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SAPIENZA
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# Chapter 3
# Transport Layer

Reti di Elaboratori

Corso di Laurea in Informatica

Università degli Studi di Roma "La Sapienza"

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Parte di queste slide sono state prese dal materiale associato al libro *Computer Networking: A Top Down Approach* , 5th edition.

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# Chapter 3 outline

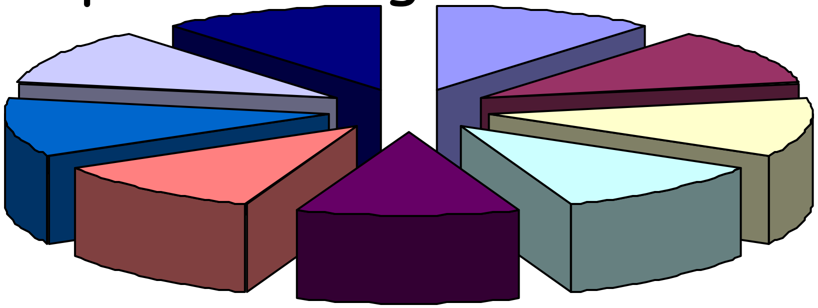
- 3.1 Transport-layer services
- 3.2 Multiplexing and demultiplexing
- 3.3 Connectionless transport: UDP
- 3.4 Principles of reliable data transfer
- 3.5 Connection-oriented transport: TCP
  - segment structure
  - reliable data transfer
  - flow control
  - connection management
- 3.6 Principles of congestion control
- 3.7 TCP congestion control

# TCP: controllo di congestione

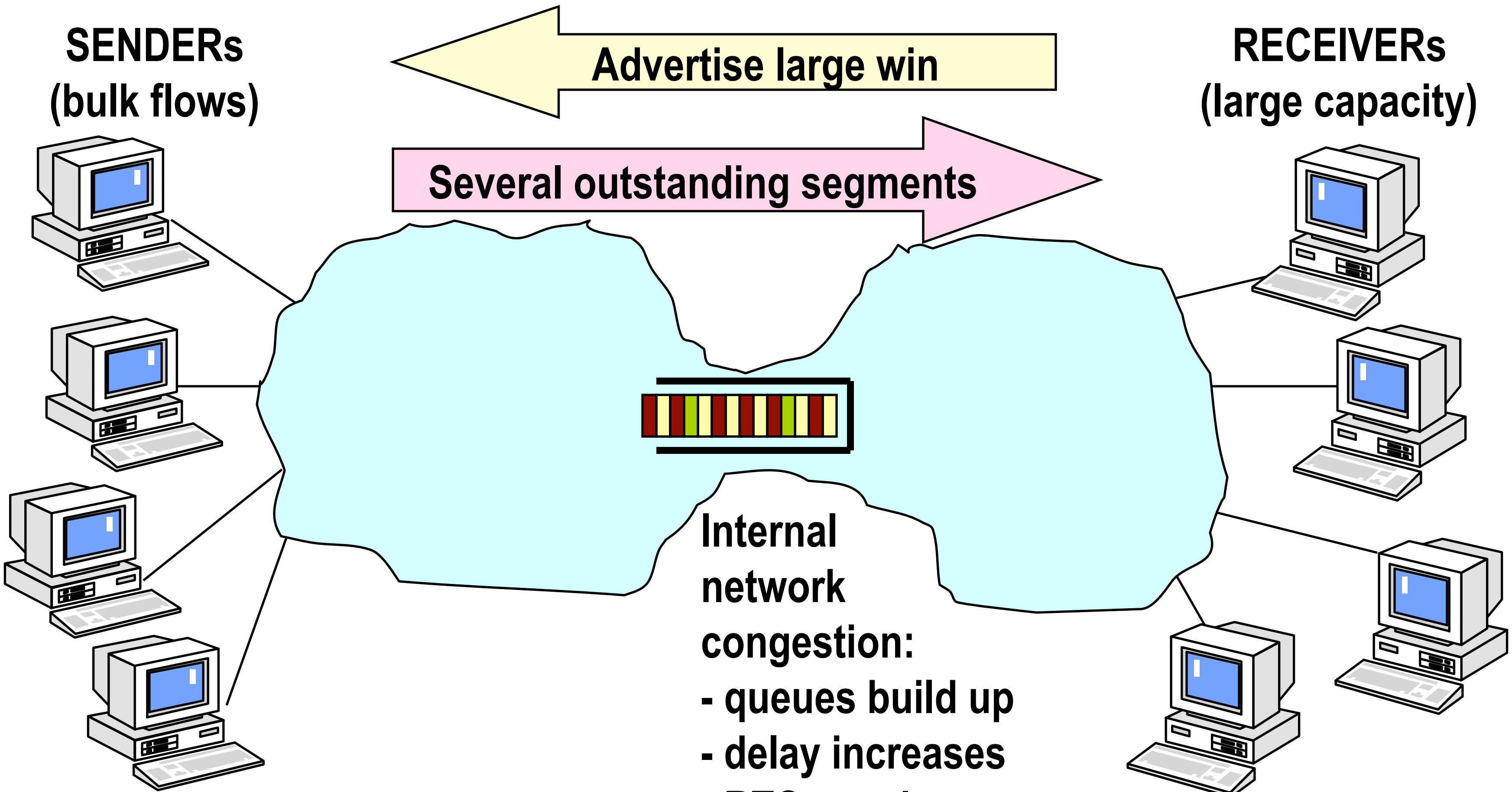
- Il TCP ha dei meccanismi di controllo della congestione
  - il flusso dei dati in ingresso in rete è anche regolato dalla situazione di traffico in rete
  - se il traffico in rete porta a situazioni di congestione il TCP riduce velocemente il traffico in ingresso
  - in rete non vi è nessun meccanismo per notificare esplicitamente le situazioni di congestione
  - il TCP cerca di scoprire i problemi di congestione sulla base degli eventi di perdita dei pacchetti

# TCP: controllo di congestione

- il meccanismo si basa ancora sulla sliding window la cui larghezza viene dinamicamente regolata in base alle condizioni in rete
- in linea di principio scopo del controllo è far si che il flusso emesso da ciascuna sorgente venga regolato in modo tale che il flusso complessivo offerto a ciascun canale non superi la sua capacità
- tutti i flussi possono essere ridotti in modo tale che la capacità della rete venga condivisa da tutti in misura se possibile uguale

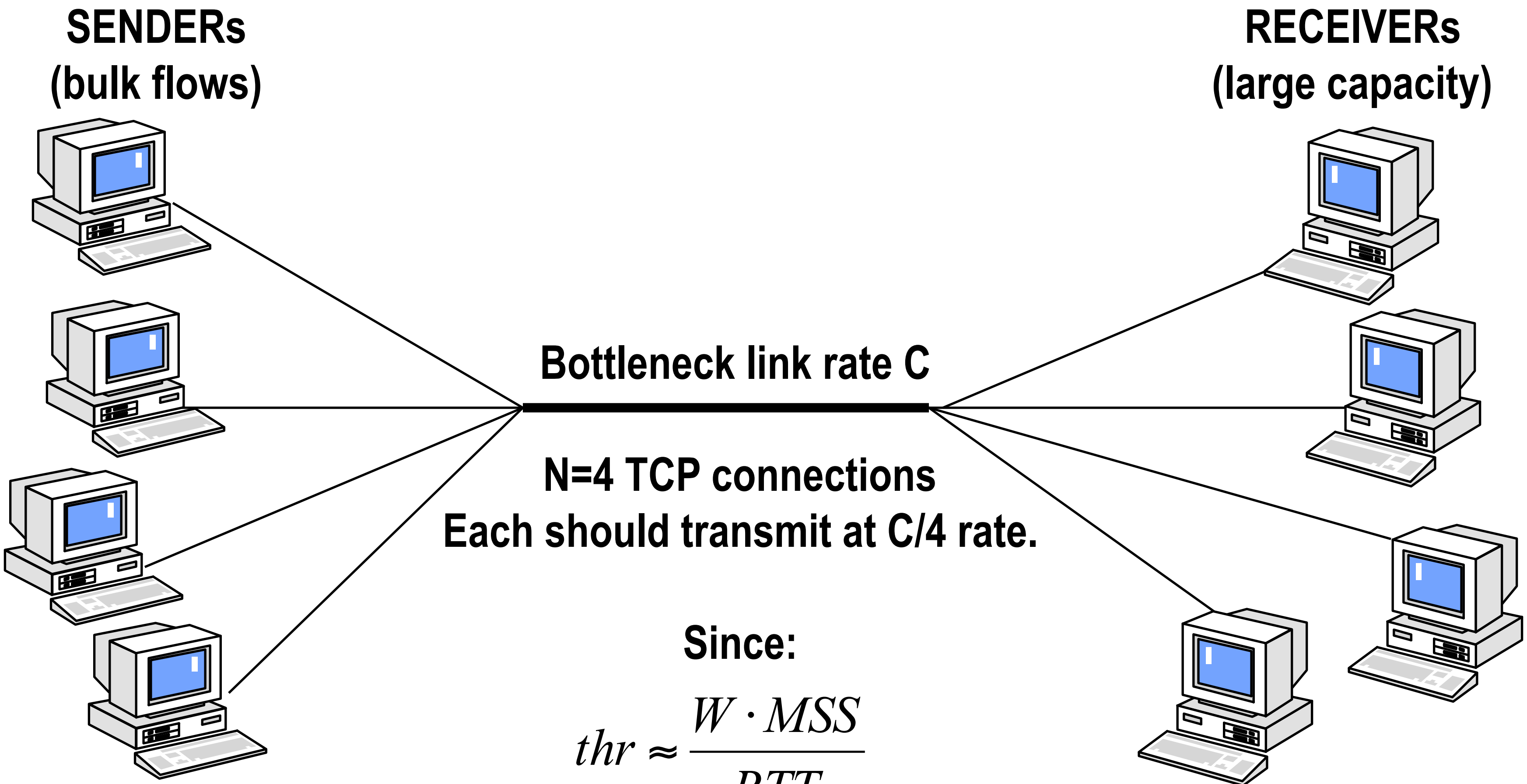


# The problem of congestion




Internal network congestion:
- queues build up
- delay increases
- RTOs expire
-more segments transmitted, more Segments retransmitted -> 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??**

# 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 means (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)

# TCP Congestion Control

- end-end control (no network assistance)
- sender limits transmission:

  **LastByteSent-LastByteAcked ≤ CongWin**
- Roughly,

$$\text{rate} = \frac{\text{CongWin}}{\text{RTT}} \text{ Bytes/sec}$$

- **CongWin** is dynamic, function of perceived network congestion

## How does sender perceive congestion?

- loss event = timeout *or* 3 duplicate acks
- TCP sender reduces rate (**CongWin**) after loss event

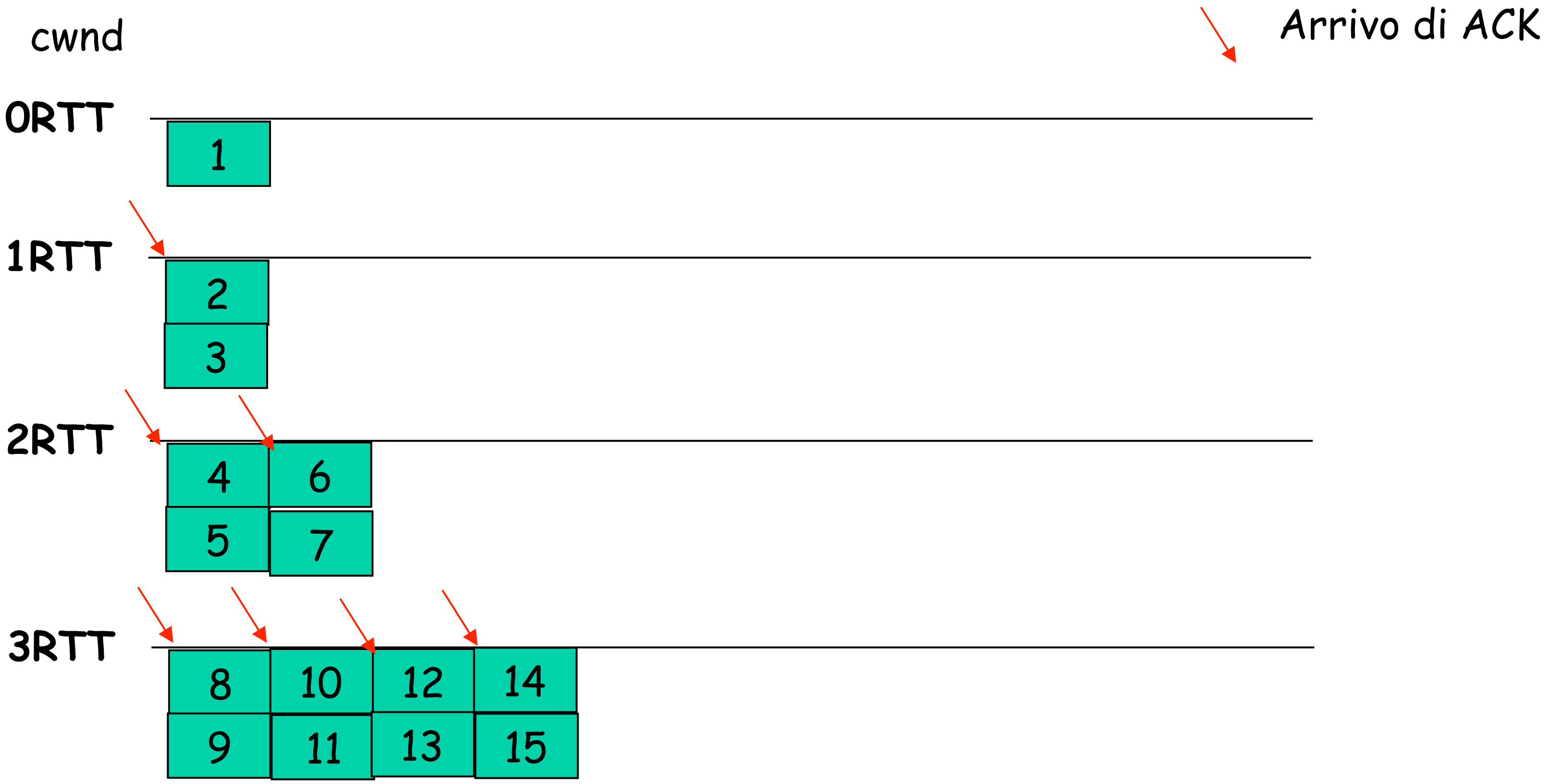
## three mechanisms:

- AIMD
- slow start
- conservative after timeout events

# 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

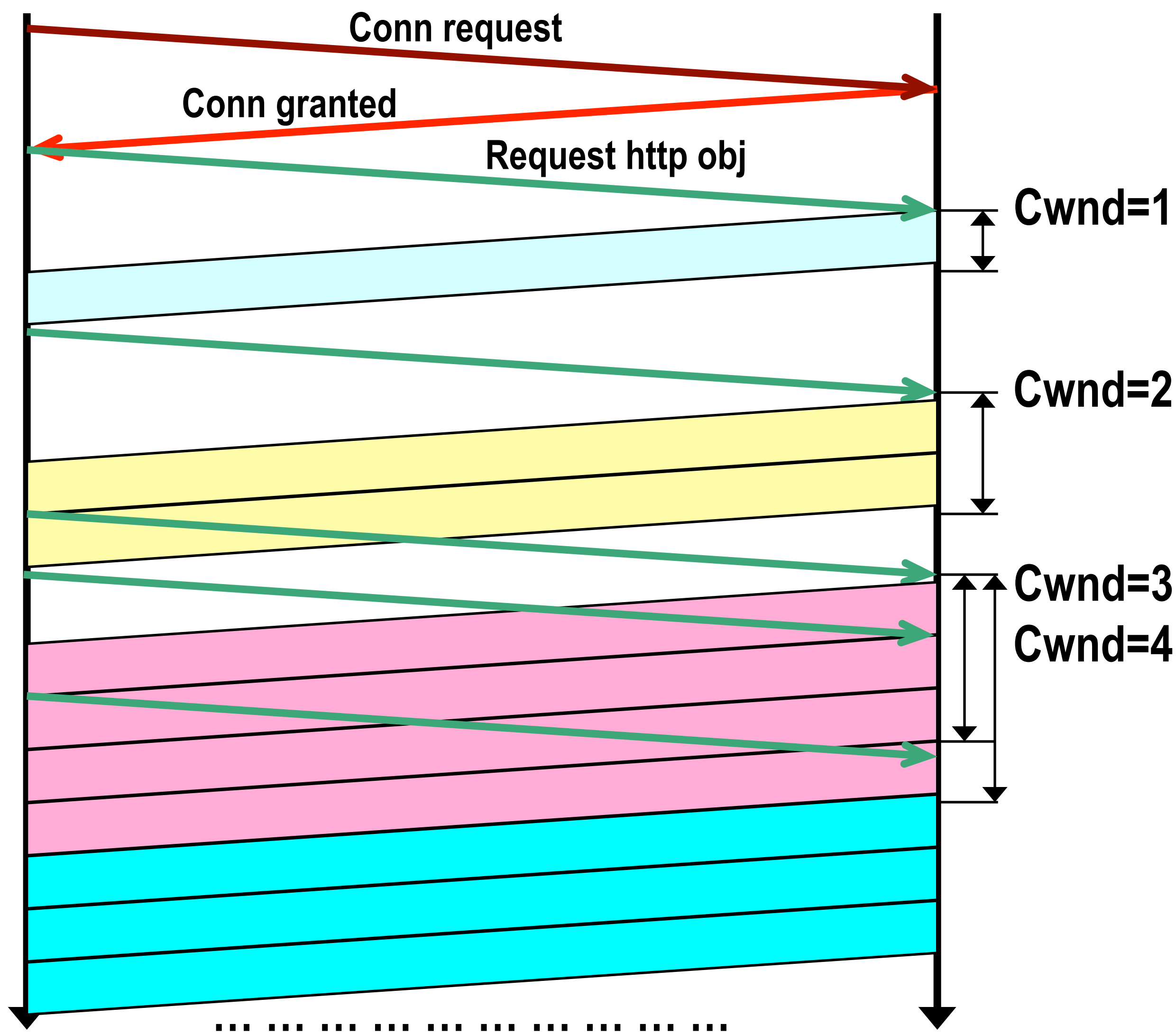
# Slow start: the idea




Si trasmette il minimo tra window e cwd pacchetti

# 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
  - ⇨ OR a segment gets lost



# 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(min(cwnd,window)/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

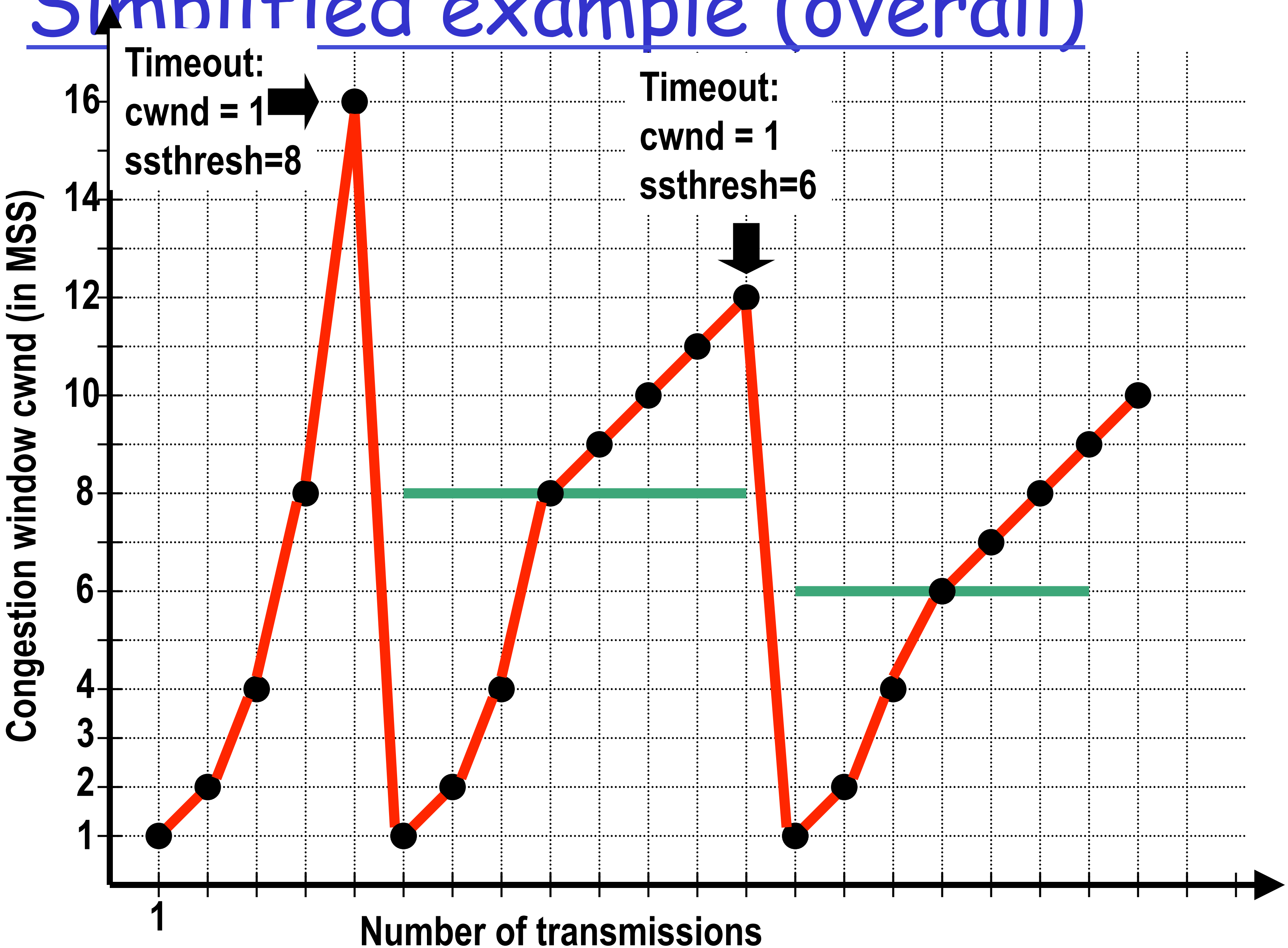
- If cwnd < ssthresh
  - Slow start region: Increase rate exponentially
- If cwnd >= ssthresh
  - Congestion avoidance region : Increase rate linearly
  - At rate 1 MSS per RTT
    - Practical implementation:
      cwnd += MSS*MSS/cwnd
    - Good approximation for 1 MSS per RTT
    - Alternative (exact)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

Corrisponde ad un segmento per finestra

- Which initial ssthresh?
  - – ssthresh initially set to 65535: unreachable!

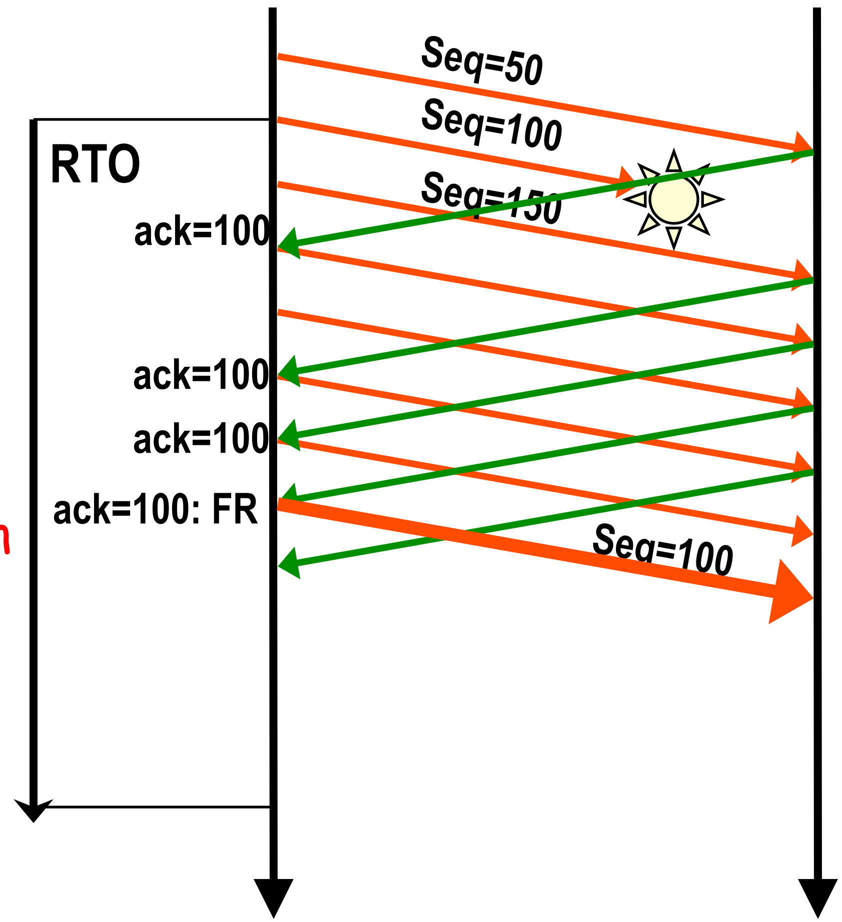
***In essence, congestion avoidance is flow control imposed by sender while advertised window is flow control imposed by receiver***

# Simplified example (overall)

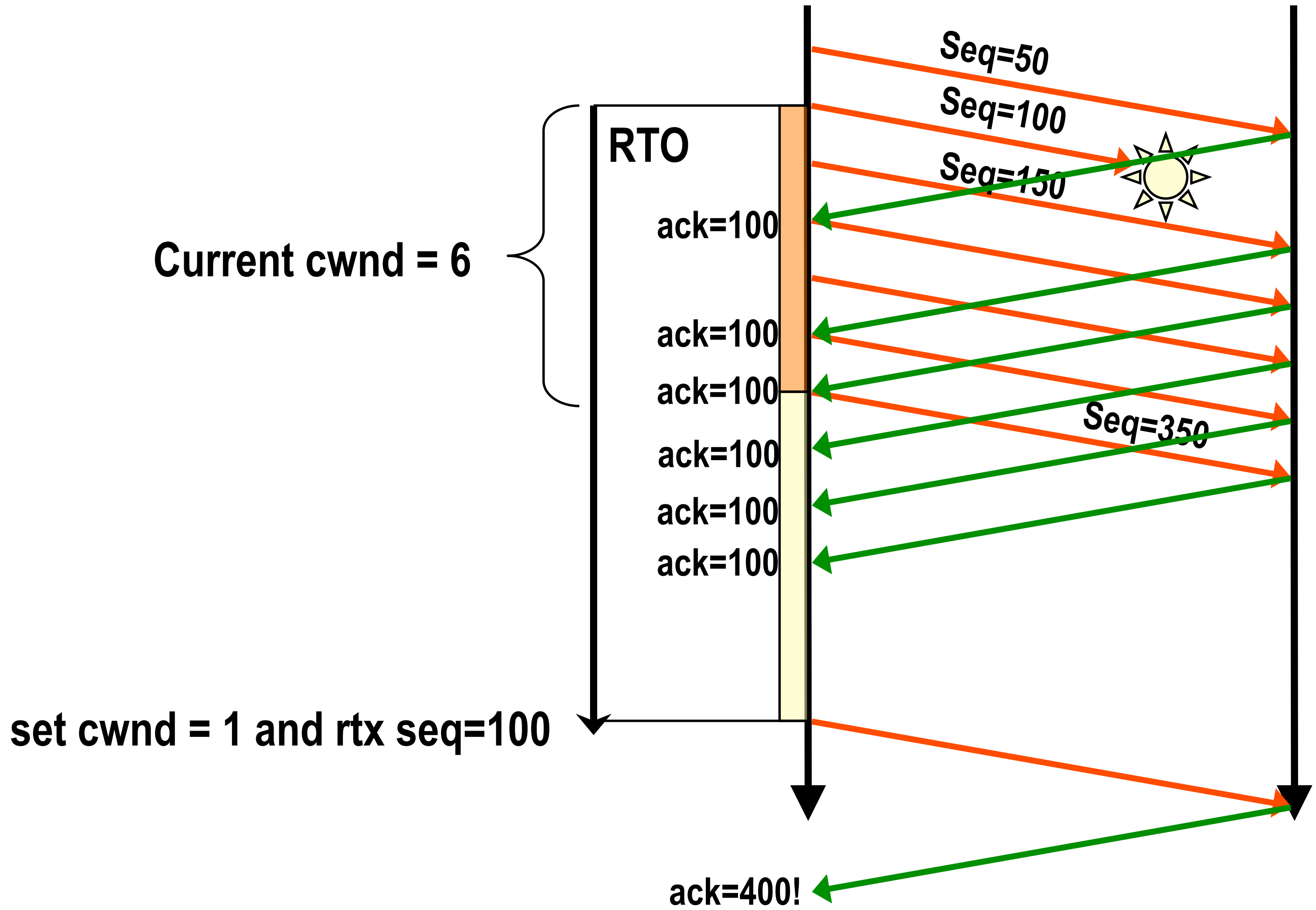


# The Fast Retransmit Algorithm

- Idea: use duplicate ACKs!
  - Receiver responds with an ACK every time it receives an out-of-order segment
  - ACK value = last correctly received segment
- FAST RETRANSMIT algorithm:
  - if 3 duplicate acks are received for the same segment, assume that the next segment has been lost. Retransmit it right away.
  - Helps if single packet lost. Not very effective with multiple losses
- And then? A congestion control issue…



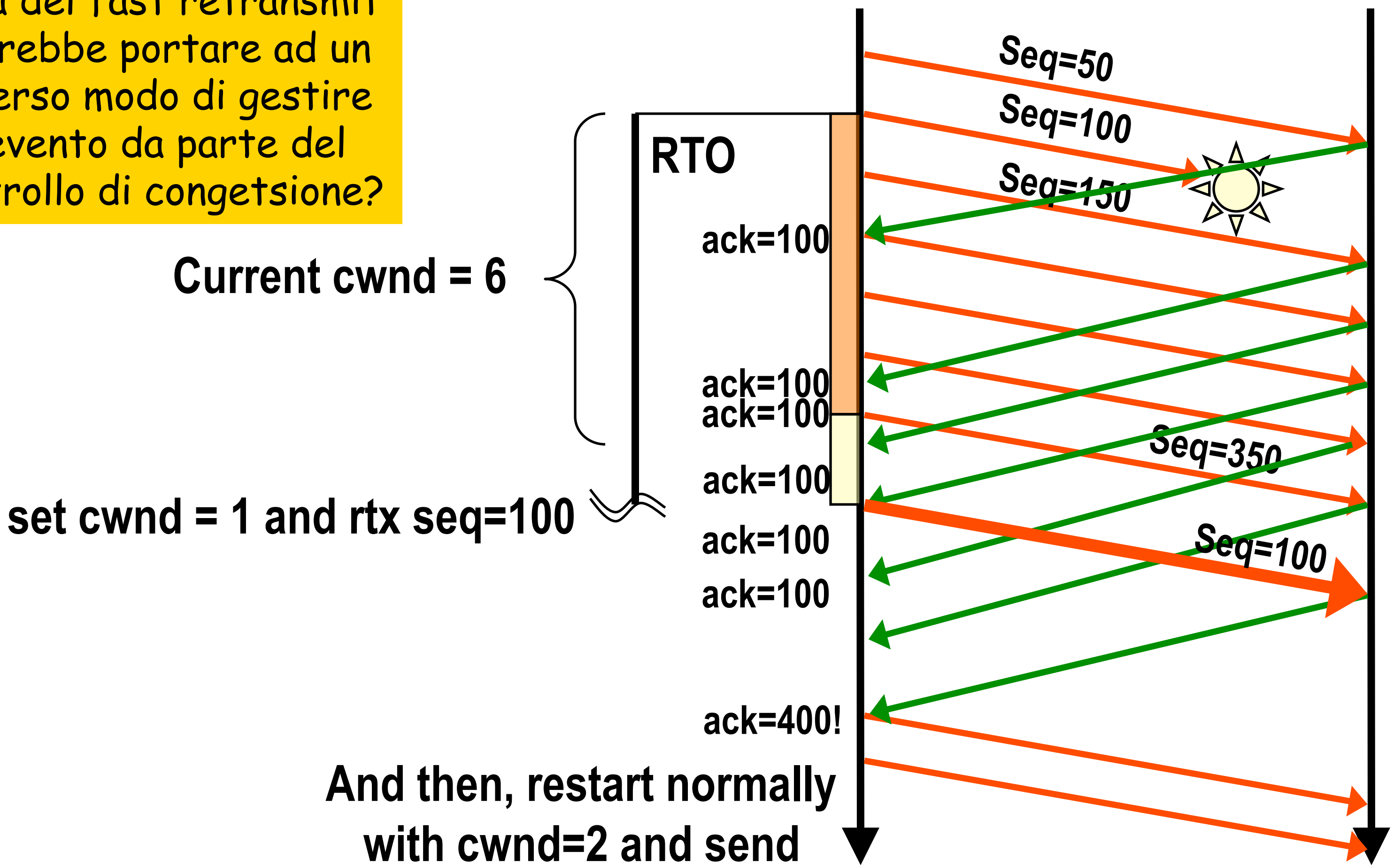
# What happens AFTER RTO? (without fast retransmit)




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

# TCP RENO
## (with fast retransmit)

Idea del fast retransmit
Dovrebbe portare ad un
Diverso modo di gestire
L'evento da parte del
Controllo di congetsione?




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

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

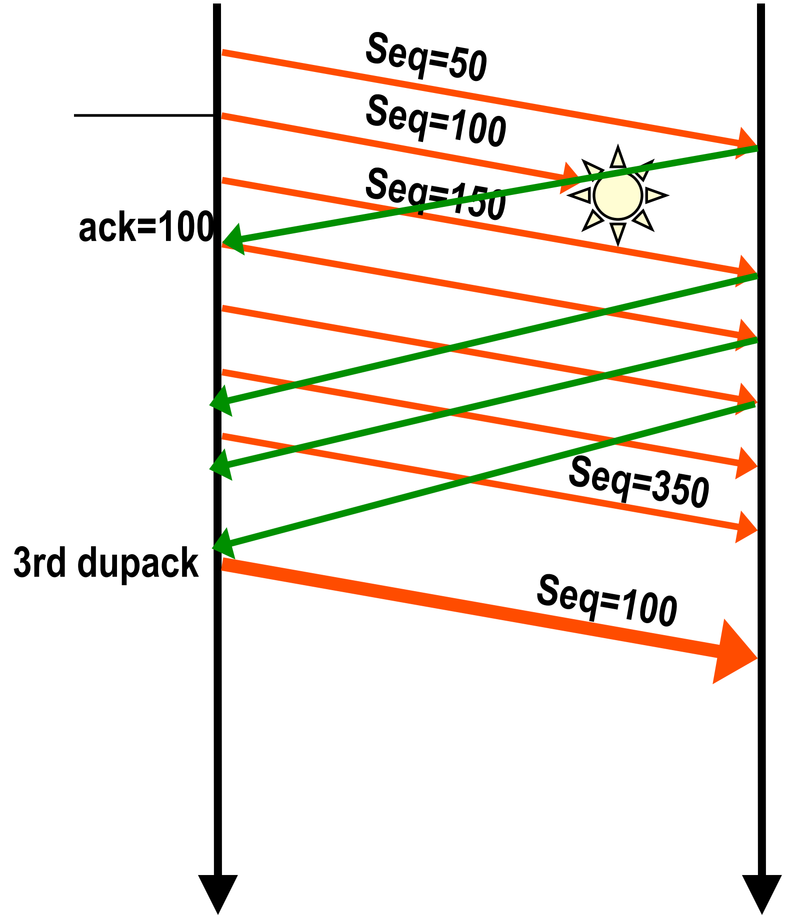
# Motivations for fast recovery

FAST RECOVERY:

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

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

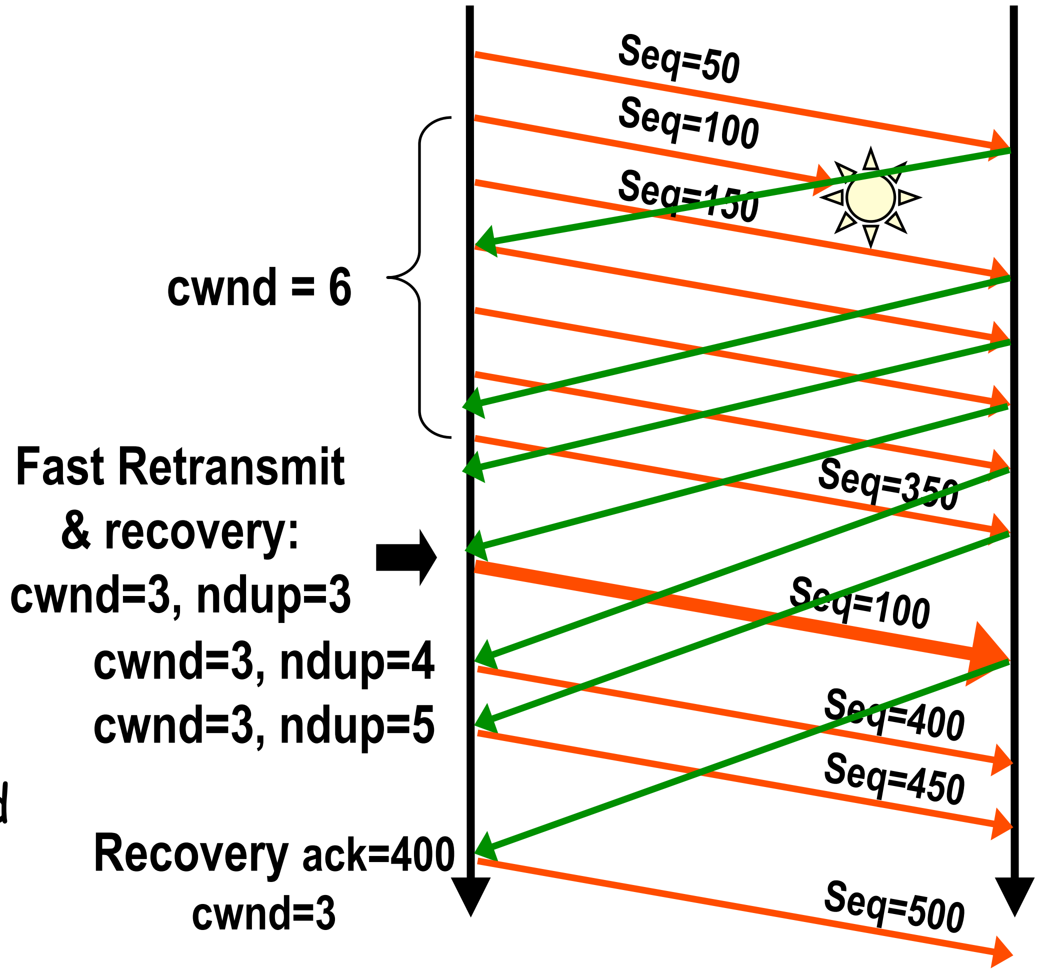
⇨ However, since subsequent acks have been received, no hard congestion situation should be present in the network: slow start is a too conservative restart!



# Fast recovery rules

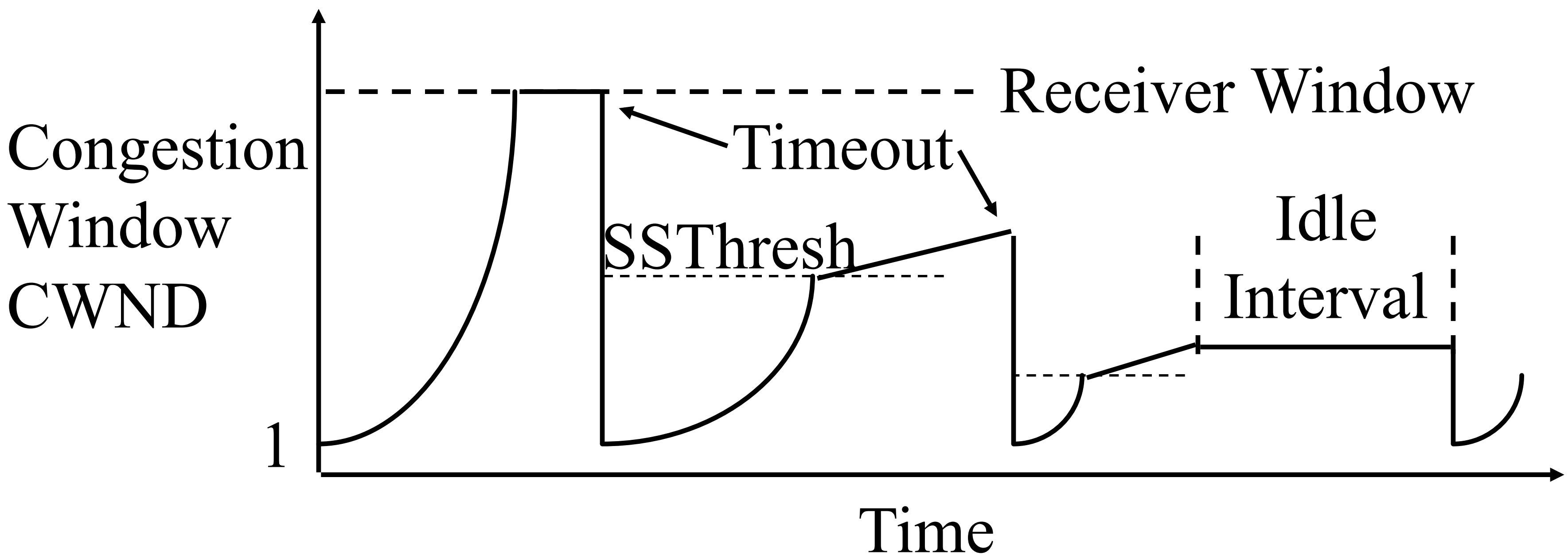
FAST RECOVERY RULES:

- ⇨ Retransmit lost segment
- ⇨ **Set cwnd = 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



# Idle periods

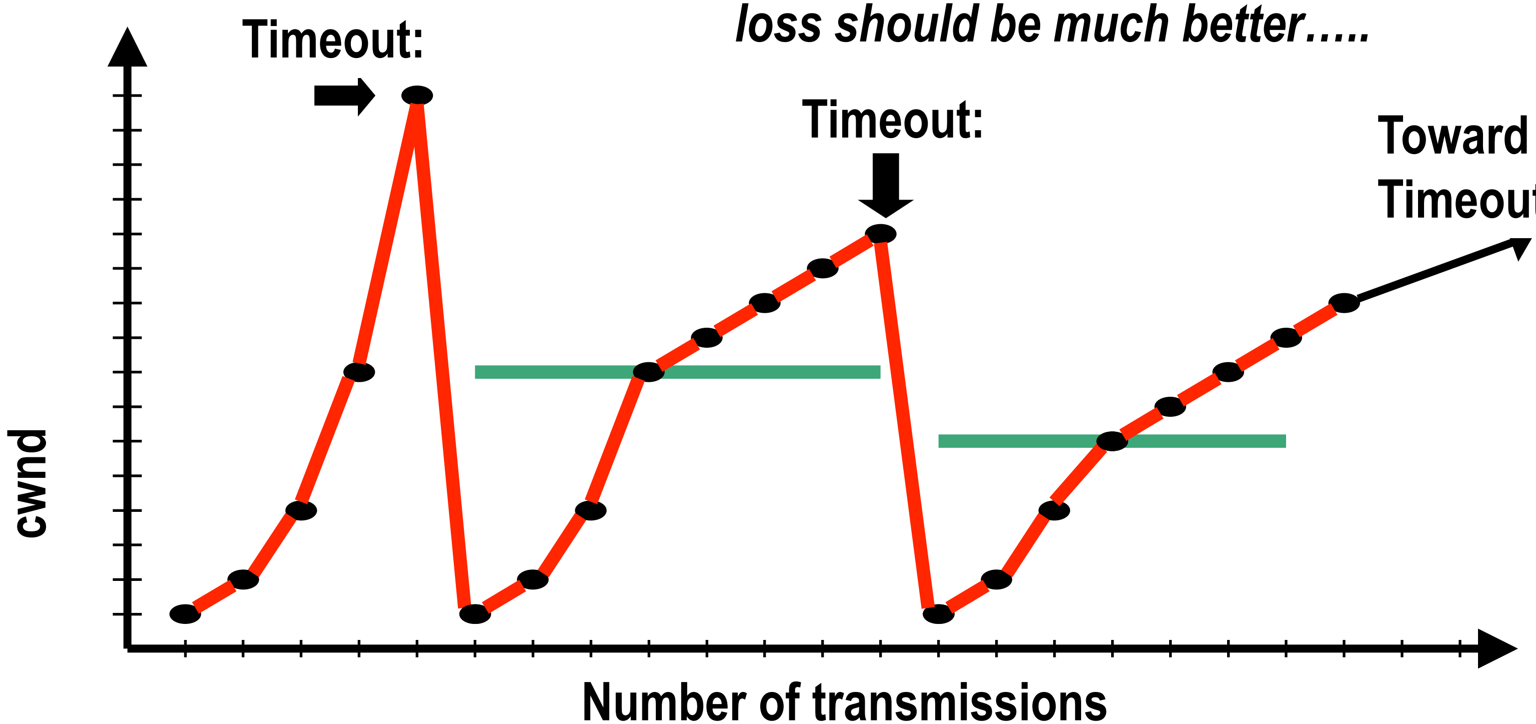
- After a long idle period (exceeding one RTO), reset the congestion window to one.



# Further TCP issues

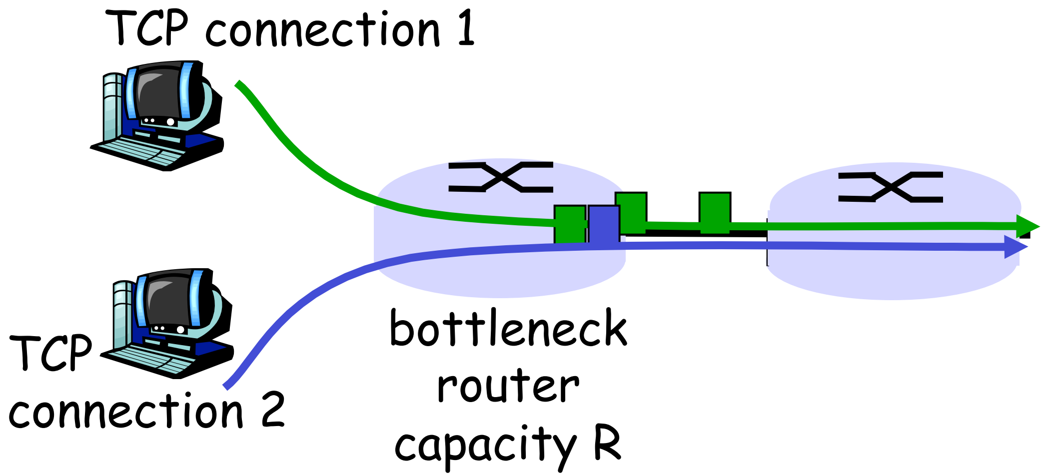
**Timeout = packet loss occurrence in an internal network router**
**TCP (both Tahoe & Reno) does not AVOID packet loss**
**Simply REACTS to packet loss**

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



# TCP Fairness

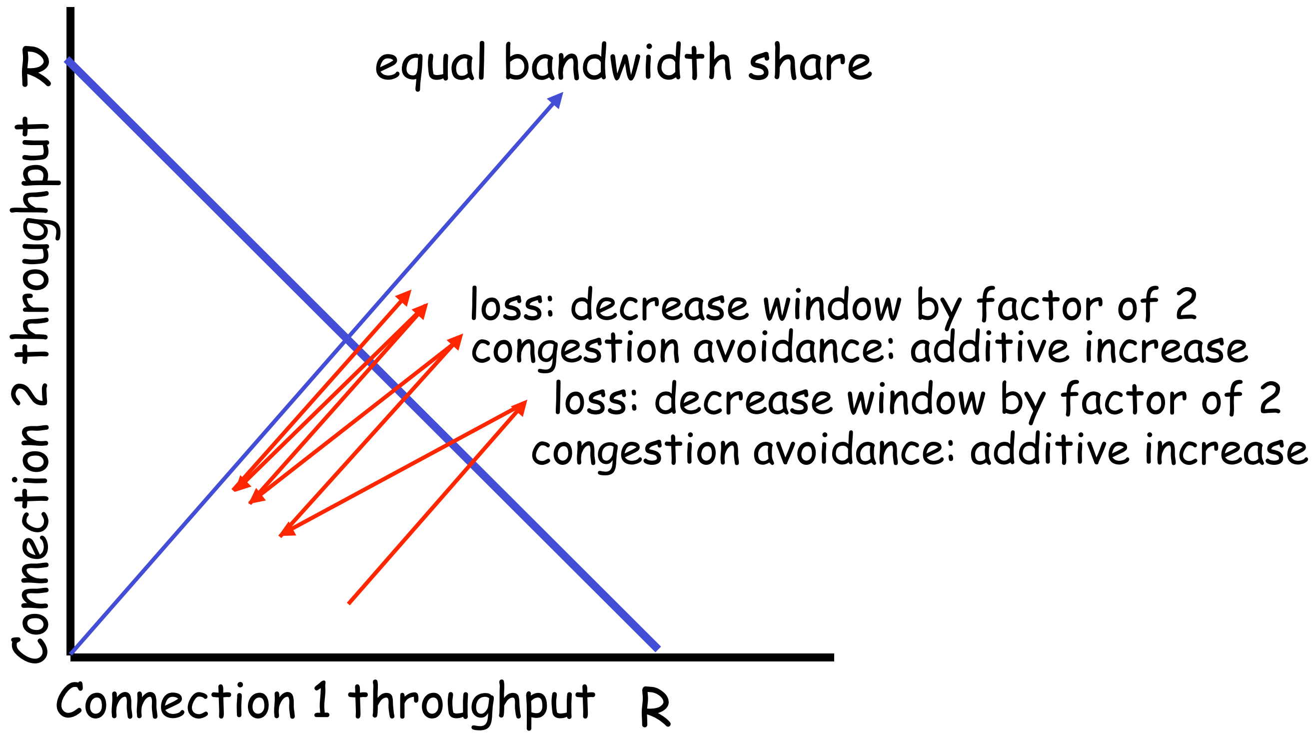
Fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



# Why is TCP fair?

Two competing sessions:

- Additive increase gives slope of 1, as throughout increases
- multiplicative decrease decreases throughput proportionally

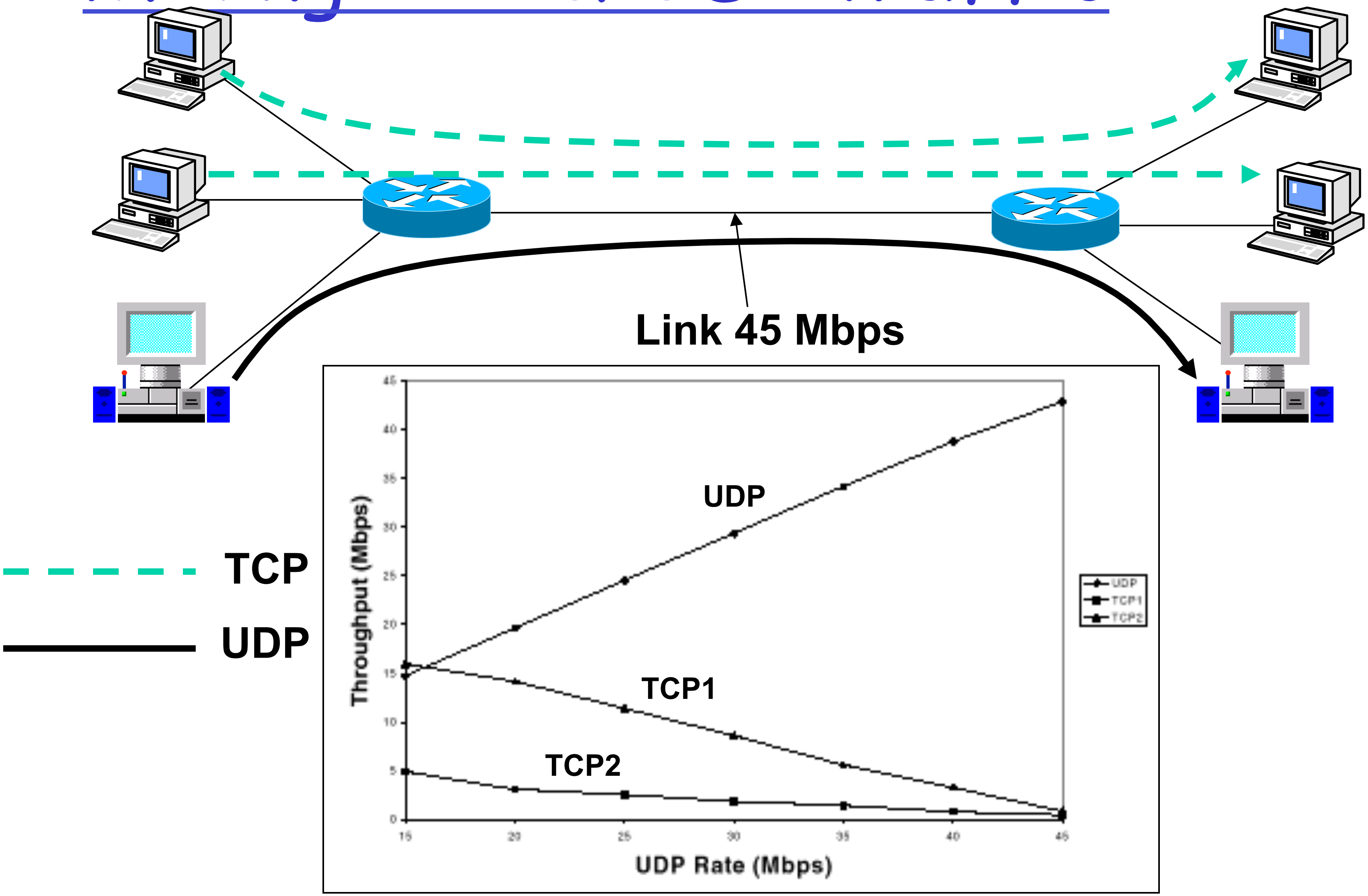


# Fairness with UDP traffic

- 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!

*This is why we still live in a World Wide Wait time (Webcams are destroying TCP traffic)*

# Mixing TCP & UDP traffic



# Fairness (more)

## Fairness and UDP

- Multimedia apps often do not use TCP
  - do not want rate throttled by congestion control
- Instead use UDP:
  - pump audio/video at constant rate, tolerate packet loss
- Research area: TCP friendly

## Fairness and parallel TCP connections

- nothing prevents app from opening parallel connections between 2 hosts.
- Web browsers do this
- Example: link of rate R supporting 9 cnctions;
  - new app asks for 1 TCP, gets rate R/10
  - new app asks for 11 TCPs, gets R/2 !