RTP retransmission framework

Draft-ietf-leon-rtp-retransmission-02.txt

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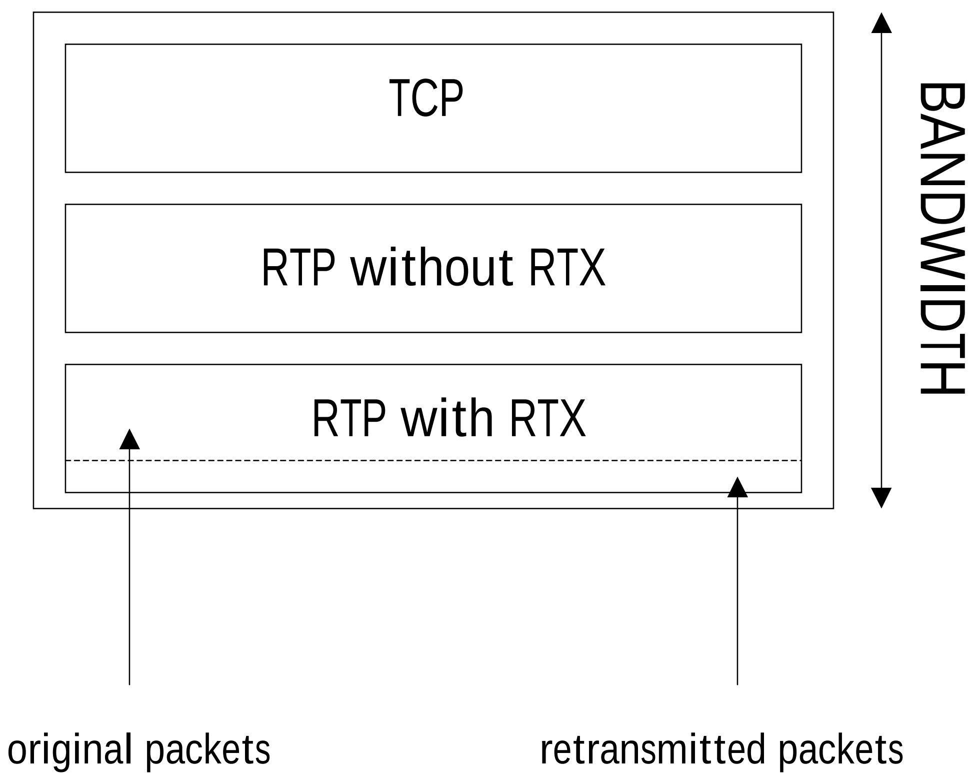
Overview

- IETF 52 comments
  - Congestion section needs expansion and clarifications
  - Piggybacking retransmitted packets (RFC 2198)
  - FEC (RFC 2733) as a retransmission payload format alternative
- New experimental results
- Way forward

Congestion Control

- An RTP Profile defines an appropriate congestion control mechanism so that an RTP sender knows its fair bitrate at a given packet loss rate
- When retransmission is used the fair bitrate includes BOTH the original data and the retransmitted data
- RTP with retransmission is then as fair as RTP without retransmission

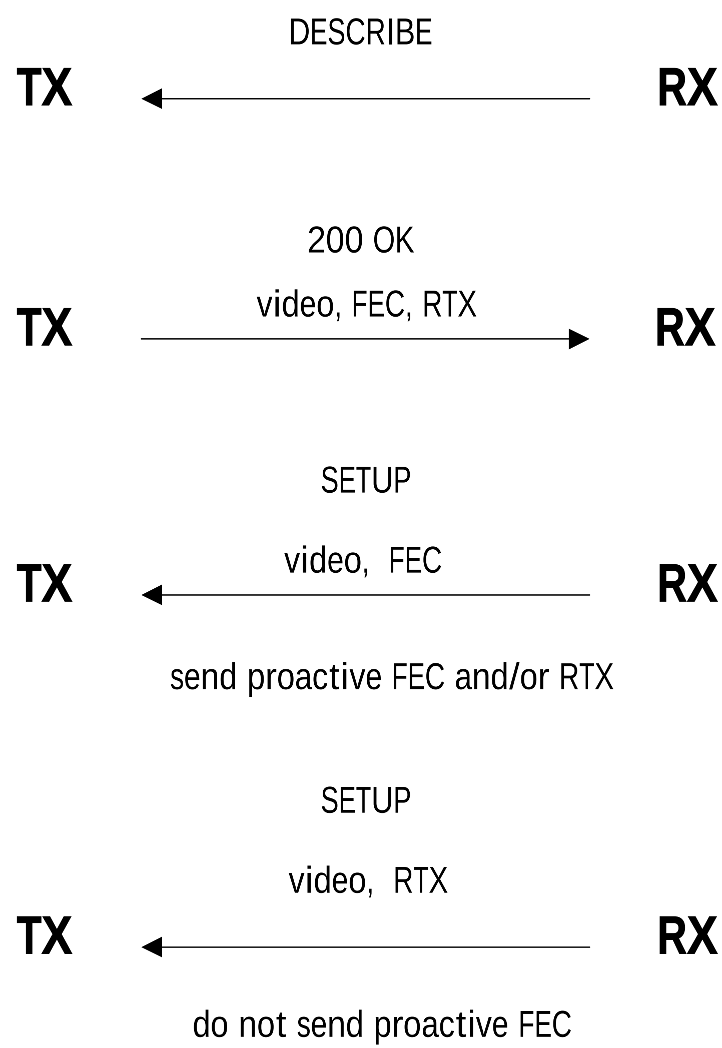


Piggybacking Retransmission

- Piggybacking allows to send the original packets and the retransmitted packets in the same RTP stream
- If RFC 2198 (redundant audio data) is used instead of the RTX payload format, original SN not available
  - necessary for some payload format such as conversational text
  - necessary to perform multiple retransmission
- RFC 2198 and RTX payload format may be used complementarily
  - marker bit and CSRC list are lost
- Multiple retransmission rules
  - If a requested packet contains both original and retransmitted data, the sender should retransmit only the original data
  - Multiple retransmission achieved by requesting the original packet SN multiple times

FEC Payload Format

- RFC 2733 (FEC) may be used to retransmit packet
  - Appropriate in particular for multicast session where a single packet may repair the loss of different packets at different receivers (scalable reliable multicast)
- Motivations for defining retransmission payload format
  - Less overhead (3 bytes vs. 12 bytes)
  - Packet TS is the media TS
  - Out-of-band signalling (session setup) of proactive FEC vs. retransmission



Comparison Early/Regular Feedback with Retransmission

PLR (%)	RTCP BW (%)	Total # of RTCP packets	# of early requests	# of regular requests	% of discarded early requests	Average waiting time (ms)	Average discarded early request waiting time(ms)	Max. Waiting Time (ms)	RTCP Mean Interval (ms)
3	5.0	501	158	156	50	118	237	544	302
5	5.0	491	203	294	59	135	228	582	307
10	5.0	480	221	737	77	203	264	598	314
			No Early Feedback						
3	5.0	496	N/A	317	N/A	144	N/A	299	305
5	5.0	489	N/A	497	N/A	147	N/A	306	308
10	5.0	478	N/A	976	N/A	150	N/A	319	316

- Video (48 kbps, 7.5 fps)
- RTCP session bw 5%
- Congestion packet loss (error pattern for Internet experiments ITU SG 16)
- No randomisation of RTCP reporting interval to facilitate interpretation

- Waiting time: time between detecting packet loss and sending a request in an RTCP report
- Early request: packet request sent in an early RTCP report
- Regular request: packet request sent in an early RTCP report
- Discarded early request: request which could not be sent in an early report and thus sent in the next regular report

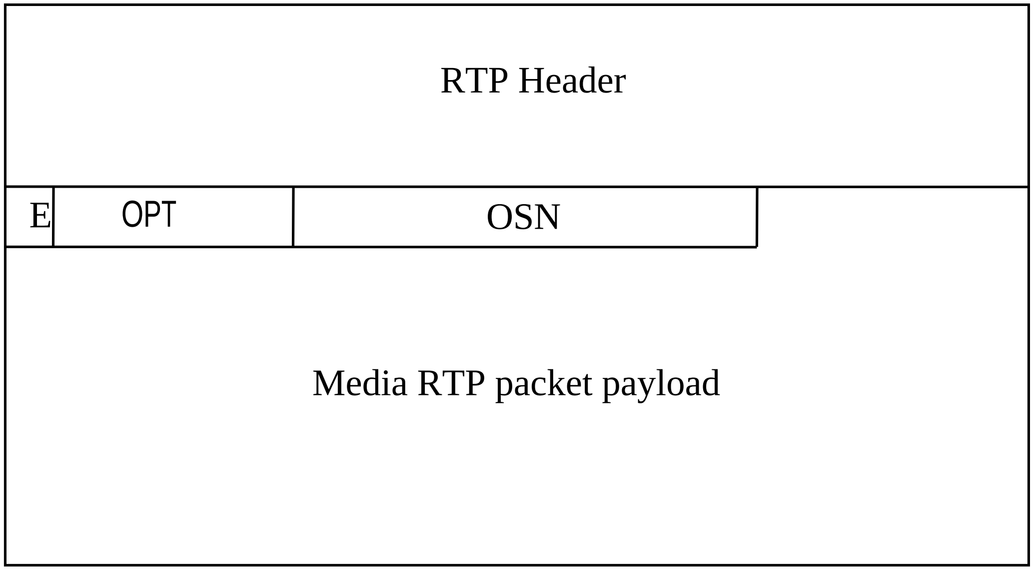
Conclusion - Way Forward

- Why do we need an RTP retransmission framework document?
  - payload format
  - protocol rules
  - congestion control
  - session setup signalling
  - comparison/use with existing payload formats (redundancy, FEC)
- Implementation, performance evaluation
- No known IPR in this framework

Backup Slides

Retransmission payload format

- RTP header TS is the original packet timestamp
- E extension bit
- OPT (7 bits) is the original packet payload type
- OSN (16 bits) is the original packet SN



IETF 51 Comments

- Associating retransmission stream and SSRC:
  - Retransmissions are required to be sent to a different RTP session (multicast group or unicast address/port) from the original data
  - The original stream and the retransmission stream should use the same SSRC

- Why not use the same SSRC and send to the same multicast group with a different PT?
  - A lost packet may be a retransmitted packet or new data. Data loss detection is necessary for some applications (e.g. conversational text RFC 2793)
  - Receiver estimation of a missing packet TS
  - RTCP jitter value is incorrect

Performance evaluation

- Implementation of retransmission draft and extended feedback profile (draft-ietf-avt-rtcp-feedback-00.txt)
- Comparison test
  - RTP without retransmission with fixed transmission rate
  - RTP with retransmission and rate adaptation (and additional buffering)
- Example test conditions
  - RTP session bw 64 kbps (include RTP/UDP/IP headers)
  - RTCP receiver session bw 1.6 kbps (5% RTP session bandwidth shared equally between sender and receiver)
  - Packet loss 5% (on both the forward and reverse path)
  - RTT 500 ms
  - Buffering delay 1.2 s
  - One retransmission attempt only
- Measured RTCP overhead
  - No retransmission
    - Average RTCP average packet size: 69.8 bytes
    - Average RTCP interval: 348 ms
  - With retransmission
    - Average RTCP average packet size: 74.3 bytes
    - Average RTCP interval: 371 ms