A report on a few steps in the evolution of congestion control

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Topics:

- High-speed TCP.
- Faster Start-up?
- AQM: Adaptive RED
- Evaluation of AQM mechanisms.
- A proposal about models and simulations
- Other open questions?

Architectural sub-themes:

- A goal of incremental deployment in the current Internet.
- Steps must go in the fundamantally correct, long-term direction, not be short-term hacks.
- Robustness in heterogeneous environments valued over efficiency of performance in well-defined environments.
- A skepticism towards simple models.
- Learning from actual deployment is an invaluable step.
- The Internet will continue to be decentralized and fast-changing.

HighSpeed TCP:

Joint work with Sylvia Ratnasamy and Scott Shenker.

Additional investigations with Evandro de Souza and Deb Agarwal.

URLs:
http://www.icir.org/floyd/papers/draft-floyd-tcp-highspeed-00c.txt
http://www.icir.org/floyd/papers/draft-floyd-tcp-slowstart-00b.txt

HighSpeed TCP: The problem.

- TCP's average congestion window is roughly $1.2/\sqrt{p}$ packets.

- Maintaining an average cwnd of at least $1.2 * 10^k$ packets requires a packet loss/corruption rate of at most 10^{-2k}.

- Given 1500-byte packets and a 100 ms RTT, filling a 10 Gbps pipe would correspond to a congestion window of $W = 83,333$ packets.
  - At least 1.6 hours between packet drops.

- We can do better, even with only the current feedback from routers.

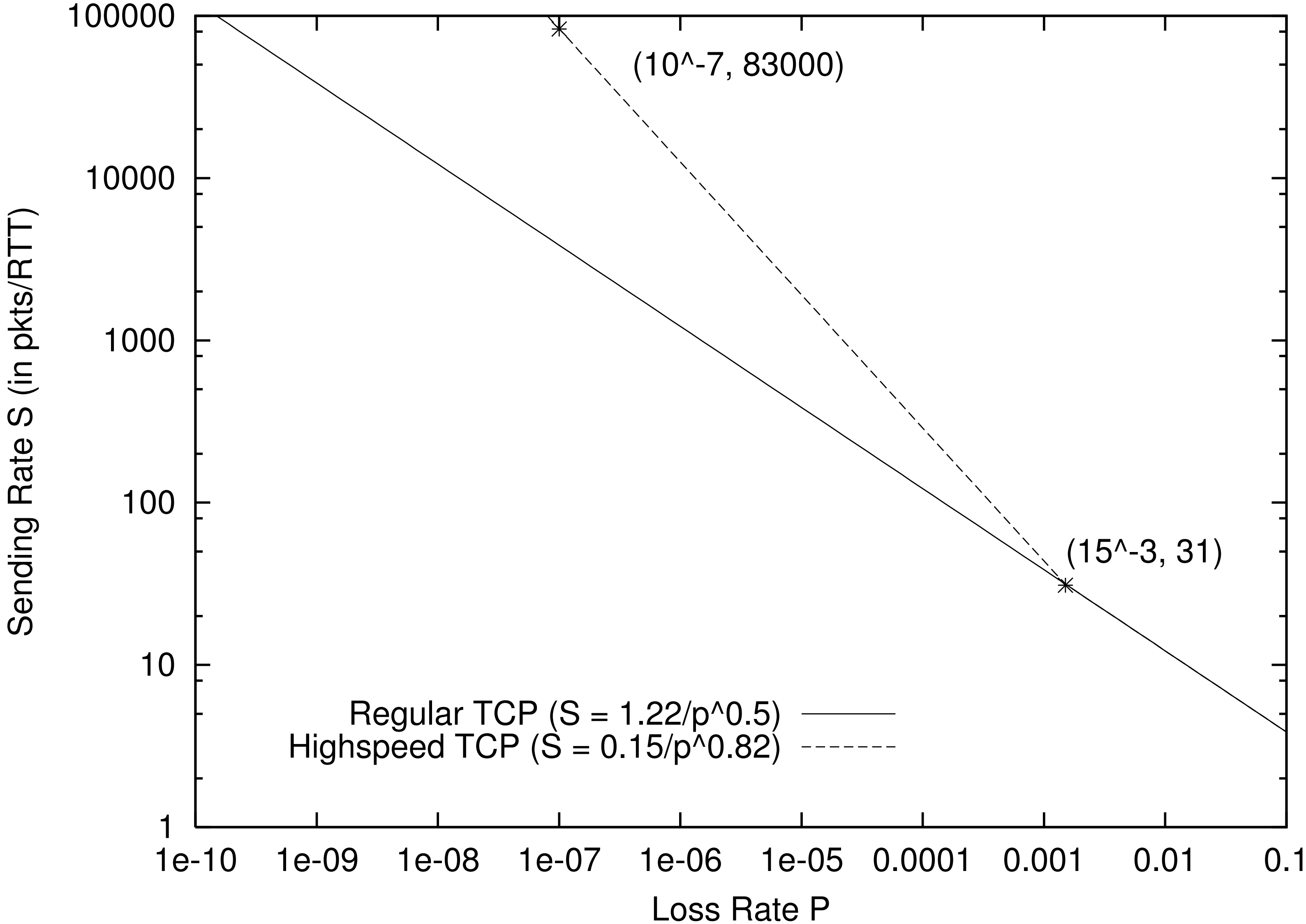
HighSpeed TCP: Is this a pressing problem?

- Nope. In practice, users do one of the following:
  - Open up N parallel TCP connections; or
  - Use MulTCP (roughly like an aggregate of N virtual TCP connections).

- However, we think it is possible to do much better, with:
  - Better flexibility (no N to configure);
  - Better scaling;
  - Better slow-start behavior;
  - Competing more fairly with current TCP

(for environments where TCP is able to use the available bandwidth).

HighSpeed TCP: use a modified response function.



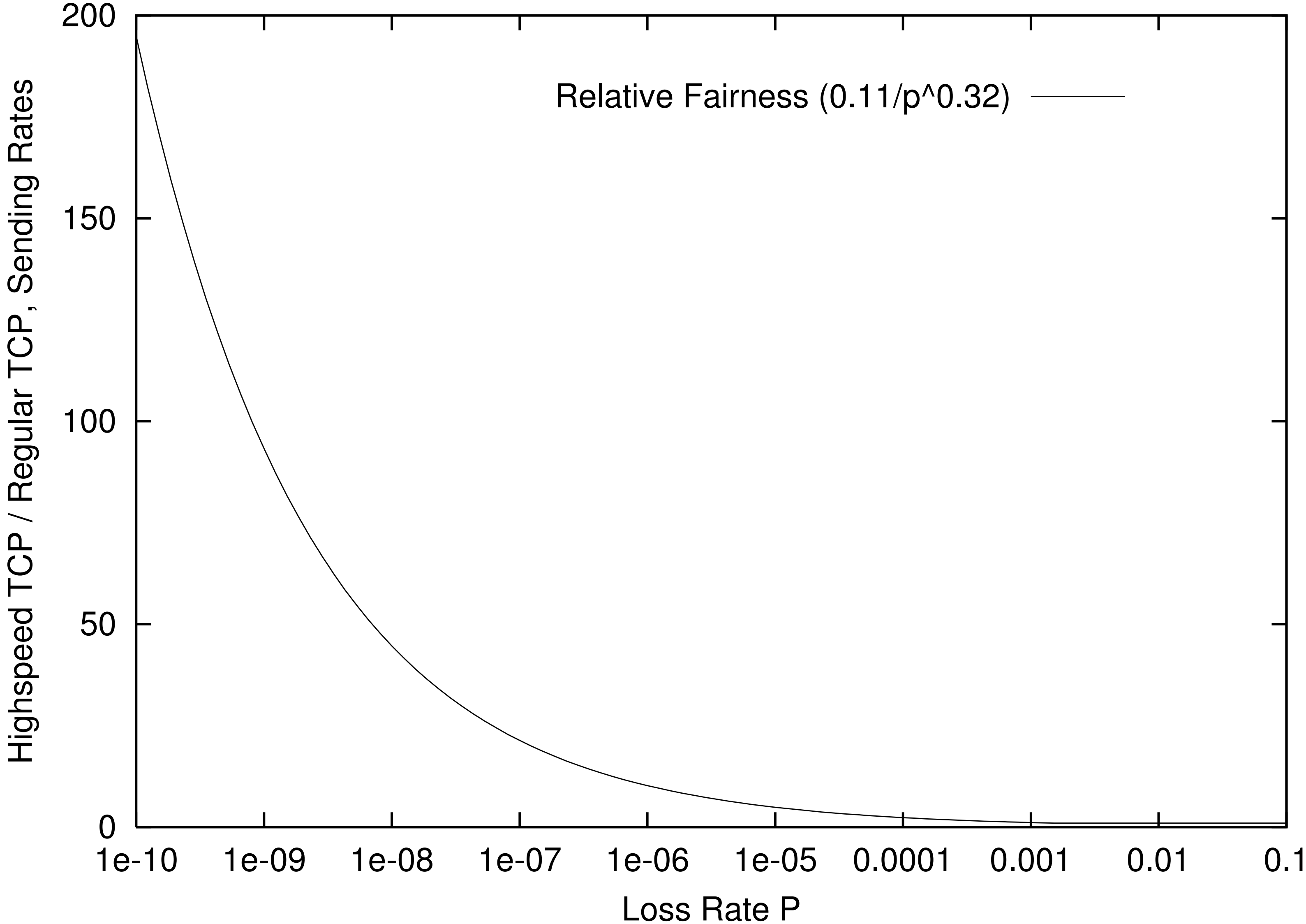
HighSpeed TCP: Simulations in NS.

- ./test-all-tcpHighspeed in tcl/test.

- The parameters specifying the response function:
  - Agent/TCP set low_window_ 31
  - Agent/TCP set high_window_ 83000
  - Agent/TCP set high_p_ 0.0000001

- The parameter specifying the decrease function at high_p_:
  - Agent/TCP set high_decrease_ 0.1

HighSpeed TCP: Relative fairness.



HighSpeed TCP: modifying slow-start:

- Slow-starting up to a window of 83,000 packets doesn't work well.
  - Tens of thousands of packets dropped from one window of data.
  - Slow recovery for the TCP connection.

- The answer:
  - Agent/TCP set max_ssthresh_ N
  - During the initial slow-start, increase the congestion window by at most N packets in one RTT.

Faster Start-up?

From a proposal by Amit Jain.

No URL yet.

Faster Start-up: Larger Initial Sending Rate

- An IP option in the SYN packet gives the sender's desired initial sending rate.
  - Routers on the path decrement a counter,
  - and decrease the allowed initial sending rate, if necessary.

- If all routers on the path participated:
  - The receiver tells the sender the allowed initial sending rate in the SYN/ACK packet, in the transport header.

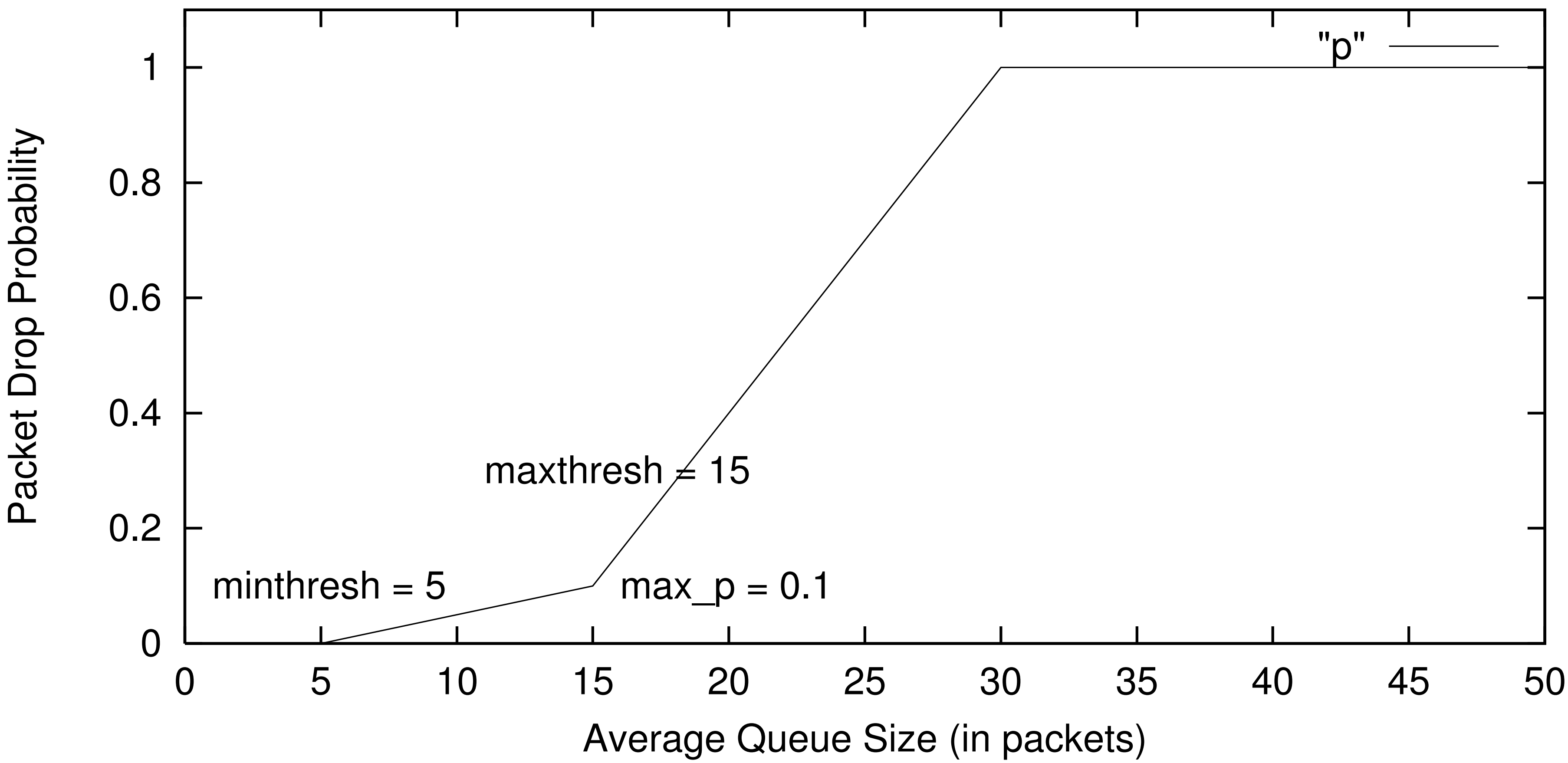
- This is from a proposal by Amit Jain (from Netscaler).

Adaptive RED

Joint work with Ramakrishna Gummadi and Scott Shenker.

URL (with simulation scripts):
http://www.cs.berkeley.edu/ ramki/adaptiveRED/

The One-Page Primer on RED:



For the average queue size:

$$ave_q \leftarrow (1 - w_q) avg_q + w_q \, q$$

Adaptive RED: adapting max_p

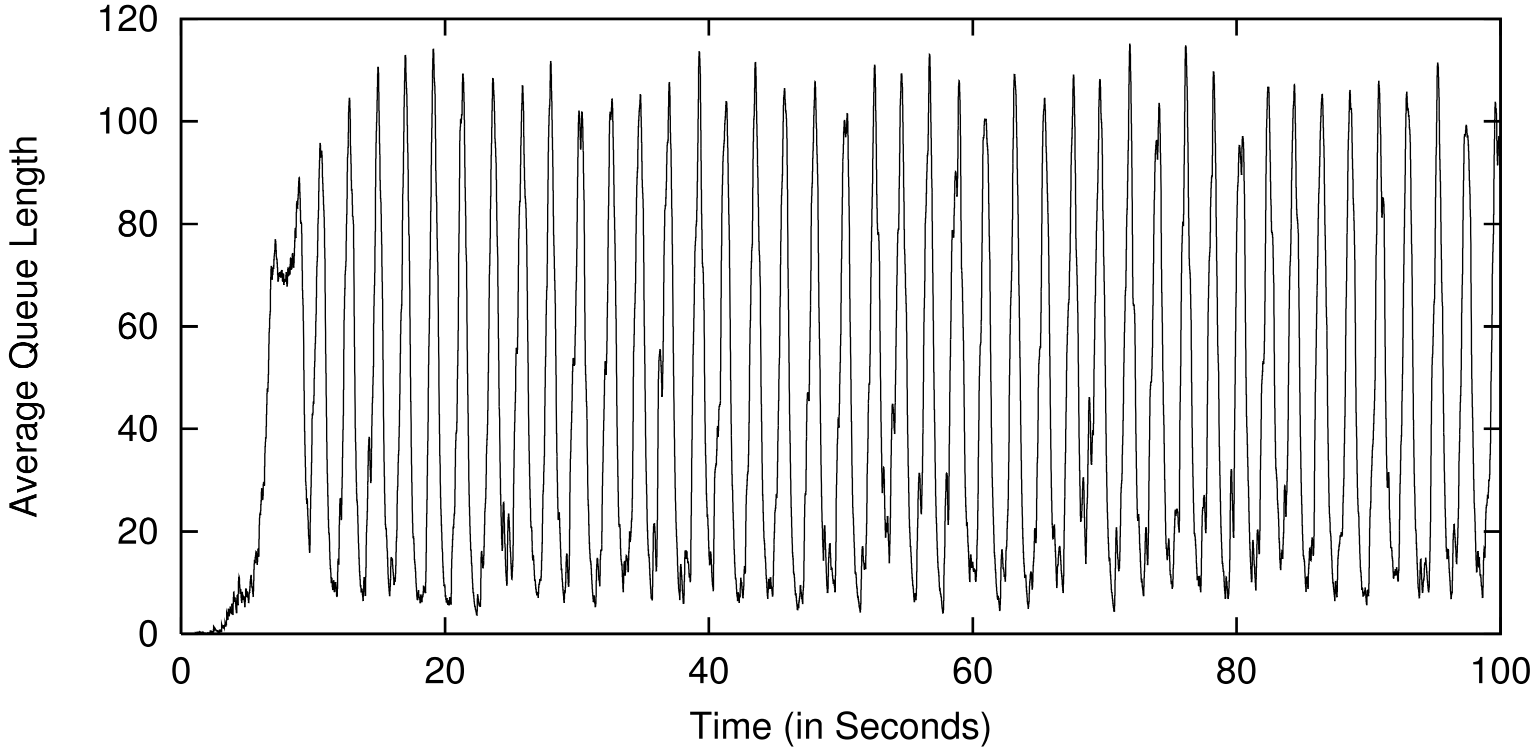
- The original Adaptive RED proposal is from Feng et al., 1997.
  - Adjusts max_p to keep the average queue between min_{th} and max_{th}.

- We have a new implementation of Adaptive RED, adapting max_p.

- Automatic setting of w_q as a function of the link bandwidth.

- Automatic setting of min_{th} and max_{th} as a function of the target queue size.

Simulation with malignant oscillations:

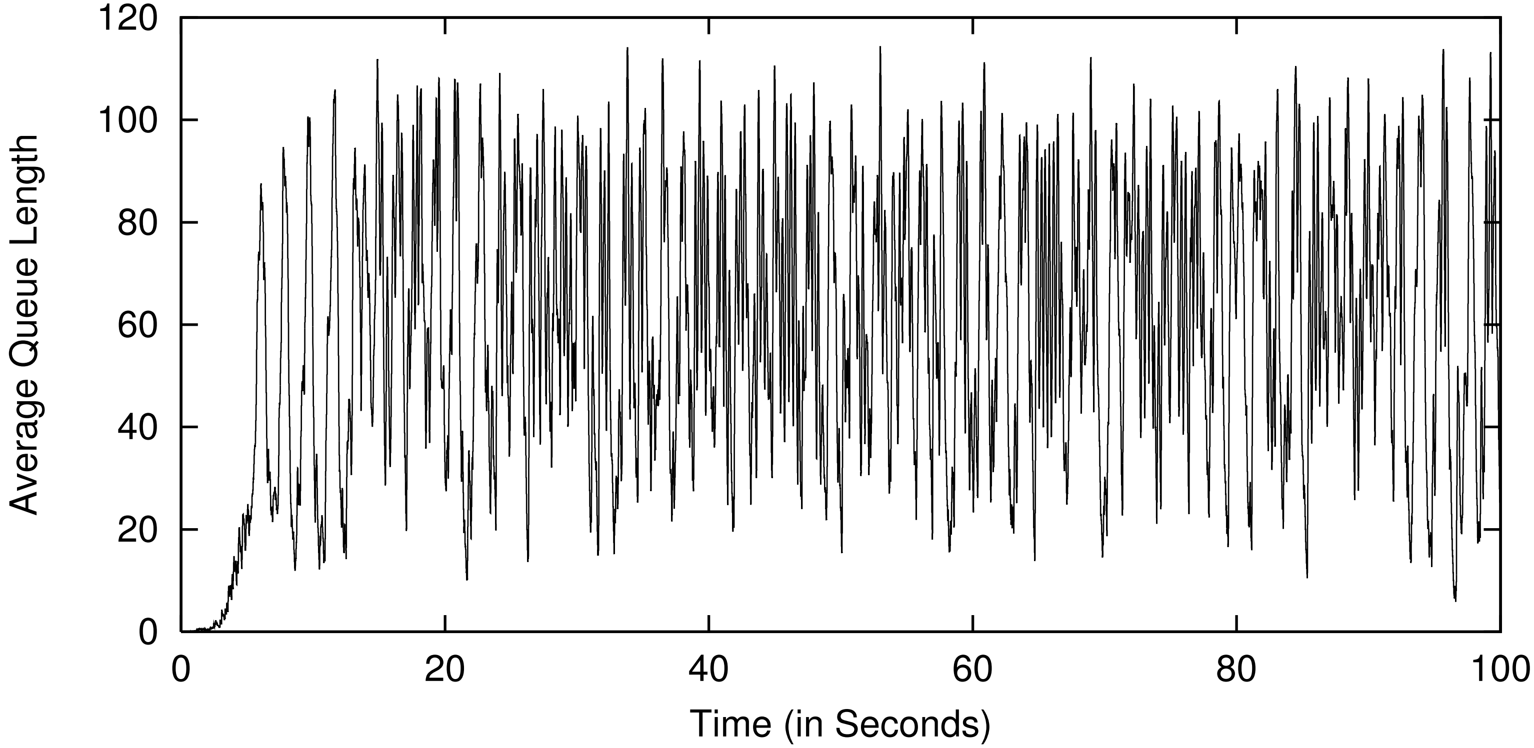


RED, one-way long-lived traffic, w_q=0.002.

No web traffic, no reverse-path traffic.

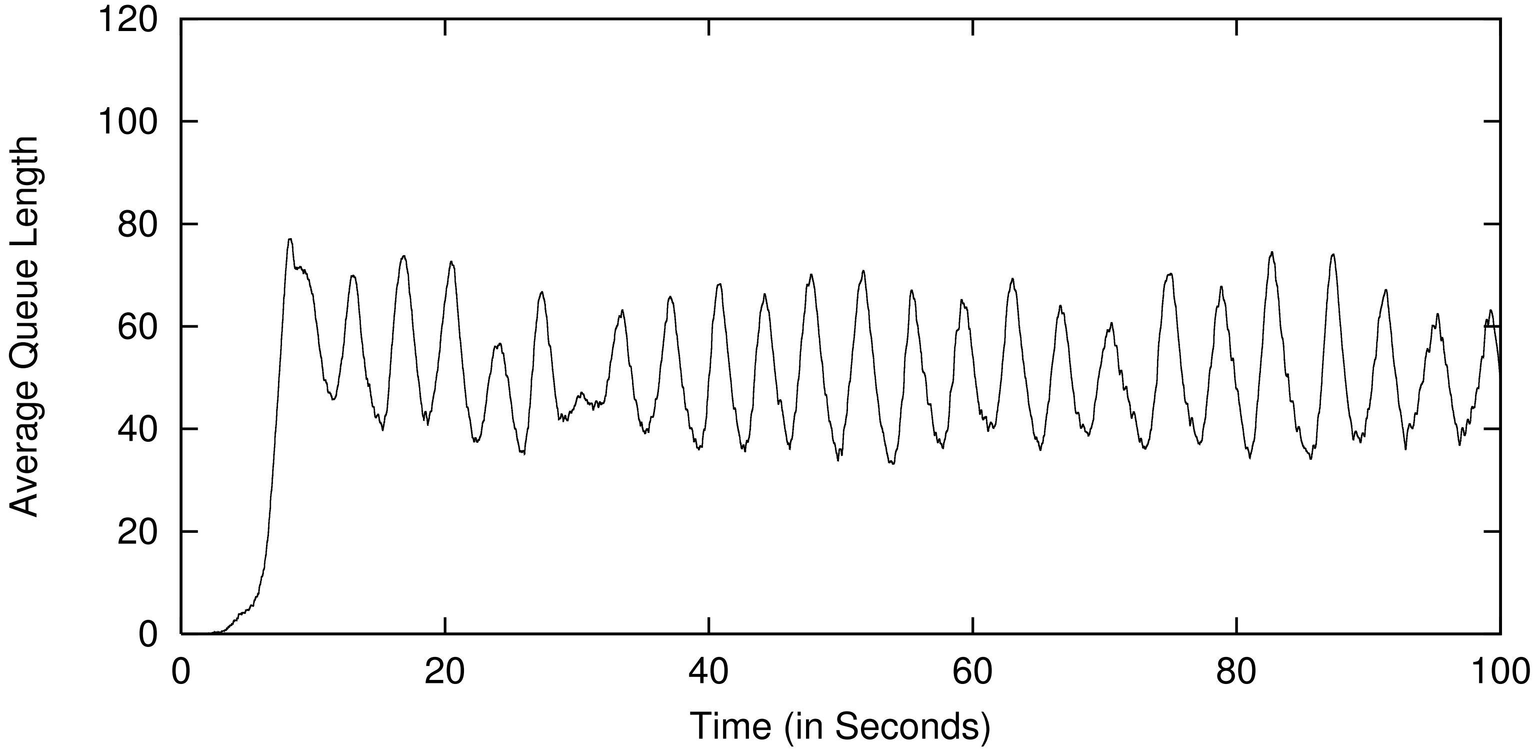
15 Mbps link, 250 ms round-trip time.

90% link utilization.



RED, mostly one-way long-lived traffic, w_q=0.002.
Web traffic and reverse-path traffic added.
15 Mbps link, 250 ms round-trip time.

Simulation with benign oscillations:

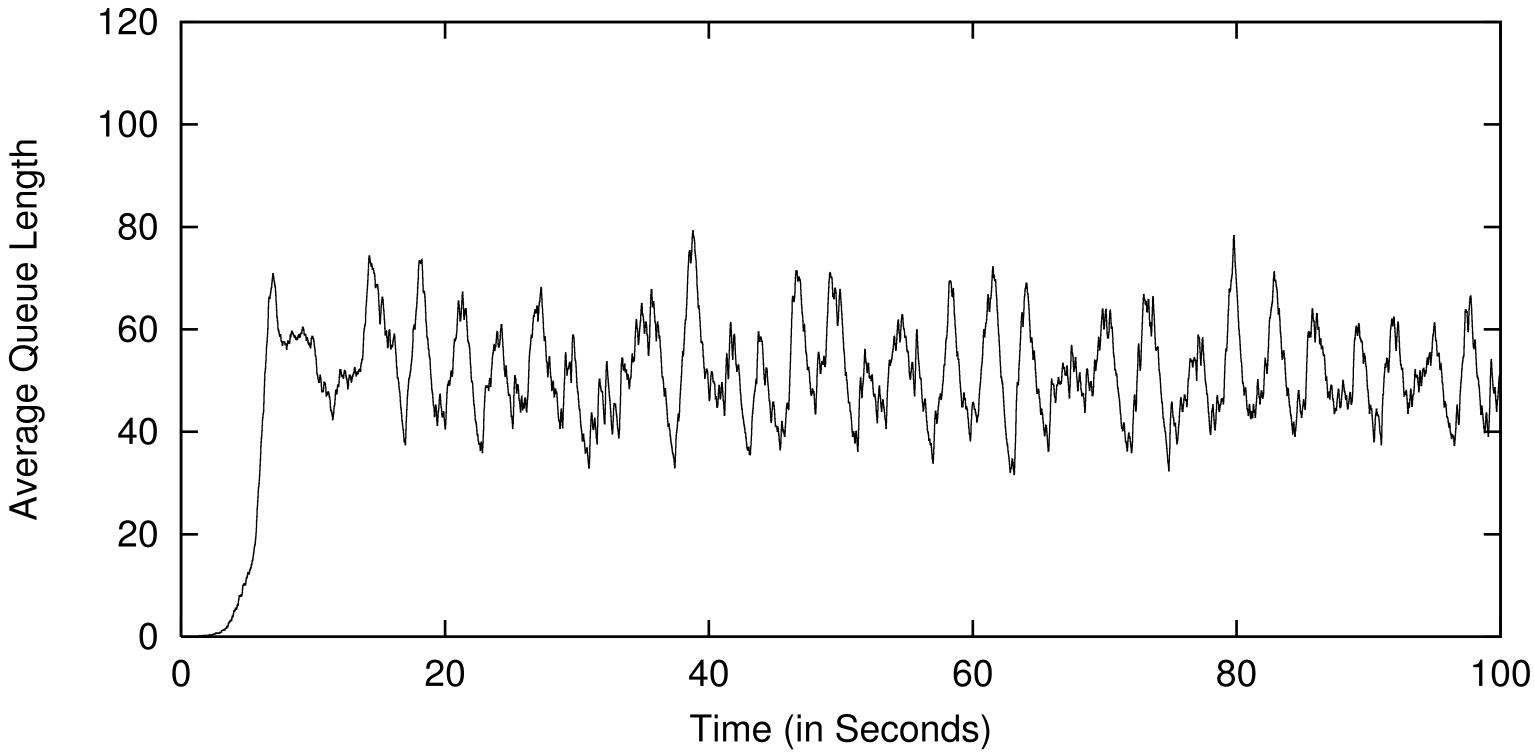


Adaptive RED, one-way long-lived traffic, w_q=0.00027.

No web traffic, no reverse-path traffic.

15 Mbps link, 250 ms round-trip time.

96.8% link utilization.



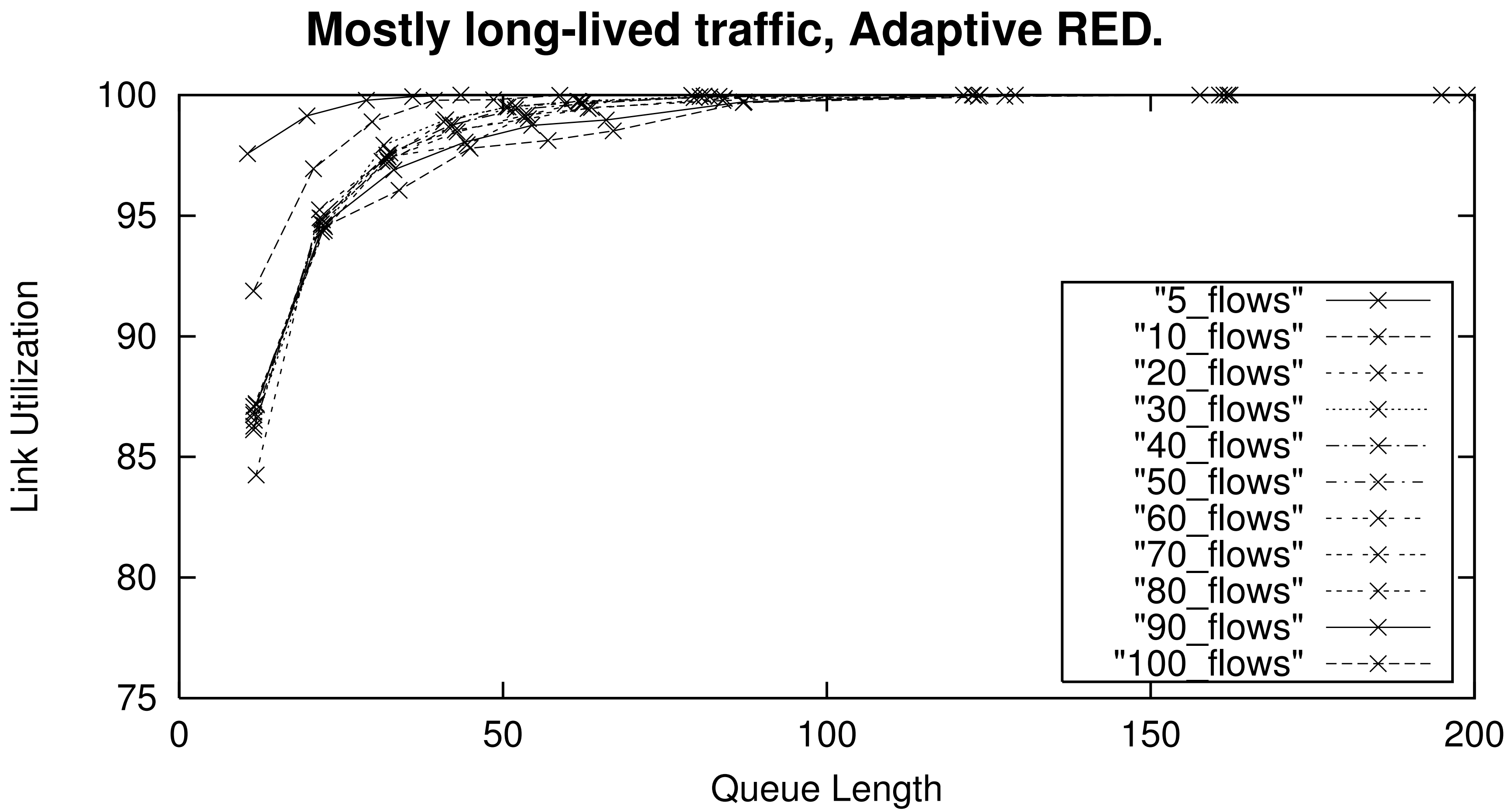
Adaptive RED, mostly one-way long-lived traffic, w_q=0.00027.
Web traffic and reverse-path traffic added.
15 Mbps link, 250 ms round-trip time.

The optimal average queue size?

- It depends on the desired tradeoff at the router between high utilization and low delay.

- It is heavily affected by traffic, topology, etc.

- We don't know the optimal average queue size.



Traffic includes some web mice, and reverse-path traffic.

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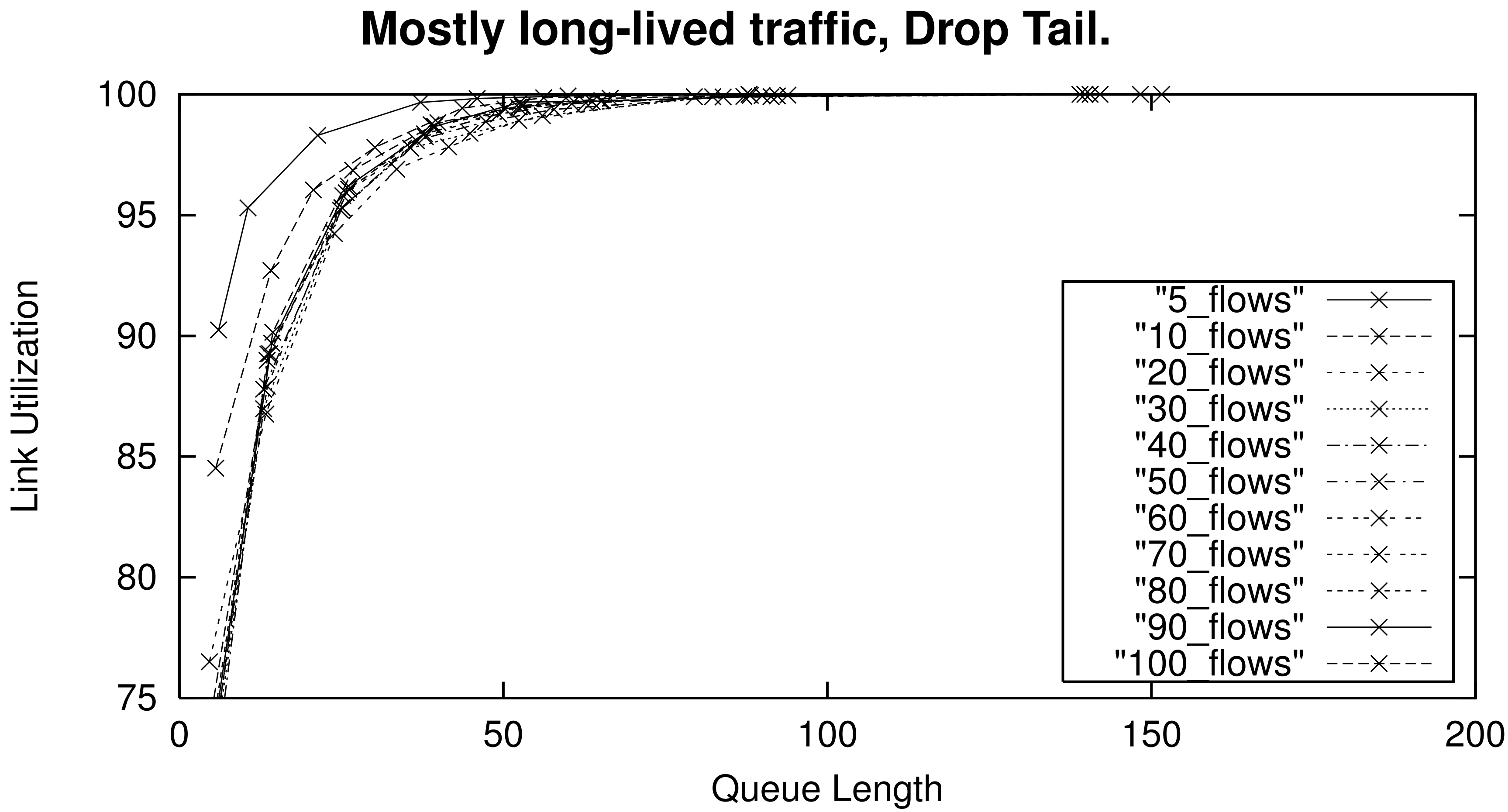
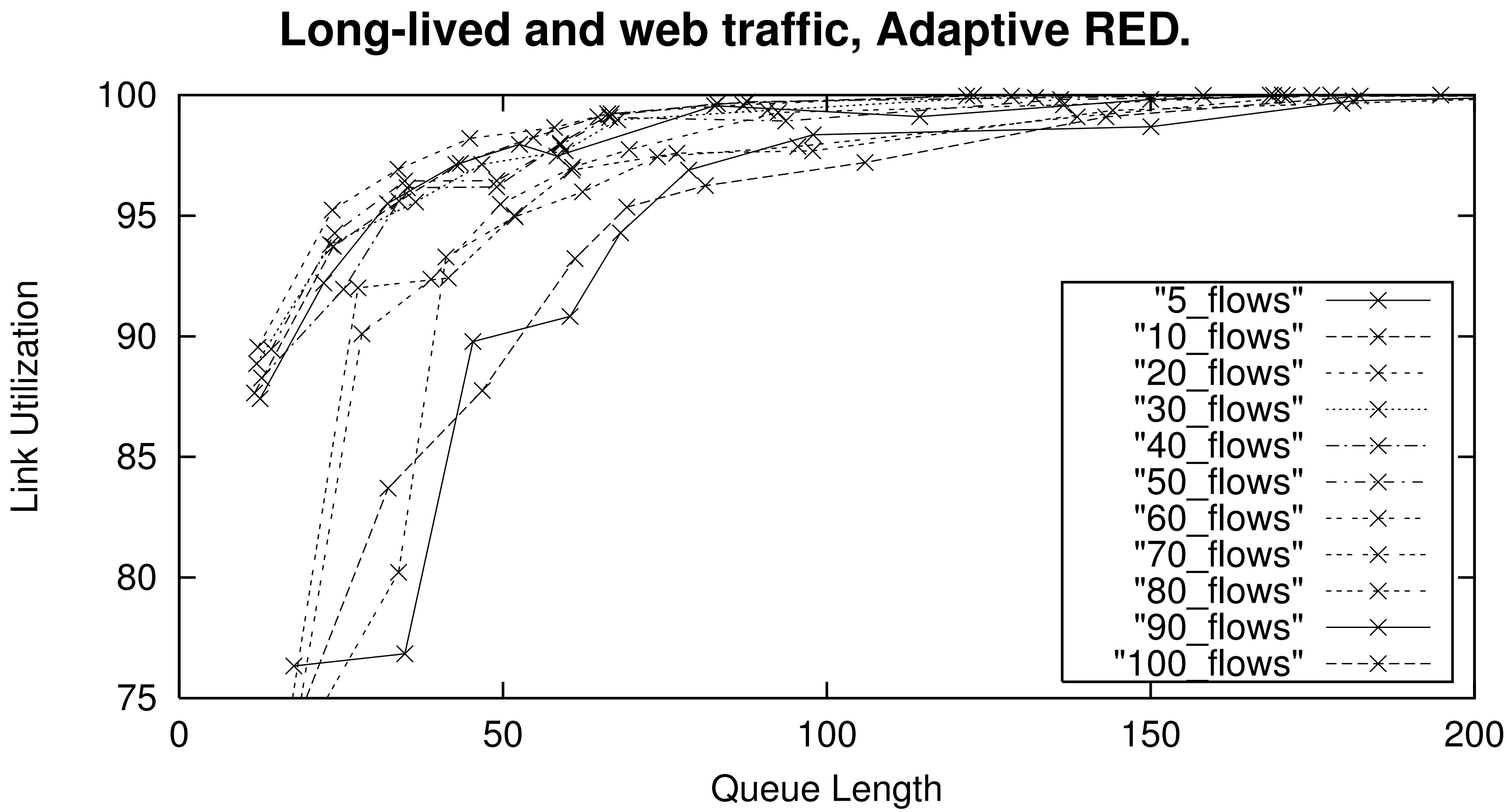


We haven't plotted anything for fairness, our third metric.

(RED and Drop Tail often differ in fairness.)



Long-lived and web traffic, Adaptive RED.
Link Utilization
Queue Length
100
95
90
85
80
75
0
50
100
150
200
"5_flows"
"10_flows"
"20_flows"
"30_flows"
"40_flows"
"50_flows"
"60_flows"
"70_flows"
"80_flows"
"90_flows"
"100_flows"

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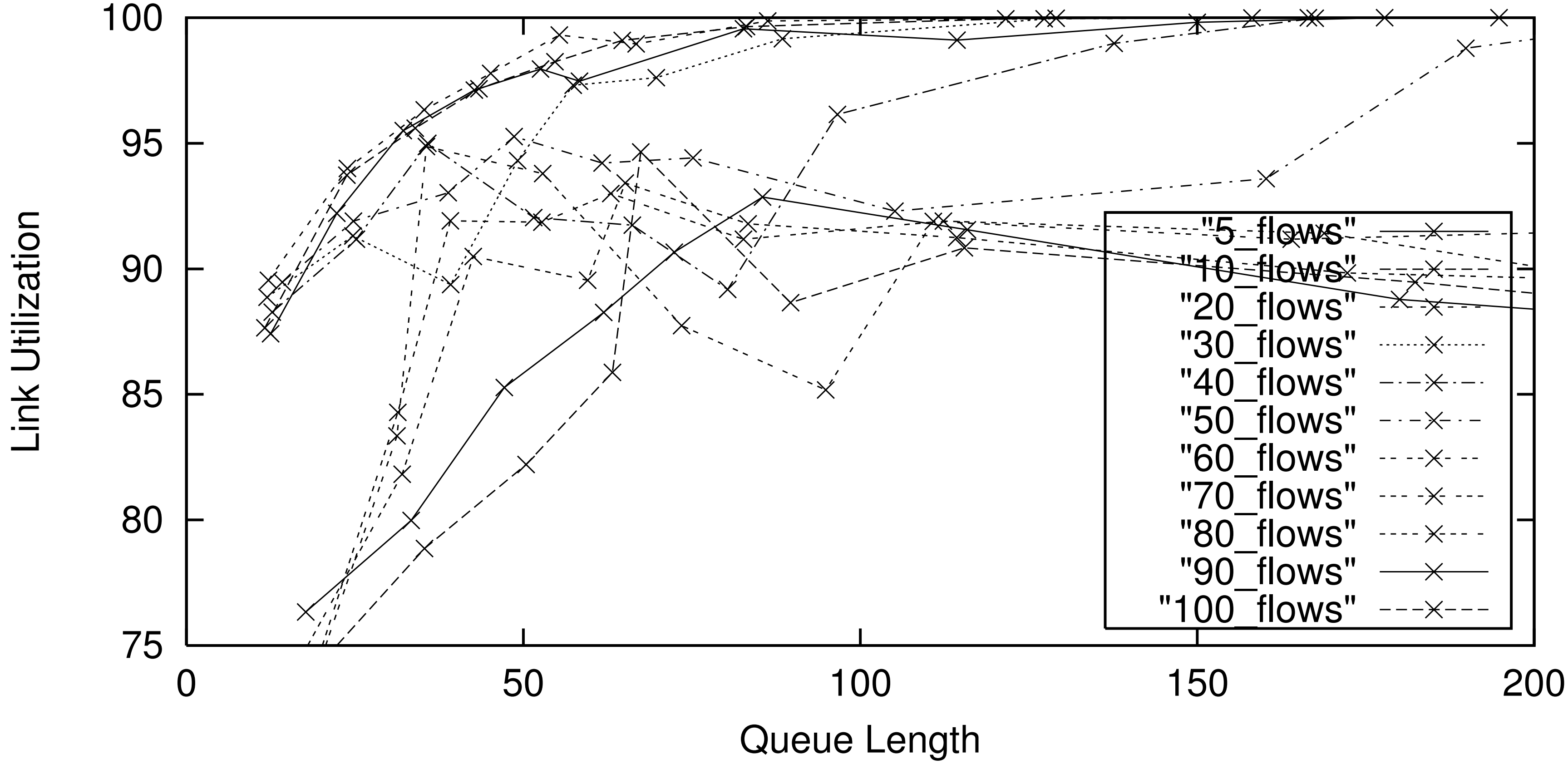
Long-lived and web traffic, Adaptive RED, with reverse-path delay.



The reverse-path queue is configured the same as the forward-path queue:

An aside: creating worst-case oscillations with TCP and AQM:
(from last night)

Assume a ”time constant” for average queue size estimator of T sec.
(T = 1 second, for Adaptive RED).
Then the ”resonant frequency” is roughly $3T$ seconds.

We want to choose the number of flows N so that the packet drop rate p gives a congestion control epoch of $3T$ seconds, so that the natural frequency of TCP matches the resonant frequency of the RED/ARED queue.

Given link bandwidth B pkts/sec, RTT R seconds, N flows:
The average bandwidth per flow is BR/N pkts/RTT.

Facts about TCP:
With packet drop rate p, the average window W is $1.2/\sqrt{p}$ pkts, and there are $\frac{2}{3}W$ RTTs in a congestion control epoch.

So we want $\frac{2}{3}W = 3T/R$, or $W = 4.5T/R$.

So choose N so that: $BR/N = 4.5T/R$, or $N = BR^2/(4.5T)$.

Conjecture: For T = 1 sec., R = 0.1 sec., B = 1000 pkts/sec (8Mbps for 1KB pkts), the worst case oscillations occur with $N = 10/0.45 = 22$ flows.

An Evaluation of AQM

Joint work with Jitendra Padhye and Scott Shenker.

No URL yet...

An Evaluation of AQM

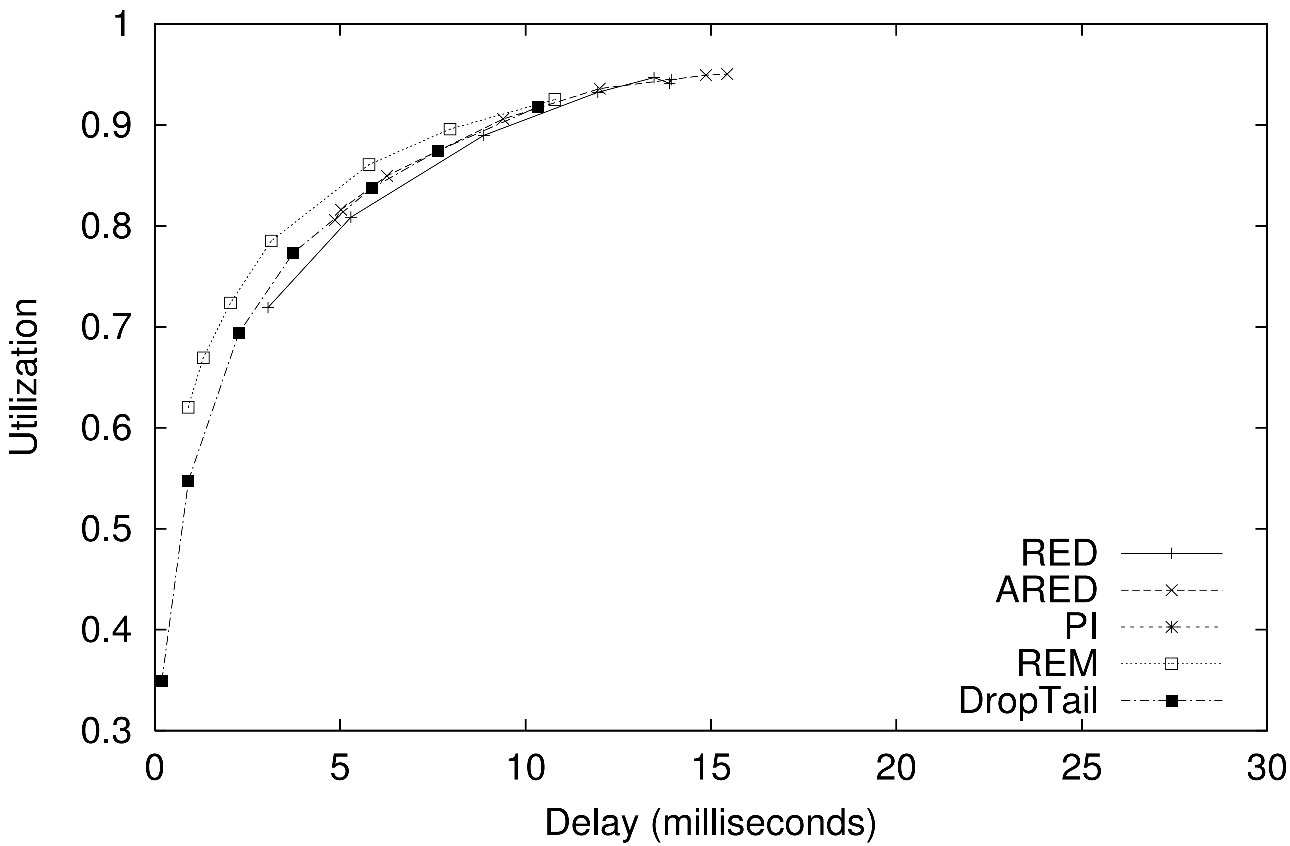
- On many simulation scenarios, Adaptive RED, AVQ, Drop-Tail PI, RED, REM give similar performance.

- Drop-Tail and AVQ generally have higher packet drop rates.

- RED and Adaptive RED can have undesirable oscillations with very large round-trip times.

- REM and PI can perform poorly in scenarios with mostly web traffic, or with changes in the level of congestion.

band=10 del=20 tcp=80 web=80 rev=1 qib=0 ecn=0



Steady-state simulations, with some web traffic and reverse-path traffic, and 40-320 ms RTTs. Queue in packets.

Developing and Evaluating Models:

Joint work with Eddie Kohler

URL for one part:
Building Models for Aggregate Traffic on Congested Links
http://www.icir.org/models/

Models: A Proposal about Developing Models and Simulation Scenarios

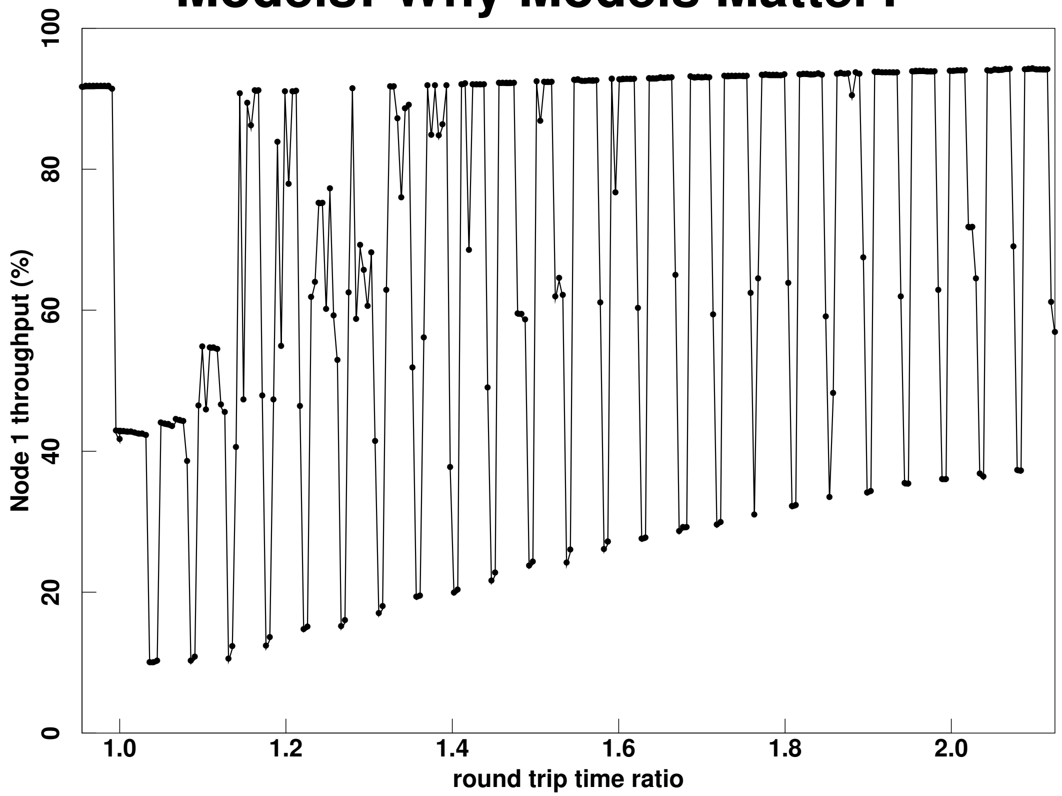
- What measurement studies are needed for improving our models?
  – E.g., where is the congestion in the Internet? What are the ranges of round-trip times? Is more needed on traffic generation? What about reverse-path congestion?

- How do we translate results from measurement studies into our models?

- How do we know what features are critical to include in our models?

- Can we do more to improve our shared understanding of best practices for models and for simulation scenarios?

Models: Why Models Matter?



Flow 1's throughput as a function of the ratio of the two flows' round-trip times, in a scenario with one-way traffic and Drop-Tail queue management. [From ”On Phase Effects”, 1992.]

Open questions?

- How do things in one part of the network (e.g., buffer sizes, flash crowds) affect behavior in other parts?

- How do we shed light on tradeoffs between delay and throughput?

- What are the inherent limitations of one bit of congestion feedback, if any? (E.g., in terms of how aggressive flows can be.)

- What are the inherent limitations of not making reservations, and not keeping per-flow state in the network? (E.g., in terms of how aggressive flows can be.)

- What will the tradeoffs be when we have very fast networks, often with very high available bandwidth, and flows could often send all of their data is a fraction of a RTT?