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.

14. a) false b) false c) true d) false e) true f) false g) false

15. a) 20 bytes b) ack number = 90

17. R/2

18. False, it is set to half of the current value of the congestion window.

Problem 1

	source port numbers	destination port numbers
a) $A \rightarrow S$	467	23
b) $B \rightarrow S$	513	23
c) $S \rightarrow A$	23	467
d) $S \rightarrow B$	23	513

e) Yes.
f) No.

Problem 3

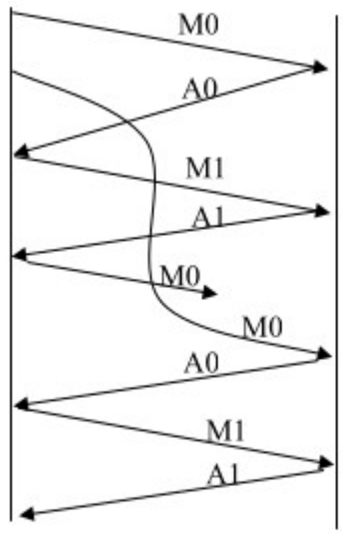
Note, wrap around if overflow.

$$\begin{array}{r} 0\ 1\ 0\ 1\ 0\ 0\ 1\ 1 \\ +\ 0\ 1\ 1\ 0\ 0\ 1\ 1\ 0 \\ \hline 1\ 0\ 1\ 1\ 1\ 0\ 0\ 1 \end{array}$$

$$\begin{array}{r} 1\ 0\ 1\ 1\ 1\ 0\ 0\ 1 \\ +\ 0\ 1\ 1\ 1\ 0\ 1\ 0\ 0 \\ \hline 0\ 0\ 1\ 0\ 1\ 1\ 1\ 0 \end{array}$$

One's complement = 1 1 0 1 0 0 0 1.
To detect errors, the receiver adds the four words (the three original words and the checksum). If the sum contains a zero, the receiver knows there has been an error. All one-bit errors will be detected, but two-bit errors can be undetected (e.g., if the last digit of the first word is converted to a 0 and the last digit of the second word is converted to a 1).

Problem 13



Problem 15

It takes 12 microseconds (or 0.012 milliseconds) to send a packet, as $1500*8/10^9=12$ microseconds. In order for the sender to be busy 98 percent of the time, we must have

$util = 0.98 = (0.012n)/30.012$

or n approximately 2451 packets.

Problem 22

a) Here we have a window size of N=3. 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 3 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

Problem 23

In order to avoid the scenario of Figure 3.27, we want to avoid having the leading edge of the receiver's window (i.e., the one with the “highest” sequence number) wrap around in the sequence number space and overlap with the trailing edge (the one with the "lowest" sequence number in the sender's window). That is, the sequence number space must be

large enough to fit the entire receiver window and the entire sender window without this overlap condition. So - we need to determine how large a range of sequence numbers can be covered at any given time by the receiver and sender windows.
Suppose that the lowest-sequence number that the receiver is waiting for is packet m. In this case, it's window is [m,m+w-1] and it has received (and ACKed) packet m-1 and the w-1 packets before that, where w is the size of the window. If none of those w ACKs have been yet received by the sender, then ACK messages with values of [m-w,m-1] may still be propagating back. If no ACKs with these ACK numbers have been received by the sender, then the sender's window would be [m-w,m-1].
Thus, the lower edge of the sender's window is m-w, and the leading edge of the receivers window is m+w-1. In order for the leading edge of the receiver's window to not overlap with the trailing edge of the sender's window, the sequence number space must thus be big enough to accommodate 2w sequence numbers. That is, the sequence number space must be at least twice as large as the window size, $k \geq 2w$.

Problem 32

a)

Denote $EstimatedRTT^{(n)}$ for the estimate after the nth sample.

$$\begin{aligned}EstimatedRTT^{(4)} = {} & xSampleRTT_1 + \\ & (1-x)[xSampleRTT_2 + \\ & (1-x)[xSampleRTT_3 + (1-x)SampleRTT_4]] \\ \\ = {} & xSampleRTT_1 + (1-x)xSampleRTT_2 \\ \\ & + (1-x)^2 xSampleRTT_3 + (1-x)^3 SampleRTT_4\end{aligned}$$

b)

$$\begin{aligned}EstimatedRTT^{(n)} = {} & x\sum_{j=1}^{n-1}(1-x)^{j-1} SampleRTT_j \\ \\ & + (1-x)^{n-1} SampleRTT_n\end{aligned}$$

c)

$$\begin{aligned}EstimatedRTT^{(\infty)} &= \frac{x}{1-x}\sum_{j=1}^{\infty}(1-x)^j SampleRTT_j \\ &= \frac{1}{9}\sum_{j=1}^{\infty} .9^j SampleRTT_j\end{aligned}$$

The weight given to past samples decays exponentially.

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 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, 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 18th 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~~ 26. Hence the threshold is ~~14~~ 13 ~~(taking lower floor of 14.5)~~ during the 24th transmission round.
h) During the 1st transmission round, packet 1 is sent; packet 2-3 are sent in the 2nd transmission round; packets 4-7 are sent in the 3rd transmission round; packets 8-15 are sent in the 4th transmission round; packets 16-31 are sent in the 5th transmission round; packets 32-63 are sent in the 6th transmission round; packets 64 – 96 are sent in the 7th transmission round. Thus packet 70 is sent in the 7th 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 4.
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.

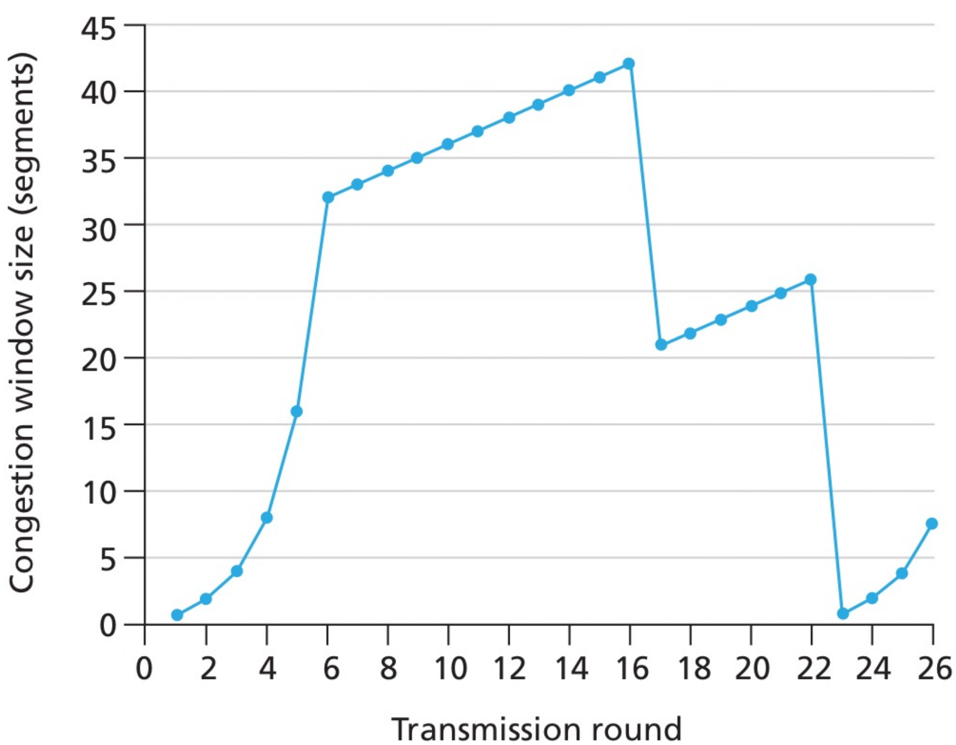


Figure 3.58 ♦ TCP window size as a function of time