

SELF-CLOCKED RATE ADAPTATION FOR CONVERSATIONAL VIDEO IN LTE

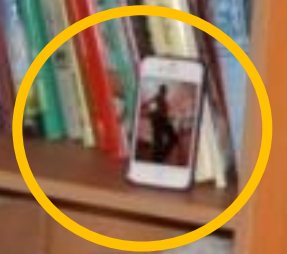
CSWS'14

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It's not only about bitrate

Latency is important too !!

Julia rehearsing for a figure skating show
with her friend Hilda over Skype



ENDPOINT CONGESTION CONTROL



- › Two types
 - Rate based congestion control
 - › GCC = Google CC (congestion control)
 - › MTSI (3GPP TS26.114)
 - Self-clocked congestion control
 - › TCP
 - › TFWC (**T**CP **F**riendly **W**indow based **C**ongestion Control)
 - › SCRaM (**S**elf-**c**locked **R**ate **A**daptation for **M**ultimedia)

ENDPOINT CONGESTION CONTROL, THE PROBLEM



- › Rate based CC seems to be unable to combat congestion in LTE well
- › Root cause, packet conservation principle is not followed.
- › ECN is one solution to problem

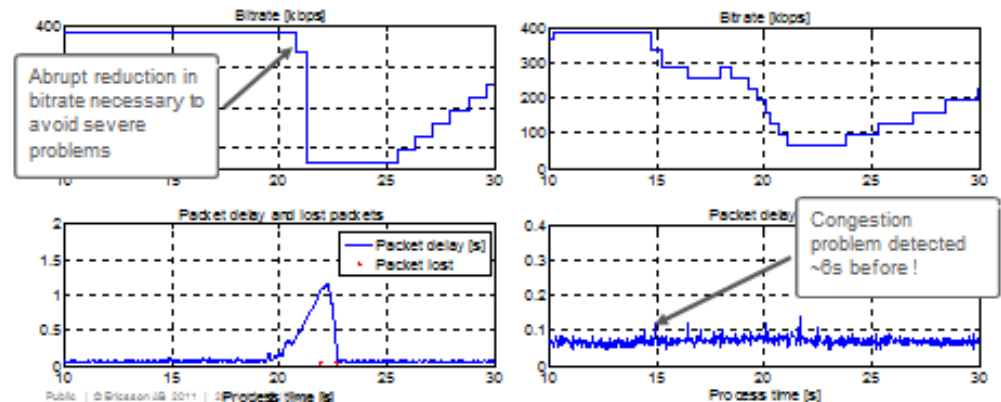
A COMPARISON *

Implicit methods are:

- › Slightly better than fixed bitrate
- › But damage is already done !
- › More sensitive tuning not helpful

Explicit methods (ECN) gives:

- › More in-advance warning
- › More users contribute to resolve congestion



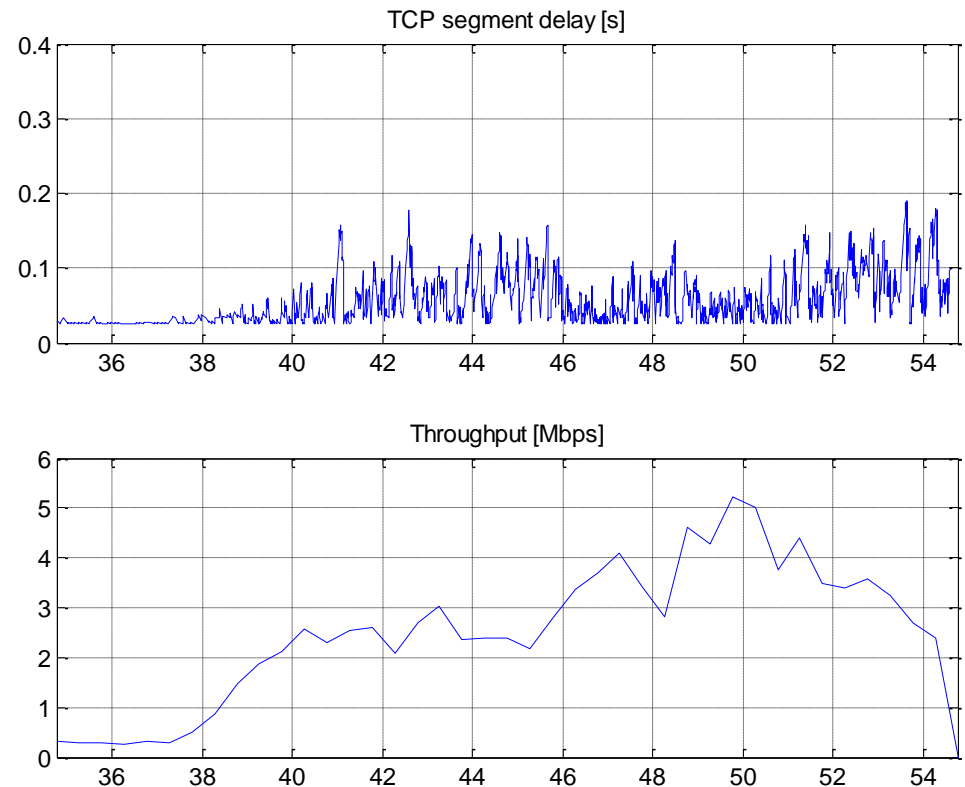
ECN :Explicit Congestion Notification

* <http://www.uppersideconferences.com/volte2011/volte2011program.html>
“Expanding VoLTE With Video Capabilities”

ENDPOINT CONGESTION CONTROL, THE SOLUTION ?



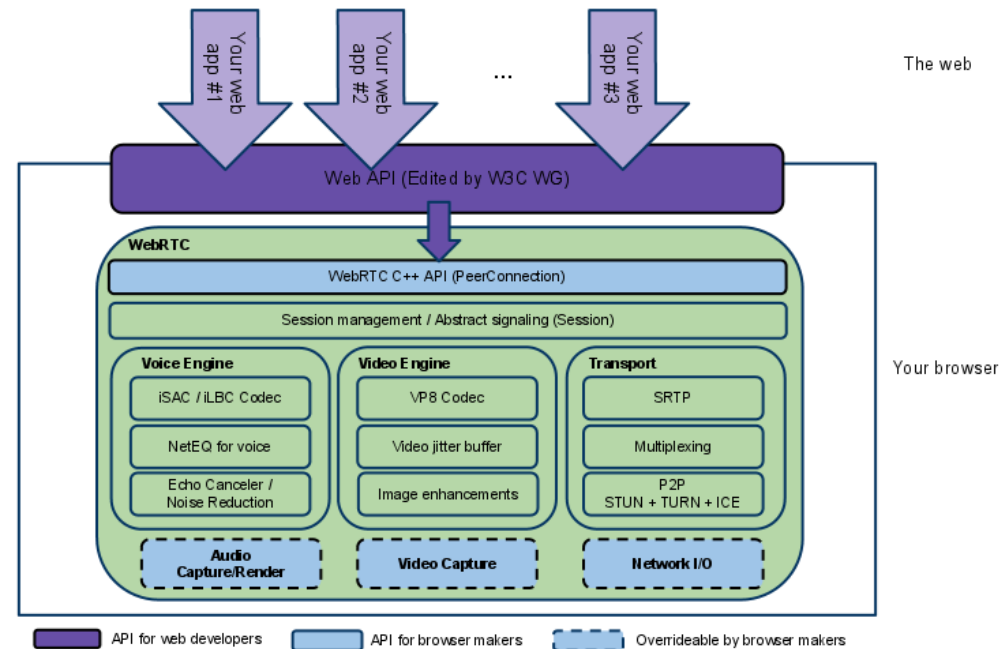
- › File transfer with TCP LEDBAT
- › Low latency despite quite a high bitrate !
- › Packet conservation principle followed
- › Can the same principles be used for conversational video ?



PROPERTIES OF CONVERSATIONAL MEDIA



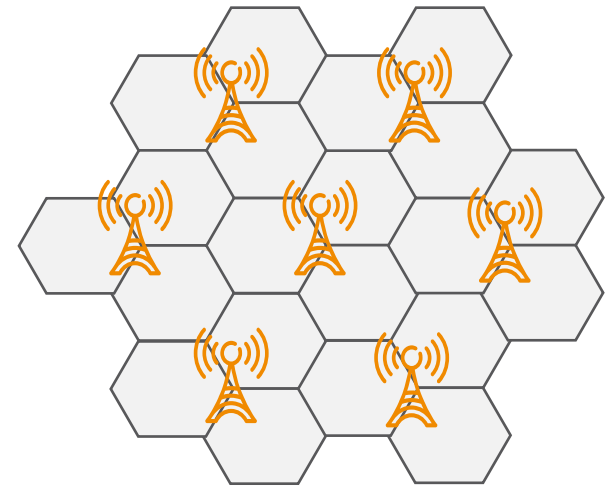
- › Example use case
 - WebRTC
- › Audio + video
- › Possibly also a data channel
- › Varying number of streams



PROPERTIES OF LTE CHANNELS



- › In general no guaranteed bitrate *
- › LTE bearers are allocated to individual terminals
 - Low statistical multiplexing
- › Throughput dictated by channel quality, number of users in cell and historical bitrate.
 - May change frequently, very little grace time to reduce bitrate



* QoS framework enables to prioritize e.g. for a given minimum bitrate

REQUIREMENTS ON A SOLUTION

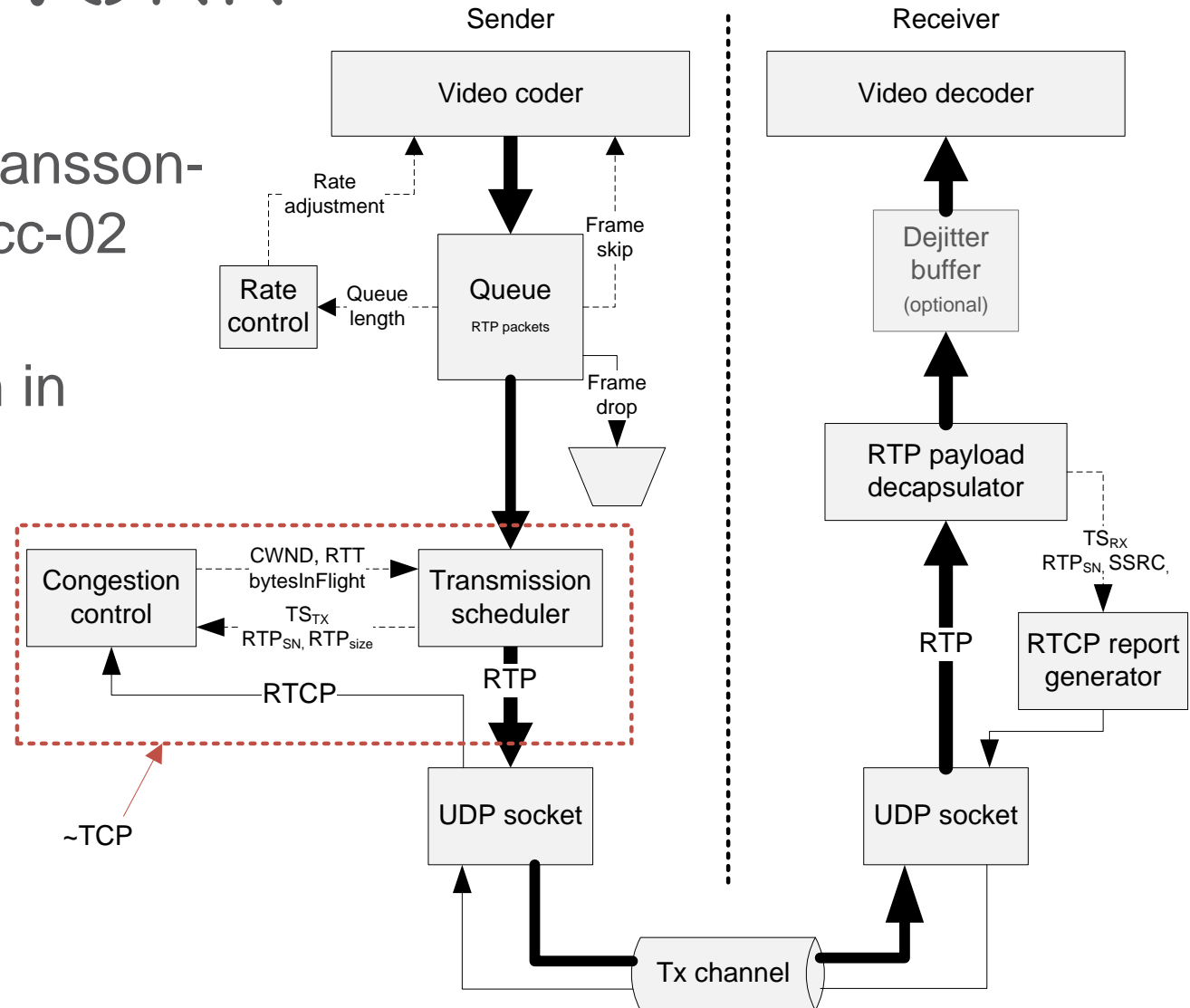


1. Low latency and packet loss
 2. Ability to coexist with other traffic over the same bearer
 3. Stable bitrate
- › #3 may be hard to fulfill given the two other requirements

THE ADAPTATION FRAMEWORK



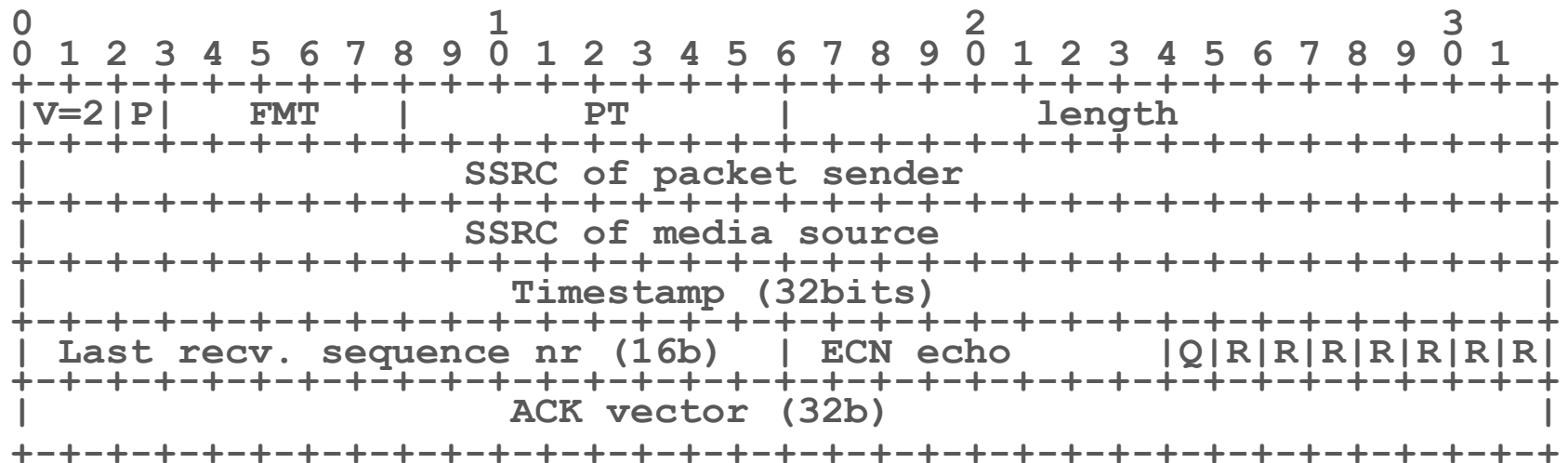
- › IETF : draft-johansson-rmcat-scream-cc-02
- › Chromium implementation in progress



RTCP FEEDBACK



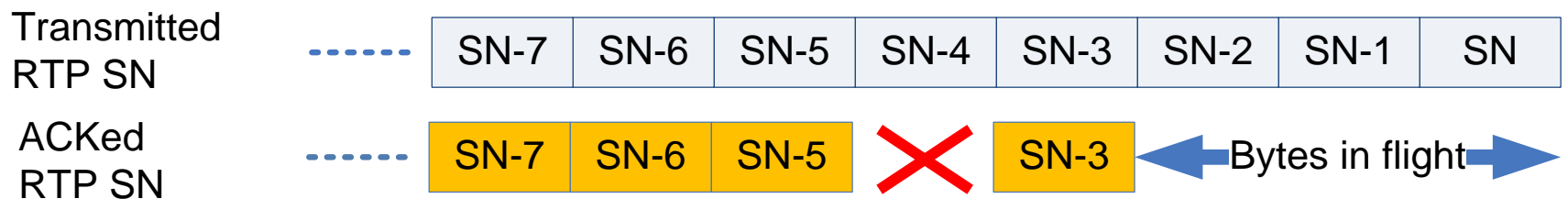
› RFC4585 Transport layer feedback message



BUZZ WORDS



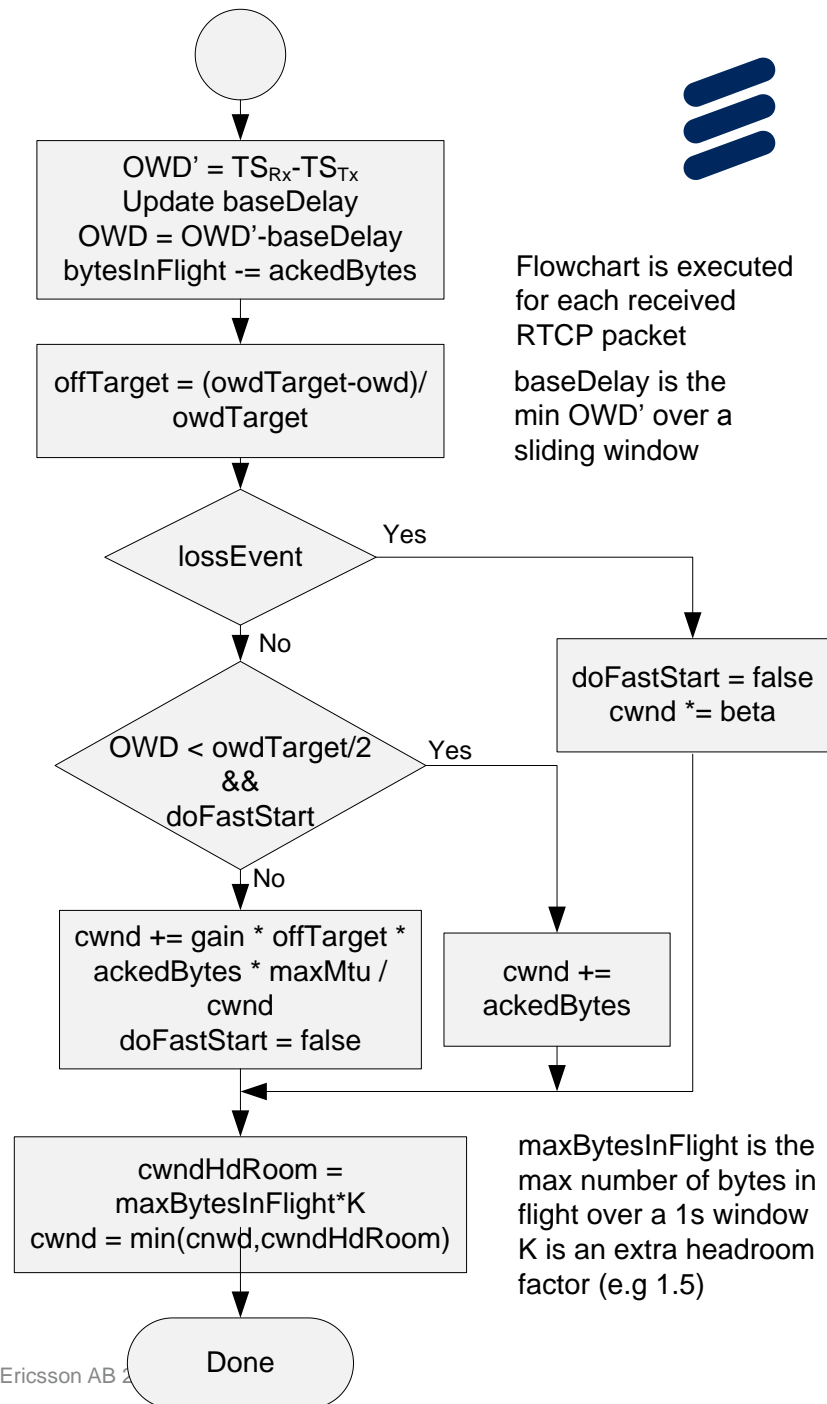
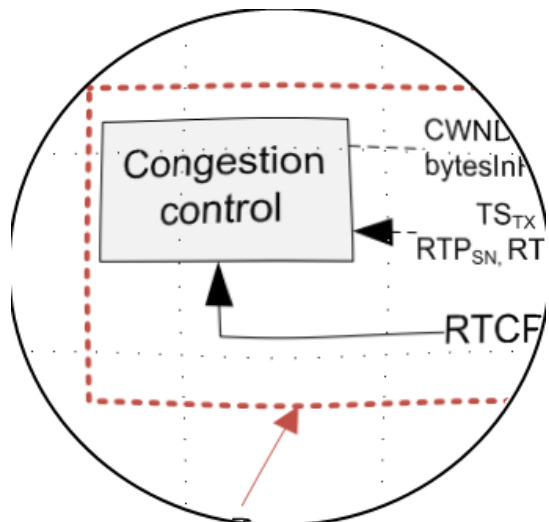
- › CWND : Congestion window, determines max number of bytes in flight, size depends on estimated network queuing delay and packet loss or ECN
- › Bytes in flight: The total size of the transmitted but not yet acknowledged RTP packets.



- › OWD : One way (extra) delay. An estimate of how much the queues in the network increase

CONGESTION CONTROL

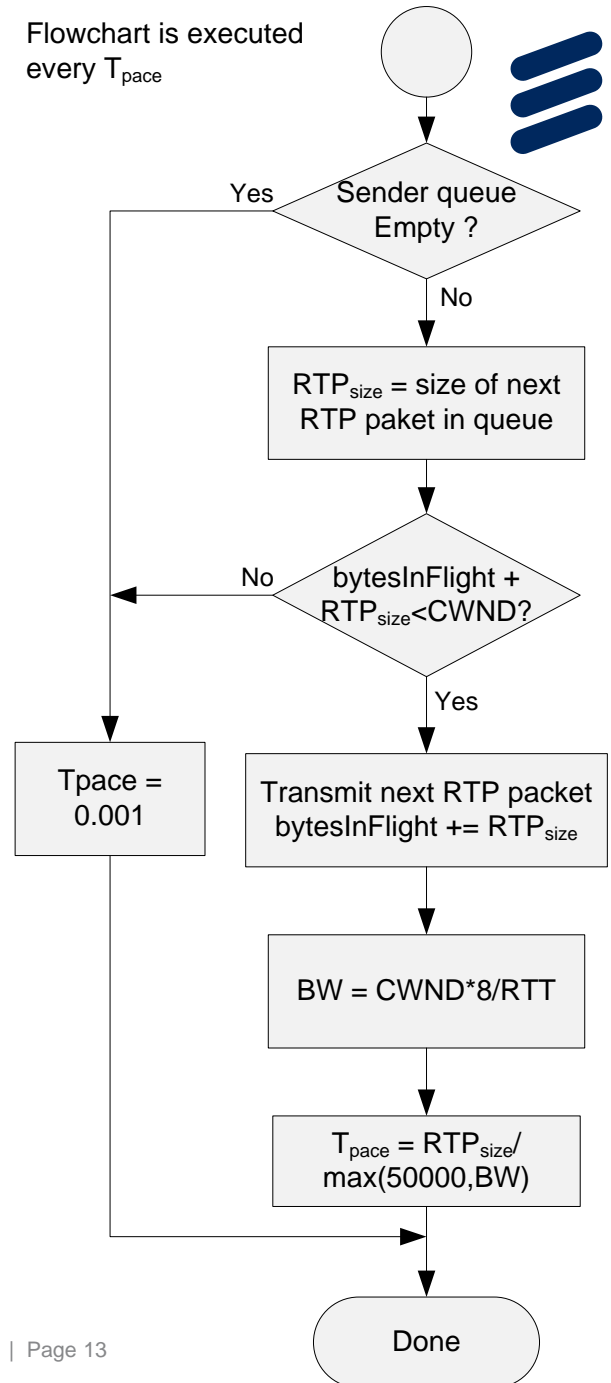
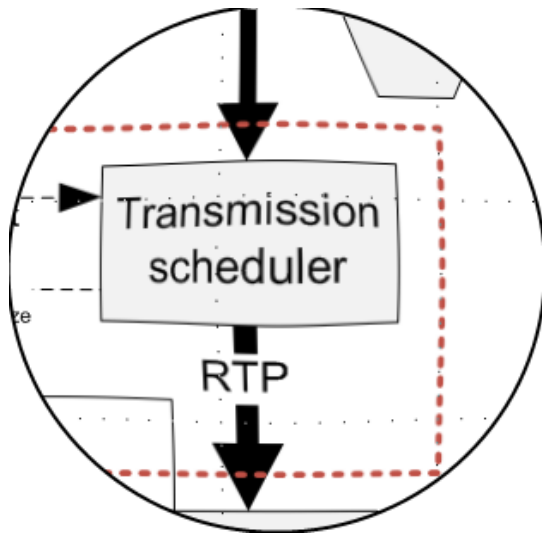
- › Inspired by TCP and LEDBAT
- › Additional features
 - Adaptive delay target
 - Random CWND reduction



TRANSMISSION SCHEDULING

› Additional feature

- Allow transmission even though bytes in flight > CWND in certain cases.

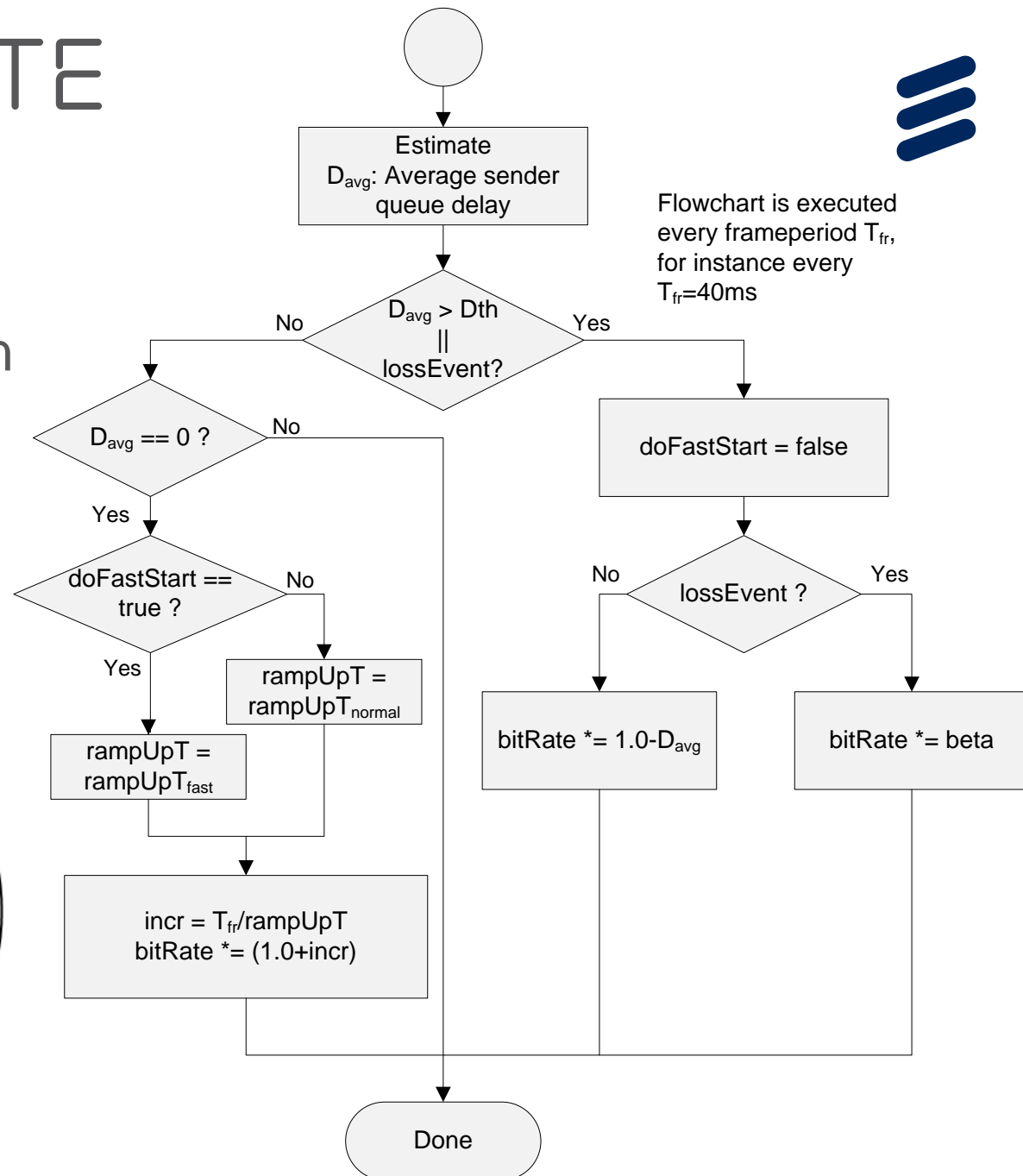
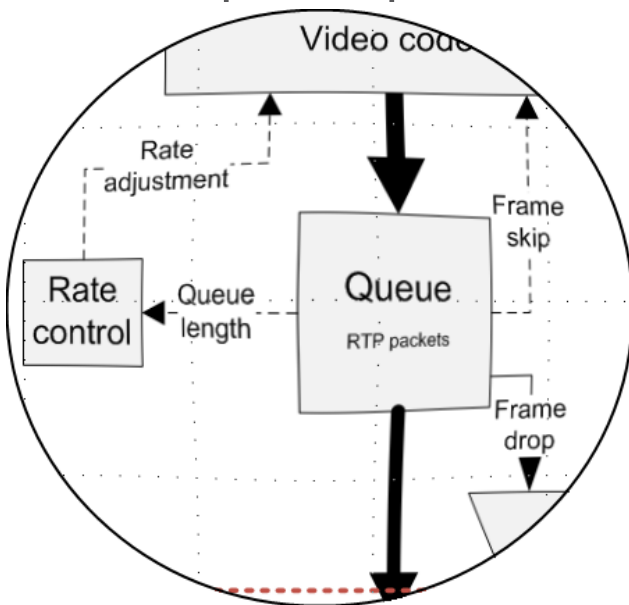


MEDIA RATE CONTROL



› Additional actions, in case sender queue grows too large

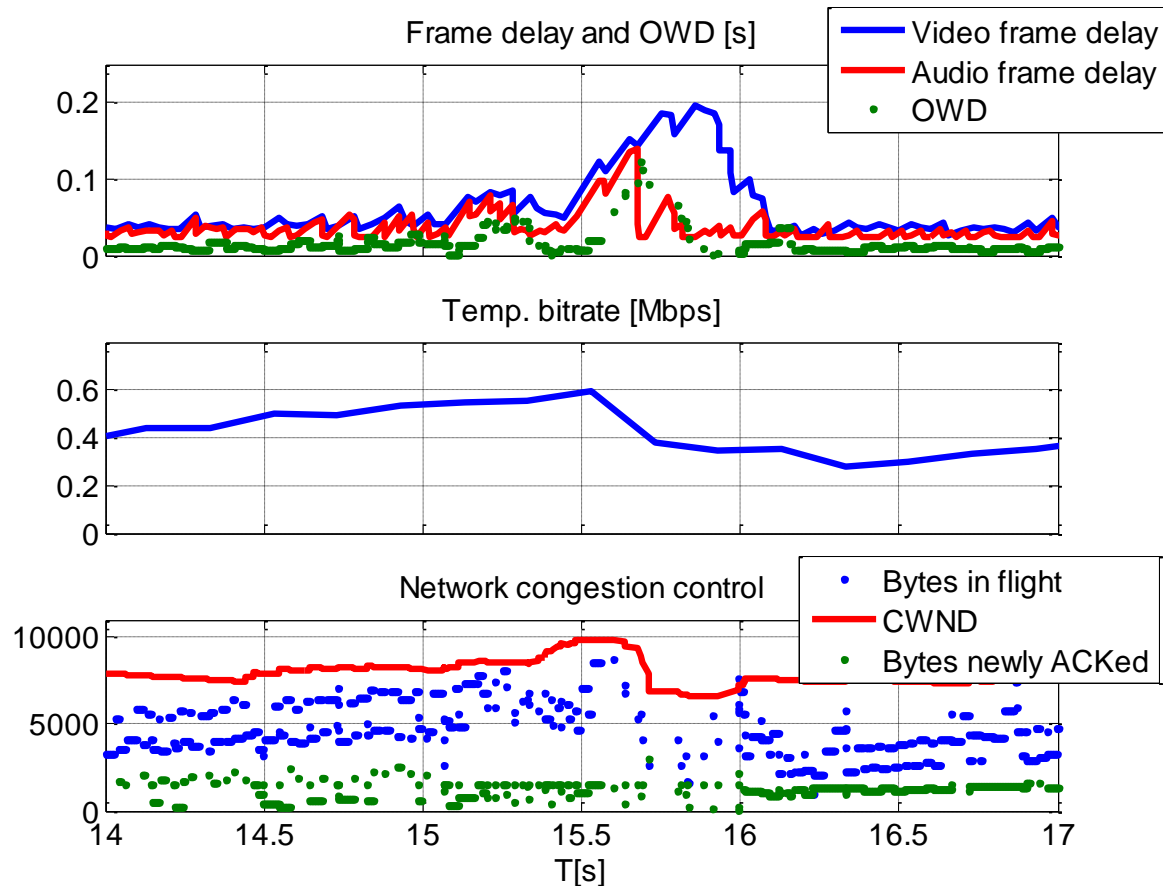
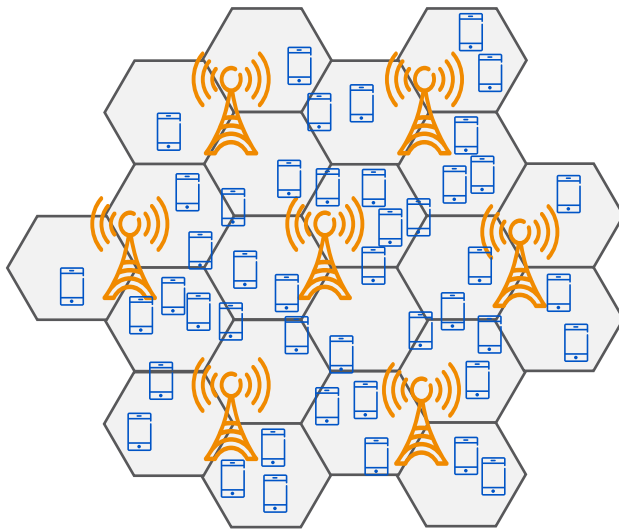
- Skip frames
- Drop RTP packets



EXAMPLE 1 LTE CHANNEL



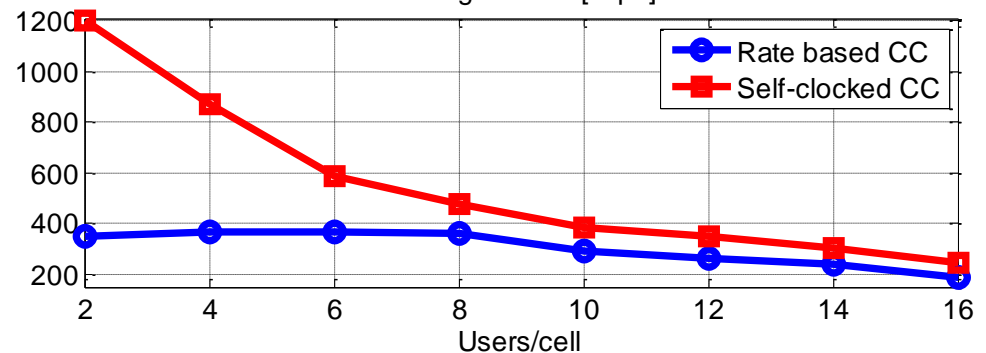
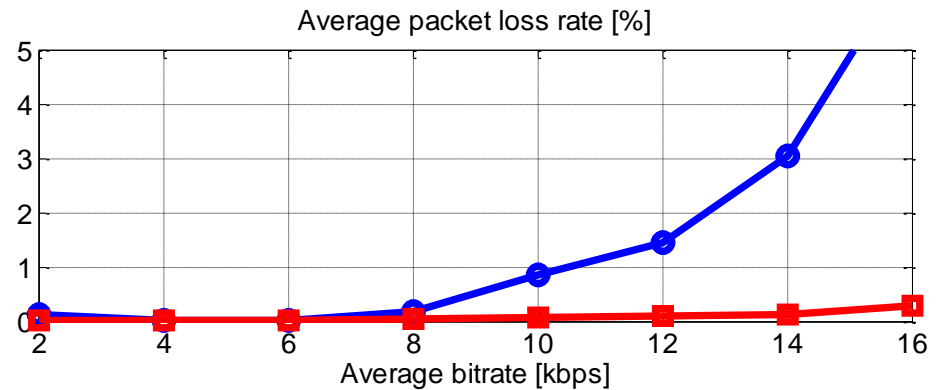
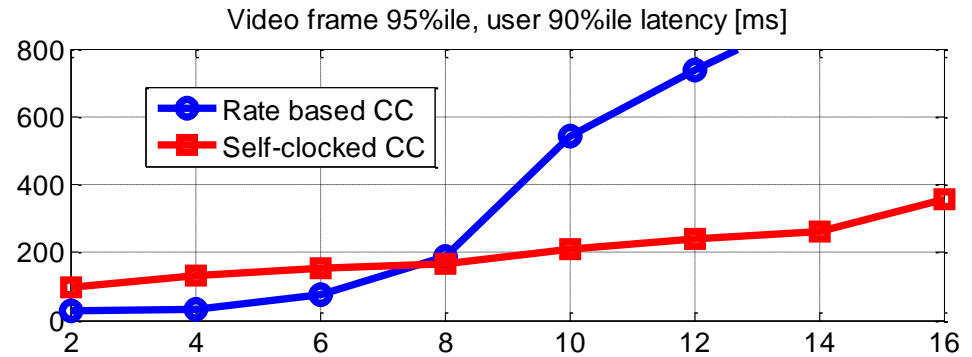
- › Note the more sparse ACKs during the congestion event
~15.5 – 16.0s



COMPARISON SELF-CLOCKED VS RATE BASED CC



- › Rate based CC increases bitrate very slowly → low average bitrate
- › Self-clocked CC increases bitrate quickly → High bitrate at low load
- › Still Self-clocked CC is more stable at higher load levels



DISCUSSION



- › Self-clocked rate adaptation performs considerably better than rate based congestion control algorithms
- › Possible remaining issues to solve
 - Over-reaction at handover
 - Over-reaction to congestion events
 - Negative impact from other cross traffic
 - Possibly frequent and large media bitrate changes
- › Possible solution to the above listed issues
 - ECN
 - Network assisted rate adaptation

CONCLUSION



- › Possible to implement well functioning conversational video over LTE
- › Self-clocking is the key !
- › Leverages on novel TCP features
- › Solution realized with RTCP feedback
 - Solution can be complemented with more network centric support for enhanced performance

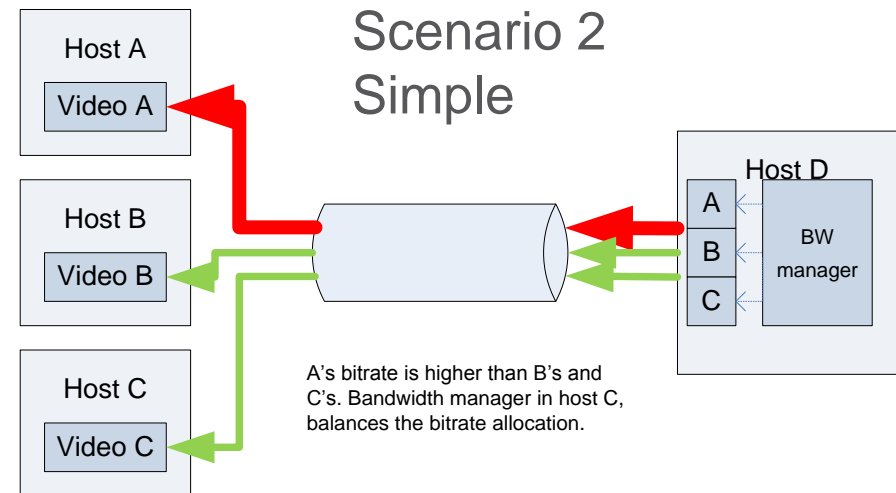
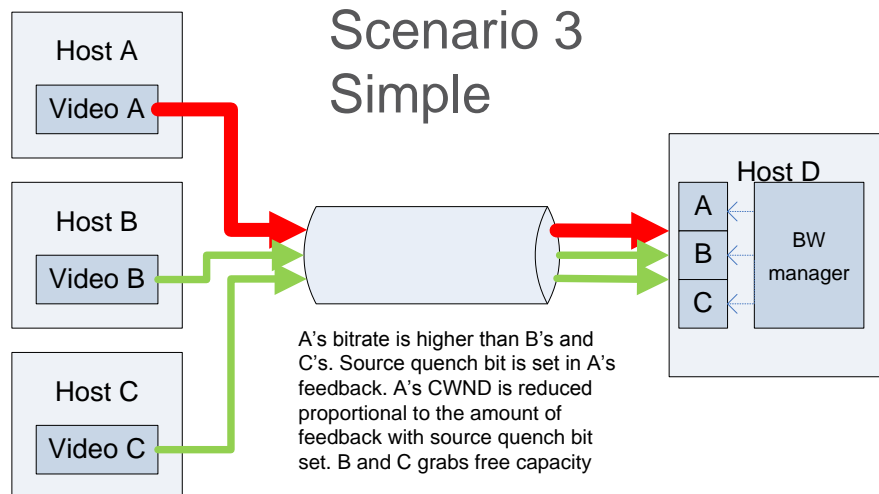
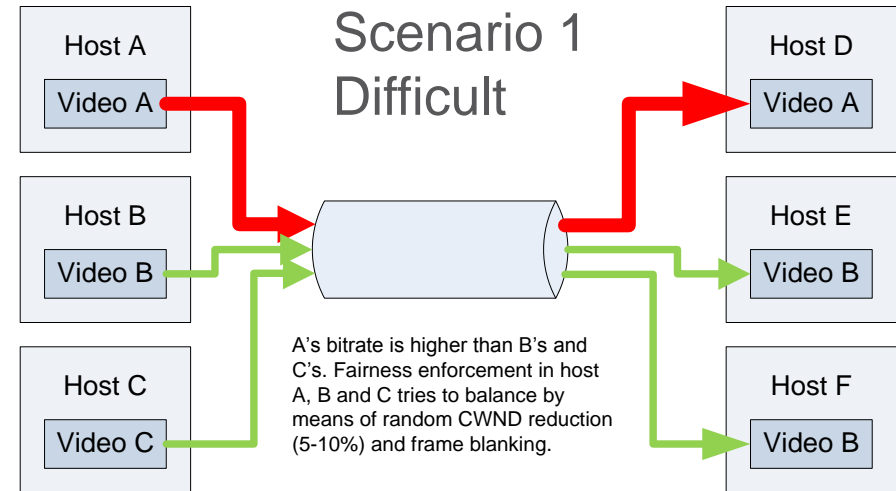


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FAIRNESS ENFORCEMENT

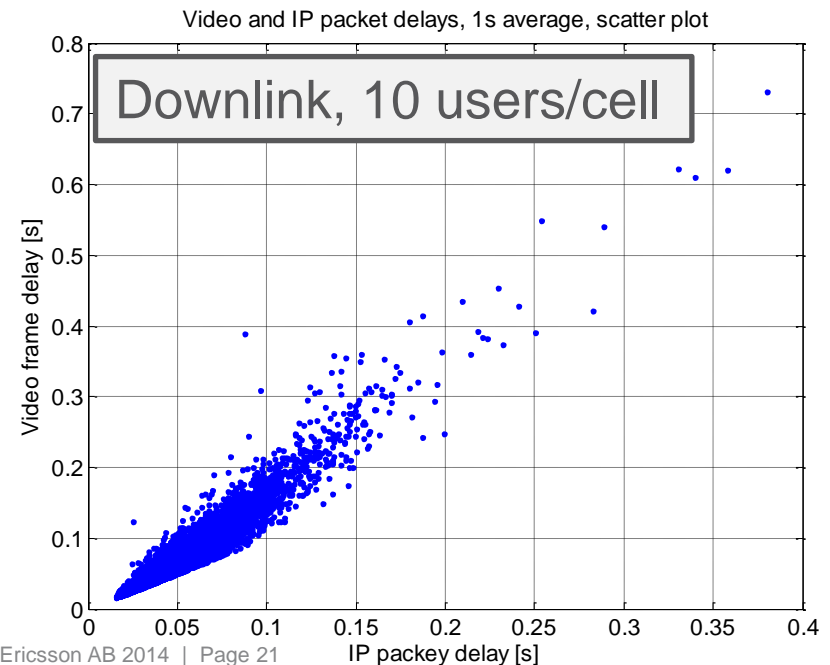
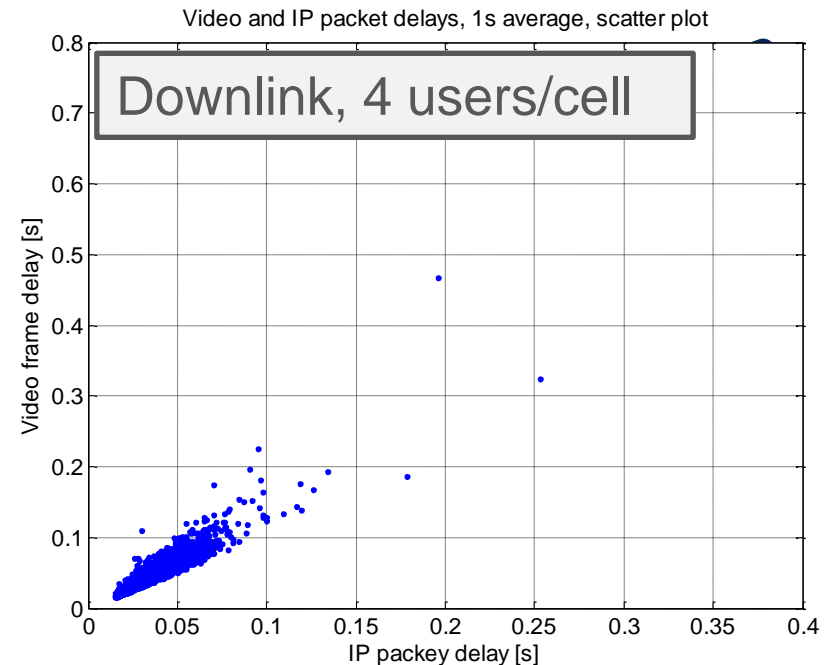


› Three scenarios



ADDITIONAL DELAY IN SENDER QUEUE

- › Scatter plots of IP packet delays vs video frame delay indicate additional queuing in sender queue is strongly correlated to network queuing.
- › Conclusion: Without a sender queue, the delay would likely occur in the network instead

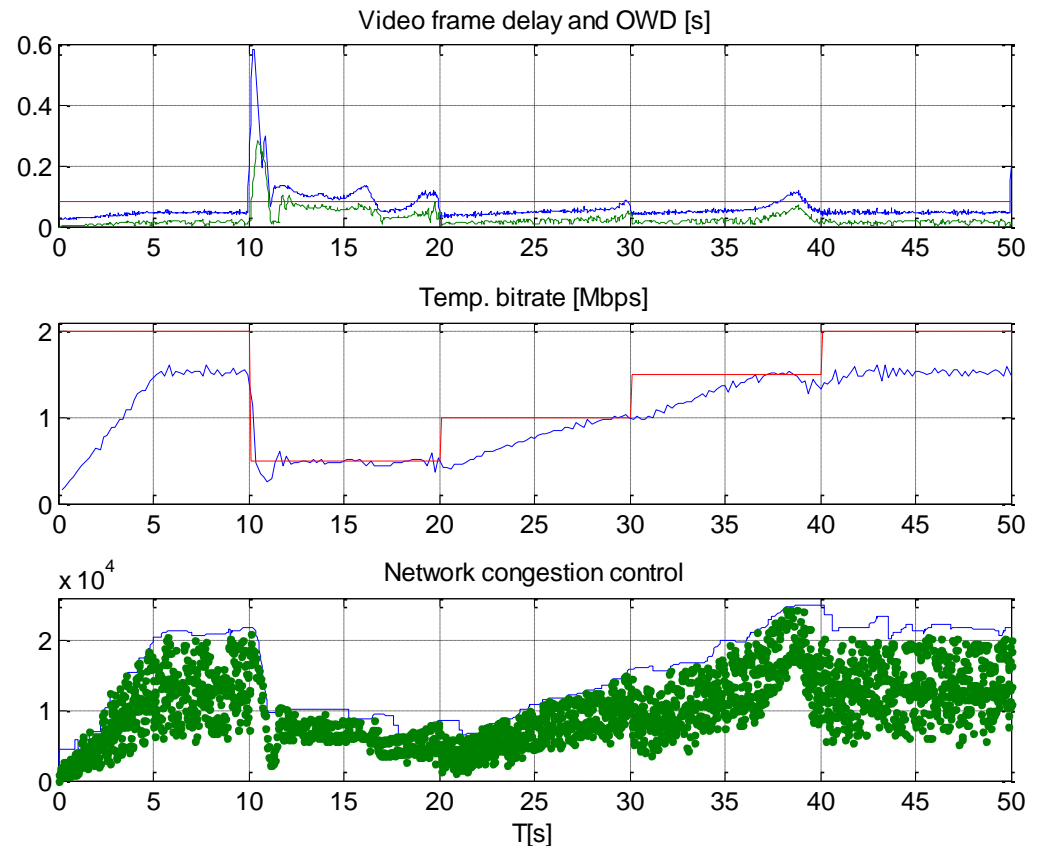


EXAMPLE 2

BOTTLENECK WITH CHANGING BW



- › SCReAM reacts quickly to reduced throughput.

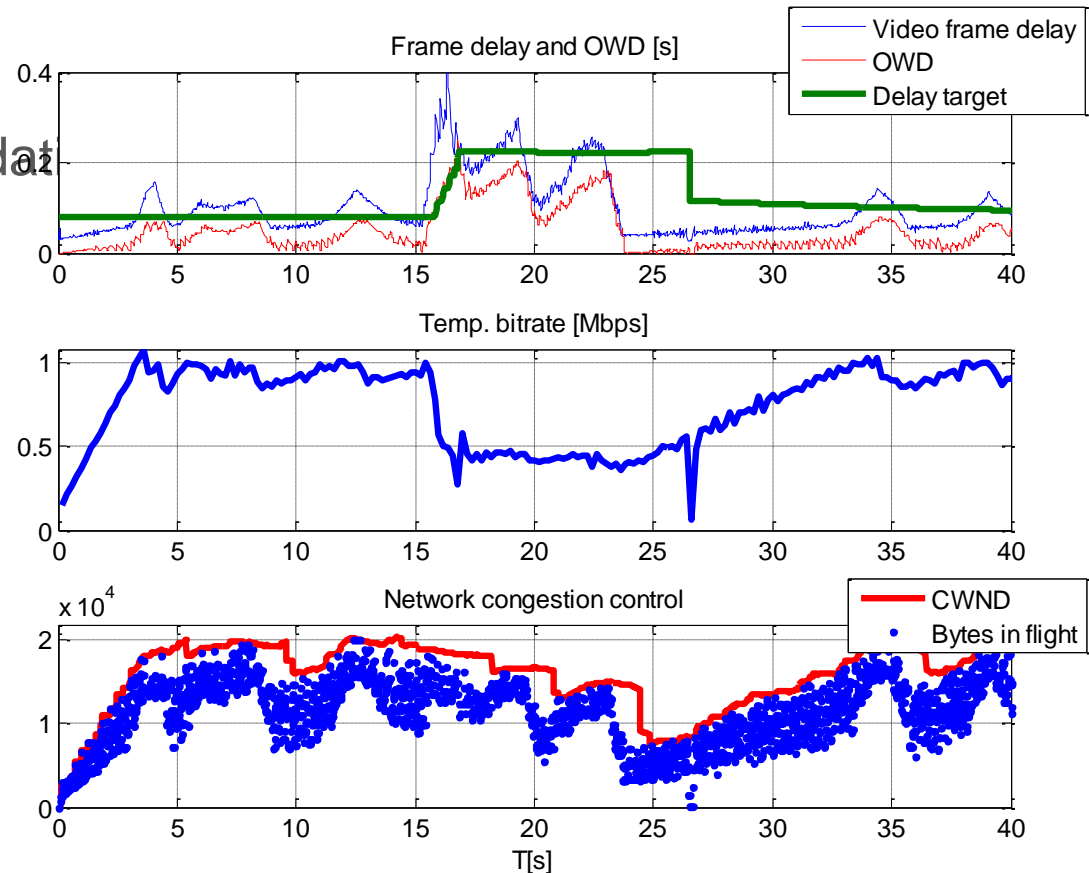


EXAMPLE 3 COMPETING FTP



› Three features :

- Fast start
- Adaptive delay target
- Congestion window validation

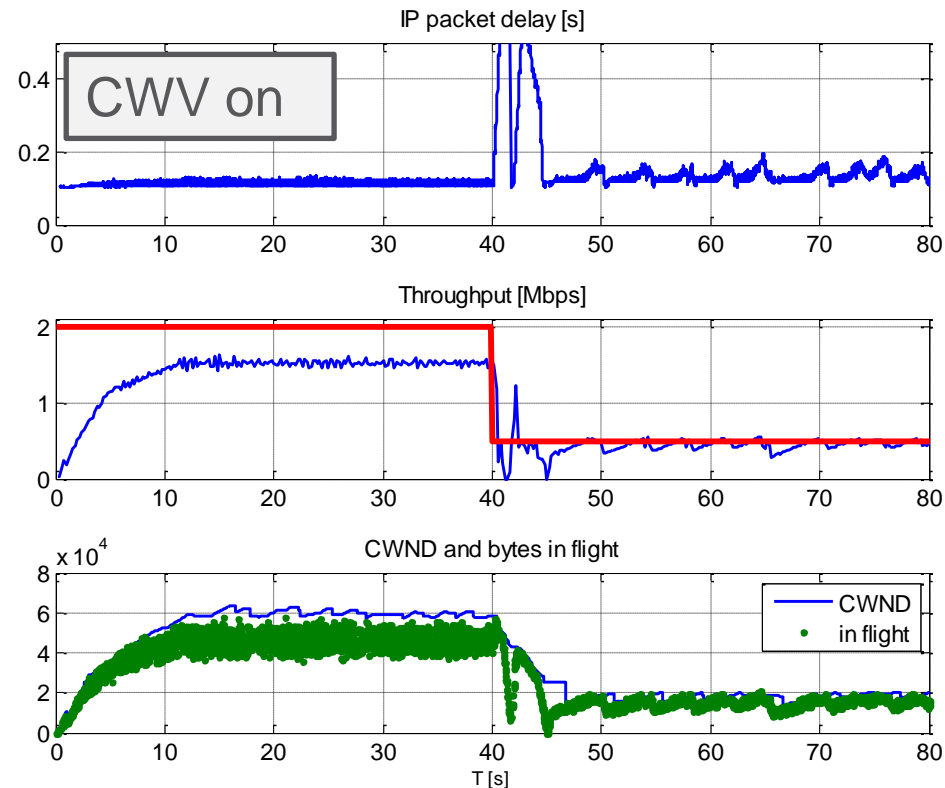
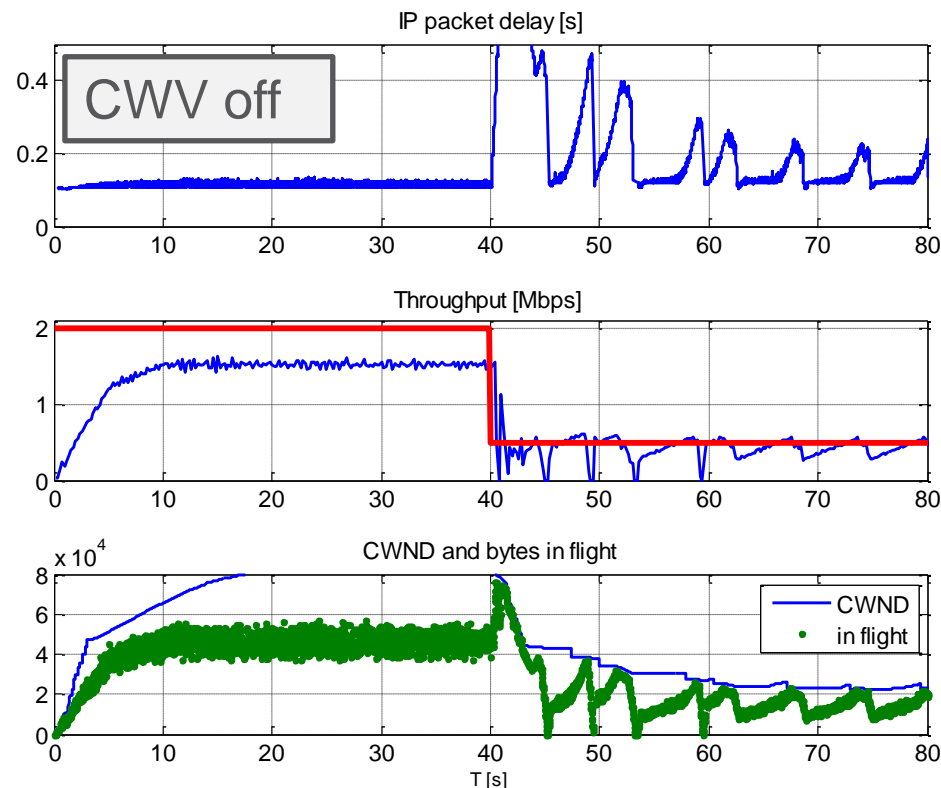


EXAMPLE 4

CONGESTION WINDOW VALIDATION



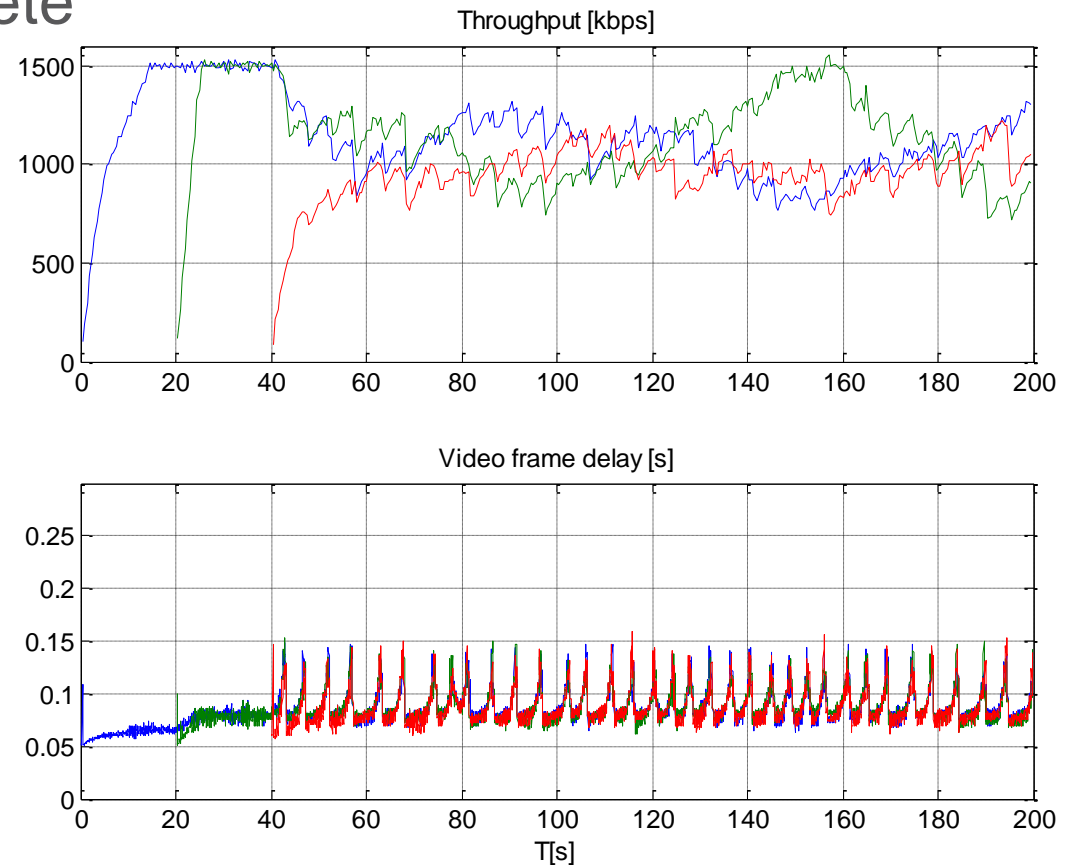
- › Congestion window validation is crucial for the stability of the algorithm



COMPETING RMCAT FLOWS



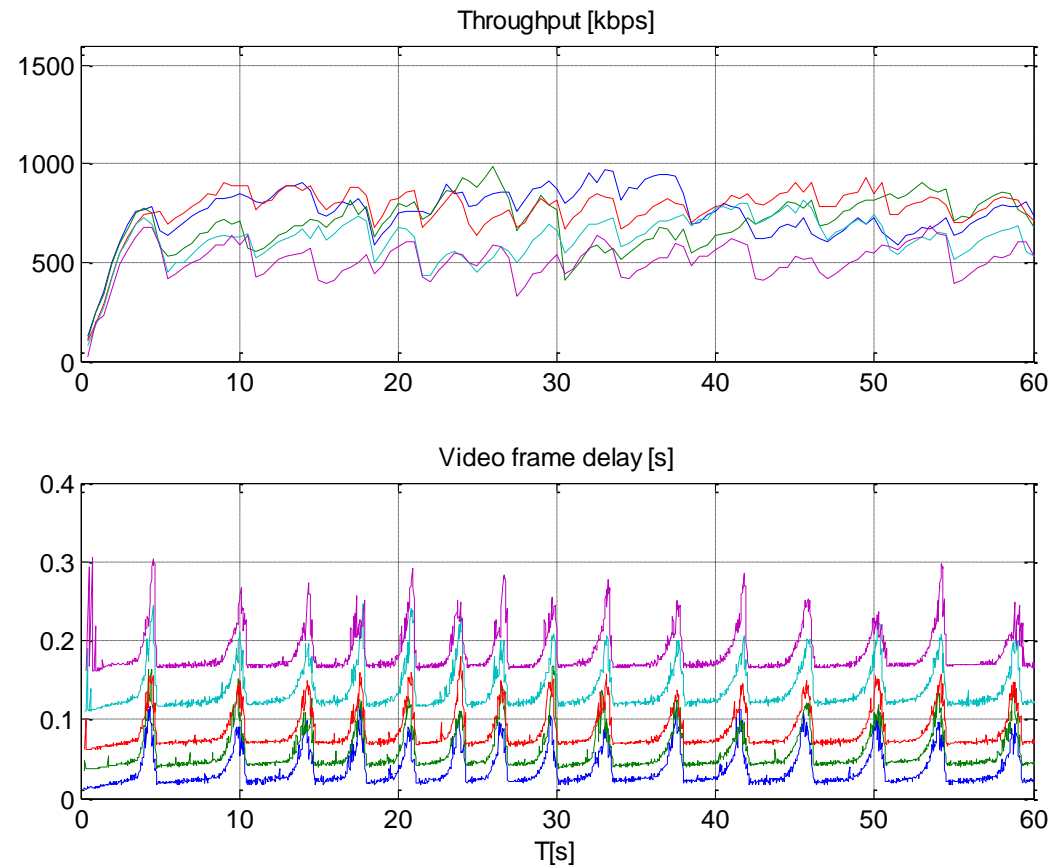
- › RTT = 100ms
- › 3 RMCAT flows compete



RTT FAIRNESS



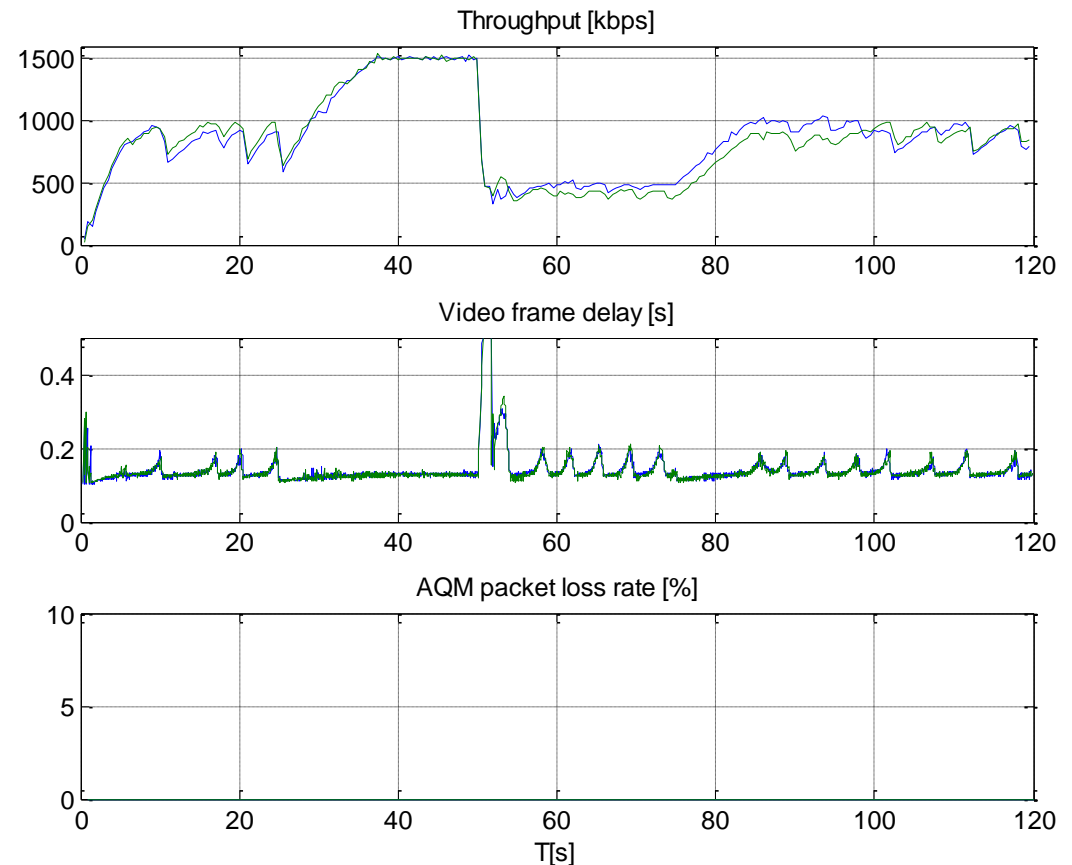
› 5 flows with different RTT



VARIABLE BW



- › Two RMCAT flows compete, changing bandwidth
- › RTT 200ms





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