Buffers and Protocols

Geoff Huston

Chief Scientist, APNIC Labs

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Networking is all about moving data

- The way in which data movement is controlled is a key characteristic of the network architecture
- The Internet Protocol architecture passed all controls to the end systems, and treated the network as a passive switching environment
- All this is changing again as we see a what could well be a new generation of flow control algorithms being adopted in the Internet

Let's talk about speed

- How fast can we push a single session to move data through the network?
- Session speed is the result of a combination of:
  - available transmission speed
  - transmission bit error rate
  - packet sizes
  - switching capacity
  - end-to-end latency
  - host buffer size
  - network buffer size
  - protocol efficiency
  - All of these factors are critical

The Evolution of Speed

1980's

- TCP rates of Kilobits per second

1990's

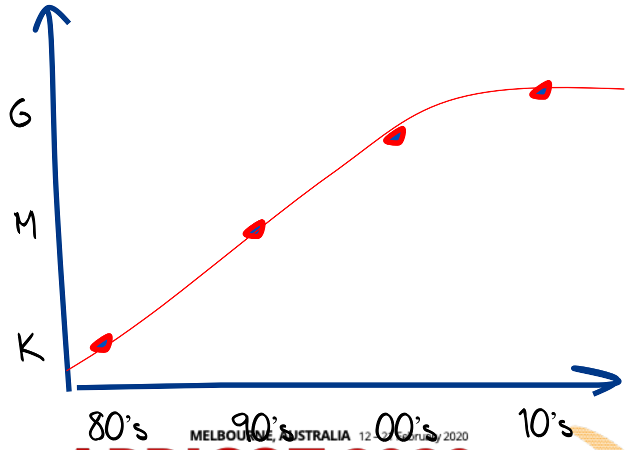
- TCP rates of Megabits per second

2000's

- TCP rates of Gigabits per second

2010's

- TCP rates of Gigabits per second



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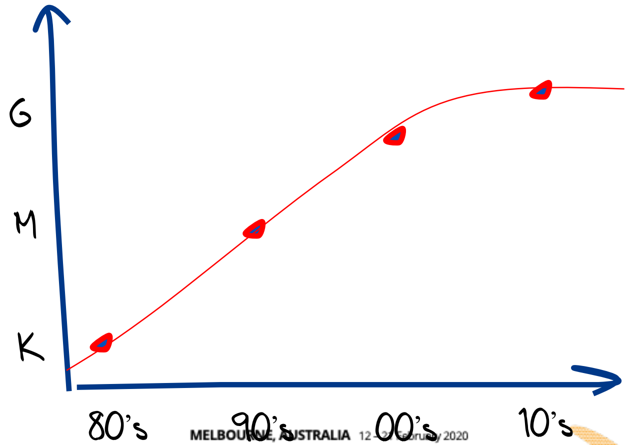
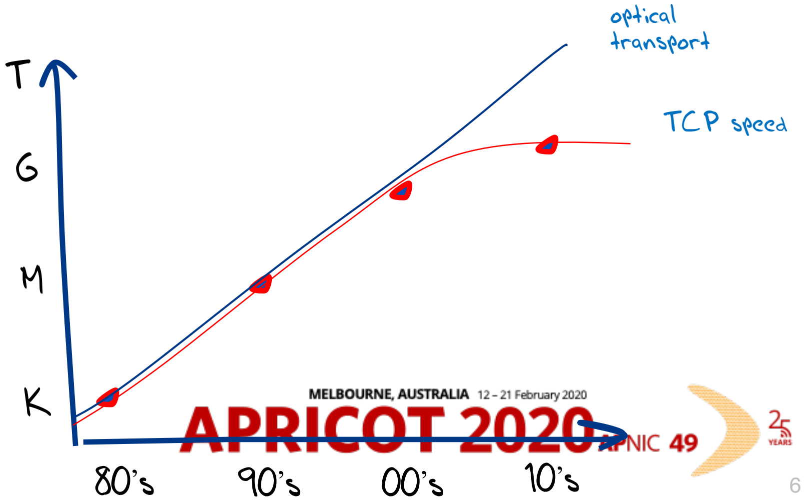


Today

- Optical transmission speeds are now in 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 **fairly** and **efficiently**

TCP Design Objectives

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

Efficient:

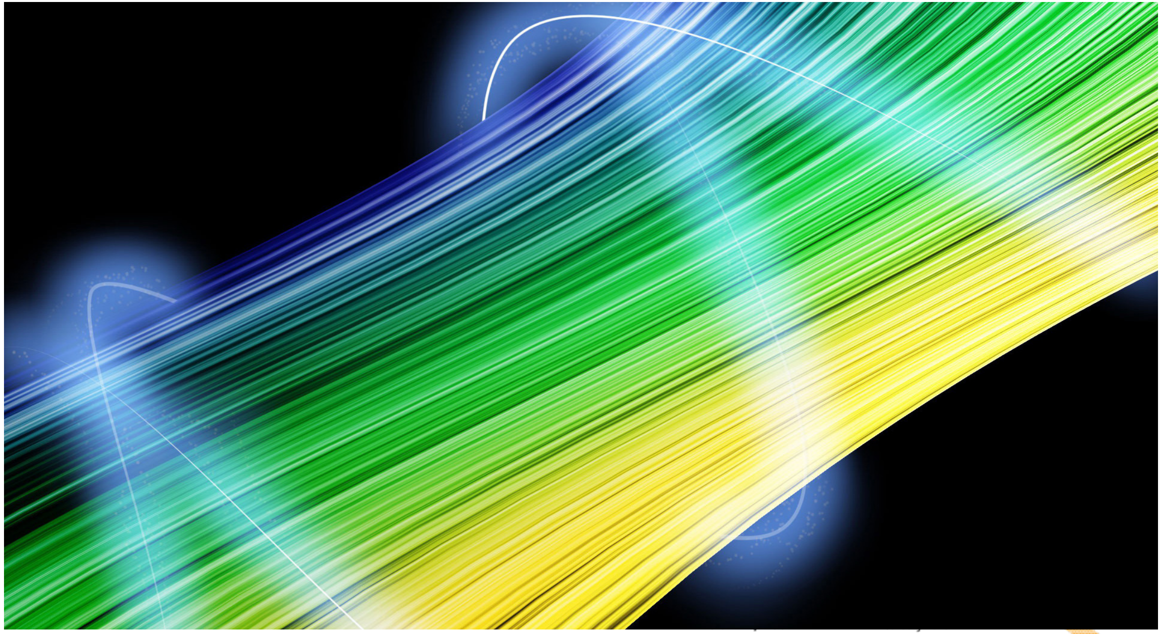
- Minimise packet loss
- Minimise packet re-ordering
- Do not leave unused path bandwidth on the table!

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

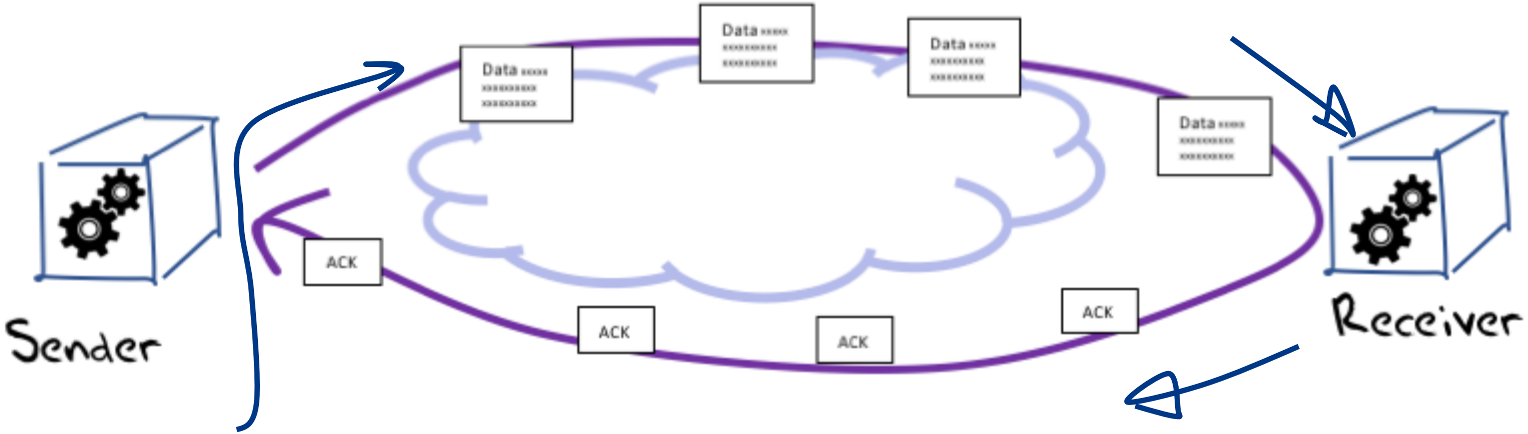


TCP Control

- 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

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



Data sending rate is matched to the ACK arrival rate

TCP Control

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

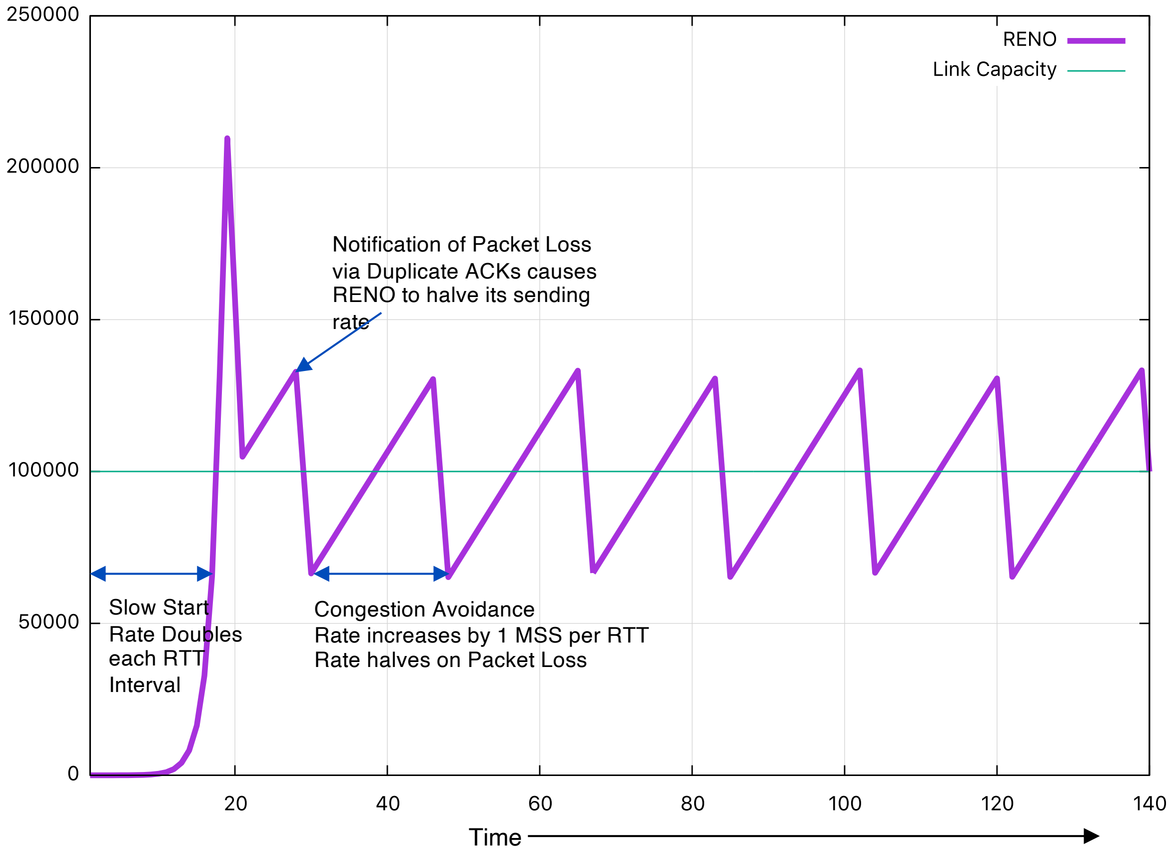
- The ACK signal shows the rate of data that left the network at the receiver that occurred at ½ RTT back in time
- So if there is data loss in the forward path, the ACK signal of that loss is already at least ½ RTT old!
  - **So TCP should react quickly to 'bad' news**
- If there is no data loss, that is also old news
  - **So 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!!!

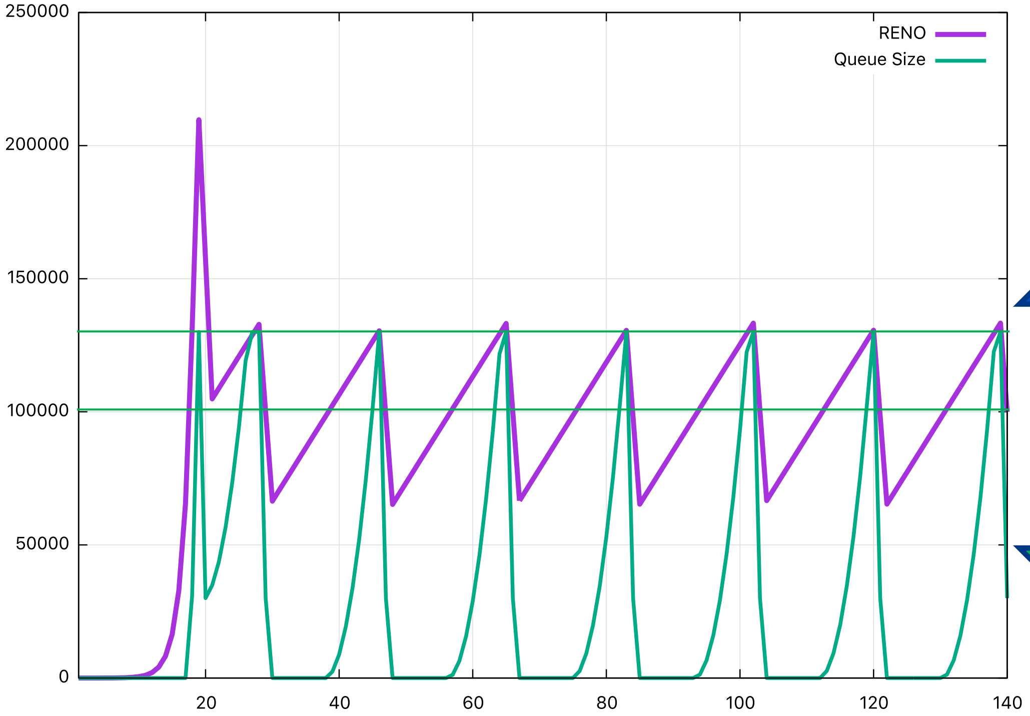
Idealised TCP Reno



TCP RENO and Idealized Queue Behaviour

Total Queue Capacity
(Onset of Packet Loss)

Link Capacity Capacity
(Onset of Queuing)



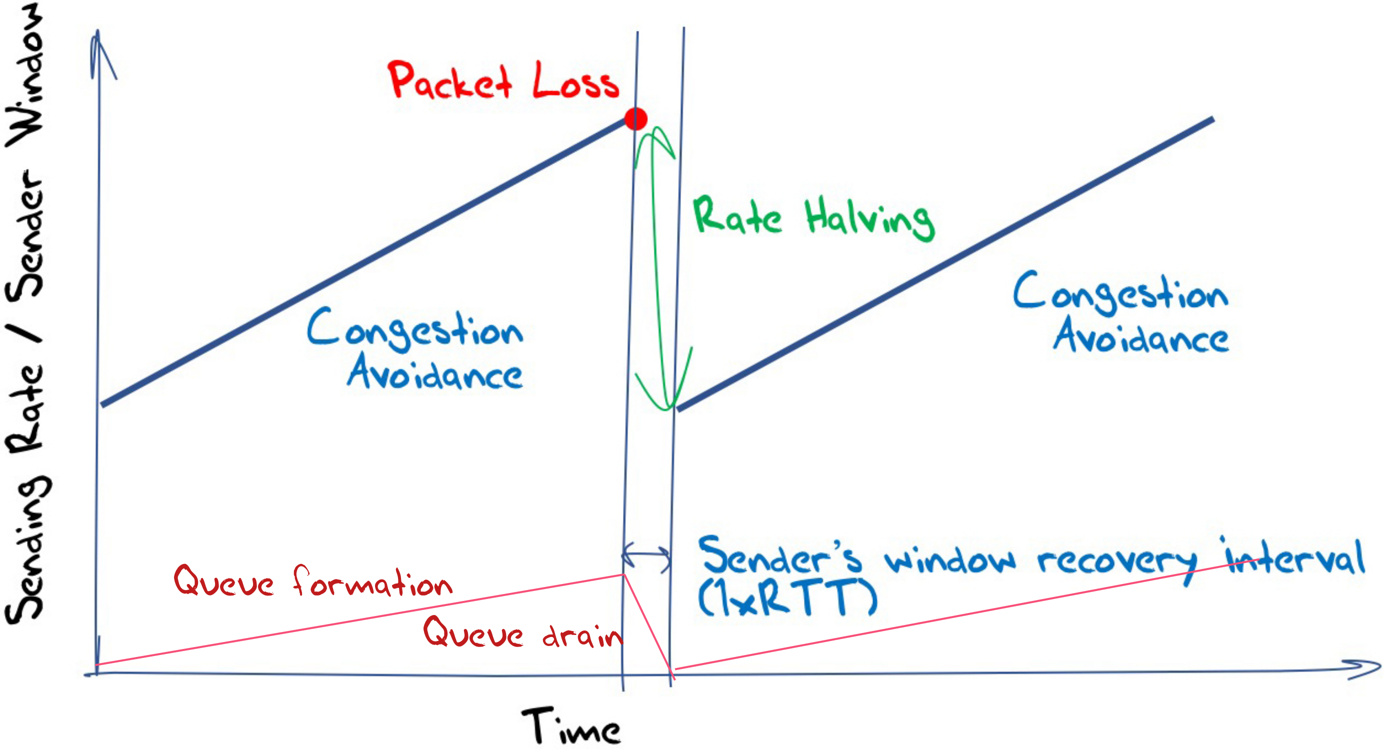
Network Buffers Fill

Network Buffers Drain

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
- **All this works with an assumption of a single queue and a single flow**

TCP and Buffers – the Theory



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

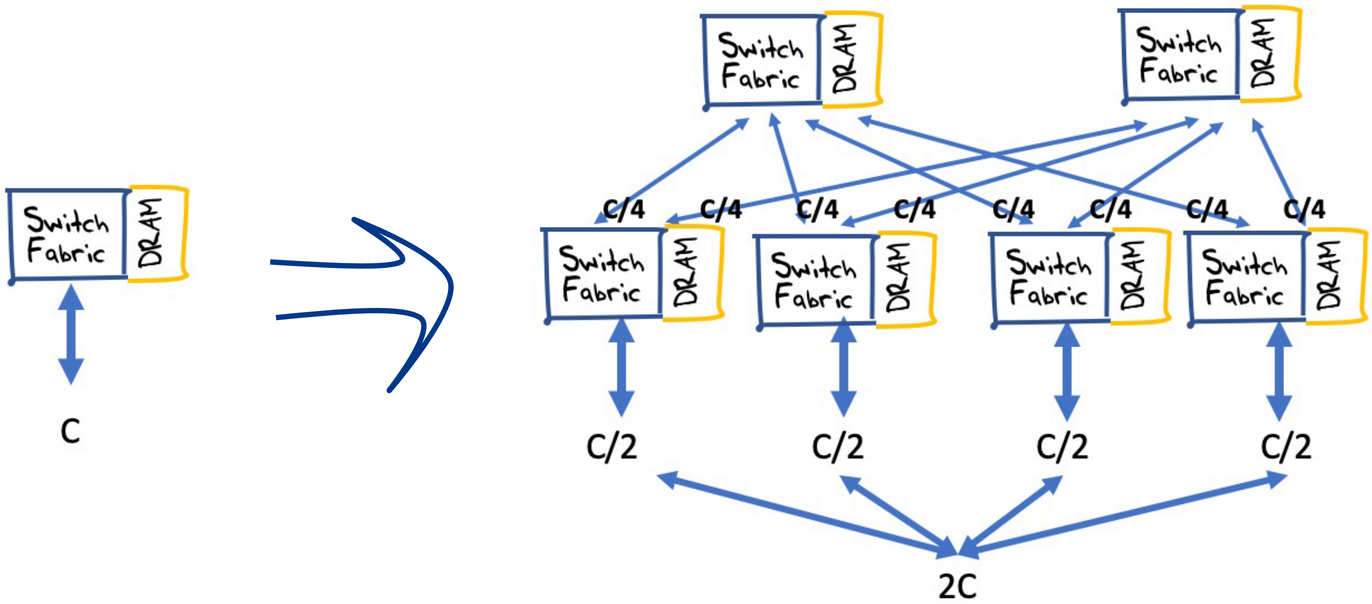
TCP and Buffers

- Larger: The queue never drains, so the buffer adds delay to the connection
- Smaller: The queue drains and the sender operates below bottlenck speed – so the link is under-used

From 1 to N - Scaling Switching

- This finding of buffer size relates to a single flow through a single bottleneck resource
- What happens with more flows and faster transmission system?
- It appears that scaling has non-linear properties

Scaling Switching

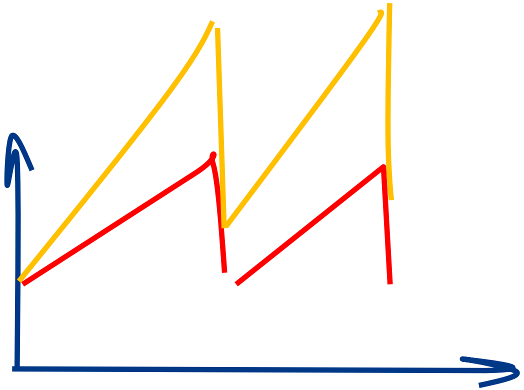


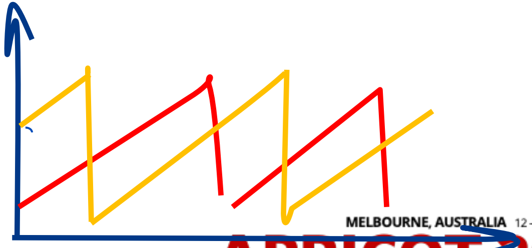
Scaling Switching

- If we can't up the clock speed of a switching chip we need to use parallel architectures
- Double the switching speed requires 6x the overall switch capacity and >4x the memory
- This non–linear scaling represents a considerable challenge in high speed systems
- Is it really necessary to use delay-bandwidth sized high speed memory interfaces?

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





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?
- Stanford 2004 study ("Sizing Router Buffers", Appenzeller, McKeown, Keslassy, SIGCOM'04)

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

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

Switching Chip Design TradeOffs

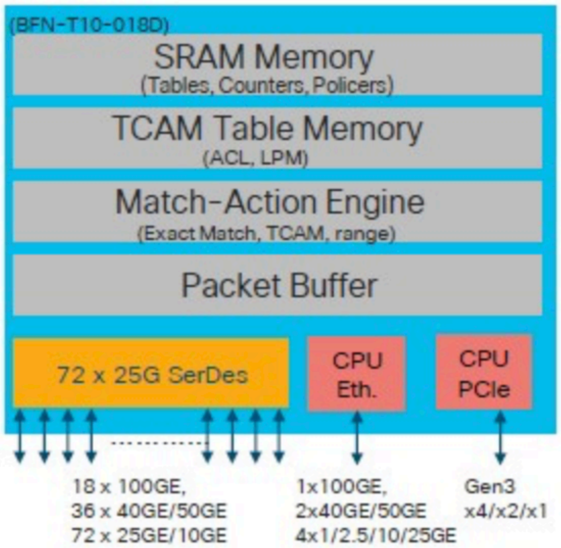
- On Chip memory is fast, but limited to between ~16M to ~64M
- A chip design can include an interface to external memory banks but the memory interface/controller also takes up chip space and the external memory is slower

- 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

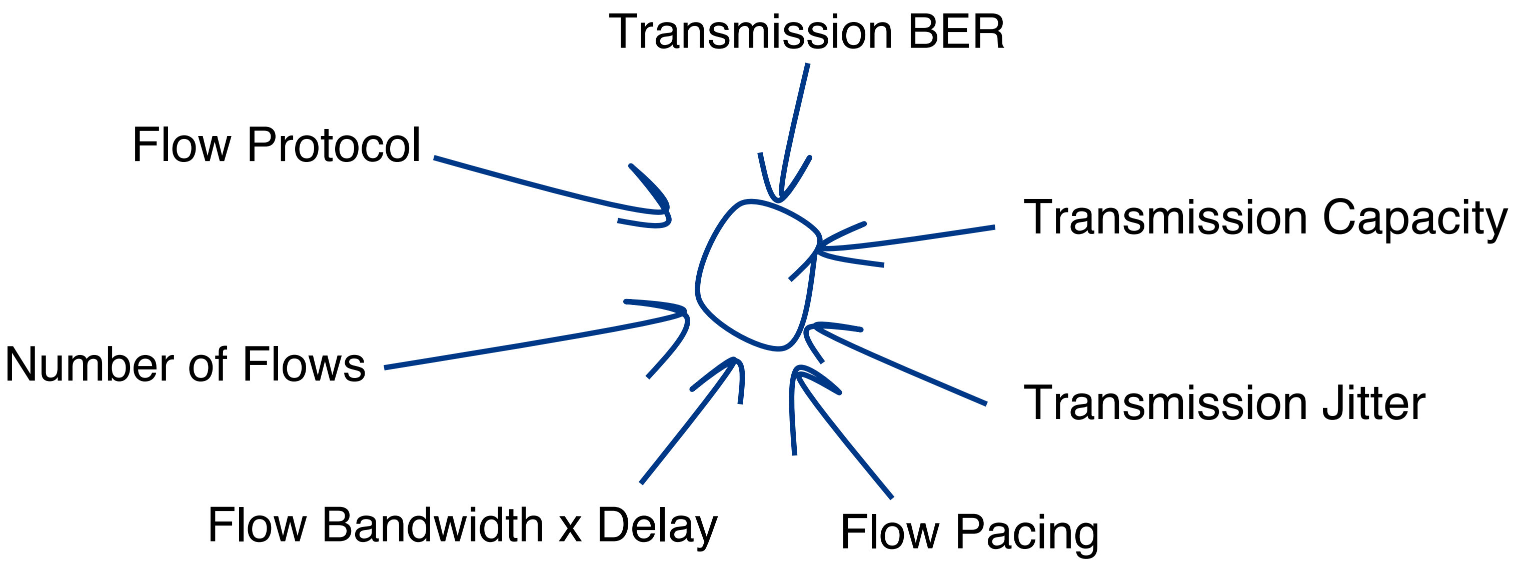
- BFN-T10-018D from Tofino family
- 1.8Tbps Single Chip Ethernet Switch
- 2 Pipes @0.9 Tbps
- P4-programmable pipeline
- Single 16 MB Unified Packet Buffer
- Inband Network Telemetry (INT)



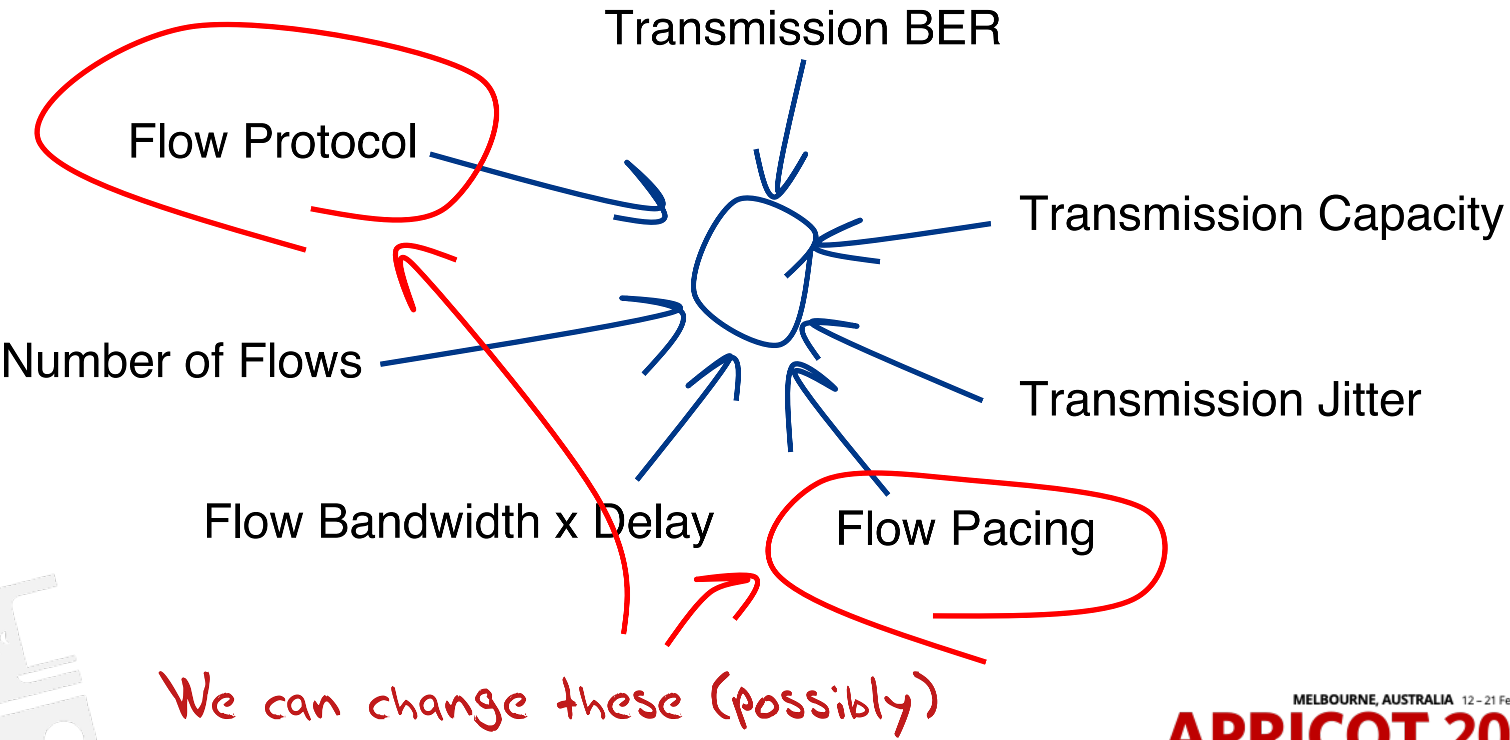
The Network Design Dilemma

- What are the acceptable tradeoffs here?
  - Larger buffers tend to create more efficient outcomes for aggregate throughput
  - Smaller buffers limit the achievable performance of some protocols

Buffer Sizing



Buffer Sizing



Protocols and Buffers

- It seems that Reno strongly influenced the design assumption that BDP buffers are necessary in the Internet
- But are there other protocols that lead to different assumptions about buffer sizes?

Refinements to RENO

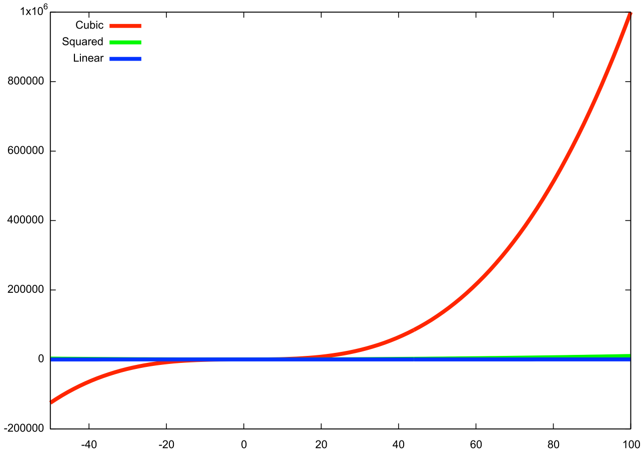
- There have been many efforts to alter RENO’s flow control algorithm
- In a loss-based AIMD control system the essential parameters are the manner of rate increase and the manner of loss-based decrease
  - For example:

    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

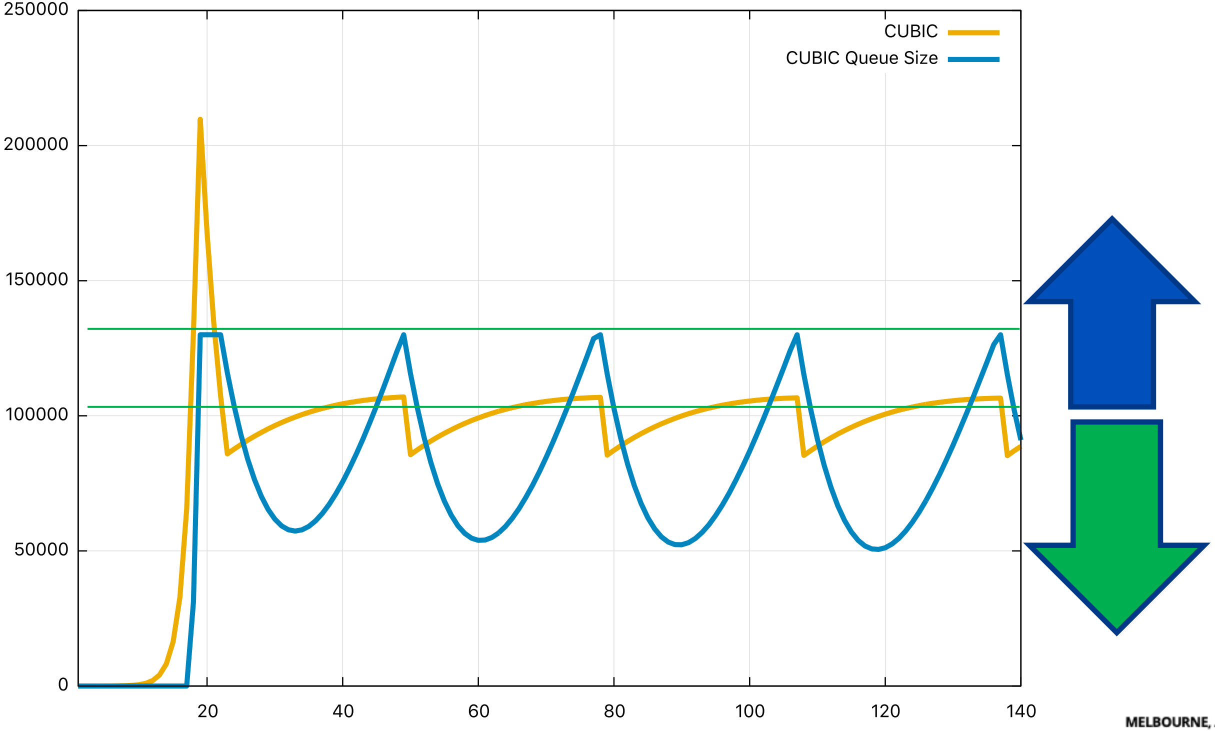
- CUBIC is designed to be useful for high speed sessions while still being 'fair' to other sessions and also efficient even at lower speeds
- Rather than probe in a linear manner for the sending rate that triggers packet loss, CUBIC uses a non-linear (cubic) search algorithm



CUBIC and Queue formation

Total Queue Capacity
(Onset of Packet Loss)

Link Capacity Capacity
(Onset of Queuing)



Network Buffers Fill

Network Buffers Drain

CUBIC assessment

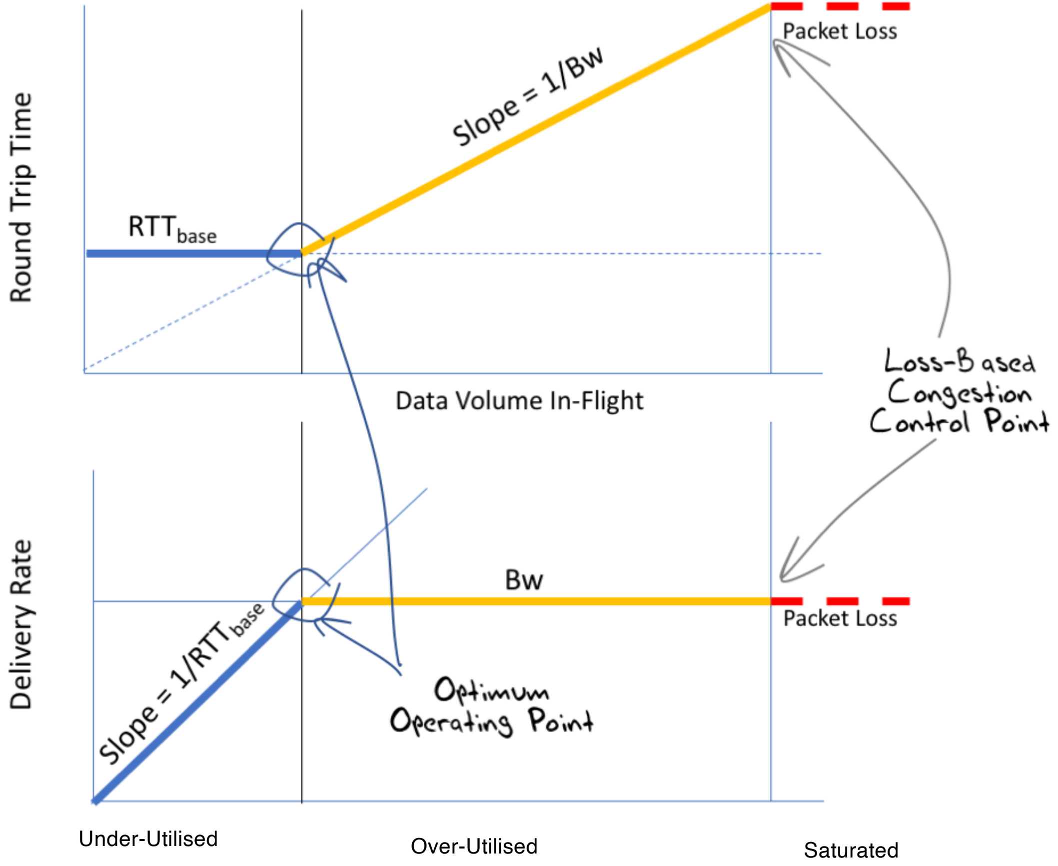
- Can react quickly to available capacity in the network
- Tends to sit for extended periods in the phase of queue formation
- Can react efficiently to long fat pipes and rapidly scale up the sending rate
- Operates in a manner that tends to exacerbate ‘buffer bloat’ conditions, but also operates efficiently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

Can we do even better?

- Lets look at the model of the network once more, and observe that there are three ‘states’ of flow management in this network:
  - 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



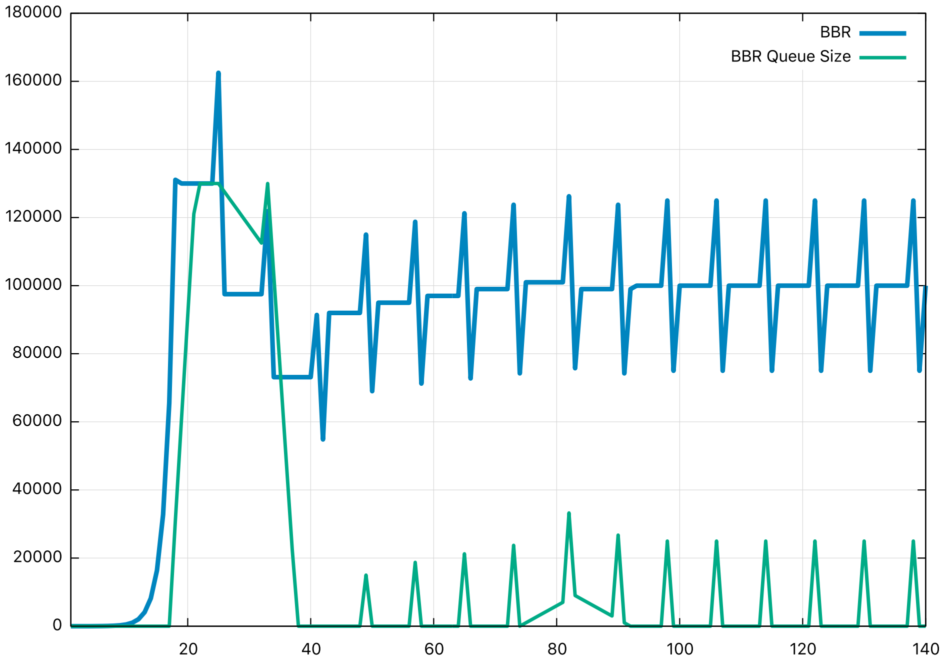
How to detect the onset of queuing?

- By carefully measuring the Round Trip Time!

BBR Design Principles

- Probe the path capacity only intermittently
- Probe the path capacity by increasing the sending rate for a short interval and then drop the rate to drain the queue:
  - If the RTT of the probe 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
- Pace the sending packets to avoid the need for network buffer rate adaptation

Idealised BBR profile



sending rate

network queues

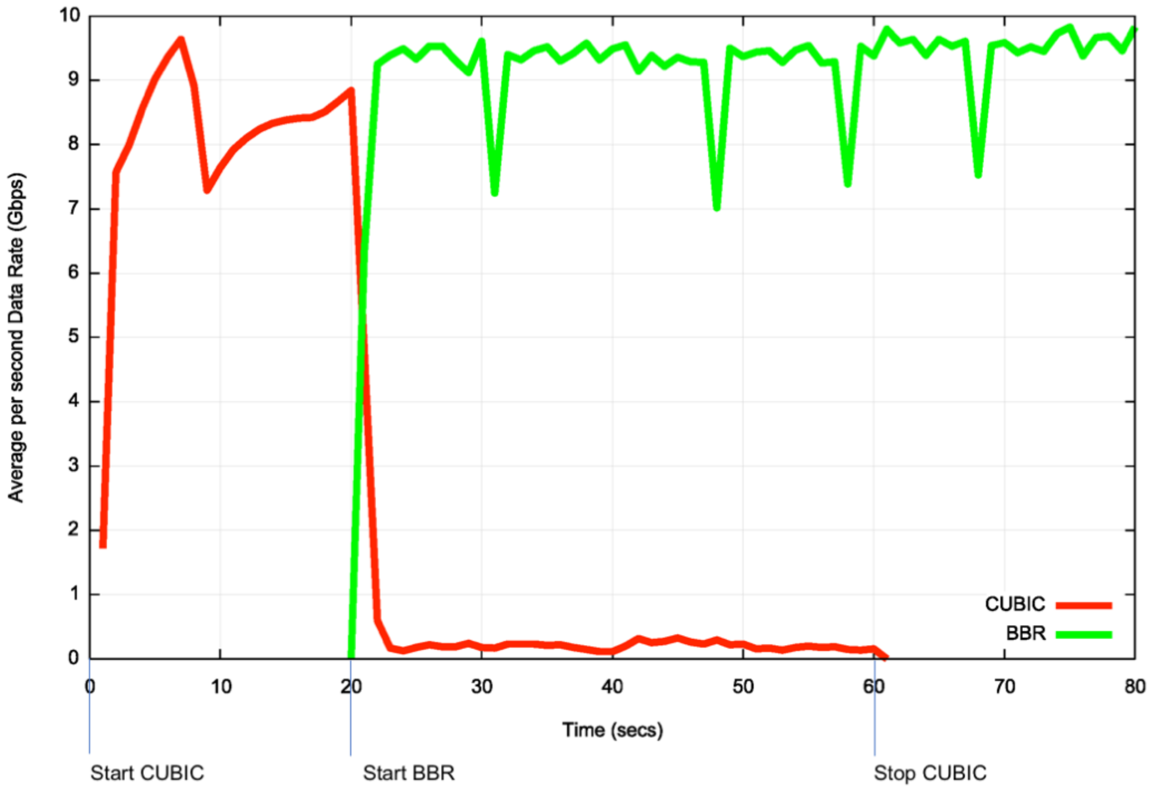
BBR Politeness?

- 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

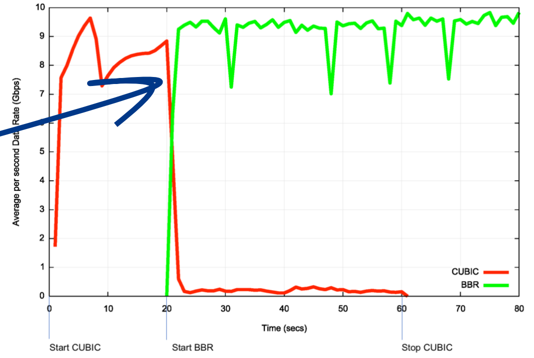
From Theory to Practice

- Lets use BBR in the wild
- I'm using iperf3 on Linux platforms (Linode)
  - The platforms are dedicated to these tests
- It's the Internet
  - The networks paths vary between tests
  - The cross traffic is highly variable
  - No measurement is repeatable to a fine level of detail

Cubic vs BBR over a 12ms RTT 10G circuit



Wow!



- That was BRUTAL!
- As soon as BBR started up it collided with CUBIC, and BBR startup placed pressure on CUBIC such that CUBIC's congestion window was reduced close to zero
- At this stage CUBIC's efforts to restart its congestion window appear to collide with BBR's congestion control model, so CUBIC remains suppressed
  – The inference is that BBR appears to be operating in steady state with an ability to crowd out CUBIC

BBR vs Cubic – second attempt

Same two endpoints, same network path across the public Internet

Using a long delay path AU to Germany via the US

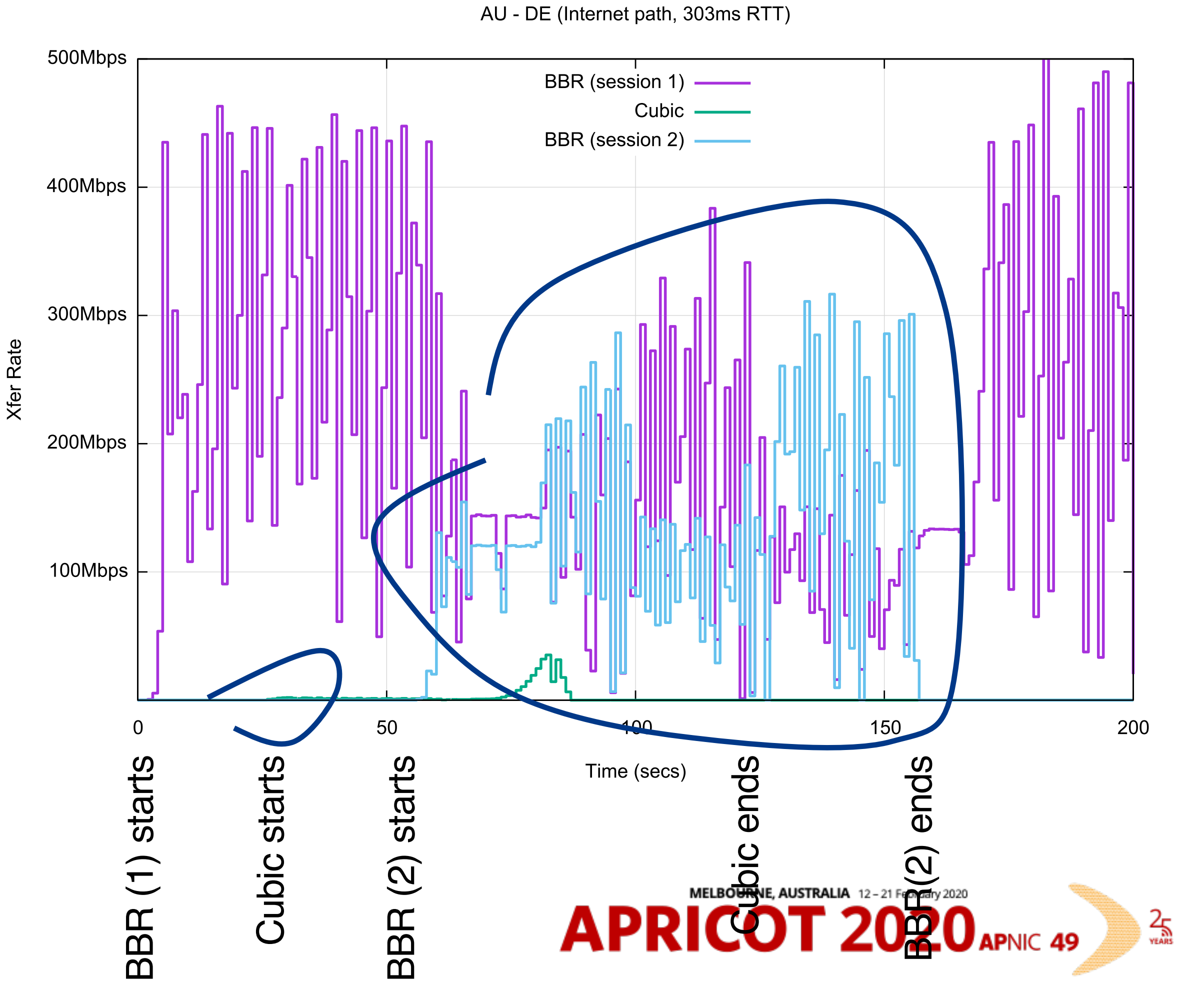
```
gih@wally:~$ traceroute testbed-de.rand.apnic.net
traceroute to testbed-de.rand.apnic.net (172.104.147.241), 30 hops max, 60 byte packets
 1  202.158.221.221 (202.158.221.221)  0.412 ms  0.410 ms  0.401 ms
 2  et-0-3-0.pe1.rsby.nsw.aarnet.net.au (113.197.15.10)  4.290 ms  4.294 ms  4.312 ms
 3  113.197.15.157 (113.197.15.157)  4.265 ms  4.273 ms  4.332 ms
 4  xe-0-2-4.bdr1.a.sjc.aarnet.net.au (202.158.194.162)  150.825 ms  150.828 ms  150.823 ms
 5  208.185.52.77.available.above.net (208.185.52.77)  150.930 ms  150.931 ms  150.926 ms
 6  ae12.cr1.sjc2.us.zip.zayo.com (64.125.25.21)  151.439 ms  151.165 ms  151.186 ms
 7  ae16.mpr3.sjc7.us.zip.zayo.com (64.125.31.13)  155.216 ms  151.696 ms  151.594 ms
 8  ix-ae-5-0.tcore1.SQN-San-Jose.as6453.net (63.243.205.9)  151.945 ms  151.966 ms  151.96
 9  if-ae-12-2.tcore1.NTO-New-York.as6453.net (63.243.128.28)  304.046 ms  304.074 ms  304.
10  if-ae-7-2.tcore1.N0V-New-York.as6453.net (63.243.128.26)  292.651 ms  292.585 ms  292.7
11  if-ae-2-2.tcore2.N0V-New-York.as6453.net (216.6.90.22)  294.608 ms if-ae-32-2.tcore2.LD
12  if-ae-15-2.tcore2.L78-London.as6453.net (80.231.131.117)  296.843 ms  296.812 ms if-ae-
13  if-ae-14-2.tcore2.AV2-Amsterdam.as6453.net (80.231.131.161)  301.732 ms  301.484 ms  29
14  if-ae-2-2.tcore1.AV2-Amsterdam.as6453.net (195.219.194.5)  293.261 ms if-ae-3-2.tcore1.
15  if-ae-11-2.tcore1.PVU-Paris.as6453.net (80.231.153.49)  308.666 ms if-ae-6-2.tcore1.FNM
16  if-ae-9-2.tcore2.FNM-Frankfurt.as6453.net (195.219.87.13)  307.218 ms if-ae-2-2.thar1.F
17  195.219.61.30 (195.219.61.30)  287.745 ms if-ae-14-3.thar1.F2C-Frankfurt.as6453.net (19
18  139.162.129.2 (139.162.129.2)  289.321 ms 139.162.129.11 (139.162.129.11)  294.369 ms 1
19  li1663-241.members.linode.com (172.104.147.241)  303.581 ms  303.558 ms 139.162.129.15
```

BBR vs Cubic

The Internet is capable of offering a 400Mbps capacity path on demand!

In this case BBR is apparently operating with filled queues, and this crowds out CUBIC

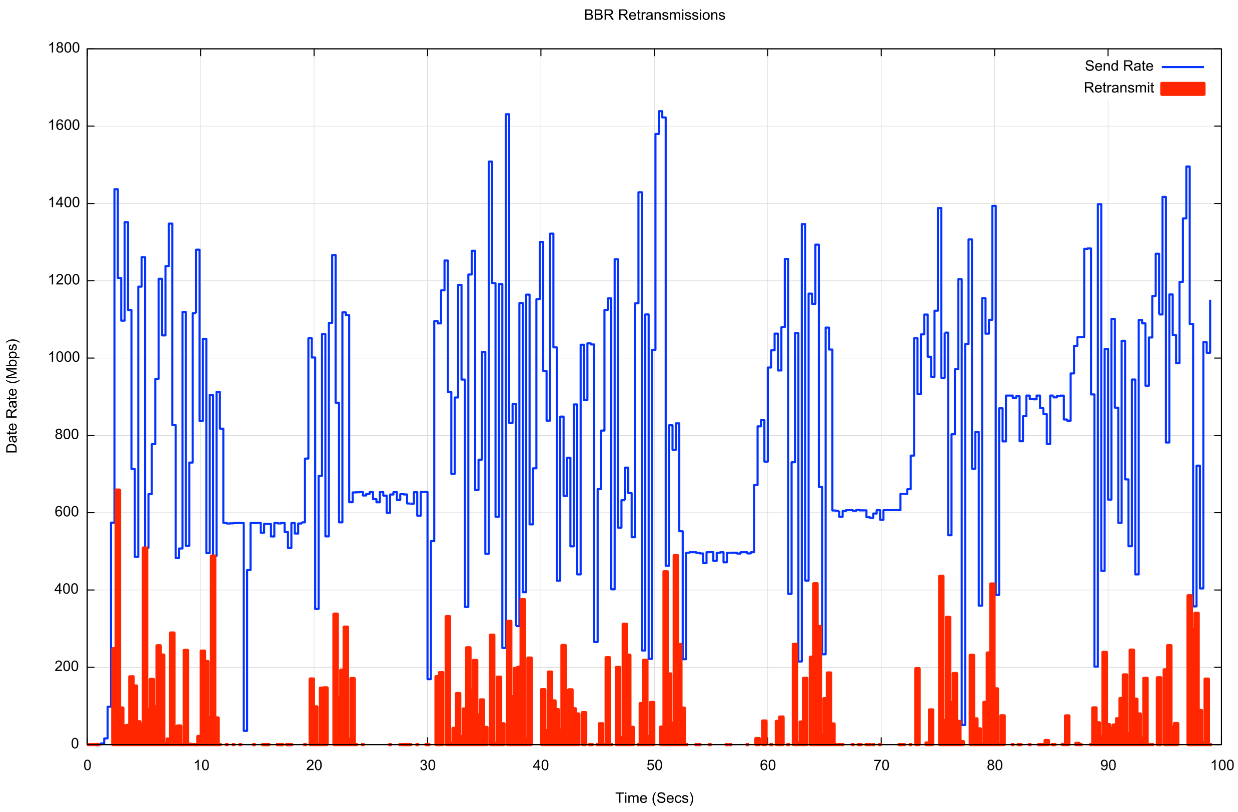
BBR does not compete well with itself, and the two sessions oscillate in getting the majority share of available path capacity



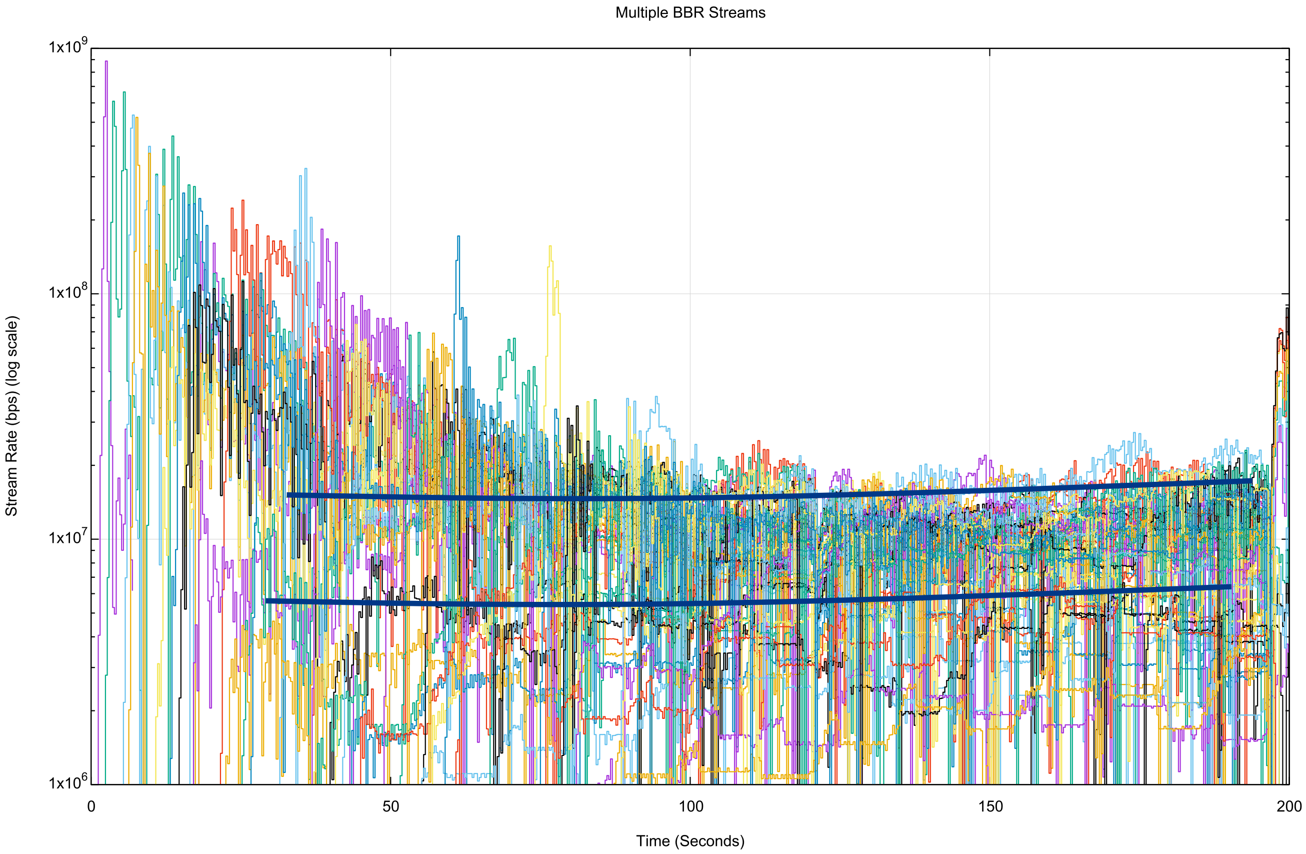
BBR and Loss Recovery

Packet loss causes retransmission that appears to occur in addition to the stable link capacity model used by BBR.

Once loss is reduced, BBR maintains a more consistent sending model



Parallel BBR Streams



We used 50 parallel BBR streams between the same two endpoints (200ms RTT)

In this larger setup the majority of sessions managed to equilibrate between each other and evenly share the path bandwidth

So what can we say about BBR?

It's "interesting" in so many ways:

- It's a move away from the more common loss-based flow control protocols
- It looks like it will operate very efficiently in a high-speed small-buffer world
  - High speed small buffer switching chips are far cheaper, but loss-based TCP reacts really badly to small buffers by capping its flow rate
- It will operate efficiently over ECMP paths, as it is relatively impervious to packet re-ordering
- It also looks as if it will operate efficiently in rate policed environments
- Unlike AIMD systems, it will scale from Kbps to Gbps over long delay paths very efficiently
- It resists the conventional network-based traffic control mechanisms

Why use BBR?

- Because it achieves
- Its incredibly efficient
- It makes minimal demands on network buffer capacity
- It's fast!

Why **not** use BBR?

- Because it **over achieves**!
- The classic question for many Internet technologies is scaling – "what if everyone does it?"
  - BBR is not a scalable approach in competition with loss-based flows
  - It works so well while it is used by just a few users, some of the time
  - But when it is active, BBR has the ability to slaughter concurrent loss-based flows
  - Which sends all the wrong signals to the TCP ecosystem
    - The loss-based flows convert to BBR to compete on equal terms
    - The network is then a BBR vs BBR environment, which is unstable

Is this BBR experiment a failure?

Is it just too ‘greedy’ and too ‘insensitive’ to other flows to be allowed out on the Internet to play?

- Many networks have been provisioned as a response to the aggregate behaviours of loss-based TCP congestion control
- BBR changes all those assumptions, and could potentially push many networks into sustained instability
- We cannot use the conventional network control mechanisms to regulate BBR flows
  - Selective packet drop just won’t create back pressure on the flow

Is BBR an outstanding success?

- We can’t achieve speed if we need also large high speed buffers in network routers
  – Loss-based flow-control systems have a sloppy control loop that is always ½ RTT late
- We can use small buffers in switches if we use sender pacing coupled with flow control systems that are sensitive to the onset of queue formation (rather than being sensitive to packet loss resulting from a full queue)
- BBR points to an approach that does not require large buffer pools in switches

Where now with BBR?

BBR 2.0

- Alter BBR's 'sensitivity' to loss rates, so that it does not persist with an internal bandwidth delay product (BDP) that exceeds the uncongested BDP

  This measure would moderate BBR 1.0's ability to operate for extended periods with very high loss levels

- Improve the dynamic sharing fairness by moderating the Bandwidth Delay Product by using an estimated 'fair' proportion of the path BDP

- Accommodate the signal distortion caused by ACK stretching middleware

- Place an upper bound on the volume of in-flight data

- Alter the +/- 25% probe factors dynamically (i.e. allow this to be less than 25% overload)

The new Network Architecture

- We are seeing a shift in end systems to assert edge-centric control and hide from network-level active middleware in the Internet
- QUIC and BBR are instances of a recent push back from the network-level QoS bandwidth control mechanisms, and result in greater levels of autonomous control being passed back to the end hosts
- For better or worse!

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 high speed networks requires a different approach to what we are doing today
- BBR seems to be a step in an interesting direction, particularly for very high speed networking
- But it still calls for more research and more testing at scale

Tiny Buffers

- Buffers in a network serve two major purposes:
  - smooth sender burstiness
  - Multiplexing
- What 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

This would allow Tbps switches to operate with on-chip memory (10’s Mb) and still allow highly efficient network utilisation

That's it!

Questions?

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