

SCReAMv2

IETF-119 CCWG

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Topics

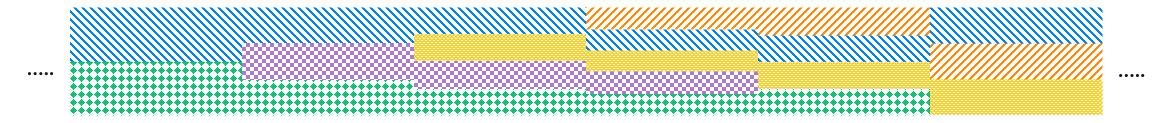


- 5G properties
- Video coders
- SCReAMv2
- Source and experiments

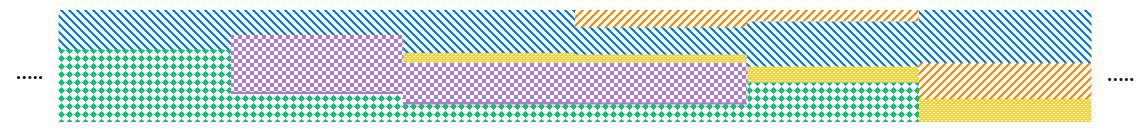
5G properties

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- Resource allocation in frequency and time (average)
 - End user applications may be bitrate limited
 - Resource allocation can drop in a few RTTs, or in an instant when other users enter in the same cell



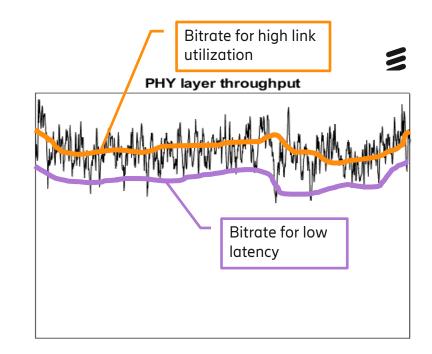
- Actual throughput (average)
 - Modulation and Coding Scheme (MCS) varies with channel quality, power limitation in uplink
 - Result, varying throughput

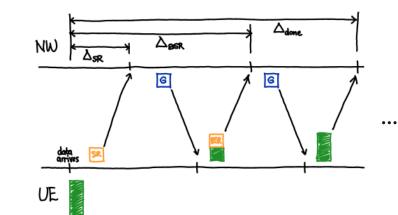


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5G properties

- Cellular transport is subject to fast fading → throughput varies on short time scale
 - Trade off between large network buffer, high link utilization and small network buffer, reduced link utilization, pick one
 - Throughput can drop quickly
 - Large dynamic range in throughput
- (Dynamic) uplink scheduling of intermittent data
 - Increased delay
 - Reduced link utilization
- Delay can occur due to
 - Congestion, scheduling
 - Hand over, battery saving (DRX)
 - Retransmission on lower radio protocol stack layers





UE = terminal NW = network SR = Scheduling request G = Scheduling grant BSR = Buffer Status Report

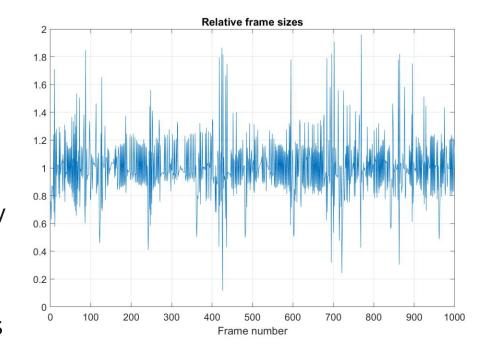
Fr 1 Fr 2 Fr 3 Fr 4

Video coder aspects

Frame sizes typically vary

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- I-Frames can be large \rightarrow GDR (Gradual Decoding Refresh) highly recommended
- Also P-frames vary in size
- Additional headroom required to cope with varying frame sizes → avoid excessive queue build-up for large frames
- Video has a max bitrate → congestion control becomes application limited
- Odd features: Systematic error in output rate, slow rate change.



Example, video frames transmission over a constrained bottleneck

Fr 1 Fr 2 Fr 5 Fr 7 Fr 3 Fr 4 Fr 6

One frame should ideally be transmitted before a new frame is generated

SCReAMv2

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- Eventually obsoletes RFC8298
- RTCP : RFC8888
- Congestion control based on
 - Estimated one way delay (similar to LEDBAT CC)
 - Packet loss detection
 - Classic ECN
 - L4S (main focus)
- Sender transmission control soft limits bytes in flight
 - Max bytes in flight = 1.5x CWND
- Media bitrate is (mainly) based on CWND and RTT
- Packet pacing rate > 1.5x media bitrate
- Congestion window validation

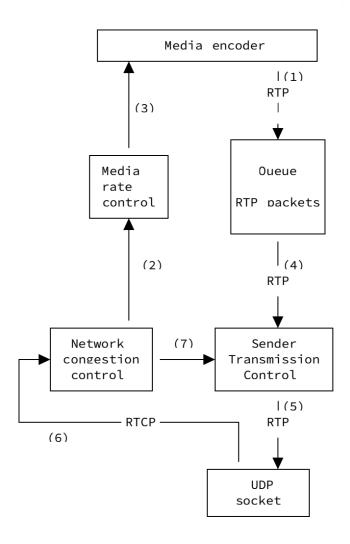


Figure 1: Sender Functional View

Congestion window update



On congestion event, max once per RTT

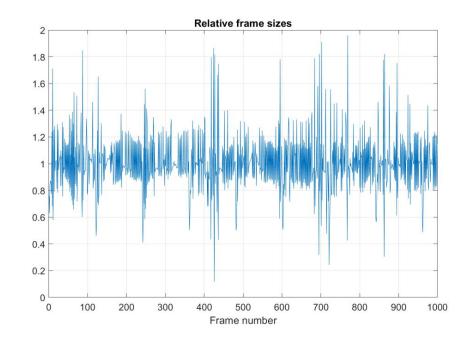
- Packet loss or classic ECN-CE marking
 - Reduce CWND by fixed value
- Delay based congestion control
 - Average of estimated queue delay to avoid over-reacting to non-congestion events like hand over
 - Virtual L4S marking based back CWND decrease when average queue delay > target delay/2
- L4S based congestion control
 - Reduce CWND proportional to fraction of packets marked (like Prague)
 - Very small CWND → limit reduction

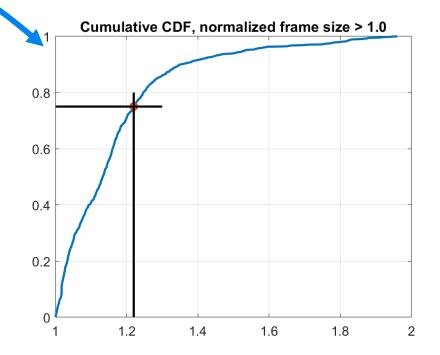
Once per RTT

- Increase CWND similar to Prague(Reno)
 - One MSS / RTT
 - In addition, multiplicative increase if uncongested
 → faster convergence to increased capacity
 - Very small CWND → limit increase
- Non-L4S. Stabilize CWND with inflexion point, similar to Cubic.
- Validate CWND for the case that max media bitrate is reached.

Target bitrate update

- Executed when congestion window is updated
- targetRate = CWND*8/RTT
 CNWD [byte], RTT [s]
- Additional down-scaling based on 75%-ile of normalized frame size
- For not-L4S: scale down additionally when bytes in flight > CWND
- For very small CWND : Scale down target rate slightly

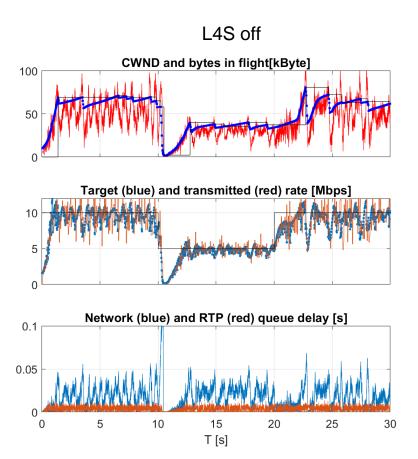


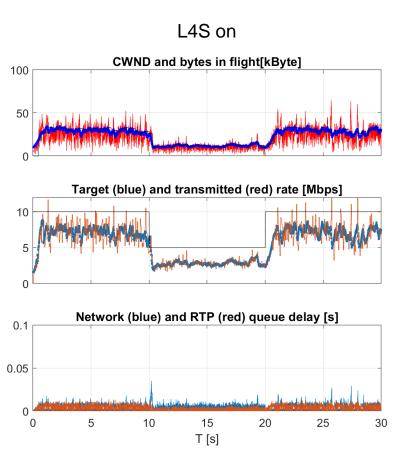






- Congestion control is delay based when L4S is not enabled or marking does not occur
- Delay based CC should be sufficiently responsive but should not over-react on non-congestion events
- Note: Bytes in flight can exceed CWND

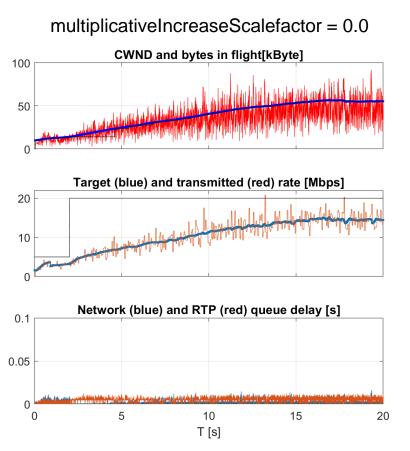




Multiplicative increase



- Multiplicative increase gives faster convergence when link throughput increases
- 0.05 = up to 5% CWND increase per RTT



multiplicativeIncreaseScalefactor = 0.05 CWND and bytes in flight[kByte] 100 Target (blue) and transmitted (red) rate [Mbps] Network (blue) and RTP (red) queue delay [s] 0.05

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T [s]

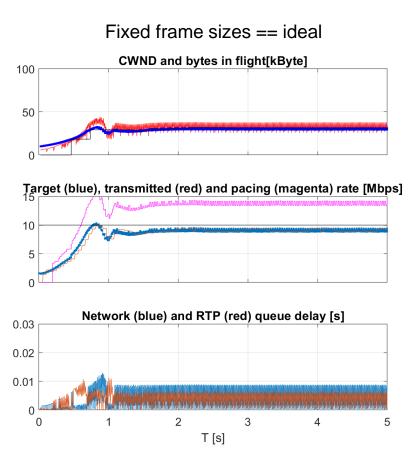
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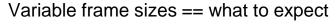
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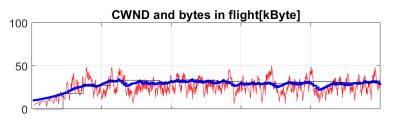
Packet pacing

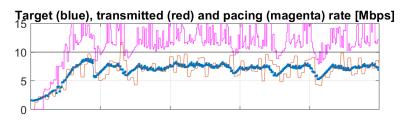
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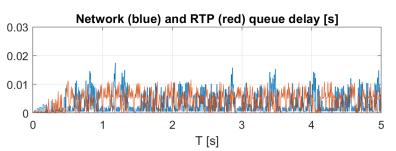
- Default pacing overhead is 50%, i.e pacing rate is 1.5x media bitrate
- In addition: Large frames→ pace even faster
- Objective : Avoid that large frames are held unnecessarily in RTP queue







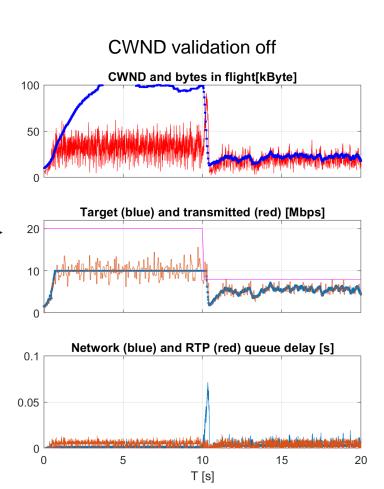


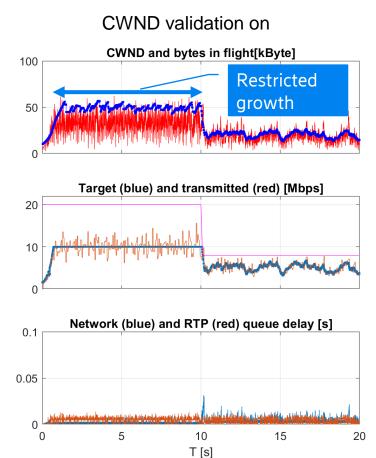


Congestion window validation



- SCReAM can often be application limited
 - Max video bitrate reached
- Congestion window growth becomes restricted when SCReAM becomes application limited
- Example, max target rate is 10Mbps
- Note, relax the restriction when uncongested <u>and</u> max target bitrate <u>not</u> reached





Sources



- IETF draft : https://datatracker.ietf.org/doc/draft-johansson-ccwg-rfc8298bis-screamv2/
- Code : https://github.com/EricssonResearch/scream
 - Continuously developed since March 2015
 - SCReAM code
 - BW test application, the plumber's aid
 - Multicamera gstreamer and C++ wrapper (multicam)
 - Complete gstreamer with multicam support (gstscream)

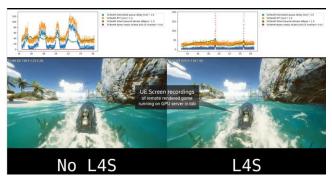
Experimentation, so far pretty much L4S-ish

- Small RC cars
 - https://www.youtube.com/watch?v=RZmS10djDEg
- Large RC Cars
 - https://www.youtube.com/watch?v=H8CBOKgHTOQ
- Boat Attack cloud rendered gaming
 - https://www.ericsson.com/en/news/2021/10/dt-and-ericssonsuccessfully-test-new-5g-low-latency-feature-for-time-criticalapplications
- On the wish list, integration into WebRTC









Questions, comments?



