BBR Update:

1: BBR.Swift; 2: Scalable Loss Handling

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Outline

- BBR.Swift: using delay as a signal in the datacenter
- Scalable loss handling: BBR, PRR, and the scalability of multiplicative decrease
- Status of the BBR code and Google deployment
- Conclusion

Target for this talk:

- Sharing our experience with experiments
- Inviting the community share feedback, test results, issues, patches, or ideas
BBR.Swift: Delay as a signal in the datacenter
Swift Congestion Control

- Swift congestion control algorithm [SIGCOMM-2020]
  - Uses network RTT and host delays as primary signals (also uses loss)
  - Potential scope:
    - Inside network with known topology/RTT properties (e.g. datacenters)
    - With NICs with hardware timestamp support
    - Where all traffic sharing bottleneck is Swift-compatible
  - Algorithm:
    - AIMD, with MD (multiplicative decrease) proportional to the excess delay
    - Uses pacing when cwnd < 1, to handle large-scale multiplexing (e.g. "incast")
  - Usage so far:
    - In Snap userspace network stack [SOSP-2019]
    - Traffic within a Google datacenter
      - Target network RTT is known
      - Known that other traffic sharing QoS queue is Swift-compatible
Swift: Delay as a Congestion Signal

- Advantages of using delay as a signal:
  - Provides richer information about the current degree of queuing
    - Allows faster reaction to long queues
    - Avoids overreaction and underutilization with short queues
  - Provides a known target latency for engineering systems
    - Helps applications to predict the latency they should expect
    - Helps set SLOs of network queuing delay for engineering, monitoring, alerting
    - By contrast, loss rate or ECN monitoring is difficult to translate into application performance (e.g. how does x% loss relate to latency?)

- Key requirement: **accurate** delay measurements of network and host
Data sender computes a Network\_RTT sample using:
- \( \text{Network\_RTT} = \text{Total\_RTT} - \text{Receiver\_ACK\_Delay} \)

- Data and ACK transmission times: measured by TCP
- Data and ACK reception times: measured by NIC (via NIC hardware rx timestamps)
- Data receiver signals \text{Receiver\_ACK\_Delay} using new timestamp option ([draft-yang-tcpm-ets-00](https://datatracker.ietf.org/doc/html/draft-yang-tcpm-ets-00), IETF 109 TCPM)
BBR.Swift: Design

- BBR.Swift is an extension of BBRv2
  - Core BBRv2 unchanged
  - Extension to BBRv2 based on Swift [SIGCOMM-2020]
  - New configuration parameter: target_RTT (e.g. O(100us) inside DC)
  - If Network_RTT > target_RTT, multiply cwnd and pacing rate by:
    \[ MD = \max\left(1 - 0.8 \times \frac{\text{Network}_\text{RTT} - \text{target}_\text{RTT}}{\text{Network}_\text{RTT}}, 0.5\right) \]

- DCTCP-style ECN response is disabled if target_RTT is used
  - For interactions between BBR.ECN WAN flows vs BBR.Swift flows:
    ■ Exploring ideas, e.g.: dynamically set target_RTT to the Network_RTT at the boundary of ECN marking
BBR.Swift: Example Performance Results in the Lab

- 3 server-class machines w/ 50G NICs on same switch; controlled lab network setting
- 2 senders, 1 receiver; 2000 bulk TCP flows;
- Each sender sends 1000 bulk netperf TCP_STREAM flows for 10 secs
  - Second sender starts ~0.2 sec after first sender
- Results reported for each data sender machine

- DCTCP issues:
  - Floor is cwnd=1
  - Big standing queue
  - Large loss rate
  - Less fair

- BBR.Swift features:
  - Pacing ~= delivery rate
  - Small queue
  - Low loss rate
  - Quite fair

<table>
<thead>
<tr>
<th>Congestion Control</th>
<th>DCTCP</th>
<th>BBRv2 (ECN)</th>
<th>BBR.Swift</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sender machine</td>
<td>1</td>
<td>2</td>
<td>1</td>
</tr>
<tr>
<td>Throughput (Gbit/sec)</td>
<td>46.6</td>
<td>3.7</td>
<td>25.0</td>
</tr>
<tr>
<td>Mean Total_RTT (us)</td>
<td></td>
<td></td>
<td>228</td>
</tr>
<tr>
<td>Mean Network_RTT (us)</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Retransmit rate</td>
<td>6.1%</td>
<td>66.2%</td>
<td>1.6%</td>
</tr>
<tr>
<td>Jain's fairness index</td>
<td>0.671</td>
<td>0.746</td>
<td>0.69</td>
</tr>
</tbody>
</table>
Preparing for production testing of BBR.Swift for Google internal traffic

- Plan to release BBR.Swift code as open source and document the algorithm in detail
  - Including implementation of [draft-yang-tcpm-ets-00](https://www.rfc-editor.org/rfc/draft-yang-tcpm-ets-00)

- Goal: we want transports to be able to use BBR.Swift as their CC
  - On connections where...
    - Target Network_RTT is known
    - It is known that other traffic sharing bottlenecks is using Swift or BBR.Swift
  - On physical machines or virtual machines
Congestion Control in Loss Recovery
Traditional TCP CC uses multiplicative decrease upon round trips with packet loss
  - e.g., Reno: 0.5x per round; CUBIC: 0.7x per round
But what if the bandwidth available to a flow suddenly drops by 1000x?
  - Theory: Reno expects log_2(old_bw / new_bw) round trips of high loss rate pain
    - e.g. 1000x cut in fair-share bandwidth can lead to 10 rounds of high loss
  - Reality:
    - With TCP loss recovery before RACK, consecutive rounds of loss => RTO
      - Before RACK, lost retransmits usually caused RTO, cwnd=1, slow-start
    - With TCP RACK but no PRR, reality matches theory
      - And the resulting high loss rate can be painful!
    - With TCP RACK and PRR, sending rate is bounded to be near delivery rate
      - And the loss rate is reasonable and tolerable

Multiplicative Decrease is Not Scalable Enough
How Should BBR Respond to Loss?

- BBRv1 used an approach inspired by PRR
  - First round of recovery: packet conservation
  - Subsequent rounds of recovery: send at 2x the per-packet delivery rate
- Initial revisions of BBRv2 simplified things by using a multiplicative decrease
  - Similar to 0.7x per round used by CUBIC
- Production experience with Google-internal datacenter RPC traffic shows initial BBRv2...
  - Approaches the painful theoretical behavior: high loss for log(old_bw / new_bw)
- Next: experimenting with various PRR-inspired responses to loss recovery for BBRv2
  - Stay tuned for production experiment results...
Wrapping up...
Status of BBR v2 algorithm and code

● TCP BBRv2 "alpha/preview" release:
  ○ Linux TCP (dual GPLv2/BSD): [github.com/google/bbr/blob/v2alpha/README.md](https://github.com/google/bbr/blob/v2alpha/README.md)

● QUIC BBR v2:
  ○ Chromium QUIC (BSD): on chromium.org in bbr2_sender. { cc, h }

● BBR v2 release is ready for research experiments
  ○ We invite researchers to share...
    ■ Ideas for test cases and metrics to evaluate
    ■ Test results
    ■ Algorithm/code ideas
  ○ Always happy to see patches or look at packet traces...

● BBR v2 algorithm was described at IETF 104 [slides | video]
● BBR v2 open source release was described at IETF 105 [slides | video]
YouTube and google.com: deployed for a small percentage of users
  ○ Reduced queuing delays: RTTs lower than BBR v1 and CUBIC
  ○ Reduced packet loss: loss rates closer to CUBIC than BBR v1

Google-internal traffic:
  ○ BBRv2 being deployed as default TCP congestion control for internal Google traffic
  ○ Used as the congestion control for most traffic within Google
    ■ Currently using bandwidth * min_rtt, loss, ECN as signals
    ■ Preparing for production testing using Network_RTT

Continuing to iterate using production experiments and lab tests
Conclusion

- Actively working on BBR v2, BBR.Swift at Google
  - Tuning performance to enable full-scale roll-out at Google
  - Improving the algorithm to scale to larger numbers of flows
  - We invite the community share test results, issues, patches, or ideas
Internet Drafts, paper, code, mailing list, talks, etc.

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Backup slides...
ECN: Responsiveness vs Utilization Trade-Offs

- **[RFC3168]-style ECN:**
  - Coarse once-per-RTT signal forces high queues or low utilization

- **DCTCP/Prague-style ECN:**
  - Per-ACK ECN signal is much better than [RFC3168], but still has a dilemma...
  - How to choose the magnitude of the multiplicative decrease (MD)?
  - MD based on EWMA of ECN mark rate ([DCTCP, TCP Prague, BBRv2]):
    - Delays response to suddenly long queues
  - MD using instantaneous ECN mark rate in current round trip ([GCN]):
    - Allows fast reaction to long/congested queues
    - But if queue is actually short or RTT is long, a big MD causes underutilization

- Ideally we'd like:
  - An MD proportional to a per-ACK signal that quantifies the magnitude of the queue
  - Adapts to use a large MD to drain long queues quickly
  - Adapts to use a small MD to allow high utilization even for short queues
Extensible Timestamp option (ETS): [draft-yang-tcpm-ets-00](https://tools.ietf.org/html/draft-yang-tcpm-ets-00) (see TCPM at IETF 109)
- Has TSval, TSecr, like [RFC7323](https://tools.ietf.org/html/rfc7323) timestamp option, but...
- Explicitly signals Receiver_ACK_Delay
- TSval, TSecr, Receiver_ACK_Delay are all in microsecond granularity

Data sender computes Total_RTT using ETS timestamp options:
- Total_RTT = (time NIC received ACK) - (scheduled pacing release time of data)

Data receiver computes Receiver_ACK_Delay:
- Receiver_ACK_Delay = (scheduled release time of ACK) - (time NIC received data)
- Data receiver sends Receiver_ACK_Delay using ETS timestamp option
Kind: 1 byte, value 254, [RFC6994] experimental option
Length: 1 byte option length, value 16 if SYN bit is set, otherwise 14 (value MAY be higher in later versions).
ExID: 2 bytes, [RFC6994] experiment ID: value 0x4554.
TSVal and TSecr: 32 bits each, have the same definition as [RFC7323] but are in microseconds.
EcrDelUnit: 2 bits; allowed values are:
0: indicates EcrDel is in microsecond units
1: indicates EcrDel is in millisecond units
2: indicates EcrDel is invalid (should be ignored)
3: reserved in this protocol version
EcrDel: 13 bits, the value of EcrDel.
Reserved: 1 bit, in this protocol version, sender MUST set to 0
And receiver MUST ignore.
MaxACKDel: 16 bits, max expected ACK delay in microseconds,
only present in SYN.