

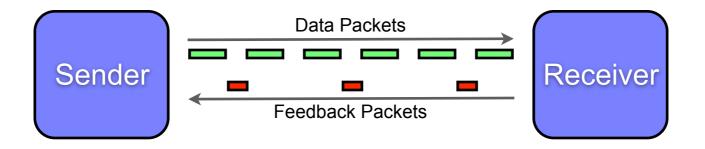
#### Using RTCP Feedback for Unicast Multimedia Congestion Control

draft-ietf-rmcat-rtp-cc-feedback-01

**Colin Perkins** 

## Motivation

• Transport protocol provides a feedback loop



- Dynamics of congestion control depend on rate of feedback, and type of information returned
- RTCP provides a feedback channel for RTP-based applications – what sort of feedback can it provide?

# TL;DR

- Questions to ask regarding congestion feedback:
  - How often is feedback needed?
  - How much overhead is acceptable?
  - How much, and what, data does each report contain?

- How often can feedback be sent in RTCP?
  - Per-packet probably not
  - Per-video frame yes, with reasonable assumptions details follow
  - Per-RTT yes in many cases, provided RTT is not too low
  - Conclusion: if configured correctly, RTCP can support congestion control, provided an appropriate feedback packet is defined

### **RTCP Feedback**

- Four types of feedback can be used:
  - Regular RTP reports [RFC 3550]
  - RTP/AVPF feedback [RFC 4584]
  - Aggregated reports [RFC 3550, draft-ietf-avtcore-rtp-multi-stream-11]
    - Avoid UDP/IP header overhead per report
  - Reporting groups [draft-ietf-avtcore-rtp-multi-stream-optimisation-12]
    - Avoid sending unnecessary reports from co-located SSRCs
- Support in WebRTC:
  - RTCP reporting groups are OPTIONAL in draft-ietf-rtcweb-rtp-usage-26, the others are required (expect aggregated RTCP reports are not widely implemented at the sender)
  - The RTCP XR report defined in draft-dt-rmcat-feedback-message-00 is assumed to be used

#### **Example Scenario**

- Point-to-point video conference
- Two parties, each sending audio and video
- Media bundled onto single 5-tuple  $\rightarrow$  4 SSRCs
  - 1 audio SSRC, 1 video SSRC, for each party

• Can we send a congestion report for every video frame using RTCP?

## Aggregation and Reporting Groups

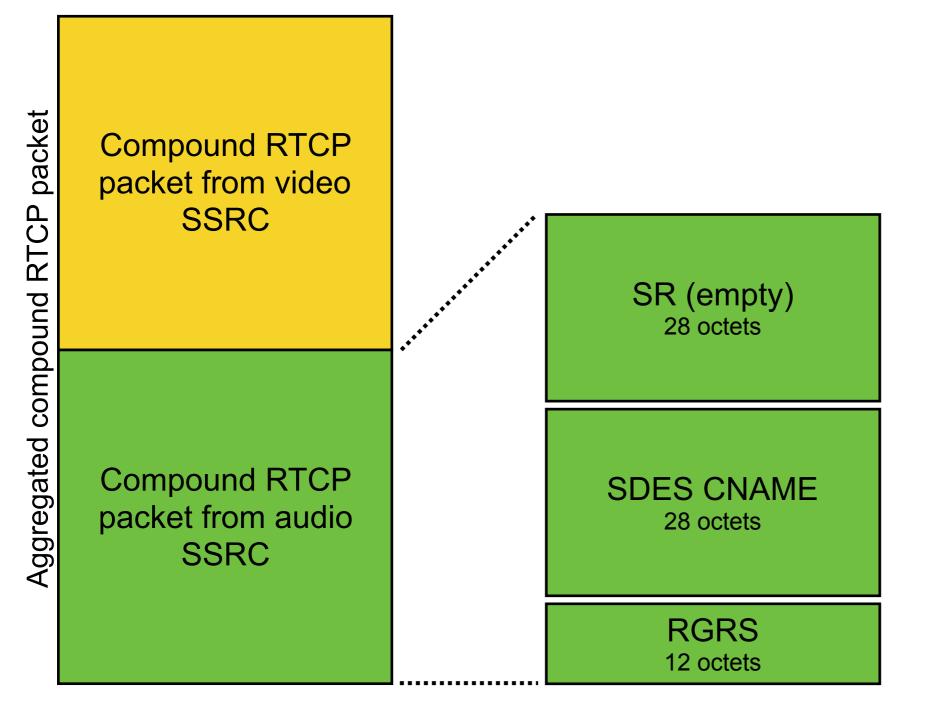


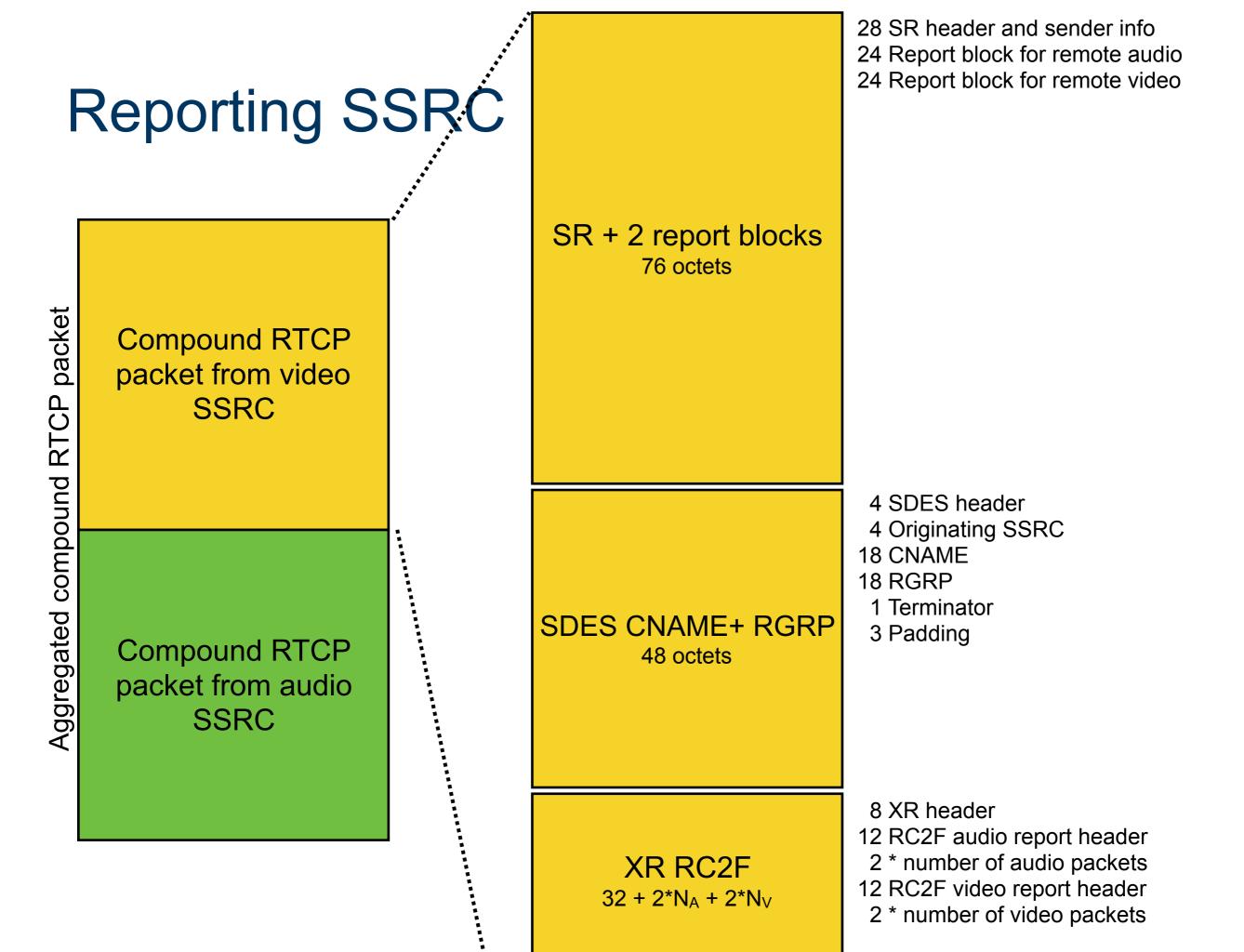
Compound RTCP packet from video SSRC

Compound RTCP packet from audio SSRC

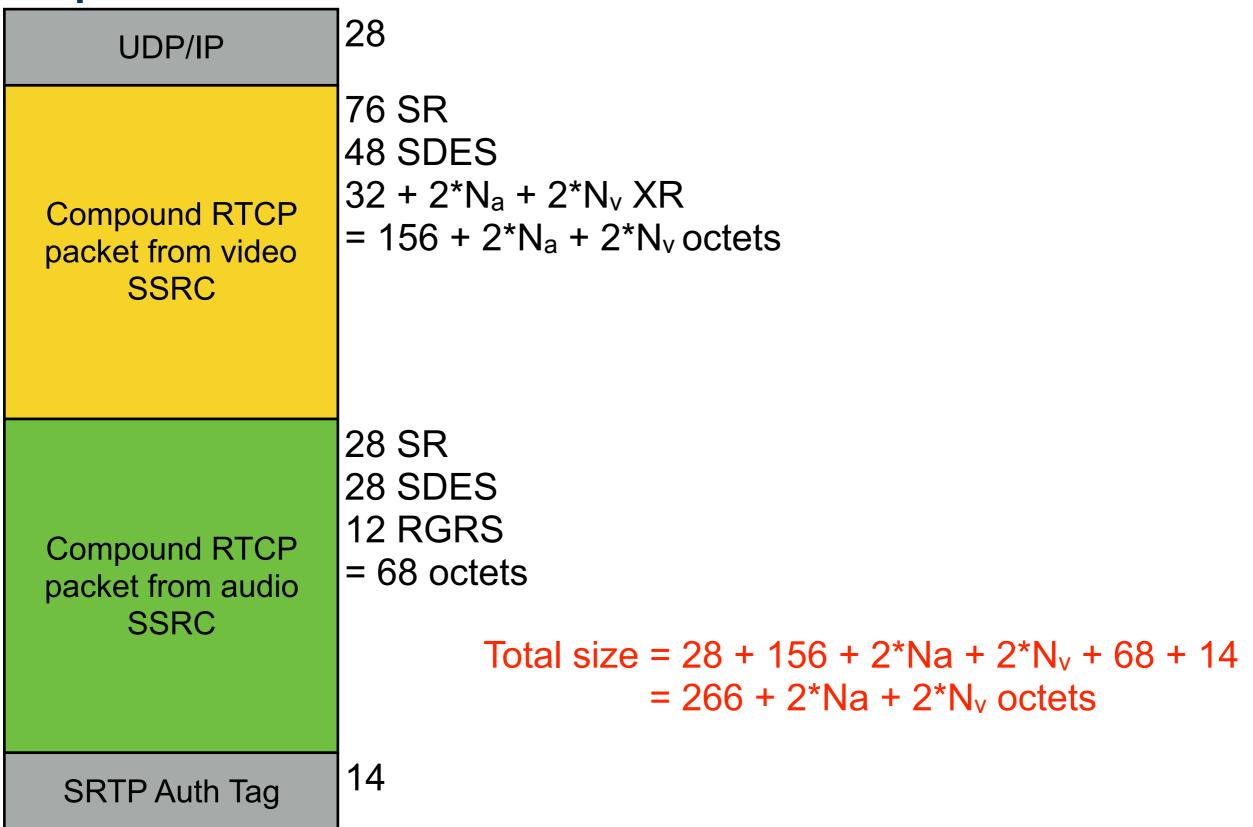
- Aggregate feedback → each RTCP packet is a compound packet, comprising a compound RTCP packet generated by the audio SSRC and a compound RTCP packet generated by the video SSRC
- RTCP reporting groups are used:
  - One SSRC is designated as the reporting SSRC
  - The other SSRC delegates its reports to that SSRC
  - The reports are aggregated, so it doesn't matter which is chosen as reporting SSRC (slides assume video SSRC is reporting SSRC)

## Non-reporting SSRC





#### **Report Size: Overall**



### Report Size: Number of Packets in Report

- What are  $N_a$  and  $N_v$ ?
- Assume:
  - video\_bit\_rate\_bps ≈ session\_bw
  - video\_packets\_per\_second = (video\_bit\_rate\_bps / 8) / mtu
  - audio\_packets\_per\_second = 50
  - N<sub>v</sub> = ceil(video\_packets\_per\_second / fps)
  - N<sub>a</sub> = ceil(audio\_packets\_per\_second / fps)
  - (these assumptions are not realistic)

### **RTCP Reporting Interval**

 Reporting interval for RTP/AVPF with T<sub>rr\_interval</sub> = 0 is rtcp\_interval = avg\_rtcp\_size \* n / rtcp\_bw

- For our scenario:
  - n = 4
  - $avg_packet_size = 266 + 2*Na + 2*N_v$
  - rtcp\_bw = 5% of session\_bw
  - Because of aggregation, avg\_packet\_size is halved [draft-ietf-avtcore-rtp-multi-stream-11]

#### Can we report per frame?

<pre>session_bw = 2.5 × 1024 × 1024/8</pre>
fps = 30
mtu = 15001,500
Na = ceil(50/fps)
Nv = ceil((session_bw/mtu)/fps)8
<pre>avg_rtcp_packet_size = (42 + 68 + 156 + (2 × Nv) + (2 × Na))/2</pre>
n = 4
<pre>rtcp_interval = (avg_rtcp_packet_size × n) / (session_bw × 0.05)0.0349121094</pre>

If session bandwidth > ~2.5Mbps we can report on each frame of 30fps video

### Summary and Next Steps

- With the RTCP congestion feedback format, and standard RTCP features, we can report on every frame of 30fps video if video bandwidth > 2.5Mbps
- Obvious ways to optimise this, without changing the congestion report format
  - RGRP extensions have high overhead

• Analysis is *very* preliminary – ongoing