3. Source port number y and destination port number x.

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.

5. Since most firewalls are configured to block UDP traffic, using TCP for video and voice traffic lets the traffic though the firewalls.

11. 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.

Problem 40

a) TCP slowstart is operating in the intervals [1,6] and [23,26]
b) TCP congestion avoidance is operating in the intervals [6,16] and [17,22]
c) After the 16^{th} 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 22^{nd} transmission round, segment loss is detected due to timeout, and hence the congestion window size is set to 1.

e) The threshold is initially 32, since it is at this window size that slow start stops and congestion avoidance begins.
f) The threshold is set to half the value of the congestion window when packet loss is detected. When loss is detected during transmission round 16, the congestion windows size is 42. Hence the threshold is 21 during the 18^{th} transmission round.
g) The threshold is set to half the value of the congestion window when packet loss is detected. When loss is detected during transmission round 22, the congestion windows size is 29. Hence the threshold is 14 (taking lower floor of 14.5) during the 24^{th} transmission round.
h) During the 1^{st} transmission round, packet 1 is sent; packet 2-3 are sent in the 2^{nd} transmission round; packets 4-7 are sent in the 3^{rd} transmission round; packets 8-15 are sent in the 4^{th} transmission round; packets 16-31 are sent in the 5^{th} transmission round; packets 32-63 are sent in the 6^{th} transmission round; packets 64 – 96 are sent in the 7^{th} transmission round. Thus packet 70 is sent in the 7^{th} transmission round.
i) The threshold will be set to half the current value of the congestion window (8) when the loss occurred and congestion window will be set to the new threshold value + 3 MSS . Thus the new values of the threshold and window will be 4 and 7 respectively.
j) threshold is 21, and congestion window size is 1.
k) round 17, 1 packet; round 18, 2 packets; round 19, 4 packets; round 20, 8 packets; round 21, 16 packets; round 22, 21 packets. So, the total number is 52.

Part 2

Q2:

As the starting byte number=500

From byte 500, we need 46 more bytes. So, the next free byte is 546. Hence, the acknowledgement that will be sent back is 546. It indicates that all bytes till 546 have been received and the next expected number is 546.

Acknowledgement number=546

Q3:

Adding the two bytes gives 11111010+01100101=01011111; 00000001+01011111(wrap around) = 01100000; The one's complement gives 10011111 Hence the check sum is 10011111. At the receiver, all 3 bytes including the 2 bytes and checksum are added. If no errors are introduced into the packet, then clearly the sum at he receiver will be 11111111. If one of the bits is a 0, then we know that errors have been introduced into the packet.

Q4:

a) 20 bytes

b) 90