

# Congestion Feedback in RTCP

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Presentation given to IETF RMCAT working group on 19 July 2017

# Congestion Feedback in RTCP

- What is the overhead of sending congestion feedback in RTCP?
- This will depend on:
  - How often you want to send congestion feedback?
  - What information is included in congestion feedback packets?
  - How that information is formatted?
  - What other information must also be included in those packets?
  - The number and format of the media streams being sent

Regular or reduced-size RTCP?  
Are SR/RR packets included?  
Are SDES packets included? What is the format of the CNAME?  
How are stream associated?  
How is cross-reporting handled?

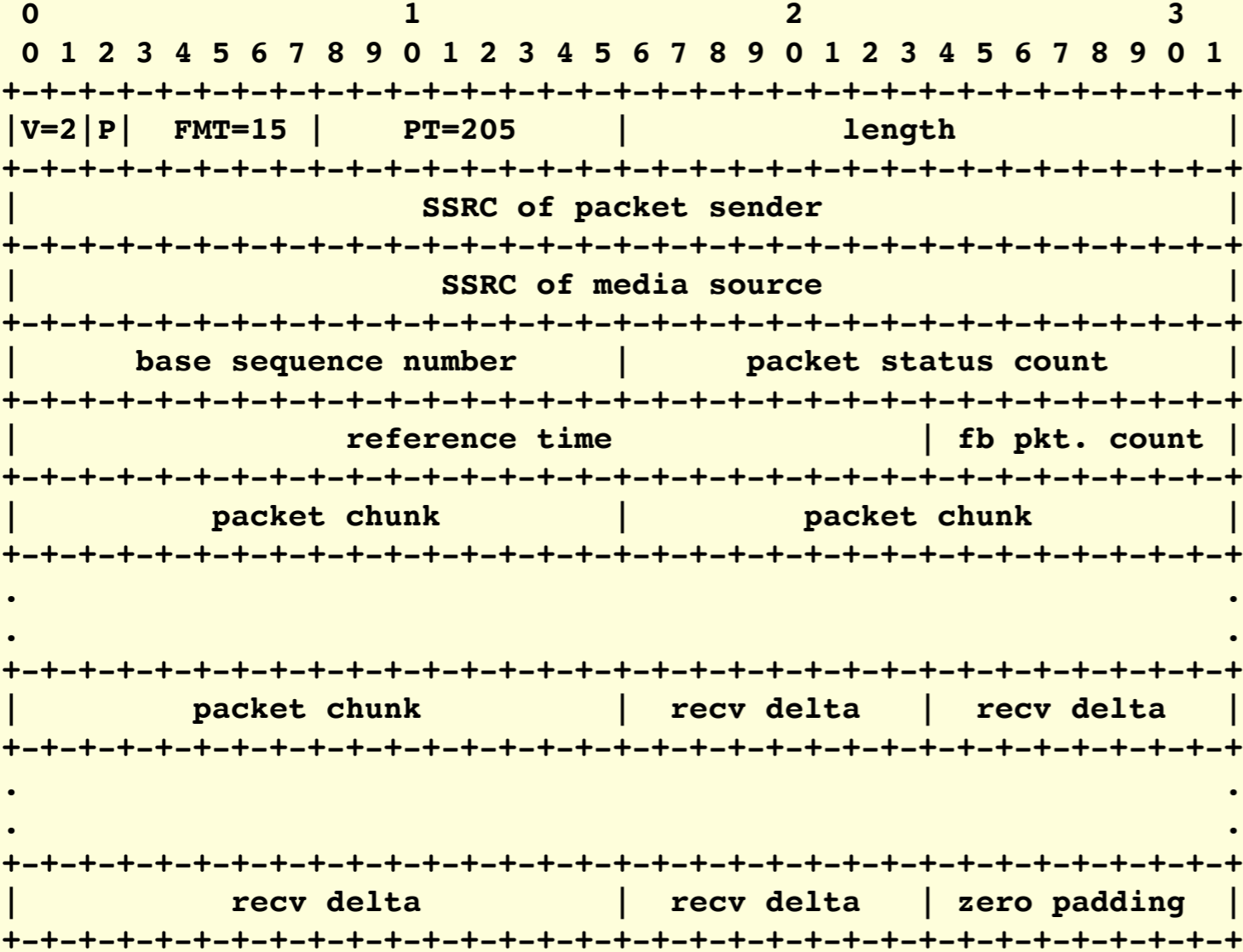
Considering two approaches:

- 1) draft-dt-rmcat-feedback-message-02
- 2) draft-holmer-rmcat-transport-wide-cc-extensions-01

# Scenario 1: VoIP

- Two-party point-to-point VoIP call
- Speech frames sent every  $T_f$  seconds; both participants sending
- Want to send congestion feedback every  $N_r$  frames
- Desire RTCP reporting interval =  $T_f \times N_r$  seconds
- Congestion feedback can be sent in regular compound RTCP packets or reduced-size packets sent using RTP/AVPF early feedback
  - Send  $N_{nc}$  non-compound packets between every compound packet

# draft-holmer-rmcat-transport-wide-cc-extensions-01

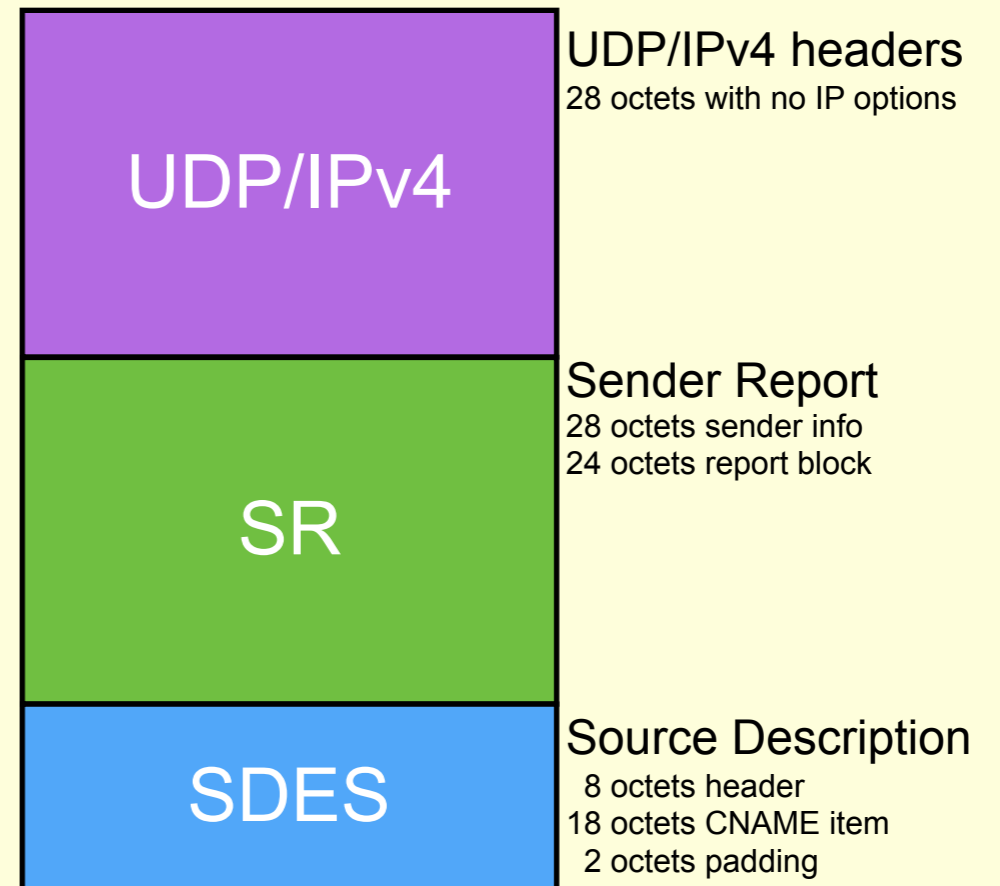


- Single packet format used, whether sent in compound or reduced size packets
- Packet chunks are either a bit vector or RLE encoded
- Recv delta fields are one or two octets, depending on packet inter-arrival time
- RTCP packet size will vary with loss pattern and inter-arrival times

Requires RTP header extension for transport-wide sequence number if more than one media stream is sent

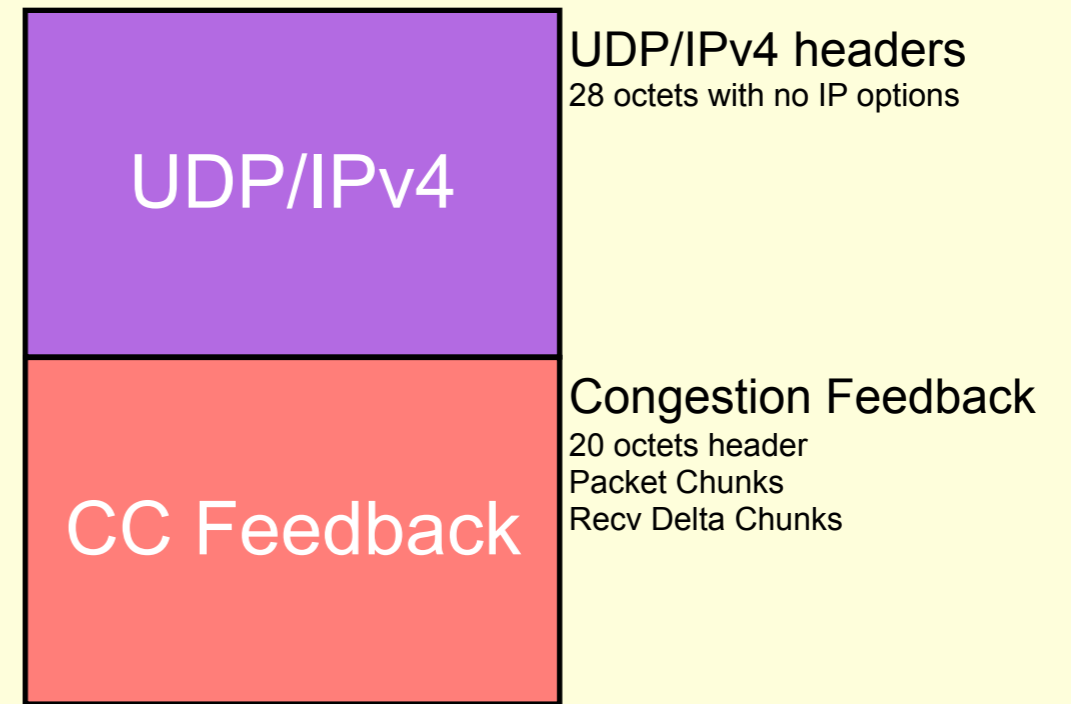
# Scenario 1: VoIP – compound RTCP packets

- Compound RTCP packets contain:
  - Sender Report (SR)
  - Source Description (SDES) with CNAME item
- Packet size,  $S_c = 108$  octets
  - UDP/IP + SR + SDES = 108 octets

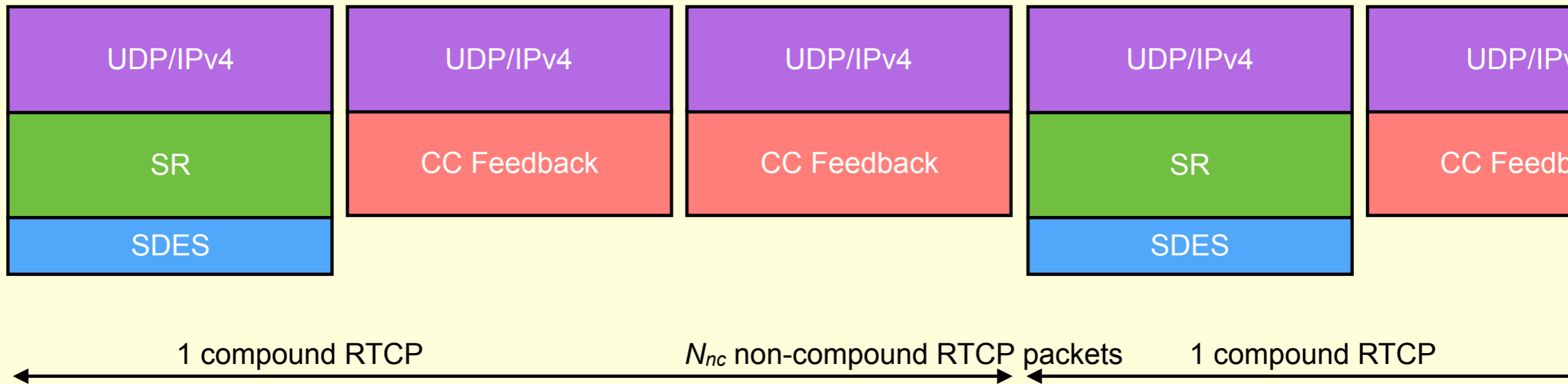


# Scenario 1: VoIP – non-compound RTCP packets

- Non-compound RTCP packets contain:
  - Congestion control feedback (draft-holmer-rmcat-transport-wide-cc-extensions-01)
- Packet size,  $S_{nc} = 48 + 2 + N_r$  octets
  - UDP/IP + RTCP congestion feedback header = 48 octets
  - Packet chunks: best case → no packets lost, single RLE packet chunk sufficient → 2 octets
  - Recv Delta chunks: best case → deltas fit one octet format →  $N_r$  octets (one recv delta per packet sent)



# Scenario 1: VoIP – average RTCP size



- Average RTCP packet size,  $S_{rtcp} = (S_c + N_{nc} \times S_{nc}) / (1 + N_{nc})$   
where  $N_{nc} = 0$  if non-compound packets are not sent

# Scenario 1: VoIP – RTCP bandwidth

- From RFC 3550: RTCP reporting interval,  $T_{rtcp} = n \times S_{rtcp}/B_{rtcp}$  where:
  - $n$  is the number of participants ( $n = 2$  in this scenario)
  - $S_{rtcp} = (S_c + N_{nc} \times S_{nc}) / (1 + N_{nc})$  is the average RTCP packet size in octets
  - $B_{rtcp}$  is the bandwidth allocated to RTCP in octets per second
- To report every  $N_r$  frames, we want  $T_{rtcp} = N_r \times T_f$ 
  - $\Rightarrow N_r \times T_f = n \times S_{rtcp}/B_{rtcp}$
  - $\Rightarrow B_{rtcp} = (n \times (S_c + N_{nc} \times S_{nc})) / (N_r \times T_f \times (1 + N_{nc}))$



# Scenario 1: VoIP – RTCP bandwidth requirements (1)

$T_f$ (seconds)	$N_r$ (frames)	$B_{rtcp}$ (kbps)
20ms	2	42.2
20ms	4	21.1
20ms	8	10.6
20ms	16	5.3
60ms	2	14.1
60ms	4	7.0
60ms	8	3.5
60ms	16	1.8

Sending only compound RTCP packets

Note: this is best case; losses increase feedback packet size → increased  $B_{rtcp}$

- Chart gives the required RTCP bandwidth,  $B_{rtcp}$ , to send a report after every  $N_r$  frames with frames being sent every  $T_f$  seconds
  - Total RTCP bandwidth for the session: each participant gets half of this
  - Compound packets only:  $N_{nc} = 0$
- Sending an RTCP report every 2nd frame with 20ms frames → 52kbps RTCP bandwidth
- Sending an RTCP report every 16th frame with 60ms frames → 2.4kbps RTCP bandwidth
  - This is 1 RTCP packet per second from each SSRC in the VoIP call

# Scenario 1: VoIP – RTCP bandwidth requirements (2)

$T_f$ (seconds)	$N_r$ (frames)	$B_{rtcp}$ (kbps)
20ms	2	31.3
20ms	4	15.8
20ms	8	8.1
20ms	16	4.2
60ms	2	10.4
60ms	4	5.3
60ms	8	2.7
60ms	16	1.4

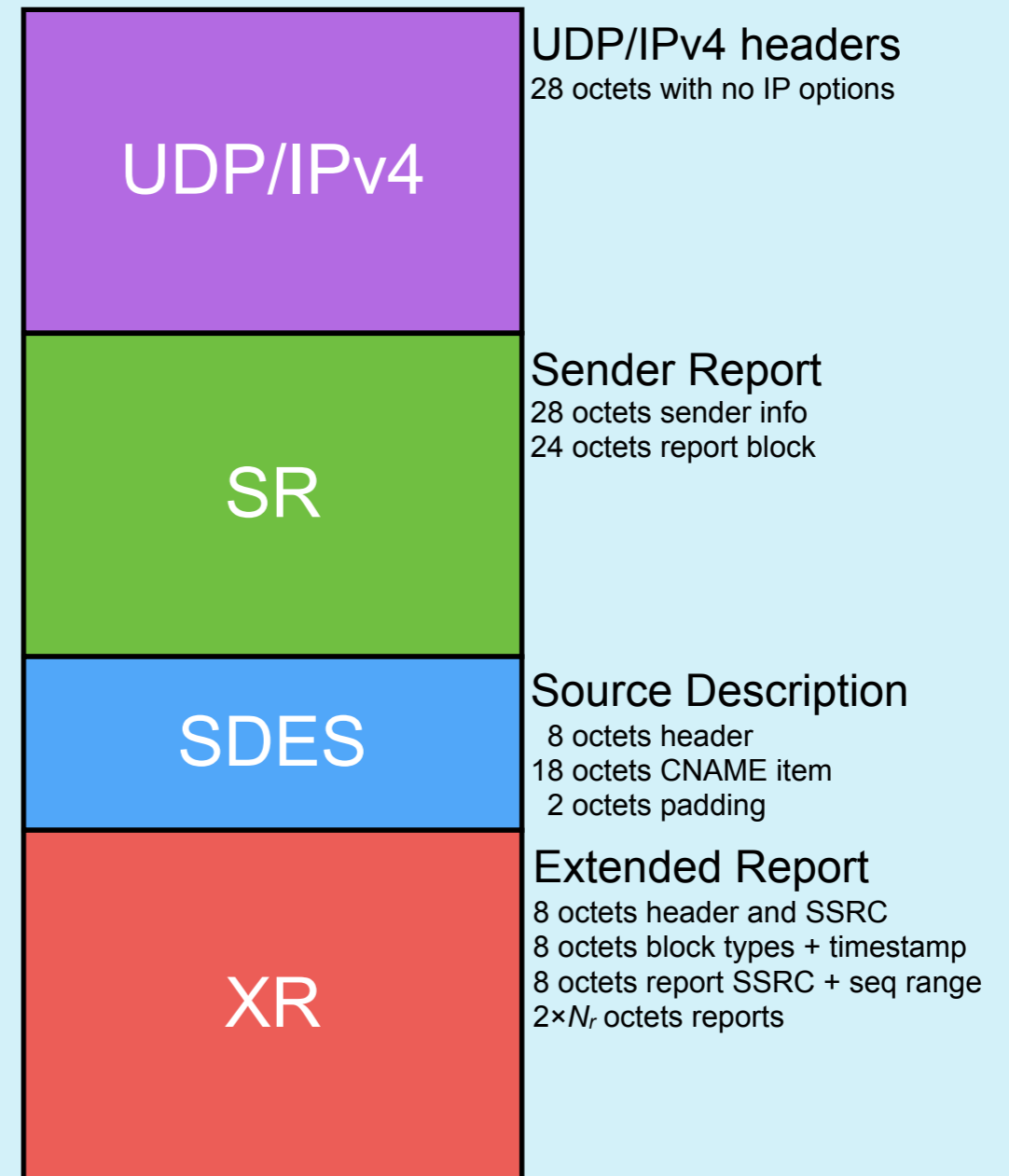
Alternating compound and non-compound RTCP

- Required RTCP bandwidth is reduced if a non-compound packet is sent between compound packets
- Reduced header overheads – due to not sending SR/RR and SDES packets in some reports

Note: this is best case; losses increase feedback packet size → increased  $B_{rtcp}$

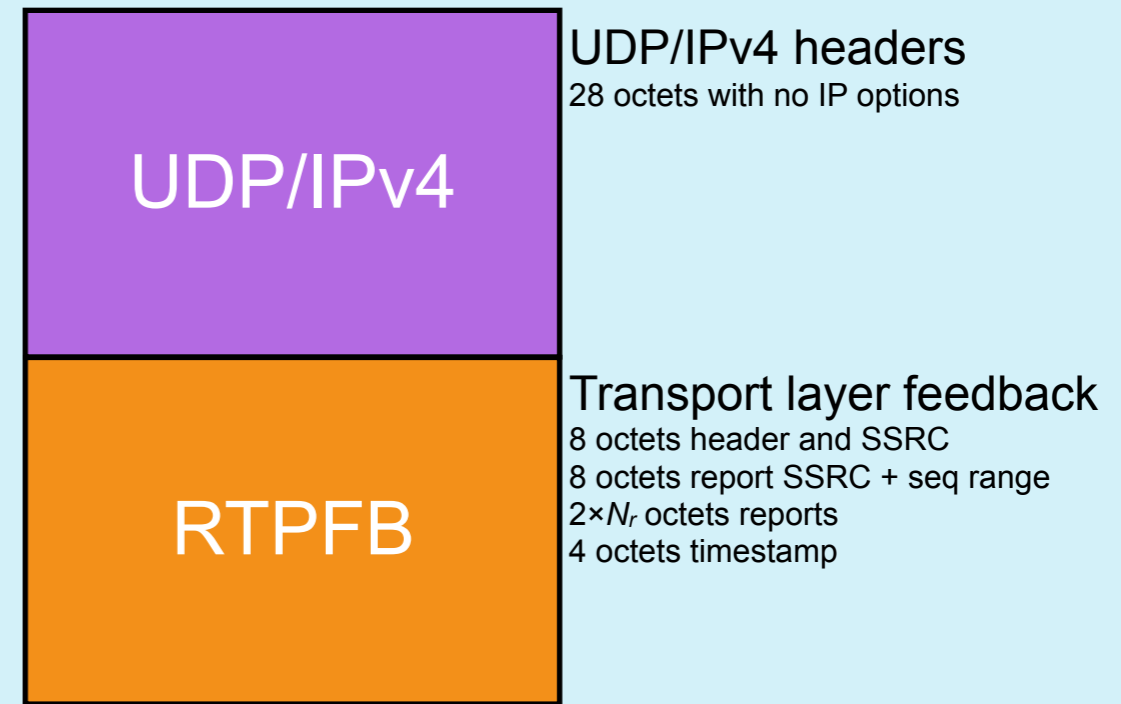
# Scenario 1: VoIP – compound RTCP packets

- Using [draft-dt-rmcat-feedback-message-02](#)
- Compound RTCP packets contain:
  - Sender Report (SR)
  - Source Description (SDES) with CNAME item
  - Extended Report (XR) with congestion control feedback ([draft-dt-rmcat-feedback-message-01](#))
- Packet size,  $S_c = 132 + 2 \times N_r$  octets



# Scenario 1: VoIP – non-compound RTCP packets

- Non-compound RTCP packets contain:
  - RTP/AVPF transport layer feedback packet (draft-dt-rmcat-feedback-message-01)
- Packet size,  $S_{nc} = 48 + 2 \times N_r$  octets



# Scenario 1: VoIP – RTCP bandwidth requirements (1)

$T_f$ (seconds)	$N_r$ (frames)	$B_{rtcp}$ (kbps)
20ms	2	53.1
20ms	4	27.3
20ms	8	14.5
20ms	16	8.0
60ms	2	17.7
60ms	4	9.1
60ms	8	4.8
60ms	16	2.7

Sending only compound RTCP packets

Note:  $B_{rtcp}$  independent of loss rate

- Chart gives the required RTCP bandwidth,  $B_{rtcp}$ , to send a report after every  $N_r$  frames with frames being sent every  $T_f$  seconds
  - Total RTCP bandwidth for the session: each participant gets half of this
  - Compound packets only:  $N_{nc} = 0$
- Sending an RTCP report every 2nd frame with 20ms frames → 53kbps RTCP bandwidth
- Sending an RTCP report every 16th frame with 60ms frames → 2.7kbps RTCP bandwidth
  - This is 1 RTCP packet per second from each SSRC in the VoIP call

# Scenario 1: VoIP – RTCP bandwidth requirements (2)

$T_f$ (seconds)	$N_r$ (frames)	$B_{rtcp}$ (kbps)
20ms	2	36.7
20ms	4	19.1
20ms	8	10.4
20ms	16	6.0
60ms	2	12.2
60ms	4	6.4
60ms	8	3.5
60ms	16	2.0

Alternating compound and non-compound RTCP

- Required RTCP bandwidth is reduced if a non-compound packet is sent between compound packets
- Reduced header overheads – due to not sending SR/RR and SDES packets in some reports

Note:  $B_{rtcp}$  independent of loss rate

# Scenario 1: VoIP – Comparison

$T_f$ (seconds)	$N_r$ (frames)	$B_{rtcp}$ (kbps)	$B_{rtcp}$ (kbps)
20ms	2	31.3	36.7
20ms	4	15.8	19.1
20ms	8	8.1	10.4
20ms	16	4.2	6.0
60ms	2	10.4	12.2
60ms	4	5.3	6.4
60ms	8	2.7	3.5
60ms	16	1.4	2.0

In best case, draft-holmer-rmcat-transport-wide-cc-extensions-01 more efficient compared to draft-dt-rmcat-feedback-message-02 – but doesn't report on ECN (saving primarily due to not including congestion reports in compound packets – still more efficient if congestion reports *are* sent in compound packets, but only just)

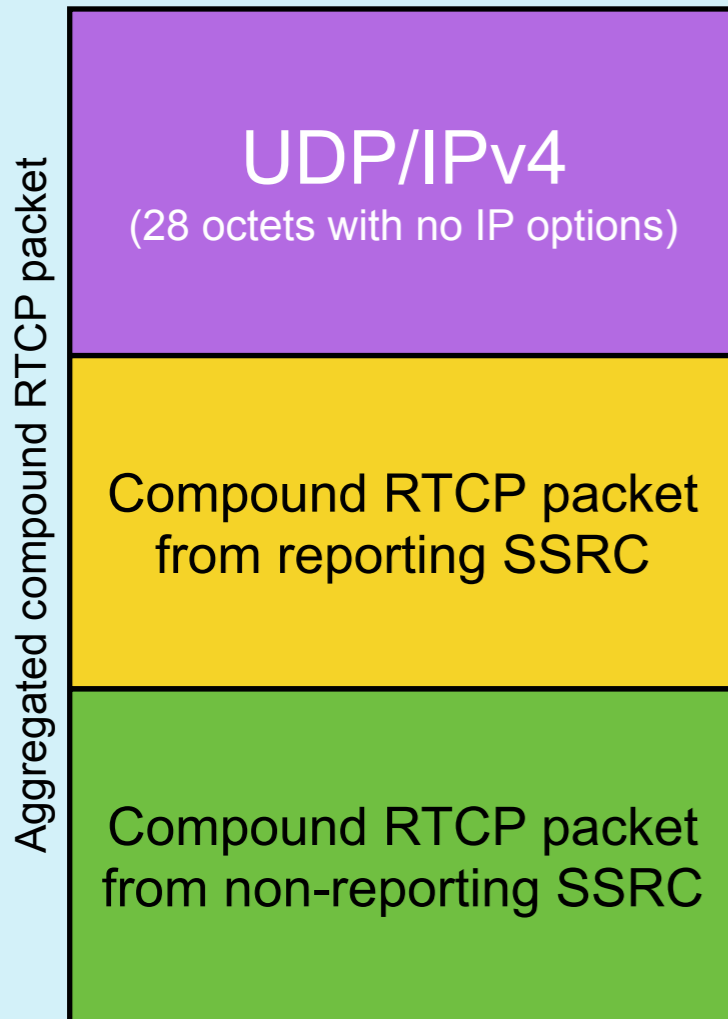
With more complex loss patterns, still saves compared to draft-dt-rmcat-feedback-message-02, but benefit is less

## Scenario 2: Video conference

- 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
  
- Video frame interval =  $T_f$  (i.e., frame rate =  $1/T_f$  frames per second)
- Desire RTCP reporting interval =  $N_r \times T_f$ 
  - If  $N_r = 1$ , report every frame
  - If  $N_r = 2$ , report every other frame
  - ...
- Packets can be sent as compound or reduced size (non-compound) RTCP packets

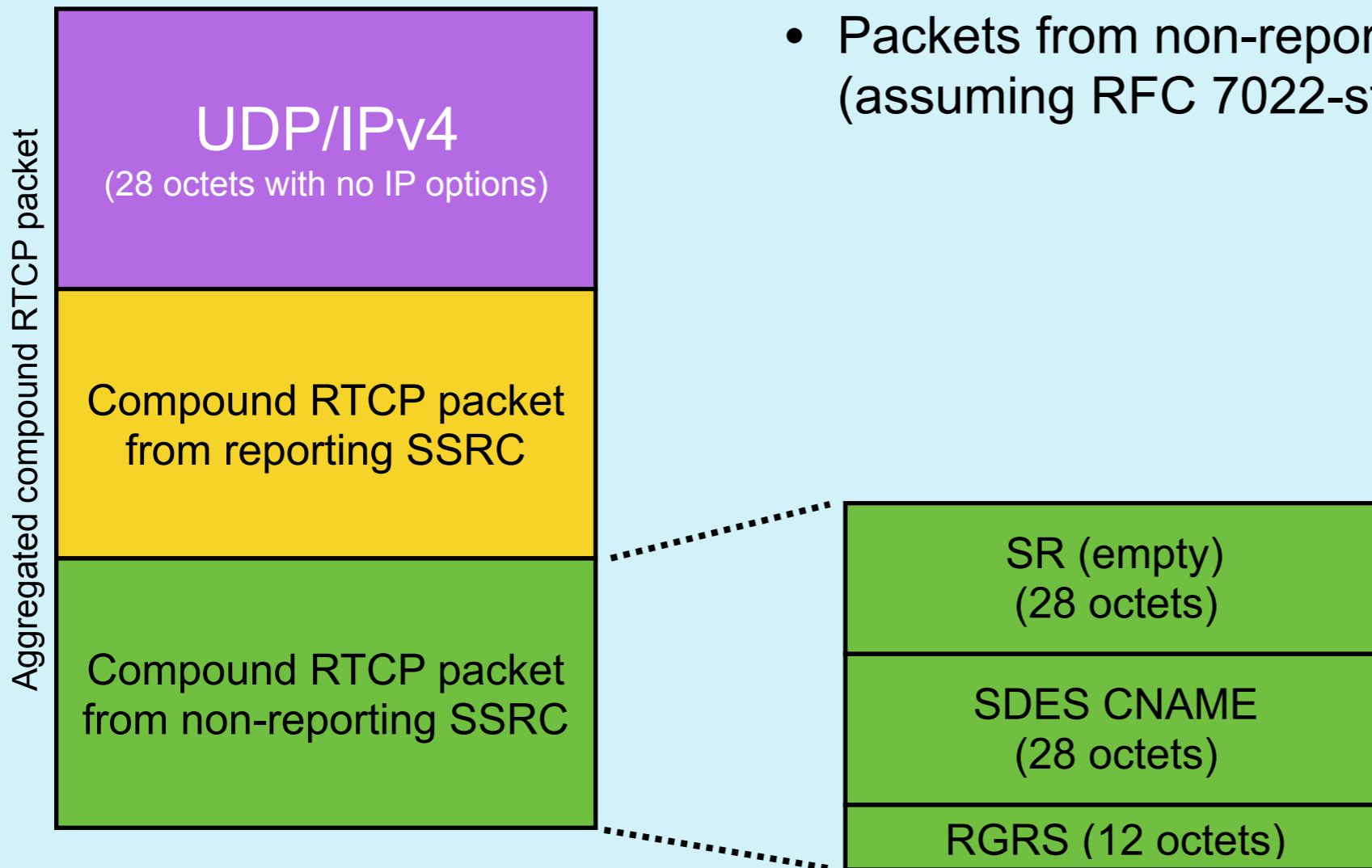


# Scenario 2: Video conference – compound packets



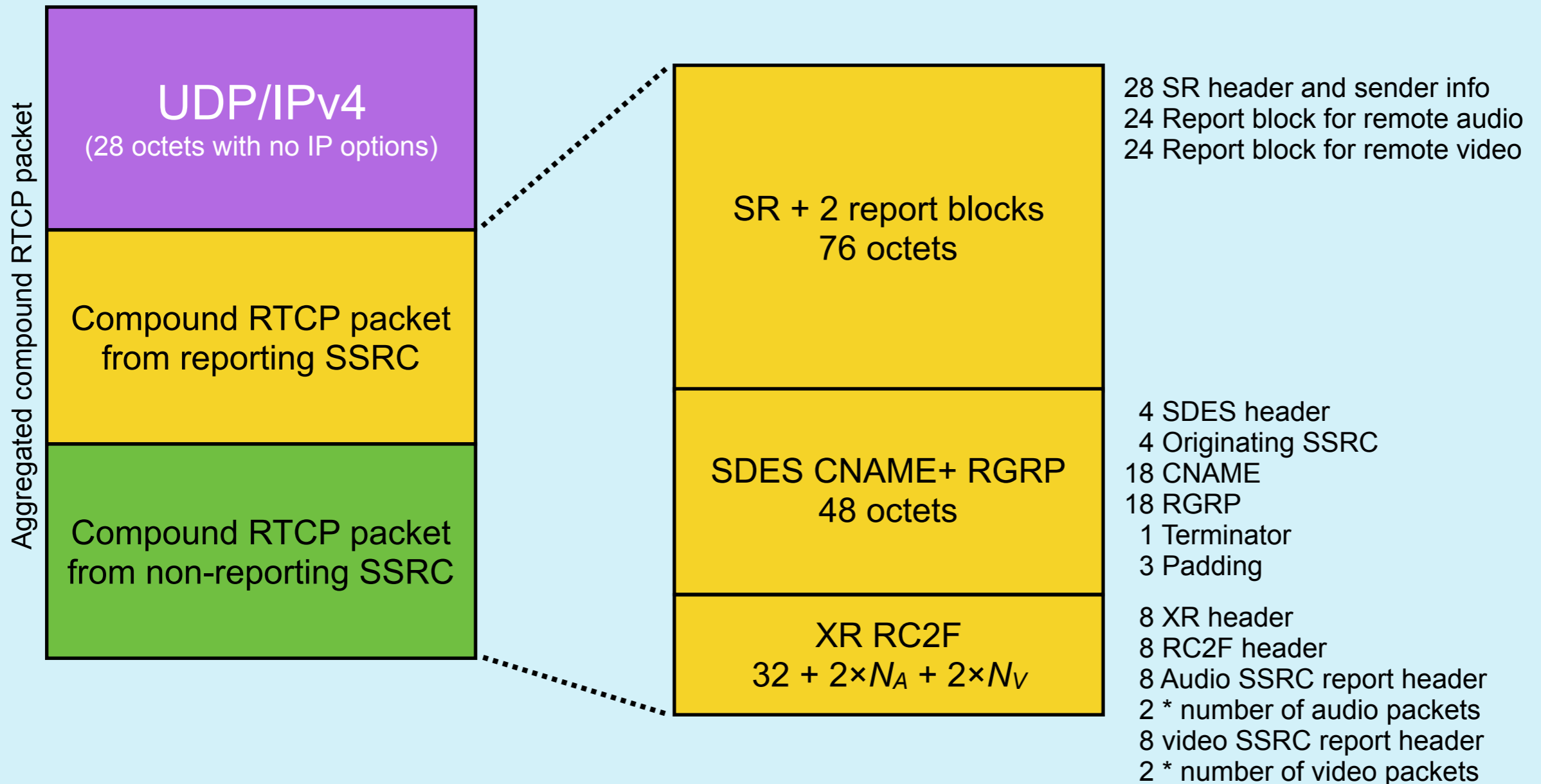
- Two SSRC → need to aggregate feedback into a single RTCP packet
  - Each packet is an aggregation of a compound RTCP packet from the audio SSRC and a compound RTCP packet from the video SSRC
- Assume RTCP reporting groups are used:
  - One SSRC is designated as the reporting SSRC
  - The other SSRC delegates its reports to that SSRC
  - The reports are aggregated, so it doesn't matter which is chosen as reporting SSRC

# Scenario 2: Video conference – compound packets



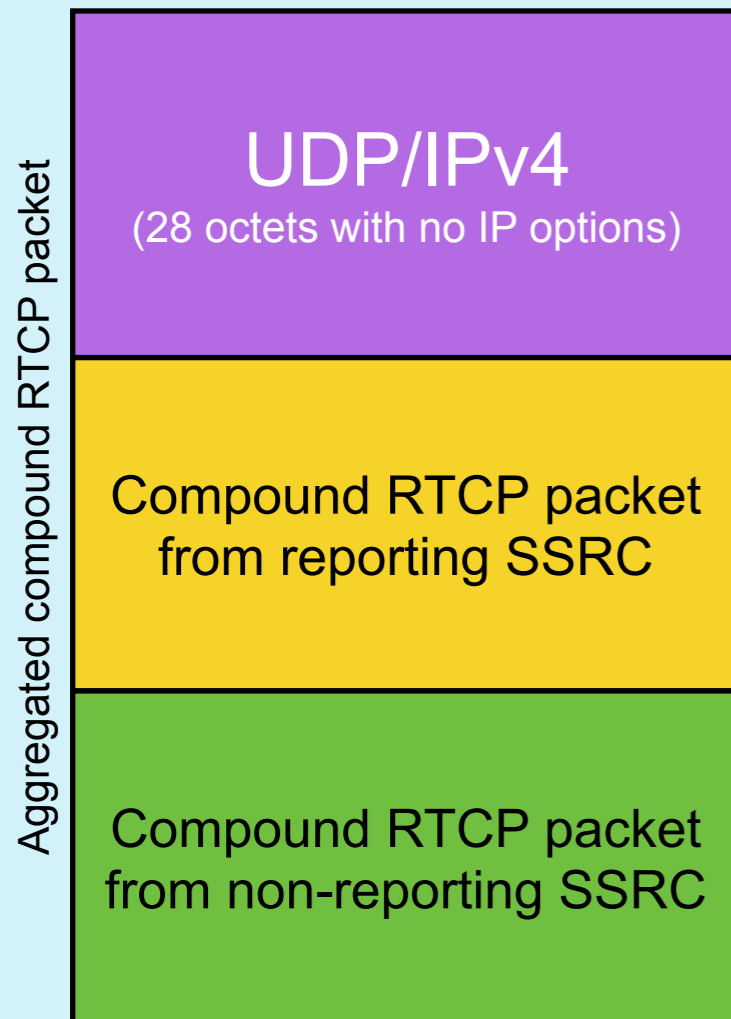
- Packets from non-reporting SSRC are 68 octets (assuming RFC 7022-style CNAME)

# Scenario 2: Video conference – compound packets



- Packets from reporting SSRC are  $156 + 2 \times N_A + 2 \times N_V$  octets

## Scenario 2: Video conference – compound packets



- 28
- $156 + 2 \times N_A + 2 \times N_V$  octets
- 68 octets
- Total =  $252 + 2 \times N_A + 2 \times N_V$  octets
- Since this reports on two SSRCs, it is halved before use:  $S_c = (252 + 2 \times N_A + 2 \times N_V) / 2$

## Scenario 2: Video conference – $B_{rtcp}$ calculation

- Assume:
  - Constant rate media
  - Video frames equal size
  - Audio at 50 packets per second (20ms frames)
  - MTU around 1500 octets
- RTCP bandwidth calculation as for scenario 1:

$$B_{rtcp} = (n \times (S_c + N_{nc} \times S_{nc})) / (N_r \times T_f \times (1 + N_{nc}))$$

with

$$S_c = (252 + 2 \times N_A + 2 \times N_V) / 2$$

$$N_{nc} = 0$$

$T_f$  based on chosen video frame rate

$$N_r = 1 \text{ (report on every frame)}$$

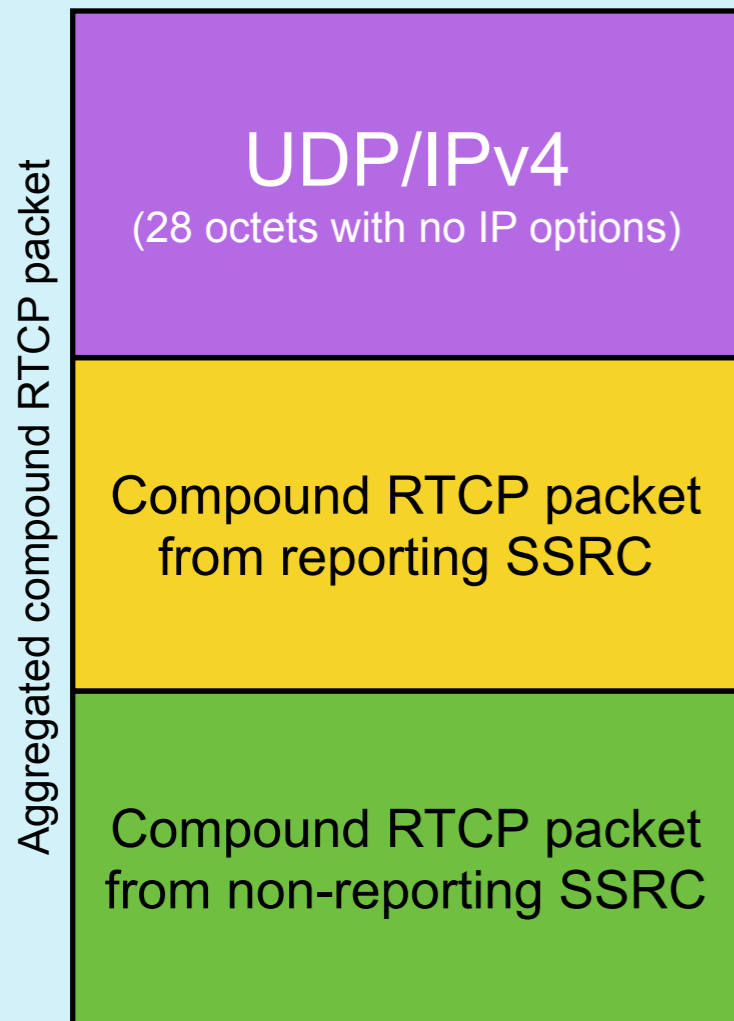
# Scenario 2: Video conference – required RTCP bandwidth

Media Rate (kbps)	Video Frame Rate ( $1/T_f$ )	Video packets per report: $N_v$	Audio packets per report: $N_a$	Required RTCP bandwidth, $B_{rtcp}$ in kbps (and as % of media rate)
100	8	1	6	33.3 (33%)
200	16	1	3	65.0 (33%)
350	30	1	2	120.1 (35%)
700	30	2	2	121.9 (17%)
700	60	1	1	240.0 (34%)
1024	30	3	2	122.8 (12%)
1400	60	2	1	241.8 (17%)
2048	30	6	2	125.6 (6%)
2048	60	3	1	243.8 (12%)
4096	30	12	2	131.3 (3%)
4096	60	6	1	249.4 (6%)

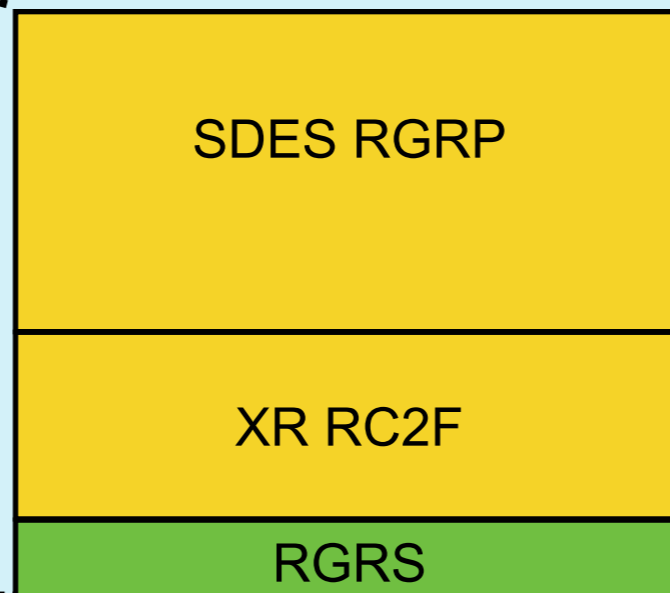
Sending only compound RTCP packets

$B_{rtcp}$  scales linearly with  $N_r$  (i.e., reporting every 2nd frame halves the required RTCP bandwidth)

## Scenario 2: Video conference – reduced size packets



- Reports from two SSRCs aggregated into compound packets irrespective of whether RFC5506 is used – *reduced size* RTCP rather than *non-compound* RTCP
- Omit SR and SDES CNAME from aggregated packet



- Gives  $S_{nc} = (96 + 2 \times N_v + 2 \times N_a) / 2$
- Repeat calculation with  $N_{nc} = 1$  indicating that we alternate regular and reduced size RTCP

# Scenario 2: Video conference – required RTCP bandwidth

Media Rate (kbps)	Video Frame Rate ( $1/T_f$ )	Video packets per report: $N_v$	Audio packets per report: $N_a$	Required RTCP bandwidth, $B_{rtcp}$ in kbps (and as % of media rate)
100	8	1	6	23.5 (23%)
200	16	1	3	45.5 (23%)
350	30	1	2	84.4 (24%)
700	30	2	2	85.3 (12%)
700	60	1	1	166.9 (24%)
1024	30	3	2	86.2 (8%)
1400	60	2	1	168.8 (12%)
2048	30	6	2	89.1 (4%)
2048	60	3	1	170.6 (8%)
4096	30	12	2	94.7 (2%)
4096	60	6	1	176.3 (4%)

Alternating regular and reduced-size RTCP packets

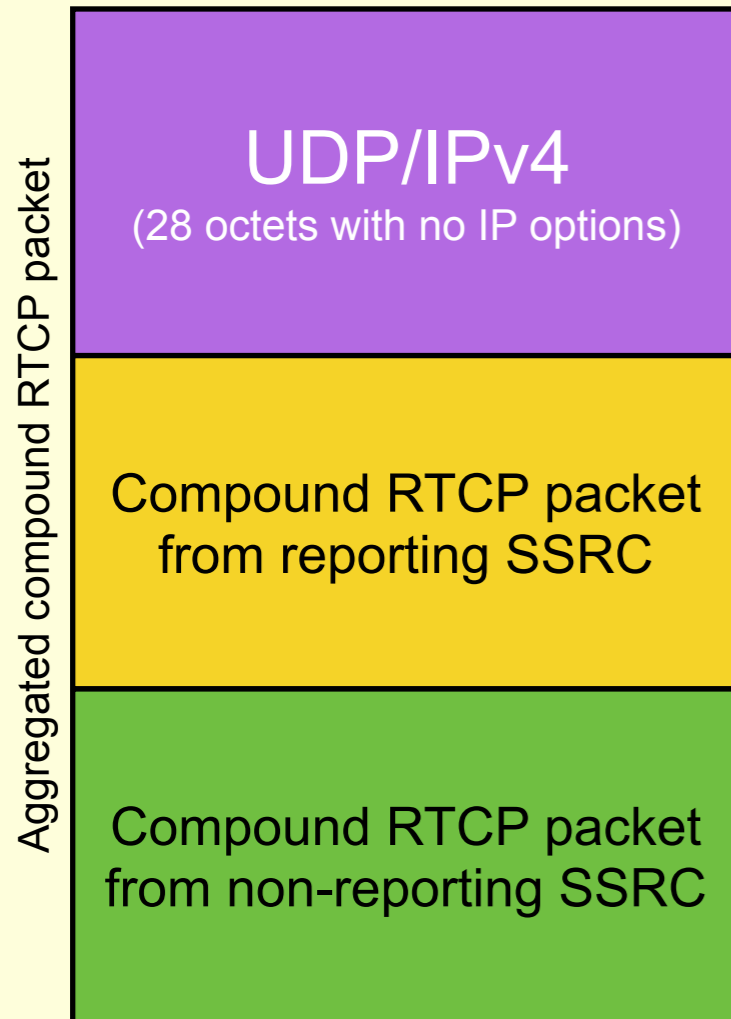
$B_{rtcp}$  scales linearly with  $N_r$  (i.e., reporting every 2nd frame halves the required RTCP bandwidth)



## Scenario 2: Video conference

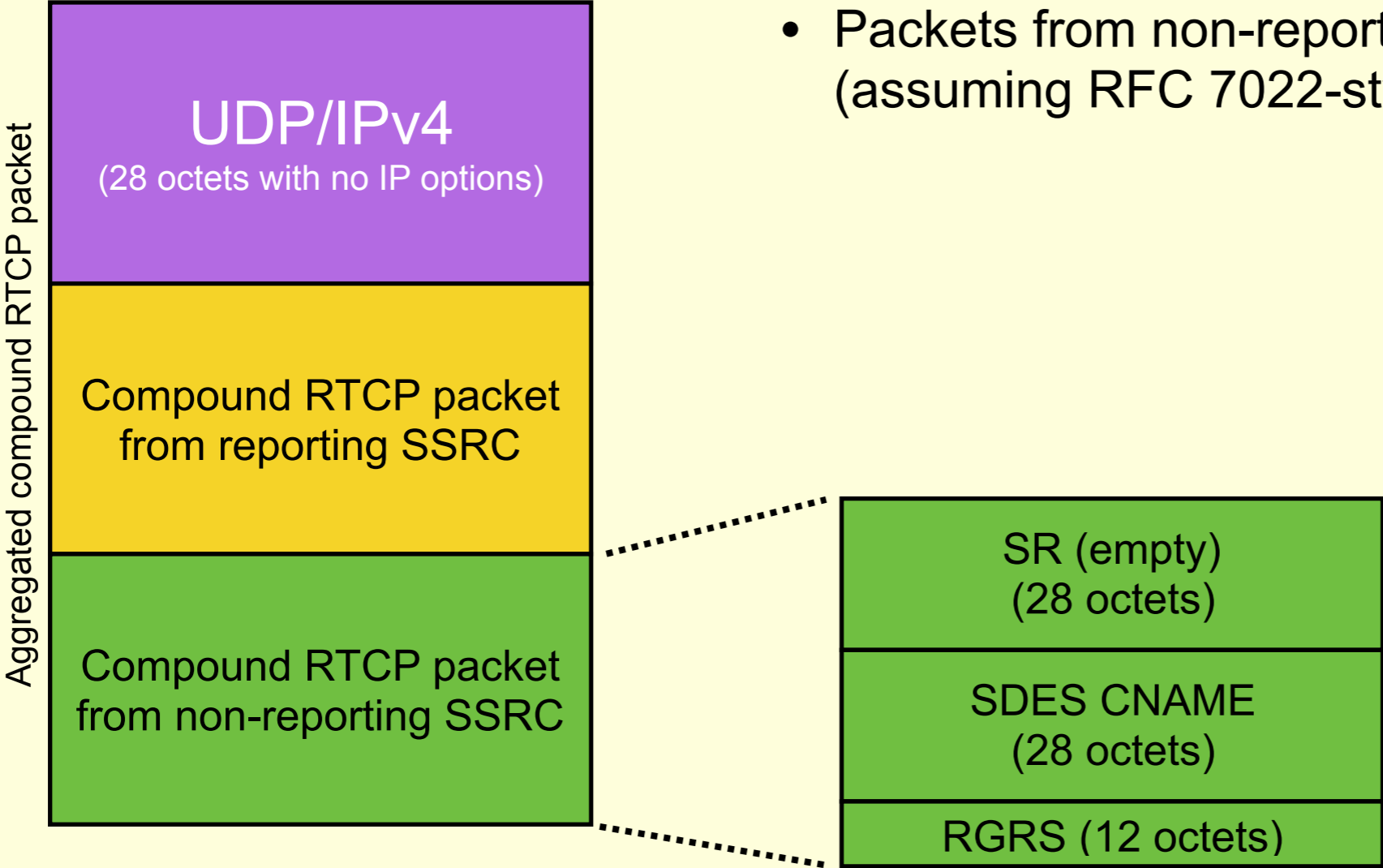
- Audio and video send regular RTCP reports as a compound packet
- Separate non-compound transport-level RTCP congestion feedback packets sent
  - RTP packets require extra header extension → 8 octets per packet

# Scenario 2: Video conference – compound packets



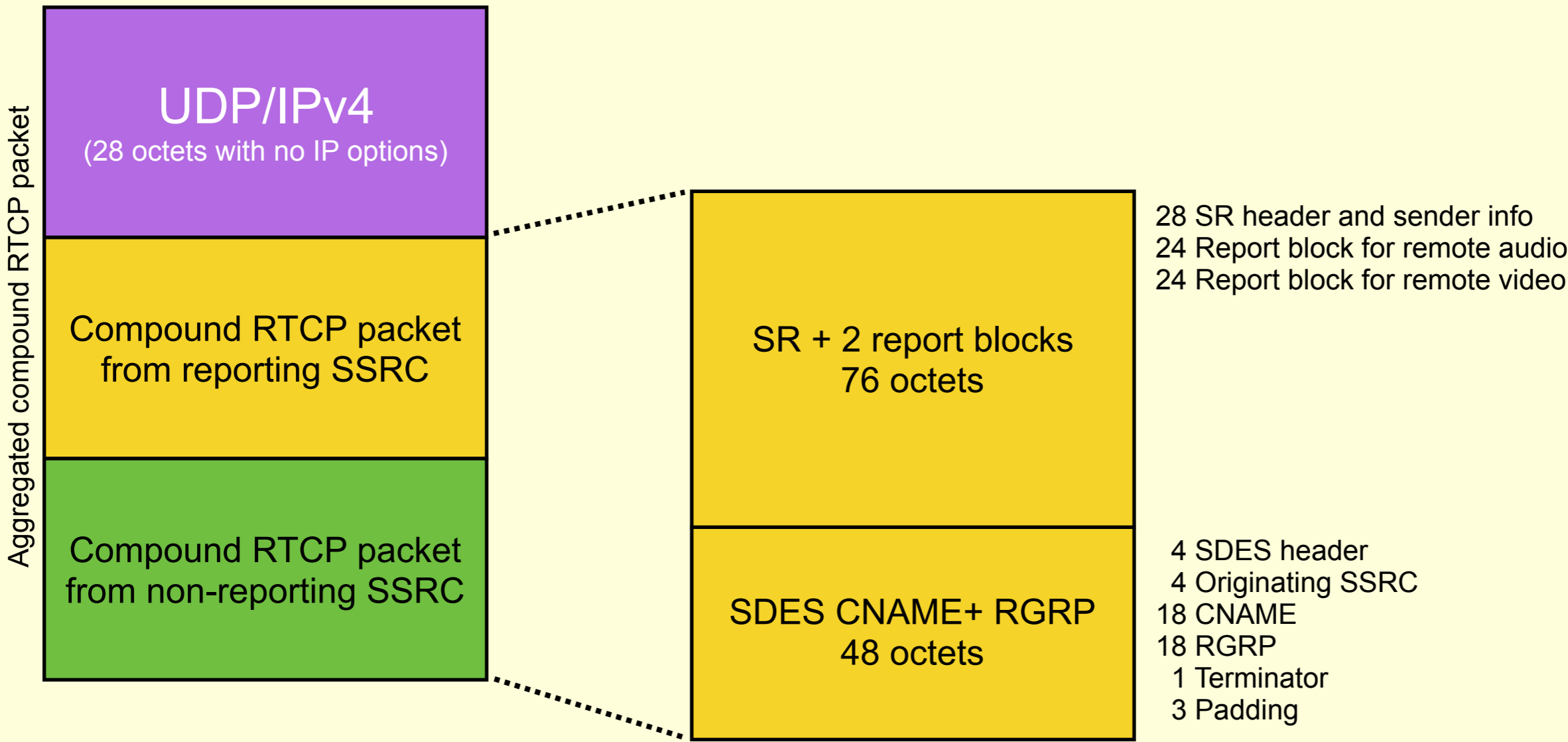
- Two SSRC → need to aggregate feedback into a single RTCP packet
  - Each packet is an aggregation of a compound RTCP packet from the audio SSRC and a compound RTCP packet from the video SSRC
- Assumes RTCP reporting groups are used:
  - One SSRC is designated as the reporting SSRC
  - The other SSRC delegates its reports to that SSRC
  - The reports are aggregated, so it doesn't matter which is chosen as reporting SSRC

# Scenario 2: Video conference – compound packets



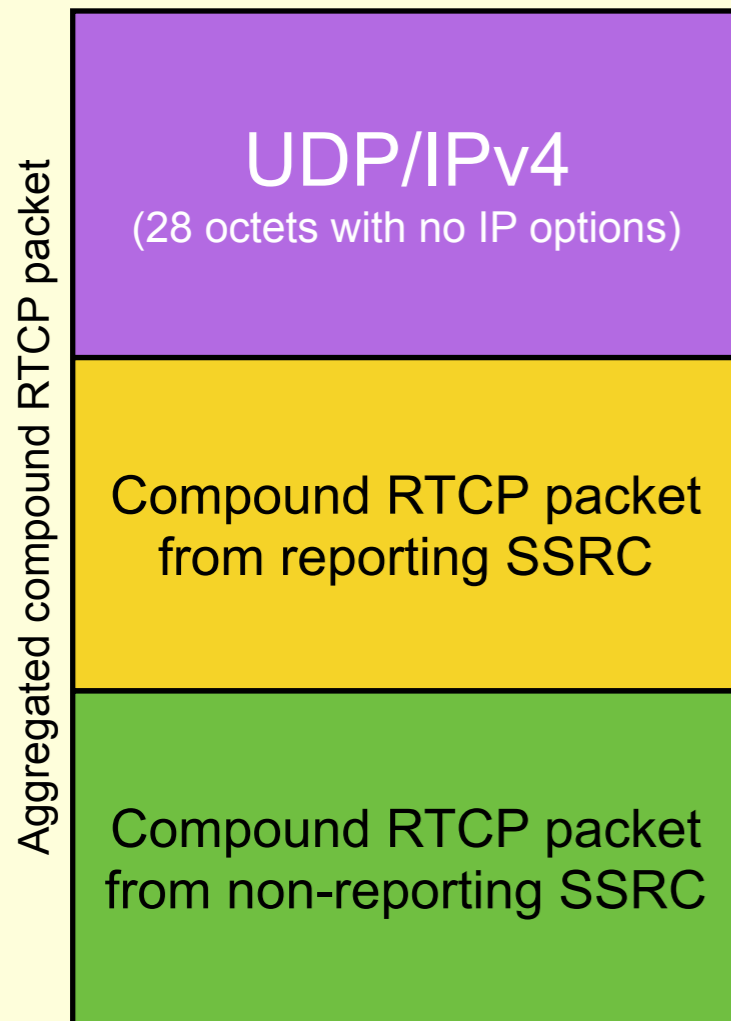
- Packets from non-reporting SSRC are 68 octets (assuming RFC 7022-style CNAME)

# Scenario 2: Video conference – compound packets



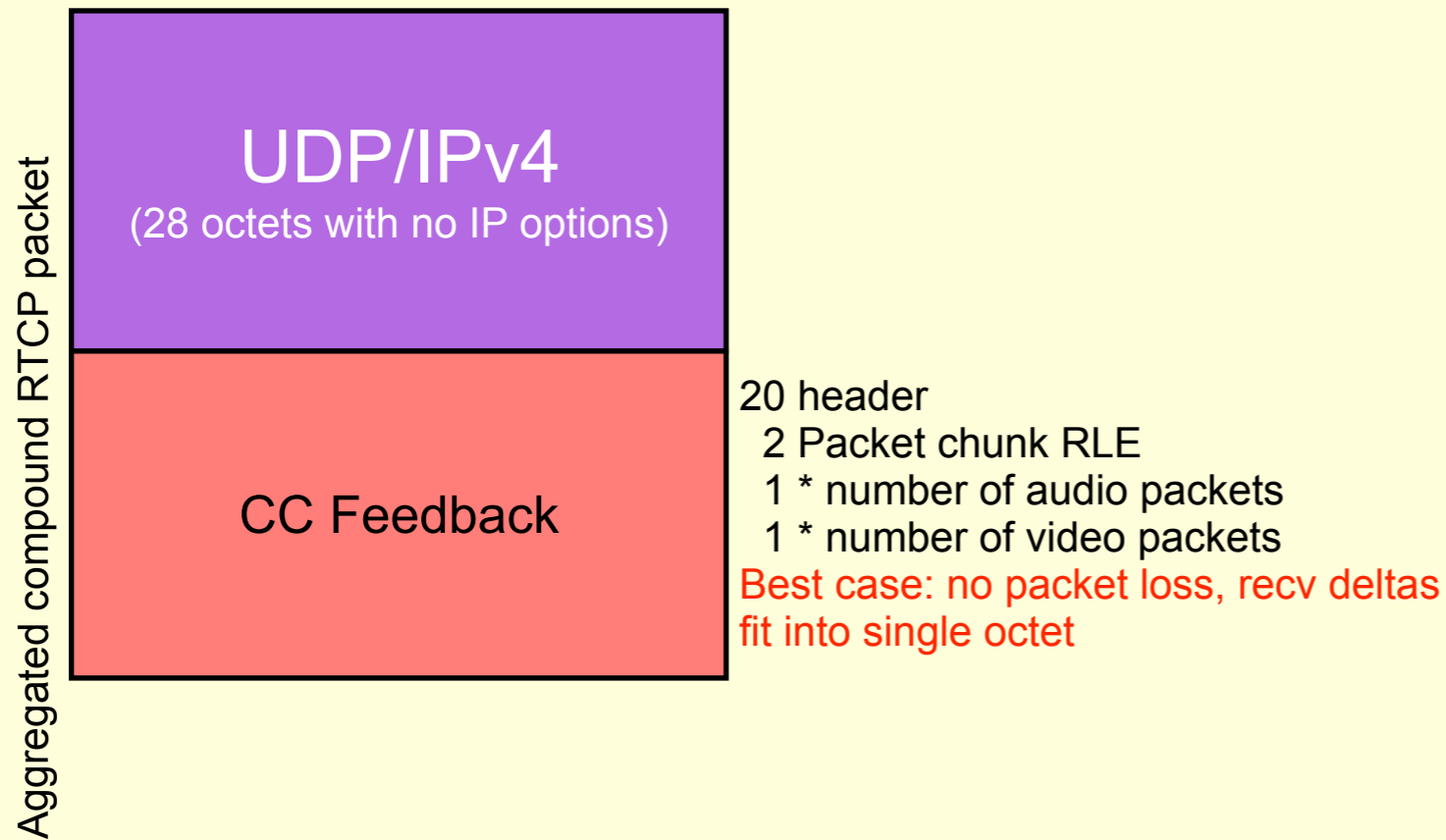
- Packets from reporting SSRC are 124 octets

# Scenario 2: Video conference – compound packets



- 28
- 124 octets
- 68 octets
- Total = 220 octets
- Since this reports on two SSRCs, it is halved before use:  $S_c = 220/2 = 110$

# Scenario 2: Video conference – reduced-size packets



- Reduced size packets:  $S_{nc} = 50 + N_a + N_v$  octets (best case)
- A single CC Feedback packet is sent, reporting on all sources

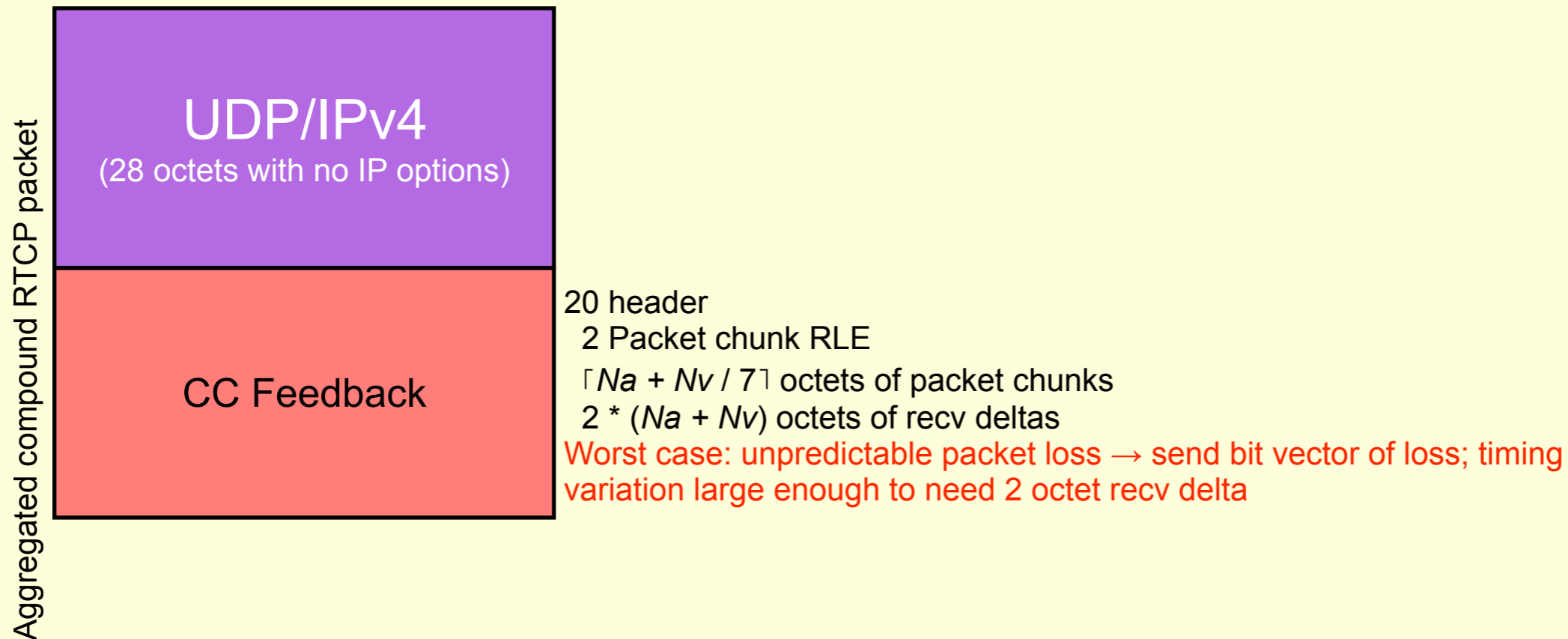
# Scenario 2: Video conference – required RTCP bandwidth

Media Rate (kbps)	Video Frame Rate ( $1/T_f$ )	Video packets per report: $N_v$	Audio packets per report: $N_a$	Required RTCP bandwidth, $B_{rtcp}$ in kbps (and as % of media rate)
100	8	1	6	20.9 (21%)
200	16	1	3	41.0 (21%)
350	30	1	2	76.4 (22%)
700	30	2	2	76.9 (11%)
700	60	1	1	151.9 (22%)
1024	30	3	2	77.3 (8%)
1400	60	2	1	152.8 (11%)
2048	30	6	2	78.8 (4%)
2048	60	3	1	153.8 (8%)
4096	30	12	2	81.6 (2%)
4096	60	6	1	156.6 (4%)

Best case – packet loss or timing variation will affect  $B_{rtcp}$

Excludes overhead of RTP header extension

# Scenario 2: Video conference – reduced-size packets



- Reduced size packets:  $S_{nc} = 50 + \lceil N_a + N_v / 7 \rceil + 2 * (N_a + N_v)$  octets (worst case)
- A single CC Feedback packet is sent, reporting on all sources



# Scenario 2: Video conference – required RTCP bandwidth

Media Rate (kbps)	Video Frame Rate ( $1/T_f$ )	Video packets per report: $N_v$	Audio packets per report: $N_a$	Required RTCP bandwidth, $B_{rtcp}$ in kbps (and as % of media rate)
100	8	1	6	21.9 (22%)
200	16	1	3	42.3 (21%)
350	30	1	2	78.3 (22%)
700	30	2	2	79.2 (11%)
700	60	1	1	154.7 (22%)
1024	30	3	2	80.2 (8%)
1400	60	2	1	156.6 (11%)
2048	30	6	2	83.4 (4%)
2048	60	3	1	158.4 (8%)
4096	30	12	2	89.1 (2%)
4096	60	6	1	165.1 (4%)

Worst case – required RTCP bandwidth increases  
 Excludes overhead of RTP header extension

# Scenario 2: Video conference – required RTCP bandwidth

Media Rate (kbps)	Video Frame Rate ( $1/T_f$ )	Video packets per report: $N_v$	Audio packets per report: $N_a$	RTCP Bandwidth (worst case)	RTCP Bandwidth (best case)	RTCP Bandwidth
100	8	1	6	21.9 (22%)	20.9 (21%)	23.5 (23%)
200	16	1	3	42.3 (21%)	41.0 (21%)	45.5 (23%)
350	30	1	2	78.3 (22%)	76.4 (22%)	84.4 (24%)
700	30	2	2	79.2 (11%)	76.9 (11%)	85.3 (12%)
700	60	1	1	154.7 (22%)	151.9 (22%)	166.9 (24%)
1024	30	3	2	80.2 (8%)	77.3 (8%)	86.2 (8%)
1400	60	2	1	156.6 (11%)	152.8 (11%)	168.8 (12%)
2048	30	6	2	83.4 (4%)	78.8 (4%)	89.1 (4%)
2048	60	3	1	158.4 (8%)	153.8 (8%)	170.6 (8%)
4096	30	12	2	89.1 (2%)	81.6 (2%)	94.7 (2%)
4096	60	6	1	165.1 (4%)	156.6 (4%)	176.3 (4%)

Results for draft-holmer-rmcat-transport-wide-cc-extensions-01 exclude overhead of RTP header extension

# Scenario 2: Video conference – required RTCP bandwidth

Media Rate (kbps)	Video Frame Rate ( $1/T_f$ )	Video packets per report: $N_v$	Audio packets per report: $N_a$	RTCP Bandwidth (worst case) + overhead	RTCP Bandwidth (best case) + overhead	RTCP Bandwidth
100	8	1	6	25.4 (25%)	24.4 (24%)	23.5 (23%)
200	16	1	3	46.3 (23%)	45.0 (23%)	45.5 (23%)
350	30	1	2	83.9 (24%)	82.0 (23%)	84.4 (24%)
700	30	2	2	79.2 (11%)	84.4 (12%)	85.3 (12%)
700	60	1	1	162.2 (23%)	159.4 (23%)	166.9 (24%)
1024	30	3	2	89.5 (9%)	86.7 (8%)	86.2 (8%)
1400	60	2	1	167.8 (12%)	164.1 (12%)	168.8 (12%)
2048	30	6	2	98.4 (5%)	93.8 (5%)	89.1 (4%)
2048	60	3	1	173.4 (8%)	168.8 (8%)	170.6 (8%)
4096	30	12	2	115.3 (3%)	107.8 (3%)	94.7 (2%)
4096	60	6	1	190.3 (5%)	182.8 (4%)	176.3 (4%)

Adding RTP header extension overhead:  $8 \times (N_a + N_v)$  octets per RTCP reporting interval

$= (8 \times (N_a + N_v)) / T_f$  octets per second (since reporting interval = frame rate)

# Conclusions

- Compared performance of draft-dt-rmcat-feedback-message-02 and draft-holmer-rmcat-transport-wide-cc-extensions-01
  - draft-holmer-rmcat-transport-wide-cc-extensions-01 needs less RTCP bandwidth for VoIP
    - Doesn't include ECN feedback
  - draft-dt-rmcat-feedback-message-02 generally needs less RTCP bandwidth to report on video once RTP header extension overhead taken into account
    - But RTCP bandwidth needed is highly dependent on data rate and packet loss patterns
    - Compatible with standard RTP use of SSRCs
    - Conveys ECN feedback