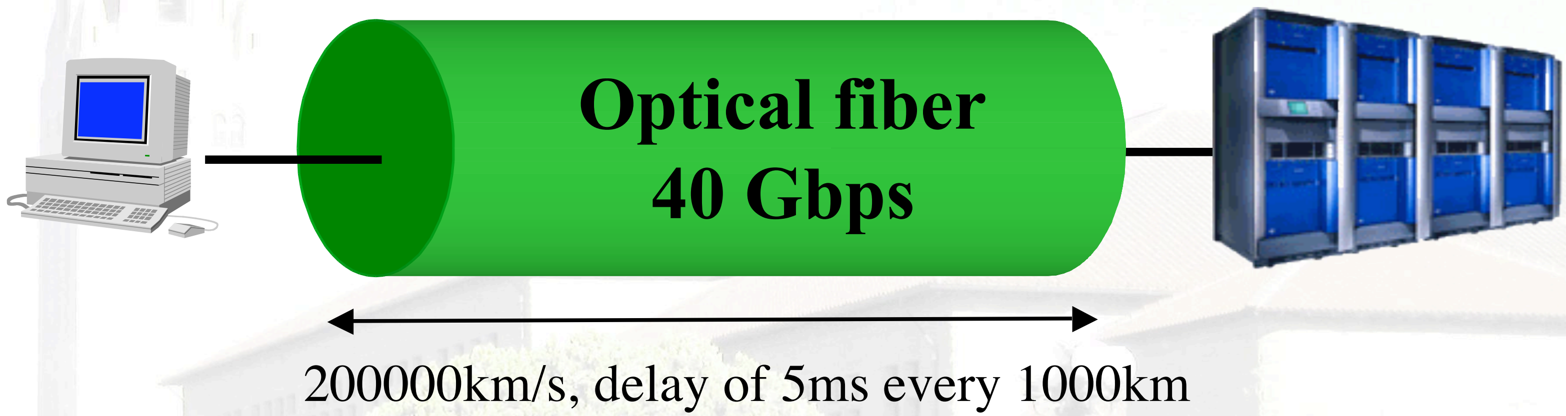
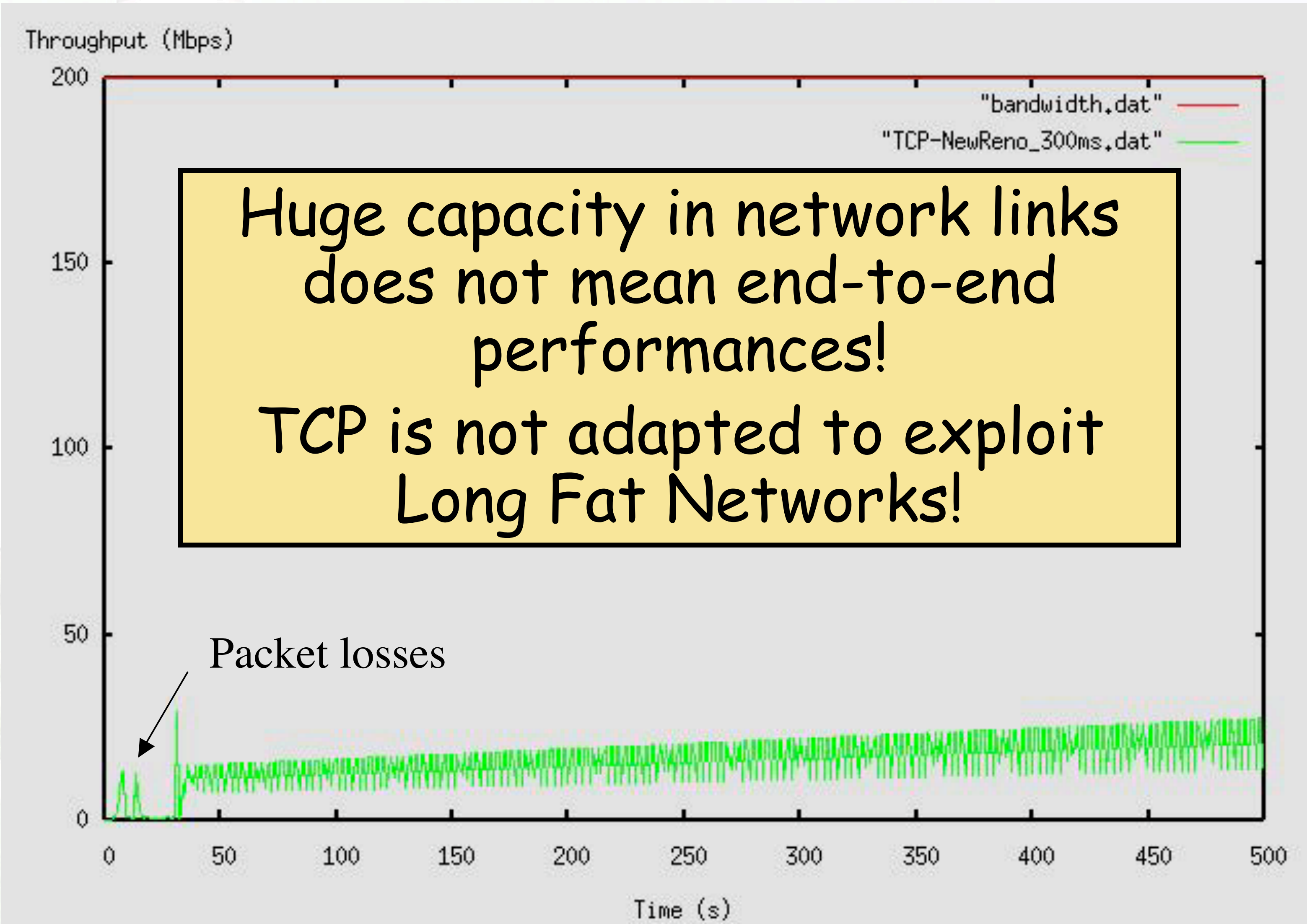
TCP & High-speed networks

NEW CHAPTER



- 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

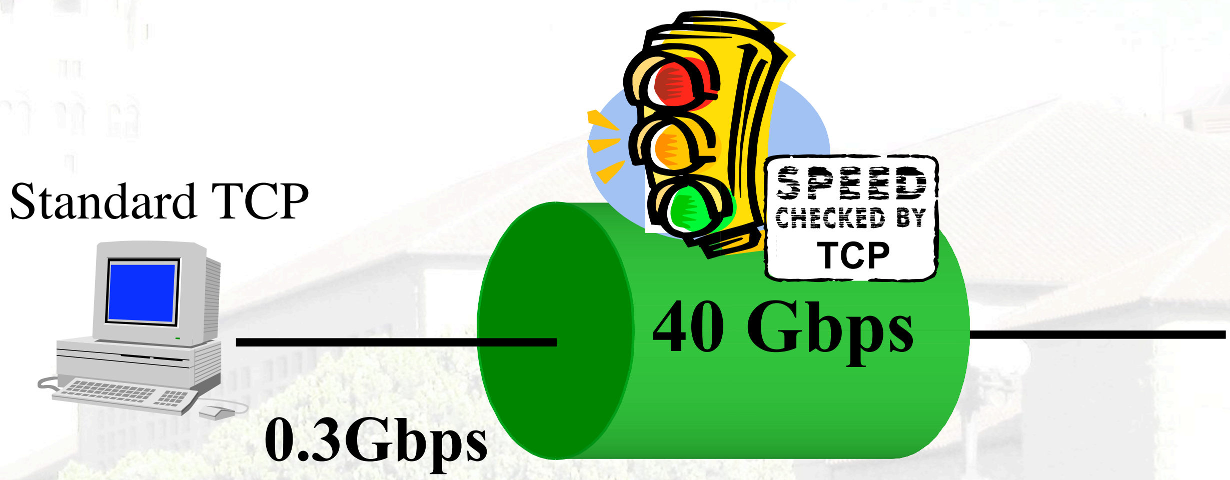
The reality check: TCP on a 200Mbps link



Huge capacity in network links does not mean end-to-end performances!

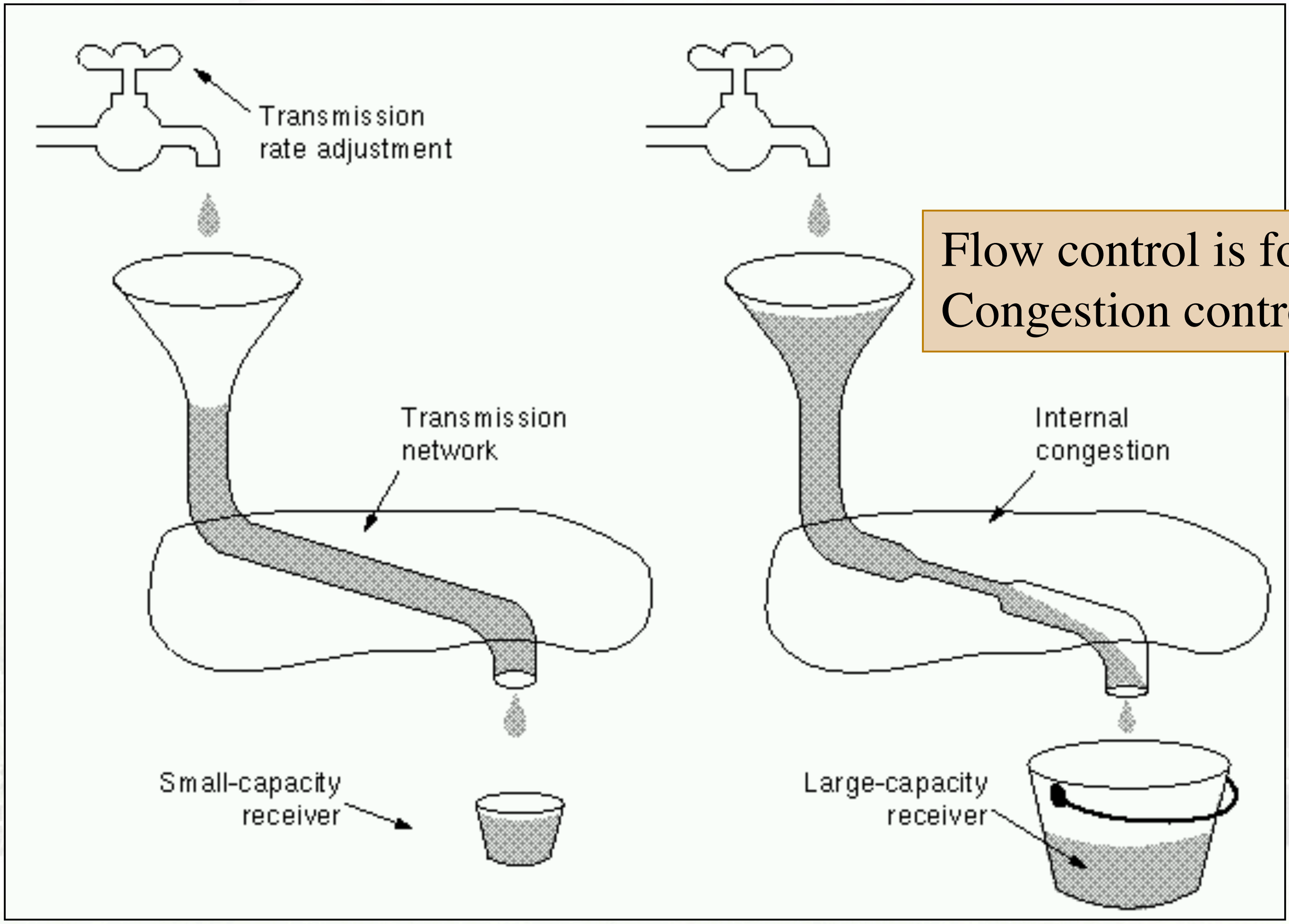
TCP is not adapted to exploit Long Fat Networks!

The things about TCP your mother never told you!



- If you want to transfer a 1Go file with a standard TCP stack, you will need minutes even with a 40Gbps (how much in $?) link!

Let's go back to the origin!



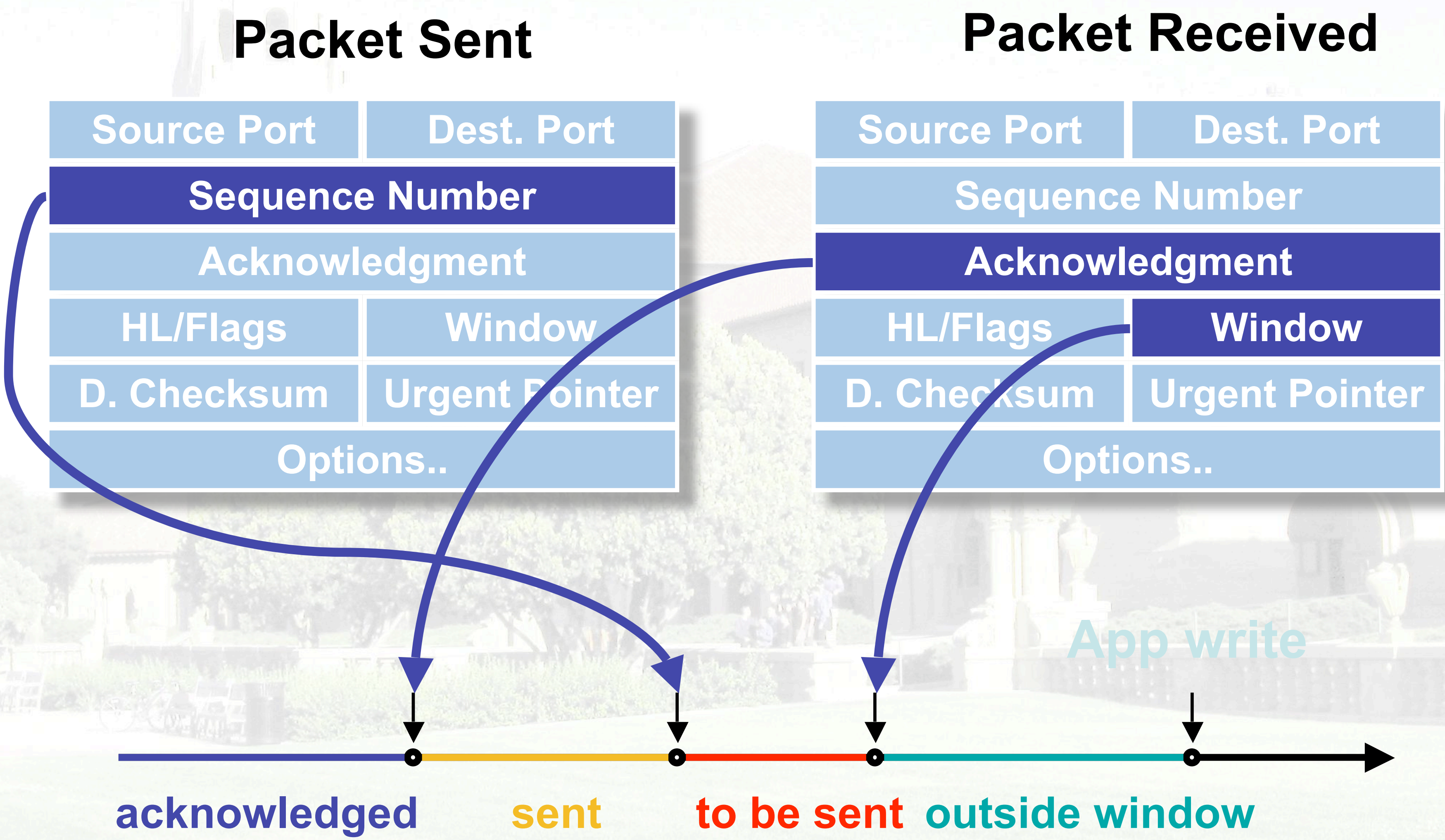
From Computer Networks, A. Tanenbaum

Flow control is for receivers
Congestion control is for the network

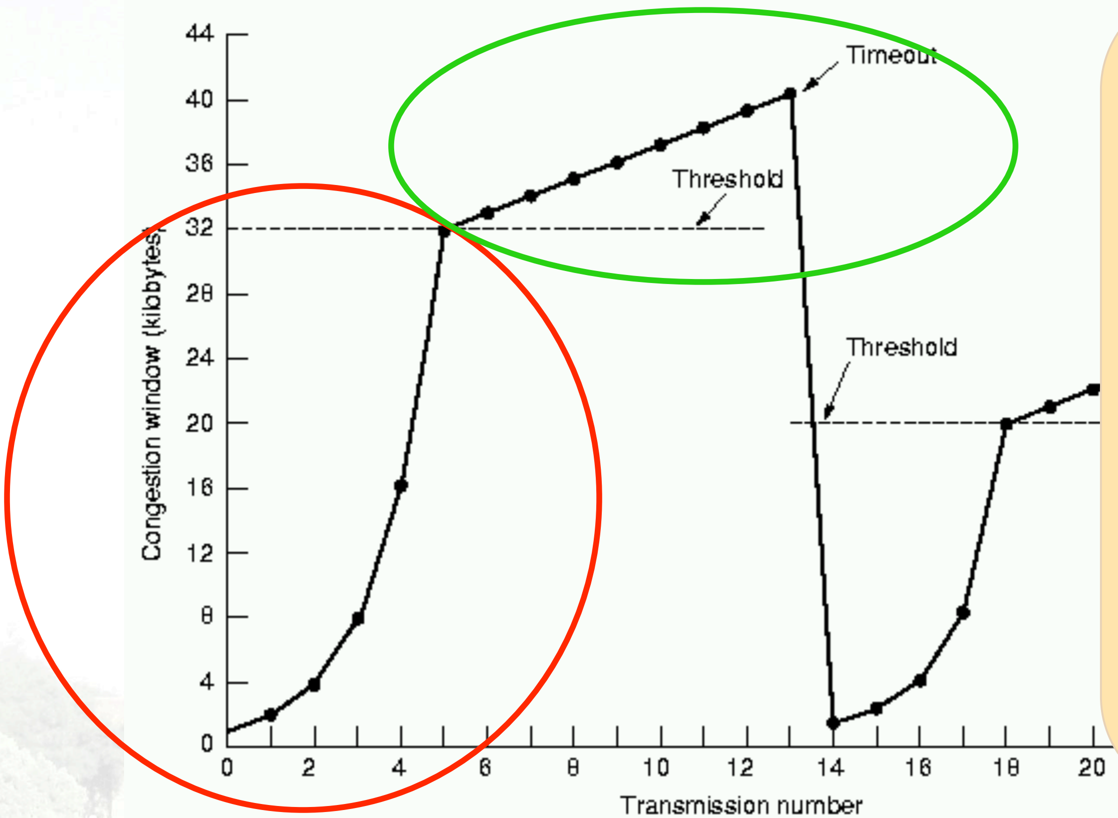
Congestion collapse was first observed in 1986 by V. Jacobson. Congestion control was added to TCP (TCP Reno) in 1988.

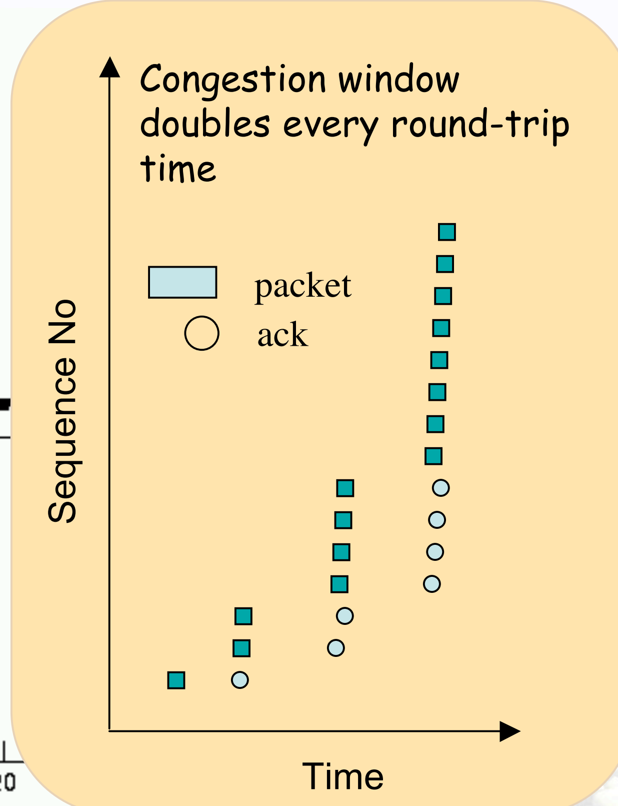
Flow control

prevents receiver's buffer overfow



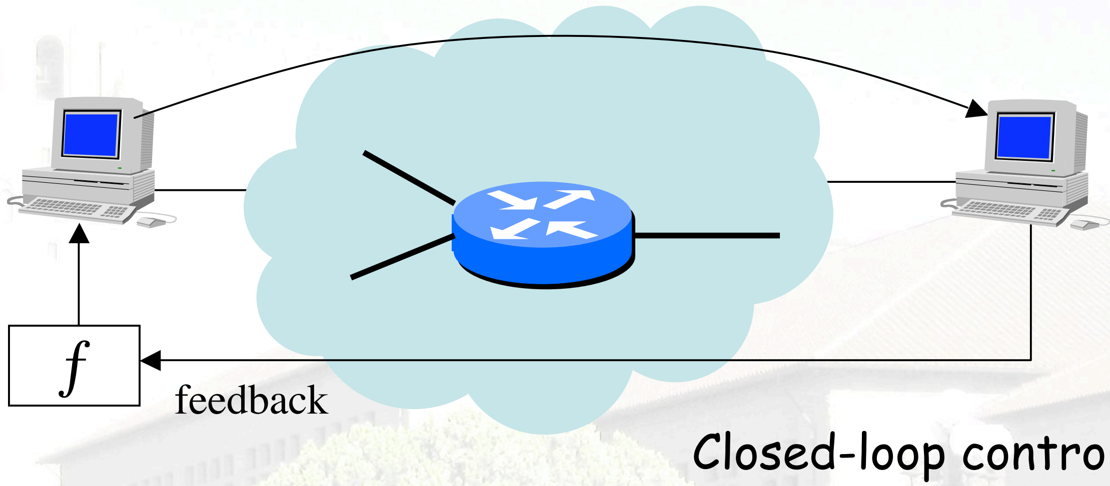
TCP congestion control: the big picture





- 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 the control theory point of view

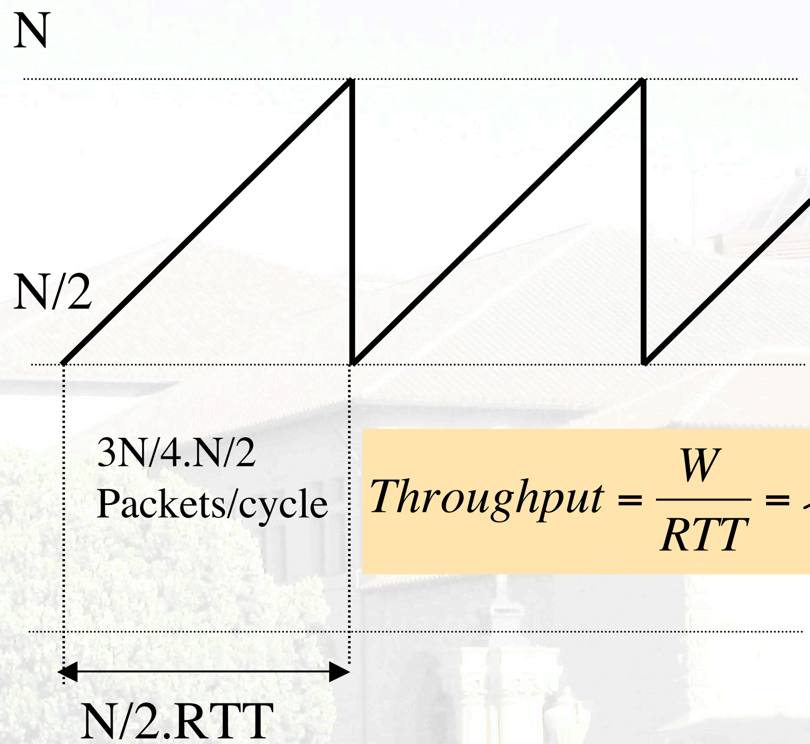


- Feedback should be frequent, but not too much otherwise there will be oscillations
- Can not control the behavior with a time granularity less than the feedback period

The TCP saw-tooth curve

TCP behavior in steady state

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



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

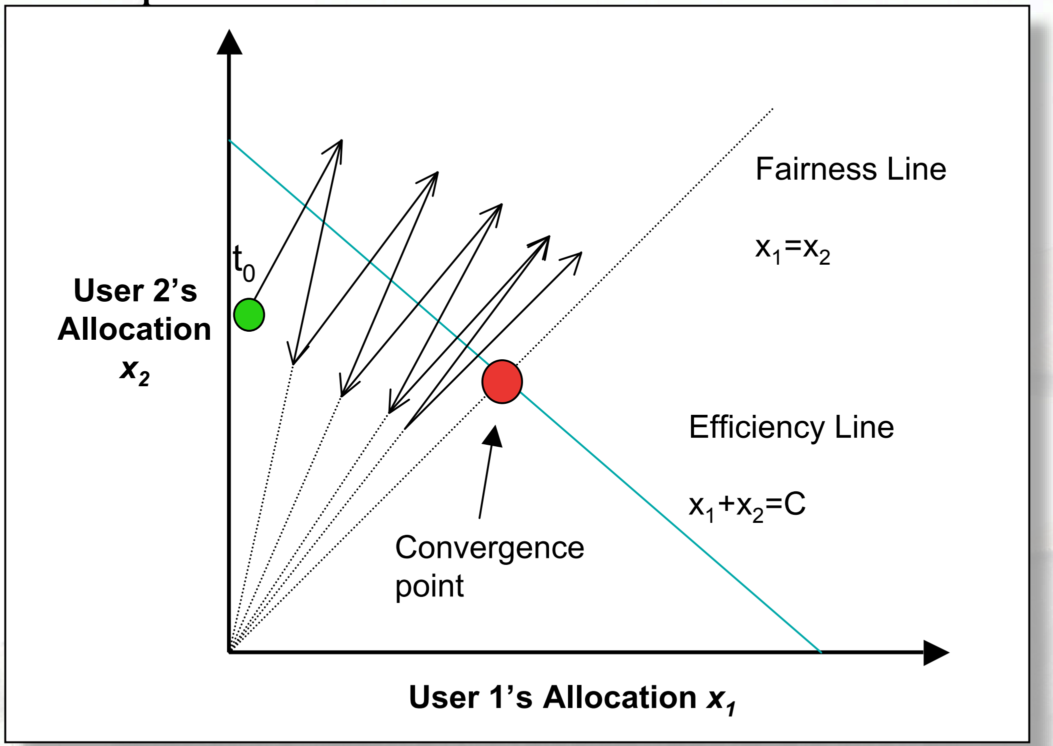
- The TCP steady-state behavior is referred to as the Additive Increase-Multiplicative Decrease process

no loss:
cwnd = cwnd + 1

loss:
cwnd = cwnd*0.5

AIMD

Phase plot

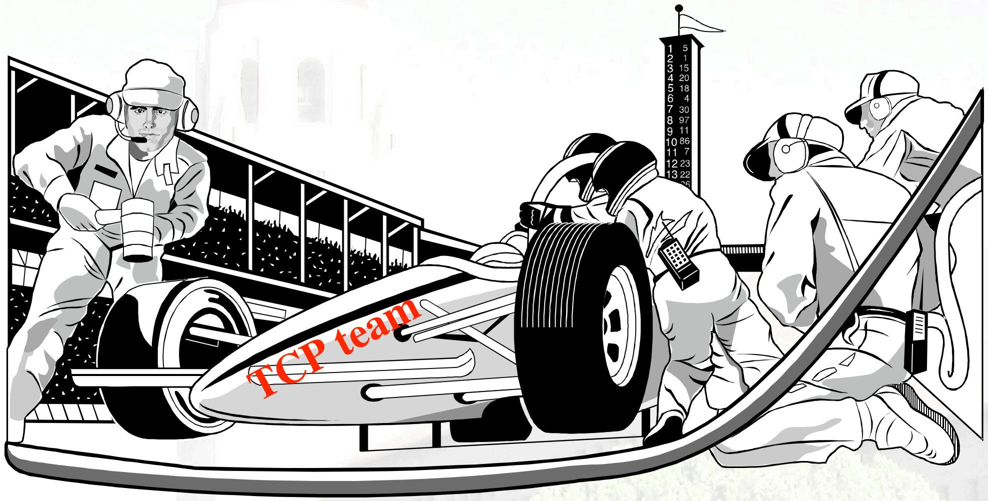


Multiplicative Decrease preserves the fairness because the user's allocation ratio remains the same

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

Tuning stand for TCP
the dark side of speed!



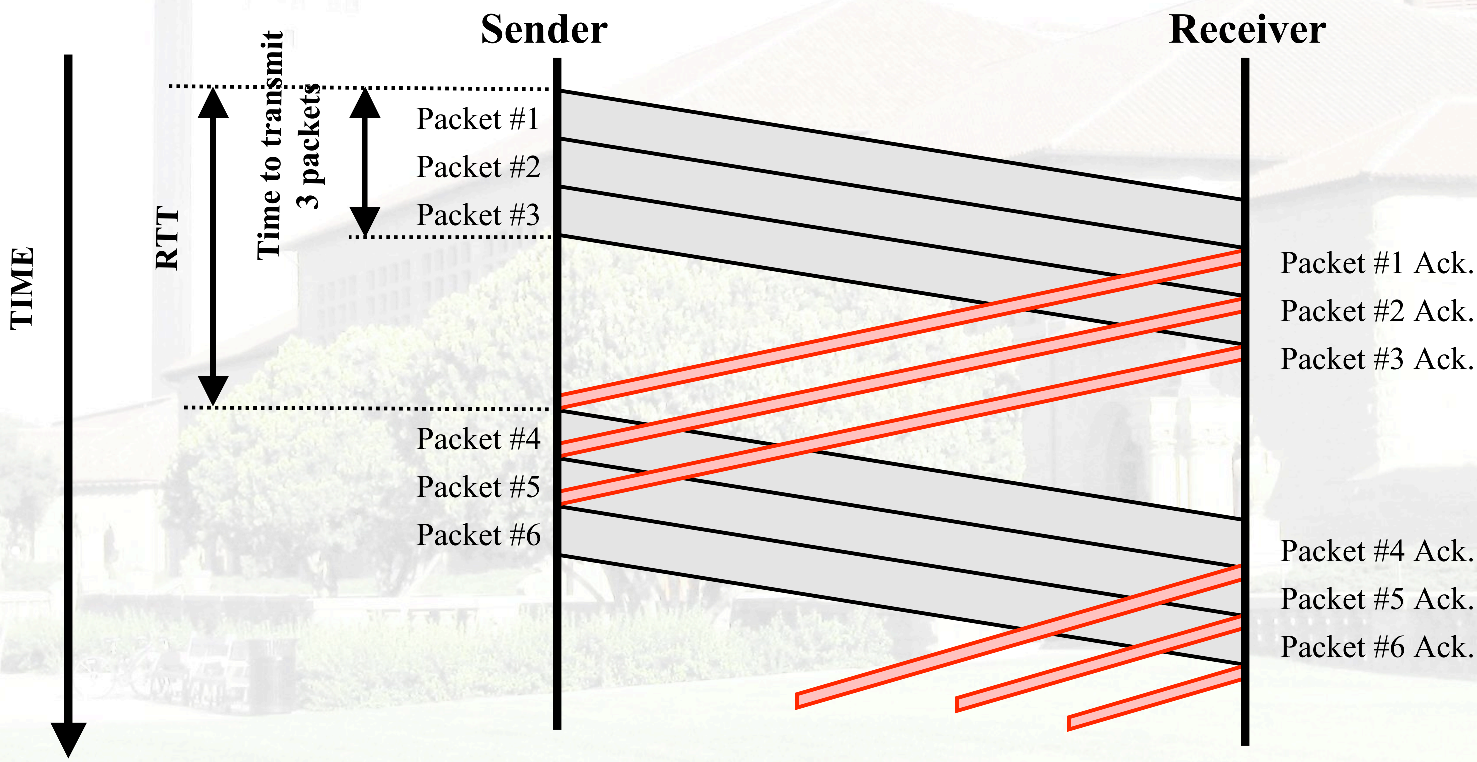
TCP performances depend on

- TCP & network parameters
  - Congestion window size, *ssthresh* (threshold)
  - RTO timeout settings
  - SACKs
  - Packet size
- System parameters
  - TCP and OS buffer size (in comm. subsys., drivers…)

NEED A SPECIALIST!

First problem: window size

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

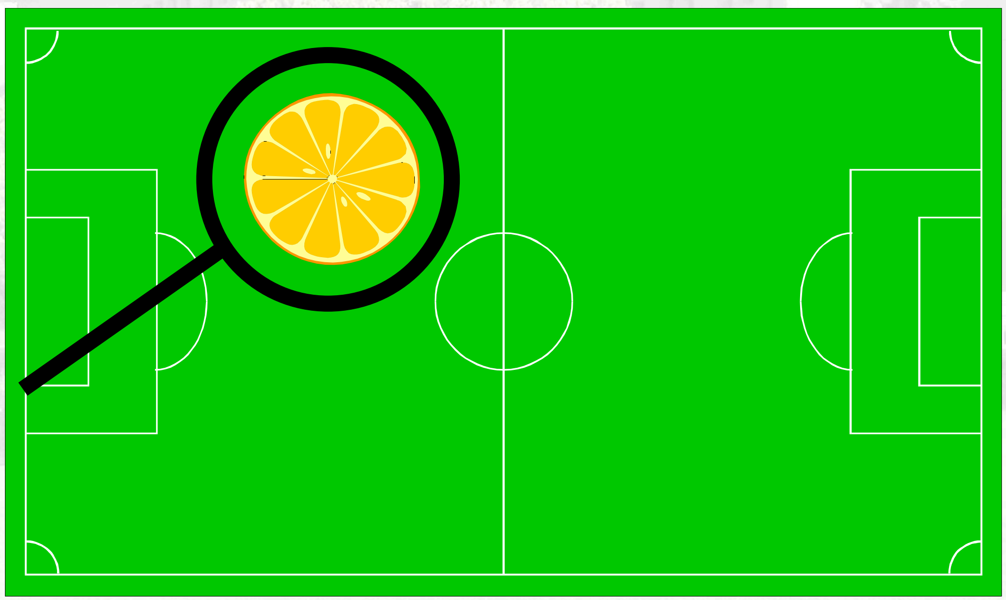


First problem: window size

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

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

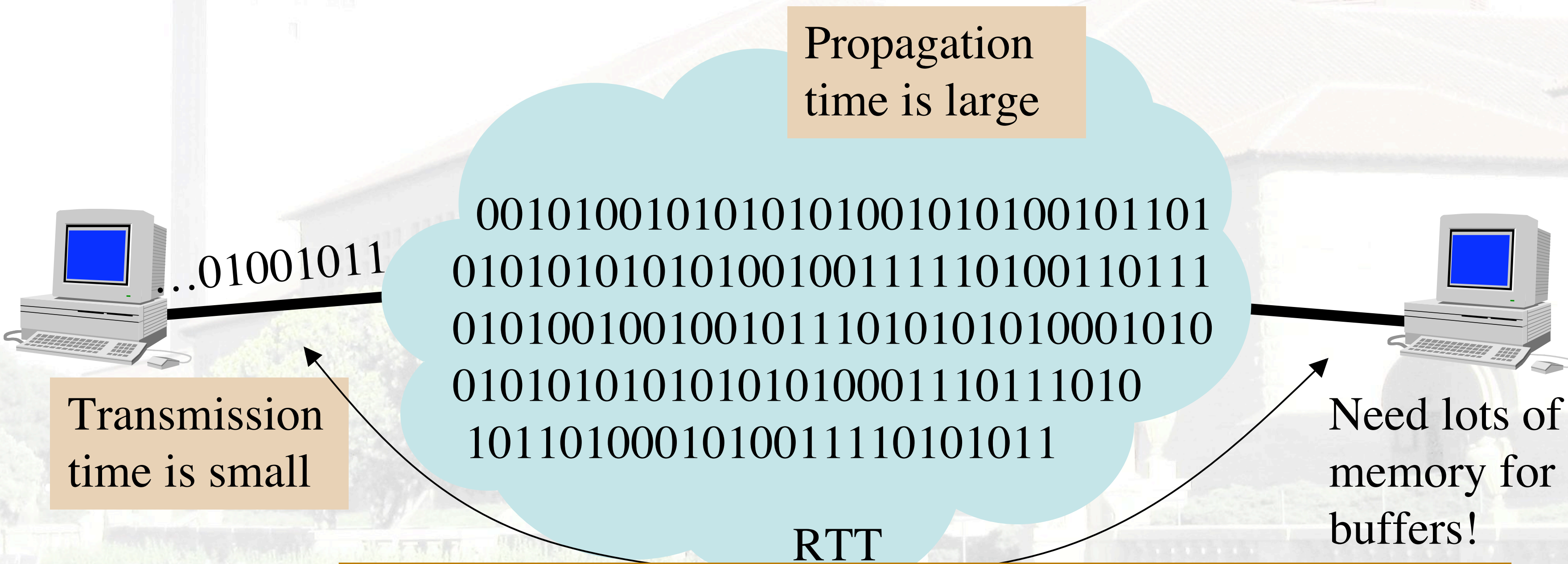
Waiting time



Rule of thumb on LFNs

capacity

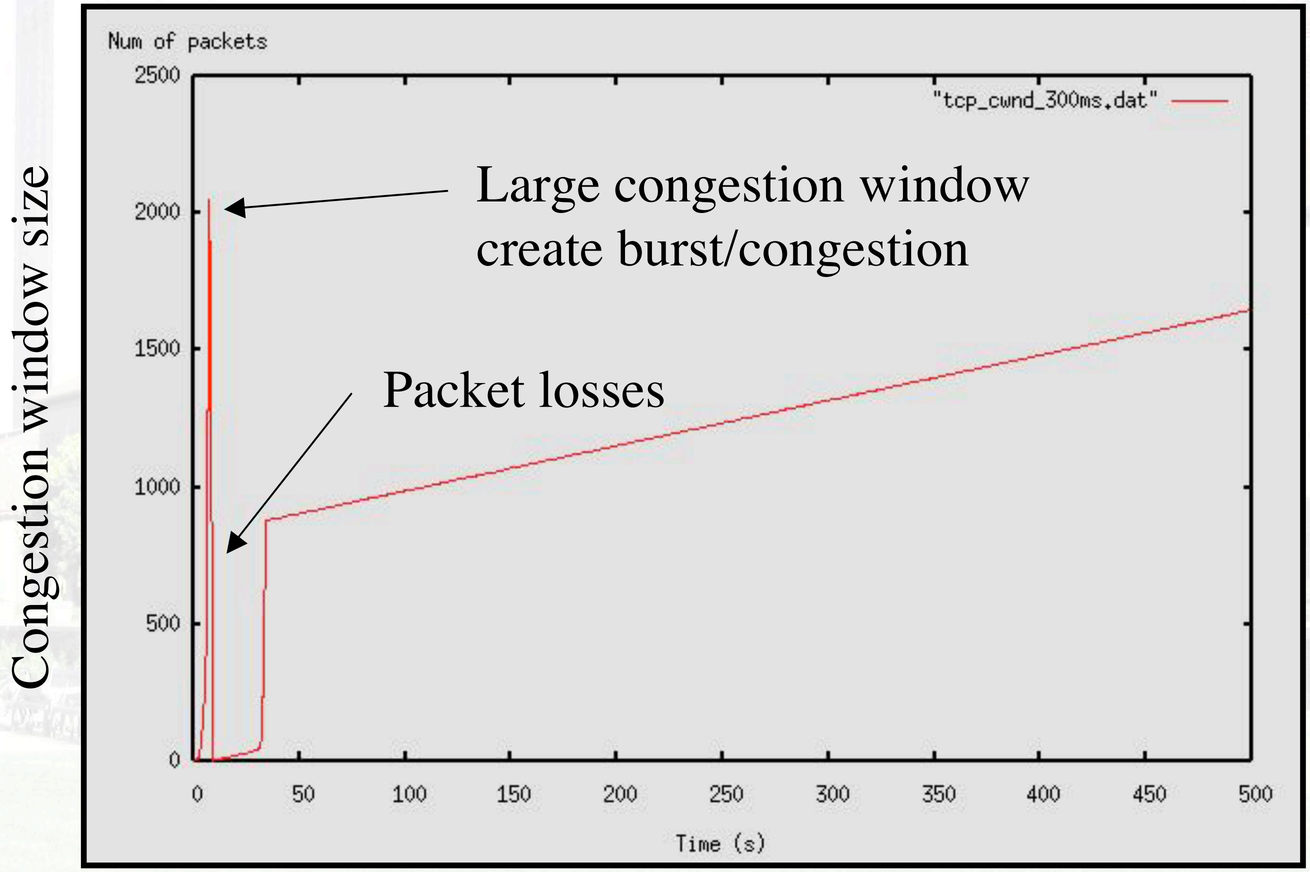
- High-~~speed~~ network



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

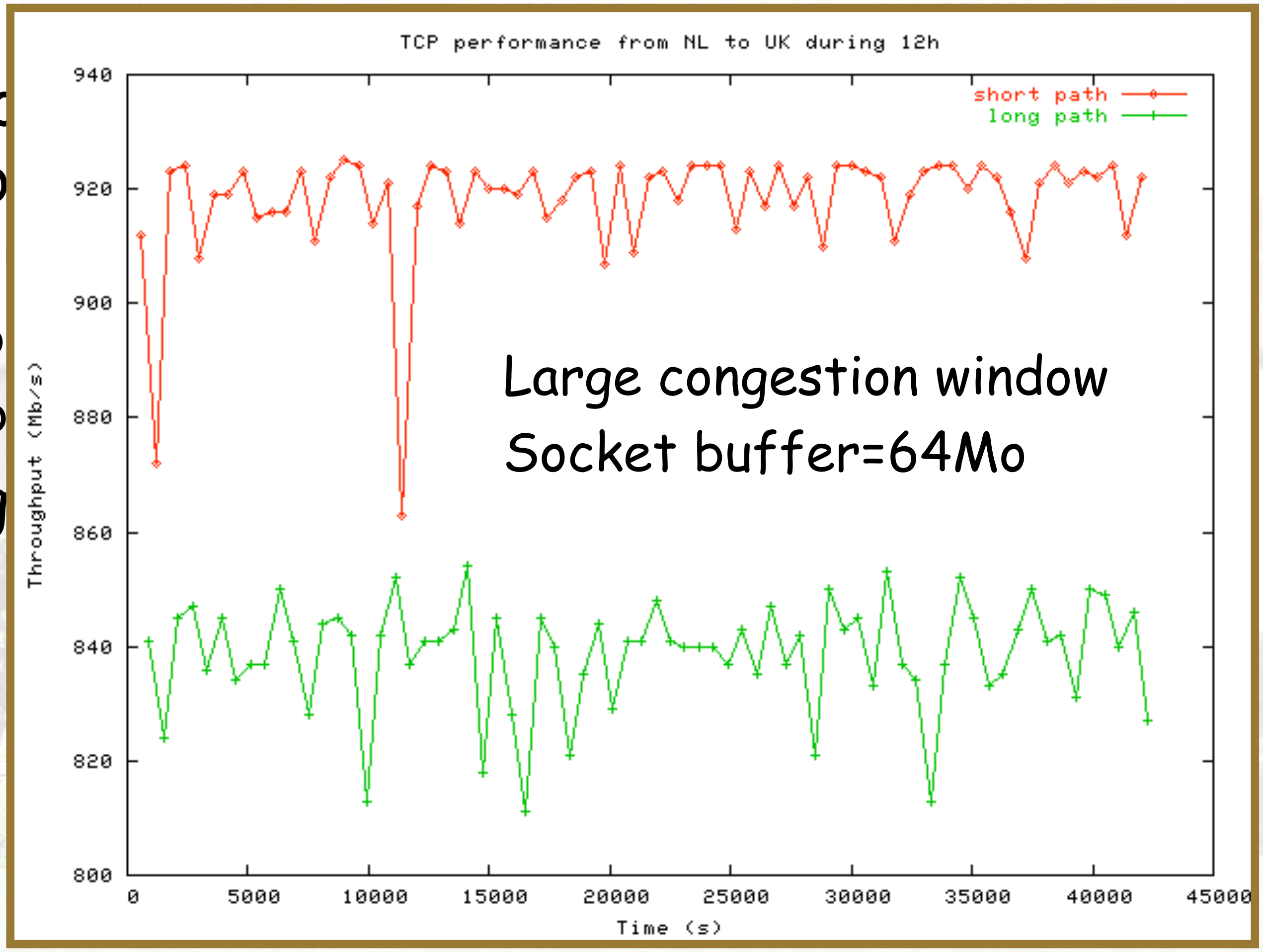


Pushing the limits of TCP

- 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 near 1Gbps if well-tuned

Pushing the limits of TCP

- Standard c
  adequate o
  - Receiver
  - System b
  - Default b
- Will manag

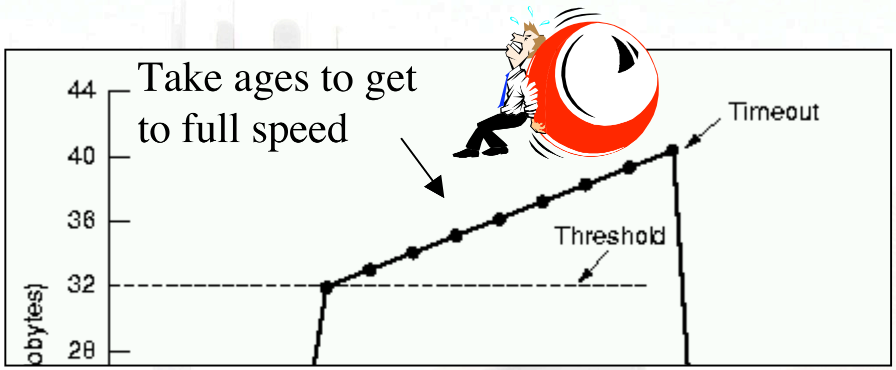


Source: M. Goutelle, GEANT test campaign

Some TCP tuning guides

- http://www.psc.edu/networking/projects/tcptune/
- http://www.web100.org/
- http://rdweb.cns.vt.edu/public/notes/win2k-tcpip.htm
- http://www.sean.de/Solaris/soltune.html
- http://datatag.web.cern.ch/datatag/howto/tcp.html

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



With 100ms of round trip time, a connection needs 203 minutes (3h23) to get 1Gbps 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:

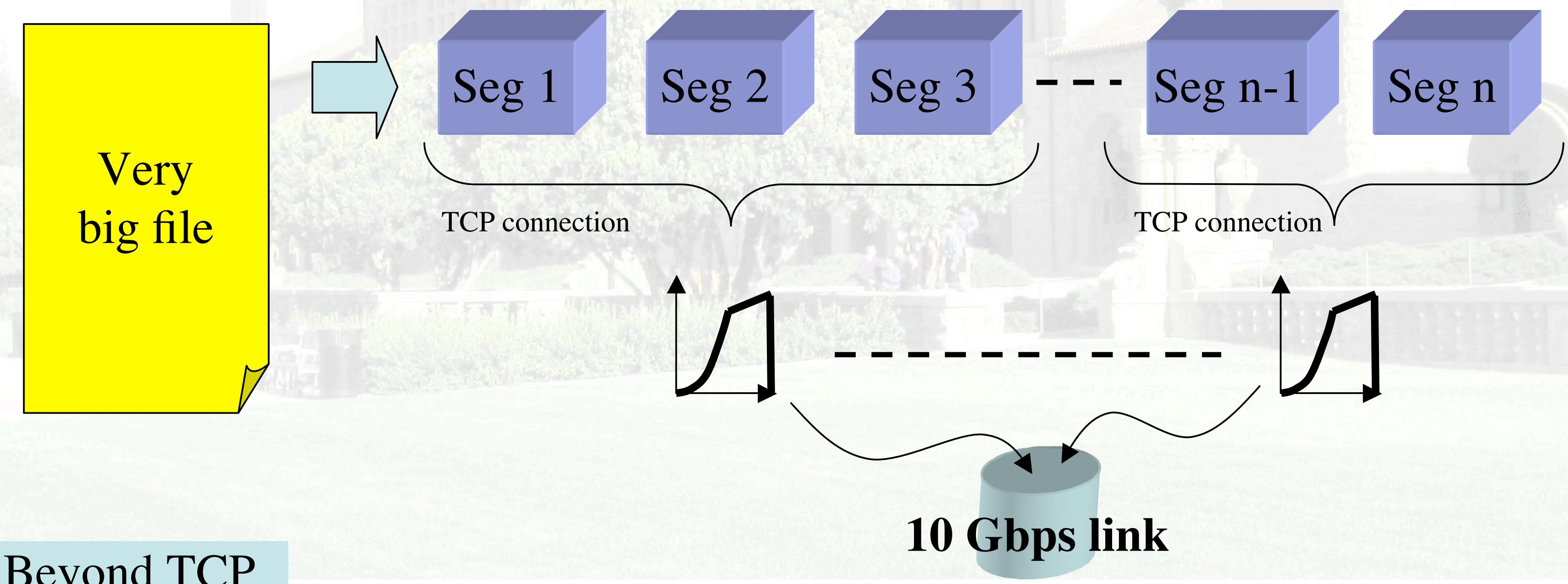
- an average congestion window of 83,333 segments;
- and at most one drop (or mark) every 5,000,000,000 packets (or equivalently, at most one drop every 1 2/3 hours).

This is not realistic.

From S. Floyd

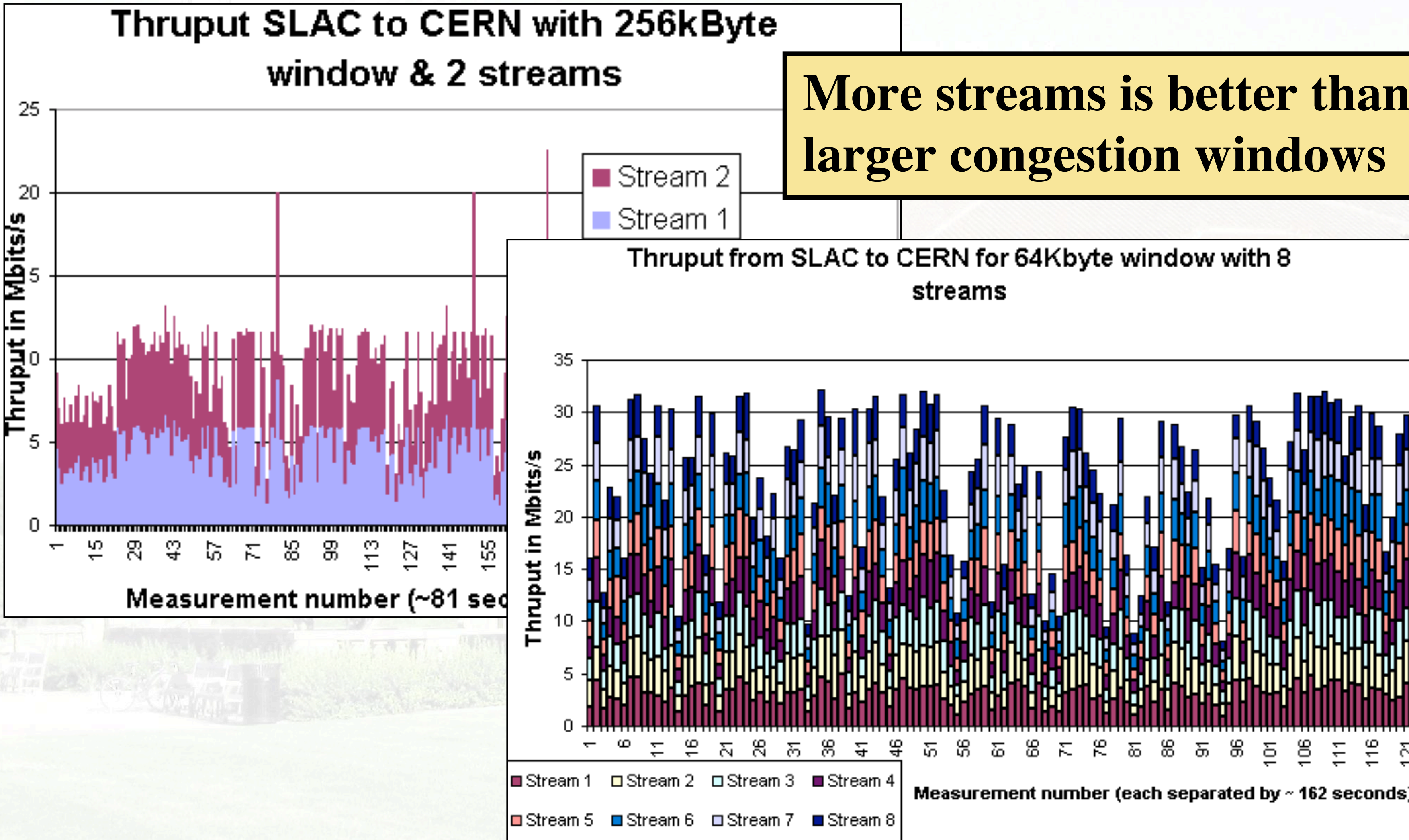
Going faster (cheating?)
n flows is better than 1

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



Some results from IEPM/SLAC

More streams is better than larger congestion windows



http://www-iepm.slac.stanford.edu/monitoring/bulk/window-vs-streams.html

Multiple streams

- No/few modifications to transport protocols (i.e. TCP)
  - Parallel socket libraries
  - GridFTP (http://www.globus.org/datagrid/gridftp.html)
  - bbFTP (http://doc.in2p3.fr/bbftp/)

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

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.

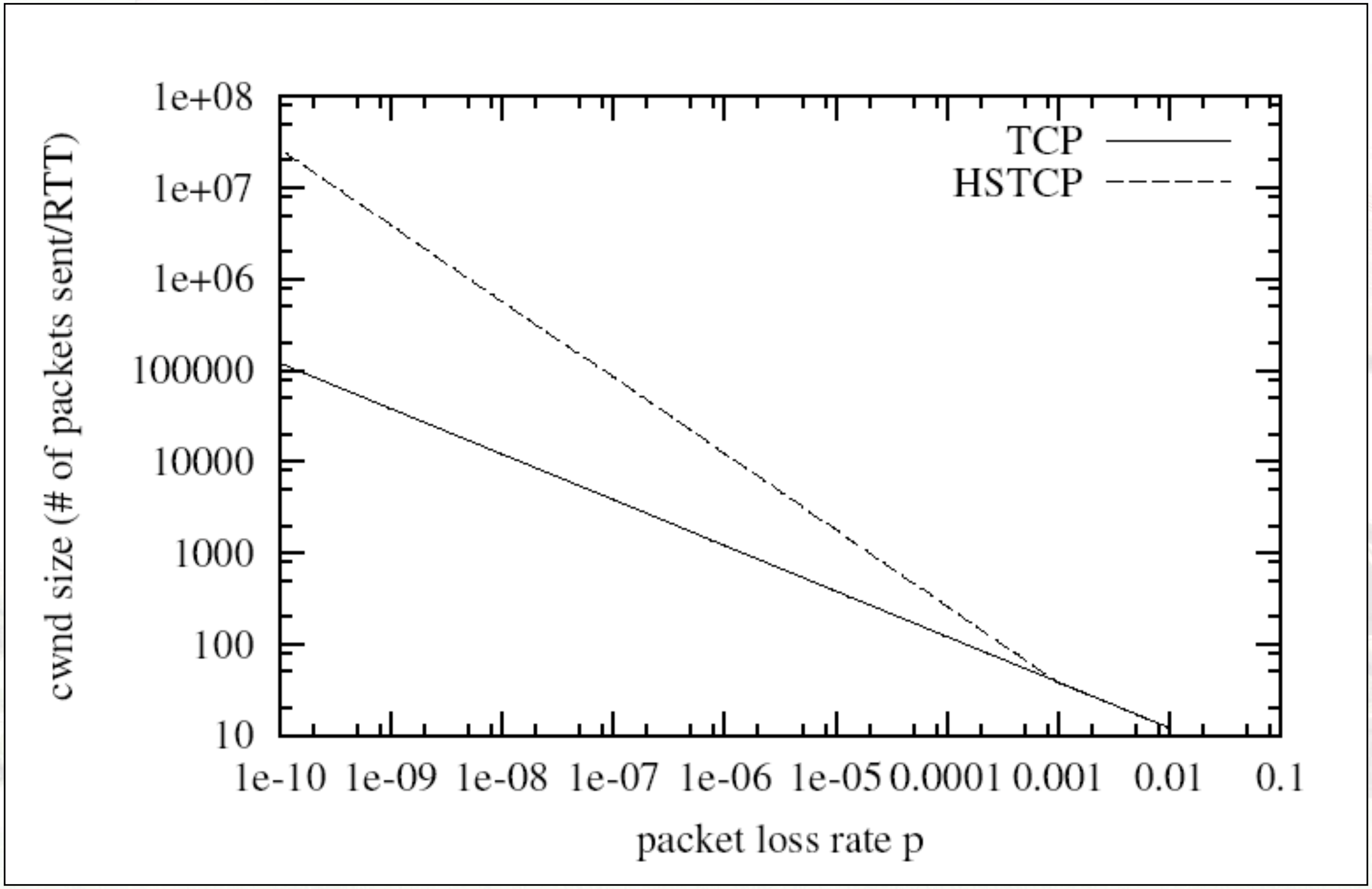
From draft-ietf-tsvwg-highspeed-01.txt

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 function when the current 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



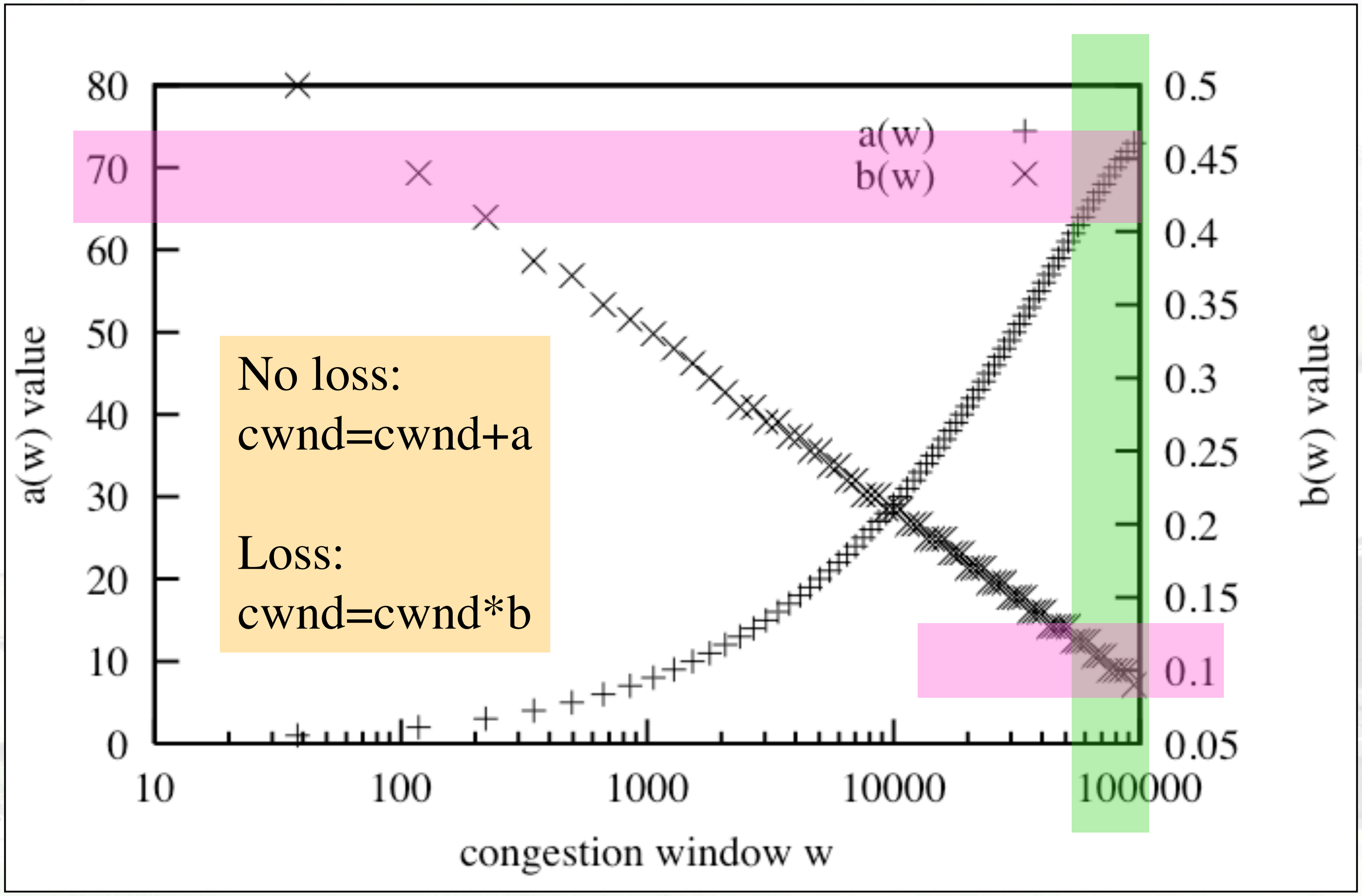
Relation with AIMD

- TCP-AIMD
  - Additive increase: a=1
  - Multiplicative decrease: b=1/2
- HSTCP-AIMD
  - Link a & b to congestion window size
  - a = a(cwnd), b=b(cwnd)

no loss:
cwnd = cwnd + 1

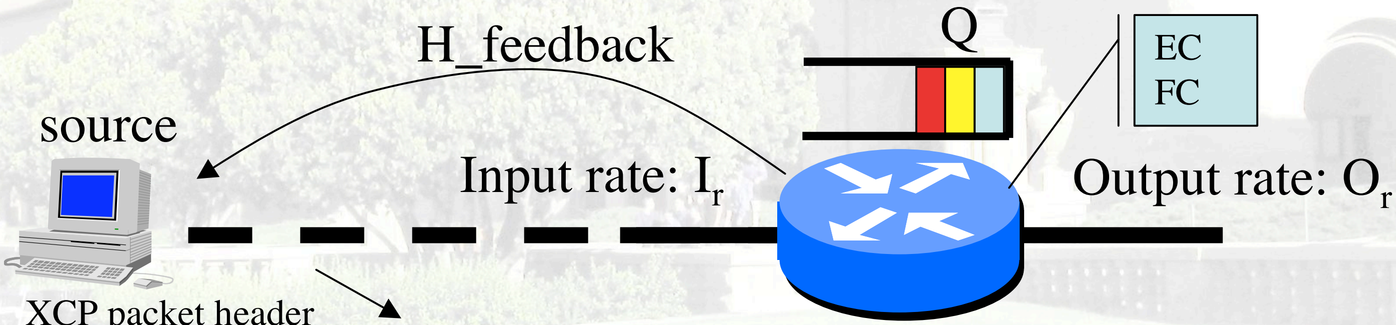
loss:
cwnd = cwnd*0.5

Quick to grab bandwidth, slow to give some back!



XCP [Katabi02]

- XCP is a router-assisted solution, generalized the ECN concepts (FR, TCP-ECN)
- XCP routers can compute the available bandwidth by monitoring the input rate and the output rate
- Feedback is sent back to the source in special fields of the packet header

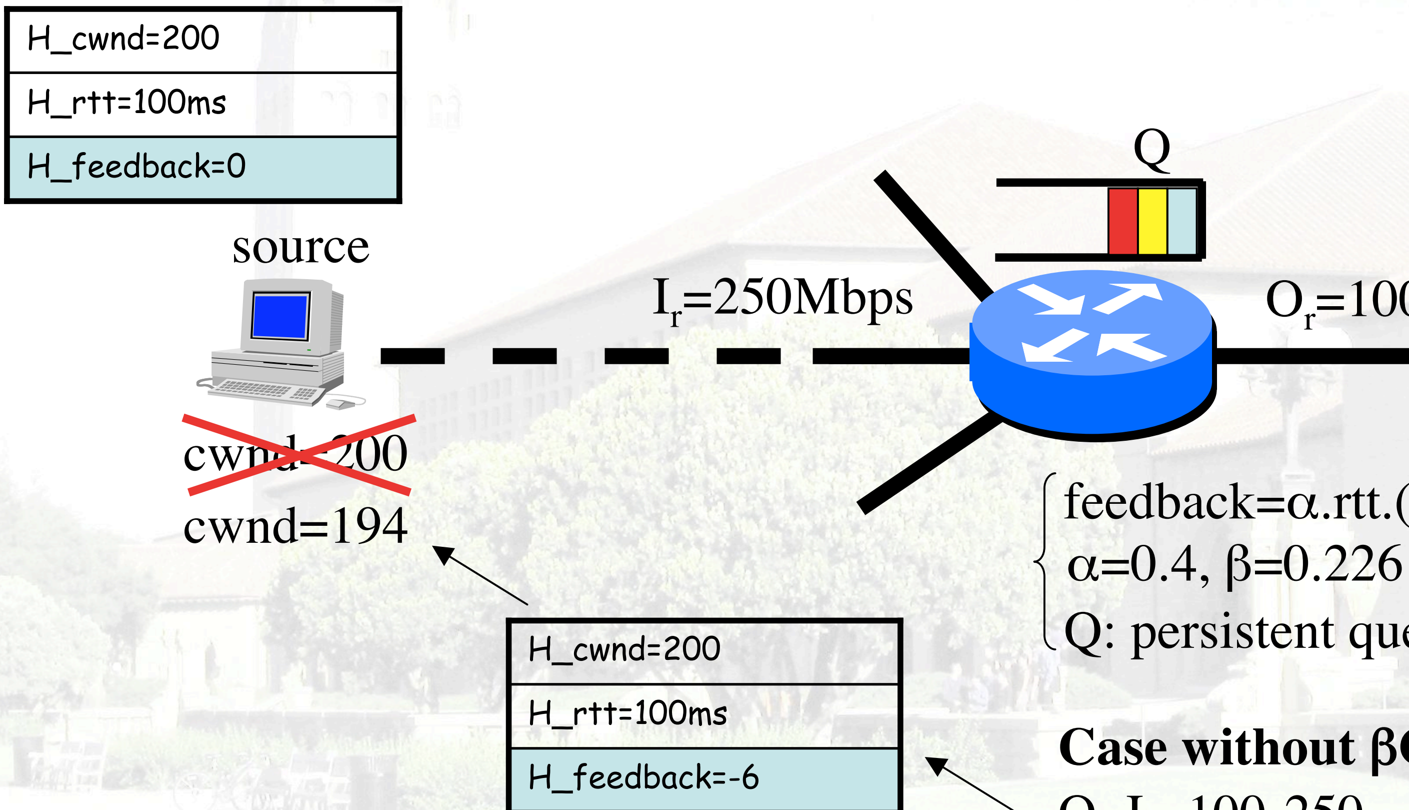


H_cwnd (set to the sender's current cwnd)
H_rtt (set to sender's RTT estimate)
H_feedback (initialized to sender's demands)

$$\begin{cases} \text{feedback} = \alpha.\text{rtt}.(O_r - I_r) - \beta Q \\ \alpha = 0.4,\ \beta = 0.226 \\ Q\text{: persistent queue size} \end{cases}$$

XCP in action

Feedback value represents a window increment/decrement



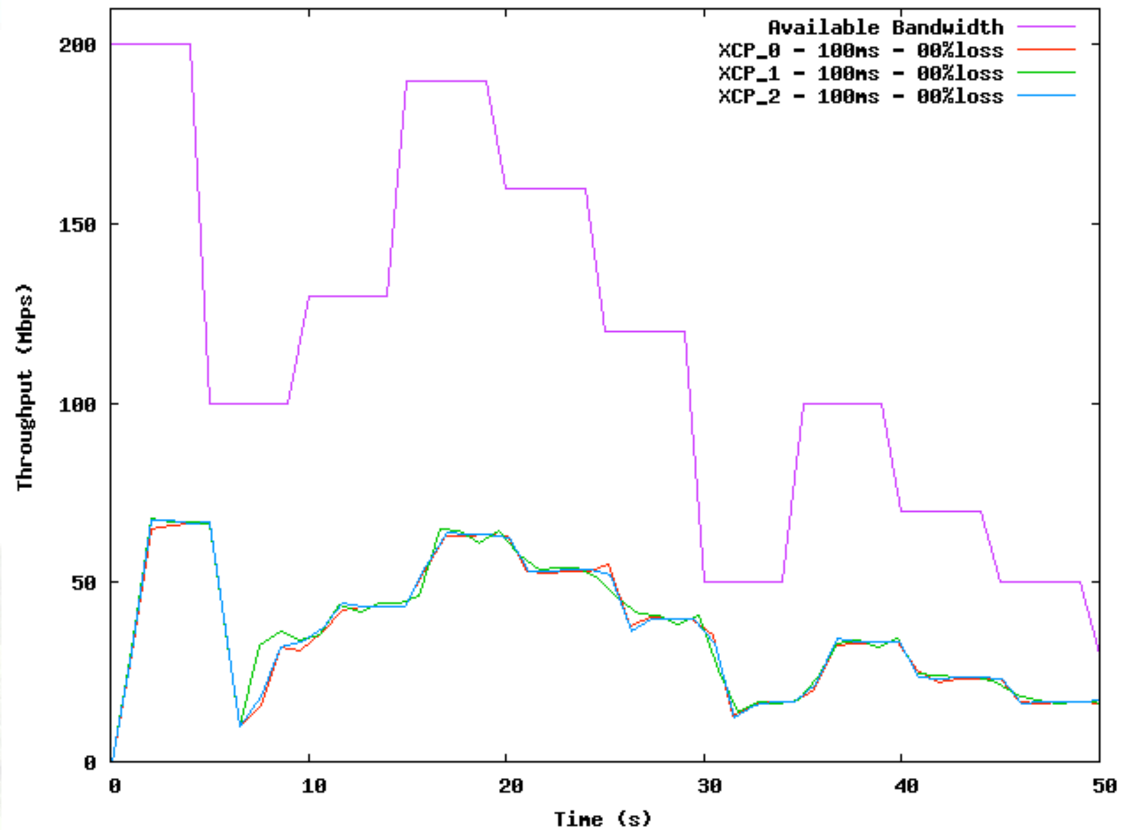
feedback=α.rtt.(O_r-I_r)-βQ
α=0.4, β=0.226
Q: persistent queue size

Case without βQ contribution

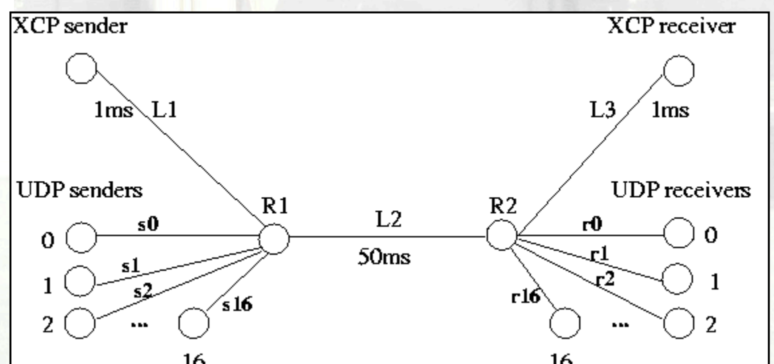
O_r-I_r=100-250=-150
feedback=-6

XCP

Variable bandwidth environments

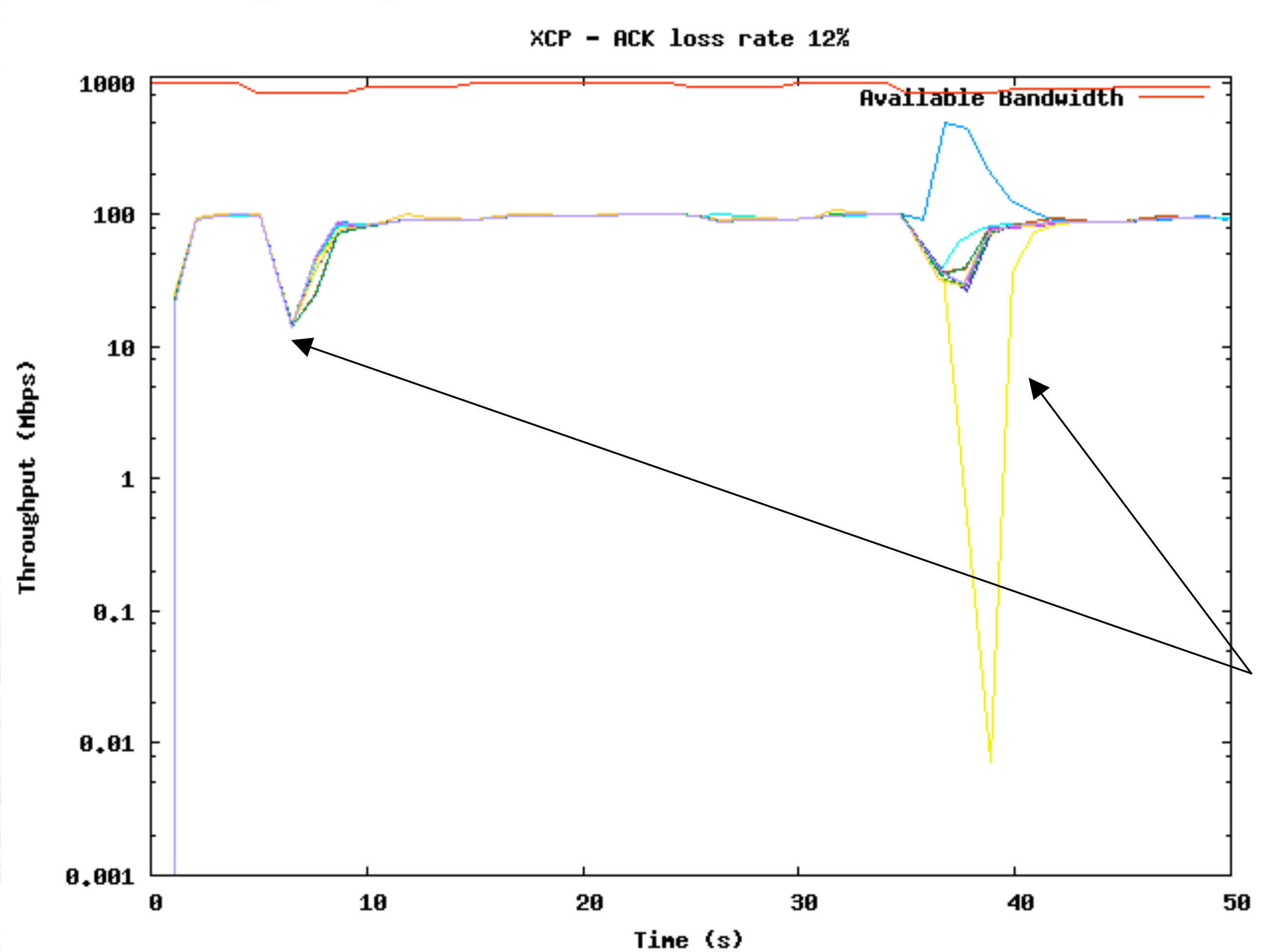


Good fairness and stability even in variable bandwidth environments



XCP-r [Pacheco&Pham05]

A more robust version of XCP

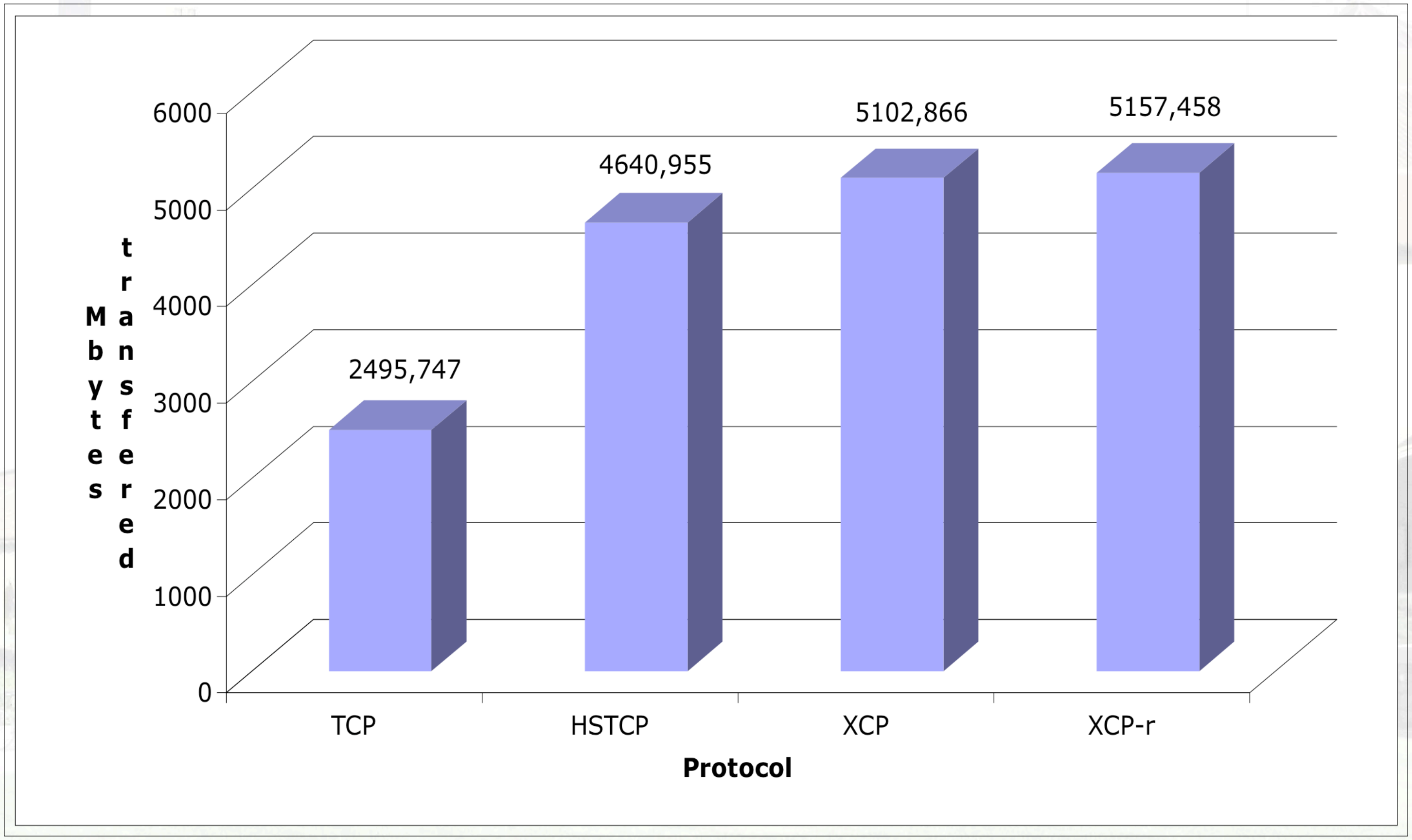


10 flows sharing a 1Gbps link

Fast recovery after the timeouts and better fairness level

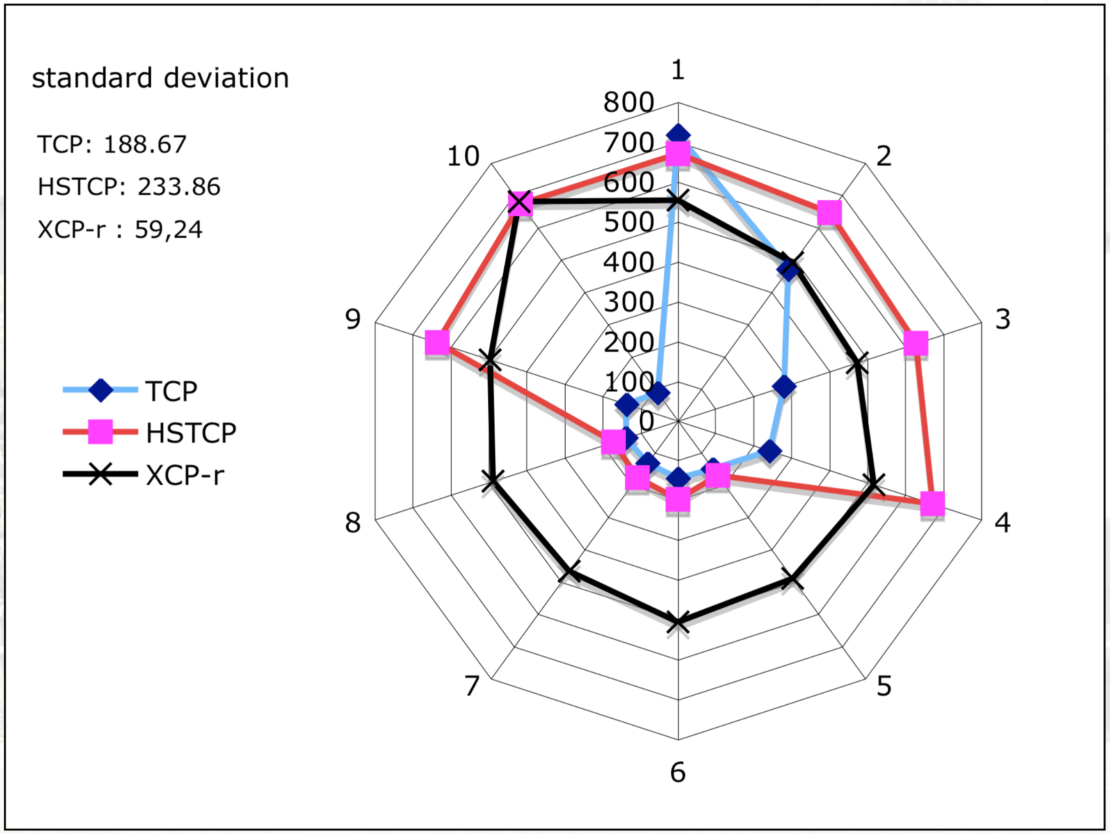
XCP-r performance

Amount of data transfered in 50s, 10 flows, 1Gbps link, 200ms RTT



XCP-r fairness

TCP and HSTCP are not really fair...



Nothing is perfect :-(

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

Where to find the new protocols?

- HSTCP
  - http://www.icir.org/floyd/hstcp.html
- STCP on Linux 2.4.19
  - http://www-lce.eng.cam.ac.uk/~ctk21/scalable/
- FAST
  - http://netlab.caltech.edu/FAST/
- XCP
  - http://www.ana.lcs.mit.edu/dina/XCP/
  - http://www.isi.edu/isi-xcp/#software

Web100 project

- www.web100.org
- « The Web100 project will provide the software and tools necessary for end-hosts to automatically and transparently achieve high bandwidth data rates (100 Mbps) over the high performance research networks »
- Actually it's not limited to 100Mbps!
- Recommended solution for end-users to deploy and test high-speed transport solutions