CS5222 Assignment 2 (Sem A, 2025-26)

Chapter 3

P8. Draw the FSM for the receiver side of protocol `rdt3.0`.

Problem 8

The sender side of protocol rdt3.0 differs from the sender side of protocol 2.2 in that timeouts have been added. We have seen that the introduction of timeouts adds the possibility of duplicate packets into the sender-to-receiver data stream. However, the receiver in protocol rdt.2.2 can already handle duplicate packets. (Receiver-side duplicates in rdt 2.2 would arise if the receiver sent an ACK that was lost, and the sender then retransmitted the old data). Hence the receiver in protocol rdt2.2 will also work as the receiver in protocol rdt 3.0.

P22. Consider the GBN protocol with a sender window size of 4 and a sequence number range of 1,024. Suppose that at time *t*, the next in-order packet that the receiver is expecting has a sequence number of *k*. Assume that the medium does not reorder messages. 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.

Problem 22

a) Here we have a window size of N=4. Suppose the receiver has received packet k-1, and has ACKed that and all other preceding packets. If all of these ACK's have been received by sender, then sender's window is [k, k+N-1]. Suppose next that none of the ACKs have been received at the sender. In this second case, the sender's window contains k-1 and the N packets up to and including k-1. The sender's window is thus [k-N,k-1]. By these arguments, the senders window is of size 4 and begins somewhere in the range [k-N,k].

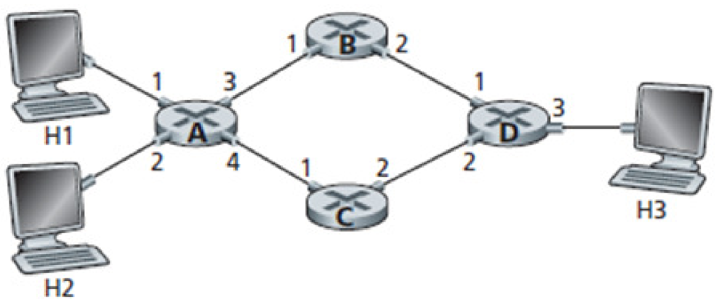
b) If the receiver is waiting for packet k, then it has received (and ACKed) packet k-1 and the N-1 packets before that. If none of those N ACKs have been yet received by the sender, then ACK messages with values of [k-N,k-1] may still be propagating back.Because the sender has sent packets [k-N, k-1], it must be the case that the sender has already received an ACK for k-N-1. Once the receiver has sent an ACK for k-N-1 it will never send an ACK that is less that k-N-1. Thus the range of in-flight ACK values can range from k-N-1 to k-1.

<u>Chapter 4 & 5</u>

P4. Consider the network below.

a. Suppose that this network is a datagram network. Show the forwarding table in router A, such that all traffic destined to host H3 is forwarded through interface 3.

b. Suppose that this network is a datagram network. Can you write down a forwarding table in router A, such that all traffic from H1 destined to host H3 is forwarded through interface 3, while all traffic from H2 destined to host H3 is forwarded through interface 4? (Hint: this is a trick question.)



Problem 4

a) Data destined to host H3 is forwarded through interface 3

Destination Address	Link Interface
H3	3

b) No, because forwarding rule is only based on destination address.

Paper reading

Please read the paper "S. Savage *et al.*, "Detour: informed Internet routing and transport," in *IEEE Micro*, vol. 19, no. 1, pp. 50-59, Jan.-Feb. 1999, doi: 10.1109/40.748796" and answer the following questions.

According to this paper:

1) What are the inefficiencies in the network-layer protocol?
2) What are the inefficiencies in the transport-layer protocol?
3) How to address these inefficiencies?

(**Note**: Please do not copy and paste the answers directly from the paper. Rather, we hope that you apply the knowledge you learn from this course to enrich your discussion here.)

Suggested solution:

1) What are the inefficiencies in the network-layer protocol?

   The network layer provides services to direct packets to a destination host on another network. To travel to other networks, the packet must be processed by a router. The role of the router is to select paths for and direct packets toward the destination host in a process known as routing. The potential sources of routing inefficiencies including: 1. poor routing metrics; 2. restrictive routing policies; 3. manual load balancing; 4. single-path routing.

2) What are the inefficiencies in the transport-layer protocol?

   The transport-layer ensures the establishment and maintenance of end-to-end connections, reliable end-to-end transmission of data packets, flow control and congestion control. The latency of network will limit host perceive changes in recipient resources. As latency increases, the host must wait longer to receive acknowledgments indicating that more data can be sent, and consequently will send more slowly. As the packet loss rate increases, the host assumes that congestion is occurring and will also send more slowly. TCP is the main transport layer protocol for wired networks. TCP adapting several mechanisms to prevent the congestion collapse events, such as slow start, congestion avoidance, time-outs and fast retransmit.

3) How to address these inefficiencies?

There are three opportunities to address routing inefficiencies:

1. Use real performance metrics to choose routes within the routing system.
2. Apply dynamic multipath routing to avoid congestion by randomly assigning flows to good paths.
3. Specialize routing decisions to the needs of different service classes.

Improving latency and packet drop rates automatically improves transport performance. TCP contains several mechanisms to prevent the congestion collapse events, such as slow start, congestion avoidance, time-outs and fast retransmit.