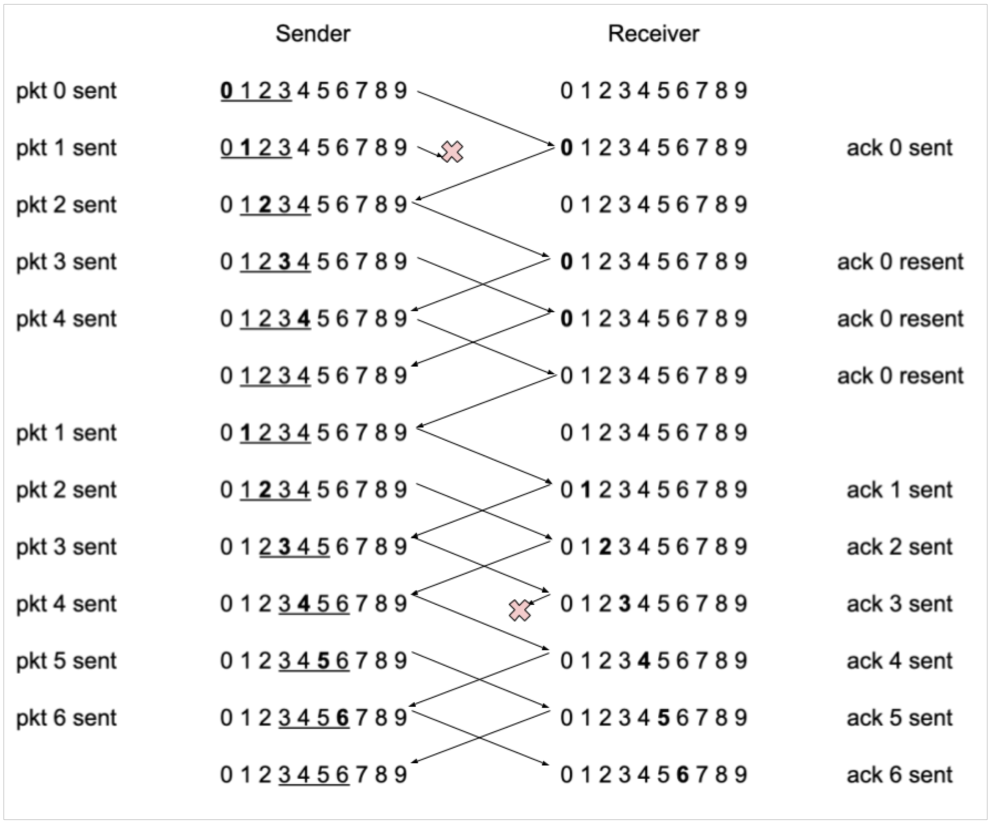
(a) (i) A timeout based retransmission mechanism can deal with both losses in sending packets and receiving ACK's. If the timer times out when waiting for ACK from receiver, it will just resend the packet. This will cover both situations where packets sent or ACKs received are lost. In addition, if the packet wasn't actually lost and it just took exceptionally long for the receiver to respond with the ACK, the packets are numbered and duplicate packets can be dealt with.

(ii) The receiver FSM of rdt3.0 should be exactly the same as that of rdt2.2

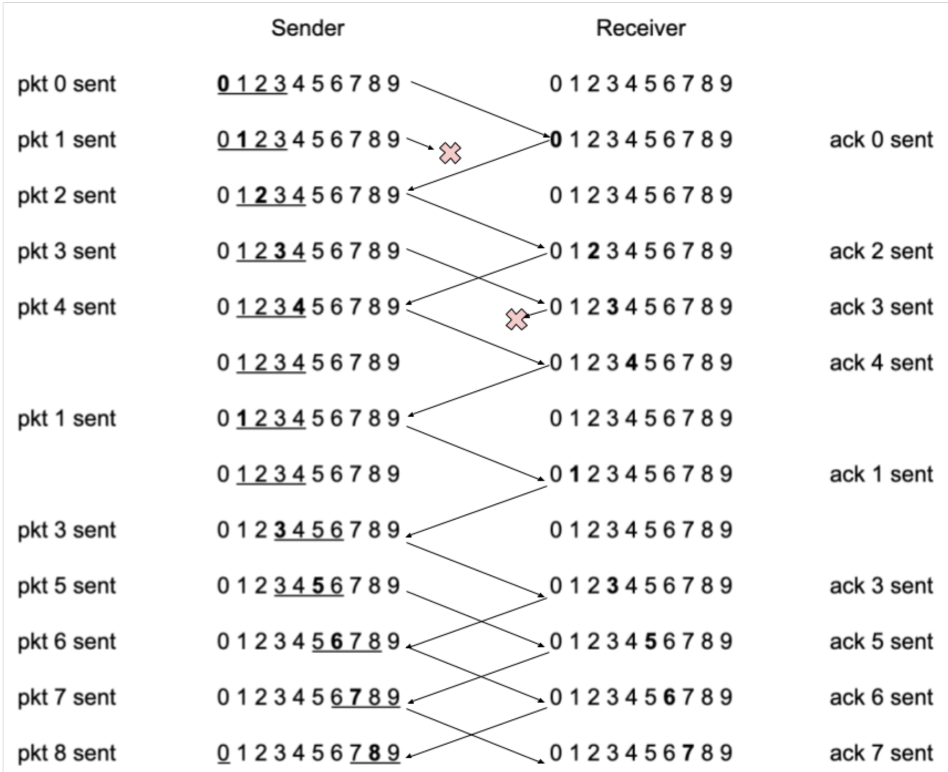


(b) The underlined numbers are the packets that are within the window, and the bolded packets are the ones sent on either side.

Go Back N:



Selective Repeat:



(c) (a) SR protocol must have a window size that less than or equal to half of n. This upper bound exists for the window size because of the worst case scenario: All of the ACK's sent by the receiver are lost. Say the window size is w, then the receiver will increment its window by w places. If $w > \frac{n}{2}$, then the receiver's window will wrap around back to sequence number 0 again. This means, the receiver will confuse the sender's inevitable retransmission of packet 0 as a new packet rather than as a retransmission. Therefore, the window size w must be less than or equal to half of n.

(b) GBN protocol must have a window size less than or equal to $n-1$ (i.e the window size must be smaller than n). This is because the receiver does not buffer any out-of-order packets. Thus in the worst case if all of the receiver's ACKs are lost, and the window size is w, the receiver will refuse to process any out of order packets (i.e sequence number that is not $w+1$). This means the sender is free to retransmit the previous packets and there is no concern of the receiver confusing the retransmitted packets as new.

(d) (i) True. If all of the receiver's ACKs are lost, then the receiver's window would have advanced, while the sender's has not. Thus the sender will retransmit those packets and those packets will be outside of the receiver's window.

(ii) True. Say the receiver's initial ACKs take a very long time to send. The sender will retransmit those packets, and say the receiver's second set of ACKs on those packets get delivered quickly, causing the sender to advance its window. Then the receiver's intial ACKs get delivered, then they fall outside of the sender's current window.

(iii) True. The situation is exactly the same as part (ii).

(iv) False, the receiver will not return ACK's for any packet sequence higher than expectedseqnum, which means there is no way the sender could advance its window above expectedseqnum.

Connectionless Transport

(a) The checksum is the 16 bit one's complement of the the one's complement sum of three things:

- Pseudo header (which contains sum of the source IP, destination IP, protocol, and UDP length)
- UDP header
- Data

This number may be padded with zero octets at the end, if necessary, to make a multiple of two octets. [1]

(b) First convert everything to hexadecimal:

$$\begin{aligned} \text{source IP} &= \texttt{0x44ACF5AA} \\ \text{destination IP} &= \texttt{0xACD90A0E} \\ \text{length} &= \texttt{0xE745} \\ \text{UDP protocol} &= \texttt{0x0011} \\ \text{source port} &= \texttt{0x10E1} \\ \text{destination port} &= \texttt{0x162E} \end{aligned}$$

Thus to calculate the 16 pseudo header, we add the source and destination IP, protocol and UDP length. Note that we split the IP addresses into two 16-bit segments:

$$\text{pseudo header} = \texttt{0x44AC} + \texttt{0xF5AA} + \texttt{0xACD9} + \texttt{0x0A0E} + \texttt{0x0011} + \texttt{0xE745} = \texttt{0x2D893}$$

Now we can calculate the sum of the pseudo header with the UDP header and data (we will ignore the data since the payload is all zeros):

$$\text{sum} = \texttt{0x2D893} + \texttt{0x10E1} + \texttt{0x162E} + \texttt{0xE745} = \texttt{0x3E6E7}$$

Now convert the above hexadecimal into a 16 bit one's complement:

$$(\texttt{0x0003} + \texttt{0xE6E7}) \oplus \texttt{0xFFFF} = \boxed{\texttt{0x1915}}$$

(c) The receiver checks for errors by computing the one's complement sum of the received data (which includes the checksum). If any of output bit's are 0, then an error is detected. The one's complement scheme makes it so that supposedly matching data should have an output of all 1's, making it very easy to computationally check.

(d) The checksum can fail to catch errors. The most obvious example would be that swapping 16-bit blocks does not change the computed checksum since summation is commutative. Thus, in part (b), if the source port and destination port were swapped (such that the destination port was 4321 and the source port was 5678), then the checksum would remain exactly the same!

[1] https://tools.ietf.org/html/rfc768

(e) The theoretical maximum datagram size is 65,535 bytes since the datagram length field is 16 bits long.

(f) The size of UDP datagram cannot reach the theoretical limit in (e) because the length includes the IP header. The specified source and destination addresses are already 8 bytes each, which will detract from the maximum possible size.

(g) DHCP uses UDP because it is connectionless, which makes is much faster rather than needing to establish a 3 way handshake for every communication, which would be infeasible for a DHCP server sending outbound packets to many hosts. Furthermore, because the source does not yet have a fully configured IP address, this can cause problems conducting the 3 way handshake.

(h) Loss based congestion control have two issues, primarily revolving around the fact that (1) lost packets do not necessarily mean traffic congestion, and also (2) congestion does not necessarily mean packet loss:[2]

- In shallow buffers, packet loss may occur from transient traffic bursts, which halves the sending rate and results in less throughput than what the link could easily handle for most of the time
- In deep buffers, congestion may occur at deep buffers in many last-mile links, and this will cause a lot of delay. Loss-based congestion control can't account for this since no packets are lost due to the deep buffers.

(i) In steady state, BBR uses the bottleneck bandwidth and round-trip propagation time to determine the rate and amount of packets to send through the network. [3]

(j) By leveraging the bottleneck bandwidth and round-trip propagation, BBR is able to ensure that the bottleneck runs at 100% utilization, and also guarantees there is enough data to "prevent bottleneck starvation but not overfill the pipe". In otherwords, the bottleneck ultimately determines the congestion level of the network, and by pacing packets to be exactly what the bottleneck can handle would maximize the number of packets sent without causing overflowing congestion.[4]

Connection-Oriented Transport

(a) If Host Y receives the first segment normally, the ACK will return the number number 100. If Host Y receives the second segment first, the ACK will return the number 80.

(b) (i) From the table below, the EstimatedRTT is $\boxed{89.99}$ ms.

[2] https://cloud.google.com/blog/products/gcp/tcp-bbr-congestion-control-comes-to-gcp-your-internet-just-got-faster
[3] https://cacm.acm.org/magazines/2017/2/212428-bbr-congestion-based-congestion-control/fulltext
[4] Ibid.

Time	SampleRTT	EstimatedRTT
1	120	120
2	25	110.5
3	35	102.95
4	40	96.66
5	30	89.99

SampleRTT5 has the largest impact on the EstimatedRTT because the alpha value is 0.1, giving 9 times more weight to the previous estimate than to the current sample. This disproportionately adds weight to prior estimates.

(ii) Given n SampleRTT's, with SampleRTT_1 being the most recent one:

$$\text{EstimatedRTT} = \left(\alpha \sum_{i=1}^{n-1} (1-\alpha)^{i-1} \text{SampleRTT}_i\right) + (1-\alpha)^{n-1} \text{SampleRTT}_n$$

Given $\alpha = 0.1$:

$$\text{EstimatedRTT} = \left(0.1 \sum_{i=1}^{n-1} 0.9^{i-1} \text{SampleRTT}_i\right) + 0.9^{n-1} \text{SampleRTT}_n$$

(c) The information in the last packet when establishing a connection is the client's ACK for the server's initial sequence number. This packet is necessary as a TCP connection is meant to provide for bi-directional communication, and thus without it, the server hasn't finished synchronizing its initial sequence number with the client (it hasn't been ACK'ed yet). This prevents the server from sending communication over the connection, and hence breaks the bidirectionality of a TCP connection.

(d) In the client's first packet to the server, it will set the SYN bit in the flag field to be 1. Additionally, the sequence number field will also be set to the client's choosing.

In the server's return packet to the client, the SYN bit in the flag field is also set to 1. The acknowledgement field in the header is set to be the client's ISN + 1. The sequence number field in the header is also set to the server's choosing.

In the third packet of the handshake (client's second packet to the server), the acknowledgement field in the header is set to be the server's ISN + 1. The SYN bit in the flag field is also set to 0.

(e) The FIN flag is used to close a connection. The other end sends an acknowledgement to complete the closure.

(f) Only two handshake packets are needed to establish a unidirectional, reliable connection-oriented protocol. This is because only A's ISN needs to be synchronized across the two servers in order for B to reliably receive A's messages. The process would look like the following: A will send a SYN packet with its ISN to B, and then B needs to respond with an ACK packet to A. Now A knows that its can reliably send messages to B starting at that ISN, and communciation and henceforth begin.

(g) Only two steps are needed to close the connection started by A, since only one set of buffers and resources needs to be deallocated. It would just be one FIN packet sent by the initiator of the closure, and an ACK packet returned.

(h) The TCP sender only maintains the smallest sequence number of a transmitted but unACK'ed byte as well as the sequence number of the next byte to be send, making the sender side look a lot like the GBN protocol.

(i) On the receiver's side, many TCP implementations will correctly buffer out-of-order segments that are received, which is similar to a Selective Repeat behavior. On the sender's side, retransmission also works like Selective Repeat, where it would only the specific retransmit segments that are timed out.

(j) Just using NAK's is not sufficient to allow A to tell the status of the segments. In the case where B is only expecting seg1, and say receives seg1 and seg2 (this means seg3 and seg4 dropped). B will not respond with any NAK's, since it doesn't know that it was supposed to receive seg3 and seg4, and B's lack of response makes A believe that all of the segments were successfully sent, when in reality 2 of the segments actually dropped.

(k) The super fast retransmission does not work well in situations where multiple packets are lost successively. For example, let's say seg2, seg3, seg4, and seg5 were all dropped (so the packet status is 100001). The receiver will send an ACK for seg1 after receiving seg1. Then once the receiver gets seg6, it will do one of the following:

- In the super fast retransmission protocol, the receiver will return a duplicate ACK for seg1 since it received an out-of-order segment with a higher than expected sequence number (seg6). This would be a duplicate ACK on the sender's end, so it would send seg2, which the receiver would ACK once it receives. Then the sender's timer for seg3, seg4, seg5 and seg6 would expire and it would retransmit those packets. The receiver receives all of those packets and then ACK's all of them accordingly.
- In the NAK protocol, the receiver will send a NACK for segments 2 through 5, since it received segment 1 and segment 6 (and knows that it is out of order). The sender will receive the NACK, and then retransmit segments 2 through 5. The receiver will receive them and ACK all of them.

From the two cases above, the NACK protocol improved overall performance because it retransmitted fewer segments (it did not need to retransmit segment 6 twice as it did in the super fast retransmission case), and also it started retransmission earlier for segments 3 through 5 – i.e when it received the NACK rather than when it waited for the timer to expire in the super fast retransmission case.

(l) The above case worked because propagation delay is larger than transmission delay, which means that the sender would likely have sent all of the segments out before hearing any response from the receiver. This makes multiple sequential dropped segments even more costly because the sender was unable to receive feedback as it was sending each individual segment (and thus ultiamtely relying on the timer to figure out seg3 through seg5 were dropped).

The above case is unlikely and limited because it required the propagation delay to be so much larger than transmission delay that all of the sender's segments were enroute by the time the first response reached the sender. The above case also assumed that NACK's can't be dropped enroute, and also ignores cases when the later segments were dropped (without the receiver's knowledge that it was supposed to receive them)

Methods of Congestion Control and avoidance

(a) (1) Flow control prevents the receiver's buffer from being overwhelmed (sender bears the responsibility), while congestion control prevents the network (router buffers) from being overwhelmed (transport layer bears responsibility).

(2) The point of congestion control is to ensure everyone on the network has fair access to the resources, while the point of congestion control is ensure smooth communication between two people

(b) This would technically work but be terribly inefficient. Congestion control would never replace flow control because the premise of congestion control is that the sender is unsure how congested the networks are (how many other people are on the network, how heavy is their usage, what is the state of the router buffers), and thus attempts to maximize the amount of packets sent without overloading the network. However, when just sending to another person, their buffer size is known, and the amount of remaining space on the buffer is also known. Thus, there is no need to optimize the maximum throughput, and instead the known number can just be used.

Furthermore using loss controlled flow control is also fundamentally inefficient because retransmission of the lost packets is required, when compared to a flow control algorithm that can guarantee no packet losses on the client's end.

(c) During Fast Recovery, packet retransmission occurs when the sender receives 3 duplicate ACKs. It then sends half the number of packets that it previously sent (congestion window cut in half).

During normal recovery, packet retransmission occurs when the sender witnesses a timeout occuring for a packet that it sent. It then cuts its congestion window to 1 MSS.

(d) Neither fast nor normal recovery definitely mean a packet was lost. This is because different packets may take different routes based on the state of the congestion, and hence packets may arrive out of order, causing 3 ACK's to go out before the missing packet arrives. Timeouts may also occur without packet loss due to network congestion.

(e) Congestion window is halved during fast recovery. The window is set to equal to 1 MSS during normal recovery.

(f) Relying on too few duplicate ACK's does not signal strong confidence that the packet was lost (because packets can be re-ordered), thus retransmitting in this case makes it likely that duplicate work is being done (as the packet is unlikely to have been lost).

Relying on too many duplicate ACK's waits beyond a reasonable measure of confidence that the packet was lost, effectively wasting time before starting the retransmission process.

(g) In networks where the buffer sizes and RTT's are fairly homogenous, Vegas would perform better as it would not be forced to wait for packets to be lost (which would require retransmission), and can take preventative efforts to scale back.

However, in situations where the network is very heterogenous, Reno may be better as packets can be routed differently based on congestion levels and one high RTT for a packet does not necessarily mean the network is on the brink of congestion collapse.

(h) My answer would be different, because TCP Vegas is based on the premise of RTT rather than packet loss. Looking for duplicate ACK's is a measure of confidence of packet loss rather than RTT, and thus a completely different measure than what Vegas is using.

(i) A Vegas network with a Reno network would not be fair, as the linear increase and linear decrease fails to move the connection throughout to the equal bandwidth share (the multiplicative decrease is the key that decreases the throughput proportionally to the equal bandwidth share).

For streaming videos, I would not use Vegas – I would still use UDP, as UDP does not abide by fairness congestion control and videos do not want their transmission rate to be throttled.

(j)

$$(6, 14) \text{ and } (15, 26)$$

(k) 32 MSS

(l) A packet was lost by detecting 3 duplicate ACKs and fast recovery was implemented (congestion window halved). This event definitely indicates that at least one packet was sent, as the receiver only sends a duplicate ACK when receiving an out of order segment.

(m) A packet was lost due to a timeout, thus normal recovery occured (where the congestion window was set to 1 MSS)

(n) No this does not definitely indicate that the network discarded a packet. Network congestion could have delayed the packet enroute such that the sender's timer expired before it arrived.

(o) The threshold immediately before round 27 is 20 MSS (because it is half the congestion window size at the previous loss event, which was at 40).

Round 28 is at 1 MSS. After round 28 the congestion window starts at 2 (round 29). Assuming no losses, the congestion window will grow to 4, then 8, then 16, then exceed the previous threshold on round 33.

Assuming that the packets are able to be put on the link in quick succession (i.e negligible time difference between when the packets are put on the link), each round takes one RTT to complete. It will take four rounds, and thus 4 RTT's, or **400 ms** before the congestion window size reaches the previous threshold.