CS 655 Computer Network
Lab 8 TCP Performance

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Throughput

Theoretically, the throughput of a TCP connection can be approximated by:

$$Throughput = \frac{1.22\ MSS}{RTT\ \sqrt{plr}}$$

where the Maximum Segment Size (MSS) = 1460 Bytes (1460x8=11680 bits) on a link with MTU of 1500 Bytes, and the RTT value can be calculated using the Ping application.

Experimentally, use iperf to measure the bandwidth as the throughput.

Result

Unit: Mbps
The upper one is the theoretical throughput and the lower one is the measured throughput.

	Loss = 0.01%	Loss = 0.1%	Loss = 1%	Loss = 3%	Loss = 7%	Loss = 10%
Delay: 5ms	128 275 (exp)	40 144 (exp)	12 18 (exp)	7 7.94 (exp)	4.8 3.41 (exp)	4 1.89 (exp)
Delay: 10ms	67 243 (exp)	21 46 (exp)	6.7 9.38 (exp)	3.9 4.58 (exp)	2.5 2.11 (exp)	2.1 1.12 (exp)

Comment

How close your measured throughput values are to their analytical counterparts?

1. When delay is 5ms and packet loss rate is 3%, the measured throughput is close to the analytical counterparts. When the plr gets larger or smaller, the gap between measured value and the analytical counterpart will become larger.
2. When delay is 10ms and packet loss rate is between 3% and 7%, the measure throughput is close to the analytical counterparts. When the plr gets larger or smaller, the gap between measured value and the analytical counterpart will become larger.

Try at least two higher packet loss values (e.g., 2% and 5%) and comment on the validity of the above analytical model.

The gap between the analytical model and experimental value is kind of large. Only when the loss rate and delay lie on some values, the gap will get small. The reason may be that the impact caused by packet loss can not be easily measured by a square root and a simple parameter 1.22. It should be a more complicated analytical model. TCP Reno has a mechanism to handle packet loss such as timeout retransmission and fast retransmission. When packet loss gets higher, the measured throughput will intuitively get smaller.