浙江大学《计算机网络》课程课后作业六

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1. In both parts of Fig. 6-6, there is a comment that the value of SERVERPORT must be the same in both client and server. Why is this so important?

Answer: 客户端中绑定的端口号将被送往服务端对应的端口，如果客户端和服务端的端口不一致，那么服务端在监听端口时将不会把数据包转发到对应的端口中。

2. Imagine that a two-way handshake rather than a three-way handshake were used to set up connections. In other words, the third message was not required. Are deadlocks now possible? Give an example or show that none exist.

Answer: 死锁可能存在，假设 A 发送了一个数据包给 B(Seq=x)，当 B 收到数据包时，B 给 A 回复了一个响应(Seq=y, Ack=x)，但是响应中途丢失了。则此时 A 以为已经建立了连接，于是等待 B 发送下一个数据包，而 B 却还在等待 A 发回的响应，两者陷入死锁。

3. Why does UDP exist? Would it not have been enough to just let user processes send raw IP packets?

Answer:

(1)UDP 协议能够支持无连接的传输协议，不会引入建立连接的延迟，而且 UDP 头格式较小，适用于一次性传输数据较小的网络应用以及多媒体应用。

(2)不能，如果 UDP 协议没有指明端口，那么服务端接收后将不知道要把数据交给哪个端口对应的进程。

4. A client sends a 128-byte request to a server located 100 km away over a 1-gigabit optical fiber. What is the efficiency of the line during the remote procedure call?

Answer:

准备 128B 的数据需要花费 128*8b/1Gbps=0.000001s （忽略不计）

而路上的开销为 0.0005s * 2 = 0.001s

因此总开销 T0 = 0.001s

传输效率为 0.000001/0.001 = 0.1%

5. Datagram fragmentation and reassembly are handled by IP and are invisible to TCP. Does this mean that TCP does not have to worry about data arriving in the wrong order?

Answer:

并不是。即使数据报都到达了，但是其到达的顺序仍然可能产生错误，因此 TCP 仍然需要重组数据报使其有序。

6. The maximum payload of a TCP segment is 65,495 bytes. Why was such a strange number chosen?

Answer:

因为 TCP 段的起始部分占用 20 个字节，且 IP 头占用 20 个字节，那么如果 Options 可选不占用空间，那么最大数据载荷应该是 65535-20-20=65495 字节

7. If the TCP round-trip time, RTT, is currently 30 msec and the following acknowledgements come in after 26, 32, and 24 msec, respectively, what is the new RTT estimate using the Jacobson algorithm? Use $\alpha=0.9$.

Answer:

[1] $0.9 * 30 + 0.1 * 26 = 29.6$

[2] $0.9 * 29.6 + 0.1 * 32 = 29.84$

[3] $0.9 * 29.84 + 0.1 * 24 = 29.256$

8. To get around the problem of sequence numbers wrapping around while old packets still exist, one could use 64-bit sequence numbers. However, theoretically, an optical fiber can run at 75 Tbps. What maximum packet lifetime is required to make sure that future 75-Tbps networks do not have wrap around problems even with 64-bit sequence numbers? Assume that each byte has its own sequence number, as TCP does.

Answer:

序号绕回的时间 T = (2^64)Bytes/(75/8*10^12)Bps=1967652s=22.77Days

因此最大生存周期应该是 22.77 天

9. Consider that only a single TCP (Reno) connection uses one 10Mbps link which does not buffer any data. Suppose that this link is the only congested link between the sending and receiving hosts. Assume that the TCP sender has a huge file to send to the receiver, and the receiver's receive buffer is much larger than the congestion window. We also make the following assumptions: each TCP segment size is 1,500 bytes; the twoway propagation delay of this connection is 150 msec; and this TCP connection is always in congestion avoidance phase, that is, ignore slow start.

– a. What is the maximum window size (in segments) that this TCP connection can achieve?

– b. What is the average window size (in segments) and average throughput (in bps) of this TCP connection?

– c. How long would it take for this TCP connection to reach its maximum window again after recovering from a packet loss?

Answer:

(1) MSS * W / RTT = 10Mpbs => W = (10 * 1000000) * 0.15 / 1500 / 8 = 125

(2) Average WindowSize = (W + W/2) /2 = 0.75W

Average Throughput = 0.75W * 1500 * 8 / 0.15 = 0.75 * 125 * 1500 * 8 / 0.15 = 7.5Mbps

(3)从 W/2 阈值恢复到 W 采用加法增大，因此 T=(125/2)*0.15s=9.375s