Buffers and Protocols

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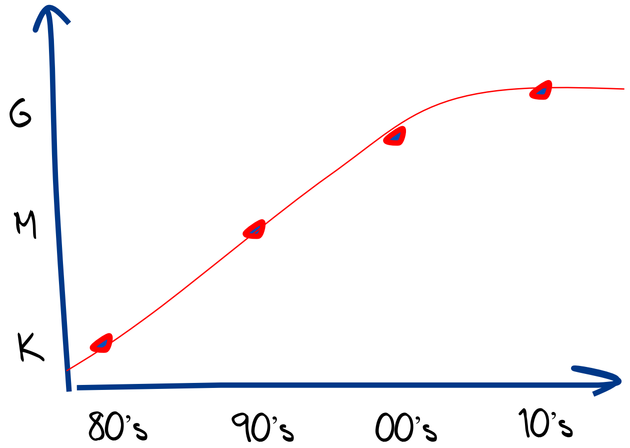
The Evolution of Speed

1980's
- TCP rates of Kilobits per second

1990's
- TCP rates of Megabits per second

2000's
- TCP rates of Gigabits per second

2010's
- TCP rates of Gigabits per second



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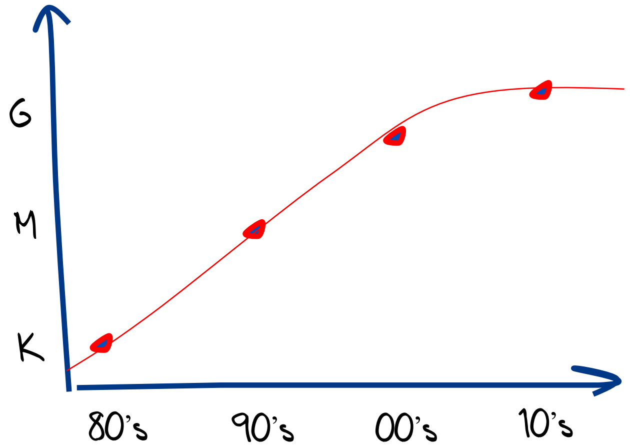
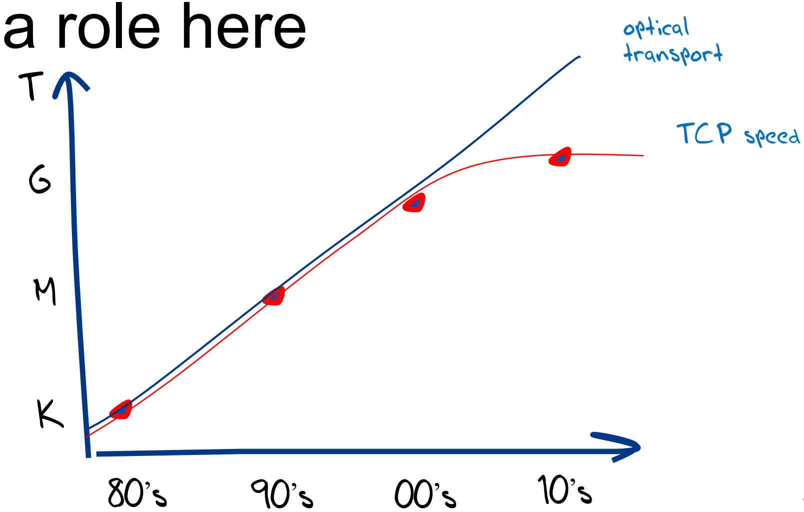


Today

- Optical transmission speeds are now edging into Terrabit capacity
- But peak TCP session speeds are not keeping up
- Its likely that network buffers play a role here
- How?



TCP

- The Transmission Control Protocol is an end-to-end protocol that creates a reliable stream protocol from the underlying IP datagram device
- TCP operates as an adaptive rate control protocol that attempts to operate **efficiently** and **fairly**

TCP Design Objectives

To maintain an average flow which is **Efficient** and **Fair**

Efficient:

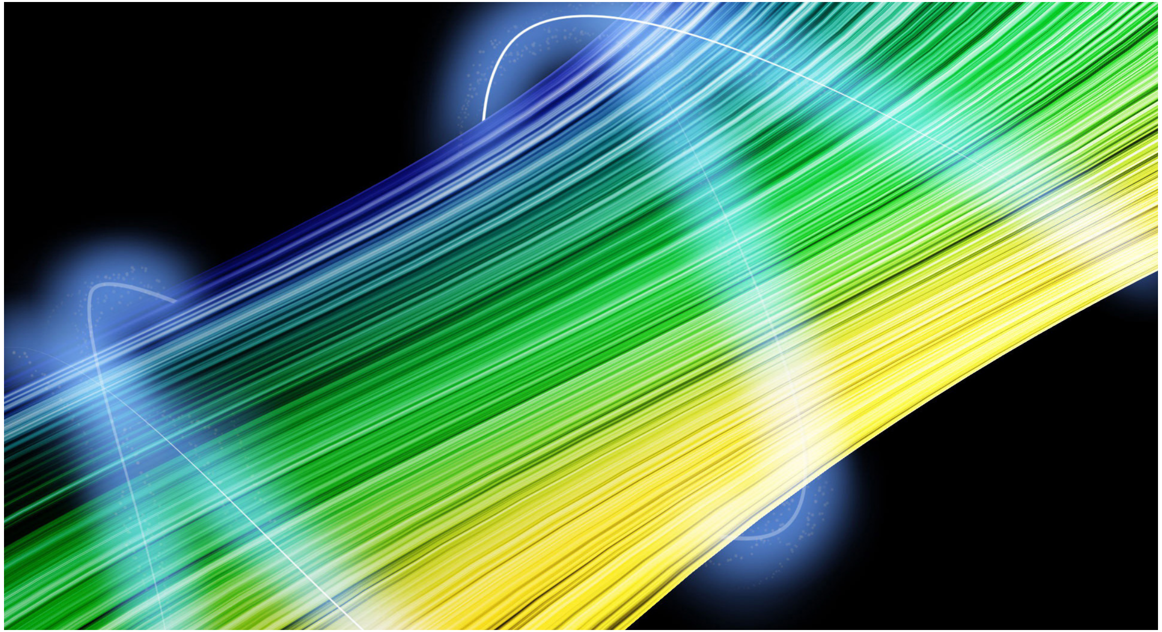
- Minimise packet loss
- Minimise packet re-ordering
- Do not leave unused path bandwidth on the table!

Fair:

- Do not crowd out other TCP sessions
- Over time, take an average 1/N of the path capacity when there are N other TCP sessions sharing the same path

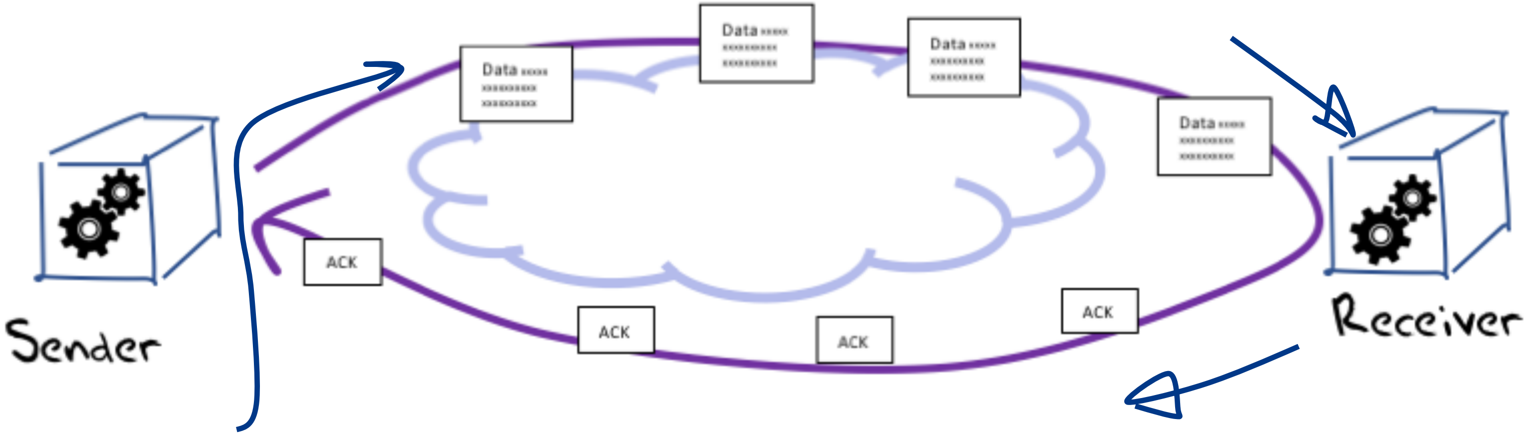
It's a Flow Control process

- Think of this as a multi-flow fluid dynamics problem
- Each flow has to gently exert pressure on the other flows to signal them to provide a fair share of the network, and be responsive to the pressure from all other flows



TCP Control

TCP is an ***ACK Pacing*** protocol



Data sending rate is matched to the ACK arrival rate

TCP Control

- Ideally TCP would send packets at a fair share of available network capacity. But the TCP sender has no idea what “available network capacity” means.
- So TCP uses ‘rate adaptation’ to probe into network, increasing the sending rate until it is ‘too fast’
- Packet drop is the conventional signal of ‘too fast”

TCP Control

ACK pacing protocols relate to a **past** network state, not necessarily the **current** network state

- The ACK signal shows the rate of data that left the network at the receiver that occurred at ½ RTT back in time

- If there is data loss in the forward path, the ACK signal of that loss is already at least ½ RTT old!

  TCP should react quickly to 'bad' news

- If there is no data loss, that is also old news

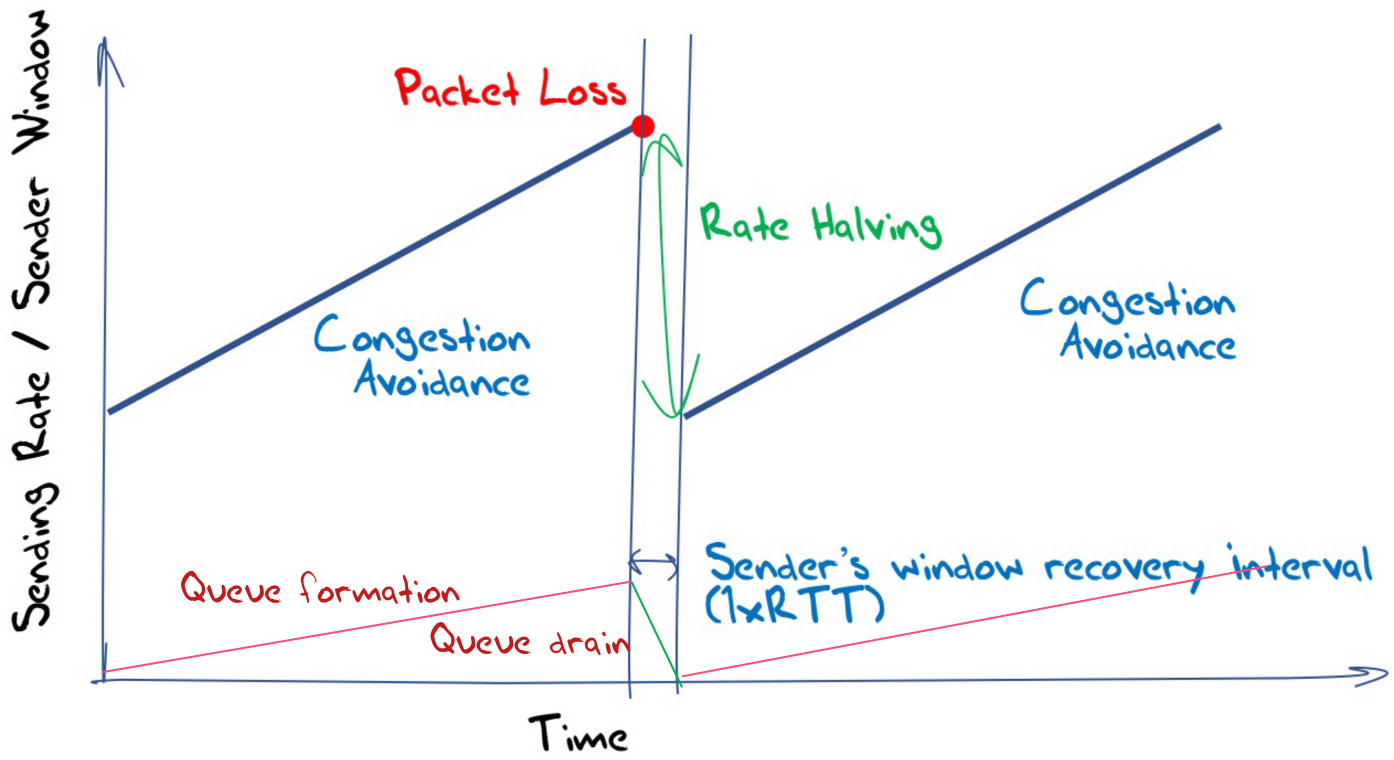
  TCP should react conservatively to 'good' news

"Classic TCP" - TCP Reno

- Additive Increase Multiplicative Decrease (AIMD)
  - While there is no packet loss, increase the sending rate by one segment (MSS) each RTT interval
  - If there is packet loss decrease the sending rate by 50% over the next RTT Interval, and halve the sender's window

- Start Up
  - Each RTT interval, double the sending rate
  - We call this "slow start" – probably because its anything but slow!!!

TCP Reno and Buffers - the Theory



TCP and Buffers - the Theory

- When a sender receives a low signal it repairs the loss and halves its sending window
- This will cause the sender to pause for the amount of time to drain halve the outstanding data in the network
- Ideally this exactly matches the amount of time taken for the queue to drain
- At the time the queue is drained the sender resumes its sending (at half the rate) and the buffer has fully drained
- For this to work, the queue size should equal the delay bandwidth product of the link it drives

TCP and Buffers – the Theory

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- ...drained the sender resumes its sending ...) and the buffer has fully drained
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All this works with an assumption of a single queue and a single flow

TCP and Buffers

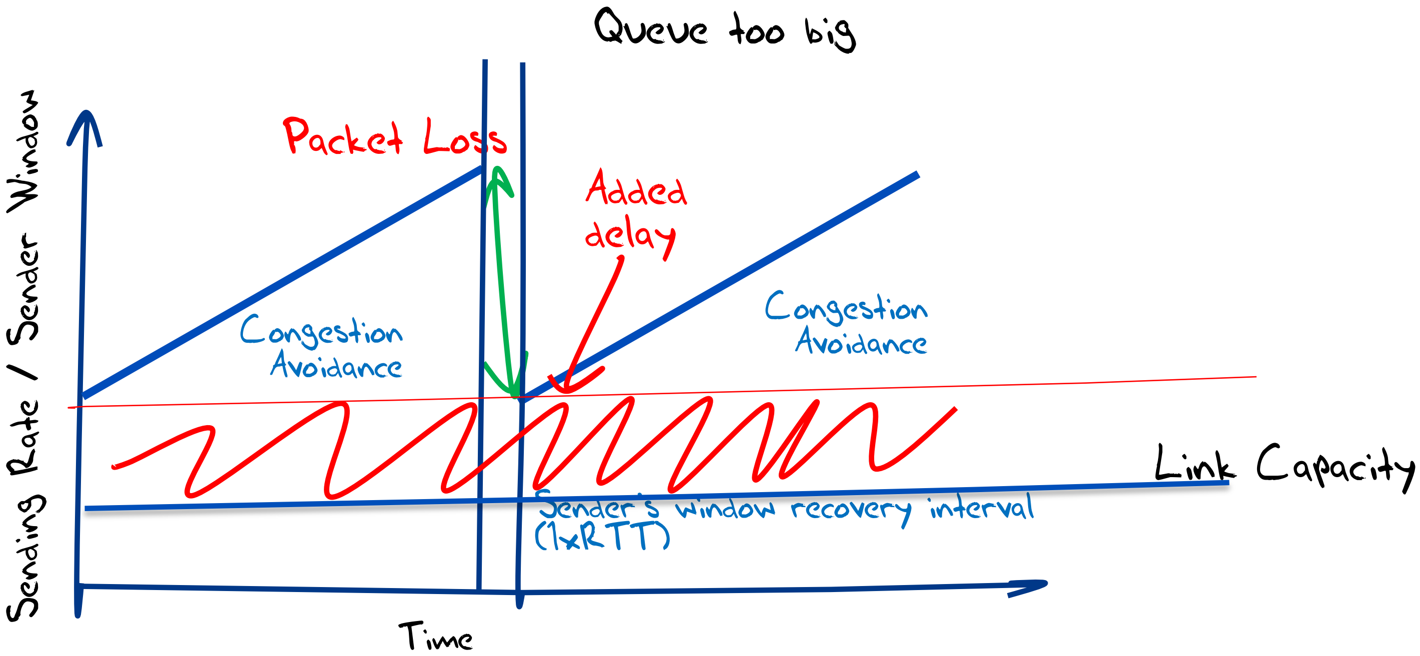
- The rule of thumb for buffer size is

$$\textbf{Size} = (\boldsymbol{BW} \cdot \boldsymbol{RTT})$$

"High Performance TCP in ANSNET"
Villamizar & Song, 1994

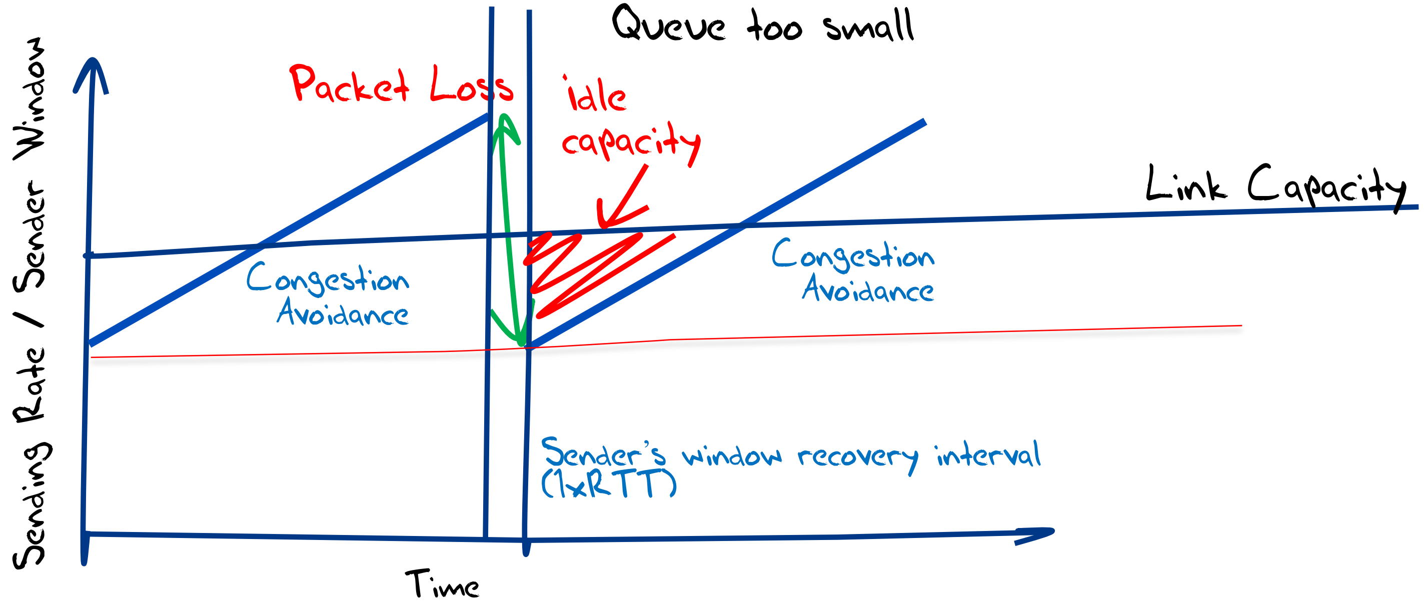
TCP and Buffers

Too Big: The queue never drains, so the buffer adds delay to the connection



TCP and Buffers

Too Small: The queue drains and the sender operates below bottleneck speed – so the link is under-used



Refinements to RENO

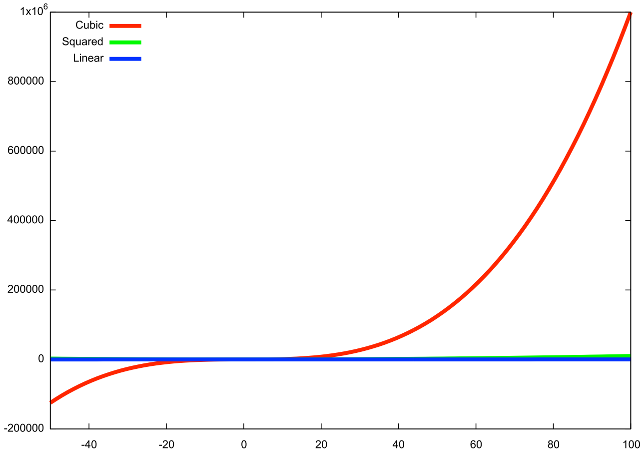
- There have been many efforts to alter RENO’s flow control algorithm
- In a loss-based AIMD control system the essential parameters are the manner of rate increase and the manner of loss-based decrease
  - For example:

    MulTCP behaves as it it were N simultaneous TCP sessions: i.e. increase by N segments each RTT and rate drop by 1/N upon packet loss

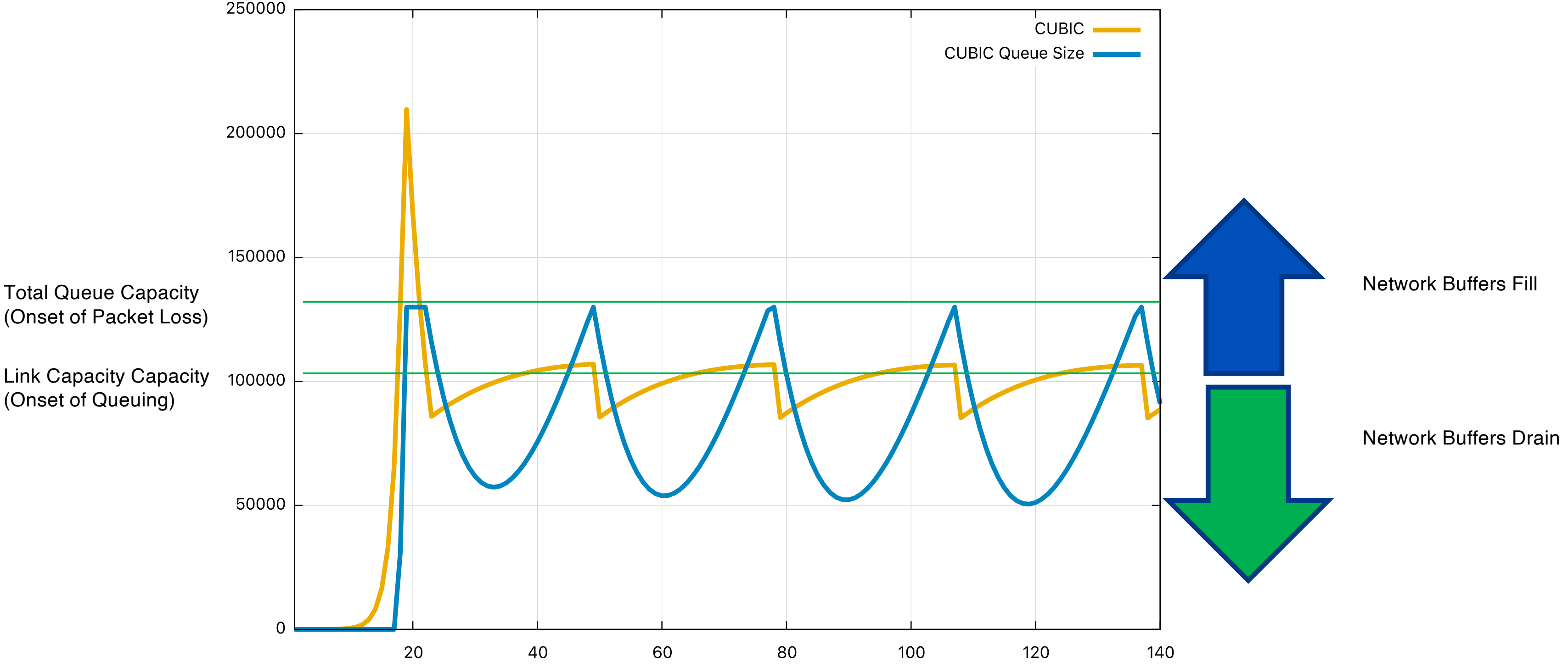
- What about varying the manner of rate increase away from AI?

Enter CUBIC

- CUBIC is designed to be useful for high speed sessions while still being ‘fair’ to other sessions and also efficient even at lower speeds
- Rather than probe in a linear manner for the sending rate that triggers packet loss, CUBIC uses a non-linear (cubic) search algorithm



CUBIC and Queue formation



CUBIC assessment

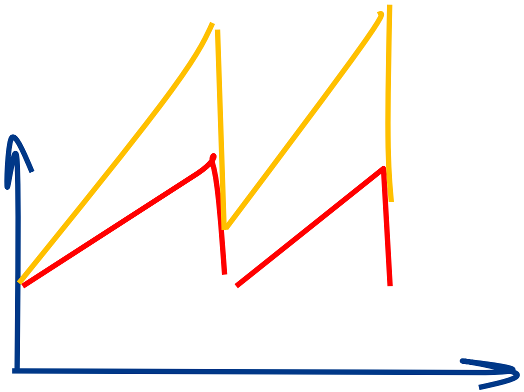
- Can react quickly to available capacity in the network
- Tends to sit for extended periods in the phase of queue formation
- Can react efficiently to long fat pipes and rapidly scale up the sending rate
- Operates in a manner that tends to exacerbate ‘buffer bloat’ conditions

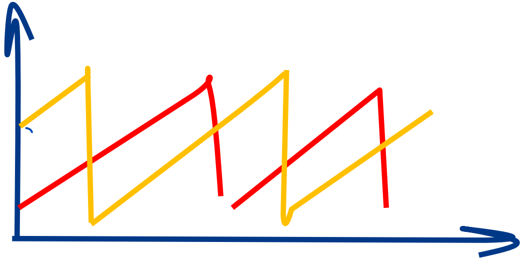
From 1 to N - Scaling Switching

- This finding of buffer size relates to a single flow through a single bottleneck resource
- What happens to buffers with more flows and faster transmission system?

Flow Mixing

- If 2 flows use a single buffer and they resonate precisely then the buffer still needs to be delay-bandwidth size
- If they are precisely out of phase the common buffer requirement is halved





Smaller Buffers?

- If 2 flows use a single buffer and they resonate precisely then the buffer still needs to be delay-bandwidth size
- If they are precisely out of phase the common buffer requirement is halved
- What about the case of N de-synchronised flows?

$$\textbf{Size} = (\boldsymbol{BW} \cdot \boldsymbol{RTT}) / \sqrt{\boldsymbol{N}}$$

Assuming that the component flows manage to achieve a fair outcome of obtaining 1/N of the resource in a non-synchronised manner, then the peak buffer resource is inversely proportionate to the square root of N

("Sizing Router Buffers", Appenzeller, McKeown, Keslassy, SIGCOM'04)

The Role of Buffers

- Buffers in a network serve two essential roles:
  - smooth sender burstiness
  - Multiplexing N inputs to 1 output

Sender Pacing

- Distribute cwnd data across the entire RTT interval
- Remove burst adaptation pressure on network buffers

Tiny Buffers?

- If all senders ‘paced’ their sending to avoid bursting, and were sensitive to the formation of standing queues then we would likely have a residual multiplexing requirement for buffers where:

$$B >= O(\log W)$$

where W is the average flow window size

Why is this important?

- Because memory speed is not scaling at the same rate as transmission or switching
- Further capacity and speed improvements in the network mandate reduced memory demands within the switch

Switching Chip Design TradeOffs

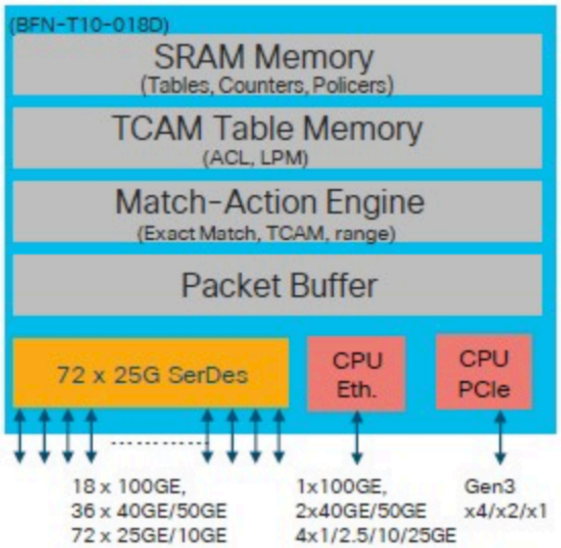
- On Chip memory is fast, but limited to between ~16M to ~64M
- A chip design can include an interface to external memory banks but the memory interface/controller also takes up chip space and the external memory is slower

- Between 20% to 60% of switch chip real estate is devoted to memory / memory control
- Small memory buffers in switch design allows for larger switch fabric implementations on the chip

Switch Design

Barefoot Tofino ASIC Architecture

- BFN-T10-018D from Tofino family
- 1.8Tbps Single Chip Ethernet Switch
- 2 Pipes @0.9 Tbps
- P4-programmable pipeline
- Single 16 MB Unified Packet Buffer
- Inband Network Telemetry (INT)

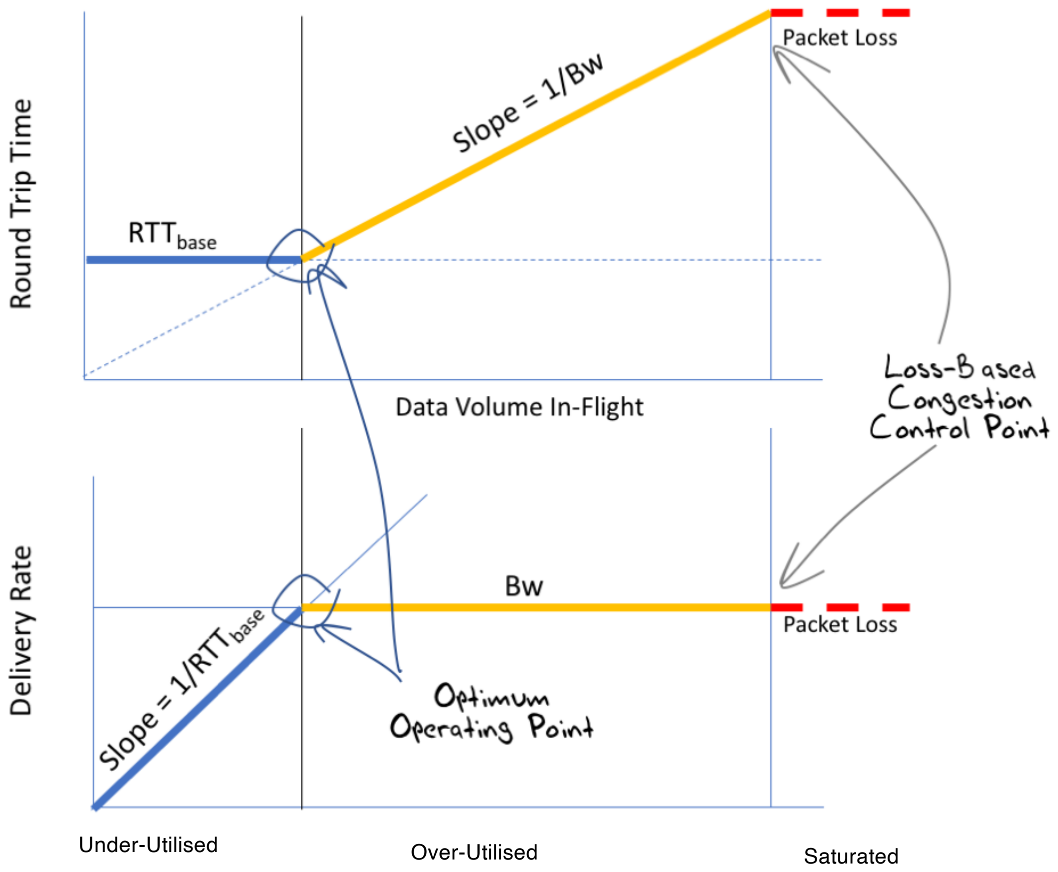


Flow States

- There are three ‘states’ of flow management:
  - **Under-Utilised** – where the flow rate is below the link capacity and no queues form
  - **Over-Utilised** – where the flow rate is greater that the link capacity and queues form
  - **Saturated** – where the queue is filled and packet loss occurs

- Loss-based control systems probe upward to the Saturated point, and back off quickly to what they guess is the Under-Utilised state in order to the let the queues drain
- But the optimal operational point for any flow is at the point of state change from Under to Over-utilised, not at the Saturated point

RTT and Delivery Rate with Queuing



How to detect the onset of queuing?

- By getting the network say when queues are forming

ICMP Source Quench Redux!

- Switch generates an ICMP message (similar to ICMP PTB)
- ICMP payload allows sender to identify TCP session

ICMP Issues

- IMCP messages are unverified
  - DOS attack vector
- ICMP messages are often filtered
  - A sender cannot rely upon the message
- Anycast can add subtle complications here!

Explicit Congestion Notification

Explicit Congestion Notification

- Sparse signal (single bit)
- Both hosts and routers need to be ECN aware
- IP level marking requires end host protocol surgery at both ends:
    - Receivers need to reflect ECN bits
    - Senders need to pass IP ECN up to the TCP session

ECN Issues

- It would be good if…
  - everyone did it!

- But they don't all do it, which means that hosts cannot rely on ECN as the only means of congestion control
- What's the value of partial adoption of ECN?

High Precision Congestion Control

- Eliminate all the guesswork out of the problem by having each switch attach the time, local queue length and link bandwidth to the IP packet!

How to detect the onset of queuing?

- By getting the network say when queues are forming

OR

- By detecting the onset of queue-based delays in the measured RTT

Flow Control Revisited

- Current flow control systems make small continual adjustments every RTT interval and a massive adjustment at irregular intervals
  - As the flow rate increases the CA adjustments of 1 segment per RTT become too small
  - Rate halving is a massive response

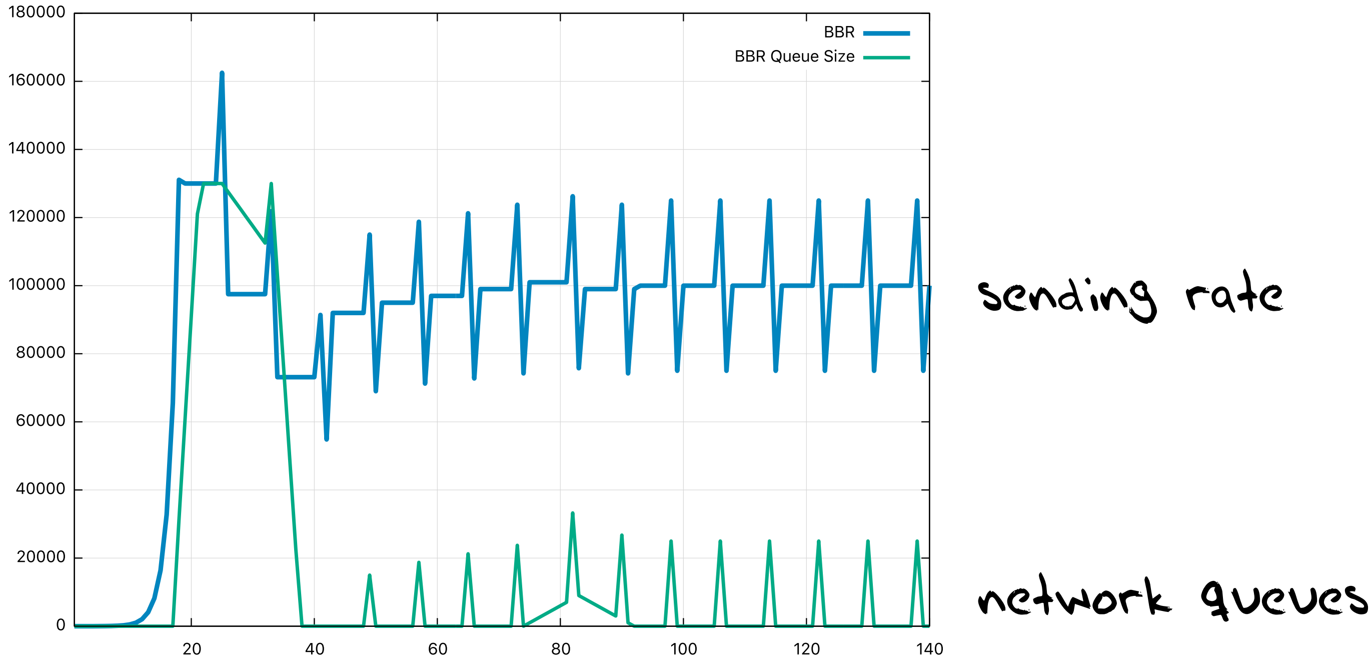
OR

- We could use a system that only made periodic adjustments every *n* RTT intervals
  - And set the adjustment to be proportionate to the current flow rate

BBR Design Principles

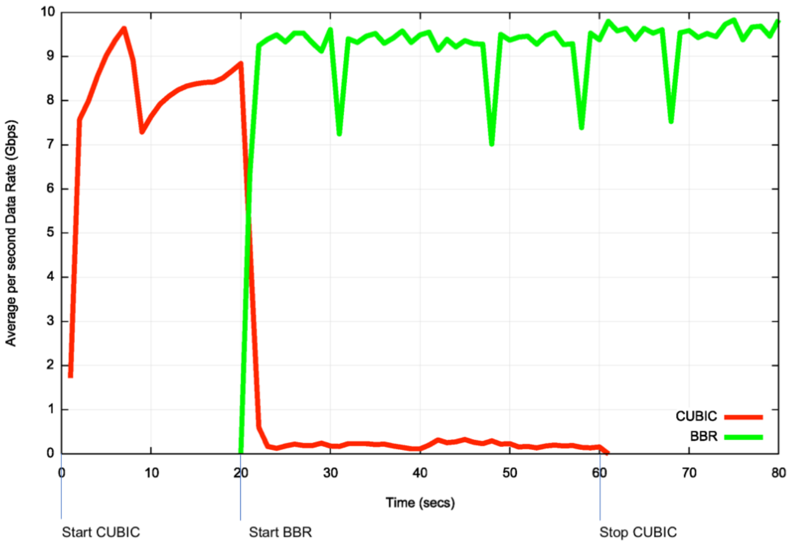
- Pace the sending packets to avoid the need for network buffer rate adaptation
- Probe the path capacity only intermittently (every *8th* RTT)
- Probe the path capacity by increasing the sending rate by *25%* for an RTT interval and then drop the rate to drain the queue:
  - If the RTT of the probe interval equals the RTT of the previous state then there is available path bandwidth that could be utilised
  - If the RTT of the probe rises then the path is likely to be at the onset of queuing and no further path bandwidth is available
- Do not alter the path bandwidth estimate in response to packet loss

Idealised BBR profile



BBR Politeness?

- BBR will probably not constantly pull back when simultaneous loss-based protocols exert pressure on the path's queues
- BBR tries to make minimal demands on the queue size, and does not rely on a large dynamic range of queue occupancy during a flow



Pulling it back together…

Where are we in networking today?

- A diverse mix of e-2-e TCP control protocols
  CUBIC, NewRENO, LEDBAT, Fast, BBR
- A mix of traffic models
  Buffer-filling streamers, flash bursts, bulk data
- A mix of active queue disciplines
  RED, WRED, CODEL, FQ, none
- A mix of media
  Wire line, mobile, WiFi
- A mix of buffer size deployments
- Sporadic ECN marking

Protocol Darwinism?

What “wins” in this diverse environment?

- *Efficiency* is perhaps more critical than *fairness* as a “survival fitness” strategy
- I suspect that protocols that make minimal assumptions about the network will be more robust than those that require certain network characteristics to operate efficiently
- Protocols that operate with regular feedback mechanisms appear to be more robust than irregular “shock” treatment protocols

What is all this telling us?

- The Internet still contains a large set of important unsolved problems
- And some of our cherished assumptions about network design may be mistaken
- Moving large data sets over very high speed networks requires an entirely different approach to what we are doing today
- BBR seems to be a step in an interesting direction, particularly for very high speed networking
- We actually don’t know much about fine-grained behaviour of large scale high capacity switching systems.
- It’s clear that more research and more testing at scale would help here!

That's it!

Questions?