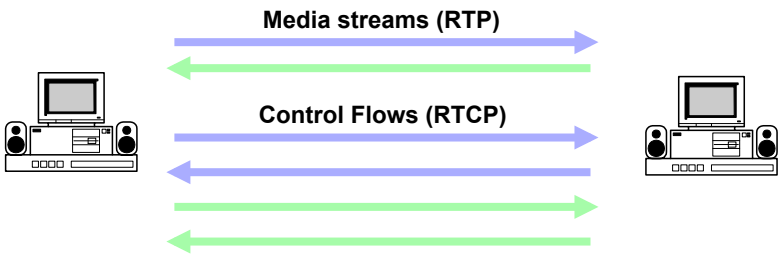
Real-Time Transport Protocol (RTP)

Real-time Transport Protocol (1)

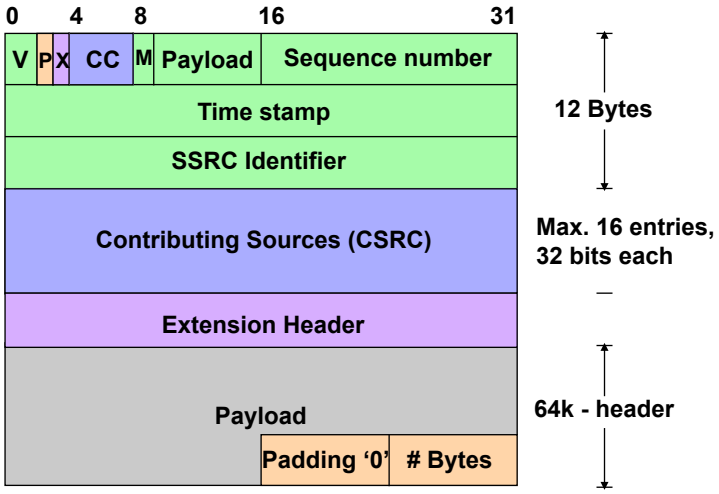
- RTP Functionality (RFC 3550)
  - framing for audio/video information streams
  - preserve intra- and inter-stream timing
  - mechanisms for awareness of others in a conference
  - → RTP sessions



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



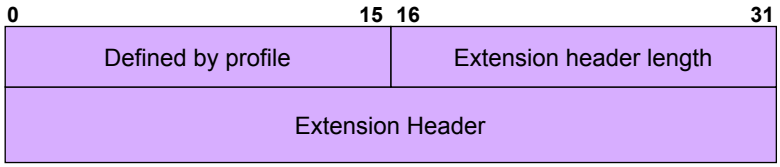
RTP Header Fields (1)

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:

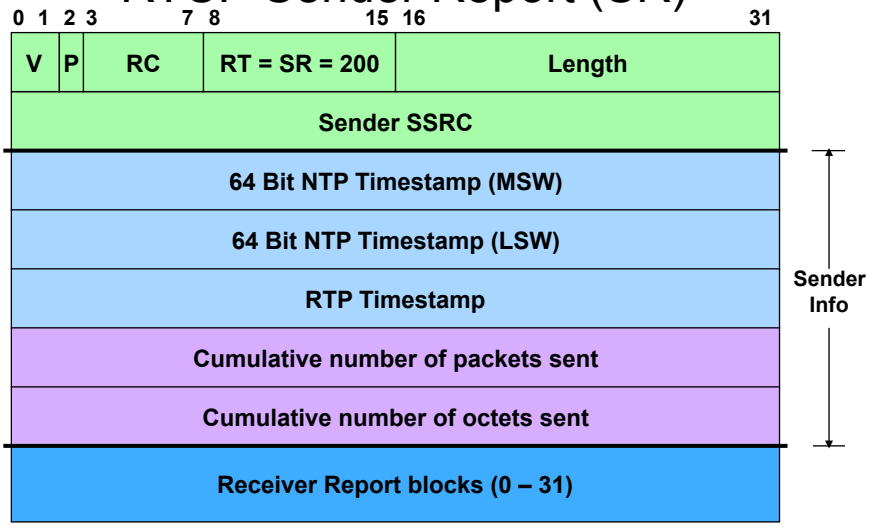
- Receivers constantly measure transmission quality
  - delay, jitter, packet loss
- Regular control information exchange between senders and receivers
  - feedback to sender (receiver report)
  - feed forward to recipients (sender report)
- Allows applications to adapt to current QoS
- Overhead limited 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

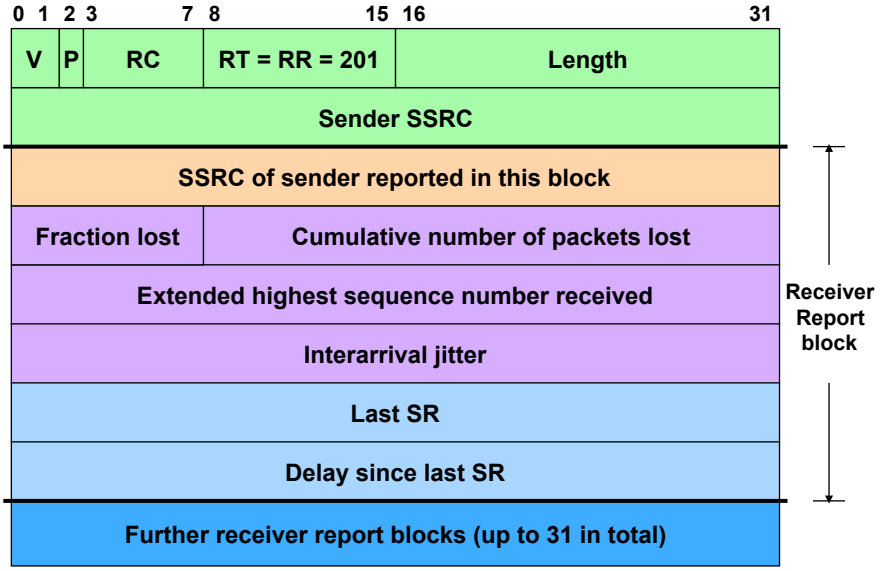
RTCP Sender Report (SR)



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)

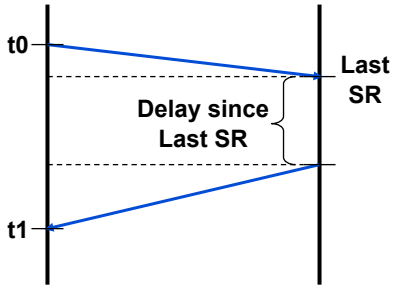


RTCP Statistics Collection (Sender)

- 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

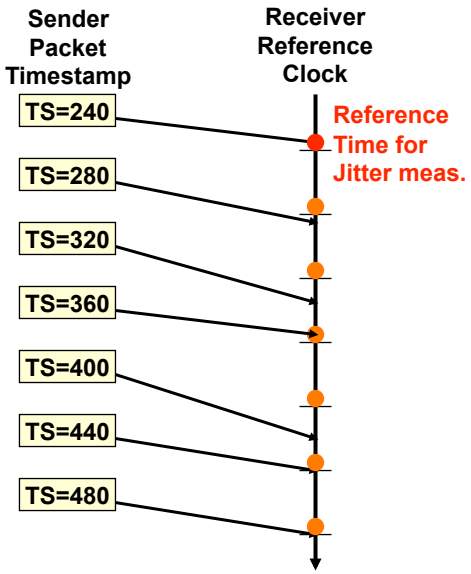
RTCP Statistics Collection (Receiver)

- 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:

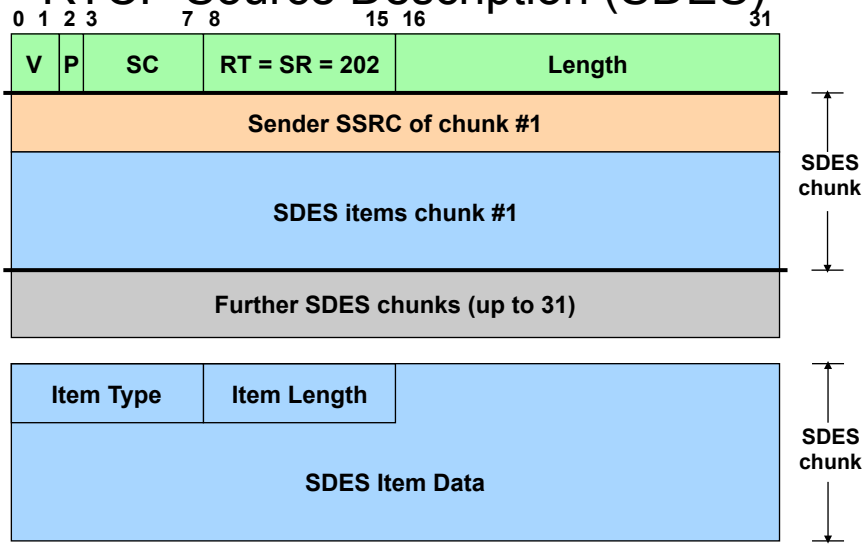
  J = J' + (D – J') / 16



RTCP Source Description (SDES)

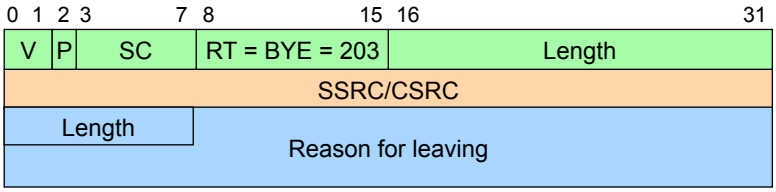
- Persistent Identification of an endpoint: Canonical Name
  - CNAME — globally unique identifier (id@host)
  - *Mandatory!*
  - Binding across RTP sessions
  - Identification across changes in the SSRC in an RTP session
- Providing additional information about an endpoint
  - NAME — Name of user (or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

RTCP Source Description (SDES)

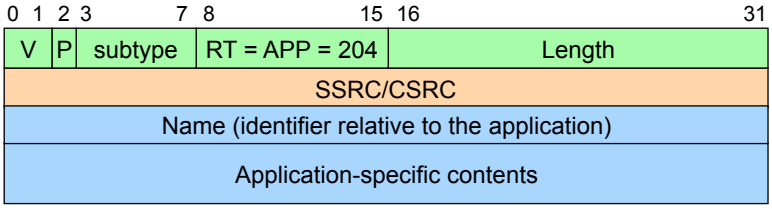


Other RTCP Packets

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



- APP: Application-specific extensions

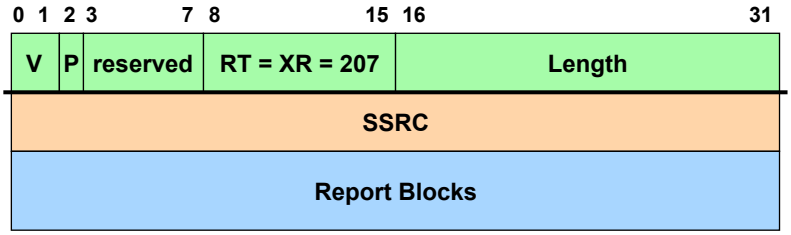


Extended RTCP Reporting (XR)

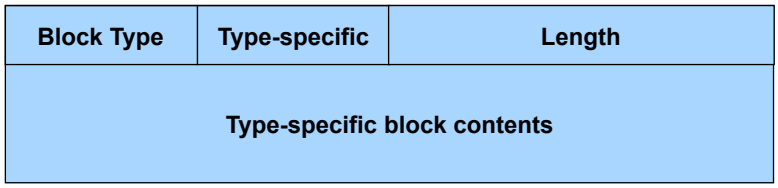
- Provide more detailed feedback (and feed forward)
  - Infer network characteristics (point-to-point and multicast)
  - Provide detailed voice quality information
- Incorporate many statistics in RTCP packets
  - Lost and duplicate packets
  - Exact packet receipt times
  - Receiver reference time and 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

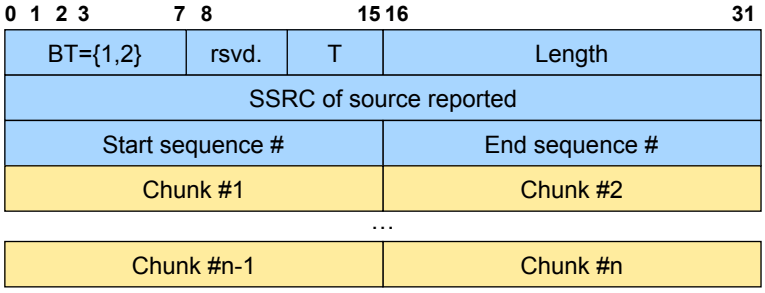


- Specific report blocks

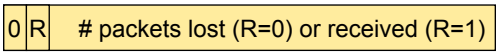


RTCP XR: Detailed Packet Reporting (1)

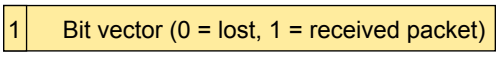
- Report (individual) lost and duplicate packets
  - Runlength encoding or bit maps of sequences (“chunks”)



- Run length:



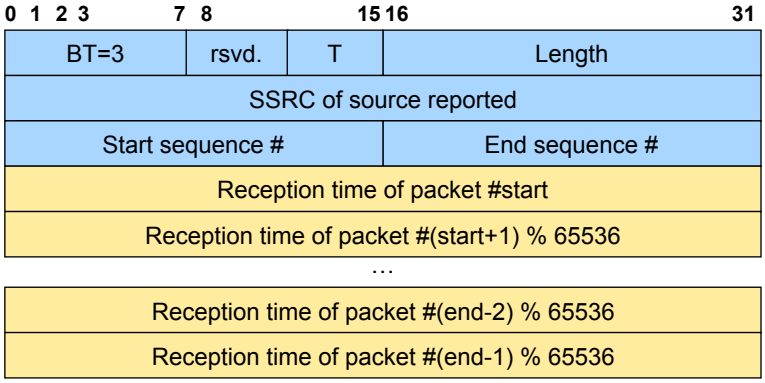
- Bit vector:



- Null chunk: 0x0000

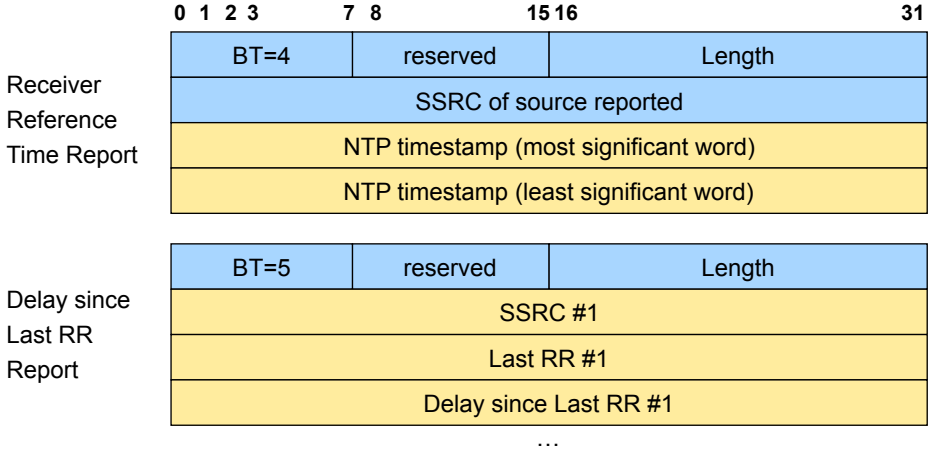
RTCP XR: Detailed Packet Reporting (2)

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



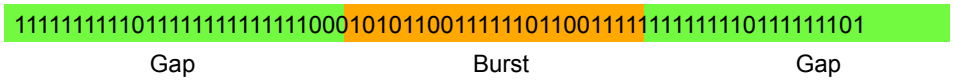
RTCP XR: Receiver Side RTT Calculation

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



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



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

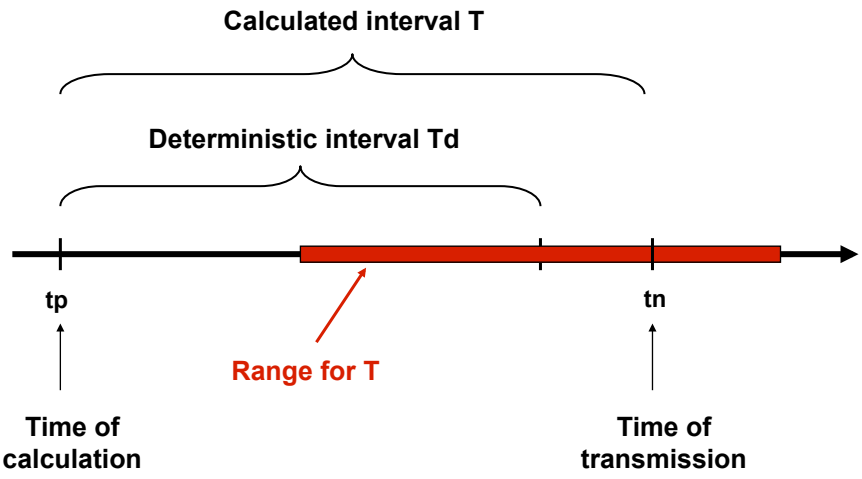
RTCP Variables for Bandwidth Calculation

- 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

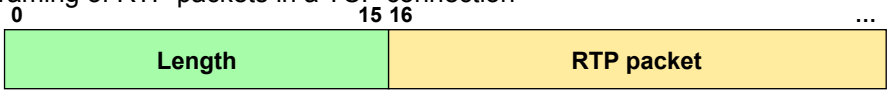
- Determine role (sender or receiver)
  - Derive n as # of relevant members for calculation
  - Derive relevant bandwidth share
- C = average RTCP size / relevant bandwidth share
- Td = max (Tmin, n*C)
- T = Random [0.5 – 1.5] * Td
- T = T / e^-1.5 (T = T / 1.21828)
  - Correction factor for timer reconsideration

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



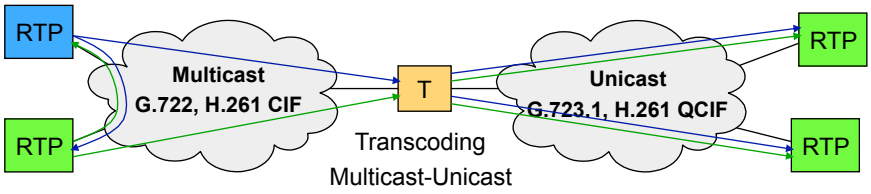
- Need to set up and tear down TCP connections for media
  - UDP is easy: just send
  - TCP: Who initiates, who accepts?
  - How to deal with accidental disconnection?

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

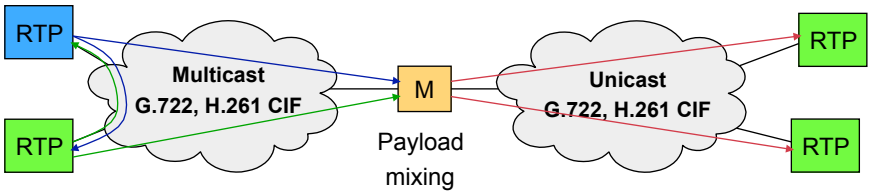
RTP Translator

- 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



See also: RFC 5117: RTP Topologies

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:”)

Payload formats defined for many audio/video encodings

- 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)

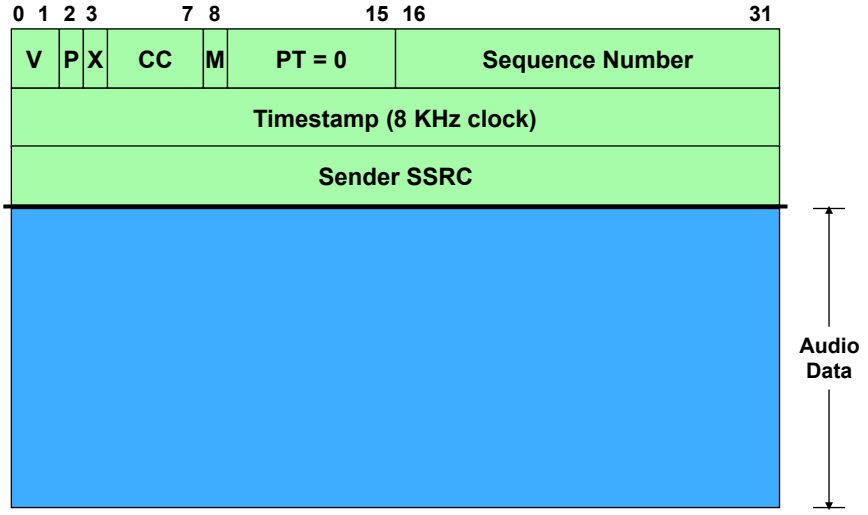
General principle:

- 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



Video over RTP: H.261

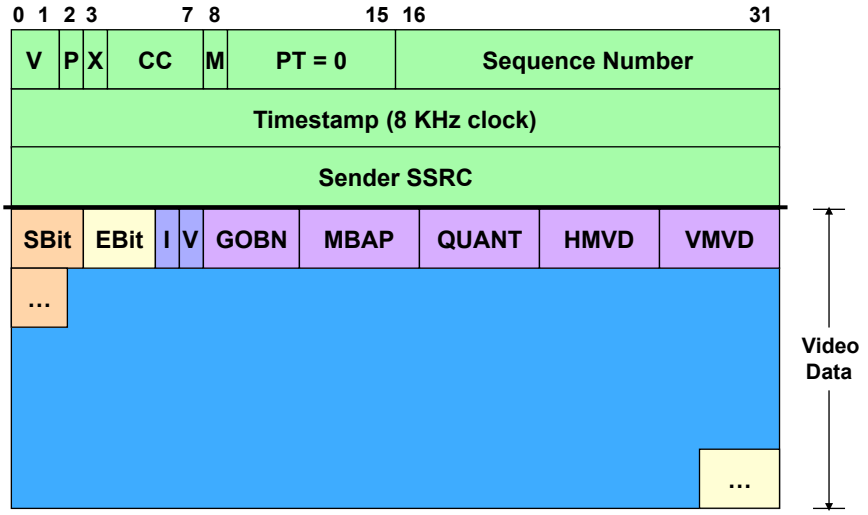
Additional payload-specific header preceeds payload

- 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)



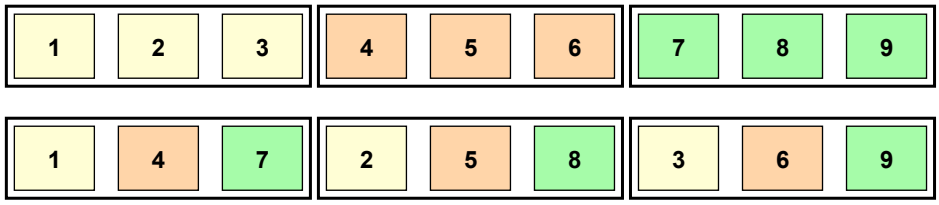
Media Packetization Schemes (2)

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, ...

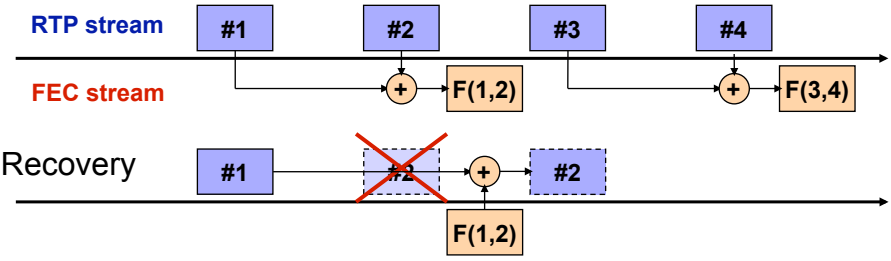
RTP Interleaving

- 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



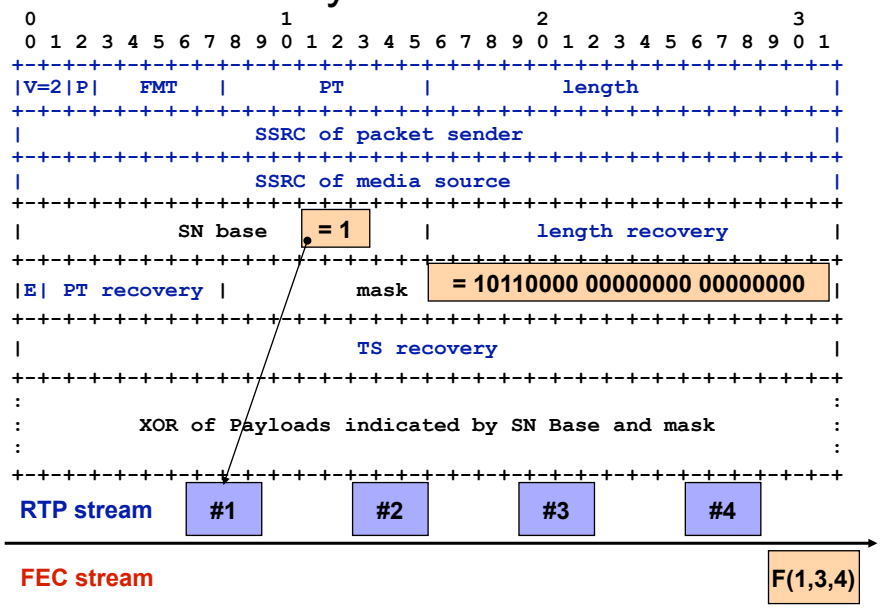
RTP FEC (RFC 2733)

- 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



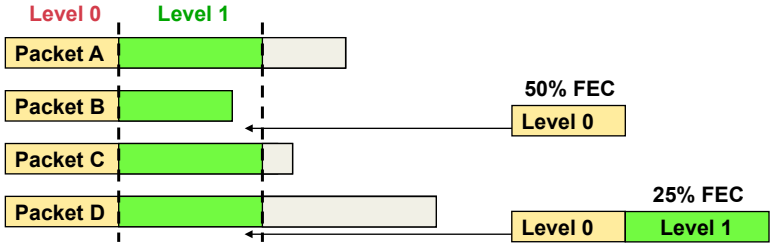
- Recovery

RTP Parity FEC Packet format



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

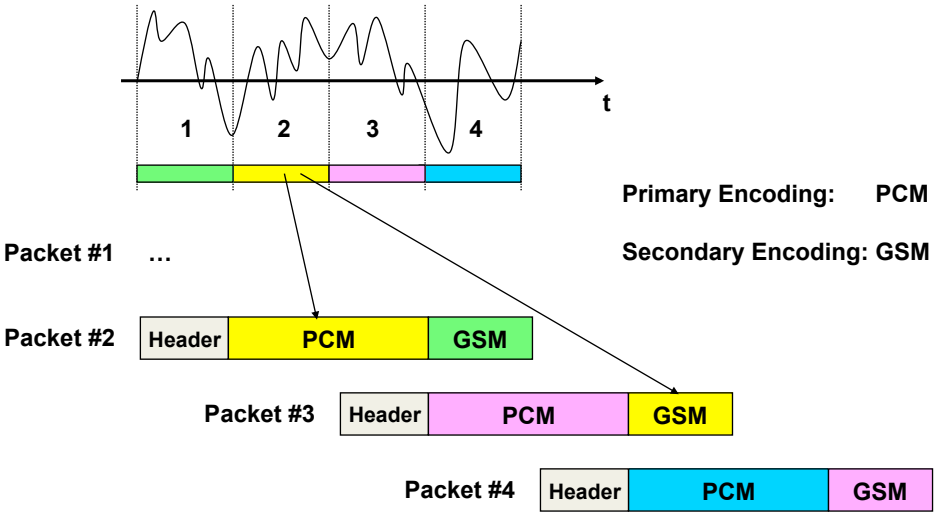


- Related thoughts: partial checksums
  - Live with bit errors in the less important parts (rather than dropping a packet)

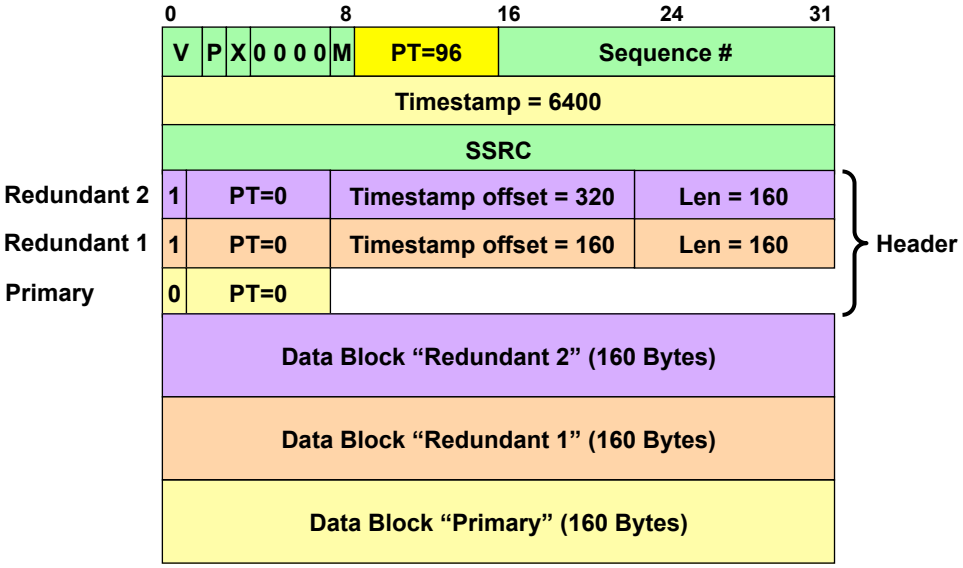
Audio Redundancy Coding (1)

- 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
  - 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)



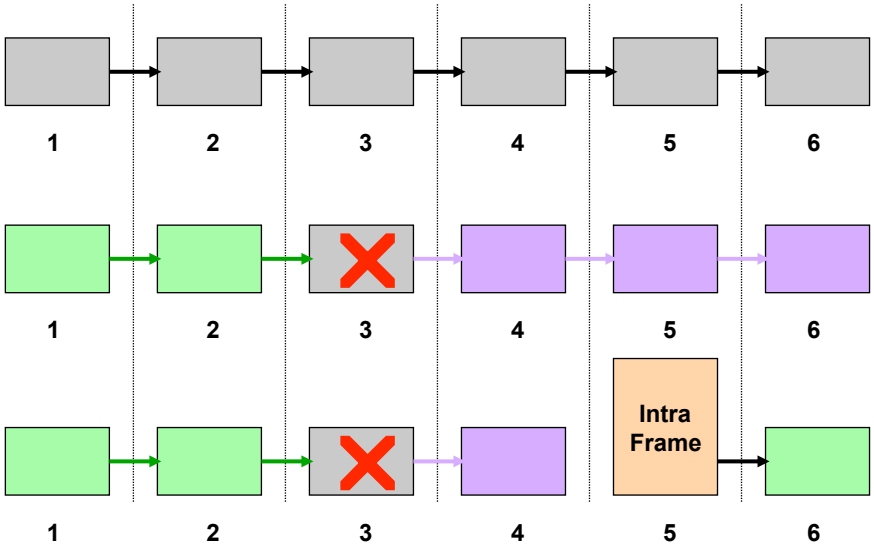
Audio Redundancy Coding (3)



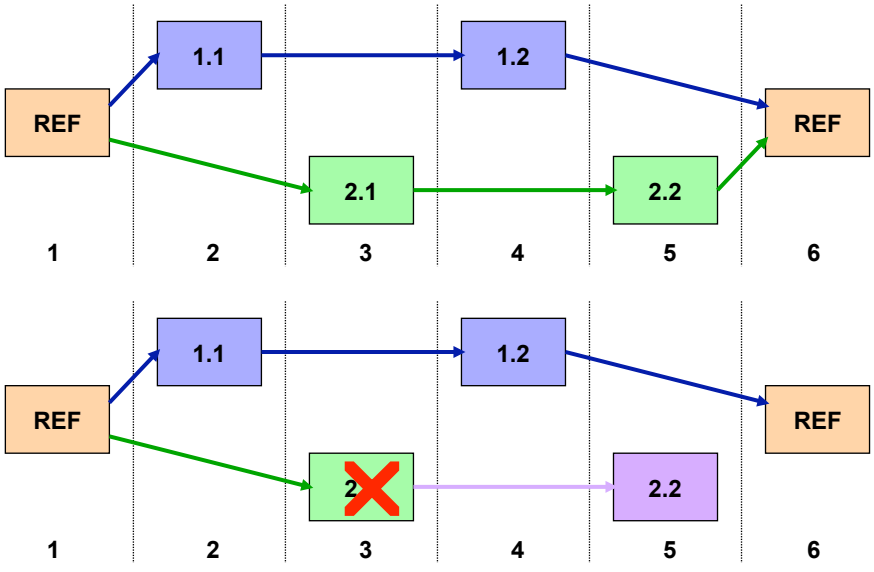
Video Redundancy Coding (1)

- 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

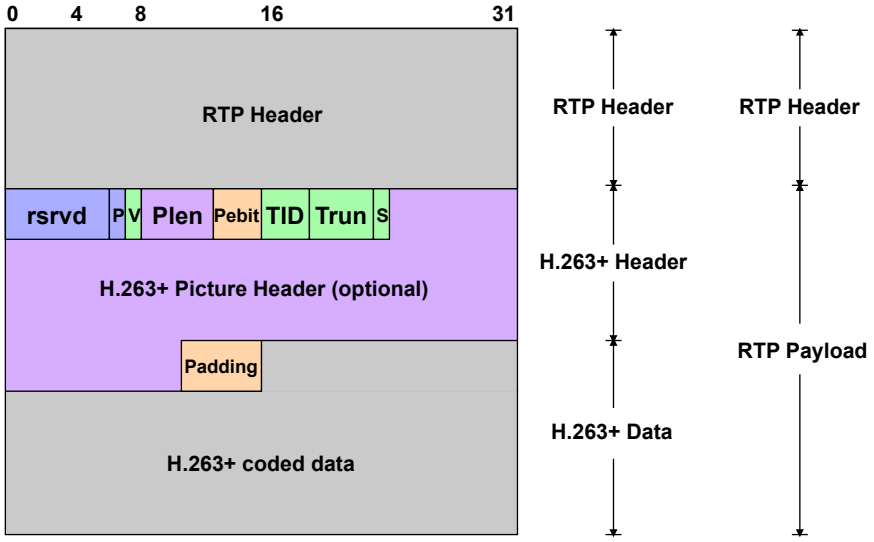
Video Redundancy Coding (2)



Video Redundancy Coding (3)



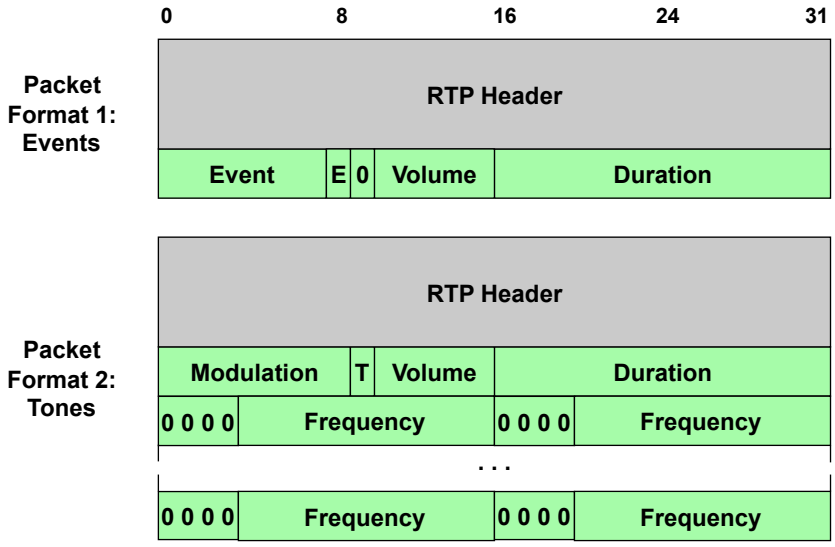
Video Redundancy Coding (4)



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)



RTP 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

RTP 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

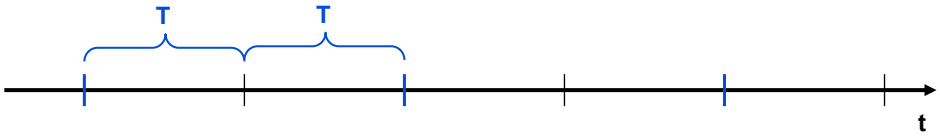
Idea:

- Packet losses are usually rare
- Provide statistical chance of virtually immediate feedback from receiver(s) to sender
- Keep the basic RTCP properties
- Eliminate Tmin

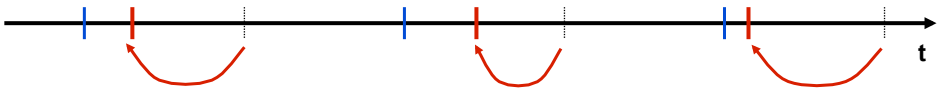
- Work most efficiently with unicast
- Also scale to moderate group sizes

Overview

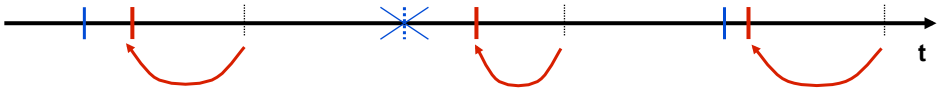
Regular RTCP operation (depicted w/o randomization, i.e. T = Td)



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

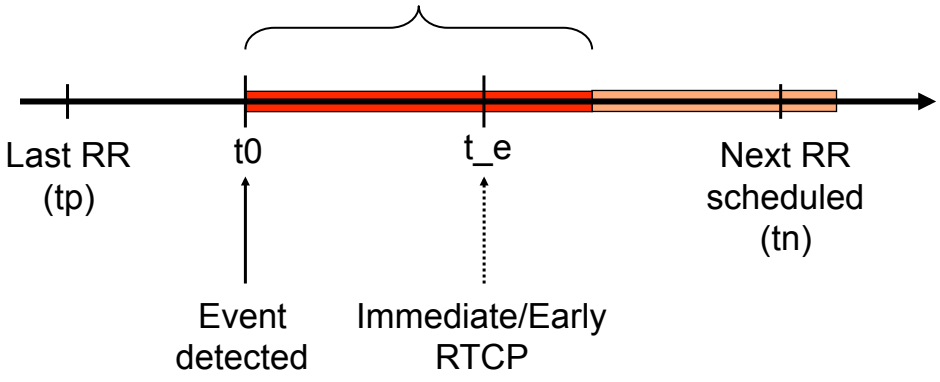


Allow to reduce the number of regular RTCP packets (w/o affecting RTCP rate)



RTCP Feedback Timing

T_dither_max = f (group size, ...)



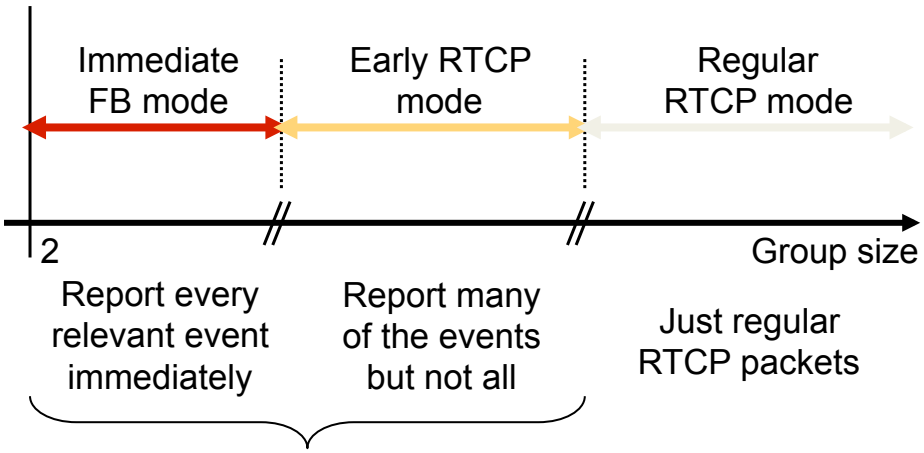
Delay calculation

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

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

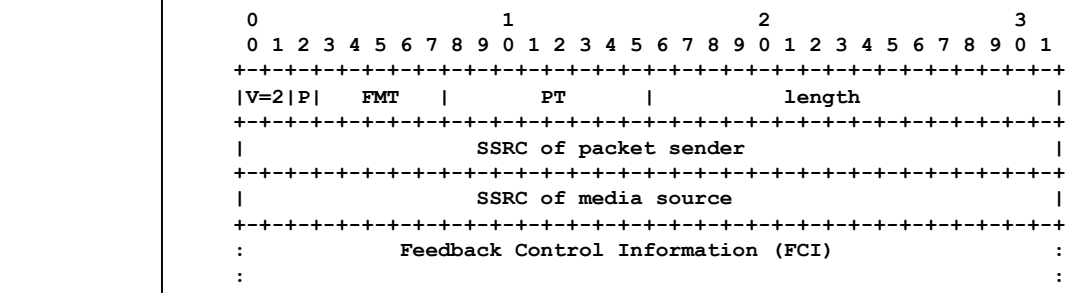


Send feedback + regular RTCP packets

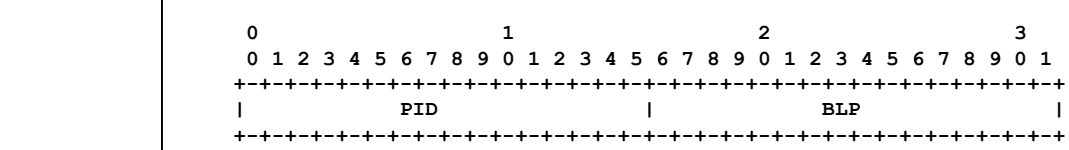
RTCP Types of Feedback

- 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

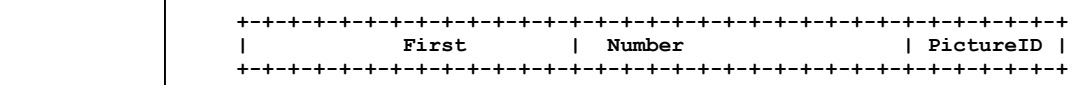


Example: Generic NACK Packet

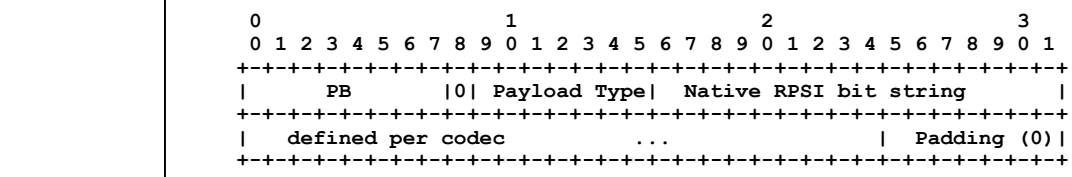


RTCP Feedback Packet Format (2)

Example: Slice lost indication



Example: Reference Picture Selection



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

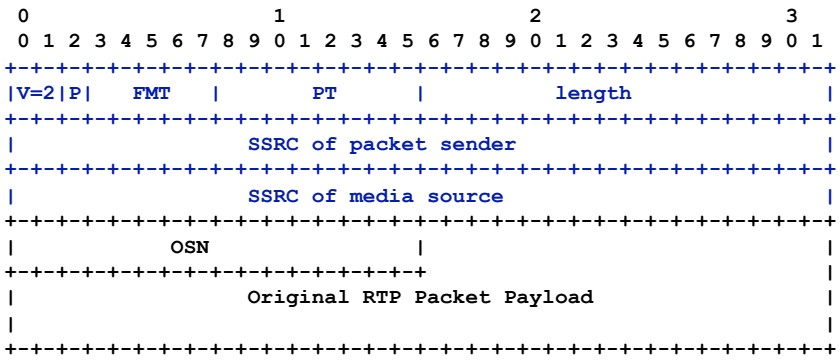
If reporting every other loss event is sufficient: ~14 receivers

Increases further if losses are correlated in some fashion

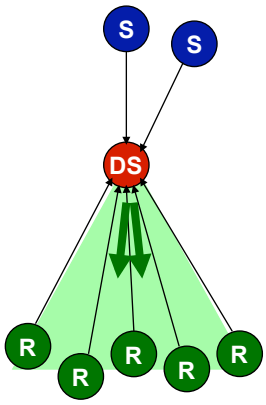
RTP Retransmissions

- 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

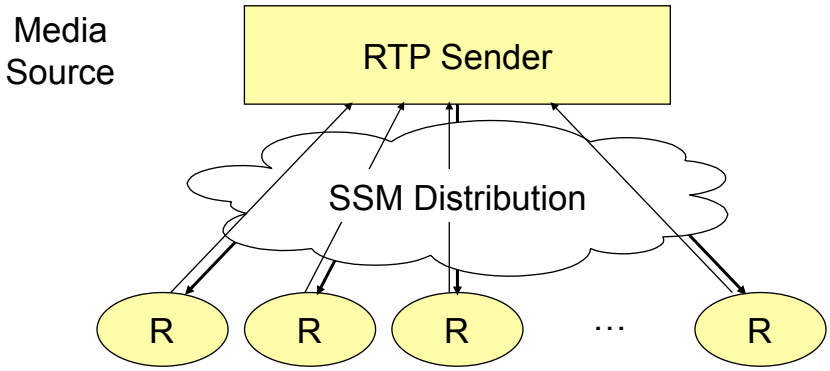


RTCP for SSM

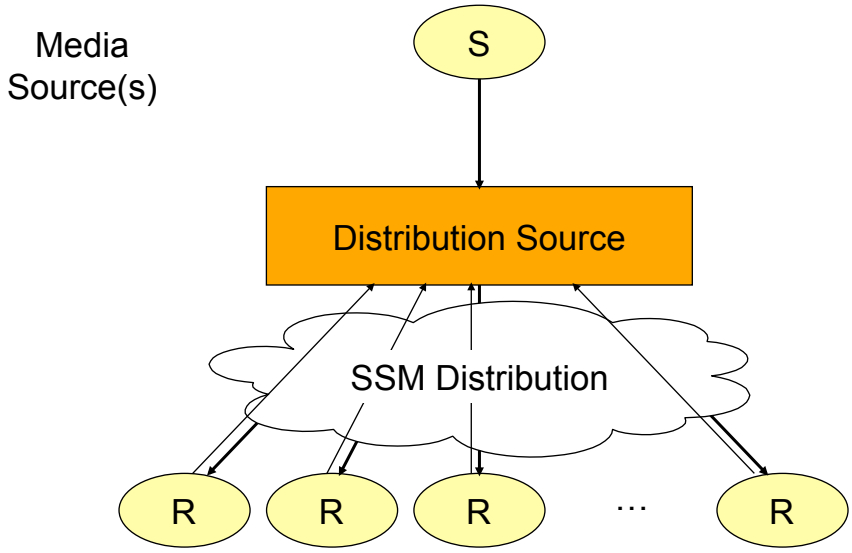


- Multicast connectivity unidirectional
  - From Distribution Source to receivers
  - Opposite direction needs to use unicasting
    - May follow different network path
- Result: no direct communication between receivers
- Adaptations required to make RTCP work
  - Estimate group size
  - Adjust timing 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

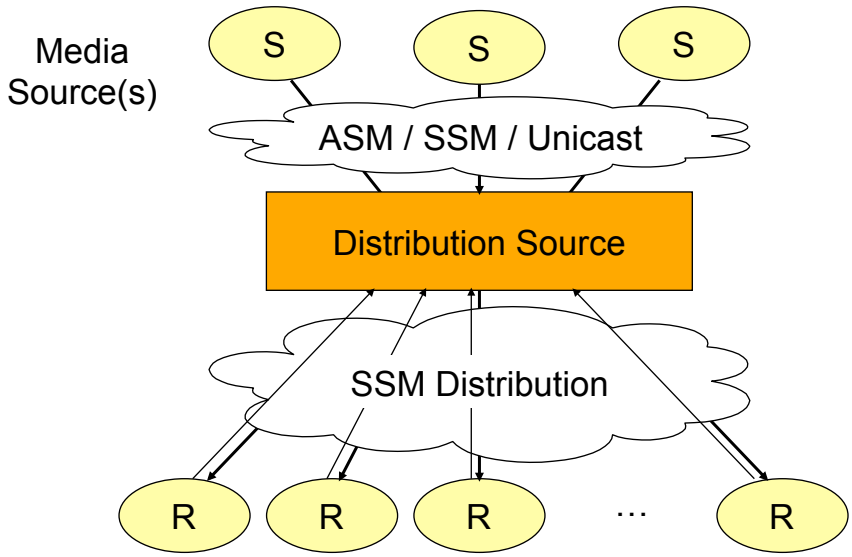
RTCP SSM Overview



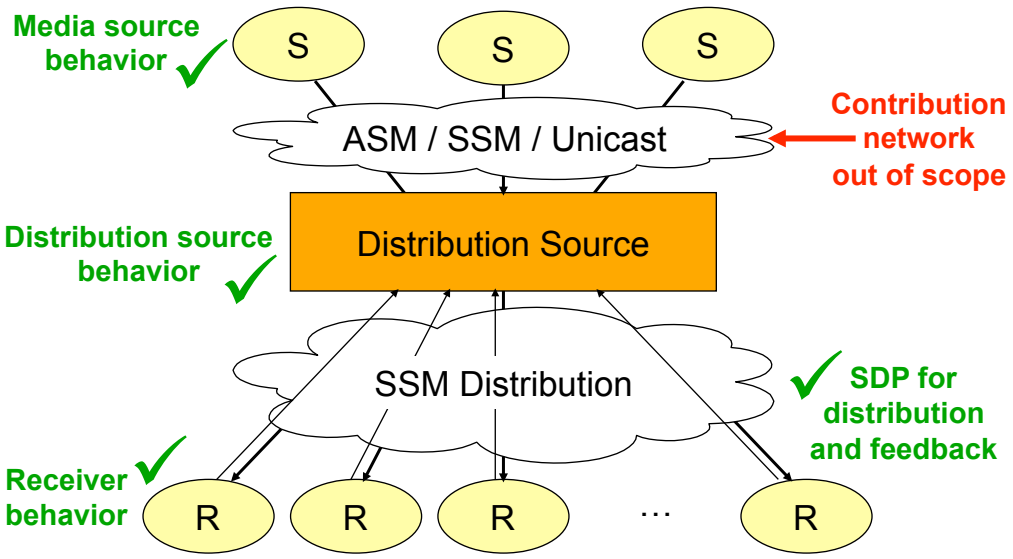
RTCP SSM Overview



RTCP SSM Overview



RTCP SSM Overview



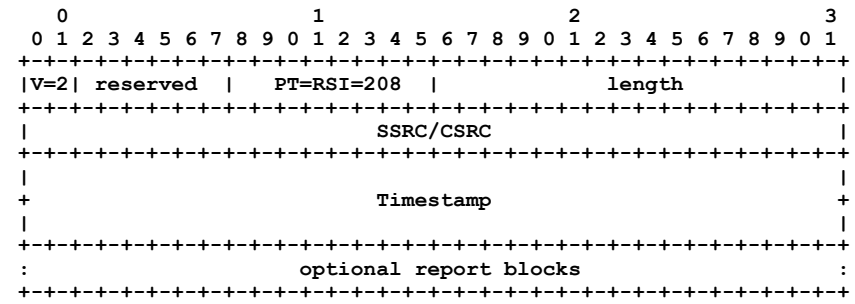
Simple Feedback Model

- 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

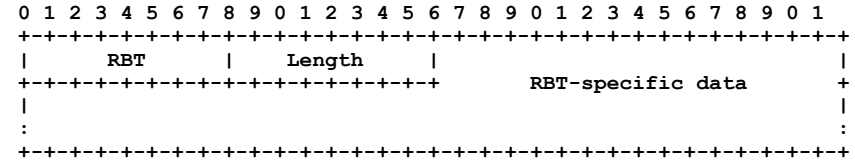
Feedback Summary Model

- 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

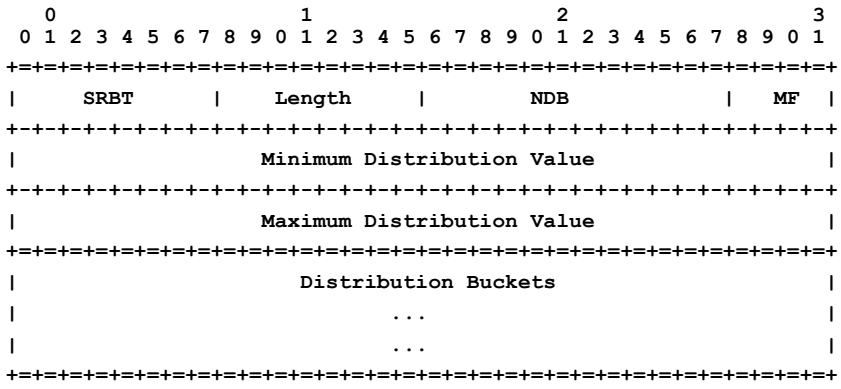
RTCP RSI Packet Format



Report Block:



Detailed Statistics Sub-Report Blocks



- 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

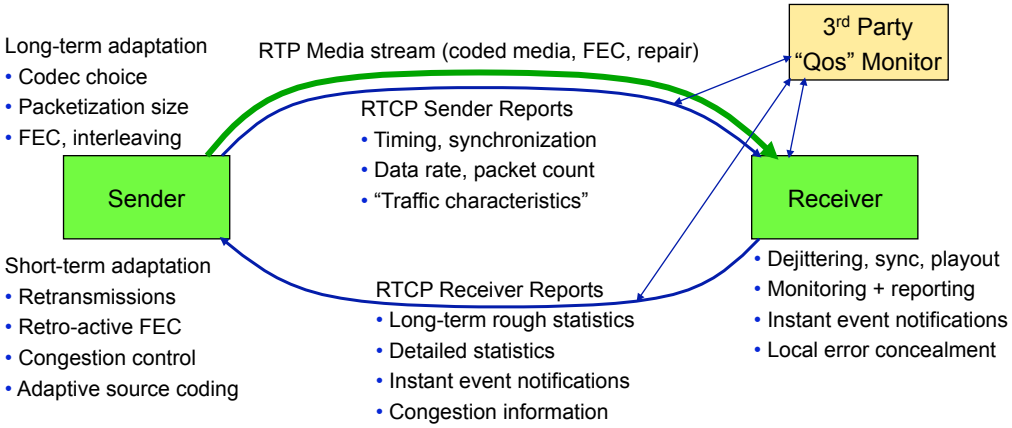
} **Pace RTCP RRs**

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