Networks and Distributed Computing — Spring 2019 — Homework 3 $\,$

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1 TCP Demo Program

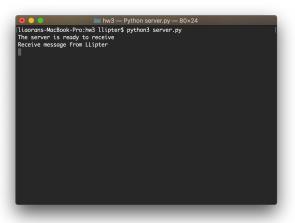


图 1: server.py

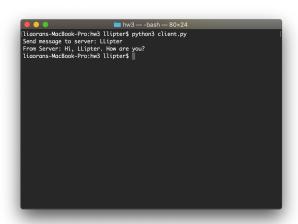


图 2: client.py

Suppose Client A requests a web page from Server S through HTTP and its socket is associated with port 33000.

- (a) What are the source and destination ports for the segments sent from A to S?

 Source port is 33000 and destination port is 80.
- (b) What are the source and destination ports for the segments sent from S to A? Source port is 80 and destination port is 33000.
- (c) Can Client A contact to Server S using UDP as the transport protocol?

 No. HTTP is built upon TCP.
- (d) Can Client A request multiple resources in a single TCP connection

 Yes. there's no limit on how many data can be transported through a single TCP connection.

Consider Figure 3.5. What are the source and destination port values in the seg- ments flowing from the server back to the clients' processes? What are the IP addresses in the network-layer datagrams carrying the transport-layer segments?

Suppose the IP addresses of the hosts A, B, and C are a, b, c, respectively.

To host A: Source port = 80, source IP address = b, dest port = 26145, dest IP address = a.

To host C, left process: Source port = 80, source IP address = b, dest port = 7532, dest IP address = c.

To host C, right process: Source port = 80, source IP address = b, dest port = 26145, dest IP address = c.

Assume that a host receives a UDP segment with 01011101 11110010 (we separated the values of each byte with a space for clarity) as the checksum. The host adds the 16-bit words over all necessary fields excluding the checksum and obtains the value 00110010 00001101. Is the segment considered correctly received or not? What does the receiver do?

As demonstrated above, the overall checksum contains 0. Therefore, it is not valid. The receiver will drop this packet and wait for retransmitting.

Host A and B are directly connected with a 100 Mbps link. There is one TCP connection between the two hosts, and Host A is sending to Host B an enormous file over this connection. Host A can send its application data into its TCP socket at a rate as high as 120 Mbps but Host B can read out of its TCP receive buffer at a maximum rate of 50 Mbps. Describe the effect of TCP flow control.

Since the link capacity is only 100 Mbps, so host A's sending rate can be at most 100Mbps. Still, host A sends data into the receive buffer faster than Host B can remove data from the buffer. The receive buffer fills up at a rate of roughly 50Mbps. When the buffer is full, Host B signals to Host A to stop sending data by setting RcvWindow = 0. Host A then stops sending until it receives a TCP segment with RcvWindow > 0. Host A will thus repeatedly stop and start sending as a function of the RcvWindow values it receives from Host B. On average, the long-term rate at which Host A sends data to Host B as part of this connection is no more than 50Mbps.

Consider Figure 3.58. Assuming TCP Reno is the protocol experiencing the behavior shown above, answer the following questions. In all cases, you should provide a short discussion justifying your answer.

- (a) Identify the intervals of time when TCP slow start is operating.
 - TCP slowstart is operating in the intervals [1,6] and [23,26]
- (b) Identify the intervals of time when TCP congestion avoidance is operating.
 - TCP congestion avoidance is operating in the intervals [6,16] and [17,22]
- (c) After the 16th transmission round, is segment loss detected by a triple duplicate ACK or by a timeout?
 - After the 16th transmission round, packet loss is recognized by a triple duplicate ACK. If there was a timeout, the congestion window size would have dropped to 1.
- (d) After the 22nd transmission round, is segment loss detected by a triple duplicate ACK or by a timeout?

After the 22nd transmission round, segment loss is detected due to timeout, and hence the congestion window size is set to 1.