P3. UDP and TCP use 1s complement for their checksums. Suppose you have the following three 8-bit bytes: 01010011, 01100110, 01110100. What is the 1s complement of the sum of these 8-bit bytes? (Note that although UDP and TCP use 16-bit words in computing the checksum, for this problem you are being asked to consider 8-bit sums.) Show all work. Why is it that UDP takes the 1s complement of the sum; that is, why not just use the sum? With the 1s complement scheme, how does the receiver detect errors? Is it possible that a 1-bit error will go undetected? How about a 2-bit error?

(1) 0101 0011 + 0110 0110 = 1011 1001
    1011 1001 + 0111 0100 = 1 0010 1101 = 0010 1101 + 1 = 00101110
    So, the 1's complement is 11010001.
(2) It's easier to detect errors with the 1's complement of the sum instead of just the sum. And we don't need to care for whether the mode of the receiver is big-endian or little-endian.
(3) To detect errors, the receiver just need to add the sum it calculated to the 1's complement it received. If the new sum contains 0, an error occurs.
(4) 1-bit error will not go undetected, but 2-bit error may. For example, the first two bytes turn to 01110011 and 01000110, the sum will be the same.

P5. Suppose that the UDP receiver computes the Internet checksum for the received UDP segment and finds that it matches the value carried in the checksum field. Can the receiver be absolutely certain that no bit errors have occurred? Explain.

No. Double-bit error may go undetected.
For example, two 8-bit bytes, 0000 1010 and 0000 0101. If they turn to 0000 1011 and 0000 0100, their sum doesn't change. Although errors occurred, they can't be detected. So we can't absolutely certain that no bit errors have occurred.

P15. Consider the cross-country example shown in Figure 3.17. How big would the window size have to be for the channel utilization to be greater than 98 percent? Suppose that the size of a packet is 1,500 bytes, including both header fields and data.

U = (nL/R) / (RTT + L/R) > 98%
n>2450.92

So, n should be 2451 at least.

P45. Recall the macroscopic description of TCP throughput. In the period of time from when the connection's rate varies from $W/(2 \cdot \mathrm{RTT})$ to W/RTT, only one packet is lost (at the very end of the period).

a. Show that the loss rate (fraction of packets lost) is equal to

$$L = \text{loss rate} = \frac{1}{\frac{3}{8}W^2 + \frac{3}{4}W}$$

b. Use the result above to show that if a connection has loss rate L, then its average rate is approximately given by

$$\approx \frac{1.22 \cdot MSS}{RTT\sqrt{L}}$$

a. packet number form W/(2*RTT) to W/RTT is:

W/2+(W/2+1)+……+W = $\frac{3}{8}W^2 + \frac{3}{4}W$

Only one packet is lost, so the loss rate is:

$$\frac{1}{\frac{3}{8}W^2 + \frac{3}{4}W}$$

b. When $\mathrm{W} \to \infty$, $W^2 \gg W$, $\mathrm{L} \approx \frac{8}{3W^2}$, $W \approx \sqrt{\frac{3}{8}L}$

The average throughput is:

$$\frac{0.75W}{RTT} = \frac{1.22MSS}{RTT\sqrt{L}}$$

P49. Let T (measured by RTT) denote the time interval that a TCP connection takes to increase its congestion window size from $W/2$ to W, where W is the maximum congestion window size. Argue that T is a function of TCP's average throughput.

TCP's average throughput:

$$\mathrm{D} = \frac{1.22MSS}{RTT\sqrt{L}}$$

$$=> \quad \mathrm{L} = (\frac{1.22MSS}{RTT\sqrt{L}})^2$$

$$\frac{MMS}{L} = T \times D$$

$$=> \quad \mathrm{T} = \frac{D \times RTT^2}{1.22^2 MMS}$$