Computer Networks
Fall 2009
Homework 2 Solutions

Question 3-1:
Source port number y and destination port number x.

Question 3-4:
An application developer may not want its application to use TCP's congestion control, which can throttle the application's sending rate at times of congestion. Often, designers of IP telephony and IP videoconference applications choose to run their applications over UDP because they want to avoid TCP's congestion control. Also, some applications do not need the reliable data transfer provided by TCP.

Question 3-5:
For each persistent connection, the Web server creates a separate "connection socket". Each connection socket is identified with a four-tuple: (source IP address, source port number, destination IP address, destination port number). When host C receives and IP datagram, it examines these four fields in the datagram/segment to determine to which socket it should pass the payload of the TCP segment. Thus, the requests from A and B pass through different sockets. The identifier for both of these sockets has 80 for the destination port; however, the identifiers for these sockets have different values for source IP addresses. Unlike UDP, when the transport layer passes a TCP segment's payload to the application process, it does not specify the source IP address, as this implicitly specified by the socket identifier.

Question 3-6:
Since most firewalls are configured to block UDP traffic, using TCP for video and voice traffic lets the traffic though the firewalls.

Question 3-7:
Yes, both segments will be directed to the same socket. For each received segment, at the socket interface, the operating system will provide the process with the IP addresses to determine the origins of the individual segments.

Question 3-9:
A timer would still be necessary in the protocol rdt 3.0. If the round trip time is known then the only advantage will be that, the sender knows for sure that either the packet or the ACK (or NACK) for the packet has been lost, as compared to the real scenario, where the ACK (or NACK) might still be on the way to the sender, after the timer expires. However, to detect the loss, for each packet, a timer of constant duration will still be necessary at the sender.

Question 3-11:
Sequence numbers are required for a receiver to find out whether an arriving packet contains new data or is a retransmission.

Question 3-14:
a) false b) false c) true d) false e) true f) false g) false.

Question 3-15:
3 segments. First segment: seq = 43, ack =80; Second segment: seq = 80, ack = 44; Third segment; seq = 44, ack = 81.

Question 3-16:
a) 20 bytes b) ack number = 90.

Question 3-17:
False, it is set to half of the current value of the congestion window.

Question 3-18:
R/2.