Section 13: Networking

CS 162

Friday 26, 2021

Contents

1 Vocabulary

- **TCP** - Transmission Control Protocol (TCP) is a common L4 (transport layer) protocol that guarantees reliable in-order delivery. In-order delivery is accomplished through the use of sequence numbers attached to every data packet, and reliable delivery is accomplished through the use of ACKs (acknowledgements).

2 Networking

a) (True/False) IPv4 can support up to 2^{64} different hosts.

False, it has 32 bits per IP address

b) (True/False) Port numbers are in the IP header.

False, they are in the transport layer. (TCP/UDP).

c) (True/False) UDP has a built in abstraction for sending packets in an in order fashion.

False, this is a part of the TCP protocol. In UDP there is no notion of a sequence number.

d) (True/False) TCP provide a reliable and ordered byte stream abstraction to networking.

True.

e) (True/False) TCP attempts to solve the congestion control problem by adjusting the sending window when packets are dropped.

True.

f) In TCP, how do we achieve logically ordered packets despite the out of order delivery of the physical reality? What field of the TCP packet is used for this?

The seqno field.

g) Describe the semantics of the acknowledgement field and also the window field in a TCP ack.

The acknowledgement field says that the receiver has received all bytes up until that number (x). The window field says how many additional bytes past x the receiver is ready to receive.

h) List the 5 layers specified in the TCP/IP model. Layering adds modularity to the internet and allows innovation to happen at all layers largely in parallel. What is the function of each layer?

1) Physical Layer: the physical layer is responsible for the delivery of raw bits from one endpoint to another. It includes ethernet, fiber, and other mediums of data transmission.
2) Datalink Layer: this layer adds the packet abstraction and is responsible for the local delivery of packets between devices within the same network (usually a LAN but can also include WANs). Devices here include switches, bridges, and network interface cards (NICs).
3) Network Layer: this layer is the skinny waist of the internet and routing algorithms here only speak IP. The network layer is responsible for global delivery of packets across one or more networks.
4) Transport Layer: this layer provides host to host or end to end communication. Because processes operate in terms of streams of data rather than individual packets, the transport layer handles the multiplexing of packets into streams, end to end reliable delivery, and introduces the concept of flows. This layer lives in the operating system, and TCP and UDP are examples of popular transport layer protocols.
5) Application Layer: the application layer provides network support for applications. The

protocols that live here include: HTTP, FTP, DNS, SSH, TLS, etc.

i) The end to end principle is one of the most famed design principles in all of engineering. It argues that functionality should **only** be placed in the network if certain conditions are met. Otherwise, they should be implemented in the end hosts. These conditions are:

- Only If Sufficient: Don't implement a function in the network unless it can be completely implemented at this level.
- Only If Necessary: Don't implement anything in the network that can be implemented correctly by the hosts.
- Only If Useful: If hosts can implement functionality correctly, implement it in the network only as a performance enhancement.

Take for example the concept of reliability: making all efforts to ensure that a packet sent is not lost or corrupted and is indeed received by the other end. Using each of the three criteria, argue if reliability should be implemented in the network.

i) Only If Sufficient

NO. It is not sufficient to implement reliability in the network. The argument here is that a network element can misbehave (i.e. forwards a packet and then forget about it, thus not making sure if the packet was received on the other side). Thus the end hosts still need to implement reliability, so it is not sufficient to just have it in the network.

ii) Only If Necessary

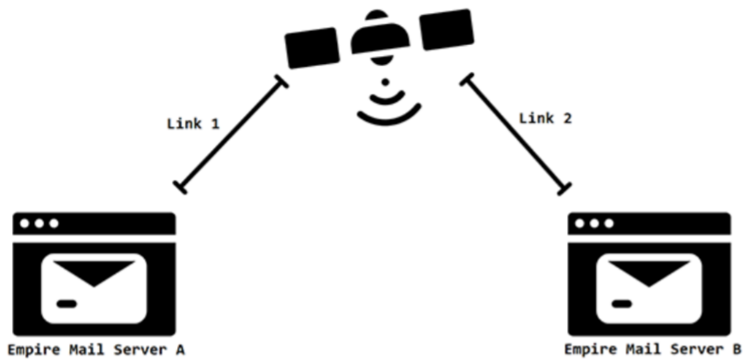
NO. Reliability can be implemented fully in the end hosts, so it is not necessary to have to implement it in the network.

iii) Only If Useful

Sometimes. Under circumstances like extremely lossy links, it may be beneficial to implement it in the network. Lets say a packet crosses 5 links and each link has a 50% chance of losing the packet. Each link takes 1 ms to cross and there is an magic oracle tells the sender the packet was lost. The probability that a packet will successfully cross all 5 links in one go is $(1/2)^5 = 3.125\%$. This means the end hosts need to try 32 times before it expects to see the packet make it through, taking up to # of tries $\times$ max # of links per try $= 32 \times 5 =$ 160 ms. Likewise at each hop, if the router itself is responsible for making sure the packet made it to the next router, each router would know if the packet was dropped on the link to the next router. Thus each router only has to send the packet until it reaches the next router, which will be twice on average. So to send this packet, it will take on average # of tries per link # number of links $= 2 \times 5 = 10$ ms. This is a huge boost in performance, which makes it useful to implement reliability in the network under some cases.

3 SP21 MT3 Stardust

Grand Moff Tarkin wants to send the blueprints over interstellar satellite connections, from Empire Mail Server A to Empire Mail Server B. Empire Mail Server A's connection to the satellite is denoted Link 1, while Empire Mail Server B's connection to the satellite is denoted Link 2.



Assume the following information:

- 120 ms latency from either server to the satellite, and vice versa (both Link 1 and Link 2)
- 10 ms latency for the router machinery within the satellite; assume that the router can process packets at full line rate (i.e. as fast as needed by the network)
- Maximum Transfer Unit of 1,000 B (both Link 1 and Link 2)
- Bandwidth over Link 1: 200 GiB/s
- Bandwidth over Link 2: 250 GiB/s
- TCP/IP Header size: 50 B

Note that the bandwidth is given in GiB/s for convenience, instead of Gb/s, which would typically be used.

1. Assuming that Empire Mail Server A and Empire Mail Server B are communicating via a mail-transfer protocol carried over TCP/IP, what is the maximum sustained communications bandwidth (excludes overhead) that the two servers could achieve over the satellite connection?

   (1000B - 50B)/1000B * min(200GiB/s, 250GiB/s) = 190GiB/s

2. Assuming that Empire Mail Server A and Empire Mail Server B are communicating via a mail-transfer protocol carried over TCP/IP. Based on the given assumptions, what is the Round Trip Time (RTT) for a 512 B packet?

   2 * (120ms + 10ms + 120ms) = 500ms