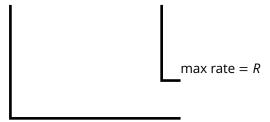
Antonio Carzaniga

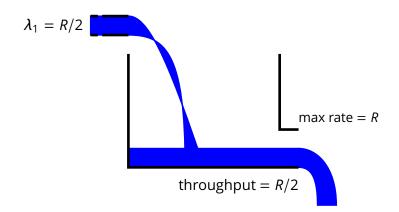
Faculty of Informatics Università della Svizzera italiana

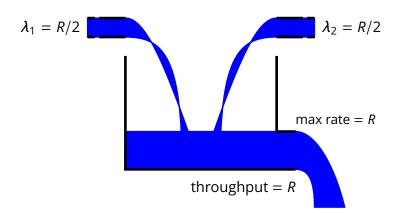
November 28, 2016

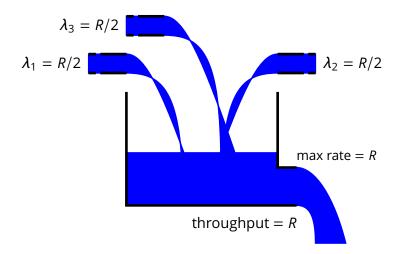
Outline

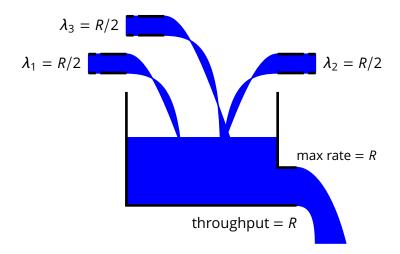
- Intro to congestion control
- Input rate vs. output throughput
- Congestion window
- "Congestion avoidance"
- "Slow start"
- "Fast recovery"

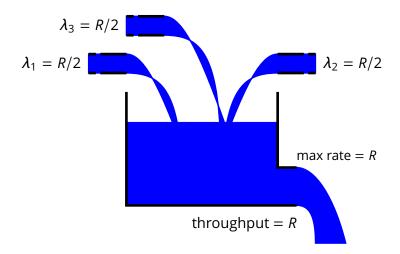


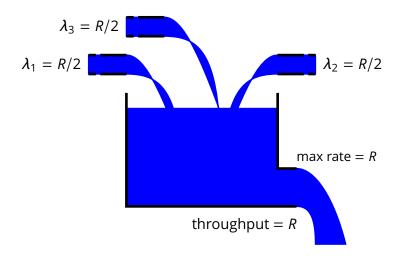


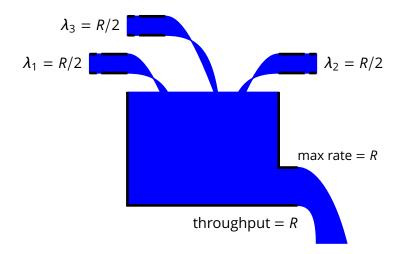














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Extreme case: constant input data rate

$$\lambda_{in} > R$$

In this case $|q| = (\lambda_{in} - R)t$ and therefore

$$d_q = \frac{\lambda_{in} - R}{R}t$$

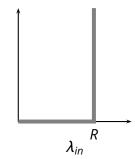


■ Steady-state queuing delay

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ideal input flow λ_{in} constant

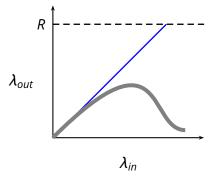
Steady-state queuing delay



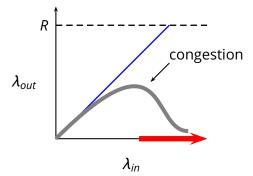
■ Conclusion: as the input rate λ_{in} approaches the maximum throughput R, packets will experience very long delays

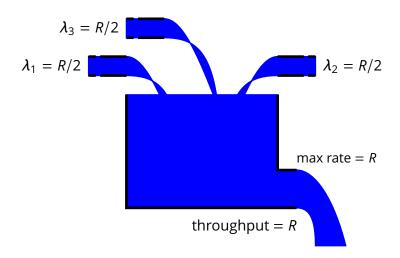
- Conclusion: as the input rate λ_{in} approaches the maximum throughput R, packets will experience very long delays
- More realistic assumptions and models
 - finite queue length (buffers) in routers
 - effects of retransmission overhead
 - full queues along multi-hops paths

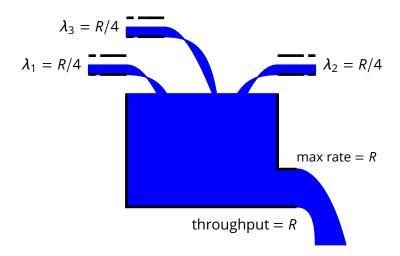
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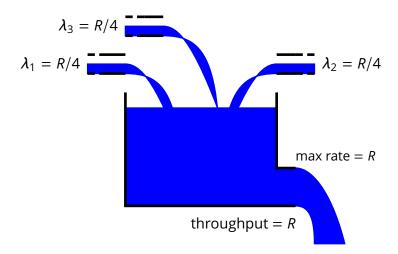


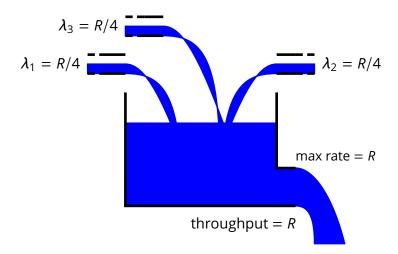
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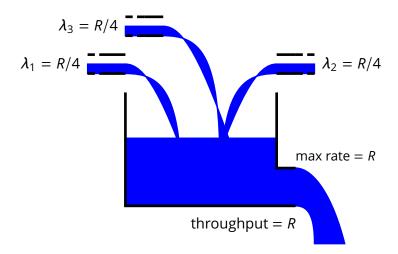














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 - i.e., how does the sender detect congestion?
- how does the sender effectively limit its output rate?
- how should the sender "modulate" its output rate?
 - i.e., what algorithm should the sender use to decrease or increase its output rate?



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- If all traffic is correctly acknowledged, then the sender assumes (quite correctly) that there is no congestion
- Congestion means that the queue of one or more routers between the sender and the receiver overflow
 - the visible effect is that some segments are dropped
- Therefore the server assumes that the network is congested when it detects a segment loss
 - time out (i.e., no ACK)
 - multiple acknowledgements (i.e., NACK)

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■ The resulting maximum output rate is roughly

$$\lambda = \frac{W}{2L}$$

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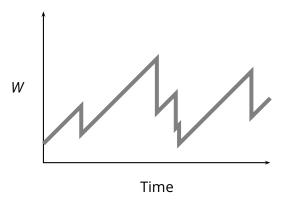
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 - e.g., suppose W = 14600 and MSS = 1460, then the sender increases W to 16060 after 10 acknowledgments acknowledgments

■ Window size W over time



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- To get quickly to a good throughput level, TCP increases its sending rate exponentially for its first growth phase, up to a *slow-start threshold* (*ssthresh*)
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- This process is called "slow start" because of the small initial value of *W*

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- So, TCP reacts differently to a timeout and to a triple duplicate ACKs

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 - run slow start up to W = ssthresh
 - then proceed with congestion avoidance
- *NACK* (i.e., triple duplicate-ack)
 - ▶ set $ssthresh = \overline{W}/2$
 - cut W in half: $W = \overline{W}/2$
 - ► run congestion avoidance, ramping up W linearly
 - ► This is called *fast recovery*

