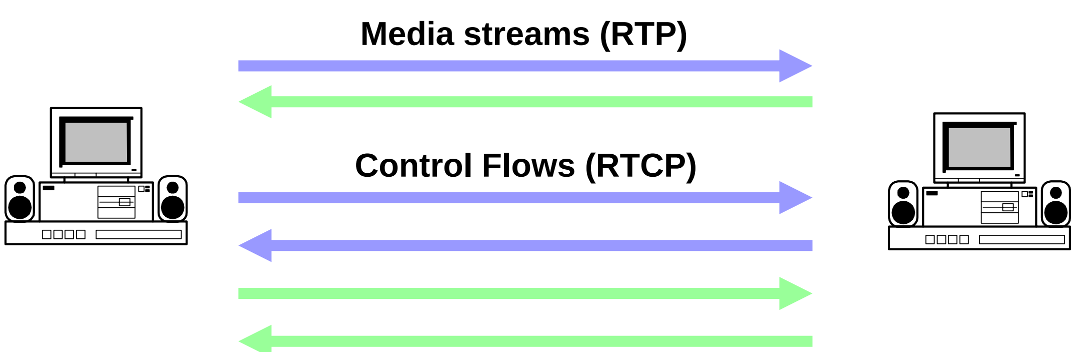
Real-Time Transport Protocol (RTP)

HELSINKI UNIVERSITY OF TECHNOLOGY
NETWORKING LABORATORY

Real-time Transport Protocol (1)

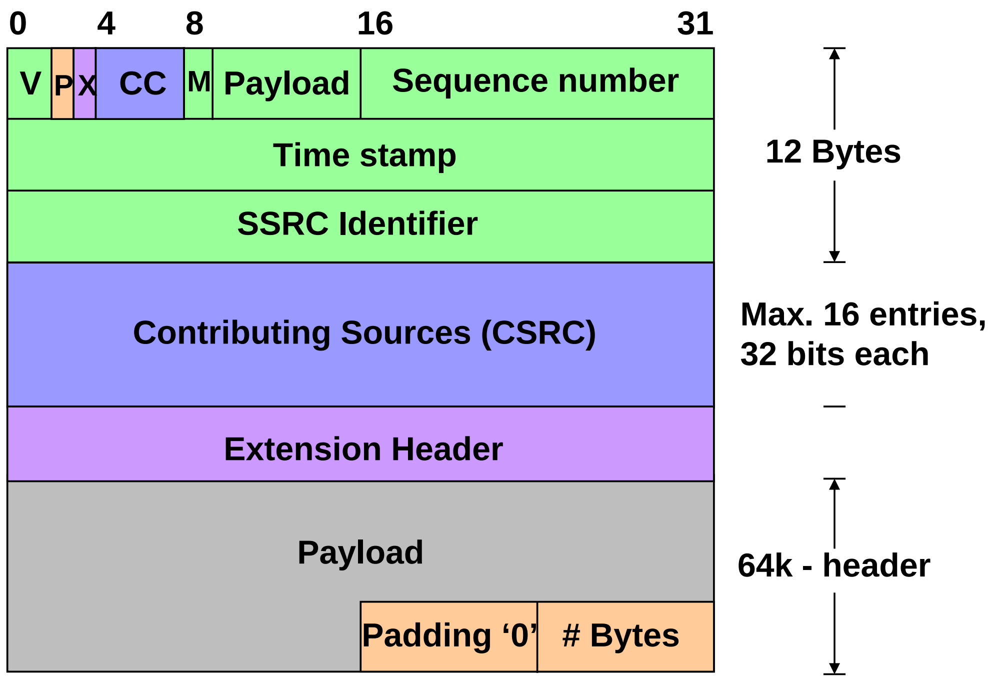
- RTP Functionality (RFC 3550)
  - framing for audio/video information streams
  - preserve intra- and inter-stream timing
  - mechanisms for awareness of others in a conference
  - ➲ RTP sessions



Real-time Transport Protocol (2)

- Standard RTP packet header
  - Independent of *payload type*
  - Possibly seconded by *payload header*
- Mechanisms
  - Detect packet loss, cope with reordering
    - sequence number per media stream
  - Determine variations in transmission delays
    - media specific time stamp (e.g., 8 kHz for PCM audio)
    - allows receiver to adapt playout point for continuous replay
  - Source identification
    - possibly mixed from several sources
  - Payload type identifier

RTP Header



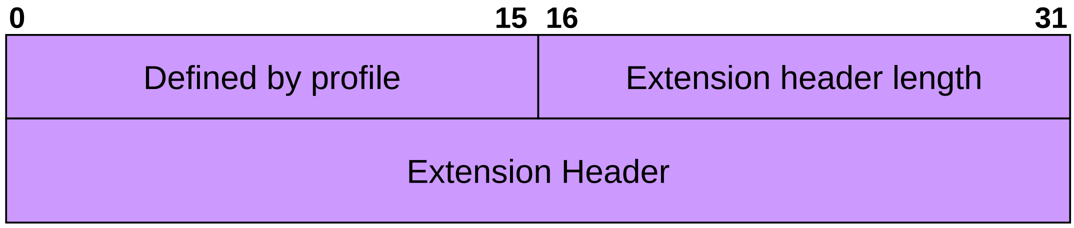
RTP Header Fields (1)

V: Version — version 2 defined in RFC 1889

P: Padding — indicates padding
bytes indicated in last byte

X: eXtension bit — extension header is present

Extension header — single additional header (TLV coded)



CC: CSRC count — # of contributing sources

CSRC: contributing sources —
which sources have been "mixed"
to produce this packet's contents

RTP Header Fields (2)

M: Marker bit — marks semantical boundaries in media stream (e.g. talk spurt)

Payload type — indicates packet content type

Sequence # — of the packet in the media stream (strictly monotonically increasing)

Timestamp — indicates the instant when the packet contents was sampled (measured to media-specific clock)

SSRC: synchronization source —
identification of packet originator

Real-time Transport Control Protocol

Mechanisms:

- Receivers constantly measure transmission quality
  - delay, jitter, packet loss
- Regular control information exchange between senders and receivers
  - feedback to sender (receiver report)
  - feed forward to recipients (sender report)
- Allows applications to adapt to current QoS
- Overhead limited to a small fraction (default: 5% max.) of total bandwidth per RTP session
  - members estimate number of participants
  - adapt their own transmission rate

Obtaining sufficient capacity: outside of RTP

RTCP Sender Report

- Enable cross-media stream synchronization
  - Relate stream-specific RTP time stamp to wall clock time
  - NTP timestamp + RTP timestamp
  - Playout adjustment to be performed by the receivers
- Provide data point for RTT measurement
  - NTP timestamp
- Provide feed forward about data transmitted
  - Transmit sender’s packet and byte count
  - Enable receiver to do proper loss calculation
- Include Receiver Reports for the sender as well

RTCP Sender Report (SR)

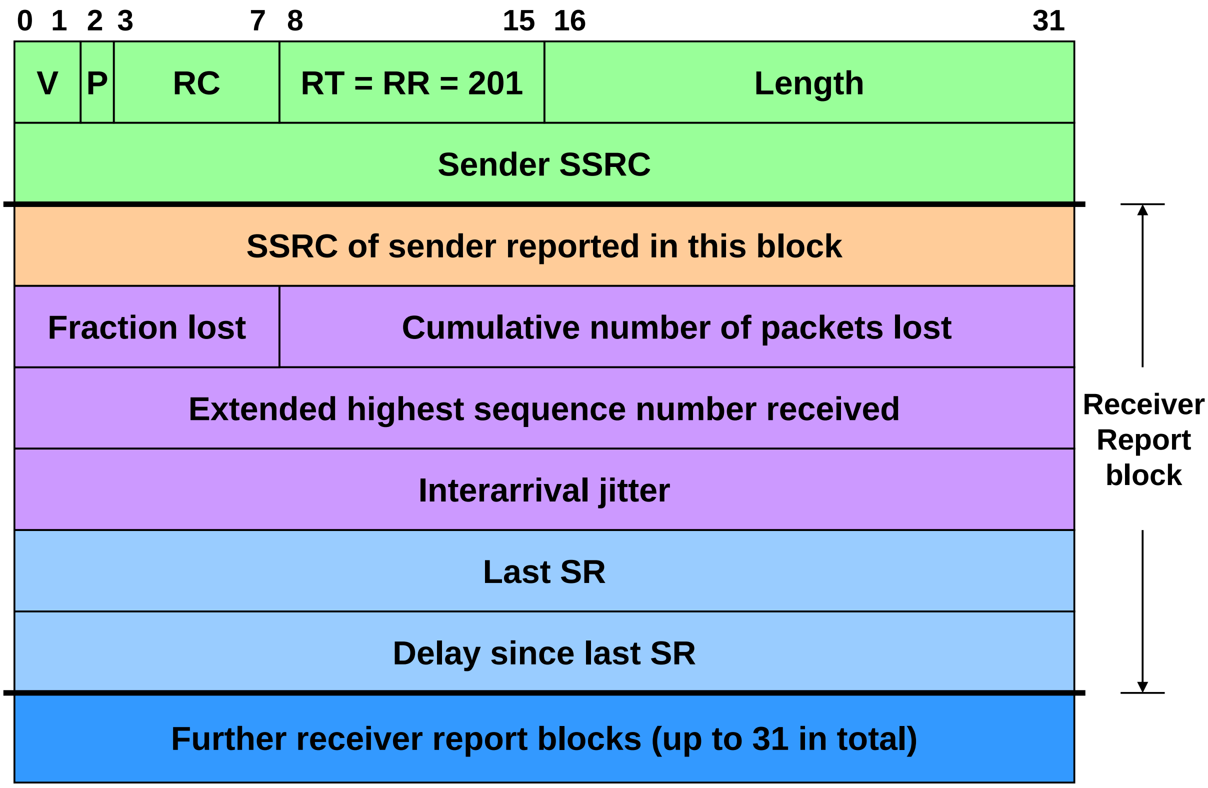
0 1	2	3 … 7	8 … 15	16 … 31
V	P	RC	RT = SR = 200	Length
Sender SSRC				
64 Bit NTP Timestamp (MSW)				
64 Bit NTP Timestamp (LSW)				
RTP Timestamp				
Cumulative number of packets sent				
Cumulative number of octets sent				
Receiver Report blocks (0 – 31)				

Sender Info: 64 Bit NTP Timestamp (MSW) through Cumulative number of octets sent

RTCP Receiver Report

- Feedback timing for RTT estimation
  - SR Timestamp
    - Middle 32 bits taken from the last SR’s NTP timestamp
  - Delay since last SR
    - Local delay at receiver between receiver SR and sending the RR block
    - Measured in units of 1 / 65556 seconds
- Provide per-sender reception statistics
  - Total number of packets lost
  - Fraction of packets lost (in units of 1 / 256)
  - Highest sequence number received so far
  - Jitter of received packets
- Enable adaptive sender behavior
  - Adjust codecs, codec parameters, transmission rate, etc.

RTCP Receiver Report (RR)



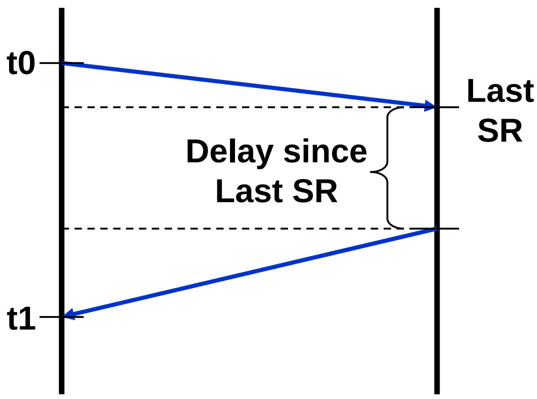
HELSINKI UNIVERSITY OF TECHNOLOGY
NETWORKING LABORATORY

RTCP Statistics Collection (Sender)

- Round-Trip Time (sender only)
  - Derived from time stamps in RR
  - Simple formula:

    RTT = t1 – t0 – DSL_SR

  - RTT may be asymmetric!



- Byte count
- Packet count

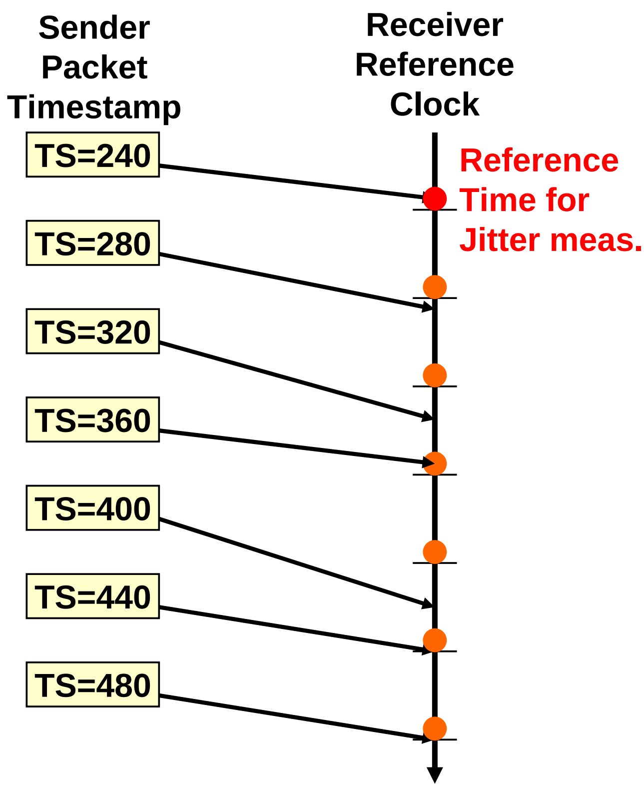
RTCP Statistics Collection (Receiver)

- Packet Loss
  - Calculated from gaps in sequence number space
    - First (lowest packet sequence number) received
  - Expected number of packets = current – lowest
  - Received number of packets
    - Count duplicates, out-of-order, and late packets as received!
  - Absolute # of lost packets = expected – received
    - May be negative!
  - Fraction of lost packet
    - Loss since last SR or RR packet was sent
  - Loss of all packets not detected!
- Extended highest sequence number received (32 bits)
- Time of last SR reception
- Jitter

RTCP Interarrival Jitter Estimation

- Receiver measures in time units of the media clock
- Relates it to local real-time clock
- Initialized through first packet received
- Derives expected reception time
- Calculates deviation D upon packet reception
- Sampled for each packet
- Jitter derived for each peer of successively received packets
  - Ordering is not relevant
- Weighing function:

  J = J' + (D – J') / 16

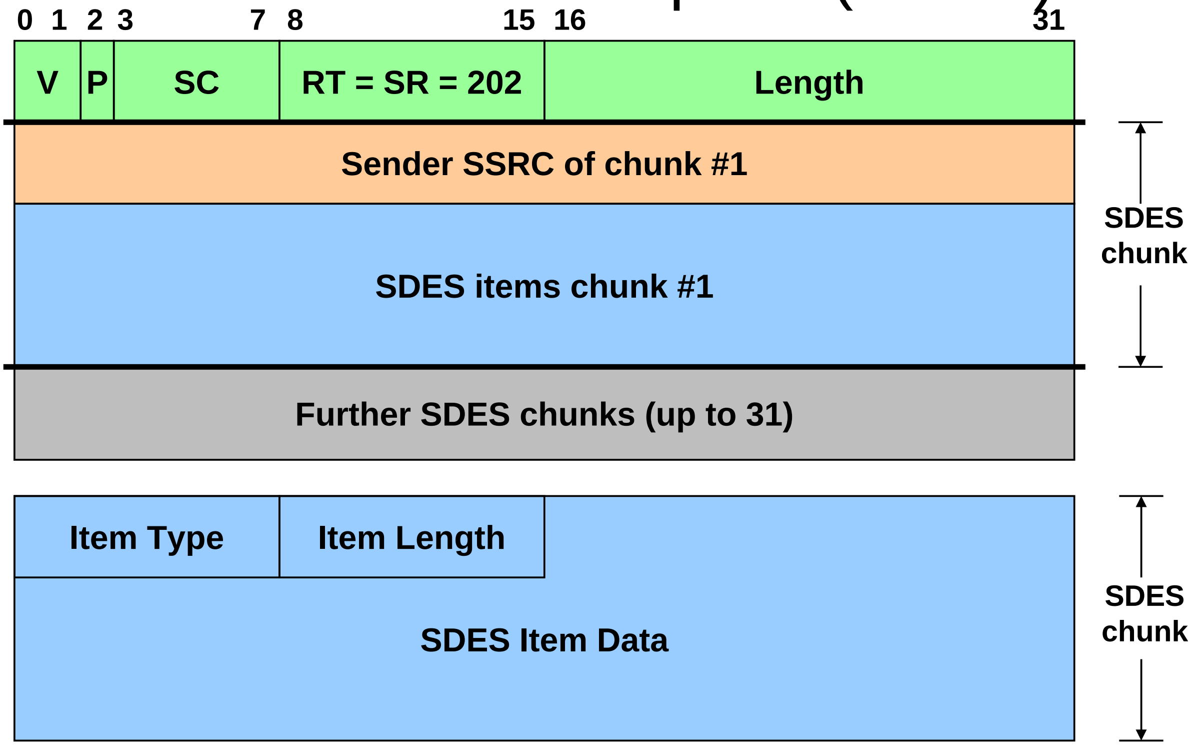


RTCP Source Description (SDES)

- Persistent Identification of an endpoint: Canonical Name
  - CNAME — globally unique identifier (id@host)
  - *Mandatory!*
  - Binding across RTP sessions
  - Identification across changes in the SSRC in an RTP session
- Providing additional information about an endpoint
  - NAME — Name of user (or system)
  - EMAIL — mailto: address
  - PHONE — phone number
  - LOC — location (no format defined)
  - TOOL — (software) client in use
  - NOTE — brief to other participants (e.g. “on the phone”)
  - PRIV — private extensions

HELSINKI UNIVERSITY OF TECHNOLOGY
NETWORKING LABORATORY

RTCP Source Description (SDES)



Other RTCP Packets

- BYE: Announce that an entity will be leaving a session
  - Optional: provide a reason phrase

0 1	2	3 7	8 15	16 31
V	P	SC	RT = BYE = 203	Length
SSRC/CSRC				
Length		Reason for leaving		

- APP: Application-specific extensions

0 1	2	3 7	8 15	16 31
V	P	subtype	RT = APP = 204	Length
SSRC/CSRC				
Name (identifier relative to the application)				
Application-specific contents				

Extended RTCP Reporting (XR)

- Provide more detailed feedback (and feed forward)
  - Infer network characteristics (point-to-point and multicast)
  - Provide detailed voice quality information
- Incorporate many statistics in RTCP packets
  - Lost and duplicate packets
  - Exact packet receipt times
  - Receiver reference time and reception information
    - for RTT measurements
  - Statistics summary
  - VoIP metrics: Burst, gaps, delay, ...
- Detailed reports may get large: thinning reports
  - Report only on every 2^T-th packet (T = 0, …, 15)
  - Indicate the thinning factor T in the packet

RTCP XR

- General report header

0 1	2	3 7	8 15	16 31
V	**P**	**reserved**	**RT = XR = 207**	**Length**
SSRC				
Report Blocks				

- Specific report blocks

Block Type	**Type-specific**	**Length**
Type-specific block contents		

HELSINKI UNIVERSITY OF TECHNOLOGY
NETWORKING LABORATORY

RTCP XR: Detailed Packet Reporting (1)

- Report (individual) lost and duplicate packets
  - Runlength encoding or bit maps of sequences (“chunks”)

0 1 2 3 7	8	15	16 31
BT={1,2}	rsvd.	T	Length
SSRC of source reported			
Start sequence #			End sequence #
Chunk #1			Chunk #2
…			
Chunk #n-1			Chunk #n

- Run length: | 0 | R | # packets lost (R=0) or received (R=1) |
- Bit vector: | 1 | Bit vector (0 = lost, 1 = received packet) |
- Null chunk: 0x0000

RTCP XR: Detailed Packet Reporting (2)

- Record individual packet reception times
  - Ideally obtained as close to the інcoming interface as possible
- Middle 32 bits of the NTP timestamp

0 1 2 3 7	8 15		16 31
BT=3	rsvd.	T	Length
SSRC of source reported			
Start sequence #		End sequence #	
Reception time of packet #start			
Reception time of packet #(start+1) % 65536			
…			
Reception time of packet #(end-2) % 65536			
Reception time of packet #(end-1) % 65536			

HELSINKI UNIVERSITY OF TECHNOLOGY
NETWORKING LABORATORY

RTCP XR: Receiver Side RTT Calculation

- Operation similar to RTCP SR+RR mechanism
- Receivers report sending and selective reception timestamps, too

Receiver Reference Time Report

0 1 2 3 7	8 15	16 31
BT=4	reserved	Length
SSRC of source reported		
NTP timestamp (most significant word)		
NTP timestamp (least significant word)		

Delay since Last RR Report

BT=5	reserved	Length
SSRC #1		
Last RR #1		
Delay since Last RR #1		
…		

RTCP XR: Statistic Summary + VoIP Metrics

- Detailed report on reception statistics for a certain packet interval
  - BT=6
  - Lost, duplicate packets
  - Min, max, mean jitter + standard deviation
- VoIP Metrics (BT=7)
  - Lost packets (network) + discarded packets (local jitter buffer = late packets)
  - Identification of (loss/discard) bursts and (loss/discard) gaps
  - Burst: first, …, last lost packet in a sequence with loss rate > threshold (Gmin)
  - Gap: Runs of packets which are not in a burst
  - Gap + Burst duration (ms) and respective packet loss rate

1111111111011111111111111	0001010110011111101100	1111111111110111111101
Gap	Burst	Gap

RTCP XR: VoIP Metrics

- Delays
  - RTT delay
  - End system delay (estimated)
- Signal information
  - Signal + noise level
- Call quality
  - R factor, extended R factor + MOS listening, conversational
- Configuration parameters
  - Gmin, packet loss concealment, jitter buffer operation (adaptiveness)
- Jitter buffer parameters
  - Delay, maximum delay (observed), absolute maximum delay (buffer size)

RTCP Operation

RTCP Transmission Interval

- Must scale with the number of group members
  - Must not take up too much network capacity (rate-limited!)
- Overall "RTP session bandwidth"
  - Includes UDP and IP header overhead
  - Provided by the application (i.e. not measured dynamically)
- Default: 5% of the session bandwidth for RTCP
  - Takes role (sender or receiver) into account
  - Up to 25% of session members are senders
    - 3.75% for receivers, 1.25% for senders
  - More than 25% of session members are senders
    - Share data rate proportionally
- May be modified by profiles
  - Parameters S and R to indicate relative share for senders/receivers
- Scalable RTCP transmission interval
  - Based upon the group size, RTCP data rate, average RTCP packet size

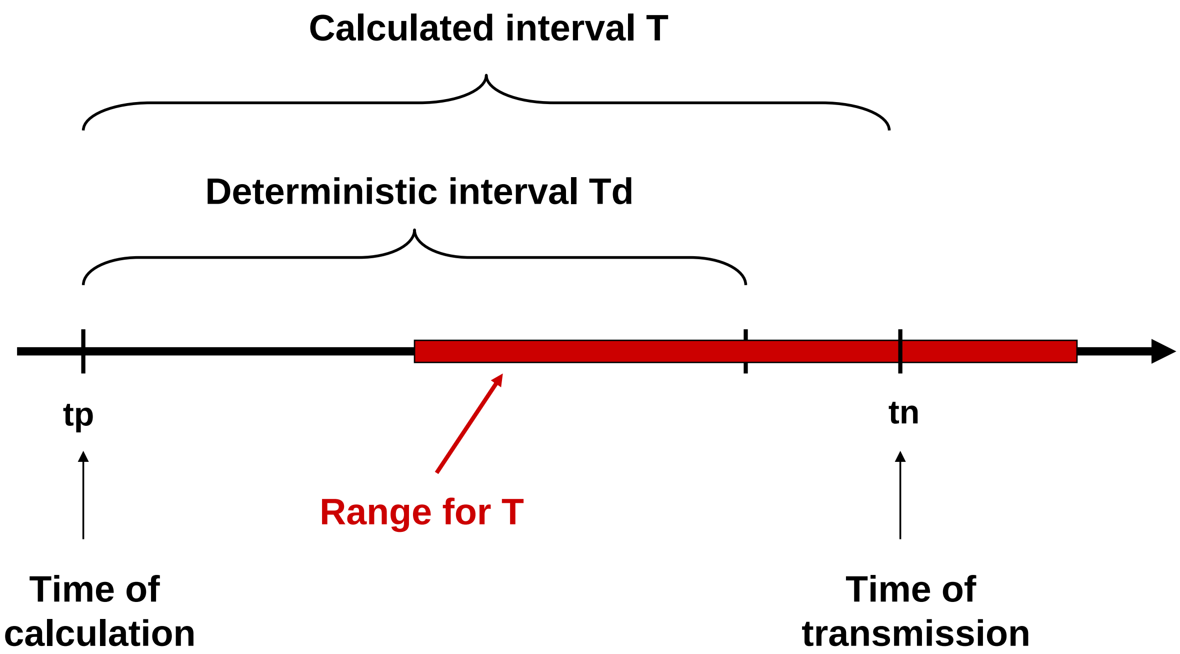
RTCP Variables for Bandwidth Calculation

- Data rate
  - Session bandwidth
  - R, S: Receiver, sender bandwidth share
  - Average RTCP packet size (moving average)
- Time
  - Tp last time an RTCP packet was sent
  - tc current time
  - tn next scheduled transmission of an RTCP packet
- Membership
  - pmembers # members when tn was last computed
  - members current # members
  - senders # senders in the session
  - n relevant # of members (depending on role, etc.)
- Intervals
  - Td Deterministic calculated interval
  - T Calculated interval
  - Tmin minimal interval between RTCP packets

Basic Operation

- Determine role (sender or receiver)
  - Derive n as # of relevant members for calculation
  - Derive relevant bandwidth share
- C = average RTCP size / relevant bandwidth share
- Td = max (Tmin, n*C)
- T = Random [0.5 – 1.5] * Td

Basic RTCP Interval Calculation



Timer Reconsideration

- The group size may change between tp and tn
- Particularly during startup and shutdown phase
  - Many users may join / leave during a short period of time
- Many joining parties: risk of RTCP implosion
- Algorithm for joining members
  - Validate the group size at time tn before transmission
  - Recalculate T as above
  - If tp + T <= tc transmit RTCP packet and update variables
  - If tp + T > tc set tn = tp + T and set timer to expire at tn
- Algorithm for leaving members
  - Adjust tp, tn according to the observed membership change
    - Factor: members / pmembers
  - Run every time a member leaves or times out

Extended Operation

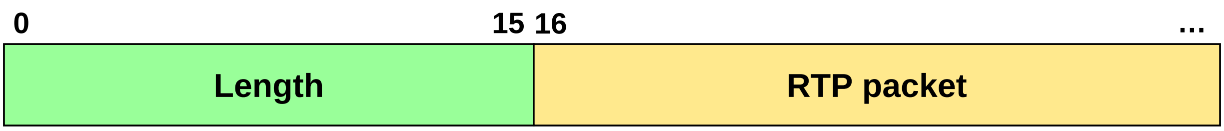
- Determine role (sender or receiver)
  - Derive n as # of relevant members for calculation
  - Derive relevant bandwidth share
- C = average RTCP size / relevant bandwidth share
- Td = max (Tmin, n*C)
- T = Random [0.5 – 1.5] * Td
- T = T / e^-1.5 (T = T / 1.21828)
  - Correction factor for timer reconsideration

RTP/RTCP Transport and Multiplexing (1)

- RTP over UDP
  - Session Identification: a pair of destination transport addresses
  - Multicasting: Common IP multicast address as destination for all RTP entities
  - Unicasting: two independent sessions
  - Usual operation: 1 transport address RTP + 1 transport address RTCP
    - Typically the same IP address + 2 port numbers to differentiate
  - Original idea: RTP port is n (even), RTCP port is n+1 (odd)
  - Issues: dynamic port assignment, NATs: ports may now be arbitrary
- Further optimization (currently discussed in the IETF)
  - Use a single port for both RTP and RTCP
  - Motivation: NATs and firewalls
    - Need to open just one pin hole
    - Need to maintain just one port binding
  - Payload type name space allows for easy differentiation
  - Raises architectural issues though

RTP/RTCP Transport and Multiplexing (2)

- RTP over connection-oriented transport: TCP (or SCTP)
  - TCP is obviously suboptimal for real-time traffic
  - Yet: many media streaming applications use TCP (also w/o RTP)
  - Works if delay is acceptable (one-way streaming)
    - Sufficient data can be buffered to account for later retransmissions
    - If necessary, media playback is paused
  - Last resort if UDP does not work (e.g., due to firewalls)
    - In many cases, connectivity is just good enough
- Framing of RTP packets in a TCP connection



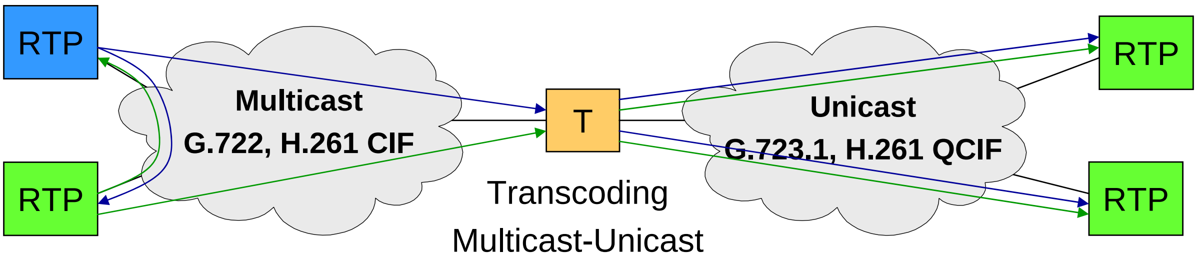
- Need to set up and tear down TCP connections for media
  - UDP is easy: just send
  - TCP: Who initiates, who accepts?
  - How to deal with accidental disconnection?

RTP and Congestion Control

- TCP-friendly RTP profile (RTP/AVPFCC) [in flux]
  - Adaptive transmission behavior compliant to the TCP-friendly rate control
    - Based upon Padhye equation for TCP throughput (RFC 3448)
    - Targeted at unicast sessions only
  - Modified RTP packet header
    - Includes 32 bit sender timestamp
    - Optional 32 bit RTT indicator (only included if RTT has changed)
    - Reduced payload type field: 6 bits
  - RTCP TFRC-FB (feedback) message
    - Reception timestamp of last packet from sender + delay since reception
    - Observed loss event rate (as defined in TFRC)
  - Control loop between sender and receiver: feedback once per RTT
- Possible Alternative: RTP over DCCP (RFC 4340)
  - Make use of congestion control characteristics of underlying transport
  - Congestion control ID 3 (RFC 4342): TFRC

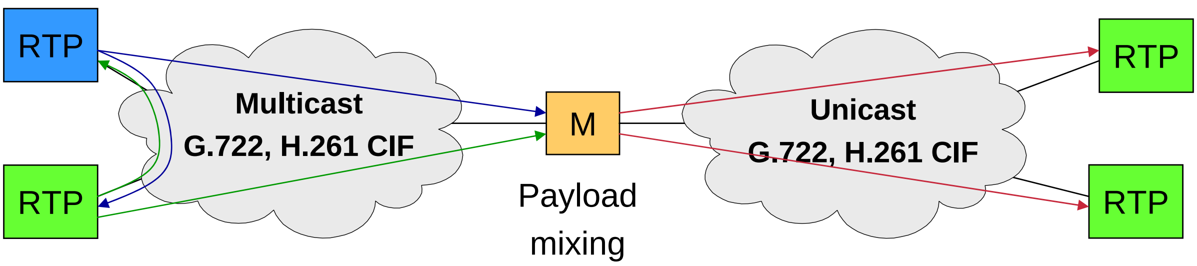
RTP Translator

- Intermediate system in an RTP session
- Operates at the transport level
- Connects two or more RTP clouds
- Leaves SSRC intact
  - Shared global SSRC space per session; end-to-end conflict resolution
- May operate on the payload, the packet size, the transport
  - IPv4 to IPv6 translation typically transparent to RTP



RTP Mixer

- Another intermediate system in an RTP session
- Creates a new media stream from one or more incoming streams
  - With its own SSRC id
  - Indicates input streams (= contributing sources) in CSRC field
  - Performs local dejittering, input synchronization, etc.
- Operates on the payload and may operate on everything else
  - Reduces bandwidth demand towards each receiver
  - Typically found in IP-based conference bridges



RTP Payloads

RTP Payload Types

- 7-bit payload type identifier
  - Some numbers statically assigned
  - Dynamic payload types identifiers for extensions – mapping to be defined outside of RTP (control protocol, e.g. SDP “a=rtpmap:”)

<u>Payload formats defined for many audio/video encodings</u>

- Conferencing profile document RFC 3551
  - Audio: G.711, G.722, G.723.1, G.728, GSM, CD, DVI, …
- In codec-specific RFCs
  - Audio: Redundant Audio, MP-3, ...
  - Video: JPEG, H.261, MPEG-1, MPEG-2, H.263, H.263+, BT.656
  - Others: DTMF, text, SONET, ...
- Generic formats
  - Generic FEC, (multiplexing)

Media Packetization Schemes (1)

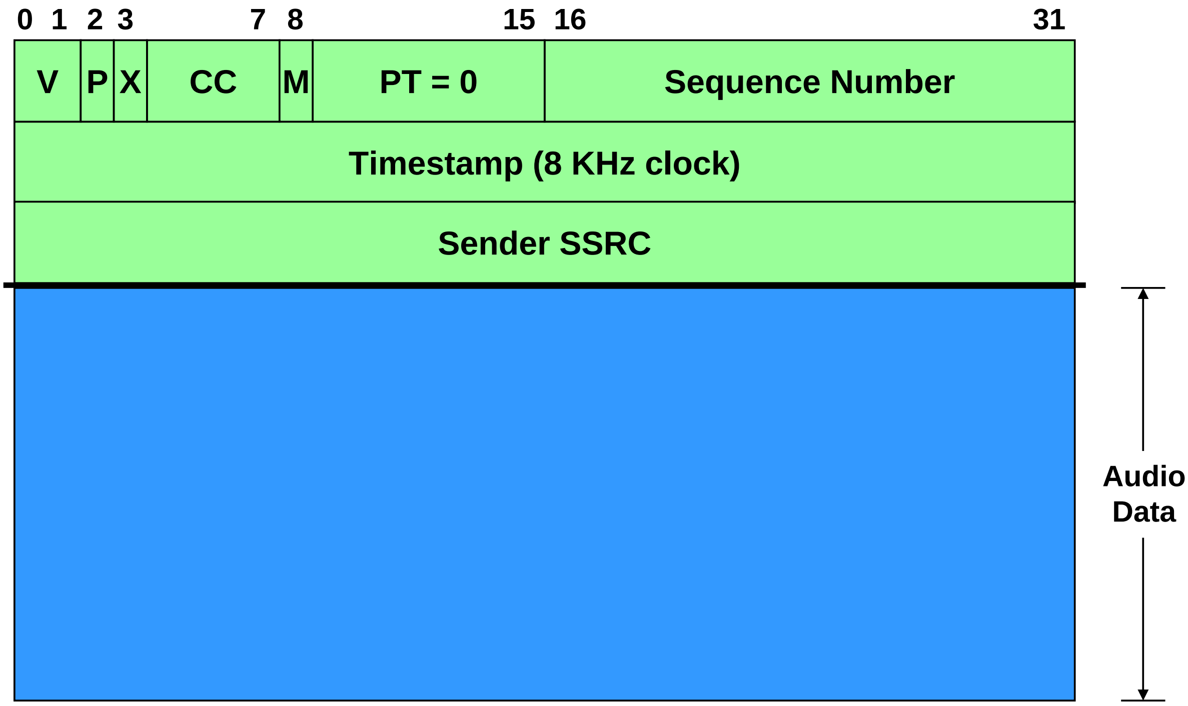
General principle:

- Payload specific additional header (if needed)
- Followed by media data
  - Packetized and formatted in a well-defined way
  - Trivial ones specified in RFC 3551
  - RFC 2029, 2032, 2035, 2038, 2190, 2198, 2250, 2343, 2429, 2431, RFC 2435, 2658, 2733, 2793, 2833, 2862, and many further ones
  - Guidelines for writing packet formats: RFC 2736
- Functionality
  - Enable transmission across a packet network
  - Allow for semantics-based fragmentation
  - Provide additional information to simplify processing and decoding at the recipient
  - Maximize possibility of independent decoding of individual packets

Sample RTP Payload Types

Illustrate a variety of approaches to deal with packet loss in the Internet

Audio over RTP: PCM



HELSINKI UNIVERSITY OF TECHNOLOGY
NETWORKING LABORATORY

Video over RTP: H.261

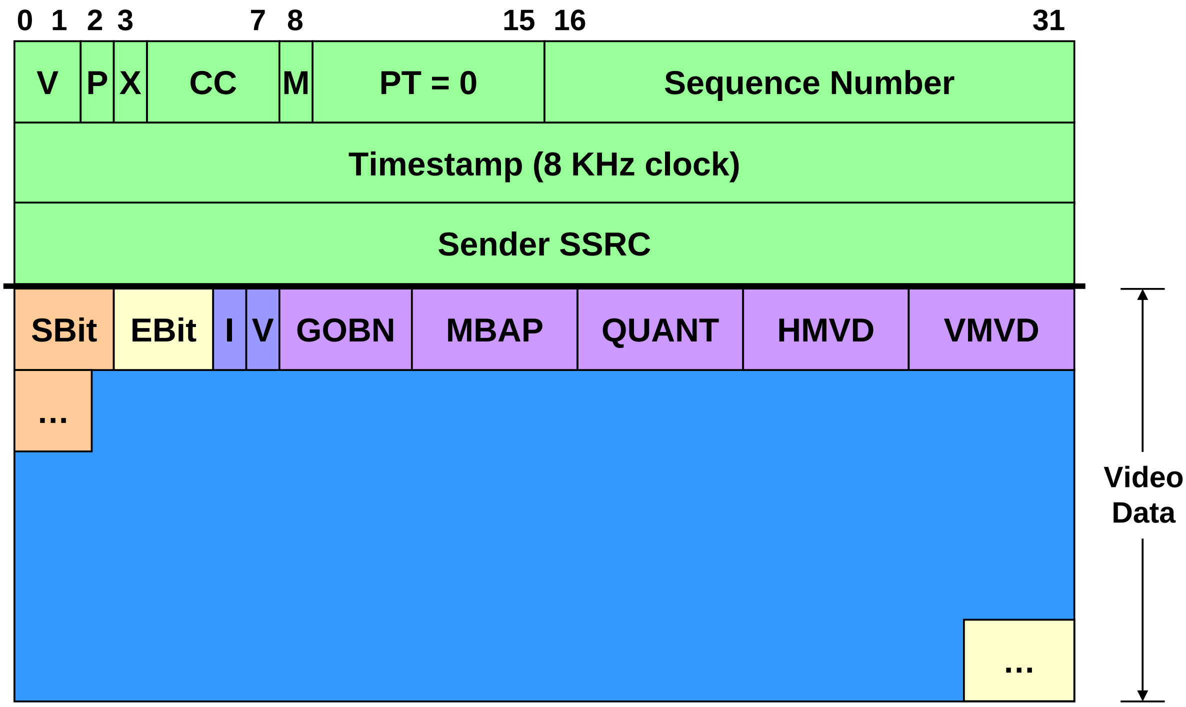
Additional payload-specific header preceeds payload

- To avoid expensive bit shifting operations
  - Indicate # invalid bits in first (SBit) and last (EBit) octet of payload
- Indicate Intra encoding (I bit)
- Indicate the presence of motion vector data (V bit)
- Carry further H.261 header information to enable decoding in the presence of packet losses

Further mechanisms for video conferencing

- FIR: Full Intra Request
  - Ask sender to send a full intra encoded picture
- NACK: Negative Acknowledgement
  - Indicate specific packet loss to sender

Video over RTP: H.261 (2)



HELSINKI UNIVERSITY OF TECHNOLOGY
NETWORKING LABORATORY

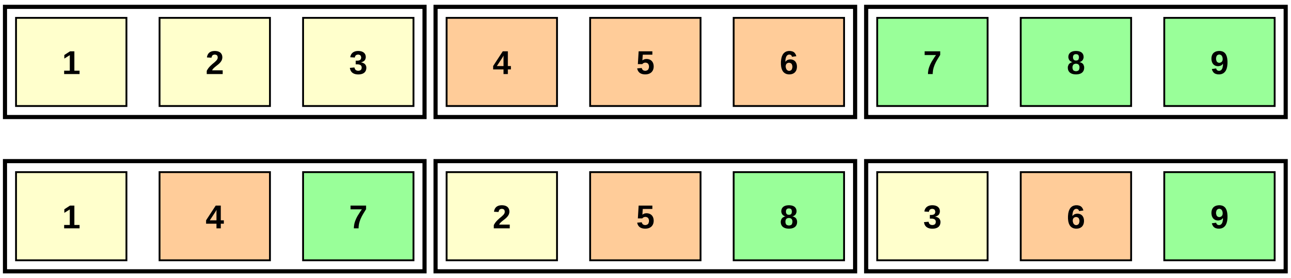
Media Packetization Schemes (2)

Error-resilience for real-time media

- Input: Observation on packet loss characteristics
- Generic mechanisms (RFC 2354)
  - Retransmissions
    - in special cases only (e.g. with no interactivity!)
  - Interleaving
  - Forward Error Correction (FEC)
    - media-dependent vs. media-independent
    - Generic FEC: RFC 2733
- Feedback loops for senders
  - based upon generic and specific RTCP messages
  - adapt transmission rate, coding scheme, error control, ...

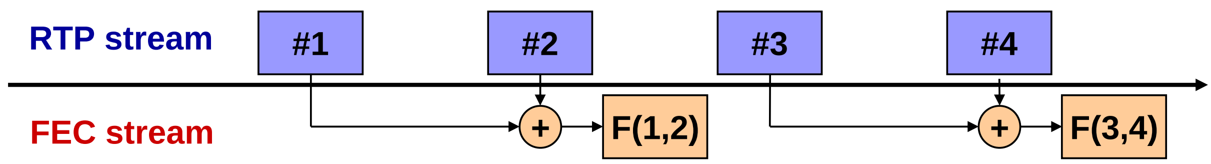
RTP Interleaving

- Distribute packets or packet contents for transmission
  - Avoid consecutive packet erasures in case of (burst) losses
  - Avoid loss of large consecutive data portions in case of single packet losses
- Motivations
  - Human perception tolerates individual losses better (with error concealment)
  - Make simple FEC schemes work better with burst losses (e.g. XOR)
- Drawback
  - Re-ordering causes additional delay

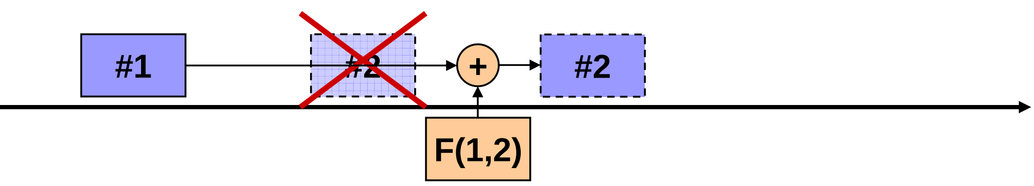


RTP FEC (RFC 2733)

- Forward Error Correction scheme for RTP packets
  - Media-independent, flexible FEC (that can be enhanced)
- Simple XOR-based (parity) FEC
  - P_fec = P1 XOR P2 XOR P3 XOR … XOR Pn
    - Allows reconstruction of any single missing packets of P1, …, Pn, P_fec
- RTP FEC stream transmitted independently of RTP stream
  - Separate transport address (IP address, port number)
  - Different SSRC

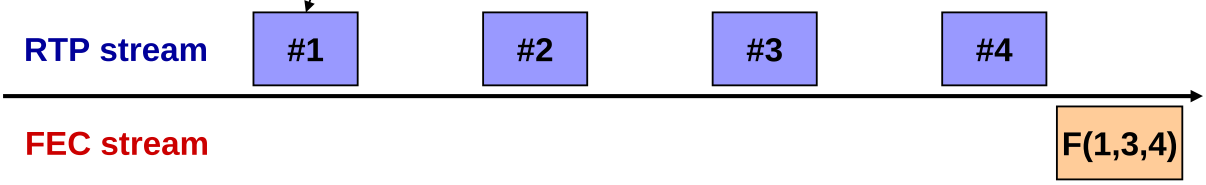


- Recovery



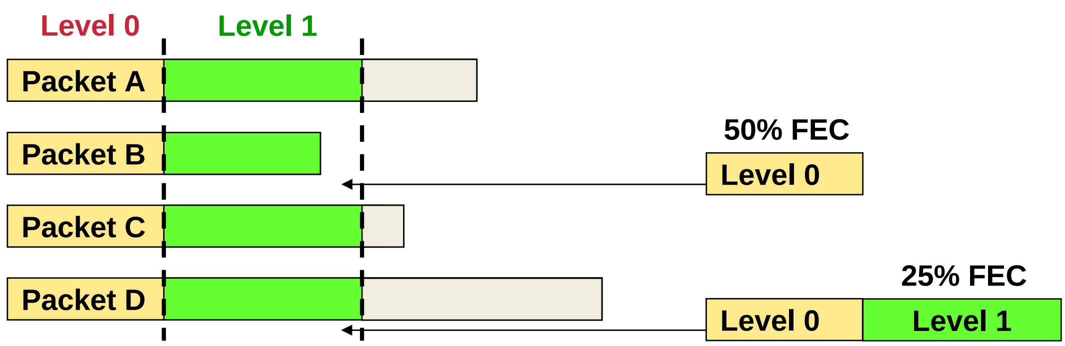
RTP Parity FEC Packet format

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|V=2|P|   FMT   |       PT      |             length            |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                  SSRC of packet sender                        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                  SSRC of media source                         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|           SN base    = 1      |        length recovery        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|E| PT recovery |       mask    = 10110000 00000000 00000000    |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                          TS recovery                          |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
:                                                               :
:       XOR of Payloads indicated by SN Base and mask           :
:                                                               :
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```



Unequal Error Protection

- Observation: not all parts of a packet are equally important
  - Beginning of packet contains headers/parameters, more relevant contents
  - Holds for both audio and video
- Uneven Level Protection (ULP)
  - Create independent parity packets for different parts of packets
  - Allows for selectively more overhead for the more important parts

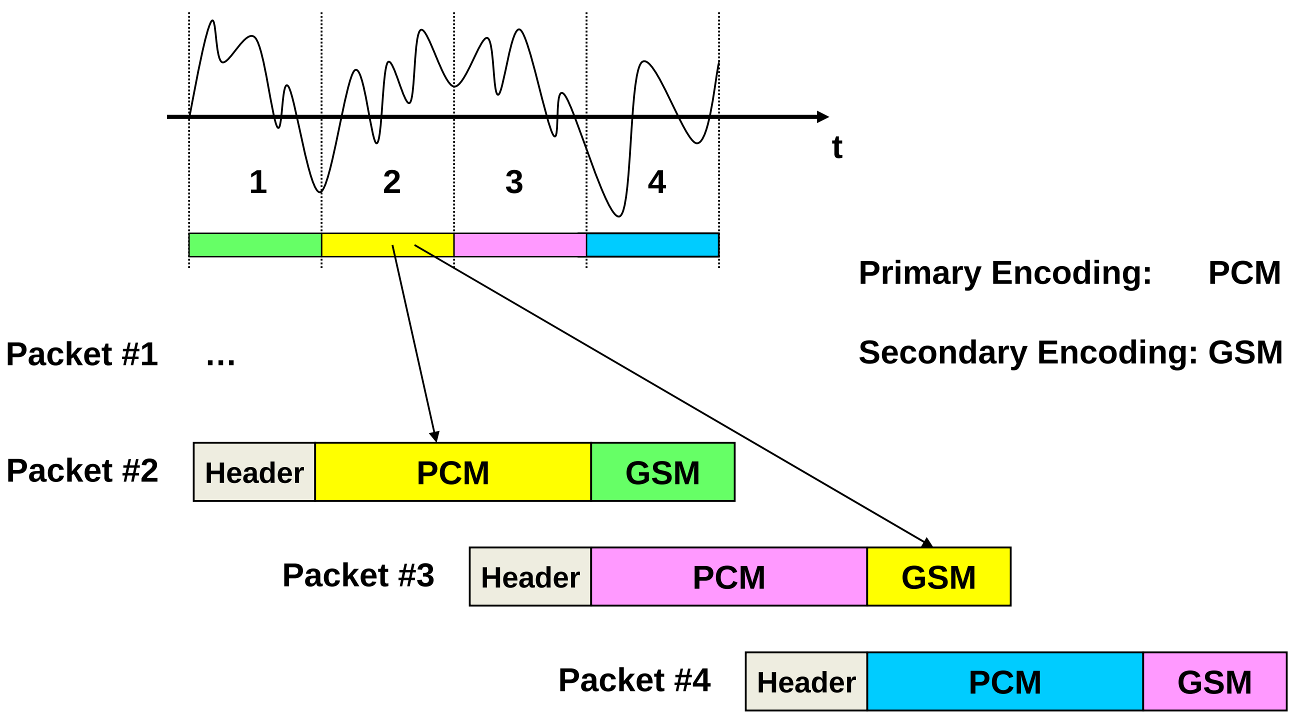


- Related thoughts: partial checksums
  - Live with bit errors in the less important parts (rather than dropping a packet)

Audio Redundancy Coding (1)

- Audio Packets are small!
  - have to be because of interactivity
    - avoid large packetization delay
  - packet loss primarily depends on packet rate
    - rather than packet size
- Payloads for multiple time slots in one packet
  - send redundant information in packet n
    to reconstruct packets k, ..., n-1 in packet n
  - redundant information typically sent at lower quality
  - details defined in RFC 2198
  - uses dynamic payload type
- Format specification, e.g. using SDP
  - m=audio 20002 RTP/AVP 96 0 0 0
  - a =rtpmap:96 red/8000/1

Audio Redundancy Coding (2)



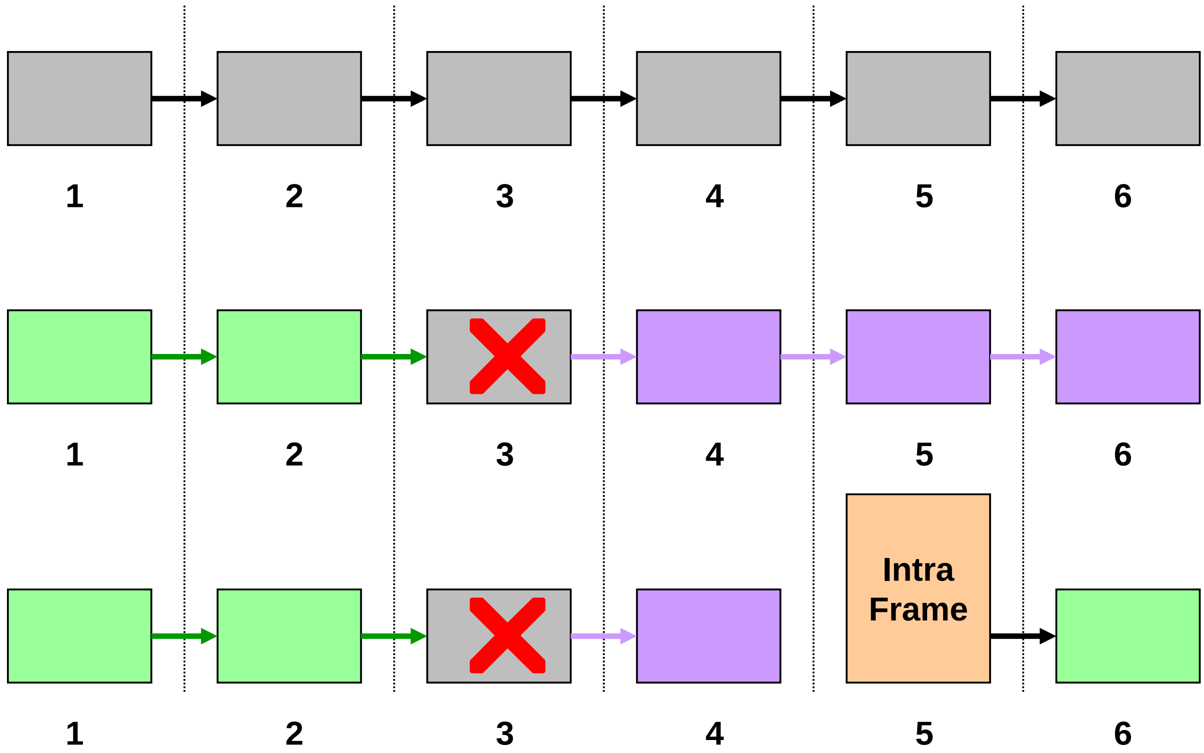
Audio Redundancy Coding (3)

	0					8	16	24	31	
	V	P	X	0 0 0 0	M	PT=96	Sequence #			
	Timestamp = 6400									
	SSRC									
Redundant 2	1	PT=0					Timestamp offset = 320		Len = 160	Header
Redundant 1	1	PT=0					Timestamp offset = 160		Len = 160	Header
Primary	0	PT=0								Header
	Data Block "Redundant 2" (160 Bytes)									
	Data Block "Redundant 1" (160 Bytes)									
	Data Block "Primary" (160 Bytes)									

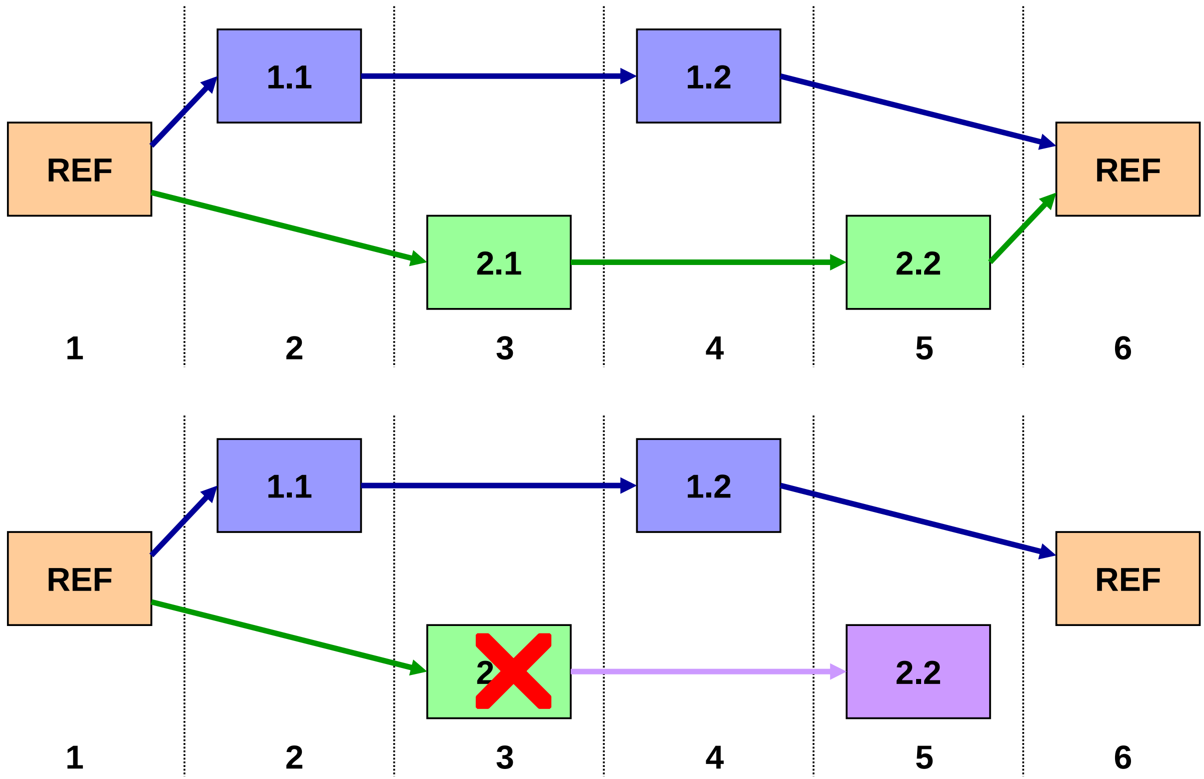
Video Redundancy Coding (1)

- Video redundancy coding
  - For H.263+ video streams
  - Transmit several interleaved sequences of predicted frames (threads) instead of one
    - improves error resilience against packet loss
- Principle
  - create several (n) independently decodable streams
  - achieved by choosing different reference pictures
  - decode only streams with no packet losses
    - reduces temporal resolution by 1/n-th per affected stream
  - bit rate penalty due to larger deltas between frames
  - RFC 2429, revised version in progress

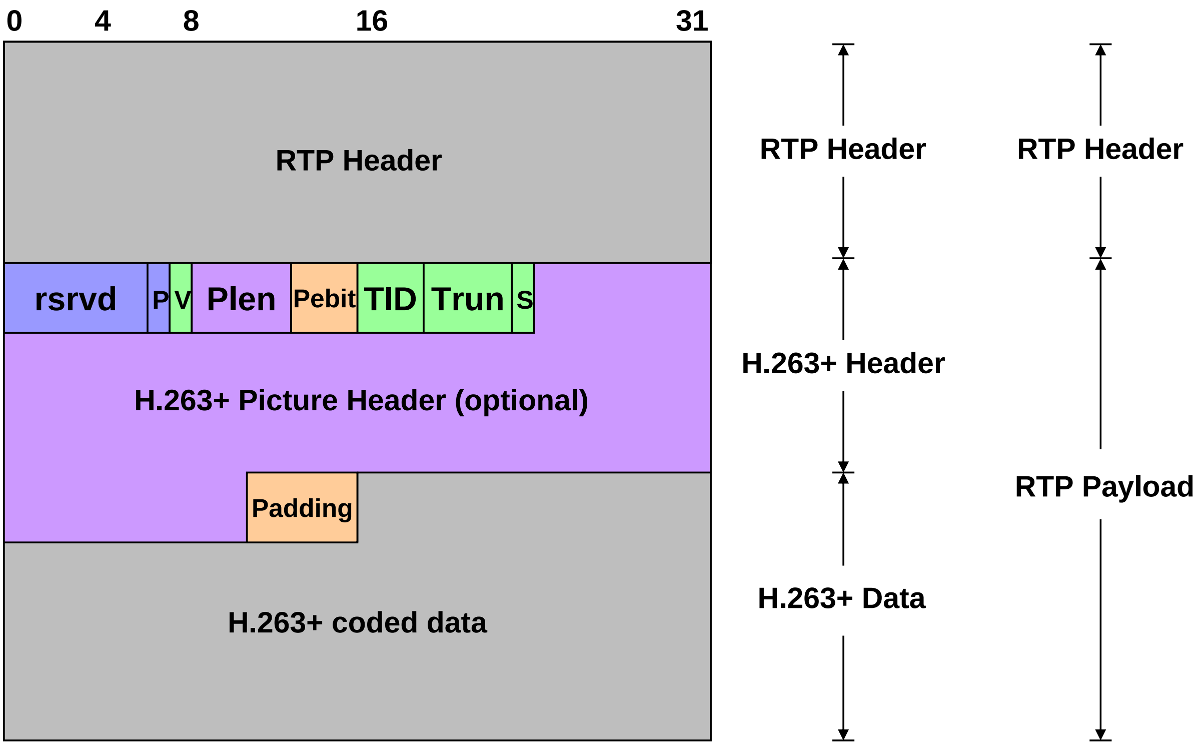
Video Redundancy Coding (2)



Video Redundancy Coding (3)



Video Redundancy Coding (4)



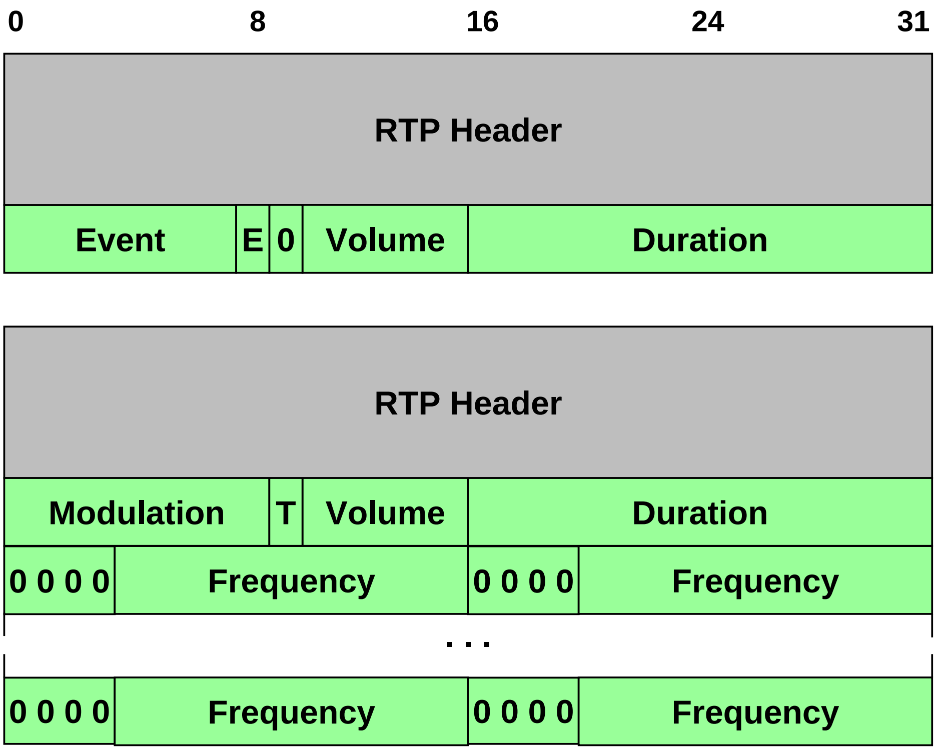
DTMF over RTP (1)

- DTMF digits, telephony tones, and telephony signals
  - two payload formats
  - 8 kHz clock by default
  - audio redundancy coding for reliability
- Format 1: reference pre-defined events
  - 0 - 9 * # A - D (Hook)Flash [17]
  - modem and fax tones [18]
  - telephony signals and line events [43]
    - dial tones, busy, ringing, congestion, on/off hook, …
  - trunk events [44]
  - specified through identifier (8-bit value), volume, duration
- Format 2: specify tones by frequency
  - one, two, or three frequencies
  - addition, modulation
  - on/off periods, duration
  - specified through modulation, n x frequency, volume

DTMF over RTP (2)

Packet Format 1: Events

Packet Format 2: Tones



HELSINKI UNIVERSITY OF TECHNOLOGY
NETWORKING LABORATORY

RTCP Payload Type Overview (1)

- RFC 3551 Collection of simple packetization formats (formerly RFC 1890)
- RFC 2029 Sun CellB Video encoding
- RFC 2032,4587 H.261 video
- RFC 2435 JPEG video (was RFC 2035)
- RFC 2250 MPEG-1/MPEG-2 video (was RFC 2038)
- RFC 2190 H.263 video (historic)
- RFC 2343 Bundled MPEG
- RFC 2429 H.263+ video & video redundancy support
- RFC 2431 BT.656 video
- RFC 2658 PureVoice audio
- RFC 2793,4103 Text conversation
- RFC 2833 DTMF, telephony tones, and telephony signals
- RFC 2862 Real-time Pointers
- RFC 3016 MPEG-4 Audio/visual streams
- RFC 3047 G.722.1 audio
- RFC 3119 Loss-tolerant format for MP3
- RFC 3189 DV video
- RFC 3190 12-bit DAT and 20-/24-bit linear audio

RTCP Payload Type Overview (2)

- RFC 3267,4352 Adaptive Multirate (AMR, AMR-WB+) audio
- RFC 3389 Comfort noise
- RFC 3497 SMPTE 292M video
- RFC 3557 ETSI Distributed speech recognition (ES 201 108)
- RFC 3558 Enhanced variable rate codecs and selectable mode vocoders
- RFC 3640 MPEG-4 elementary streams
- RFC 3952 Low Bit Rate Codec (iLBC) Speech
- RFC 3984 H.264 Video
- RFC 4040 64 kbit/s Transparent Call
- RFC 4060 Distributed speech recognition encoding (ES 202 050/211/212)
- RFC 4175,4421 Uncompressed Video
- RFC 4184,4598 AC-3 Audio, Enhanced AC-3
- RFC 4298 BroadVoice Speech codec
- RFC 4348,4424 Variable Rate Multimodal Wideband Audio (VMR-WB)
- RFC 4351 Text conversation interleaved with audio stream
- RFC 4396 3GPP Timed Text
- RFC 4425 Video Codec 1 (VC-1)
- RFC 4588 Retransmission payload format

Many more to come…

RTP Extensions

- Timely feedback from receivers to senders
- RTP Retransmissions
- Support for Source-specific Multicast (SSM)

RTCP Feedback Issues

- Senders provide regular information about media stream
  - Seems ok
- Receivers transmit RTCP at somewhat regular intervals
- RTCP RRs provide long-term statistics on reception quality
- Senders can adapt transmission strategy to receiver observations
  - Different codecs, data rate, etc.
- BUT: No short-term feedback possible
  - Error repair or mitigation impossible
  - Not suitable for congestion control
- Problem: Value of receiver feedback decreases over time
  - Repair more expensive at later times
  - Artifacts become noticeable to the user

Approach: RTCP-based Feedback

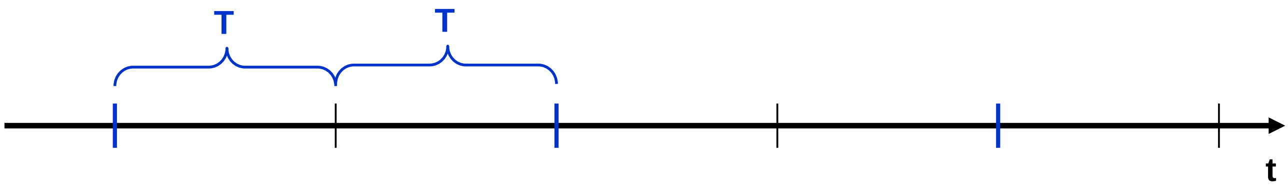
- New Profile for RTP: AVPF

Idea:

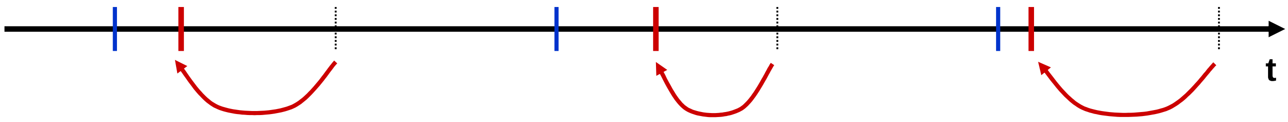
- Packet losses are usually rare
- Provide statistical chance of virtually immediate feedback from receiver(s) to sender
- Keep the basic RTCP properties
- Eliminate Tmin
- Work most efficiently with unicast
- Also scale to moderate group sizes

Overview

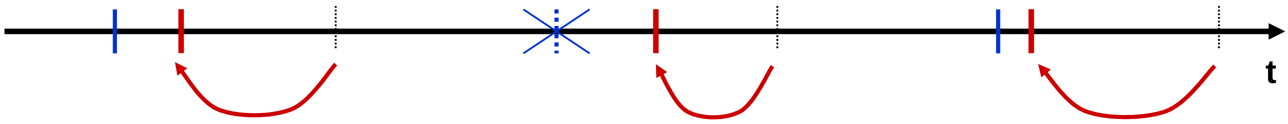
Regular RTCP operation (depicted w/o randomization, i.e. T = Td)



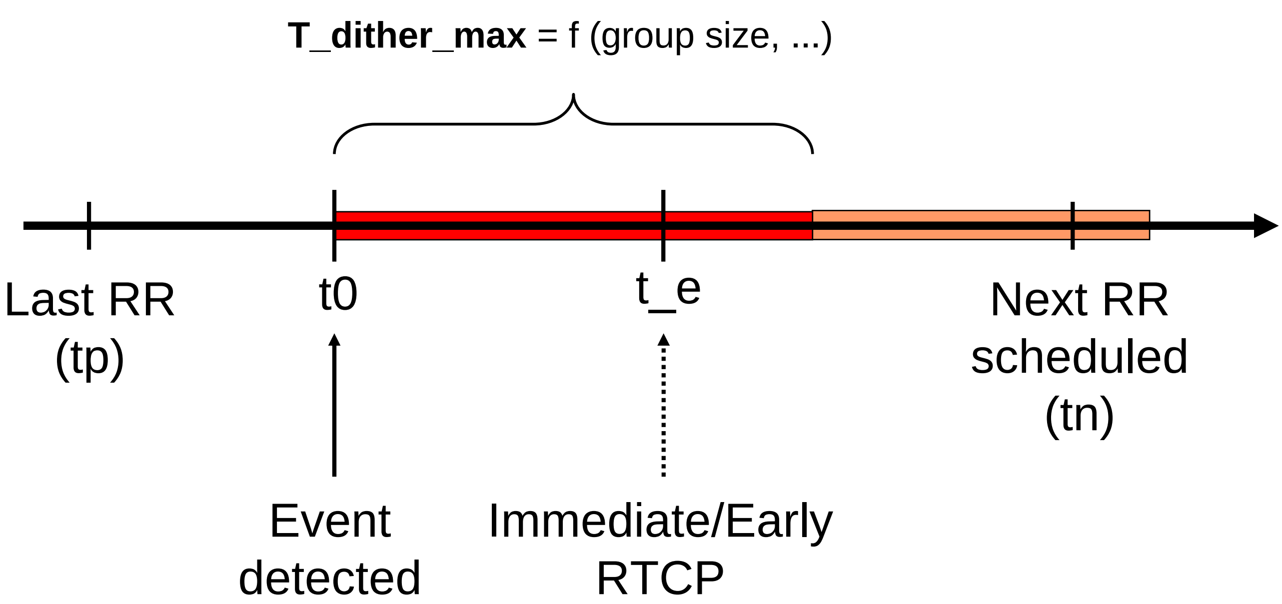
Allow (at most every other) RTCP packet to be sent earlier



Allow to reduce the number of regular RTCP packets (w/o affecting RTCP rate)



RTCP Feedback Timing



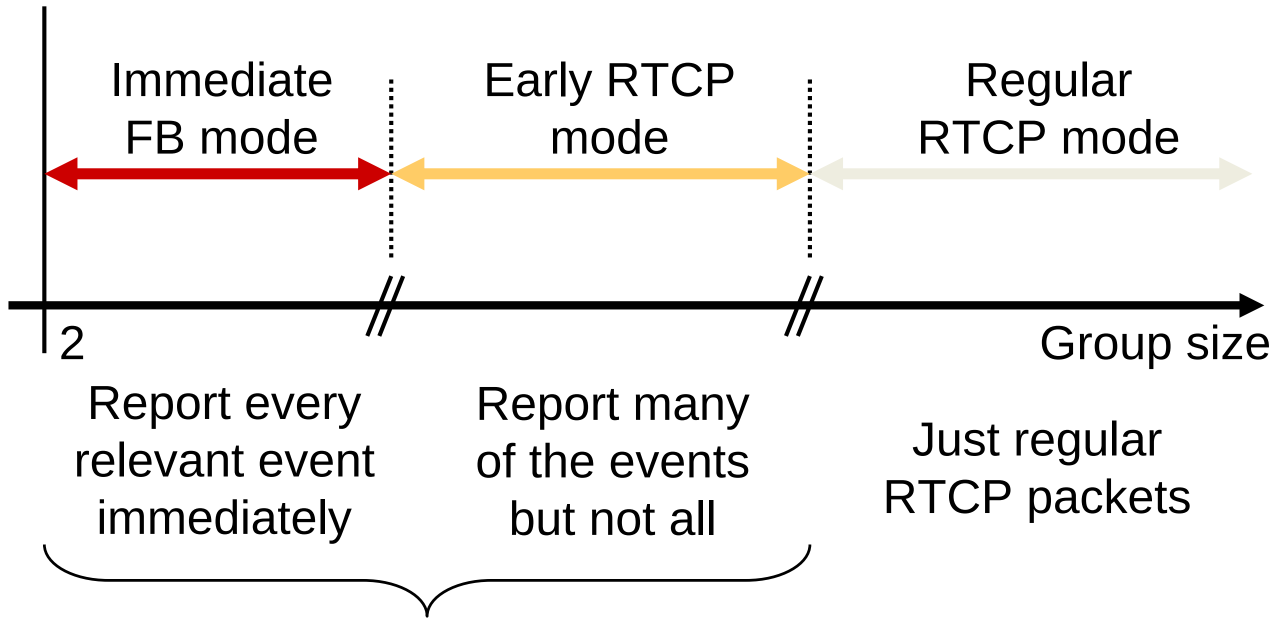
Delay calculation

$$\text{T\_dither\_max} = \begin{cases} 0 & \text{if grp size} = 2 \\ l * T & \text{otherwise} \end{cases}$$

Simulated guess: $l = 0.5$

Better approach: use RTT measurements!
But those are only available to senders…
Mixed operation (using Td and RTT) will not work.

Modes of Operation



Send feedback + regular RTCP packets

RTCP Types of Feedback

- ACK Mode
  - Positive acknowledgements for received packets
  - Restricted to point-to-point operation
- NACK Mode
  - Negative acknowledgments e.g. for missing packets or other events
  - Scalable with suppression technique
- Other types of feedback conceivable
- Transport layer feedback packets (Generic NACK)
  - Identifies missing or received packets
- Payload-specific feedback packets
  - Specific to certain codecs (e.g. video)
  - Picture / frame loss indication, reference picture selection
- Application feedback packets

RTCP Feedback Packet Format

```
  0                   1                   2                   3
  0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |V=2|P|   FMT   |       PT      |          length               |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |                  SSRC of packet sender                        |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |                  SSRC of media source                         |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 :            Feedback Control Information (FCI)                 :
 :                                                               :
```

Example: Generic NACK Packet

```
  0                   1                   2                   3
  0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |            PID                |             BLP               |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

RTCP Feedback Packet Format (2)

Example: Slice lost indication

```
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|            First        |  Number                   | PictureID |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Example: Reference Picture Selection

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|      PB       |0| Payload Type|  Native RPSI bit string       |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|   defined per codec          ...                |  Padding (0)|
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```


HELSINKI UNIVERSITY OF TECHNOLOGY
NETWORKING LABORATORY

Example for Statistical Feedback

- Applicability of feedback depends on many parameters
  - Group size, RTP & RTCP bandwidth, application requirements

256 kbit/s video stream, 30 frames per second, 1500 bytes MTU
Single sender, > 3 receivers (i.e. 3.75% RTP bandwidth for receivers)
H.263+ with approximately 1 packet per frame
5% packet loss, equally distributed, receiver independence

Statistically yields 3 losses every two seconds per receiver
3.75% * 256 kbit/s = 9.6 kbit/s for all receivers
Assuming 120 bytes (= 960 bits) per RTCP packet: 10 packets / s

If every receiver reports every loss event: 6 – 7 receivers on average

If reporting every other loss event is sufficient: ~14 receivers

Increases further if losses are correlated in some fashion

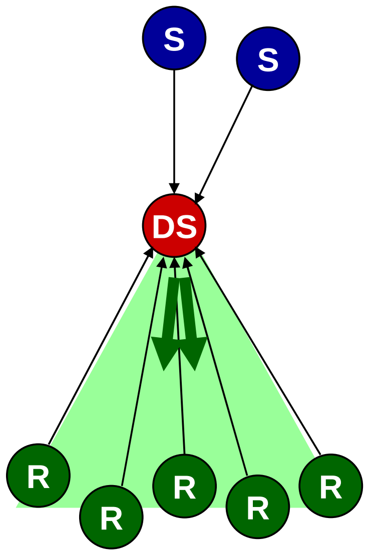
RTP Retransmissions

- Explicit repair mechanism for RTP streams
- Works for applications with acceptable higher latency
  - E.g. media streaming
- Applicable to point-to-point and small group scenarios
- Used with RTCP feedback extensions
- Approach
  - Original RTP stream
  - Augmented by retransmission RTP stream
  - Mapped to different RTP sessions or sender SSRCs
    - Use always different sessions for multicasting
  - Keeps the retransmission scheme backward compatible
  - Does not confuse RTCP statistics
  - Works with all payload types
  - Allows for multiple payload types in a session

RTCP Retransmission Packet Format

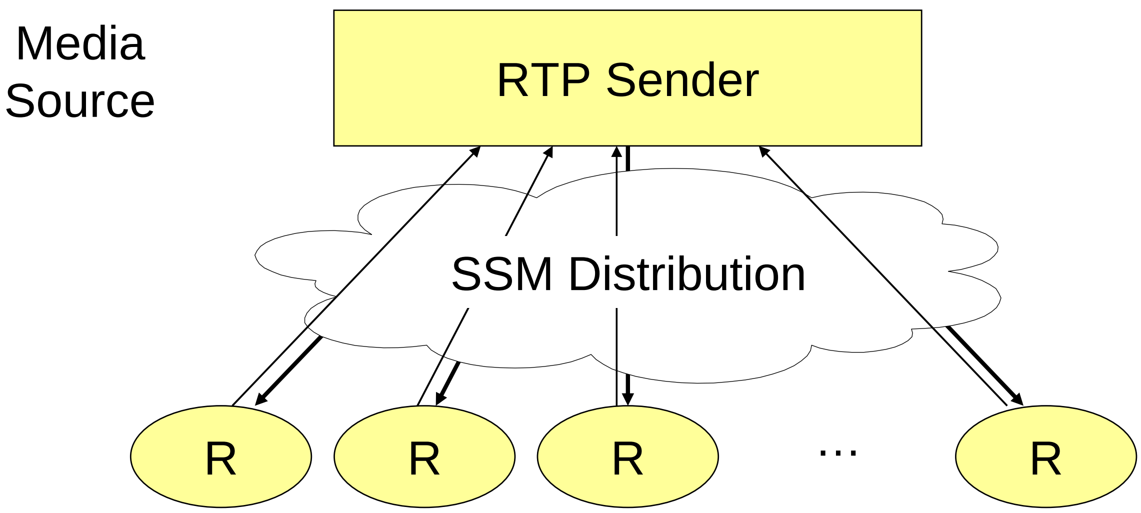
```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|V=2|P|   FMT   |       PT      |          length               |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                  SSRC of packet sender                        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                  SSRC of media source                         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|            OSN                |                               |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+                               |
|                  Original RTP Packet Payload                  |
|                                                               |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

RTCP for SSM



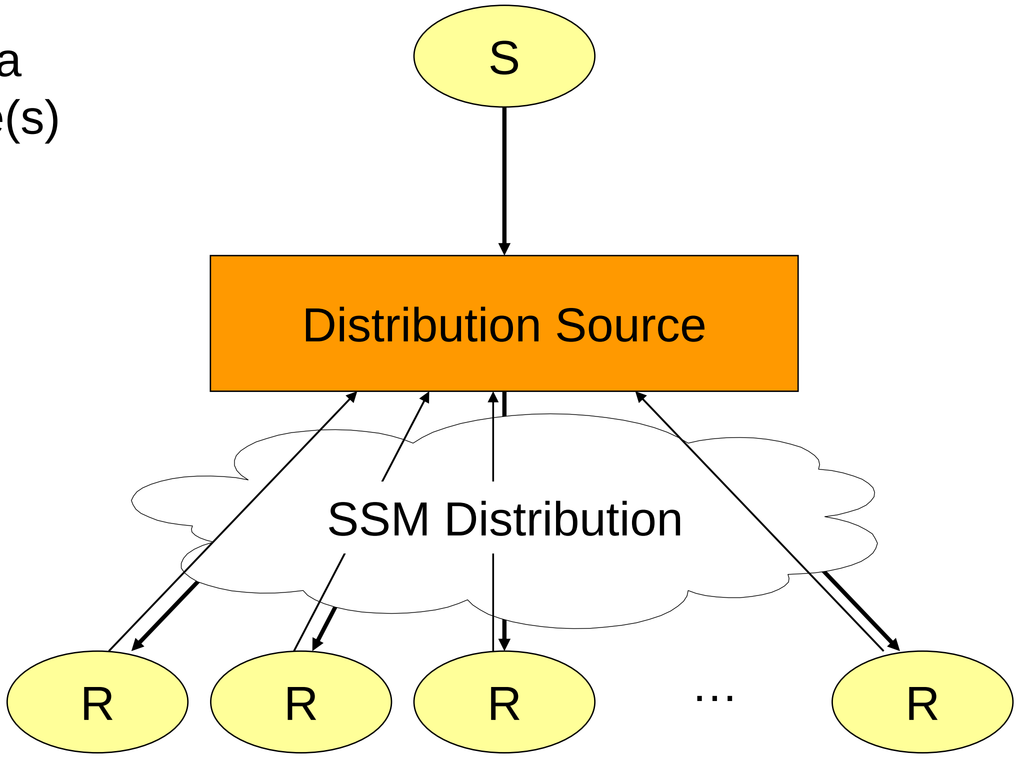
- Multicast connectivity unidirectional
  - From Distribution Source to receivers
  - Opposite direction needs to use unicasting
    - May follow different network path
- Result: no direct communication between receivers
- Adaptations required to make RTCP work
  - Estimate group size
  - Adjust timing of RTCP transmission (adhere to bandwidth limit)
  - Resolve SSRC collisions
- Two basic modes of operation
  - Make distribution source reflect RTCP traffic back to receiver
  - Provide summaries of relevant information along with sender reports

RTCP SSM Overview



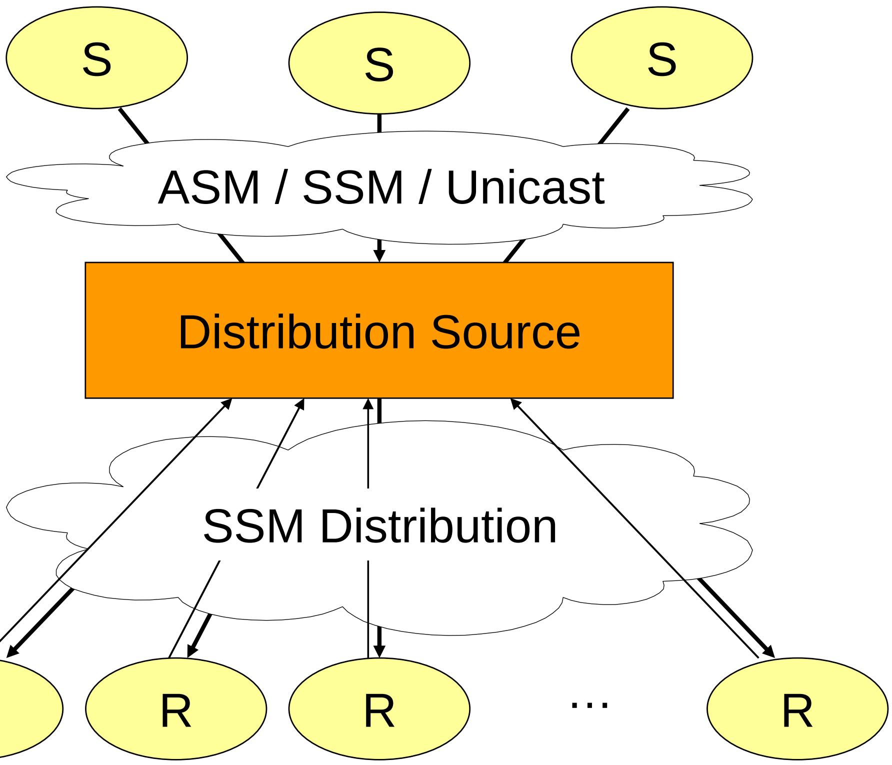
RTCP SSM Overview

Media
Source(s)

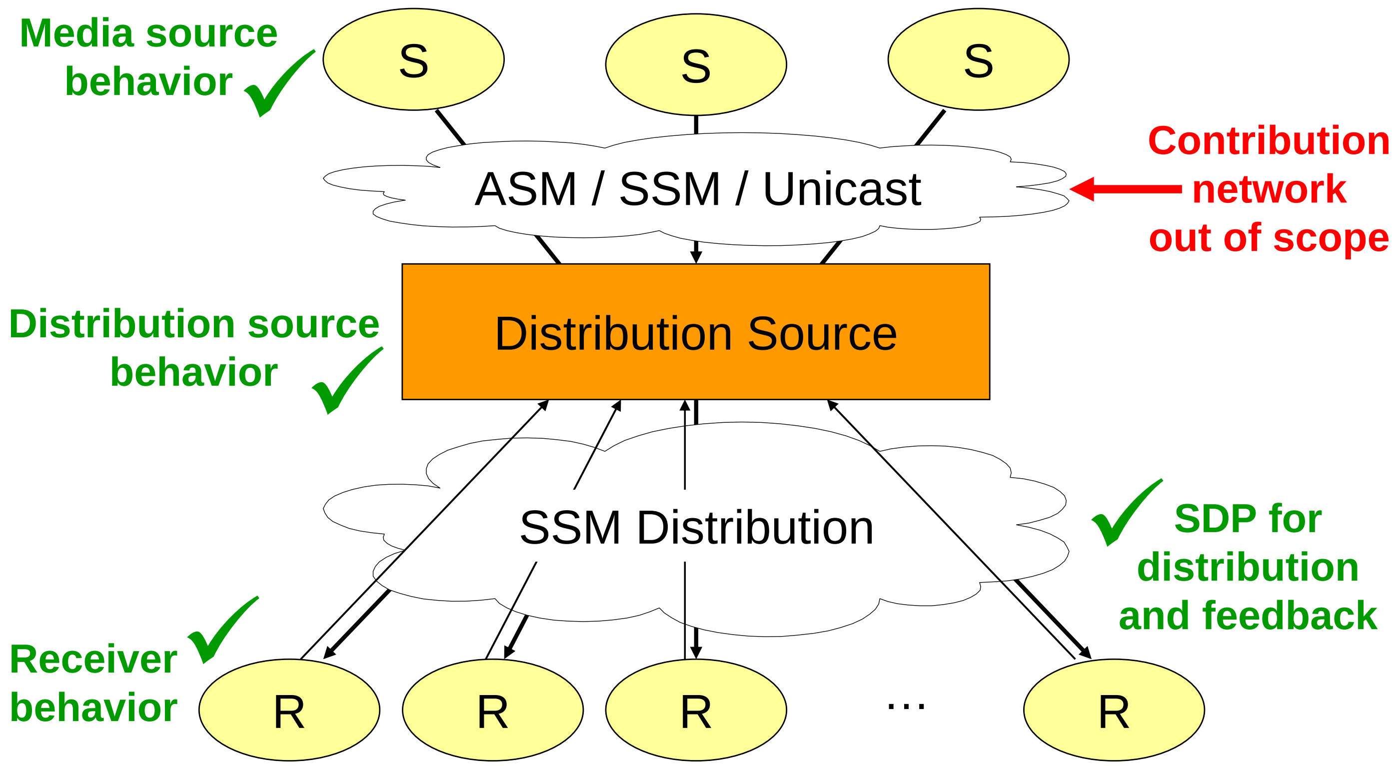


RTCP SSM Overview

Media
Source(s)



RTCP SSM Overview



Simple Feedback Model

- Distribution source reflects packets back to receivers
  - Simple mirroring at the transport / application layer
- Uses the bandwidth share for receivers for distribution
  - Not an issue: non-overlapping paths
- Increases delay for inter-receiver communication
  - Particularly with asymmetric networks
  - May impact e.g. feedback suppression
- Required for all RTCP packets that cannot be summarized
  - Unknown extensions
  - Packets that require knowledge of the originator
- Particularly applies to RTCP APP packets

Feedback Summary Model

- Distribution source collects information from receivers
- Aggregates the information over time
- Distributes representative summaries back to receivers
  - In somewhat regular intervals
  - Saves bandwidth compared to simple reflection
  - Uses (part of) receiver rate in addition to sender rate
  - Acts as another receiver from an RTP/RTCP perspective (own SSRC)
- New RTCP packet: Receiver Summary Information (RSI)
  - Contains distributions for RTCP receiver statistics
    - Relative loss, cumulative loss, RTT, jitter
    - Allows receivers to relate themselves to group reception quality
  - Simple form: general statistics report on loss and jitter
  - Feedback target address
    - Where to unicast feedback packets to
  - SSRC collision reports
  - RTCP bandwidth indication

RTCP RSI Packet Format

```
  0                   1                   2                   3
  0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |V=2| reserved |   PT=RSI=208  |             length             |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |                           SSRC/CSRC                           |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |                                                               |
 +                           Timestamp                           +
 |                                                               |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 :                     optional report blocks                    :
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Report Block:

```
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |      RBT      |    Length     |                               |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+      RBT-specific data        +
 |                                                               |
 :                                                               :
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Detailed Statistics Sub-Report Blocks

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
|     SRBT      |    Length     |          NDB          |   MF  |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                  Minimum Distribution Value                   |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                  Maximum Distribution Value                   |
+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
|                     Distribution Buckets                      |
|                             ...                               |
|                             ...                               |
+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
```

- Used for
  - Loss, Jitter, RTT, Cumulative Loss
- Reflects information collected from RTCP RRs

Other Report Blocks

- Feedback target address
  - In-band signaling for distribution source address
  - Security!
- SSRC Collision
  - Initiate selection of new SSRCs
- General statistics
  - Average loss, average jitter, highest cumulative loss
  - Calculated from received RTCP RRs
- RTCP Bandwidth indication
- Group size and average RTCP packet size

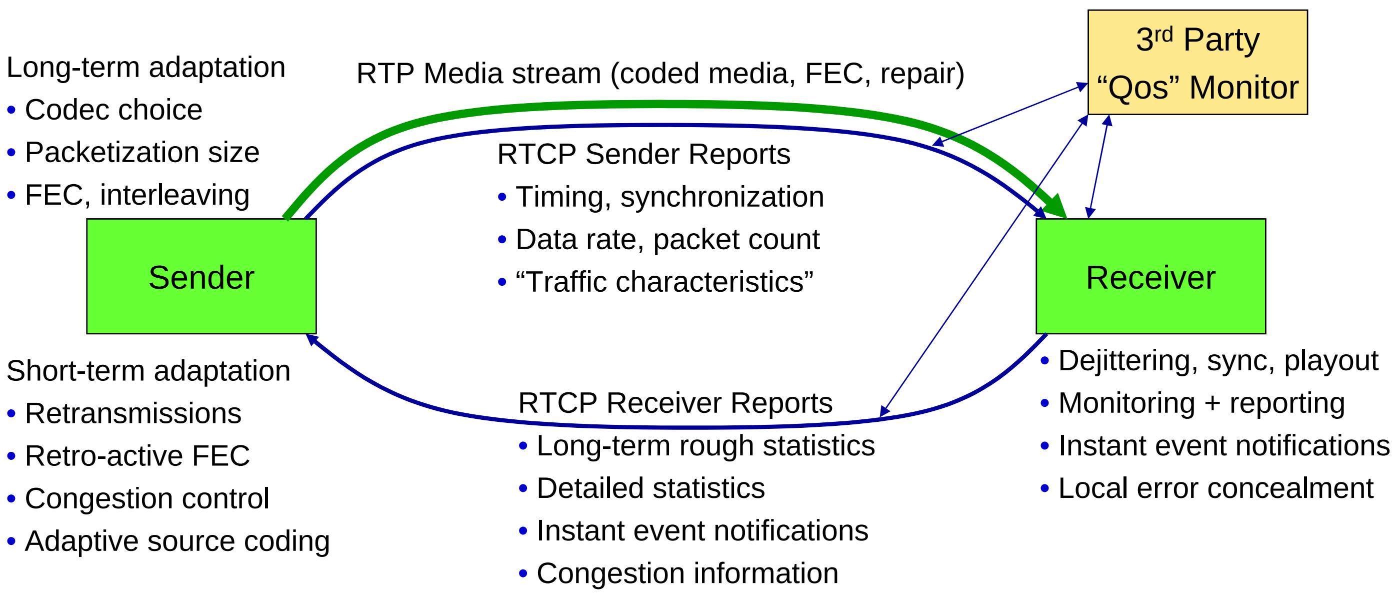
} **Pace RTCP RRs**

RTP Specs (Summary)

- RFC 3550 — Base specification (formerly RFC 1889)
- RFC 3551 — RTP Profile for Audio and Video Conference with minimal control (was RFC 1890)
- RFC 2198 — Redundant (Audio) coding
- RFC 2508 — RTP header compression for low-speed links
- RFC 2733,3009 — Generic FEC
- RFC 2736 — Guidelines for writers of RTP payload specifications
- RFC 2762 — Group membership sampling (“timer reconsideration”)
- RFC 3095 — Robust header compression for RTP (among others)
- RFC 3096 — Requirements for robust IP/UDP/RTP header compression
- RFC 3158 — RTP testing strategies
- RFC 3242 — Link-layer assisted profile for IP/UDP/RTP header compression
- RFC 3243 — Requirements & assumptions for 0-byte IP/UDP/RTP header compression
- RFC 3409 — Lower-layer guidelines for robust IP/UDP/RTP header Compression
- RFC 3545 — Enhanced compressed RTP (CRTP) for high-delay links
- RFC 3555 — MIME registrations of RTP payloads
- RFC 3611 — RTCP XR extension
- RFC 3711 — Secure RTP (SRTP)
- RFC 4362 — Robust Header Compression for IP/UDP/RTP
- RFC 4383 — TESLA for SRTP
- RFC 4571 — Framing RTP over Connection-oriented Transport
- RFC 4585,4586 — RTCP Feedback
- RFC 4588 — RTP Payload format for retransmissions

Summary: Applying RTP

- Adaptive real-time applications
  - Tunable feedback loop for individual and group communications
  - From reporting per 5s and more to event-driven to once per RTT



Present Issues and Concluding Remarks

- Implementing RTCP?
  - Yes—obviously it helps implementing good real-time applications
  - Yet, many VoIP applications don't do it
- Signaling: RTP vs. RTCP
  - RTCP sent infrequently—sufficient for signaling?
    - Frequency of RTCP vs. overhead
    - RTP level (e.g., congestion control) vs. application level (tunneled signaling protocol)
  - Shim layer in RTP?
    - Unidirectional media streams?
    - Demultiplexing?
- Reliability in RTP and RTCP
  - Retransmissions and FEC for RTP
  - Positive acknowledgements for RTCP?
    - Explicit messages vs. implicitly derived from data
- Maintaining group communication capabilities in RTP/RTCP
  - Various exceptions defined
- Important: Maintaining RTP's architectural integrity

