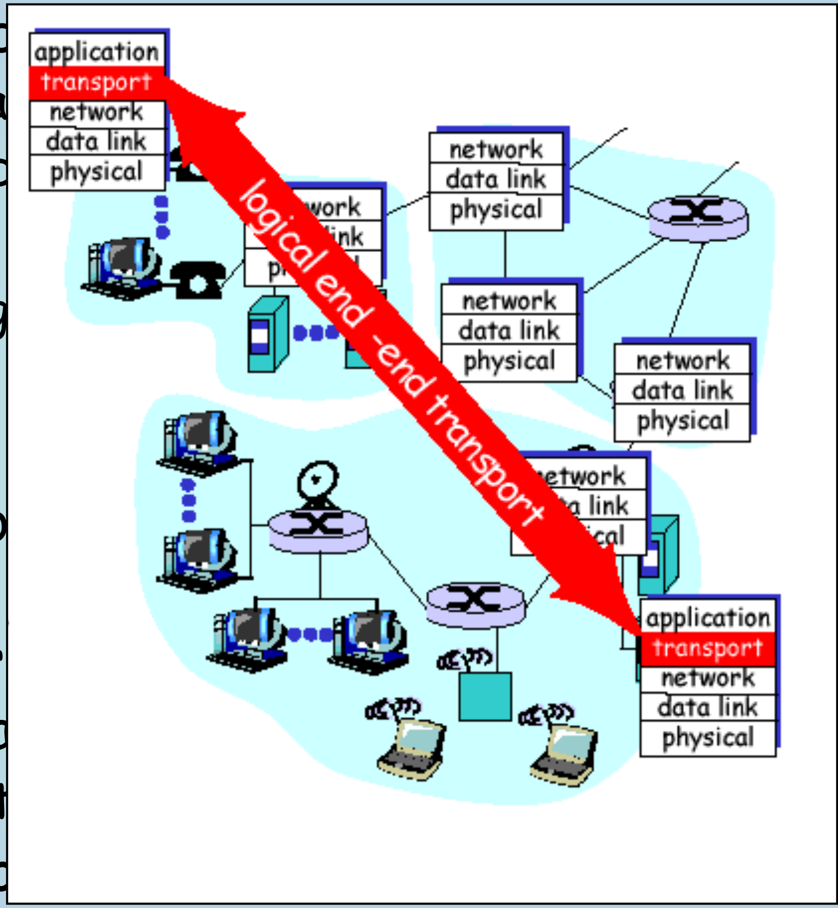
The dark side of TCP

understanding TCP on very high-speed networks

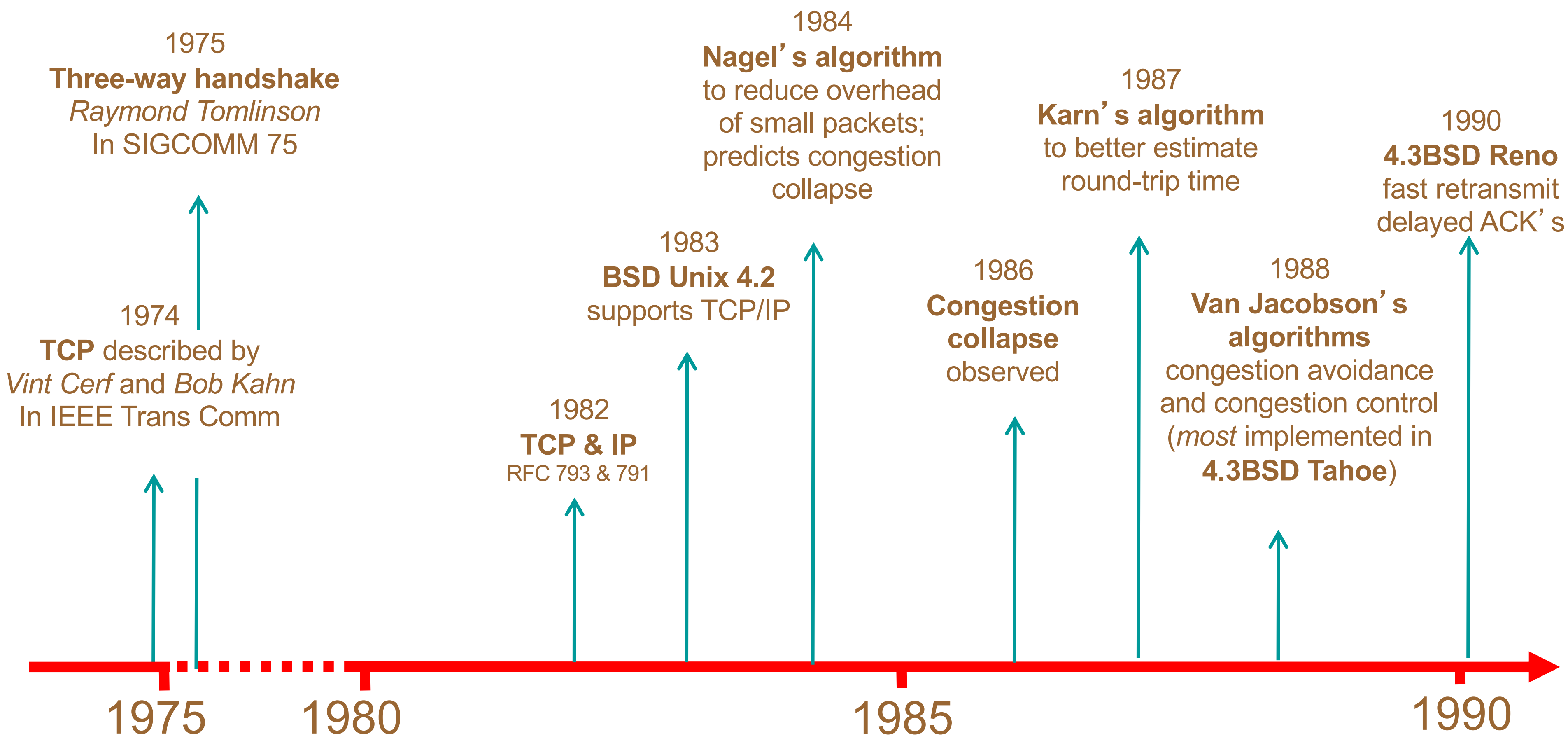
C. Pham
http://www.univ-pau.fr/~cpham
University of Pau, France
LIUPPA laboratory

What TCP brings

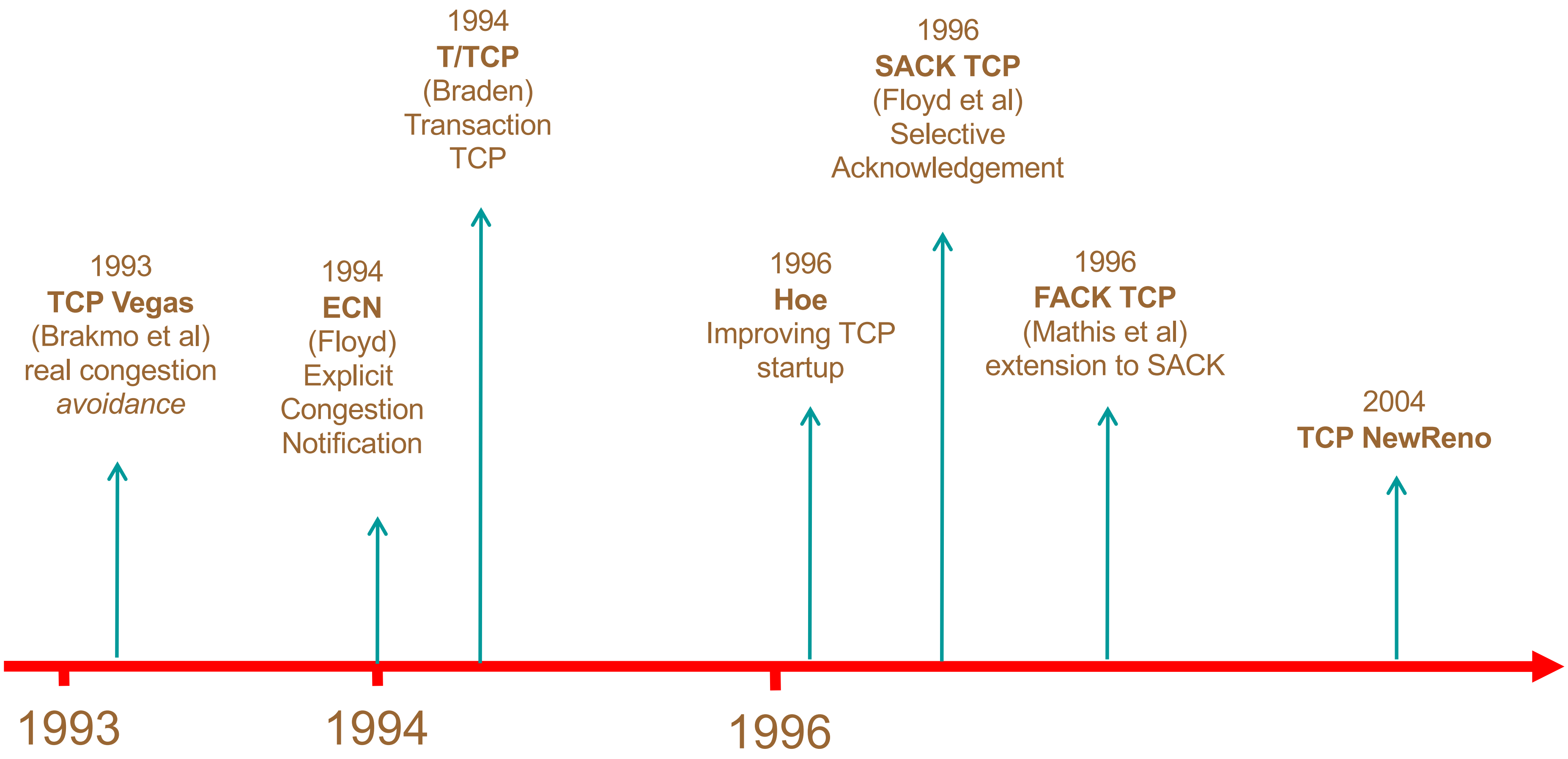
- stream-based
  - segments a … umbers
  - only conse…
- reliability
  - missing seg… g) and retransmitted
- flow control
  - receiver is … based)
- congestion co…
  - network is … based)
  - protocol tr…
- connection h…
  - explicit est…
- full-duplex c…
  - an ACK can be a data segment at the same time (piggybacking)



A brief history of TCP



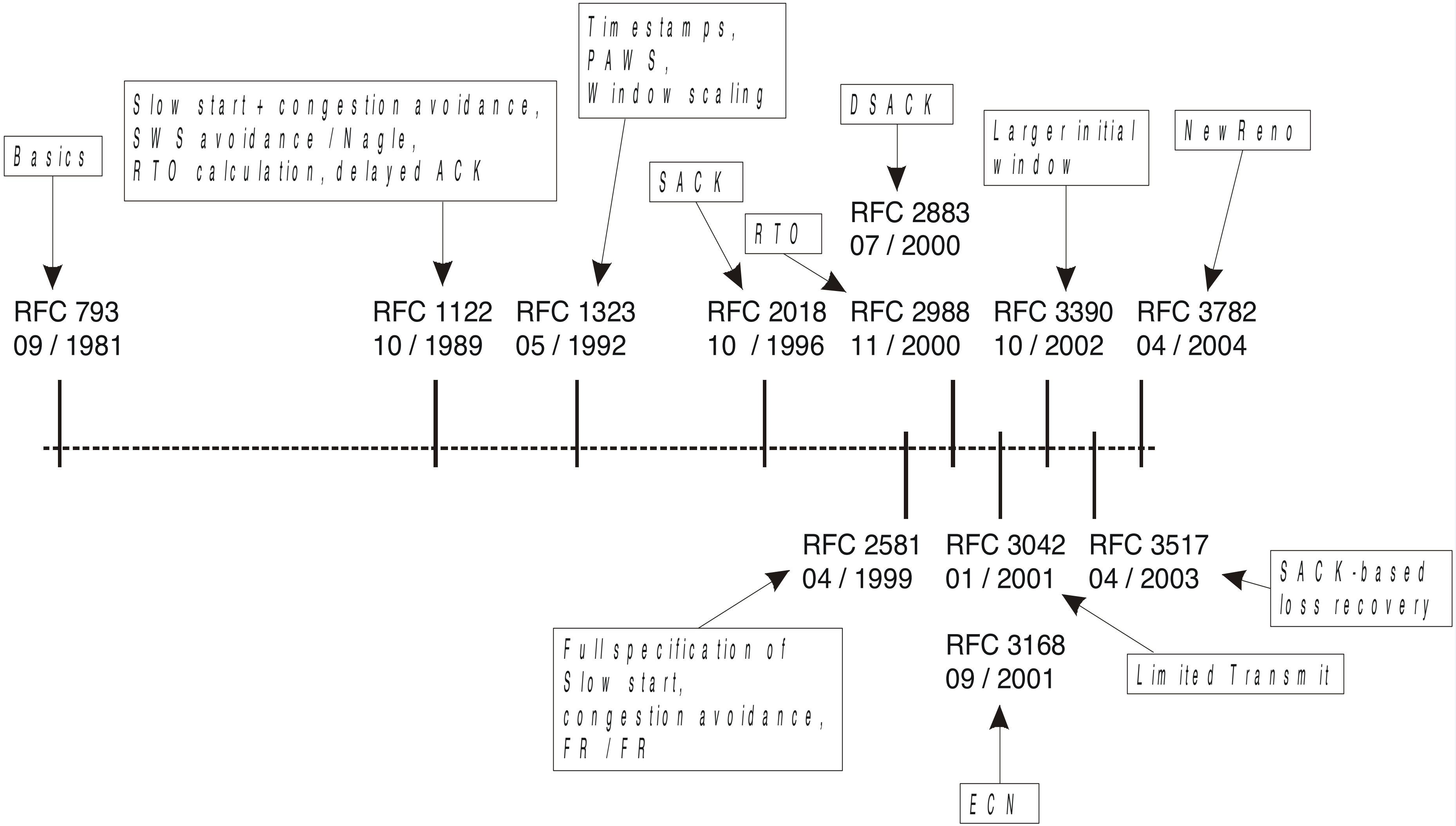
…in the nineties



From Mickeal Welzt

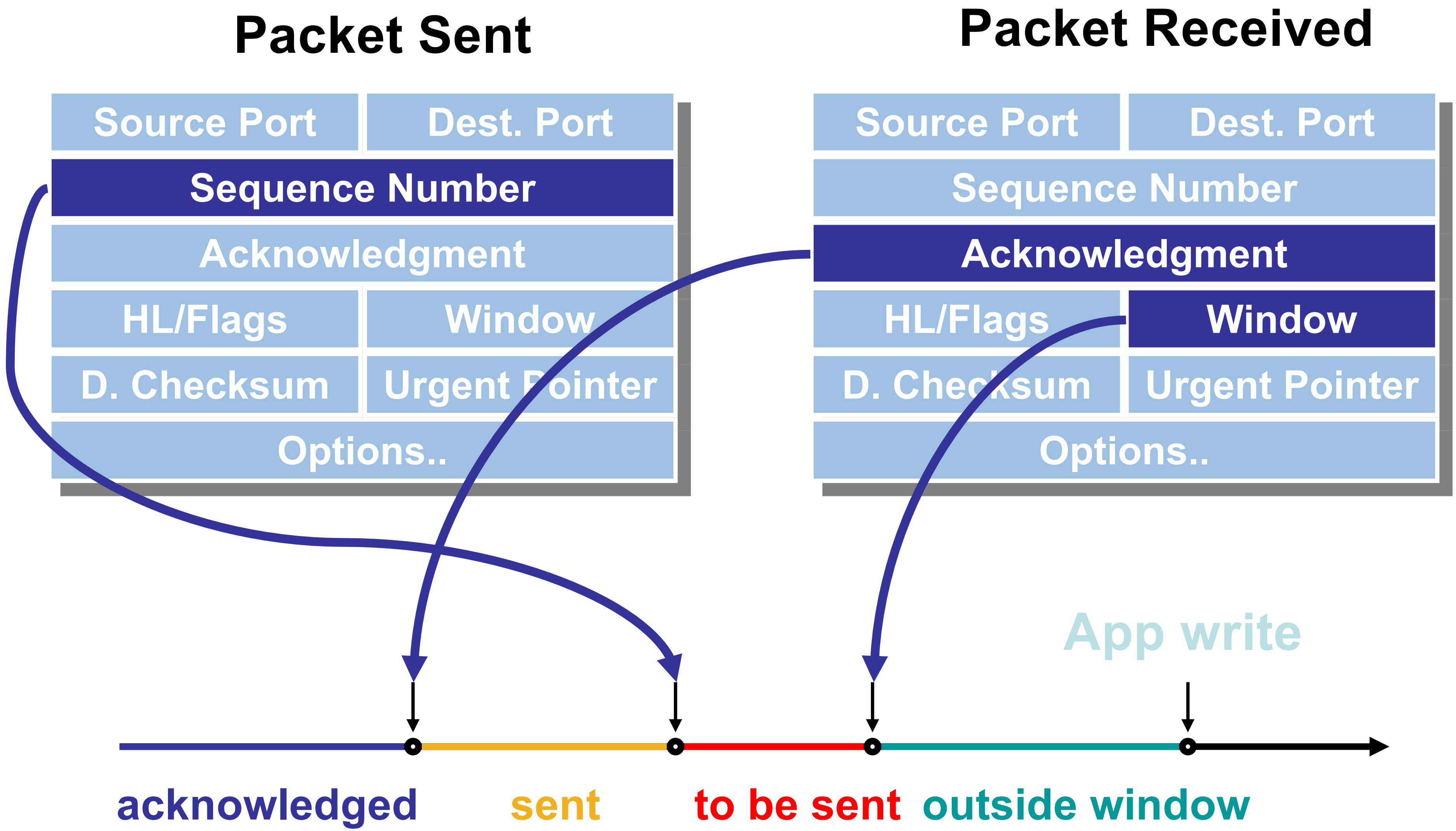
TCP History in RFC

Standards track TCP RFCs which influence when a packet is sent
(status: early 2005)

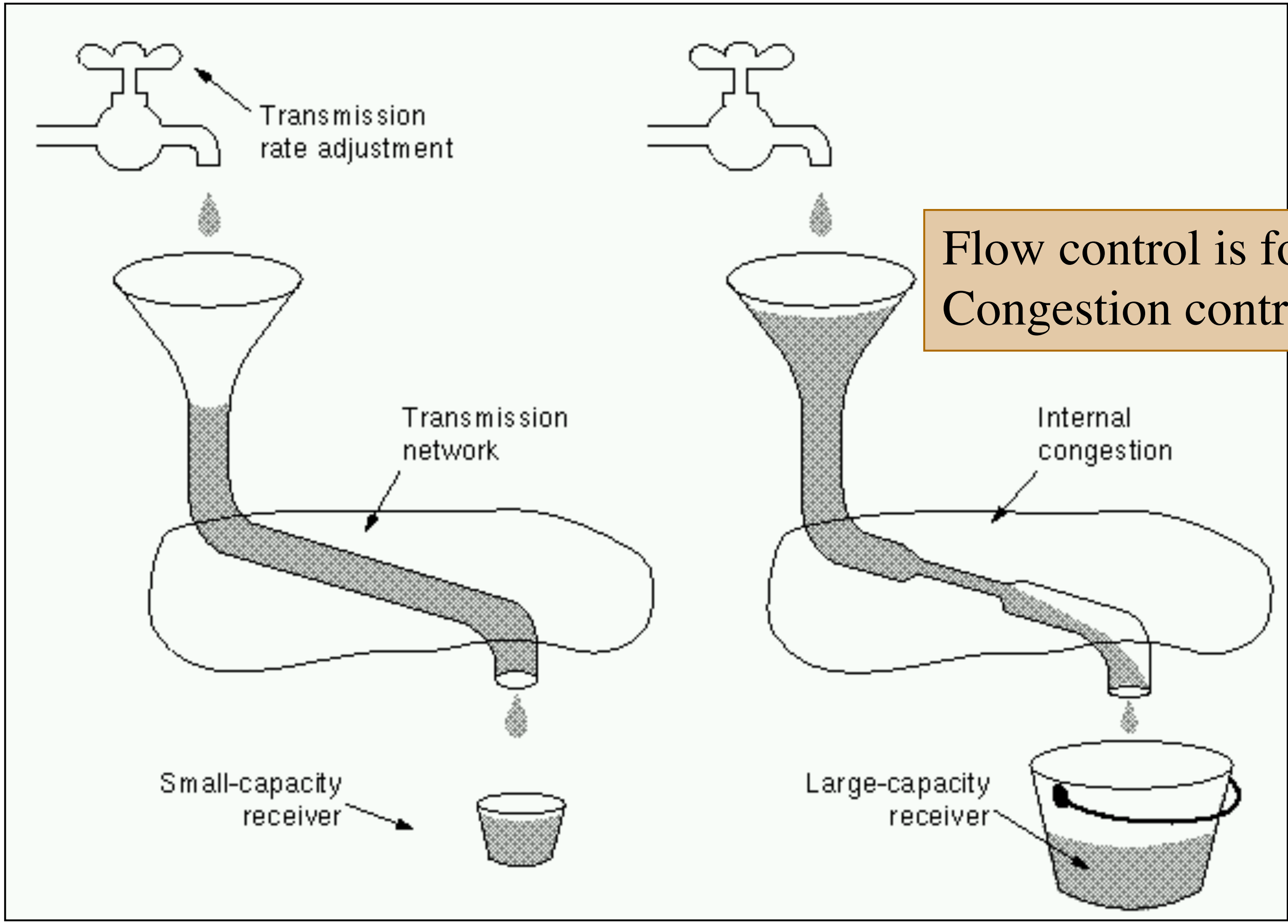


Flow control

prevents receiver's buffer overfow



Congestion control vs flow control



From Computer Networks, A. Tanenbaum

Flow control is for receivers
Congestion control is for the network

Congestion collapse was first observed in 1986 by V. Jacobson. Congestion control was added to TCP (TCP Reno) in 1988.

From Mickeal Welzt

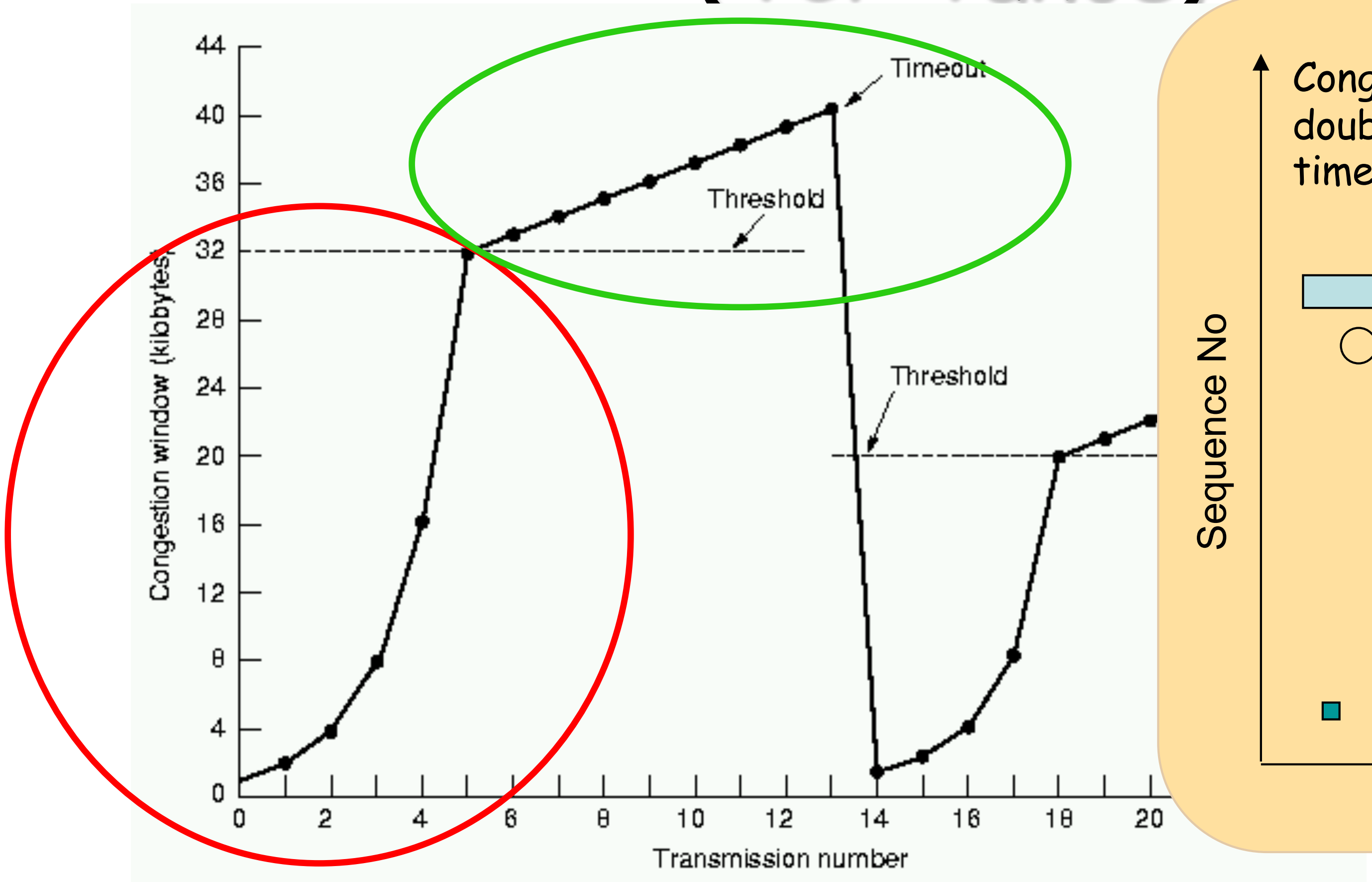
Internet congestion control: History

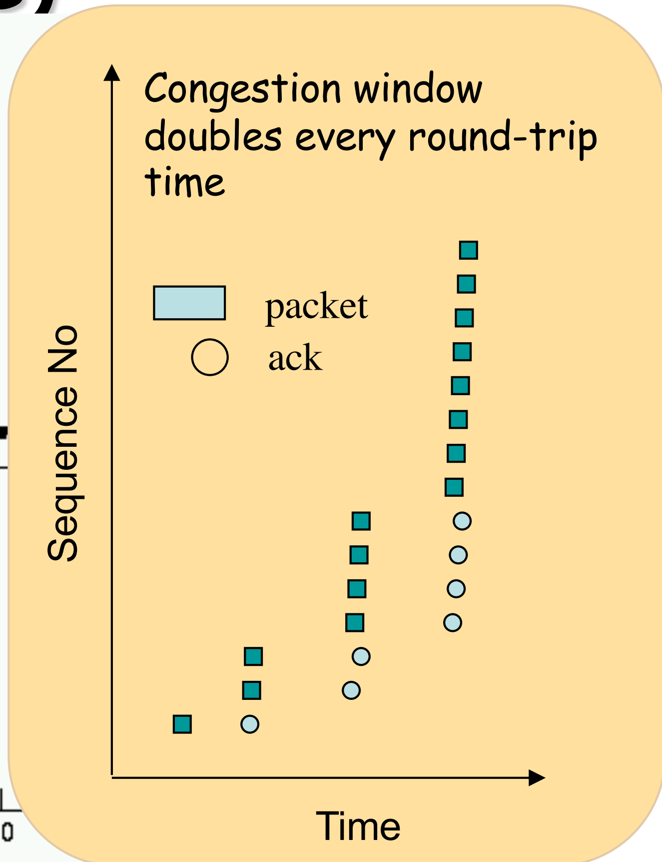
- **1968/69:** dawn of the Internet
- **1986:** first congestion collapse
- **1988:** "Congestion Avoidance and Control" (Jacobson)
  Combined congestion/flow control for TCP
  (also: variation change to RTO calculation algorithm)

- Goal: stability - in equilibrum, no packet is sent into the network until an old packet leaves
  - ack clocking, “conservation of packets“ principle
  - made possible through window based stop+go - behaviour

- Superposition of stable systems = stable → network based on TCP with congestion control = stable

TCP congestion control: the big picture (TCP Tahoe)





- cwnd grows exponentially (slow start), then linearly (*congestion avoidance*) with 1 more segment per RTT
- If loss, divides threshold by 2 (multiplicative decrease) and restart with cwnd=1 packet

From Mickeal Welzt

Fast Retransmit / Fast Recovery (Reno)

Reasoning: slow start = restart; assume that network is empty
But even similar incoming ACKs indicate that packets arrive at the receiver!
Thus, slow start reaction = too conservative.

1. Upon reception of third duplicate ACK (DupACK): ssthresh = FlightSize/2

2. Retransmit lost segment (fast retransmit);
   cwnd = ssthresh + 3*SMSS
   ("inflates" cwnd by the number of segments (three) that have left the network and which the receiver has buffered)

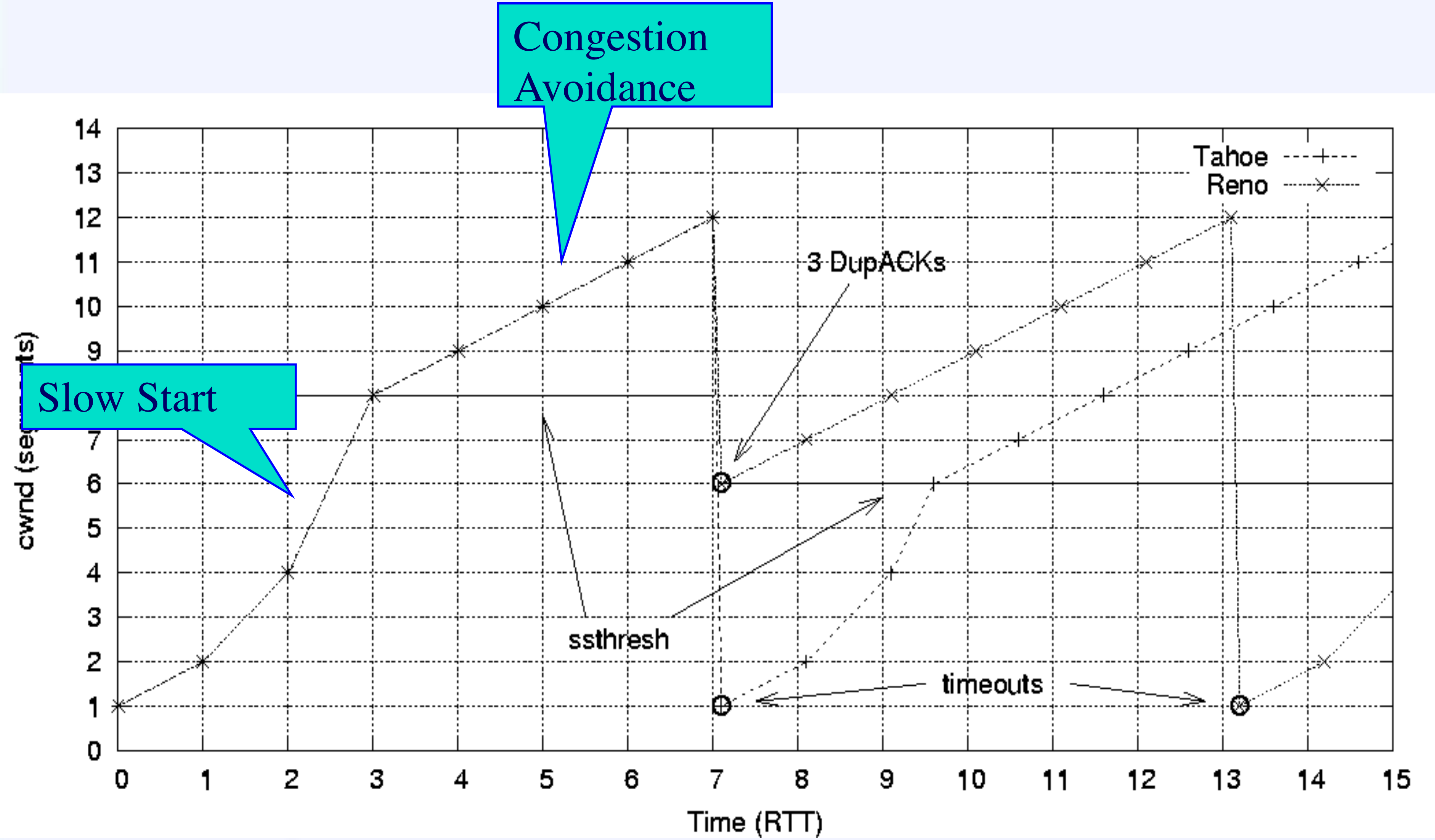
3. For each additional DupACK received: cwnd += SMSS
   (inflates cwnd to reflect the additional segment that has left the network)

4. Transmit a segment, if allowed by the new value of cwnd and rwnd

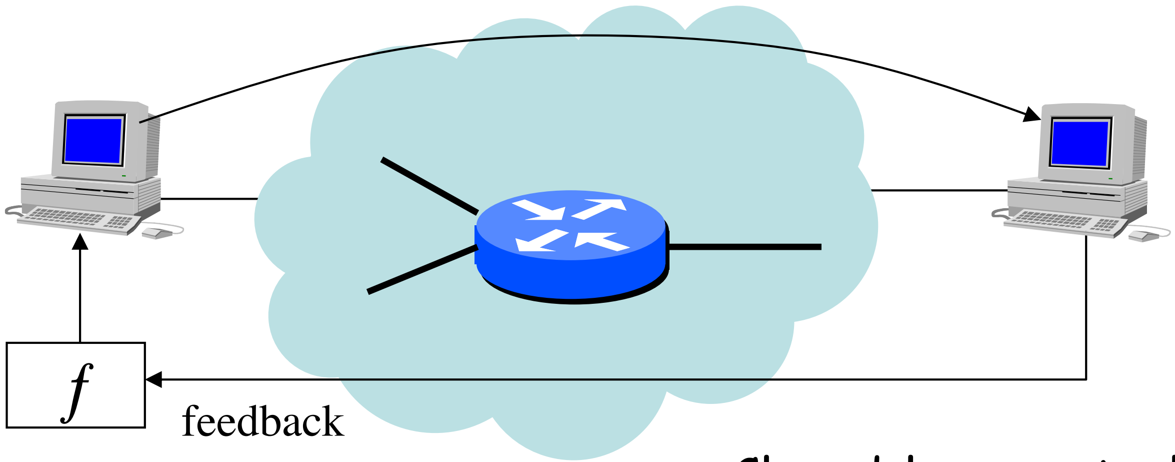
5. Upon reception of ACK that acknowledges new data (“full ACK“):
   "deflate" window: cwnd = ssthresh (the value set in step 1)

From Mickeal Welzt

Tahoe vs. Reno



From the control theory point of view



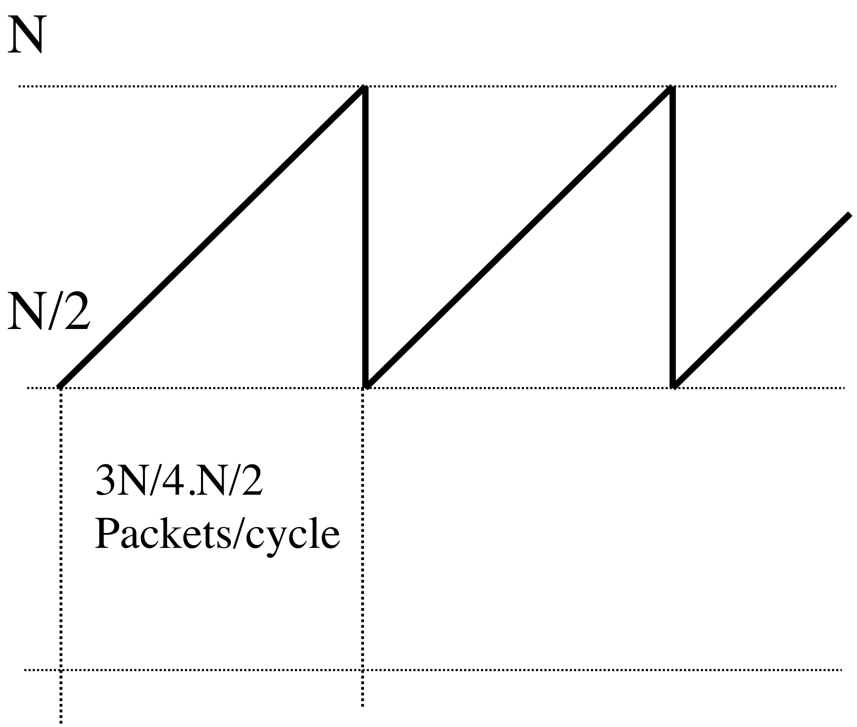
Closed-loop control

- Feedback should be frequent, but not too much otherwise there will be oscillations
- Can not control the behavior with a time granularity less than the feedback period

The TCP saw-tooth curve

TCP behavior in steady state

Isolated packet losses trigger the fast recovery procedure instead of the slow-start.

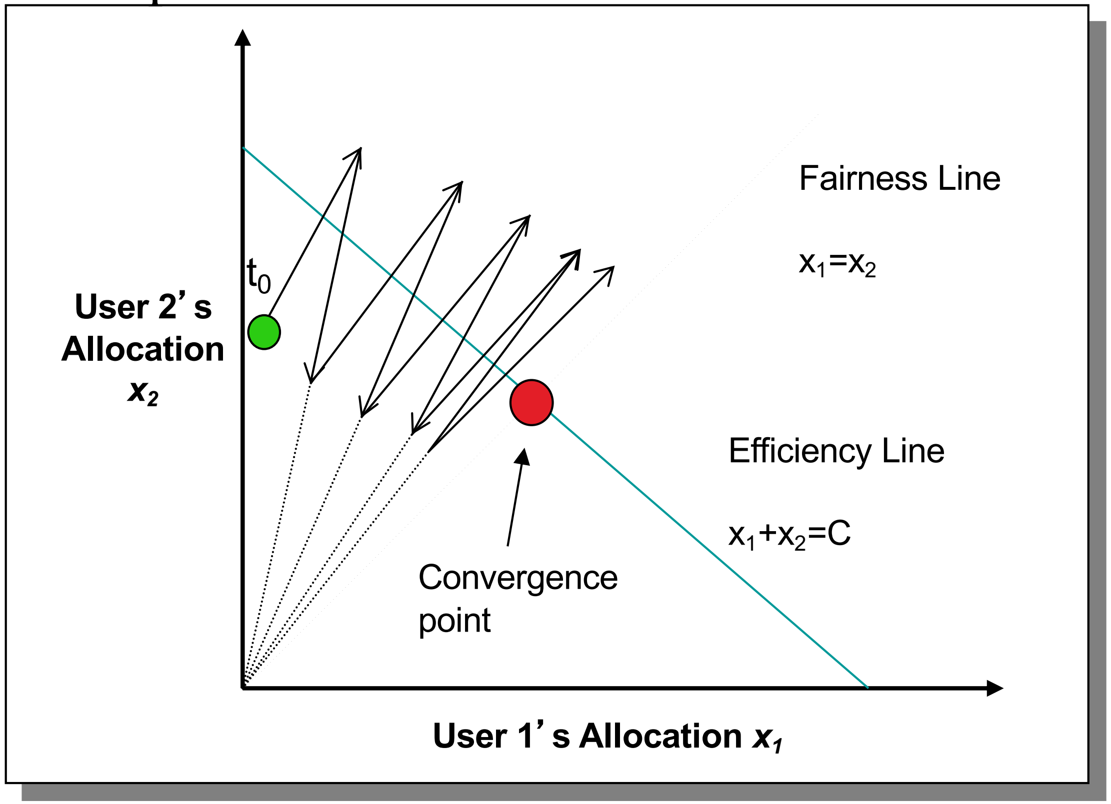


- The TCP steady-state behavior is referred to as the Additive Increase-Multiplicative Decrease process

no loss:
cwnd = cwnd + 1
loss:
cwnd = cwnd*0.5

AIMD

Phase plot

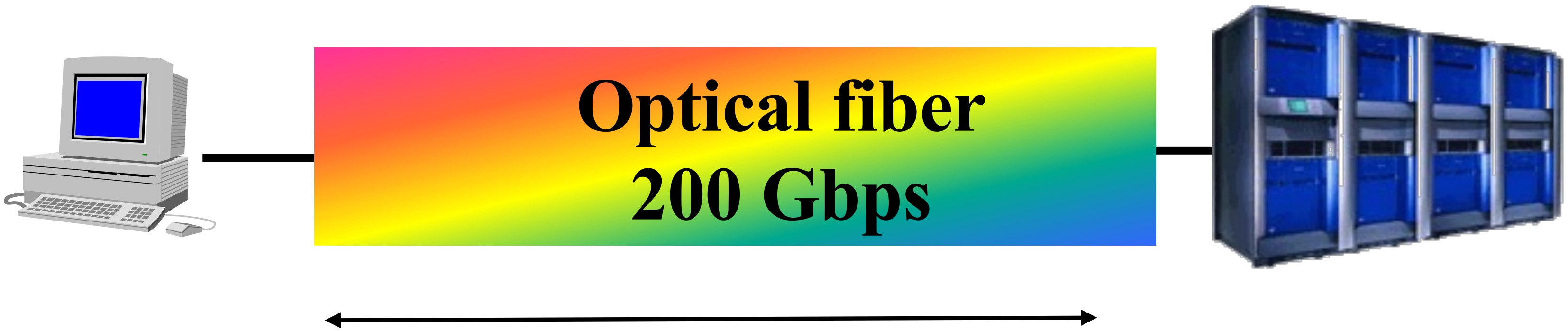


Fairness is preserved under Multiplicative Decrease since the user's allocation ratio remains the same

Ex: $$\frac{x_2}{x_1} = \frac{x_2 b}{x_1 b}$$

- Assumption: decrease policy must (at minimum) reverse the load increase over-and-above efficiency line
- Implication: decrease factor should be conservatively set to account for any congestion detection lags etc

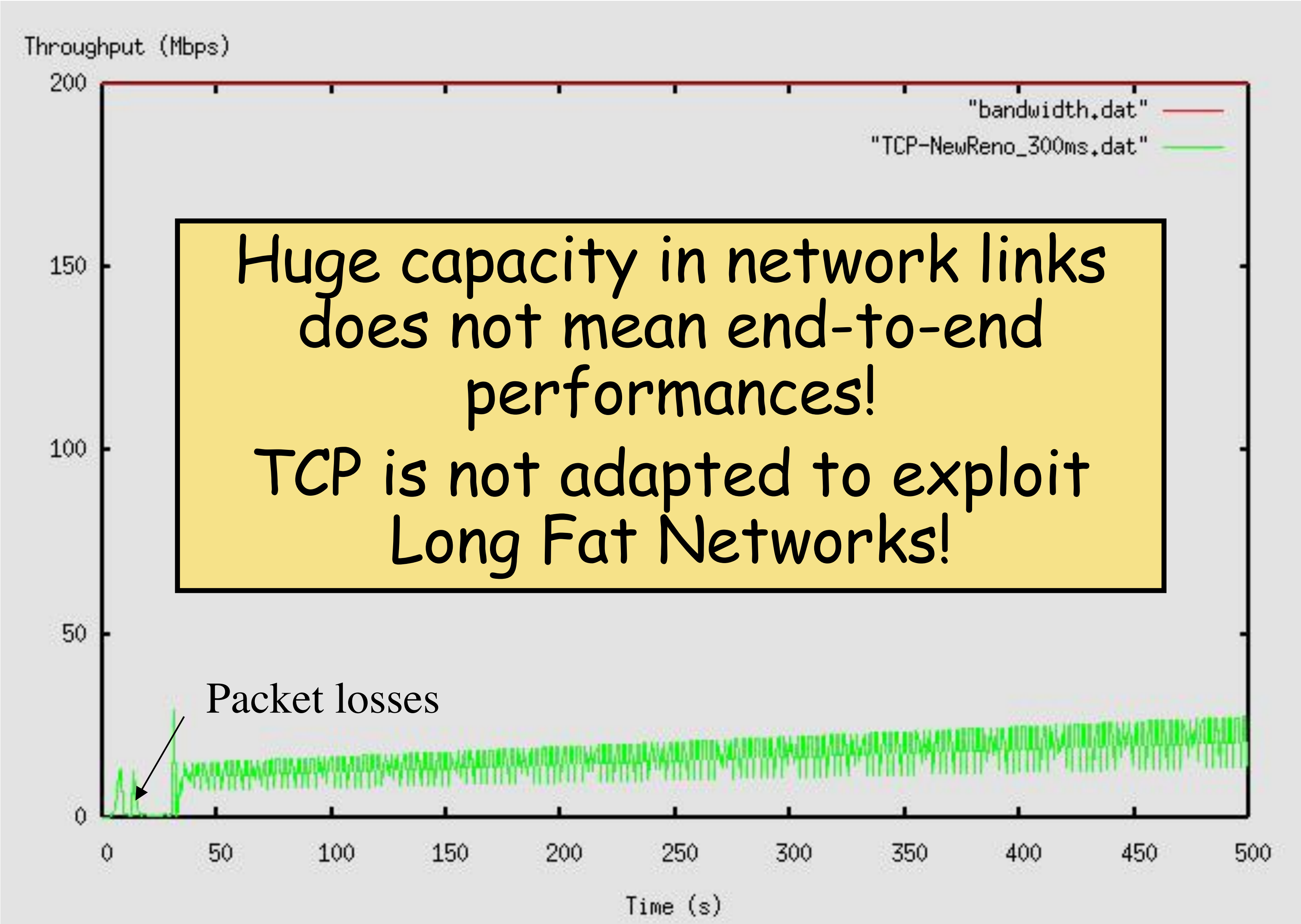
Very High-Speed Networks



200000km/s, delay of 5ms every 1000km

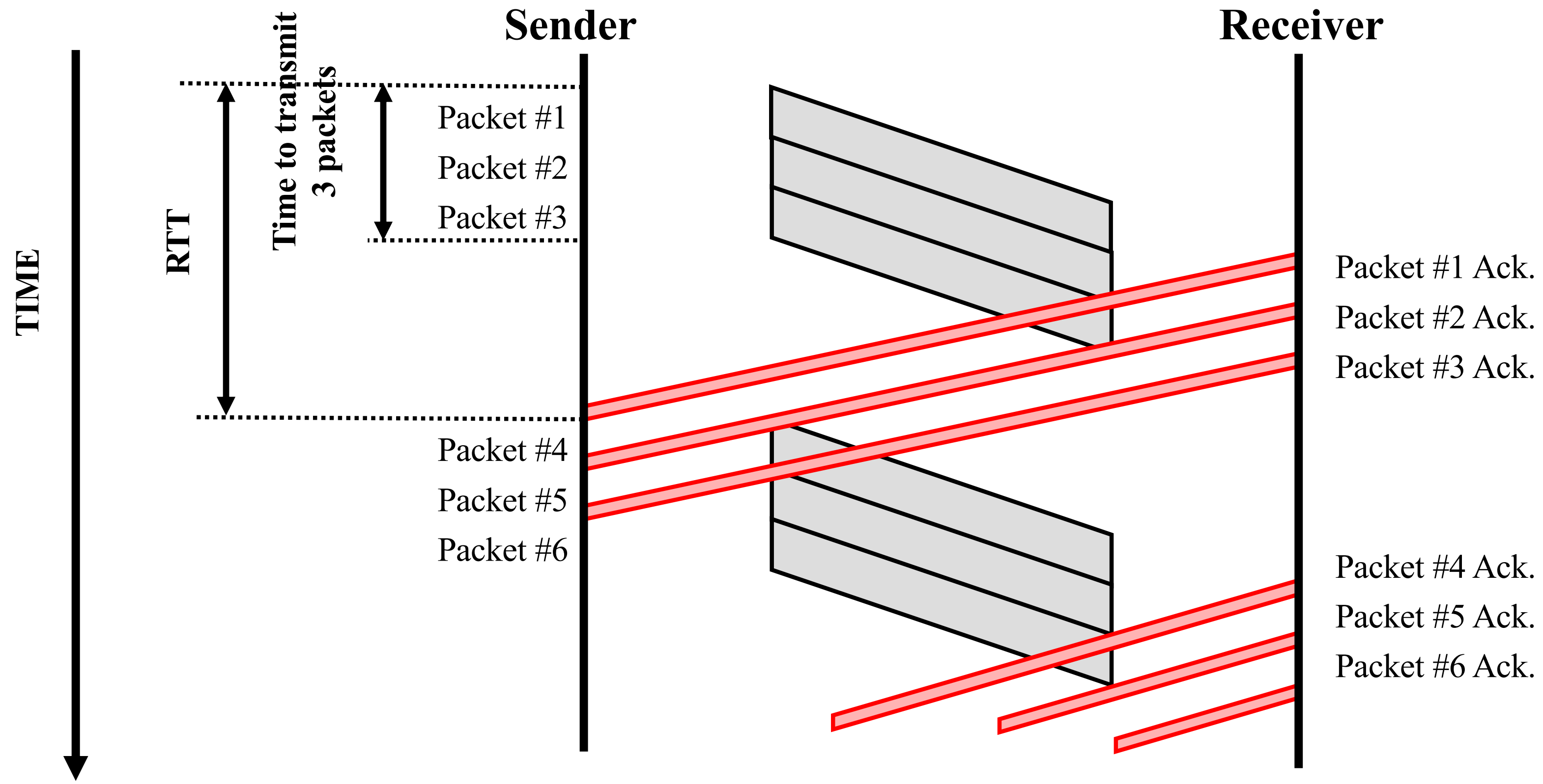
- Today's backbone links are optical, DWDM-based, and offer gigabit rates
- Transmission time <<< propagation time
- Duplicating a 10GB database should not be a problem anymore

The reality check: TCP on a 200Mbps link



First problem: window size

- The default maximum window size is 64Kbytes. Then the sender has to wait for acks.

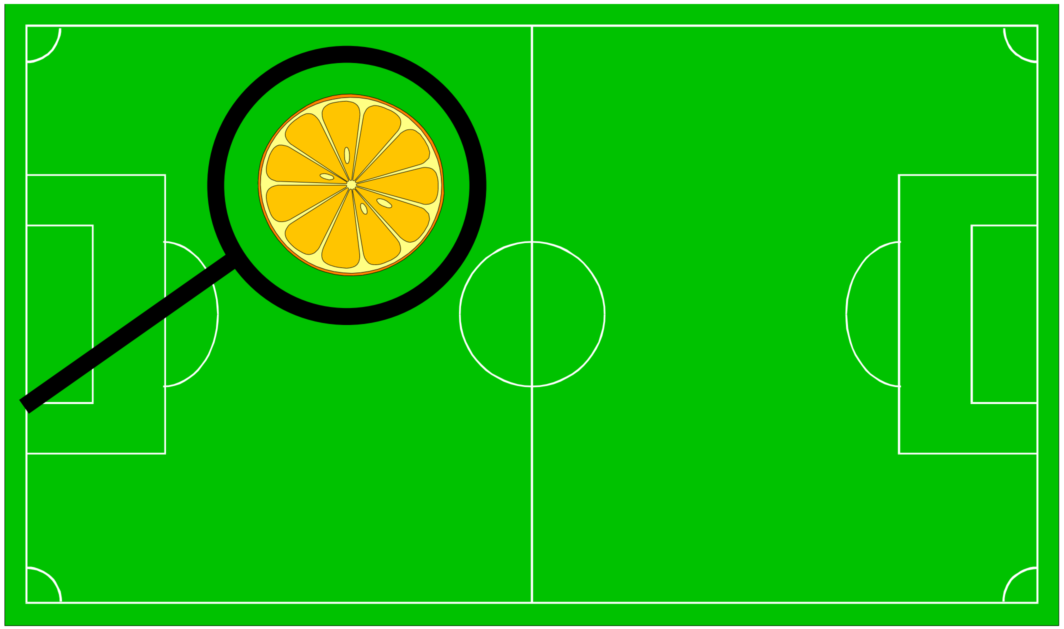


First problem: window size

- The default maximum window size is 64Kbytes. Then the sender has to wait for acks.

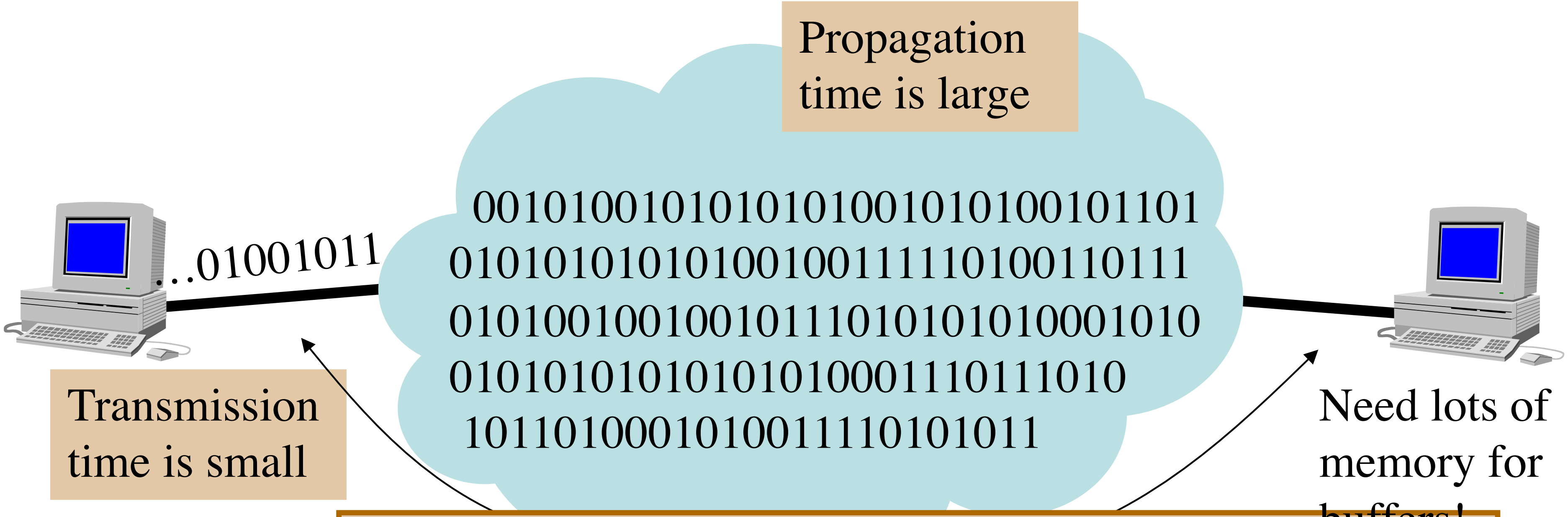
RTT=200ms Link is 0C-48 = 2.5 Gbps

Waiting time



Rule of thumb on Long Fat Networks

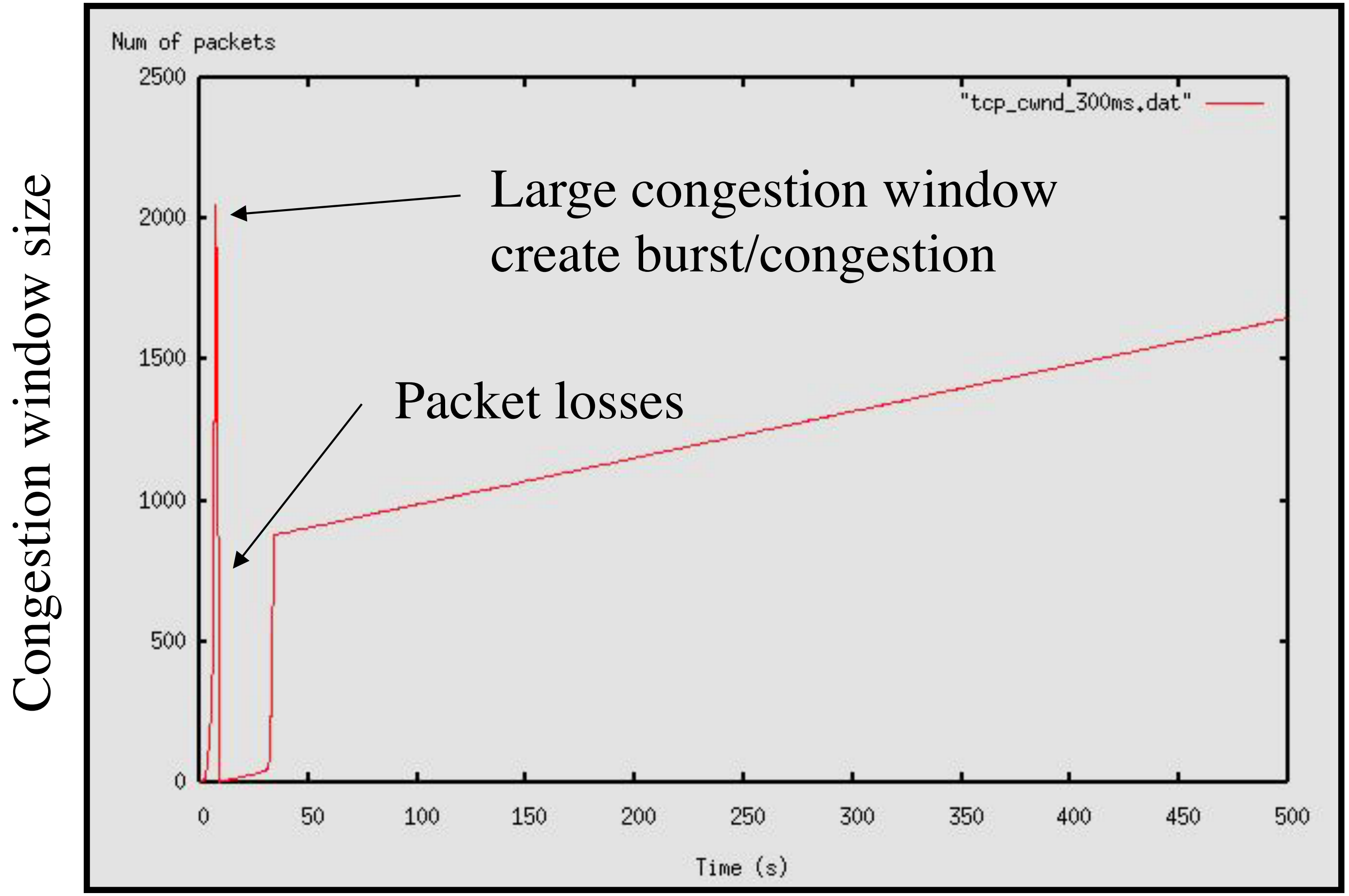
- High-~~speed~~ capacity network



The optimal window size should be set to the bandwidthxRTT product to avoid blocking at the sender side

Side effect of large windows

TCP becomes very sensitive to packet losses on LFN

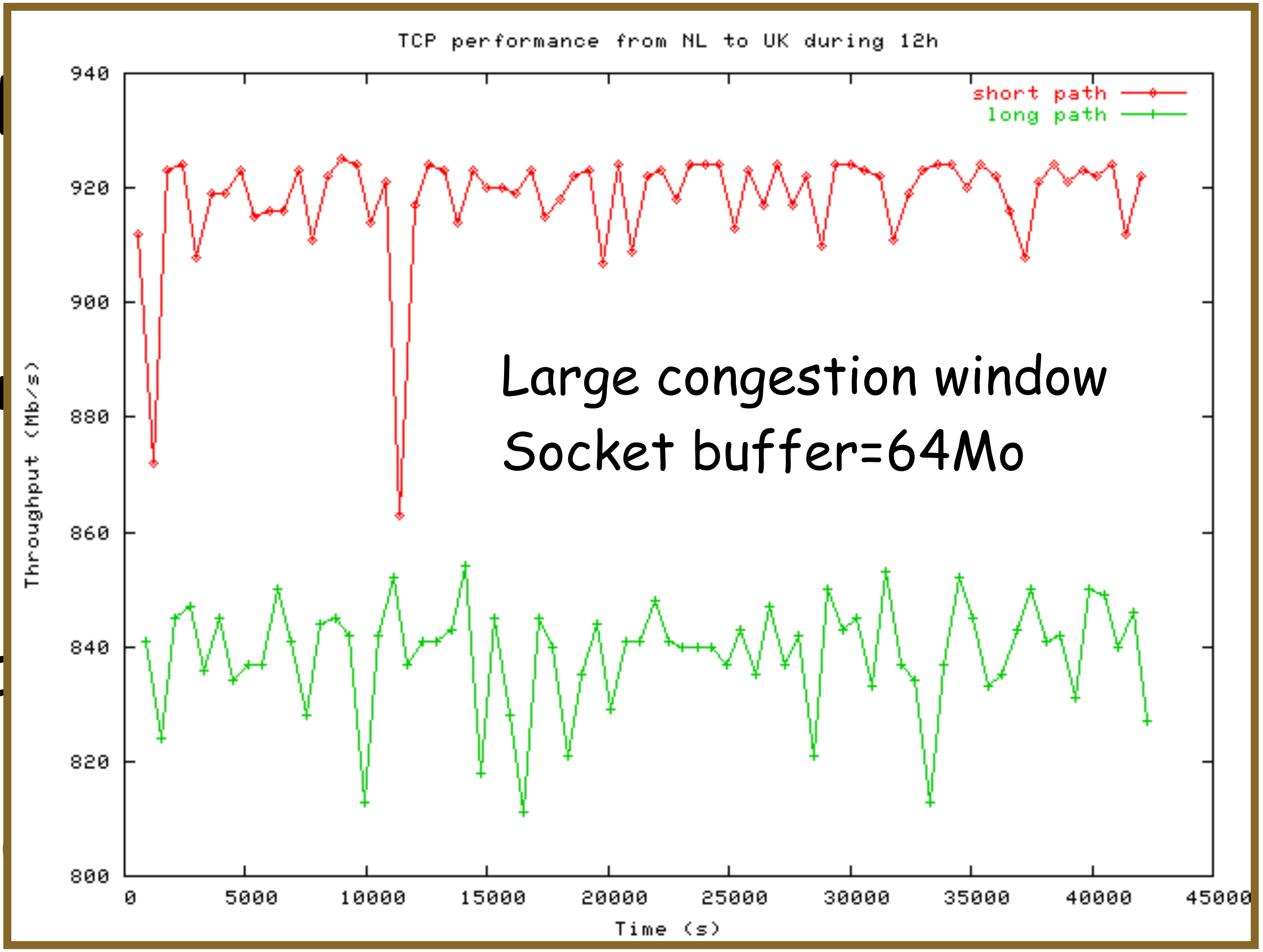


Pushing the limits of TCP

- Standard configuration (vanilla TCP) is not adequate on many OS, everything is under-sized
  - Receiver buffer
  - System buffer
  - Default block size
- Will manage to get between 1Gbps and 2Gbps if well-tuned AND if the RTT is small enough!

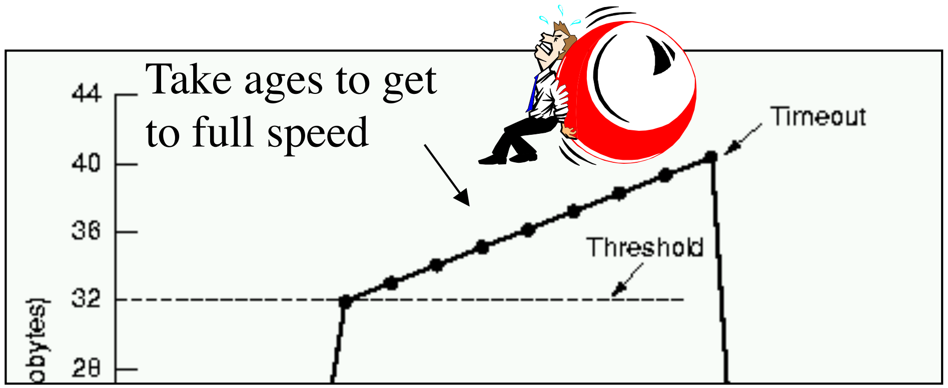
Pushing the limits of TCP

- Standard… adequate… sized
  - Receive…
  - System…
  - Default…
- Will man… 2Gbps if… small eno…



Source: M. Goutelle, GEANT test campaign

Problem on high capacity link? Additive increase is still too slow!



With 100ms of round trip time, a connection needs 203 minutes (3h23) to send at 10Gbps starting from 1Mbps!

Once you get high throughput, maintaining it is difficult too!

- Sustaining high congestion windows:

A Standard TCP connection with:

- 1500-byte packets;
- a 100 ms round-trip time;
- a steady-state throughput of 10 Gbps;

would require:

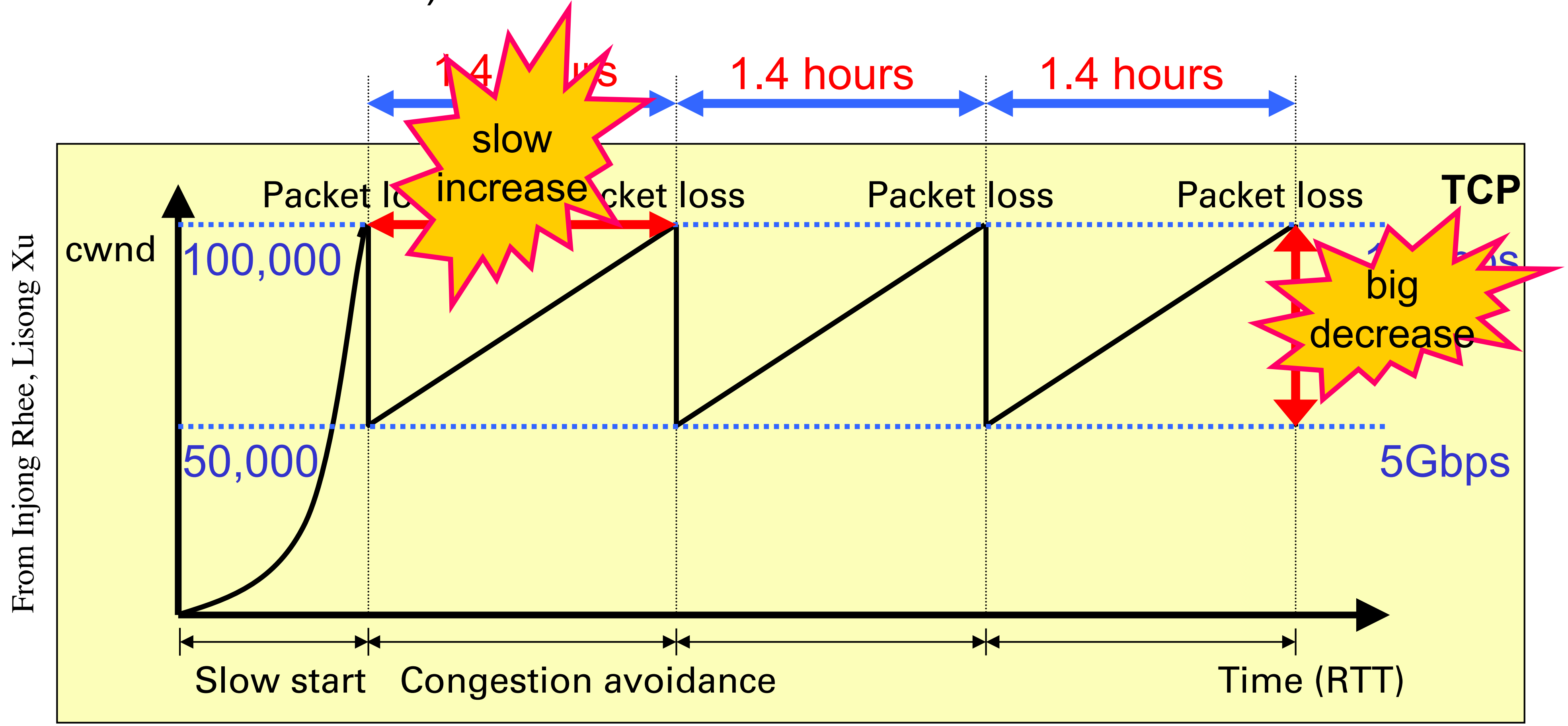
- an average congestion window of 83,333 segments;
- and at most one drop (or mark) every 5,000,000,000 packets (or equivalently, at most one drop every 1 2/3 hours).

This is not realistic.

From S. Floyd

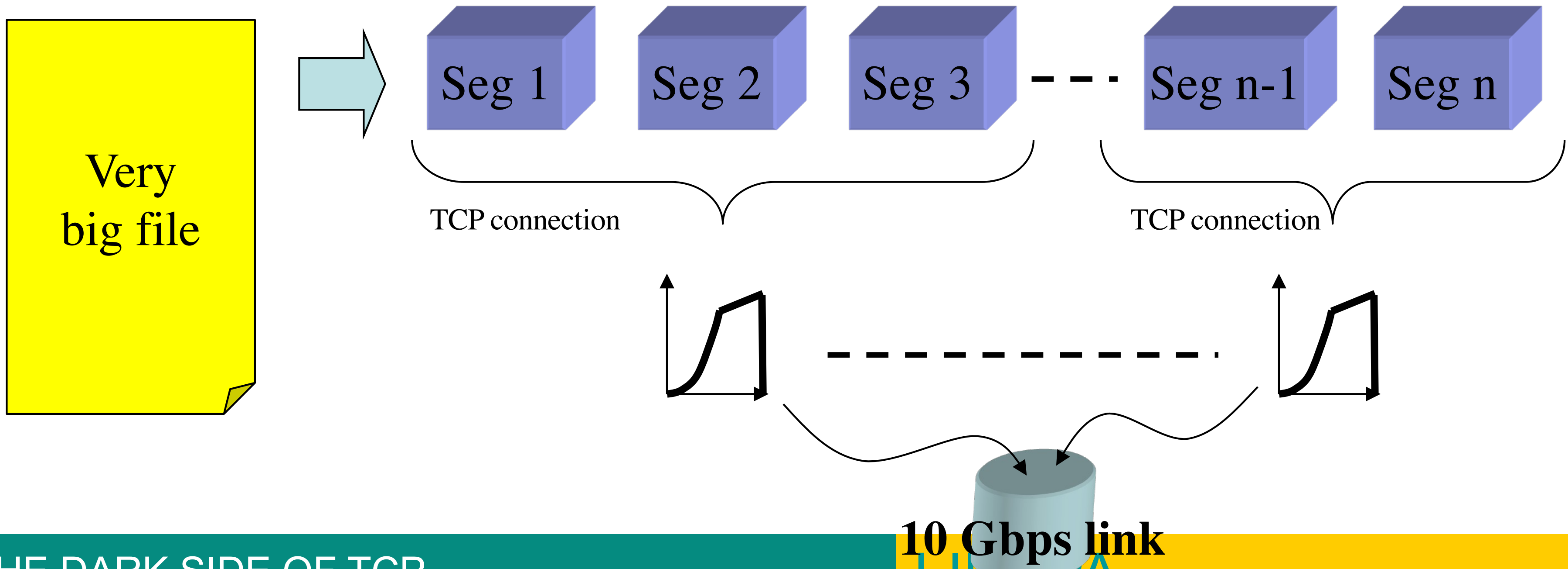
TCP rules: slow increase, big decrease

A TCP connection with 1250-Byte packet size and 100ms RTT is running over a 10Gbps link (assuming no other connections, and no buffers at routers)



Going faster (cheating?)
n flows is better than 1

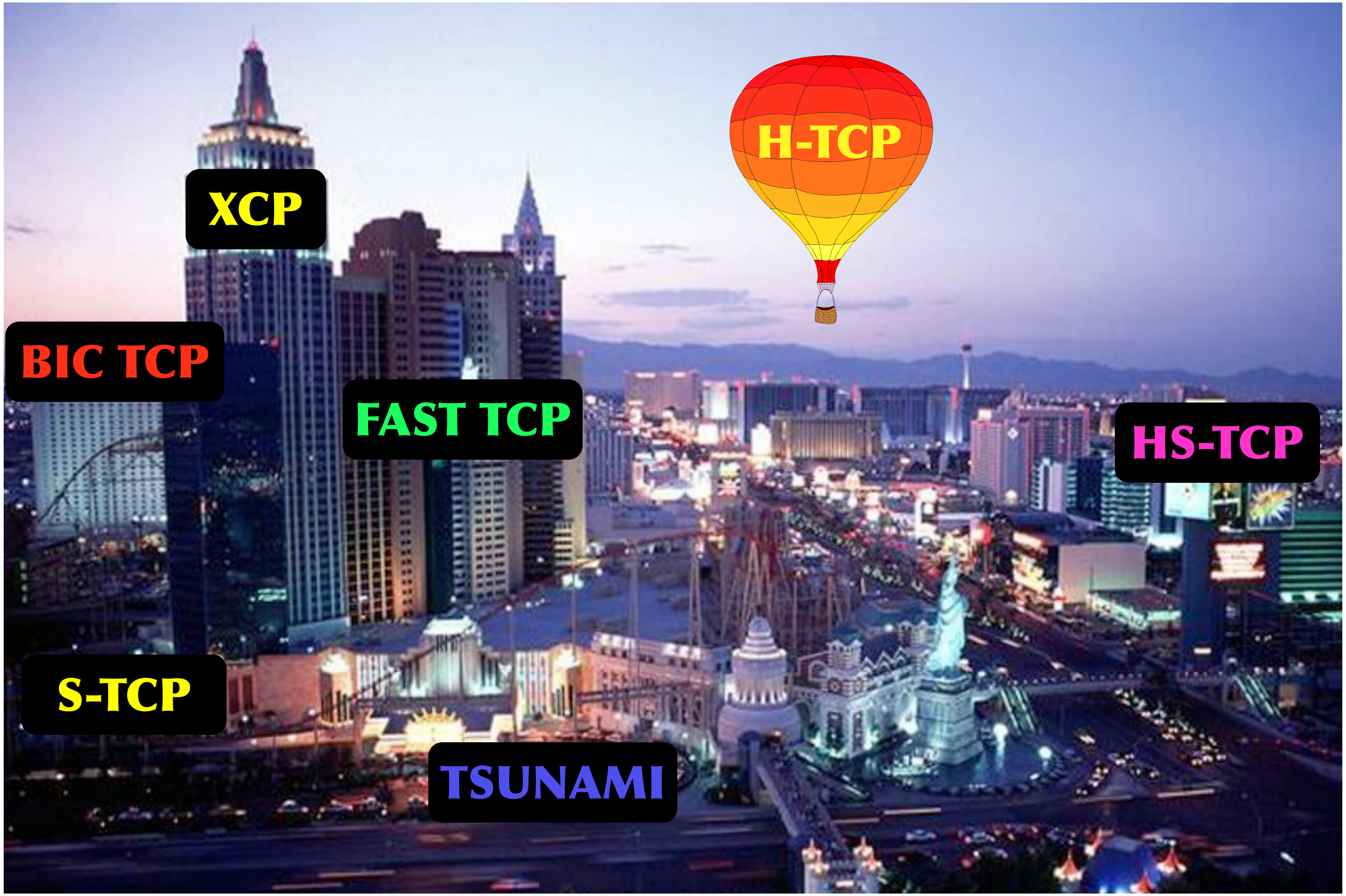
- The CC limits the throughput of a TCP connection: so why not use more than 1 connection for the same file?



New transport protocols

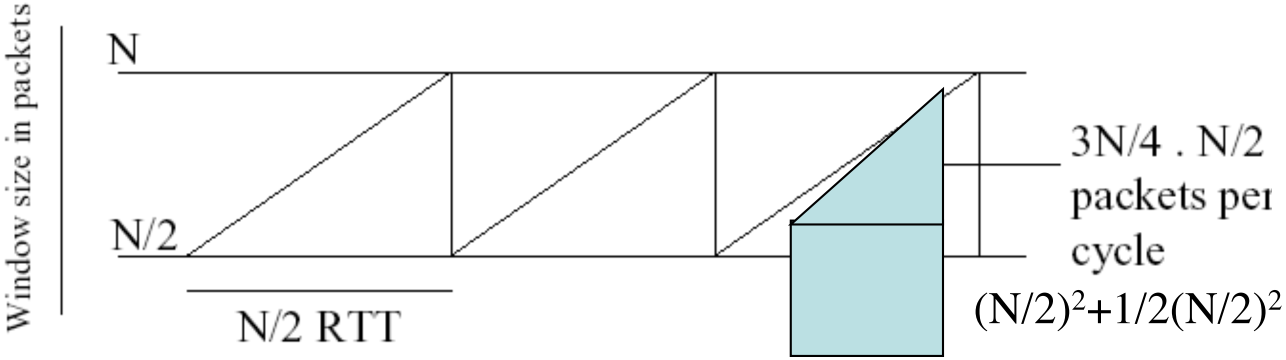
- New transport protocols are those that are not only optimizations of TCP
- New behaviors, new rules, new requirements! Everything is possible!
- New protocols are then not necessarily TCP compatible!

The new transport protocol strip



Response function

- Throughput = f(p, RTT)
- TCP's response function



Average window size (in packets) = W = 3N/4 , from (N+N/2)/2

Number of packets per cycle = 3N/4 . N/2 = $3N^2/8$ = 1/ p

- Where p is the packet loss ratio (which should remain small enough)
- So $N = \sqrt{8/3p}$

Average throughput (in packets/sec) = B = W / RTT = 3N / 4 RTT

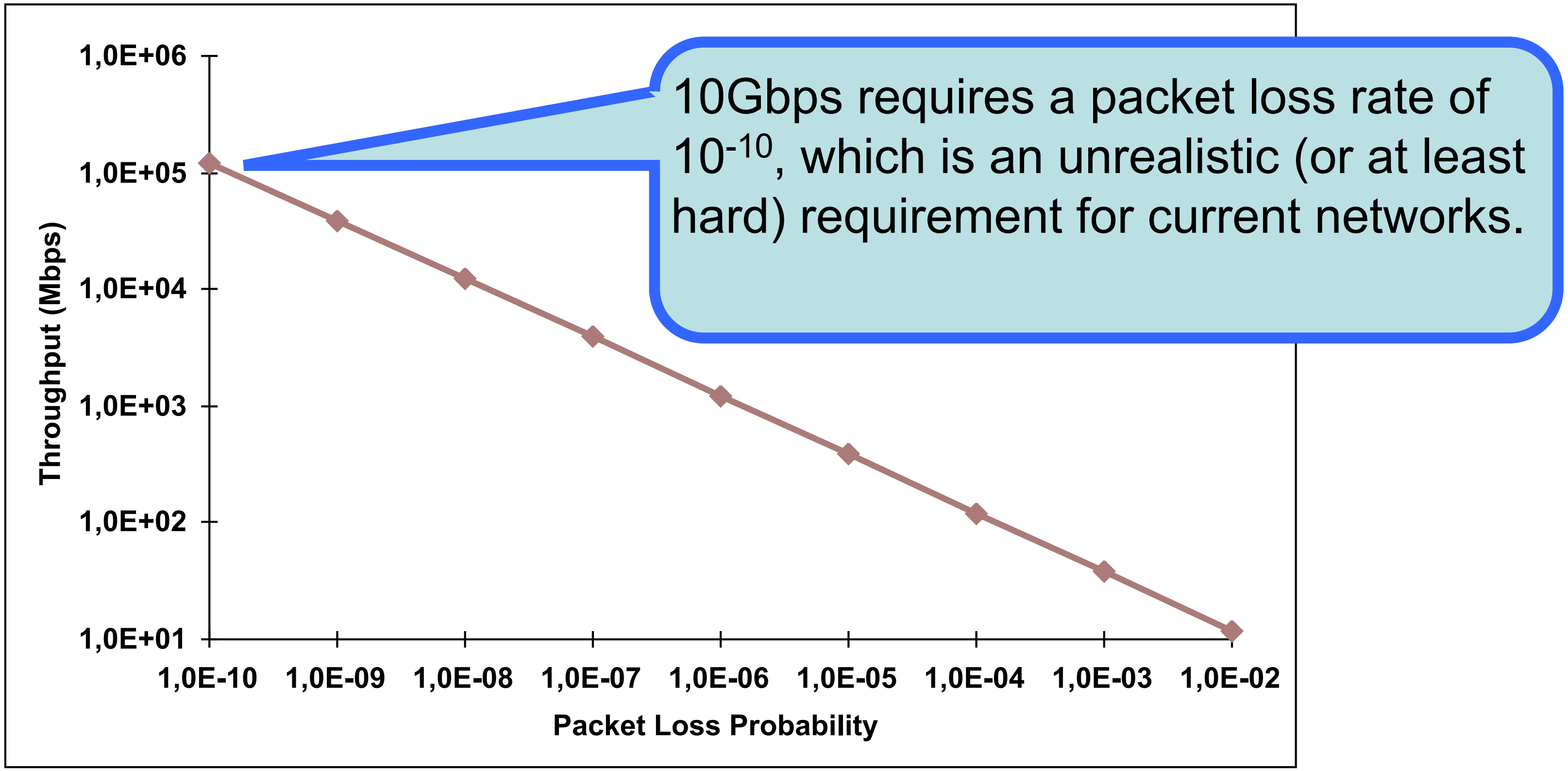
$$= \sqrt{3/2}\,\frac{1}{RTT\sqrt{p}}$$

$$Throughput = \frac{W}{RTT} = \sqrt{\frac{3}{2}}\,\frac{MTU}{RTT\sqrt{p}}$$

TCP's response function in image

$$Throughput = \frac{W}{RTT} = \sqrt{\frac{3}{2}}\frac{MTU}{RTT\sqrt{p}}$$

MTU: Packet Size
RTT: Round-Trip Time
P : Packet Loss Probability

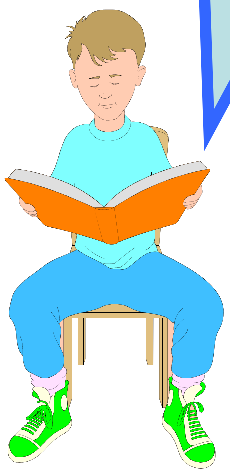


From Injong Rhee, Lisong Xu

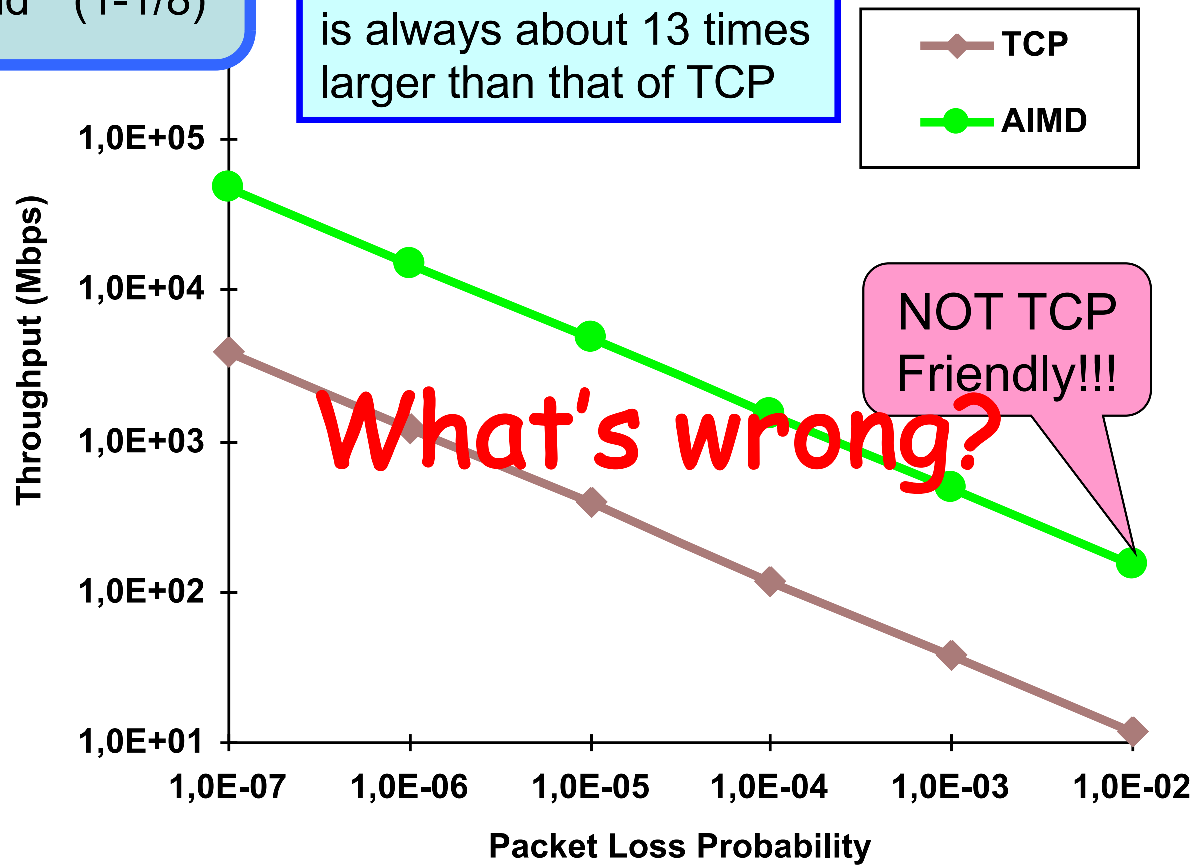
AIMD, general case

cwnd = cwnd + 1
⇩
cwnd = cwnd + 32

cwnd = cwnd * (1-1/2)
⇩
cwnd = cwnd * (1-1/8)



The throughput of AIMD is always about 13 times larger than that of TCP



- TCP: $R = \frac{MSS}{RTT}\frac{1.2}{p^{0.5}}$

- AIMD: $R = \frac{MSS}{RTT}\frac{15.5}{p^{0.5}}$

Inspired from Injong Rhee, Lisong Xu

High Speed TCP [Floyd]

- Modifies the response function to allow for more link utilization in current high-speed networks where the loss rate is smaller than that of the networks TCP was designed for (at most 10^{-2})

TCP Throughput (Mbps)	RTTs Between Losses	W	P
1	5.5	8.3	0.02
10	55.5	83.3	0.0002
100	555.5	833.3	0.000002
1000	5555.5	8333.3	0.00000002
10000	55555.5	83333.3	0.0000000002

Table 1: RTTs Between Congestion Events for Standard TCP, for 1500-Byte Packets and a Round-Trip Time of 0.1 Seconds.

From draft-ietf-tsvwg-highspeed-01.txt

Modifying the response

Packet Drop Rate P	Congestion Window W	RTTs Between Losses
10^-2	12	8
10^-3	38	25
10^-4	120	80
10^-5	379	252
10^-6	1200	800
10^-7	3795	2530
10^-8	12000	8000
10^-9	37948	25298
10^-10	120000	80000

Table 2: TCP Response Function for Standard TCP. The average congestion window W in MSS-sized segments is given as a function of the packet drop rate P.

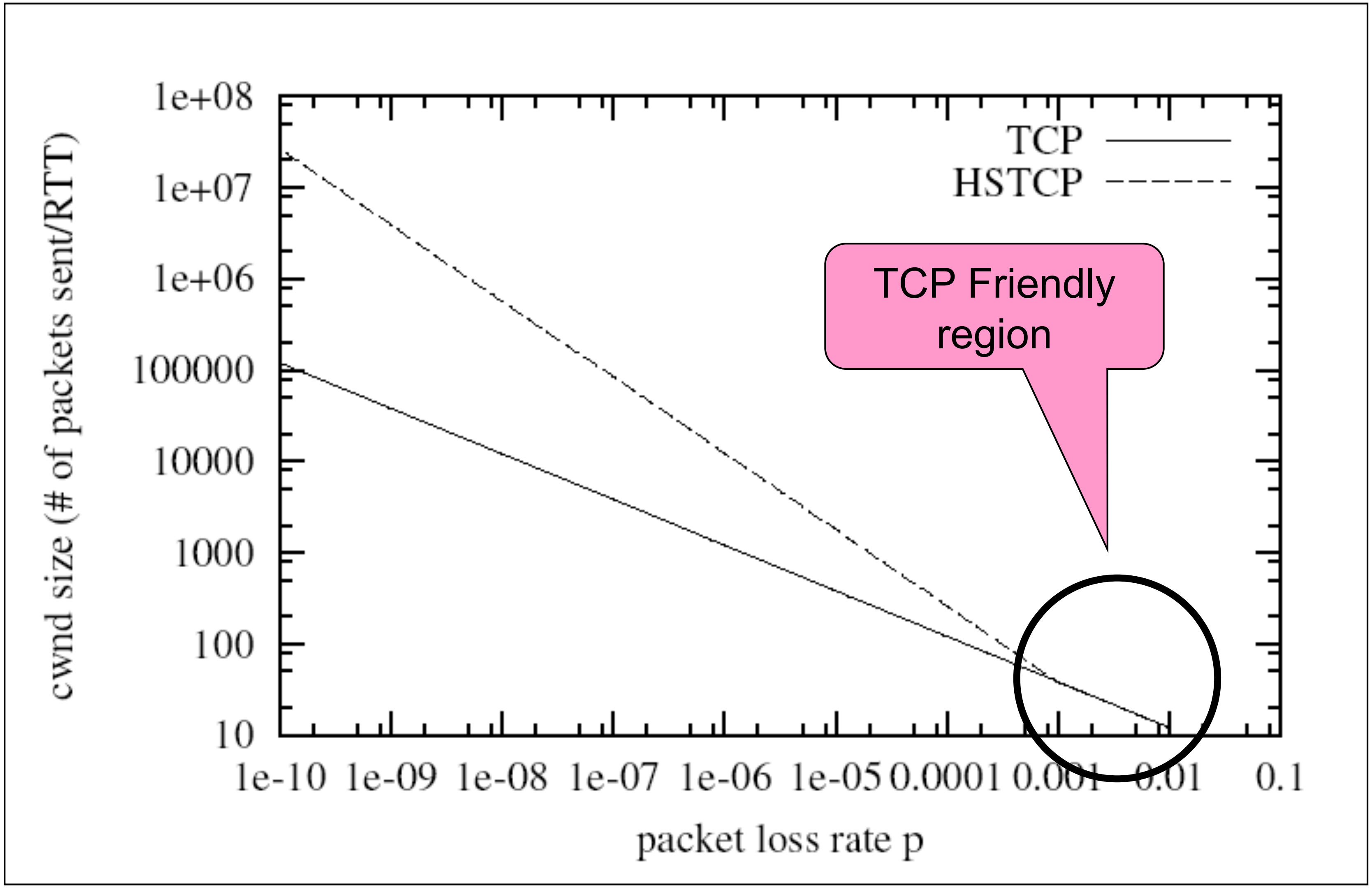
From draft-ietf-tsvwg-highspeed-01.txt

To specify a modified response function for HighSpeed TCP, we use three parameters, Low_Window, High_Window, and High_P. To Ensure TCP compatibility, the HighSpeed response function uses the same response function as Standard TCP when the current congestion window is at most Low_Window, and uses the HighSpeed response function when the current congestion window is greater than Low_Window. In this document we set Low_Window to 38 MSS-sized segments, corresponding to a packet drop rate of 10^-3 for TCP.

Packet Drop Rate P	**Congestion Window W**	**RTTs Between Losses**
10^-2	**12**	**8**
10^-3	**38**	**25**
10^-4	**263**	**38**
10^-5	**1795**	**57**
10^-6	**12279**	**83**
10^-7	**83981**	**123**
10^-8	**574356**	**180**
10^-9	**3928088**	**264**
10^-10	**26864653**	**388**

Table 3: TCP Response Function for HighSpeed TCP. The average congestion window W in MSS-sized segments is given as a function of the packet drop rate P.

See it in image



Relation with AIMD

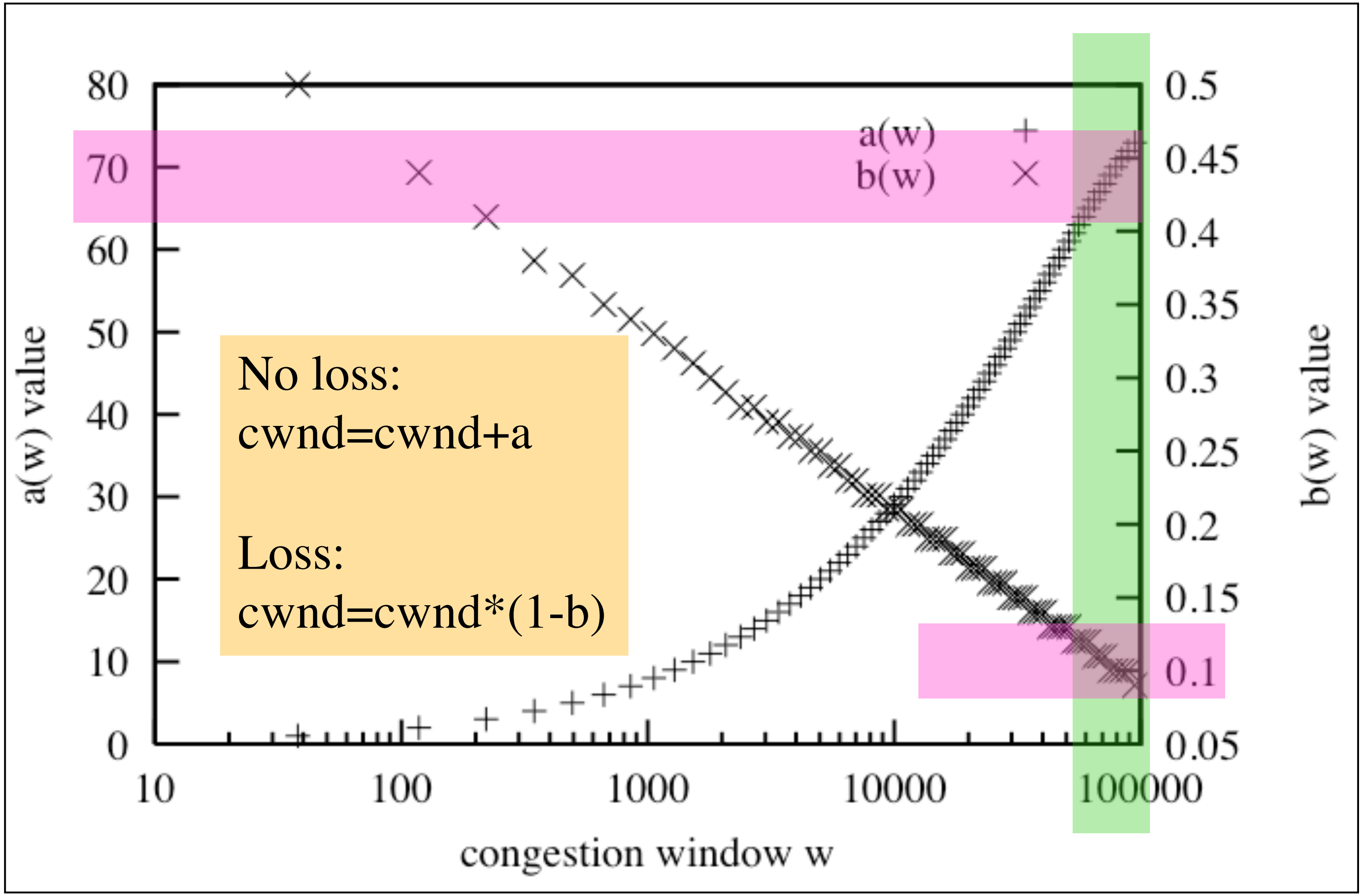
- TCP-AIMD
  - Additive increase: a=1
  - Multiplicative decrease: b=1/2

no loss:
cwnd = cwnd + 1

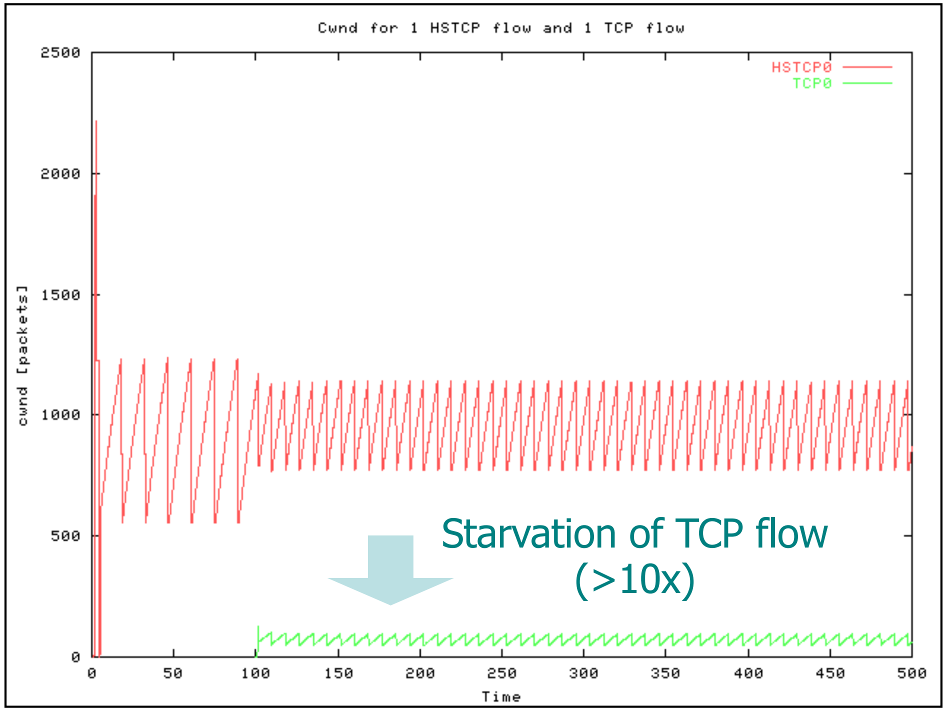
loss:
cwnd = cwnd*0.5

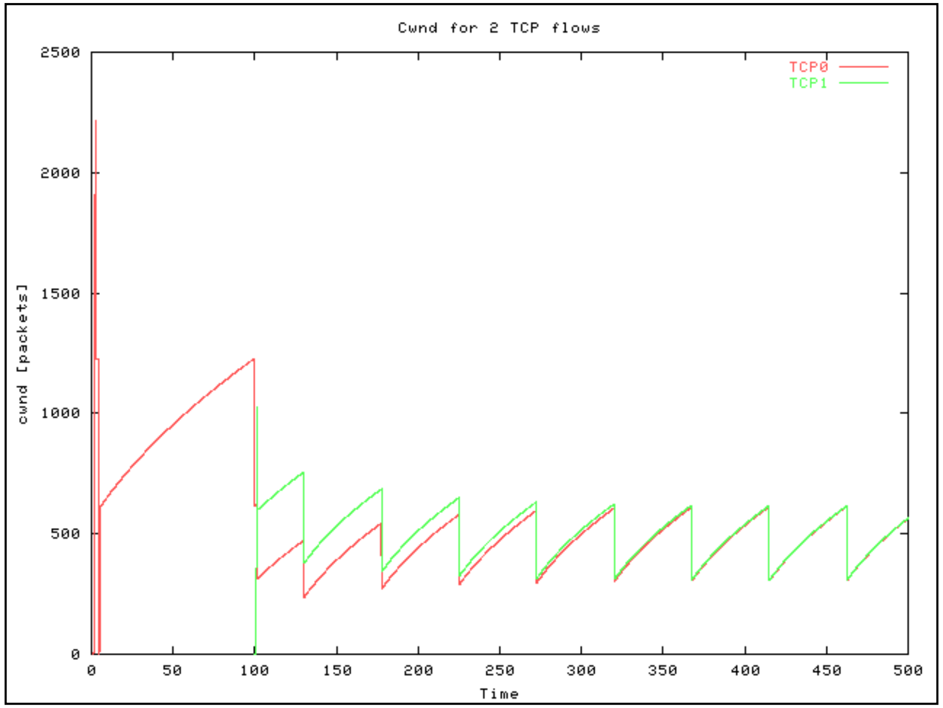
- HSTCP-AIMD
  - Link a & b to congestion window size
  - a = a(cwnd), b=b(cwnd)
  - General rules
    - the larger cwnd, the larger the increment
    - The larger cwnd, the smaller the decrement

Quick to grab bandwidth, slow to give some back!



Talking about dark side…





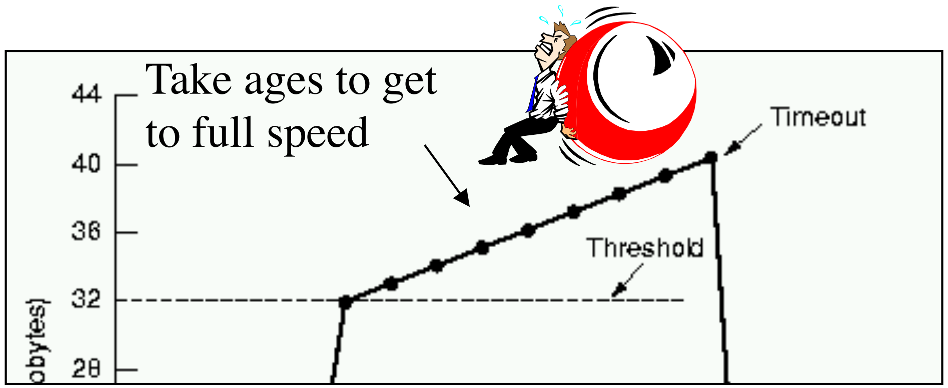
1 HSTCP and 1 TCP flow

2 TCP flows

SETUP RTT=100ms
Bottleneck BW=50Mbps
Qsize=BW*RTT
Qtype=DropTail

It's a search problem!

- Get to the available bandwidth: how to get there efficiently?



Linear increase not optimal

« Small jumps » strategy

Binary Search with Smax and Smin

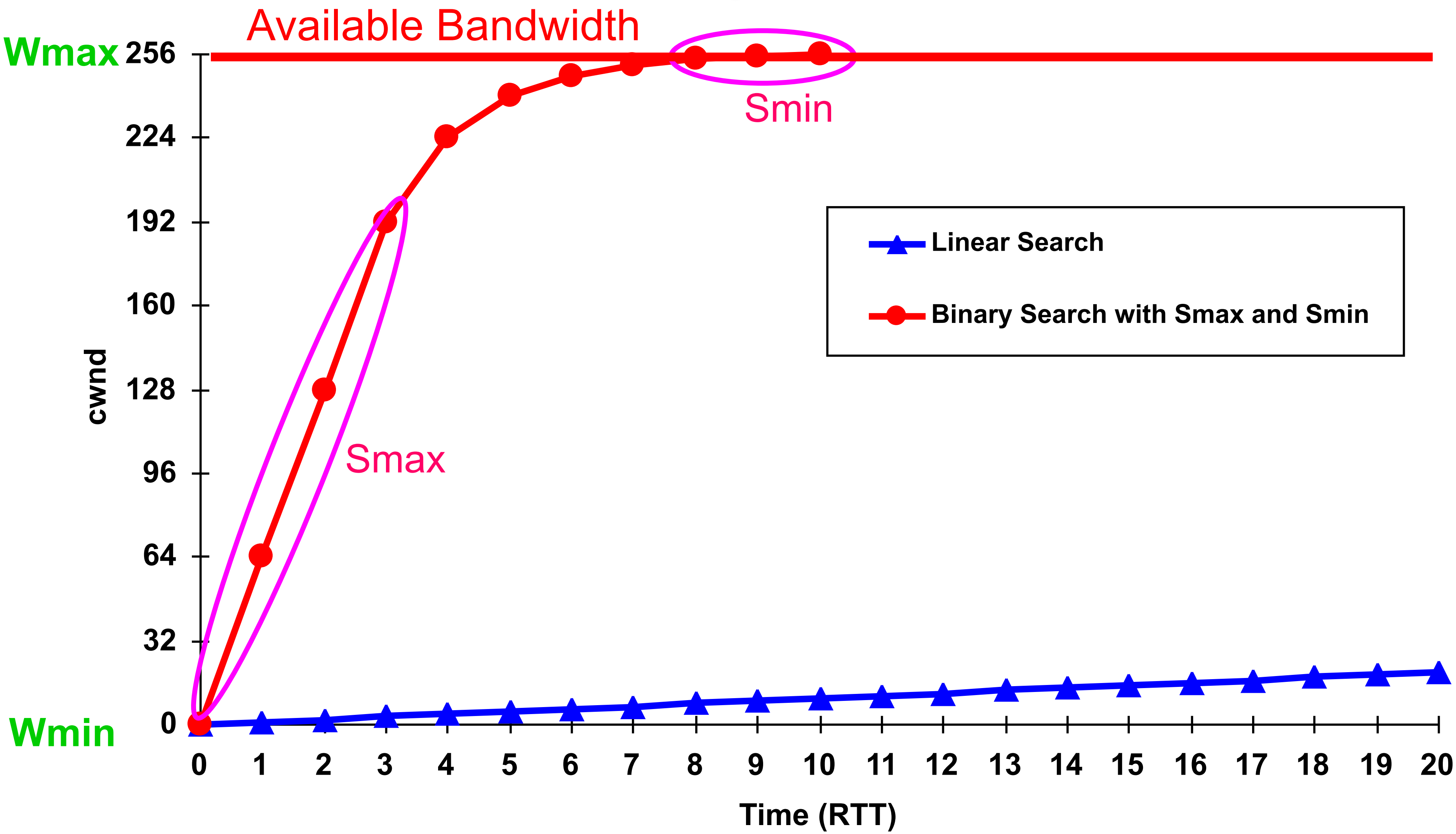
- Binary search

```
cwnd=1;
while (Wmin <= Wmax) {
    inc = (Wmin+Wmax)/2 - cwnd;
    if (inc > Smax)
            inc = Smax;
    else if (inc < Smin)
            inc = Smin;
    cwnd = cwnd + inc;
    if (no packet losses)
            Wmin = cwnd;
    else break;
}
```

- Wmax: Max Window
  - Usually the last cwnd value before packet drops (last fast recovery)
- Wmin: Min Window
- Smax: Max Increment
- Smin: Min Increment

Source Injong Rhee, Lisong Xu

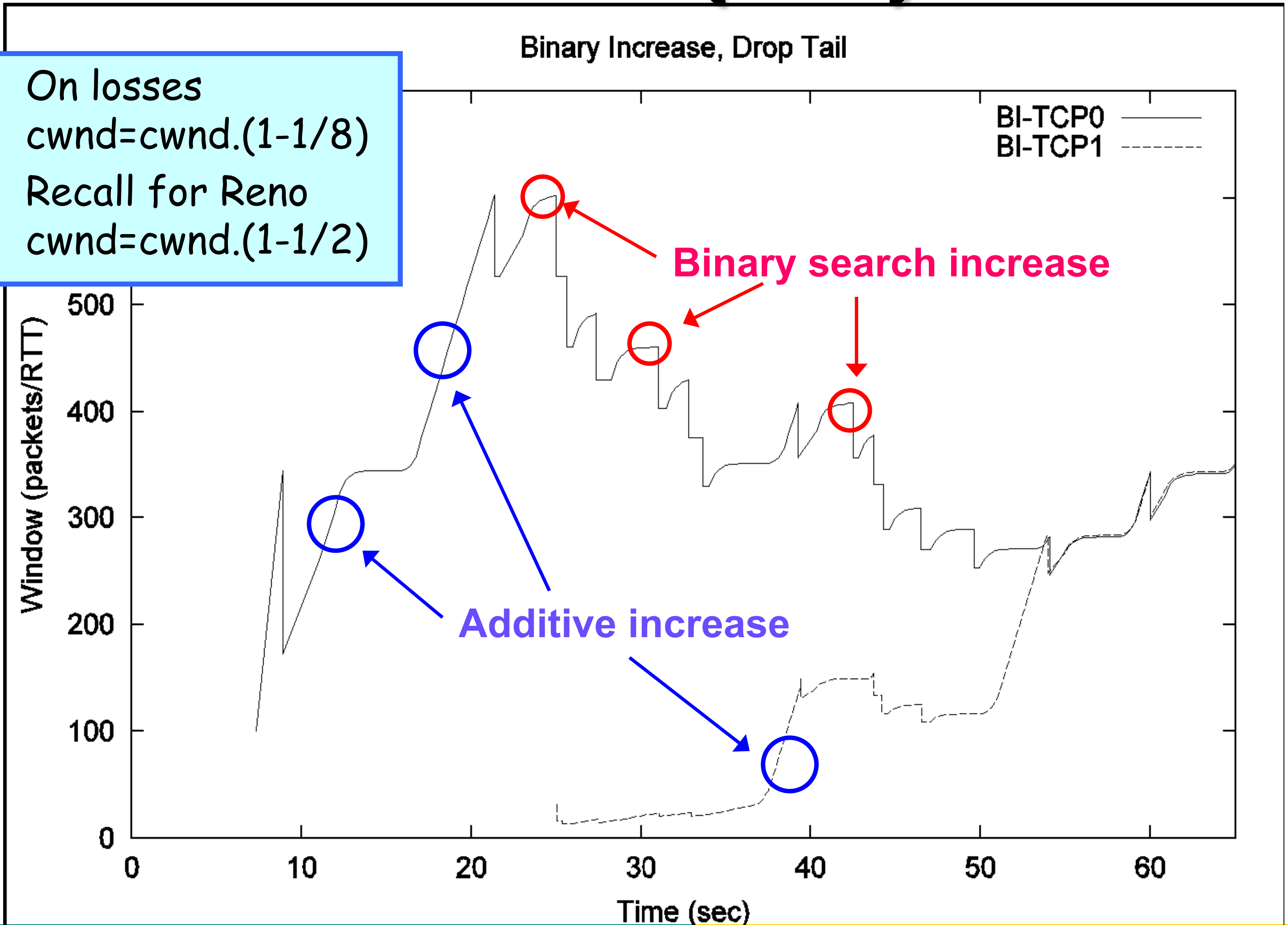
Binary Search with Smax and Smin



Source Injong Rhee, Lisong Xu

Binary Increase Congestion Control (BIC)

- On losses cwnd=cwnd.(1-1/8)
- Recall for Reno cwnd=cwnd.(1-1/2)



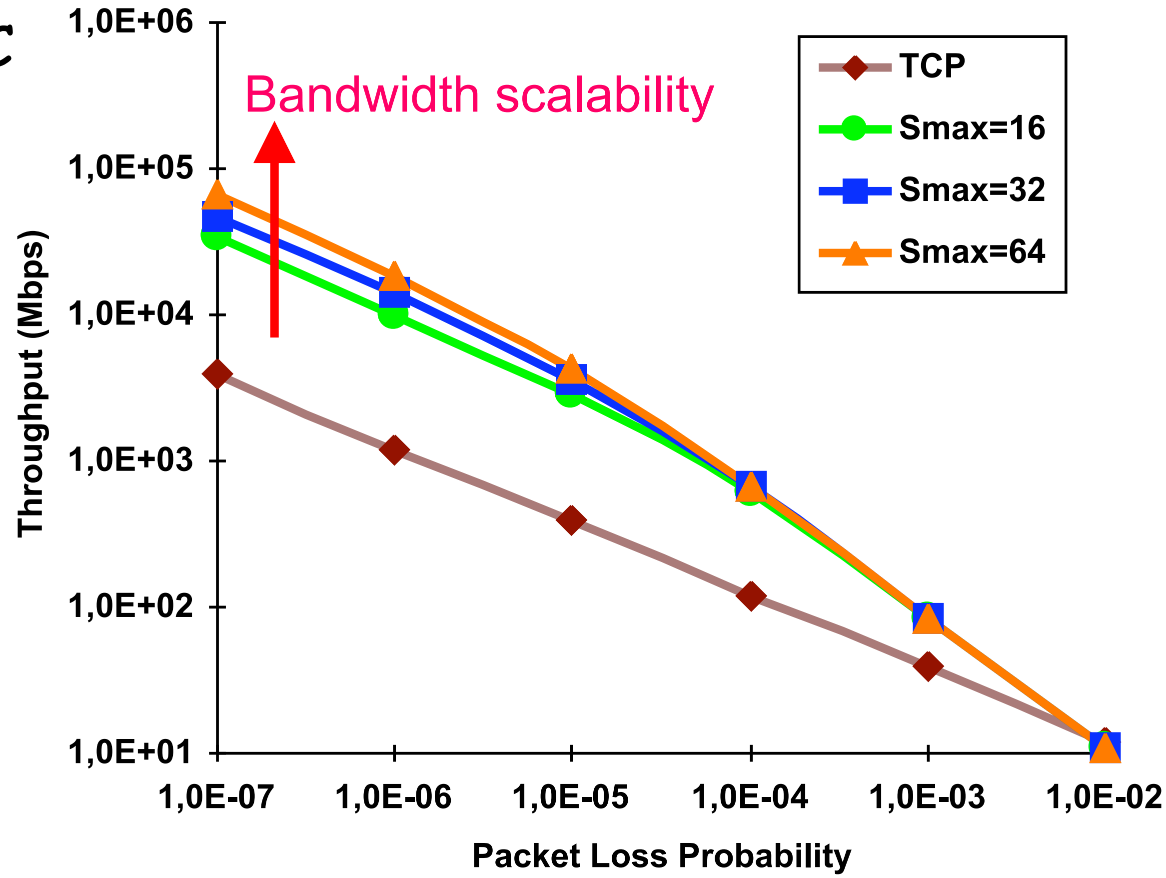
Source Injong Rhee, Lisong Xu

Setting Smax

- Response Function of BIC on high-speed networks

$$R = \frac{MSS}{RTT}\frac{2.7\sqrt{S_{\max}}}{p^{0.5}}$$

- Bandwidth scalability of BIC depends only on Smax
- RTT Fairness of BIC on high-speed networks is the same as that of AIMD



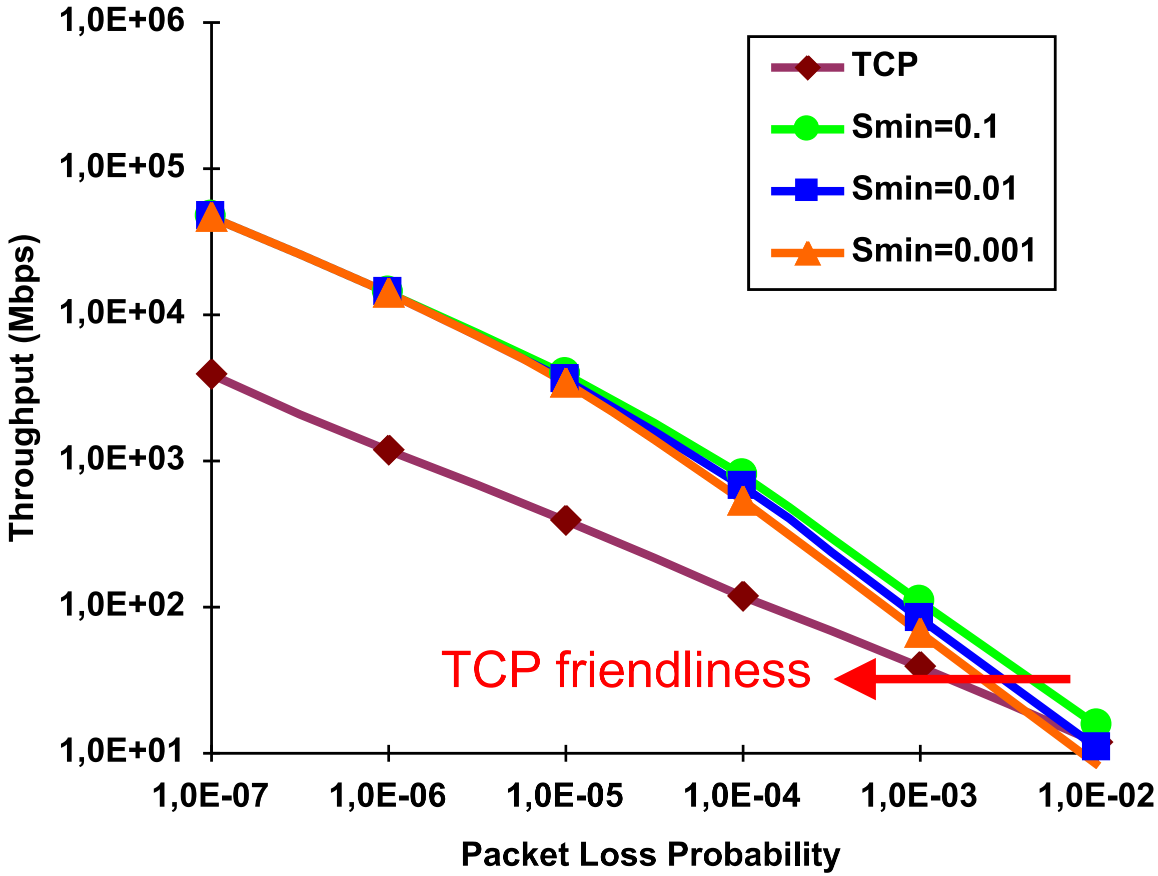
Source Injong Rhee, Lisong Xu

Setting Smin

- Response Function of BIC on low-speed networks

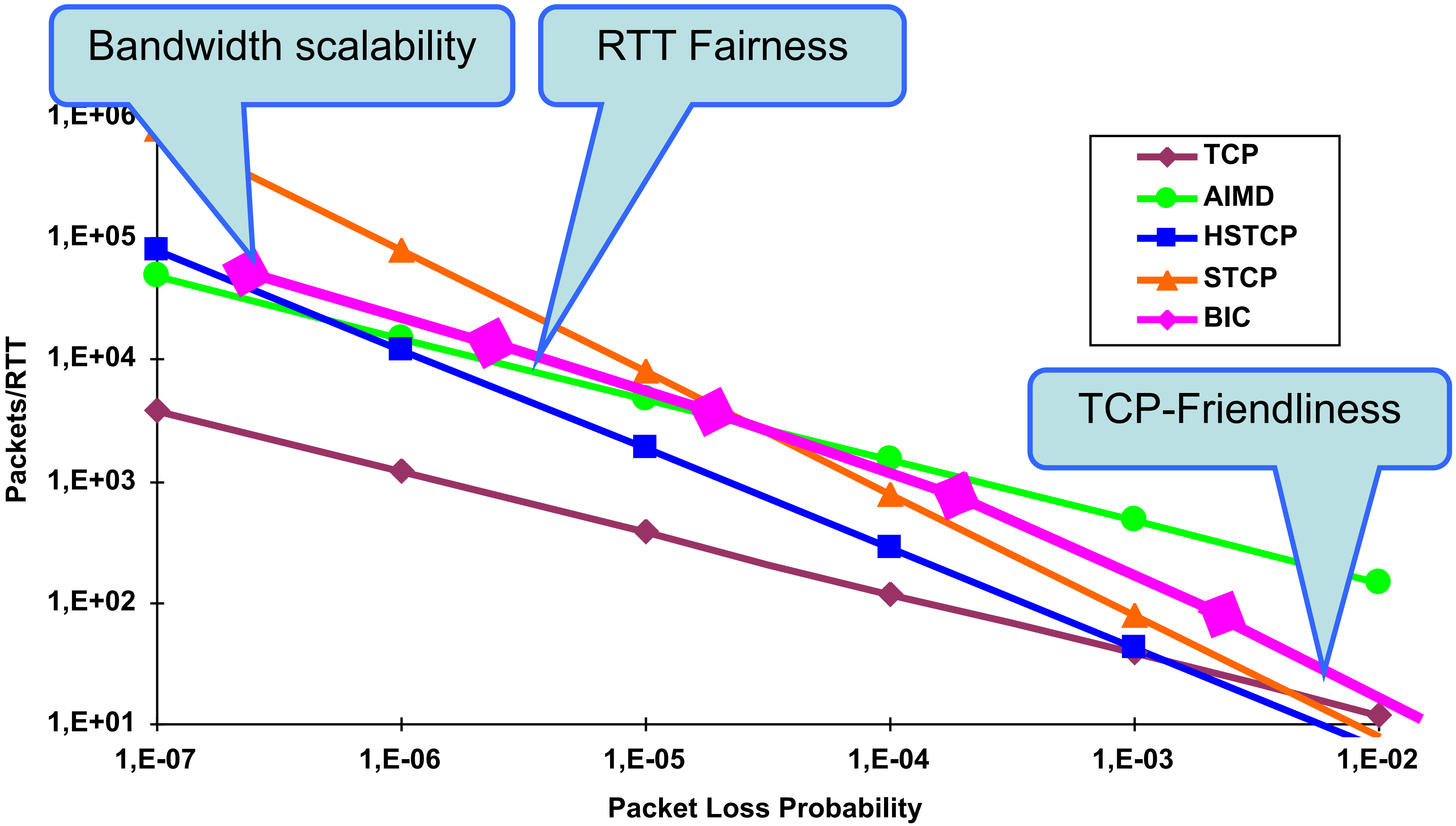
$$R = \frac{MSS}{RTT} f\left(p, S_{\min}\right)$$

- TCP-friendliness of BIC depends only on Smin



Source Injong Rhee, Lisong Xu

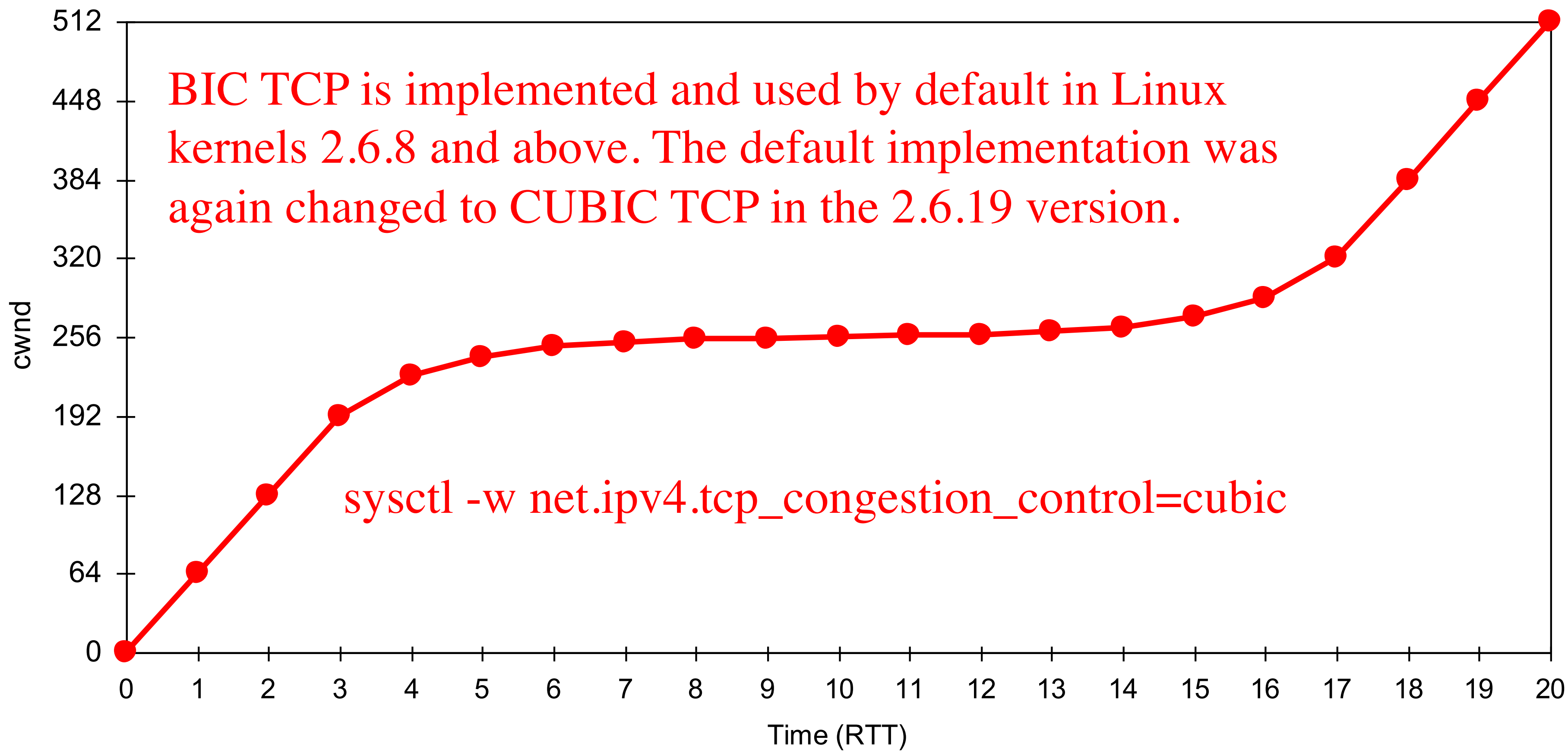
Response Functions



Source Injong Rhee, Lisong Xu

CUBIC

$$Cwnd = W_{\max} + C(t-K)^3$$

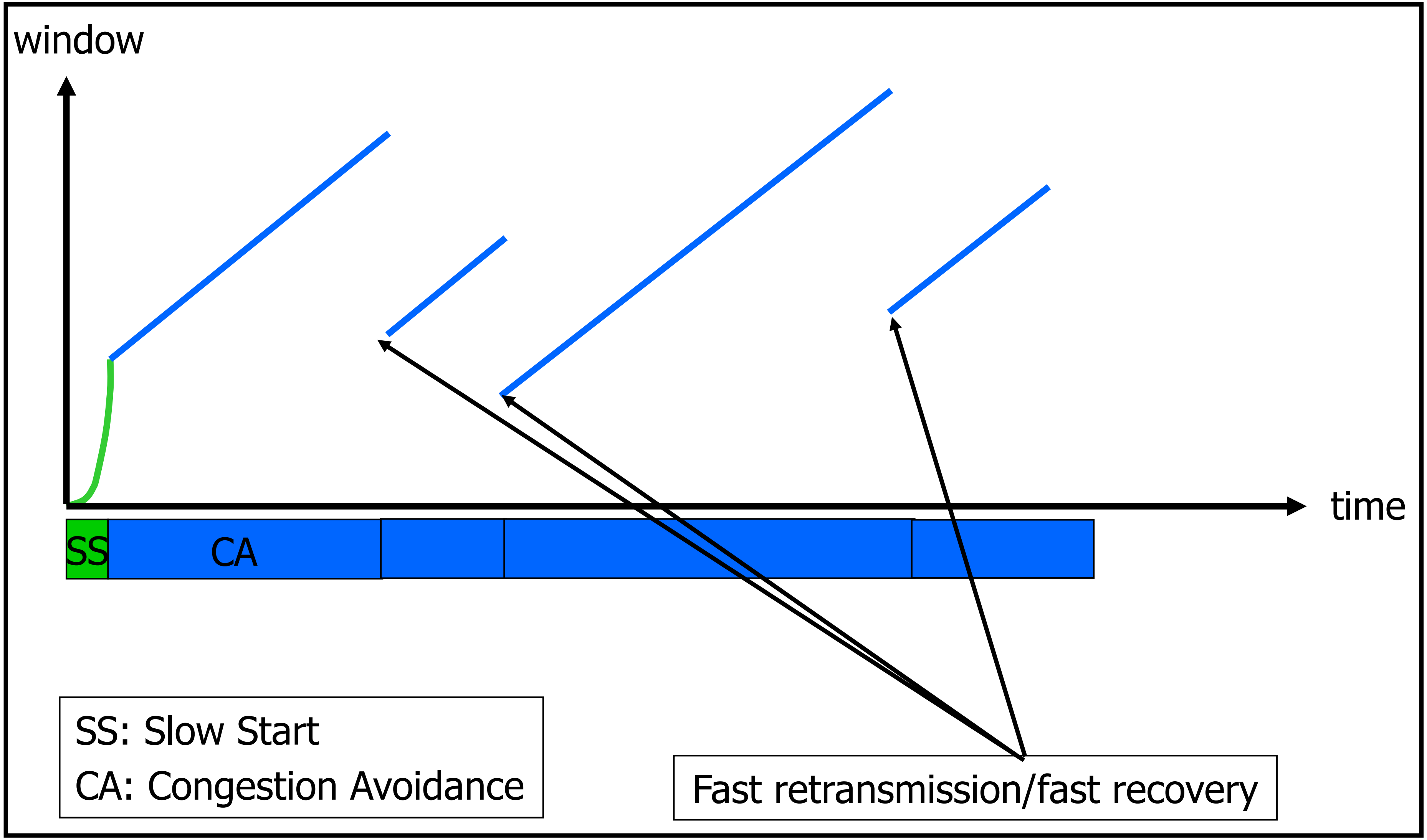


Source Injong Rhee, Lisong Xu

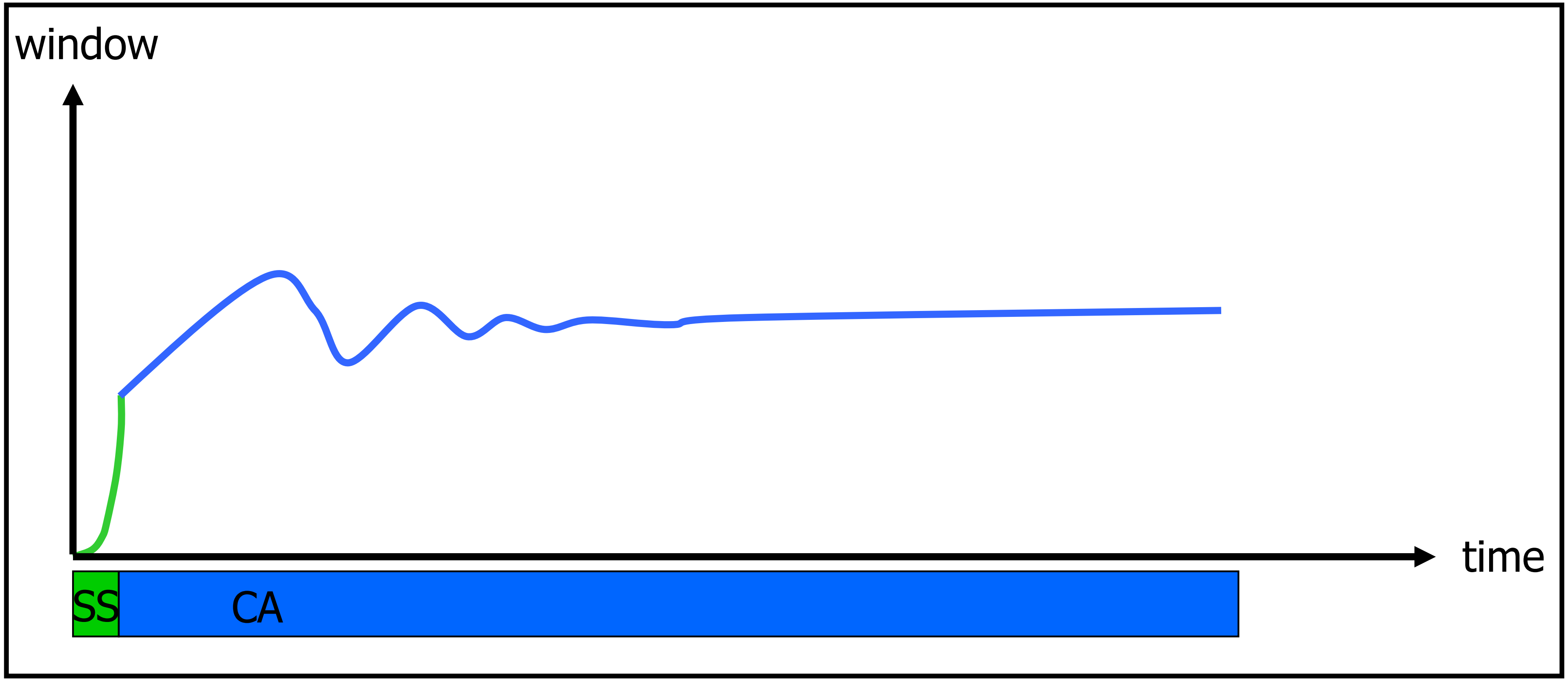
Loss-based vs Delay-based

- Most of TCP approaches uses loss-based factor to control cwnd's growth (TCP, HSTCP, BIC)
- A delay-based approach typically uses the RTT increases/decrease to decrease/increase cwnd
- When RTT increases, there is a high probability that packets are backlogged in router's buffer, indicating congestion in a near future

Loss-Based: TCP Reno



Delay-based: TCP Vegas (Brakmo & Peterson 1994)



- Converges, no retransmission
- … provided buffer is large enough

Compound TCP

- Compound TCP incorporates a delay-based factor in addition to the loss-based factor
- 2 window state variables
  - Cwnd
  - Dwnd: delay window
- Win=min(cwnd+dwnd, $a_{dvertised}$wnd)
- Cwnd updated as standard TCP

Congestion Control in CTCP (1)

- Calculate diff (backlogged pkts) samely as in TCP Vegas

$$Expected = win / baseRTT$$
$$Actual = win / RTT$$
$$Diff = (Expected - Actual) \cdot baseRTT$$

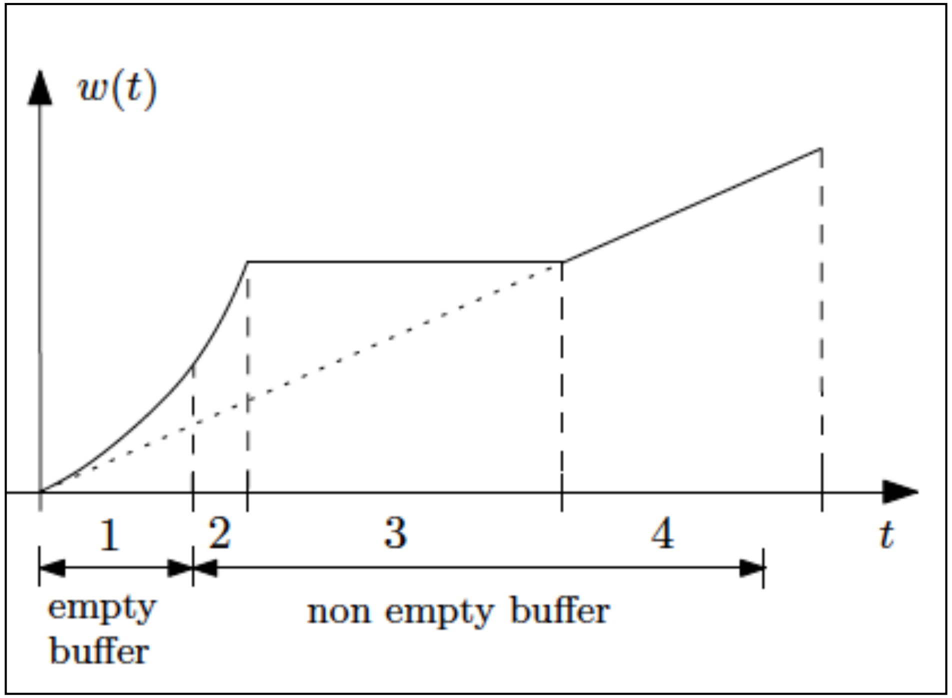
- Control functions

$$dwnd(t+1) = \begin{cases} dwnd(t) + (\alpha \cdot win(t)^k - 1)^+, \text{if } diff < \gamma \\ \left(dwnd(t) - \zeta \cdot diff\right)^+, \text{if } diff \geq \gamma \\ \left(win(t) \cdot (1-\beta) - cwnd / 2\right)^+, \text{if loss is detected} \end{cases}$$

$(.)^+ = \max(.,0)$

Congestion Control in CTCP (2)

- Reno
  - $W_{i+1}=W_i+1$
- CTCP ($\xi=1$)
  - $W_{i+1}=W_i+\alpha W_i^k$, ❶❷
  - $W_{i+1}=W_i$, ❸
  - $W_{i+1}=W_i+1$, ❹
- Δ_i: queue size estimation
- If $\Delta_i > \gamma$, move from ❷ to ❸.



CTCP and Windows Vista

- CTCP is enabled by default in computers running beta versions of Windows Server 2008 and disabled by default in computers running Windows Vista. CTCP can be enabled with the command

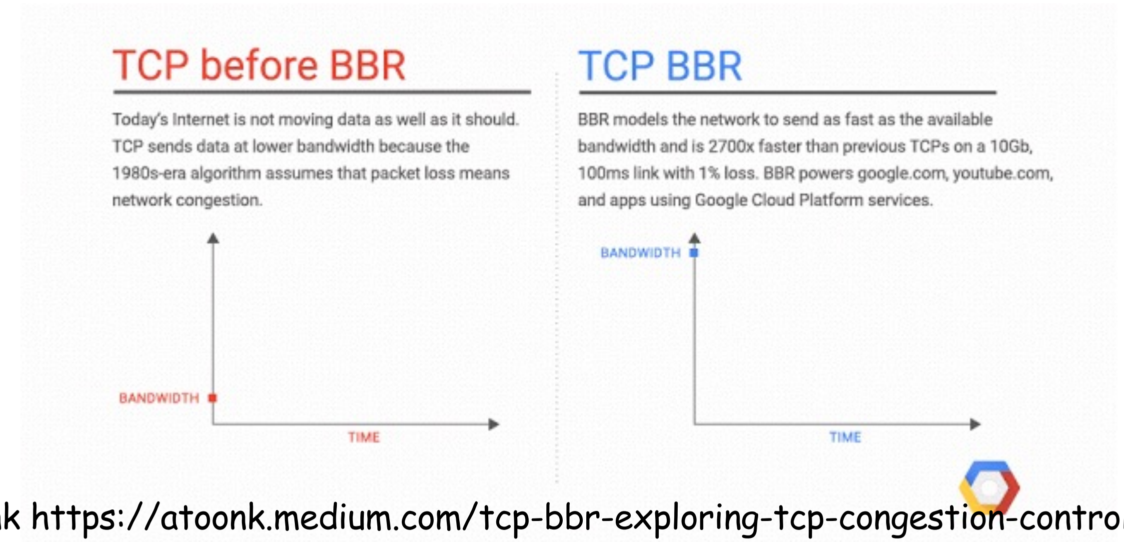
```
netsh interface tcp set global congestionprovider=ctcp
```

BBR – by Google (2016)

- Bottleneck Bandwidth and Round-trip propagation time
- uses latency, instead of lost packets as main factor to find sending rate
- top priority is to reduce queue usage
- BBR in Linux kernel since version 4.9
- https://www.youtube.com/watch?v=fLZEYiSCviE

BBR - con't

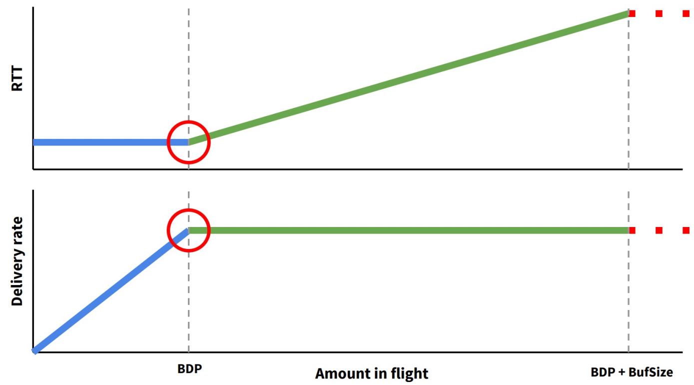
- in addition to Google, Dropbox and Spotify are two other examples where BBR is being used or experimented with



Source Andree Toonk https://atoonk.medium.com/tcp-bbr-exploring-tcp-congestion-control-84c9c11dc3a9

BBR - principles

- Try to keep near the optimal operating point
- -> "jump" faster to the bottleneck point

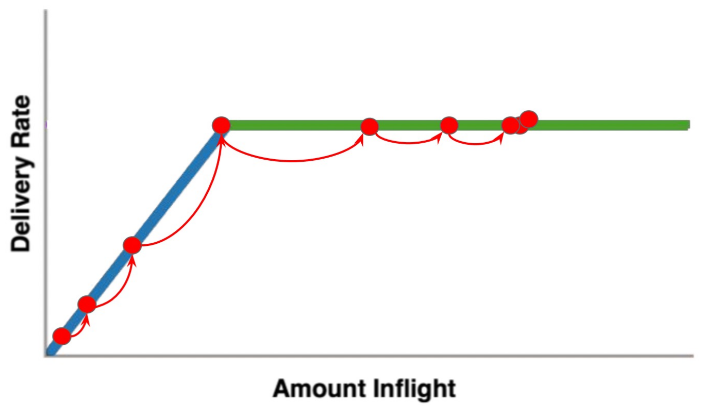


- Model network, update windowed max BW and min RTT estimates on each ACK

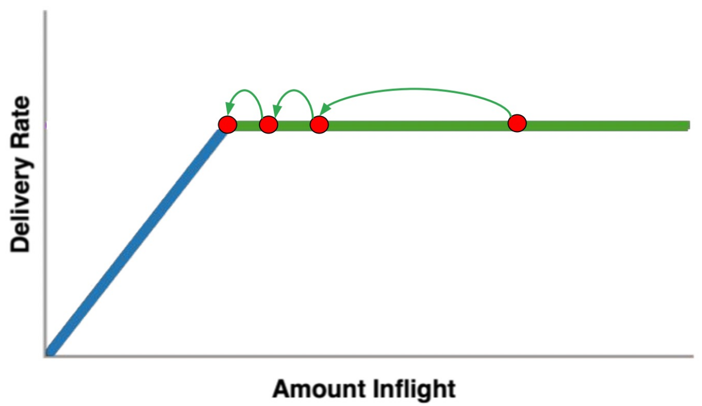
Source https://www.ietf.org/proceedings/97/slides/slides-97-iccrg-bbr-congestion-control-02.pdf

BBR - phases

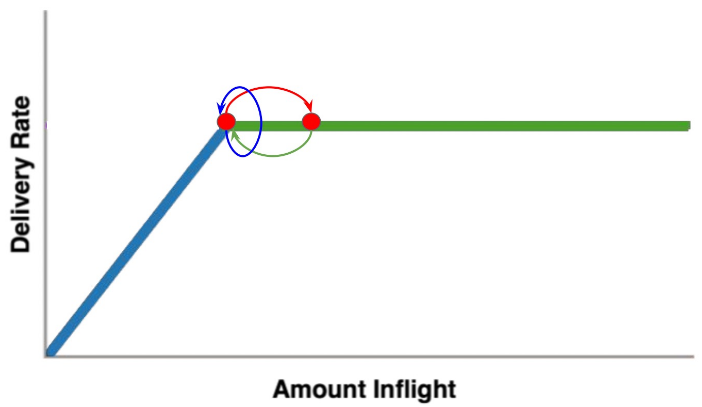
STARTUP: exponential BW search



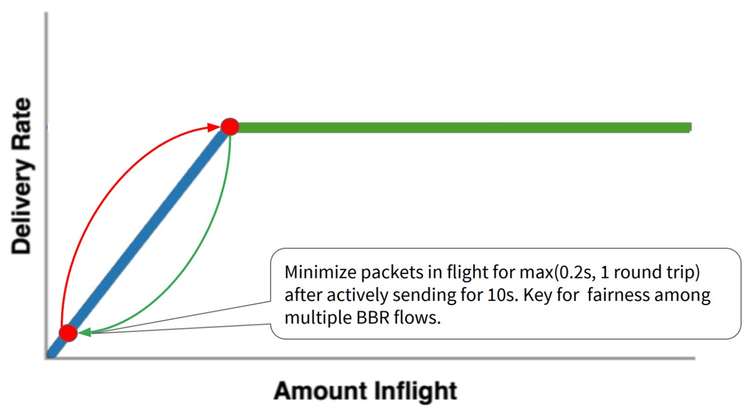
DRAIN: drain the queue created during startup



PROBE_BW: explore max BW, drain queue, cruise

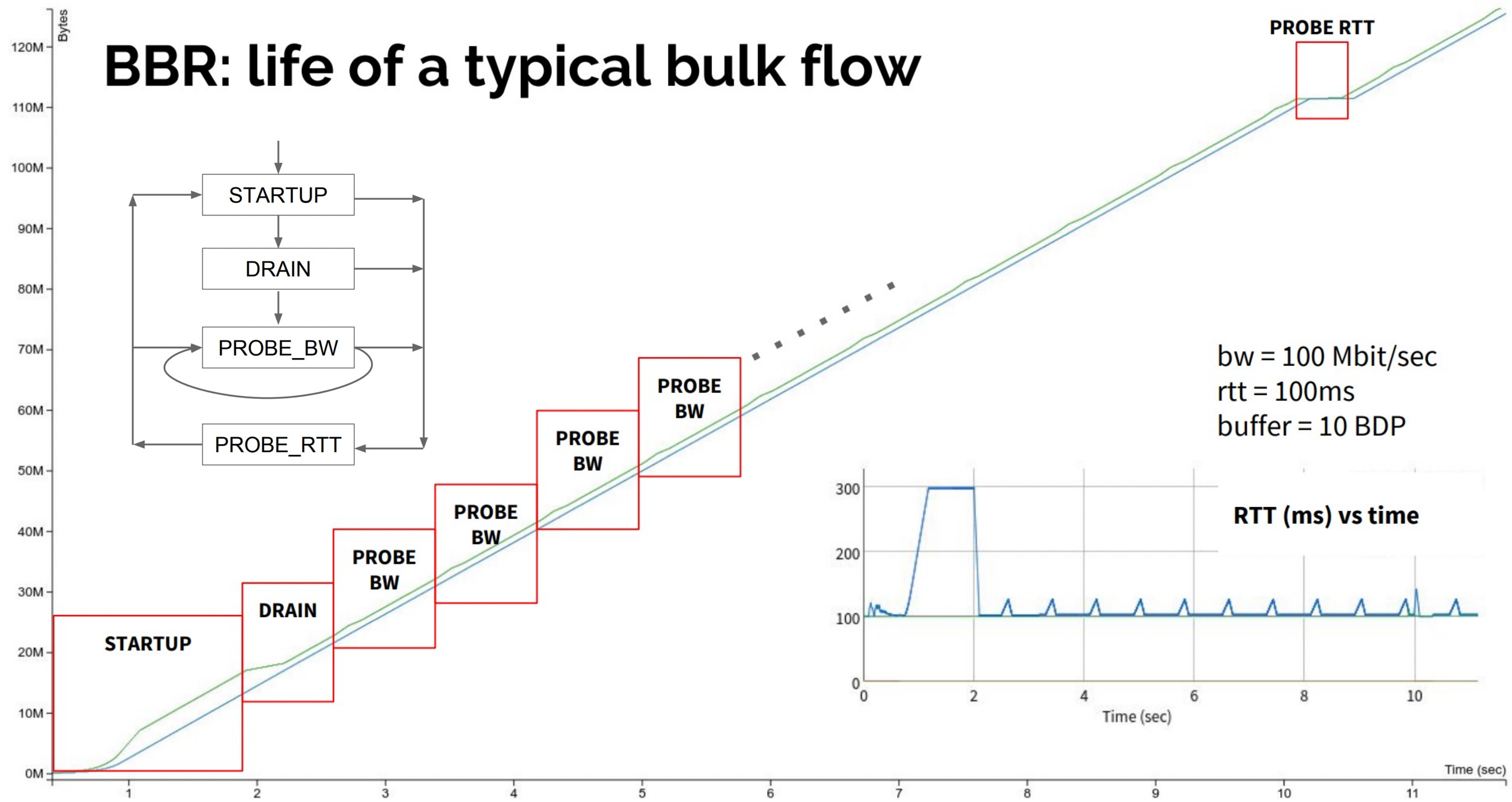


PROBE_RTT drains queue to refresh min_RTT



Source https://www.ietf.org/proceedings/97/slides/slides-97-iccrg-bbr-congestion-control-02.pdf

BBR – global view



Source https://www.ietf.org/proceedings/97/slides/slides-97-iccrg-bbr-congestion-control-02.pdf

BBR – some perf

Throughput	Congestion control algorithm (sender)	latency	loss
2.35 Gb/s	Cubic	<1ms	0%
195 Mb/s	Reno	140ms	0%
347 Mb/s	Cubic	140ms	0%
344 Mb/s	Westwood	140ms	0%
340 Mb/s	BBR	140ms	0%
1.13 Mb/s	Reno	140ms	1.5% (sender > receiver)
1.23 Mb/s	Cubic	140ms	1.5% (sender > receiver)
2.46 Mb/s	Westwood	140ms	1.5% (sender > receiver)
160 Mb/s	BBR	140ms	1.5% (sender > receiver)
0.65 Mb/s	Reno	140ms	3% (sender > receiver)
0.78 Mb/s	Cubic	140ms	3% (sender > receiver)
0.97 Mb/s	Westwood	140ms	3% (sender > receiver)
132 Mb/s	BBR	140ms	3% (sender > receiver)

Source Andree Toonk https://atoonk.medium.com/tcp-bbr-exploring-tcp-congestion-control-84c9c11dc3a9

BBR – want to know more?

- Slides: https://www.ietf.org/proceedings/97/slides/slides-97-iccrg-bbr-congestion-control-02.pdf
- Youtube from Matt Mathis: https://www.youtube.com/watch?v=6umI08w35VY

Downsides of BBR

- There are downsides!
- Very aggressive compare to TCP and its variants BIC/CUBIC
- Somehow similar to HSTCP in getting bandwidth aggressively

Nothing is perfect :-(

- Multiple or parallel streams
  - How many streams?
  - OS high overheads
  - Tradeoff between window size and number of streams
- New protocol
  - Fairness issues?
  - Deployment issues?
  - Still too early to know the side effects

Hostile environments

- Asymetric networks
  - Satellite links & terrestrial links
- Wireless (WiFi, WiMax, 5G)
  - High loss probability
  - Losses ≠congestions
- Ad-Hoc
  - Small capacity
  - High mobility
- Wireless Sensor Networks/IoT
  - **High resource constraints**