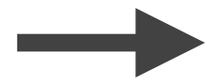
6.033 Spring 2018

Lecture #11

- **Reliable Transport**
- **Window-based Congestion Control**

Internet of Problems

How do we **route** (and address) scalably, while dealing with issues of policy and economy?



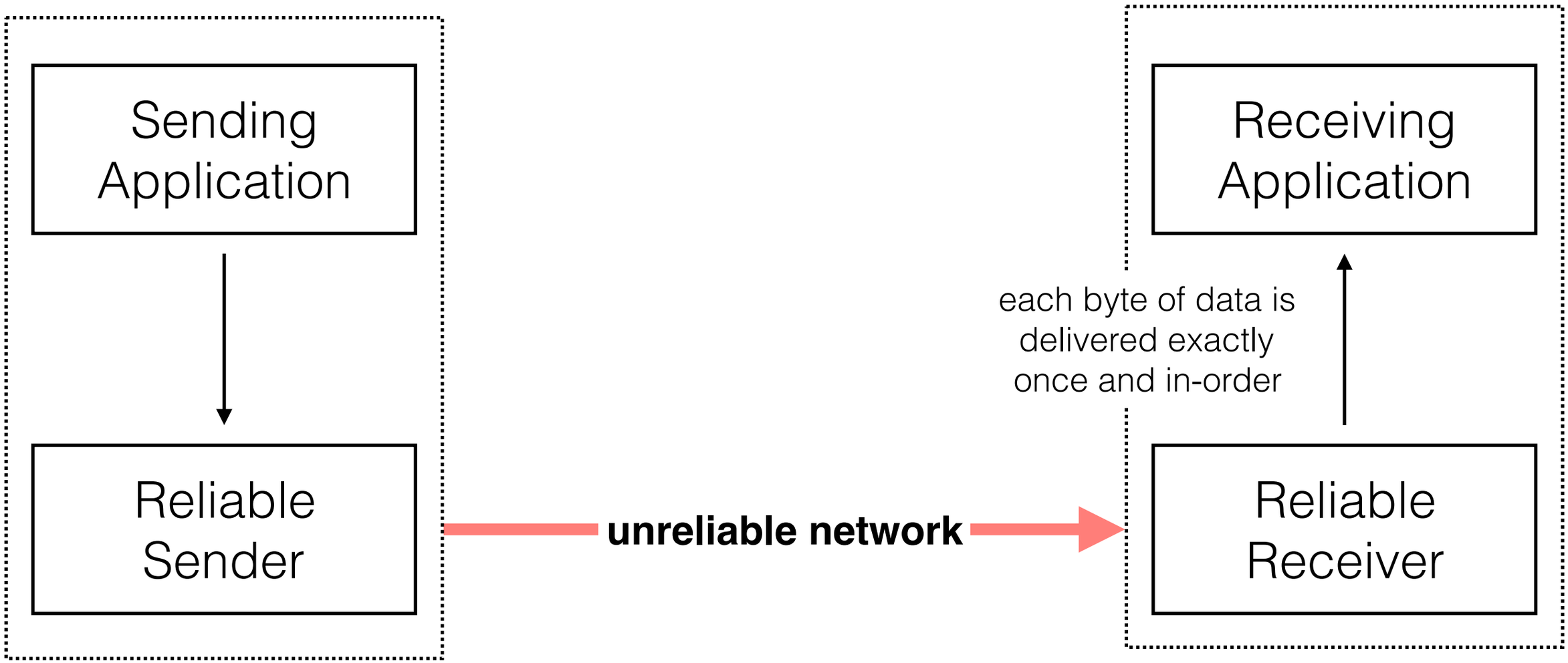
BGP

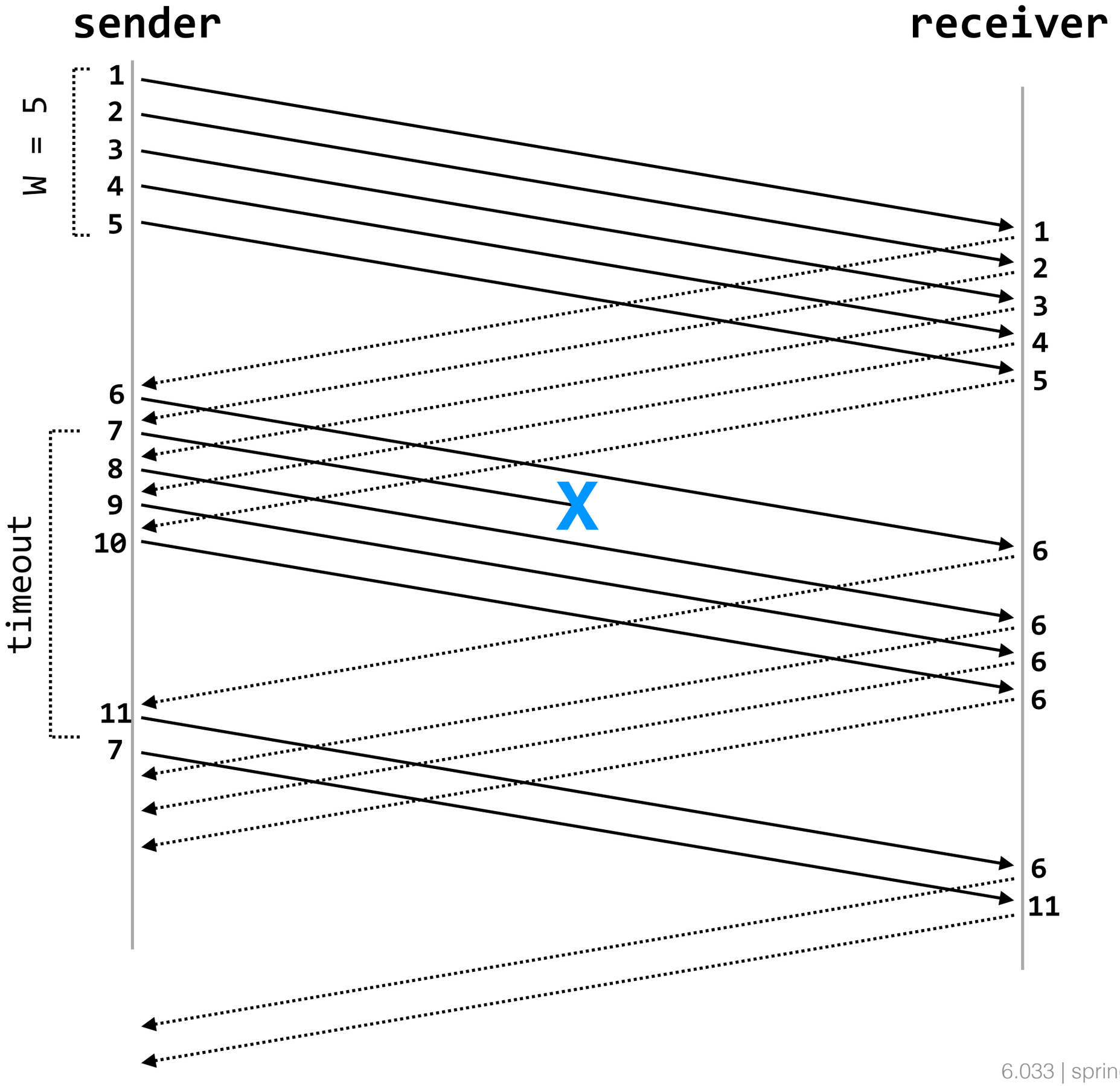
How do we **transport** data scalably, while dealing with varying application demands?

TCP

How do we **adapt** new applications and technologies to an inflexible architecture?

Reliable Transport




sender
receiver
W = 5
timeout
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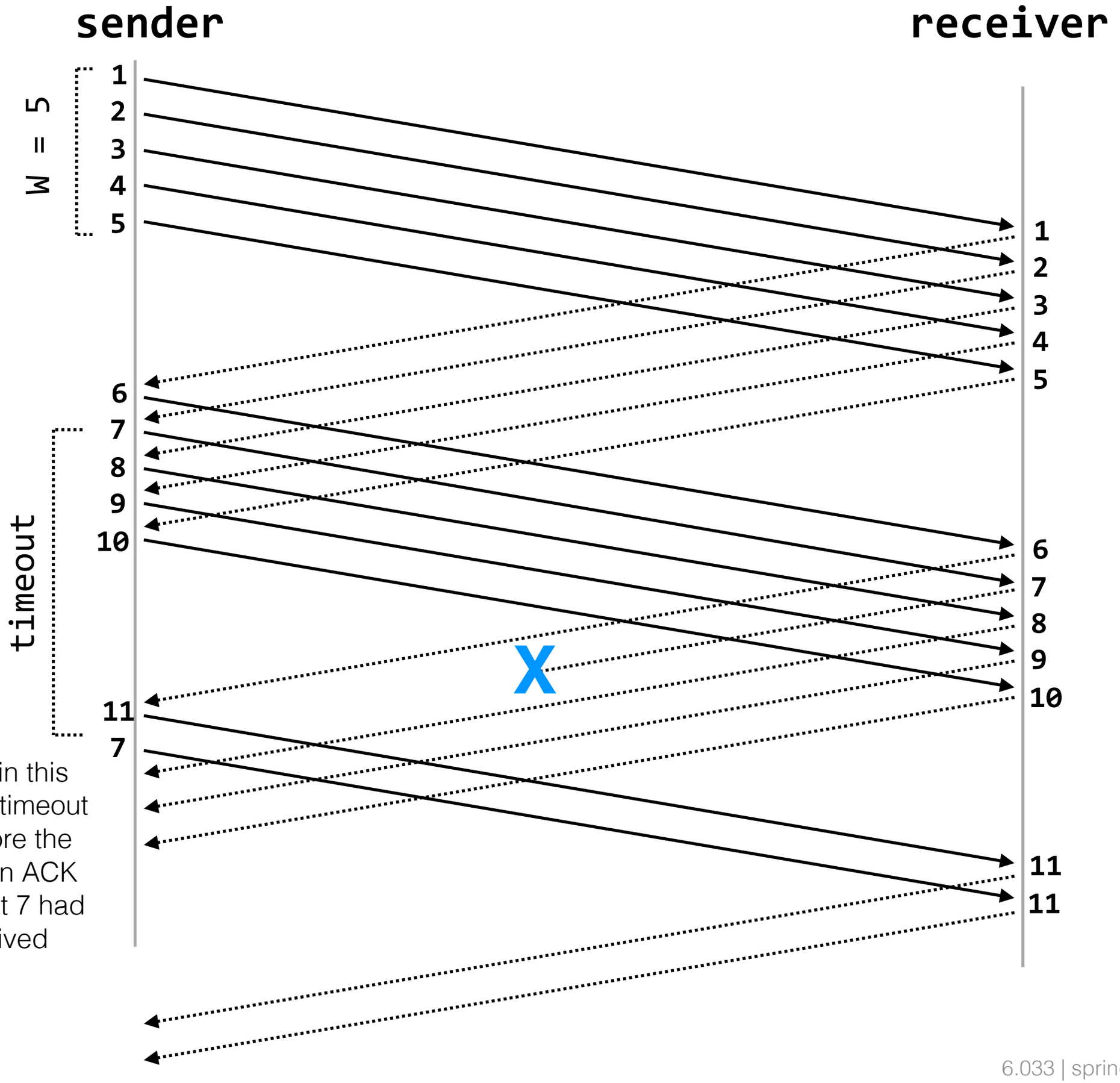
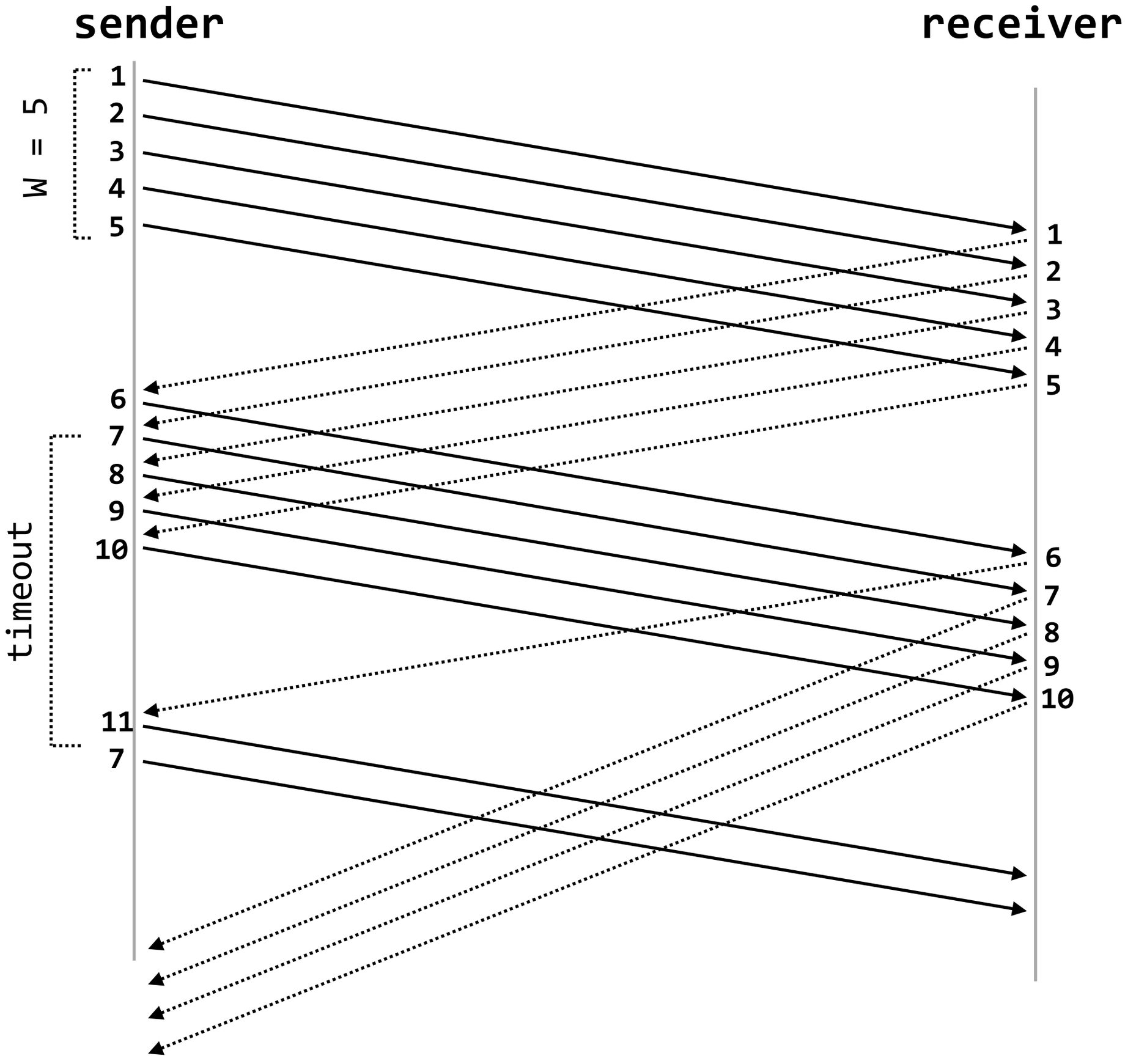


notice that (in this example) the timeout expired before the sender got an ACK indicating that 7 had been received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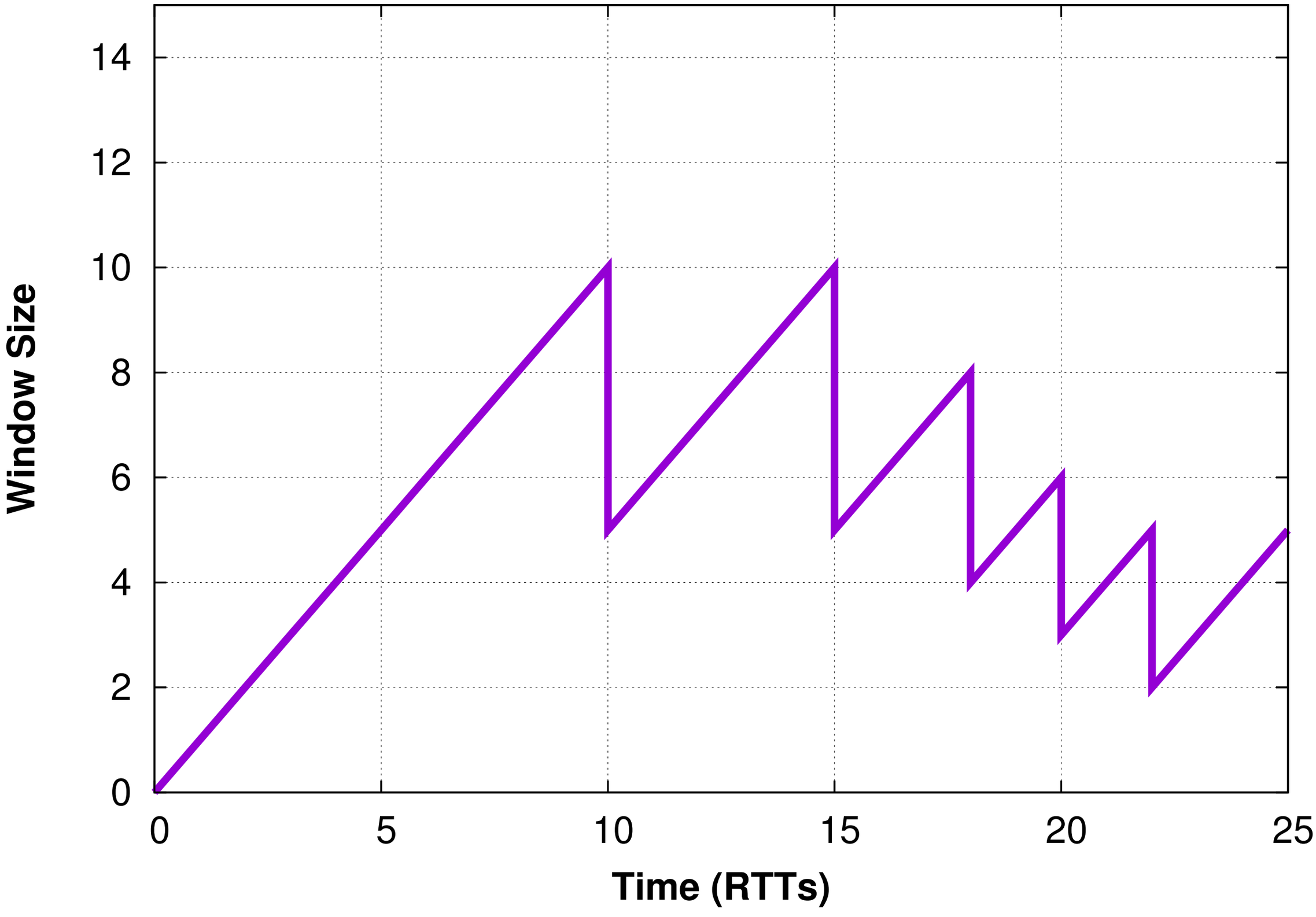
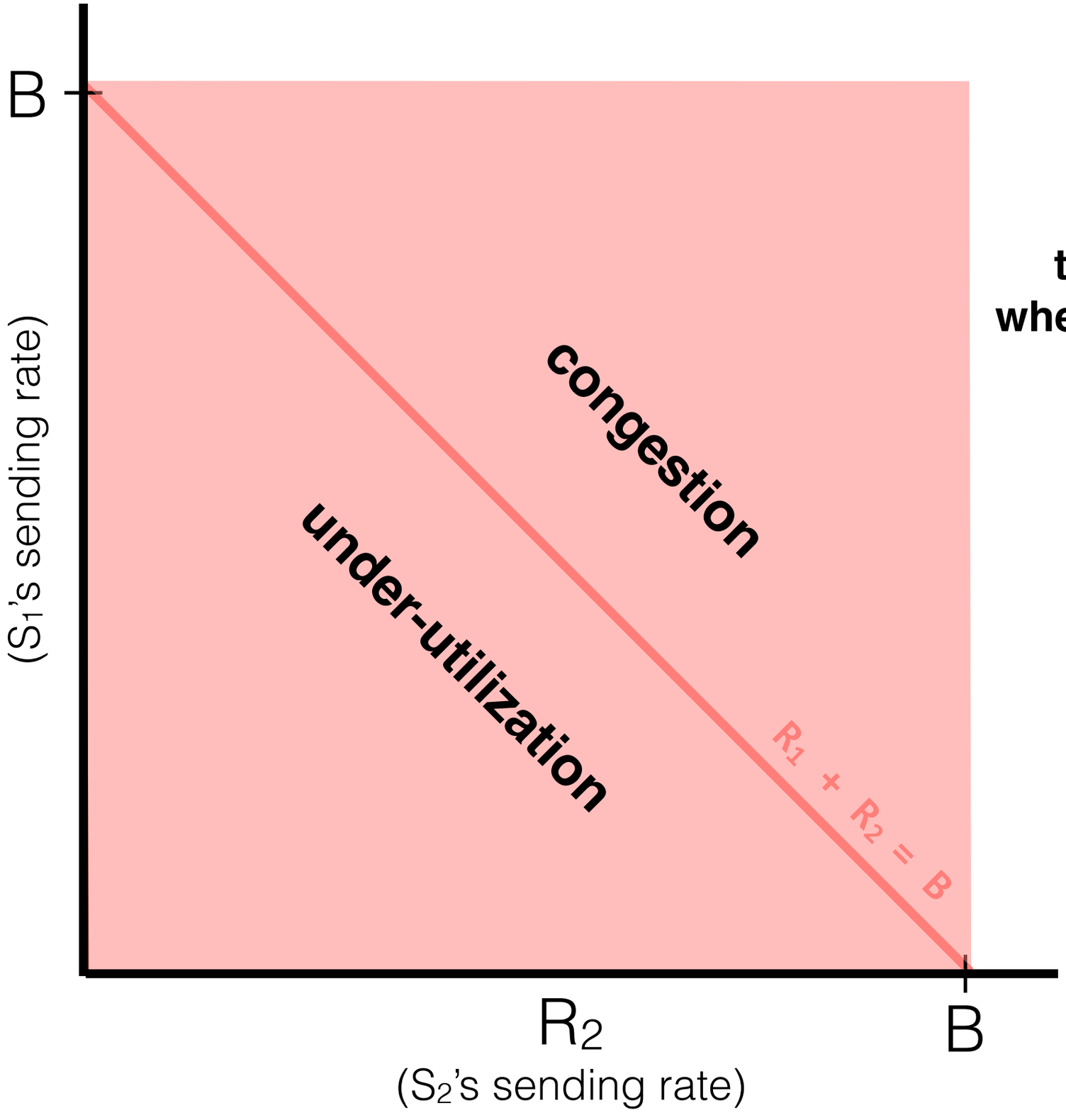
question: what is the correct value for `W`?

too small → underutilized network
too large → congestion

question: how can a single reliable sender, using a sliding-window protocol, set its window size to maximize utilization — but prevent congestion and unfairness — given that there are many other end points using the network, all with different, changing demands?

AIMD

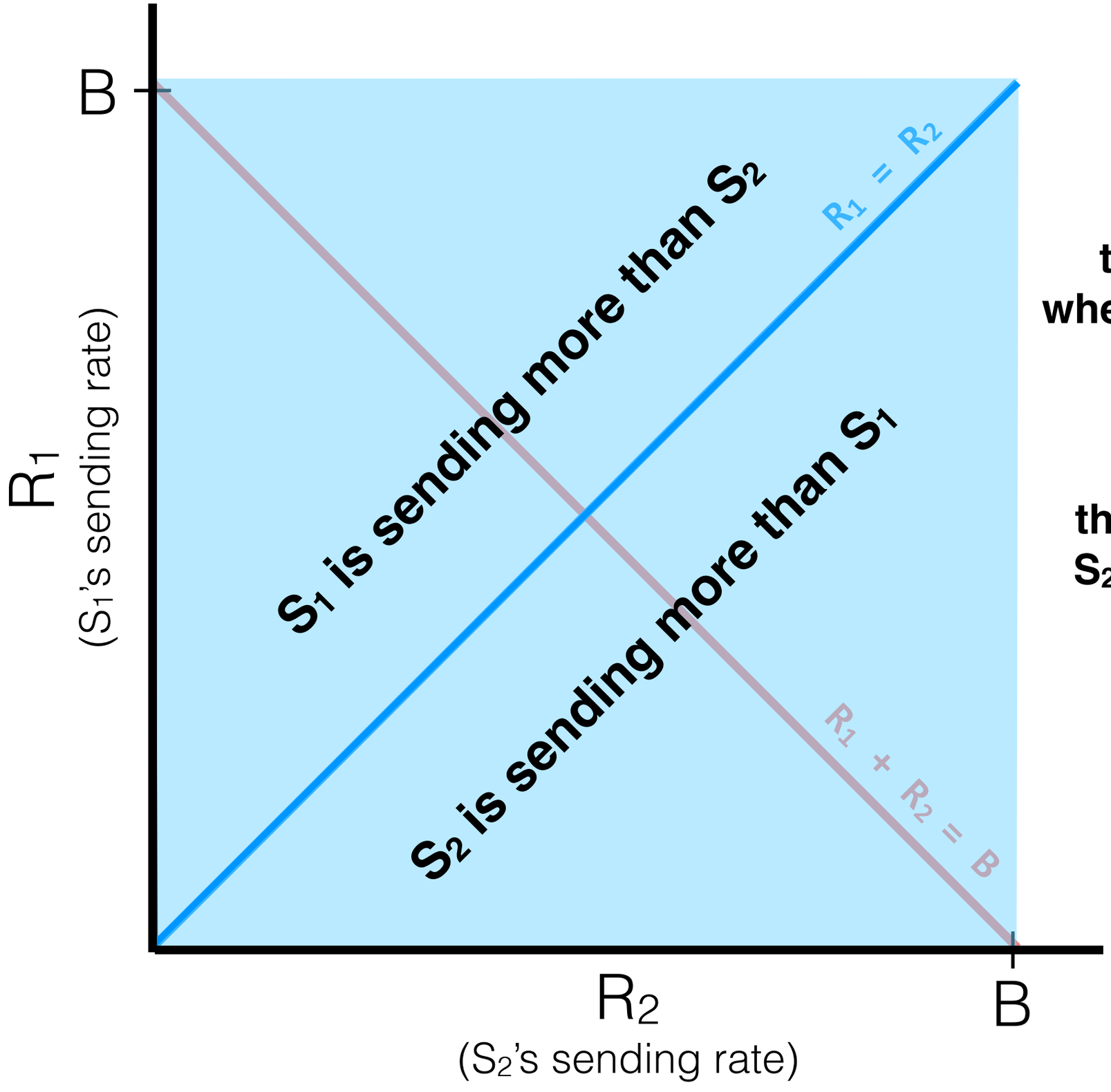




efficiency

(utilization)

the network is fully utilized when the bottleneck link is "full"



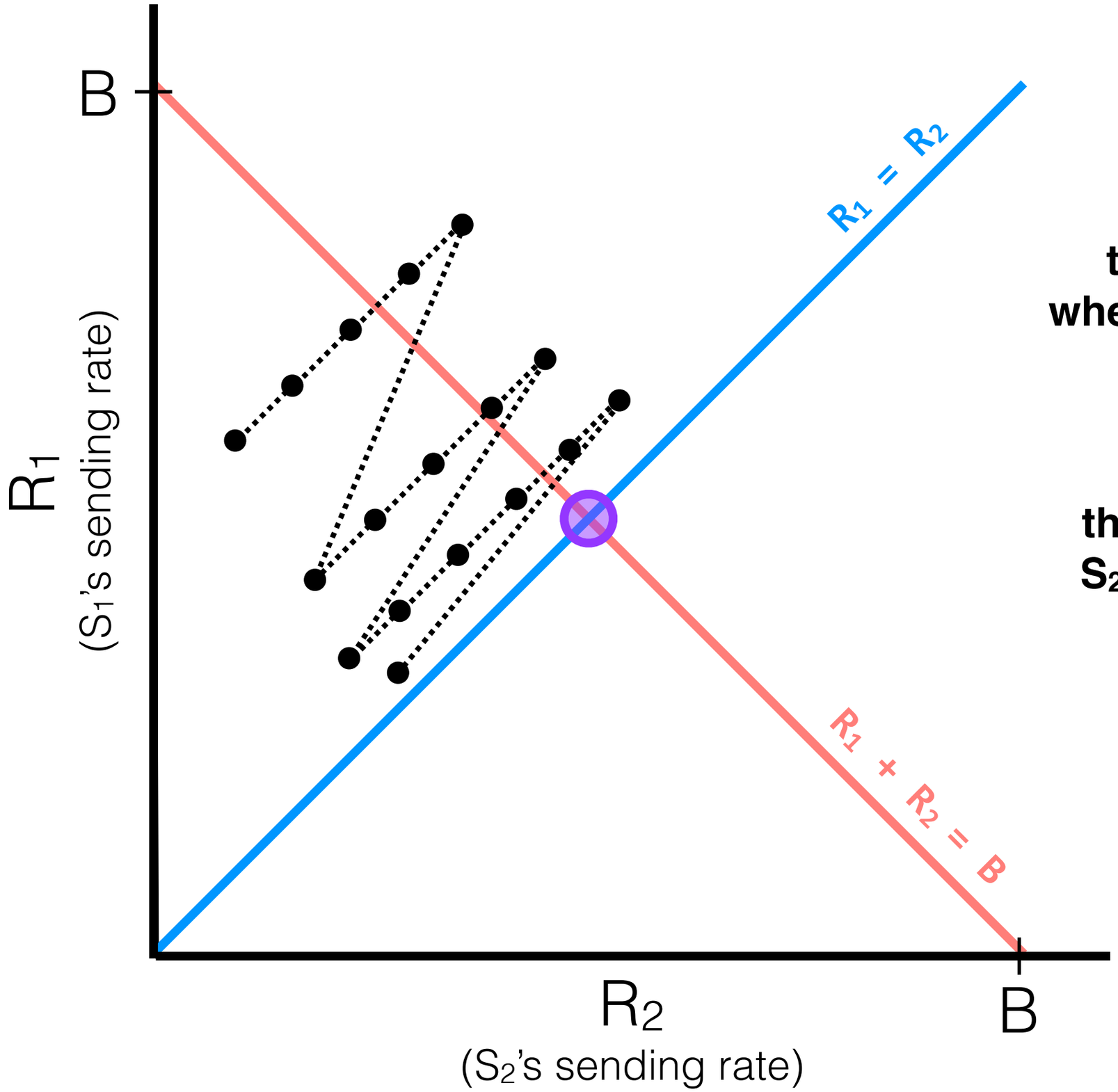
efficiency

(utilization)

the network is fully utilized when the bottleneck link is "full"

fairness

the network is fair when S_1 and S_2 are sending at the same rate



efficiency

(utilization)

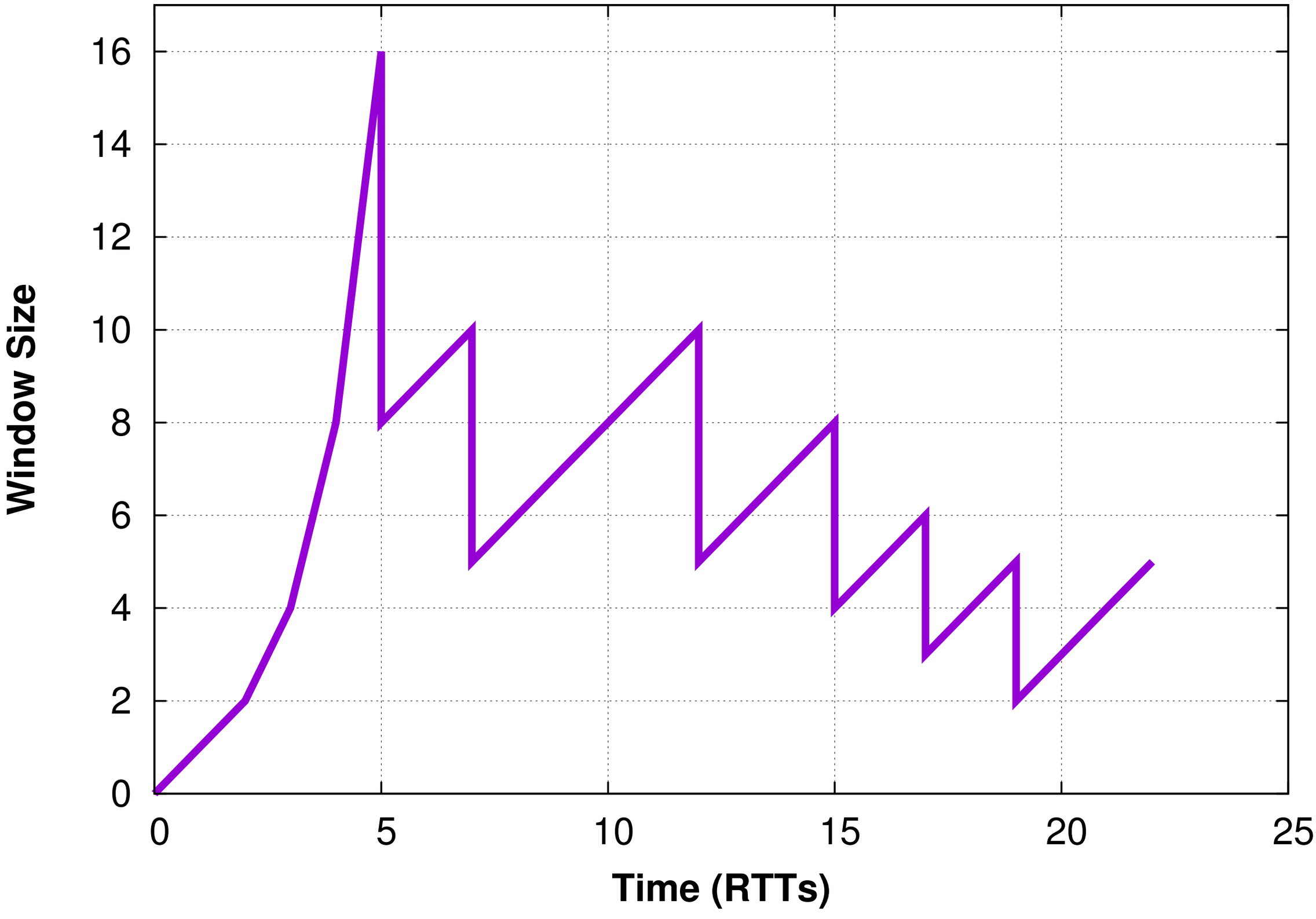
the network is fully utilized when the bottleneck link is "full"

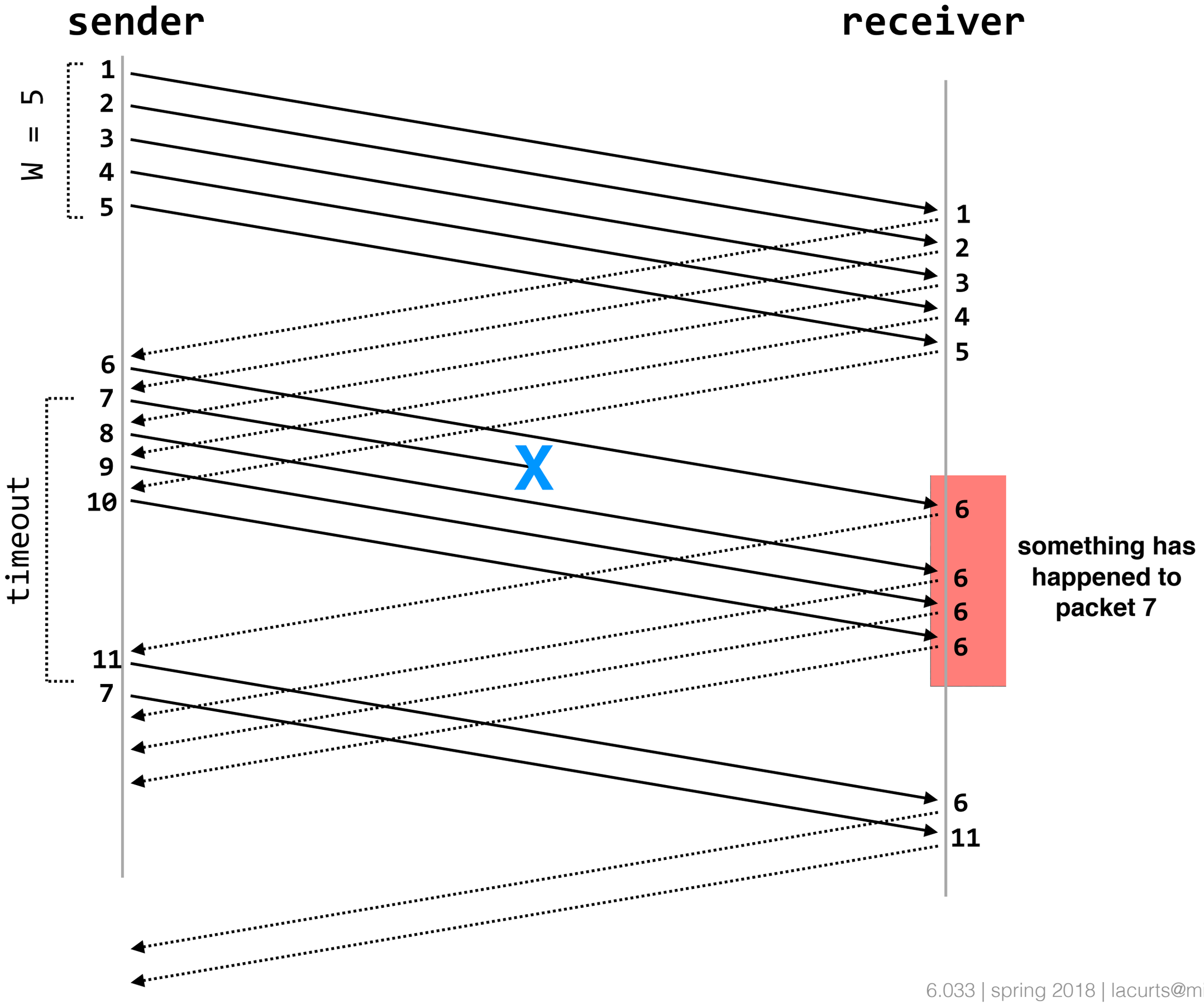
fairness

the network is fair when S_1 and S_2 are sending at the same rate

eventually, R1 and R2 will come to oscillate around the fixed point

AIMD + Slow Start




sender
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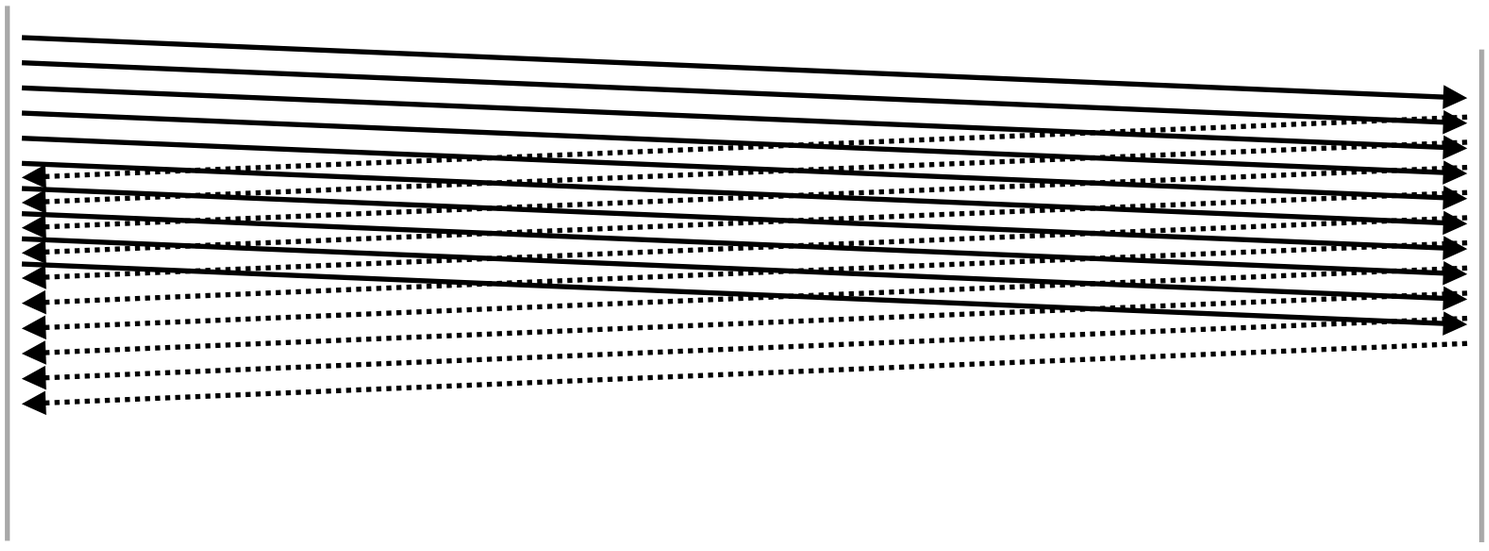
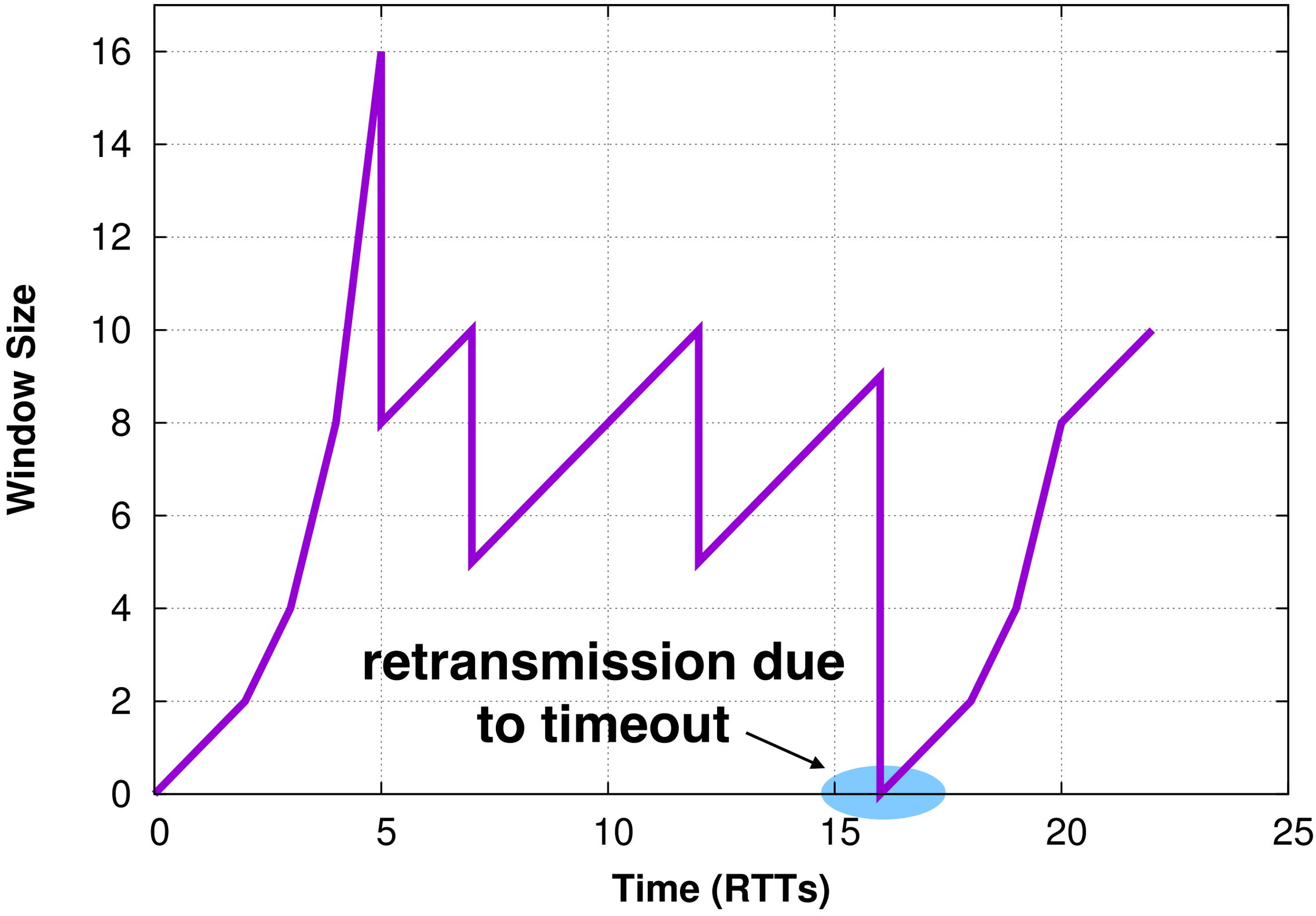
something has happened to packet 7



in practice, if a single packet is lost, the three “dup” ACKs will be received before the RTO for that packet expires

AIMD + Slow Start



- **TCP** provides **reliable transport** along with **congestion control**: senders increase their window additively until they experience loss, and then back off multiplicatively. Senders also use slow-start and fast-retransmit/fast-recovery to quickly increase the window and recover from loss.

- TCP has been a massive success, but **senders don't react to congestion until queues are already full**. Is there a better way?