Network Systems (201600146/201600197), Test 1

February 17, 2017, 13:45–15:15

Answers

1. Protocols and performance

2 pt (a) Propagation time = $\frac{\text{distance}}{\text{speed}} = \frac{1200}{200\,000}$ = 0.006 s = 6 ms

2 pt (b) Transmission time = $\frac{\text{number of bits}}{\text{bit rate}} = \frac{1000}{1000\,000}$ = 0.001 s = 1 ms

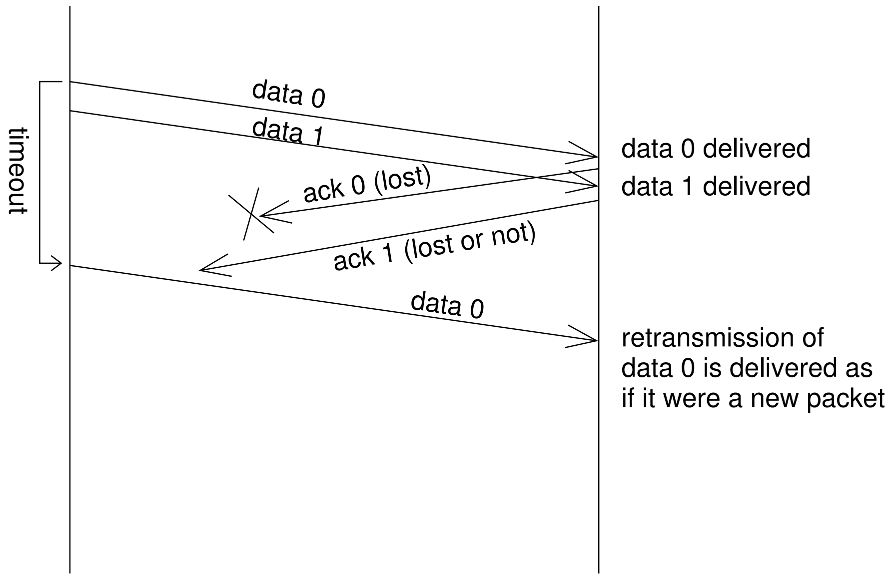
2 pt (c) 1 ms after the start of a data packet, its last bit has left the sender. 6 ms later it arrives at the receiver, which sends an ack, which takes 6 ms to reach the sender. So only 13 ms after the start of a packet can we move the window to no longer include that packet. During that time, 13 packets can be sent, so in order to transmit without interruptions, the minimum sufficient SWS is 13.

Although strictly not correct, we've also accepted 12.
Also acceptable is 14, if you assume the acknowledgment packet is equally long as a data packet.
Many students forgot to double the transmission delay to find the RTT, thus ending up with a window of 6 or 7; this was worth at most 1 point.

2 pt (d) 4 bits are needed for distinguishing SWS+RWS=13+1 sequence numbers, since $2^3 < 14 \leq 2^4$.

1 pt (e) Answer D is correct.

3 pt (f)



Many variations are possible. The essential point is that the receiver must genuinely think the retransmission of packet 0 is actually a new one with the same sequence number. So it must have acked the previous one with that number (although that ack was lost), and it must have received a packet with number 1 (otherwise, it would not be willing to accept a packet with number 0 again).

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2. Information theory and error-correcting codes

3 pt (a) Use the formula: $H = 0.3 \log_2 \frac{1}{0.3} + 0.3 \log_2 \frac{1}{0.3} + 0.19 \log_2 \frac{1}{0.19} + 0.19 \log_2 \frac{1}{0.19} + 0.02 \log_2 \frac{1}{0.02} = 2.07$ bits per word.

1 pt (b) In both cases we would already know more about his words than when they are random, so the amount of information in them is less. So answer B is correct.

2 pt (c) Code A is not uniquely decodable; codes B and D are too long on average; codes C and E are good.

1 pt (d) Answer D is correct. (Basically, this is a matter of "garbage in, garbage out". The error-correcting code does not know anything about the *meaning* of the bits; it can only correct errors that happened in transport.)

2 pt (e) Use the formula: $C = 100 \cdot (1 - 0.95 \log_2 \frac{1}{0.95} - 0.05 \log_2 \frac{1}{0.05}) = 71.36$ bits per second.

1 pt (f) Since 70 bits/s is less than the channel's capacity, error correcting codes can make the error rate arbitrarily low; so indeed, codes exist which make the error rate less than 0.01%: answer C.

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3. Applications

3 pt (a) 1 RTT to set up the connection.
1 RTT + 8 ms transmission time to fetch the HTML file.
1 RTT + 80 ms + 80000 ms transmission time for the image and movie.
Total 3×2 + 80088 = 80094 ms = 80.094 s.

1 pt (b) In that case, separate connections would need to be used for each file, so each file would need 1 RTT for set up, and 1 RTT + transmission time for the download. Total is 6 RTT + transmission time, that's 3 RTT more than in (a).

1 pt (c) Answer C is correct.

3 pt (d) We can use the formula from the handout, with $d = u_s = u_i = 10^6$ bit/s, $F = 8 \cdot 10^7$ bits, and $N = 100$: minimal download time = $\max\left(\frac{F}{u}, \frac{F}{d}, \frac{NF}{(N+1)u}\right) = \max(80, 80, 79.2) = 80$ seconds.

1 pt (e) Looking at the previous calculation, we see that the first and the second term in the $\max(\dots)$ calculation are the largest, so they are the limiting factors: the server's upload speed (first term) and the peers' download speed (second term). Answer B and C.

1 pt (f) Answer C is correct.