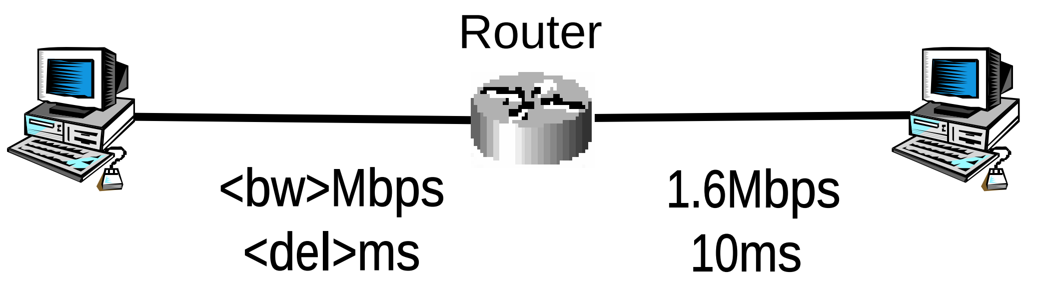
Pipelining examples and self-clocking

G.Bianchi, G.Neglia, V.Mancuso

Università degli Studi di Palermo

Pipelining

Packet size: 1000bytes



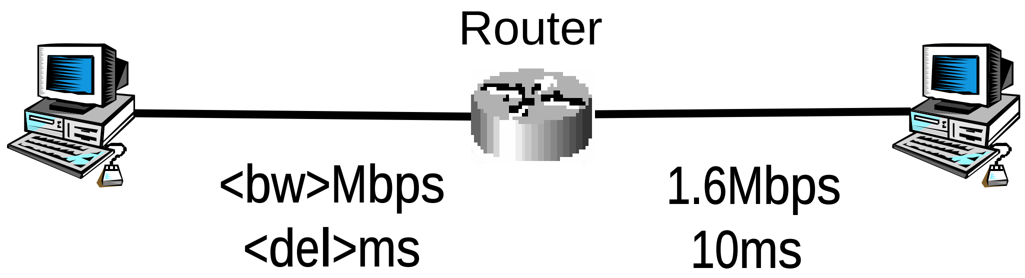
- ➔ **ns ack_clock2.tcl <bw> <del> <qu> <W> <stop>**
- ➔ **ns ack_clock2.tcl 1.6Mb 30ms 100 10 1**
- ➔ **ns ack_clock2.tcl 1.6Mb 30ms 100 17 1**
- ➔ **ns ack_clock2.tcl 1.6Mb 30ms 100 18 1**
- ➔ **ns ack_clock2.tcl 1.6Mb 30ms 100 36 1**

Self-clocking

- ➔**The ACK policy makes the protocol *self-clocking*:**
  - ⇨it dynamically adapts its transmission speed
  - ⇨*trying* to satisfy a conservation principle: a new packet for each old one leaving the network

Self-clocking (example)

Packet size: 1000bytes



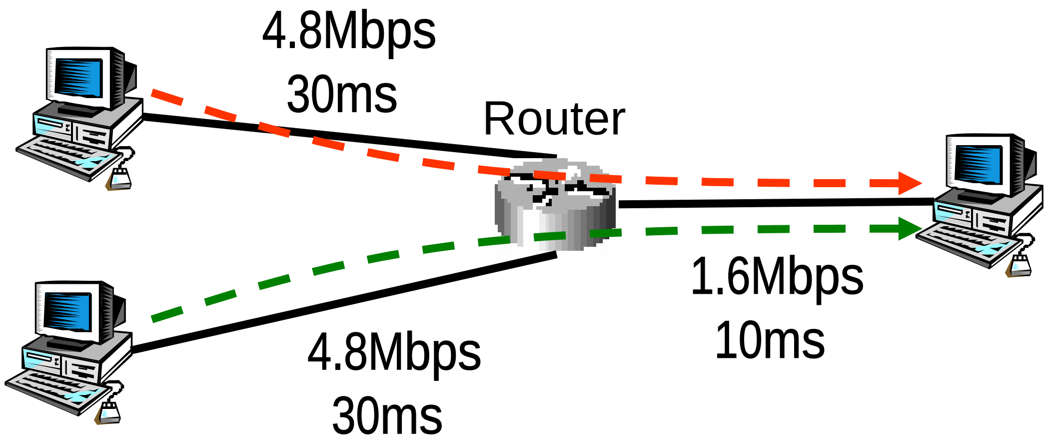
➔ **ns ack_clock2.tcl 4.8Mb 30ms 100 17 1**

➔ **ns ack_clock2.tcl 4.8Mb 30ms 100 18 1**

➔ **ns ack_clock2.tcl 4.8Mb 30ms 100 36 1**

Self-clocking: is it enough?

Packet size: 1000bytes



→ **ns congavd_motivation2.tcl 100 DropTail false false 4**

but…

→ **ns ack_clock2.tcl 4.8Mb 30ms 10 36 1**

→ **ns congavd_motivation2.tcl 10 DropTail false false 4**

TCP
congestion control

(very good summary in RFC 2581)

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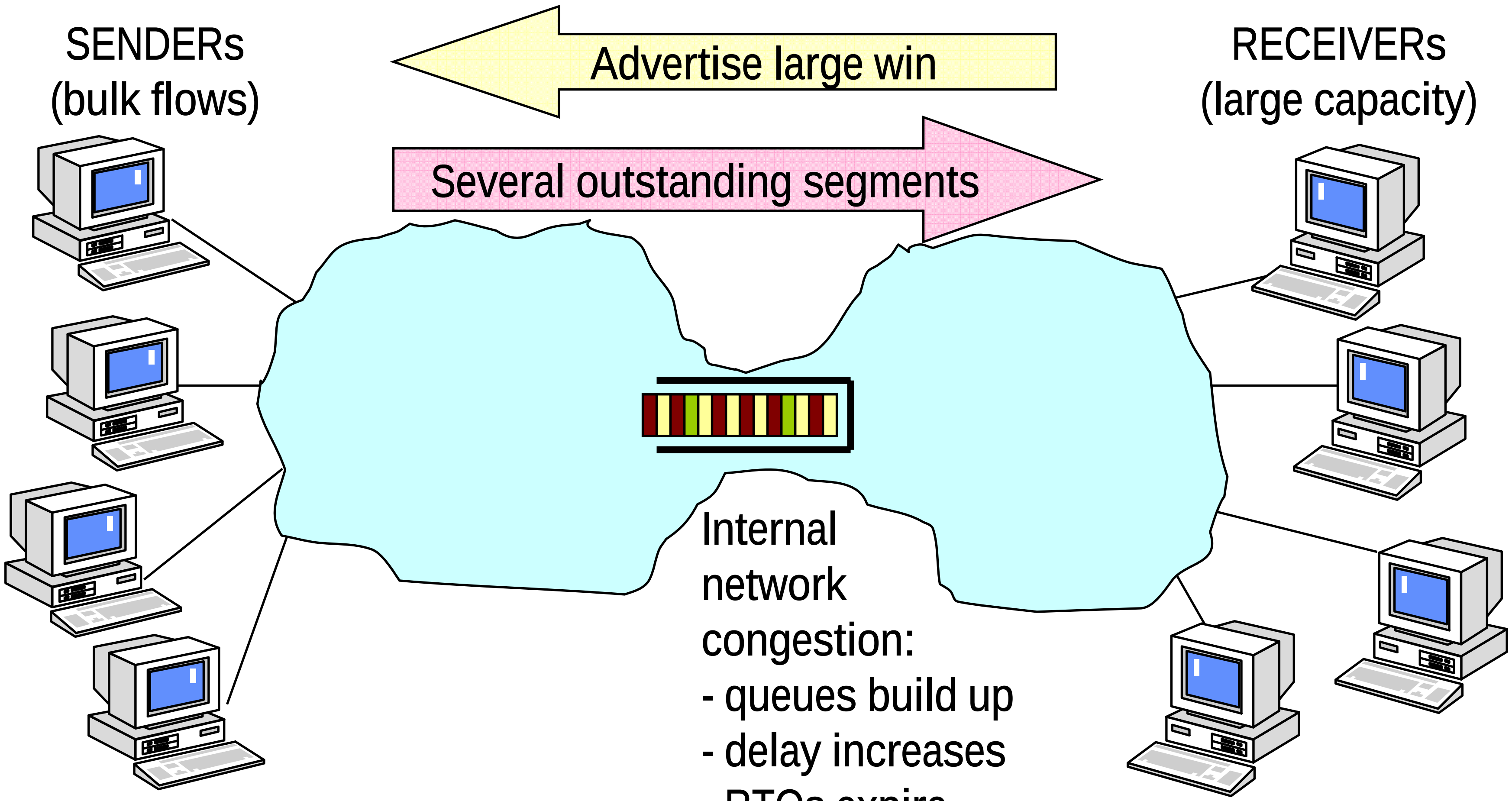
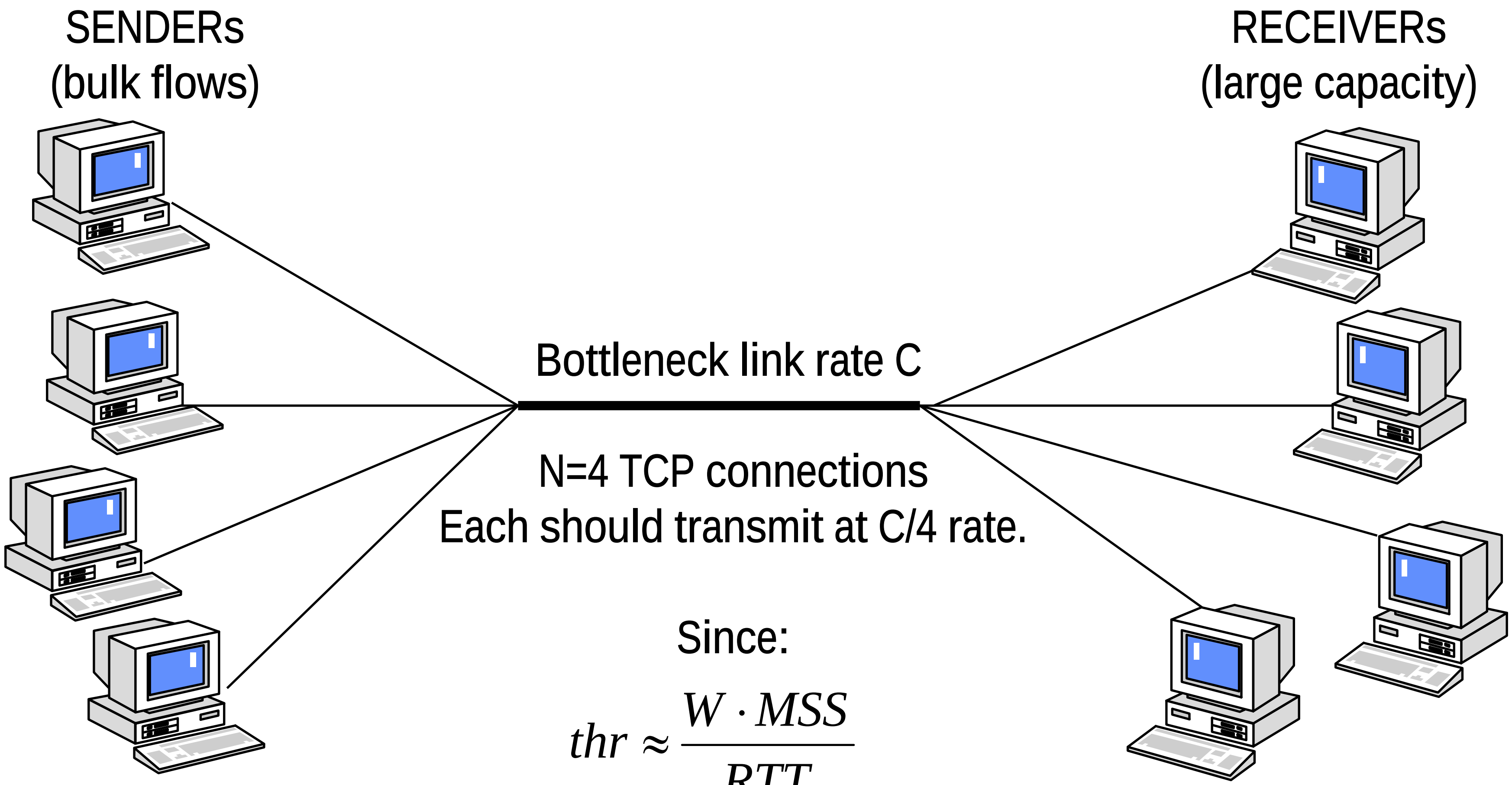
The problem of congestion



Internal network congestion:
- queues build up
- delay increases
- RTOs expire
- more segments transmitted -> more congestion!

The goal of congestion control



N=4 TCP connections
Each should transmit at C/4 rate.

Since:

$$thr \approx \frac{W \cdot MSS}{RTT}$$

Each should adapt W accordingly…
How sources can be lead to know the RIGHT value of W??

History of congestion control

- ➔ **Before 1986: the Internet meltdown!**
  - ⇨ No mechanisms employed to react to internal network congestion
- ➔ **1986: Slow Start + Congestion avoidance**
  - ⇨ Van Jacobson, TCP Berkeley
  - ⇨ Proposes idea to make TCP reactive to congestion
- ➔ **1988: Fast Retransmit (TCP Tahoe)**
  - ⇨ Van Jacobson, first implemented in 1988 BSD Tahoe release
- ➔ **1990: Fast Recovery (TCP Reno)**
  - ⇨ Van Jacobson, first implemented in 1990 BSD Reno release
- ➔ **1995-1996: TCP NewReno**
  - ⇨ Floyd (based on Hoe's idea), RFC 2582
  - ⇨ Today the de-facto standard

TCP approach for detecting and controlling congestion

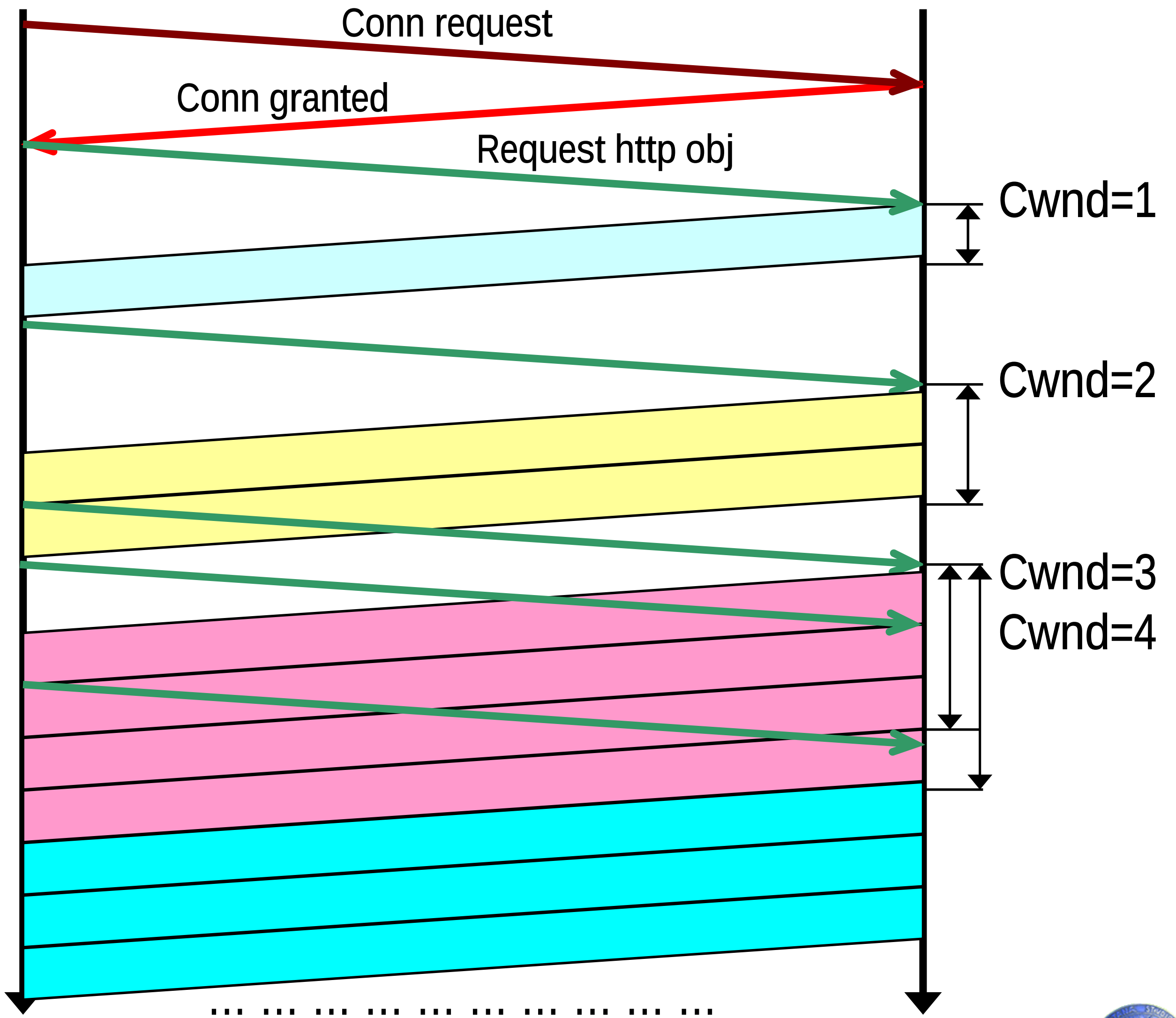
- ➔ **IP protocol does not implement mechanisms to detect congestion in IP routers**
  - →Unlike other networks, e.g. ATM
- ➔ **necessary indirect ways (TCP is an end-to-end protocol)**
- ➔ **TCP approach: congestion detected by lack of acks**
  - » couldn't work efficiently in the 60s & 70s (error prone transmission lines)
  - » OK in the 80s & 90s (reliable transmission)
  - » what about wireless networks???
- ➔ **Controlling congestion: use a SECOND window (congestion window)**
  - →Locally computed at sender
  - →Outstanding segments: min(receiver_window, congestion_window)

Starting a TCP transmission

- **A new offered flow may suddenly overload network nodes**
  - receiver window is used to avoid recv buffer overflow
  - But it may be a large value (16-64 KB)
- **Idea: slow start**
  - Start with small value of cwnd
  - And increase it as soon as packets get through
    - Arrival of ACKs = no packet losts = no congestion
- **Initial cwnd size:**
  - Just 1 MSS!
  - Recent (1998) proposals for more aggressive starts (up to 4 MSS) have been found to be dangerous

Slow start – exponential increase

- ➔ First start: set congestion window
  cwnd = 1MSS
- ➔ send cwnd segments
  - ⇨ assume cwnd <= receiver win
- ➔ upon successful reception:
  - ⇨ Cwnd +=1 MSS
  - ⇨ i.e. double cwnd every RTT
  - ⇨ until reaching receiver window advertisement
  - ⇨ <u>OR a segment gets lost</u>



Detecting congestion and restarting

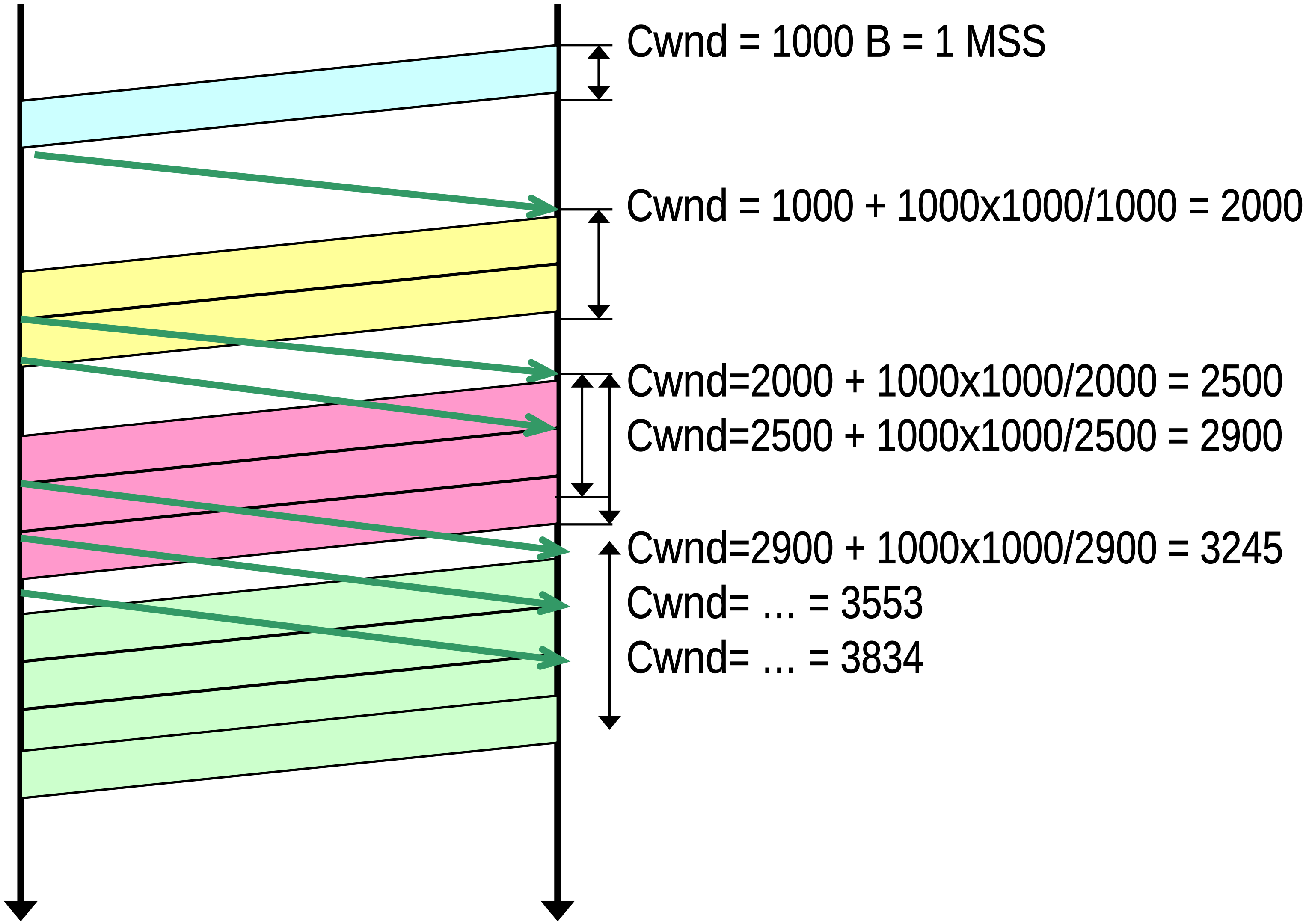
- ➔ **Segment gets lost**
  - ⇨ Detected via RTO expiration
  - ⇨ Indirectly notifies that one of the network nodes along the path has lost segment
    - » Because of full queue
- ➔ **Restart from cwnd=1 (slow start)**
- ➔ **But introduce a supplementary control: slow start threshold**
    - →sstresh = max(cwnd/2, 2MSS)
  - ⇨ The idea is that we now KNOW that there is congestion in the network, and we need to increase our rate in a more careful manner…
  - ⇨ ssthresh defines the "congestion avoidance" region

Congestion avoidance

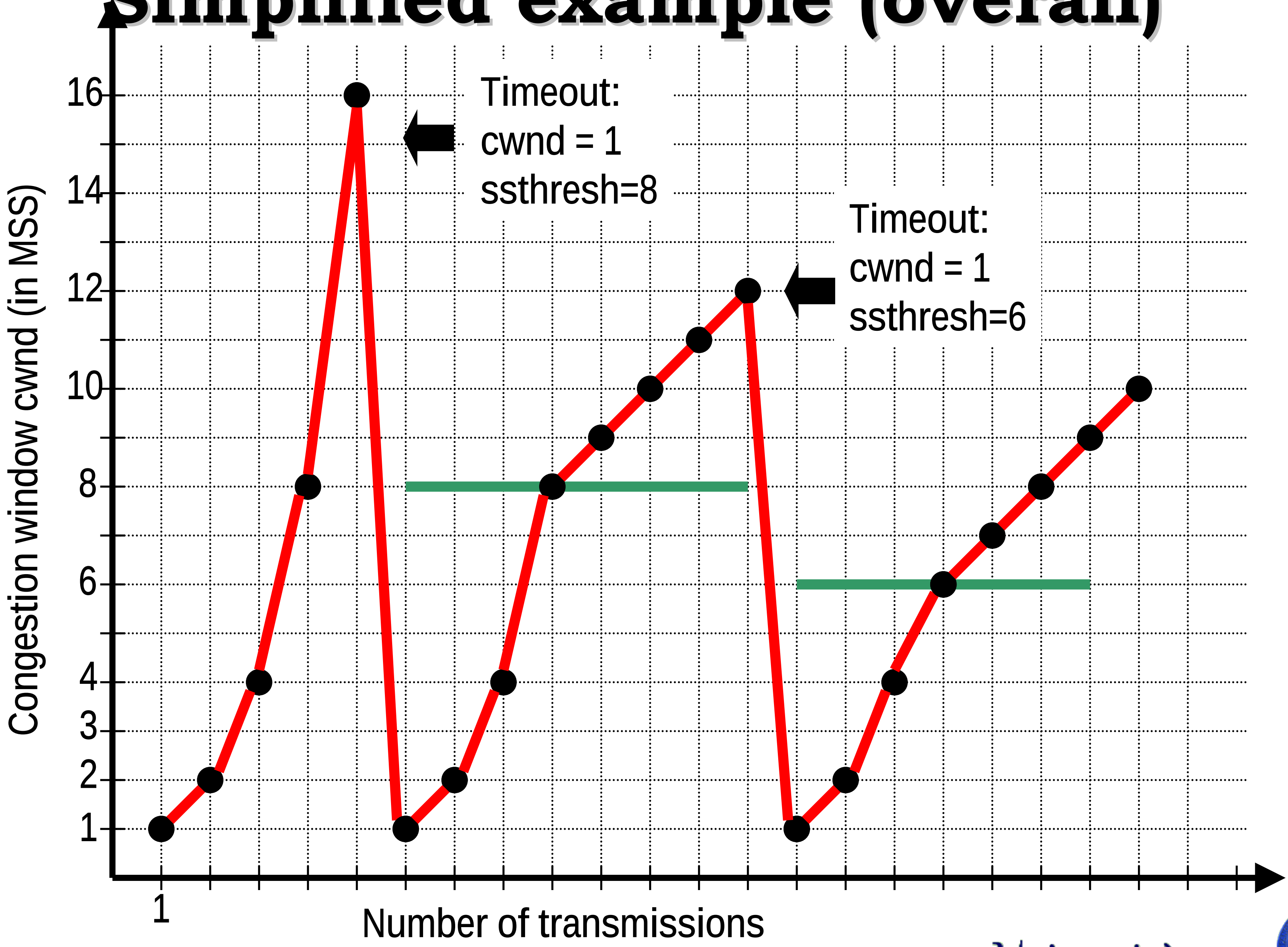
- **If cwnd < ssthresh**
  - Slow start region: Increase rate exponentially
- **If cwnd >= ssthresh**
  - Congestion avoidance region : Increase rate linearly
  - At rate 1 MSS per RTT
    - Practical implementation:
      cwnd += MSS*MSS/cwnd
    - Good approximation for 1 MSS per RTT
    - Alternative (exact) implementations: count!!
- **Which initial ssthresh?**
  - » ssthresh initially set to 65535: unreachable!

In essence, congestion avoidance is flow control imposed by sender while advertised window is flow control imposed by receiver

Congestion avoidance example



Simplified example (overall)



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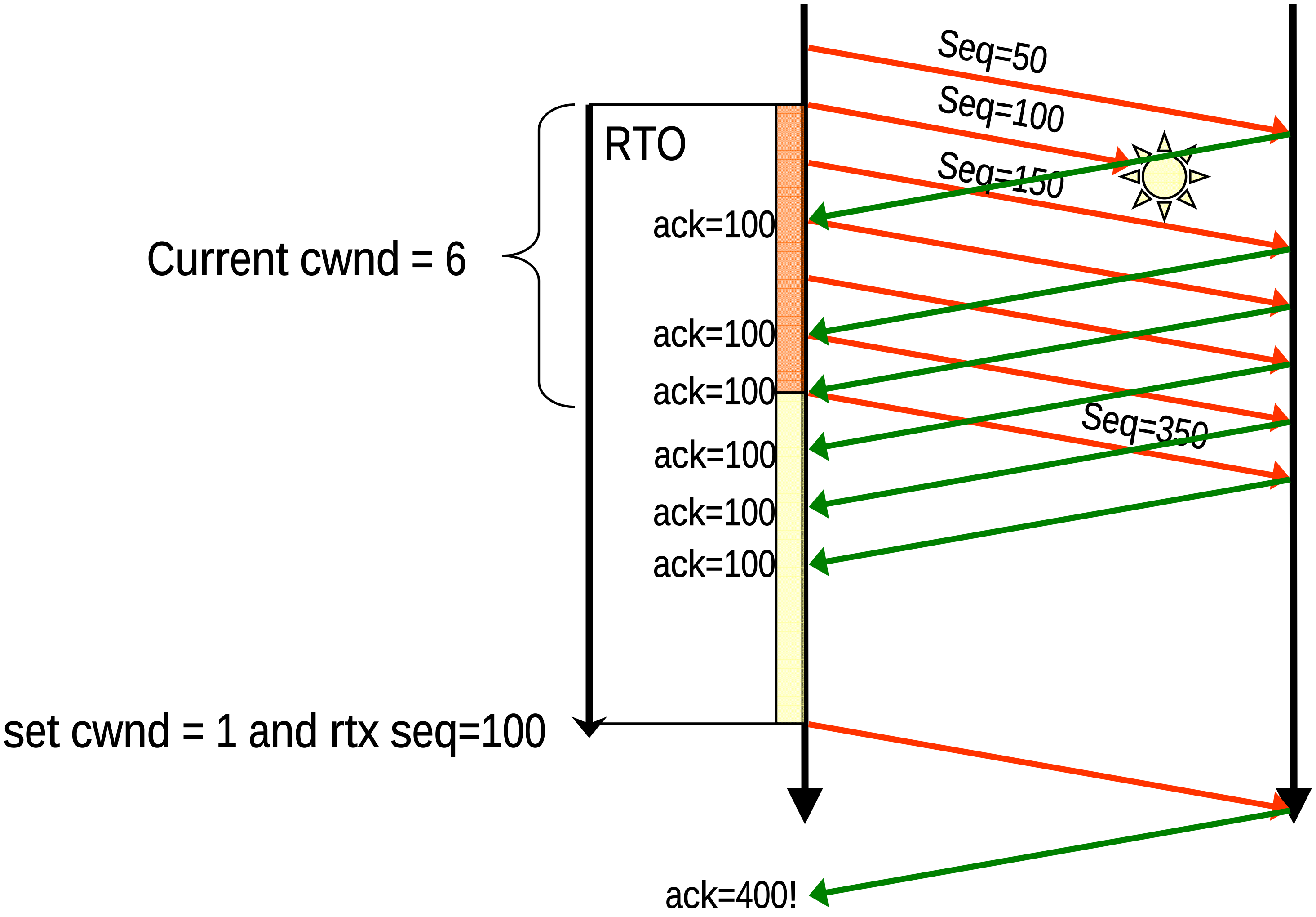
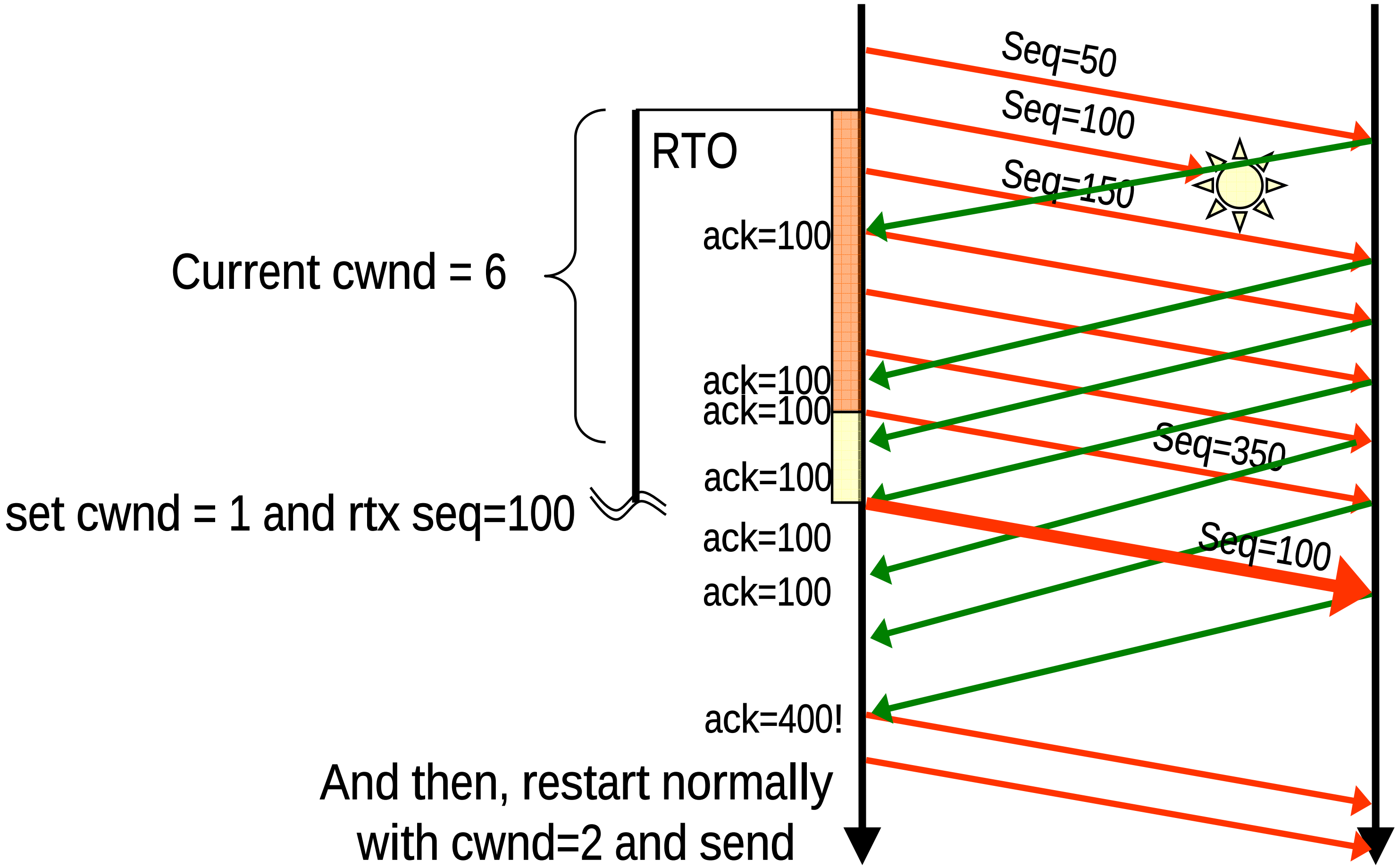
What happens AFTER RTO?
(without fast retransmit)



And then, restart normally with cwnd=2 and send seq=400,450

TCP TAHOE
(with fast retransmit)



And then, restart normally
with cwnd=2 and send
seq=400,450

Same as before, but shorter time to recover packet loss!

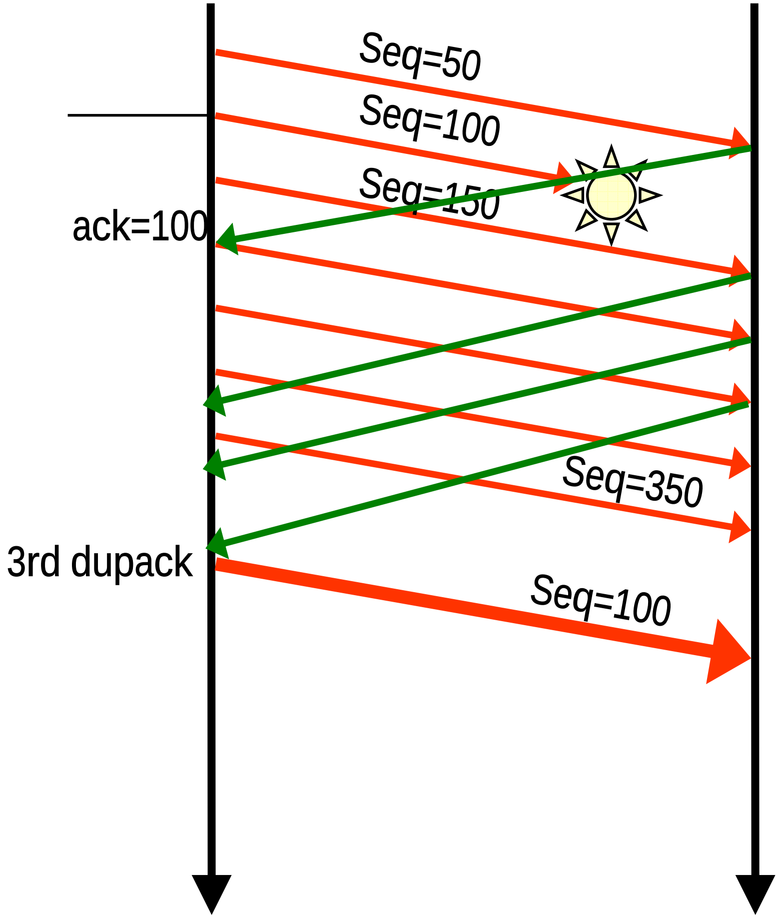
Motivations for fast recovery

FAST RECOVERY:

- ⇨ The phase following fast retransmit (3 duplicate acks received)

⇨ TAHOE approach: slow start, to protect network after congestion

⇨ However, since subsequent acks have been received, no hard congestion situation should be present in the network: slow start is a too conservative restart!



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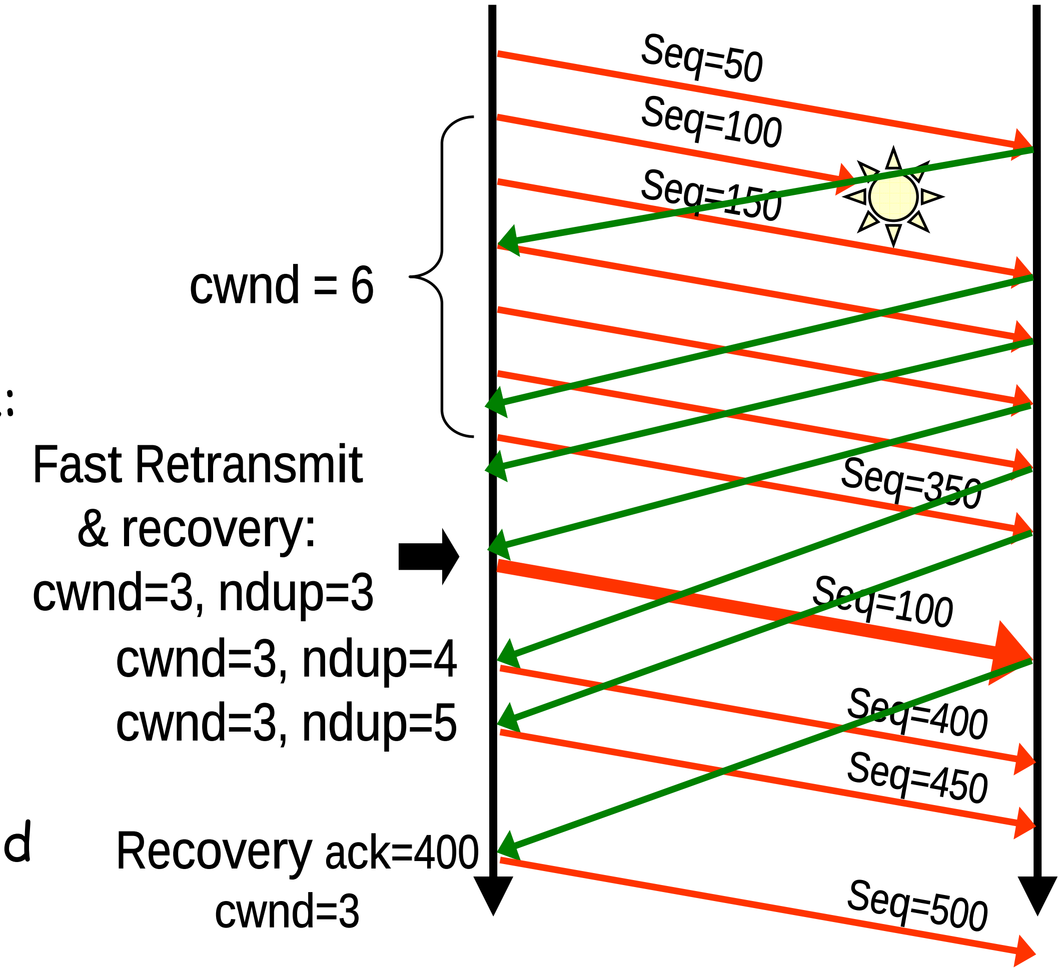
Fast recovery rules

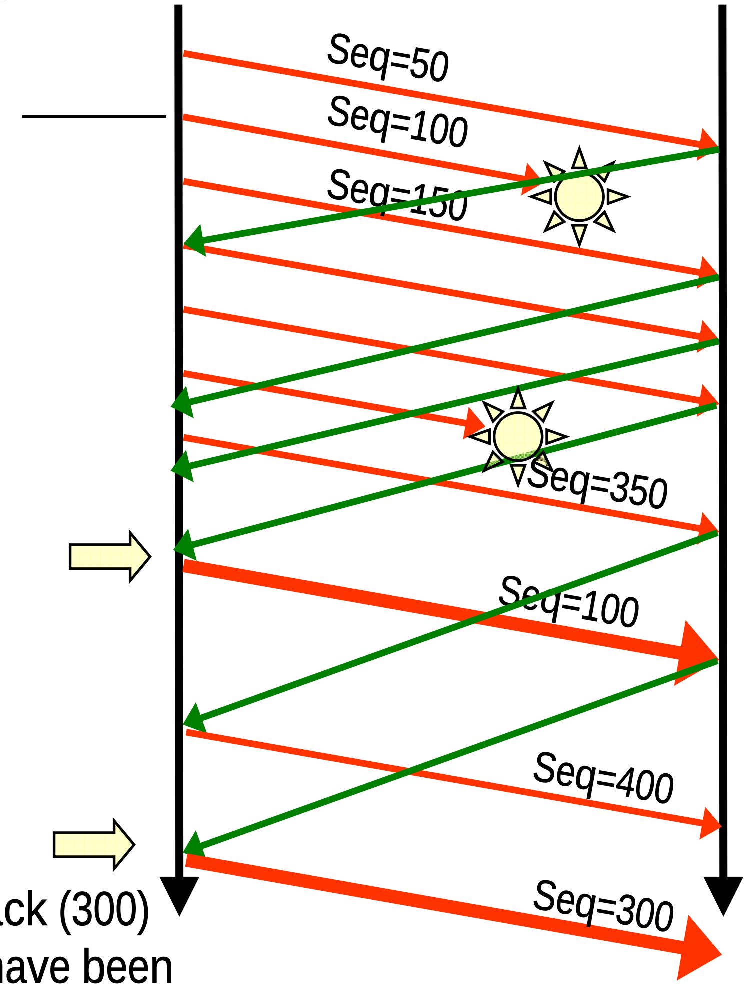
FAST RECOVERY RULES:

- ⇨ Retransmit lost segment
- ⇨ **Set cwnd = ssthresh = cwnd/2**
- ⇨ **Restart with congestion avoidance (linear)**
- ⇨ start fast recovery phase:
  - ⇨ Set counter for duplicate packets ndup=3
  - ⇨ Use "inflated" window: w = cwnd+ndup
  - ⇨ Upon new dup_acks, increase ndup, not cwnd (and send new data)
  - ⇨ Upon recovery ack, "deflate" window setting ndup=0



What about multiple losses?

- ➔ **TCP Reno optimized for single loss**
- ➔ **Performance drawbacks with multiple losses in same window**
- ➔ **Improvement: NewReno**
  - ⇨ Distinguish recovery ack from partial ack
    - →Equal to the recovery ack, but does not recover for all ndup segments
  - ⇨ Does not exit fast recovery when partial ack received
  - ⇨ Retransmit segment immediately following partial ack, assuming it was lost



Partial ack (300)
Would have been
recovery ack if ack=400

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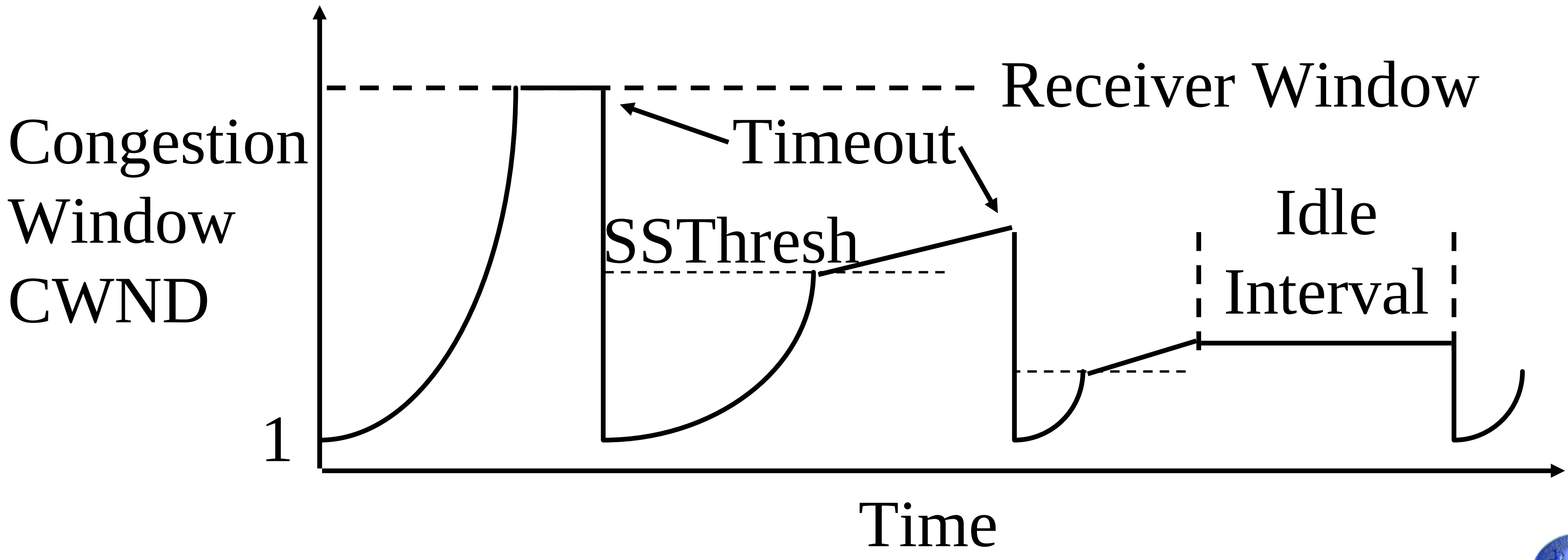
Idle periods

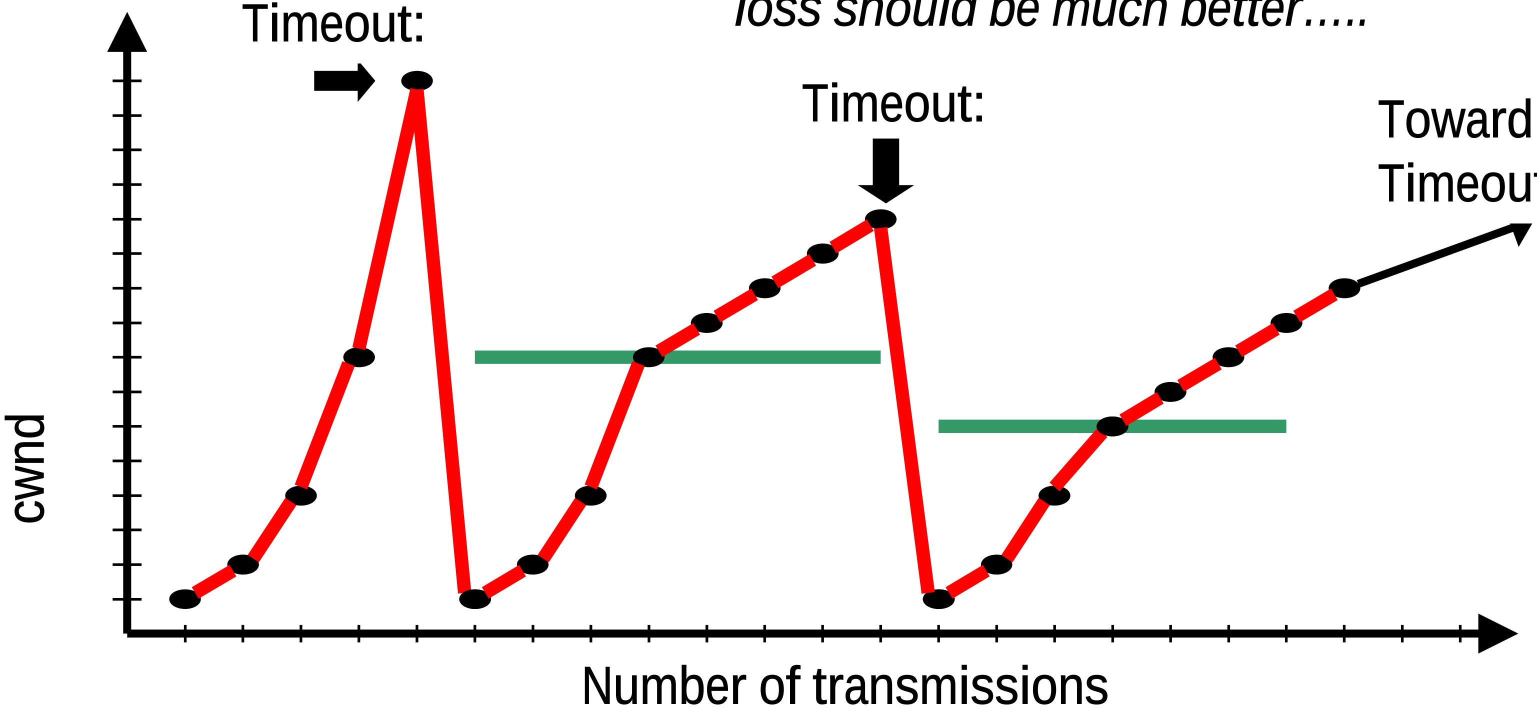
➔**After a long idle period (exceeding one RTO), reset the congestion window to one.**



Further TCP issues

Timeout = packet loss occurrence in an internal network router
TCP (both Tahoe & Reno) does not AVOID packet loss
Simply REACTS to packet loss

CONCLUSION: a TCP able to AVOID packet loss should be much better…..



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TCP Vegas (1995)

➔ **Avoids packet loss by predicting it!**
- ⇨ Approach: monitor RTT
- ⇨ when RTT shows increase, deduce that congestion is going to occur
- ⇨ and thus preventively reduce cwnd
- ⇨ but not down to as low as slow start

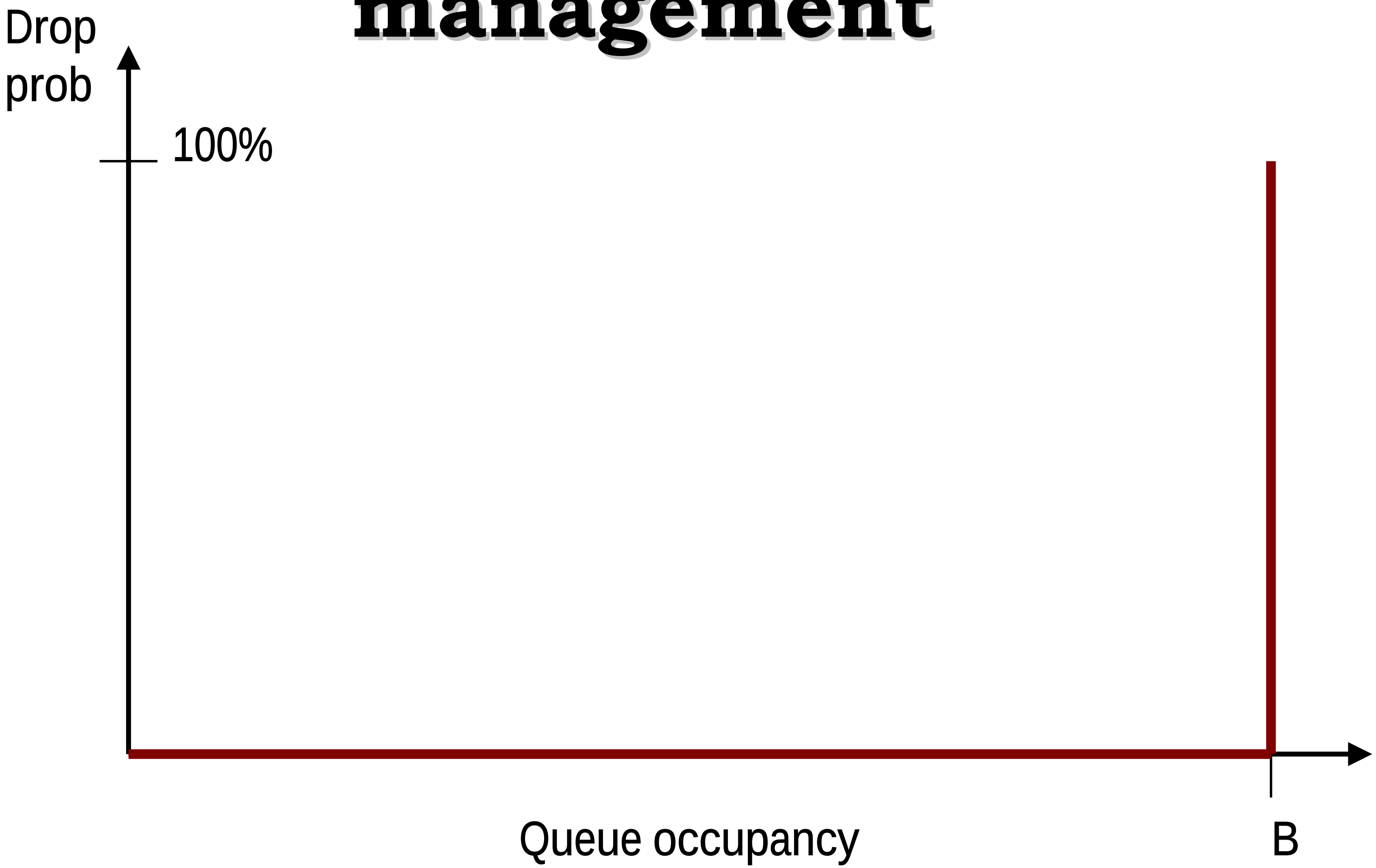
➔ **A problem: DOES NOT WORK WHEN OTHER TERMINALS USE TAHOE/RENO!!!!**
- ⇨ Vegas reduces rate to avoid congestion
- ⇨ while Tahoe/Reno grab the available bandwidth!!

A typical problem in Internet Protocol design: need to live with legacy apps and protoc

Recent Trends in congestion control

- ➔**End to end TCP congestion control not sufficient!**
- ➔**Active Queue Management (1994, 1998+)**
  - ⇨RED queueing discipline

Standard (Drop-tail) buffer management



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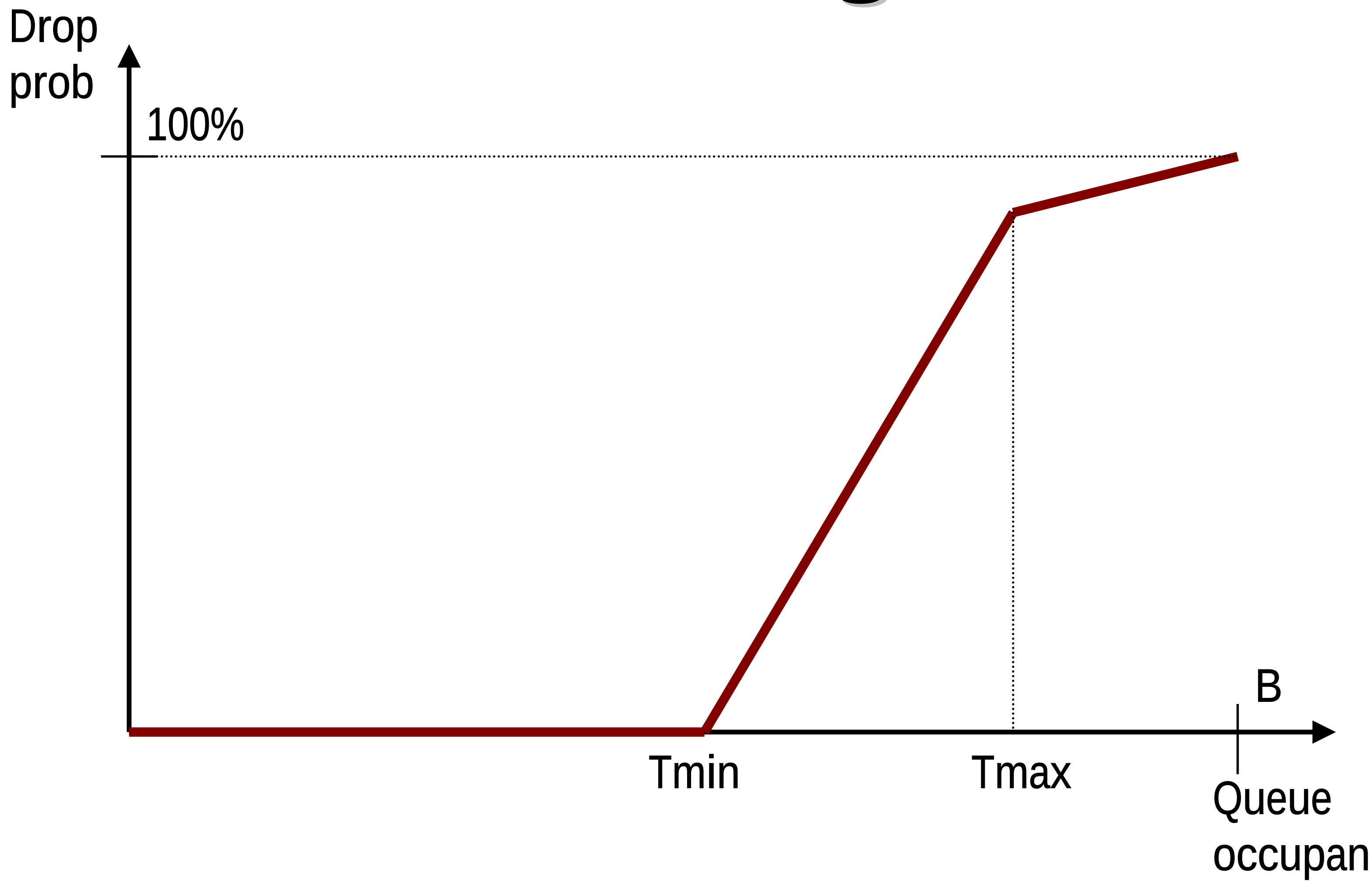
RED buffer management



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Fairness with UDP traffic

➔A serious problem for TCP

- ⇨in heavy network load, TCP reduces transmission rate. Non congestion-controlled traffic does not.
- ⇨Result: in link overload, TCP throughput vanishes!

Mixing TCP & UDP traffic