Multimedia Networking

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Departamento de Informática

Transport Protocols
Real-time Support
RTP/RTCP

Main Sources:
- *Computer Networking: A Top Down Approach* 7th edition. Jim Kurose, Keith Ross, Addison-Wesley, 2017.
- RFC3550, RFC5506

Internet transport protocols

RTP motivation

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- TCP not suited to real-time and interactive apps
  - point-to-point so not suitable for multicast
  - no way to associate timing with segments
  - causes arbitrary delays
- SCTP is multistream but has some of TCP drawbacks
- UDP does not include timing information nor any support for real-time apps. DCCP provides some support for buffer control but no media timing.

- Real-Time Protocol (RTP) aims to support the transport of *real-time* data, audio and video

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Goal: provides mechanisms for end-to-end delivery of real-time data (audio, video)

- Typical apps: videoconferencing, IP telephony, etc.
- Supports unicast or multicast sessions
- Uses a data channel (RTP) and a control channel (RTCP)
- RTCP (Real Time Control Protocol) reports the status of the data channel
- RTP (RFC 3350)

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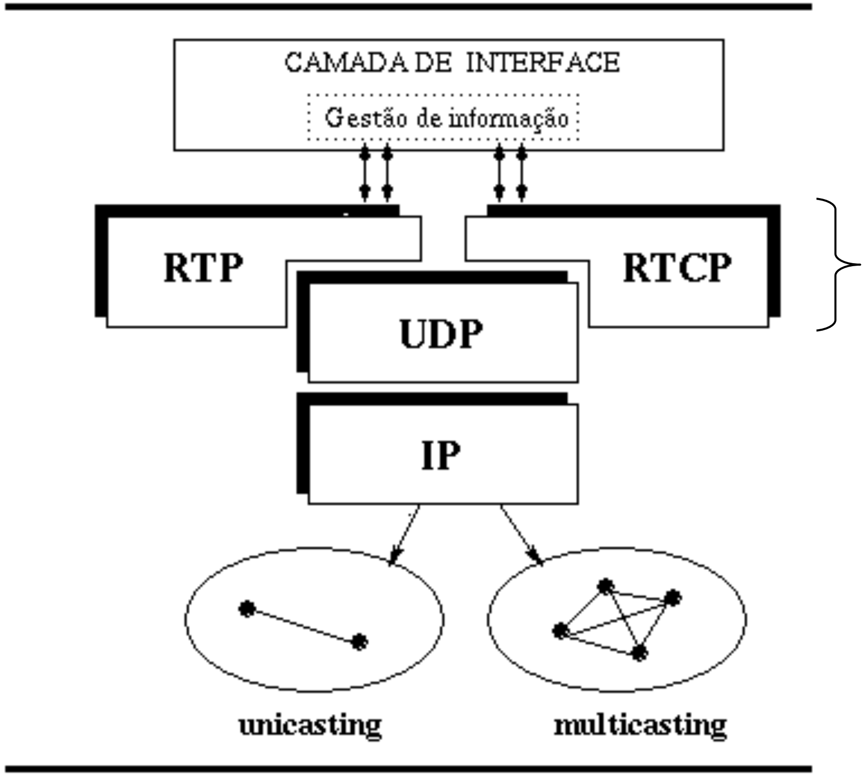
- Support for application real-time data delivery



- Support to control application real-time data

- RTP in the protocol stack

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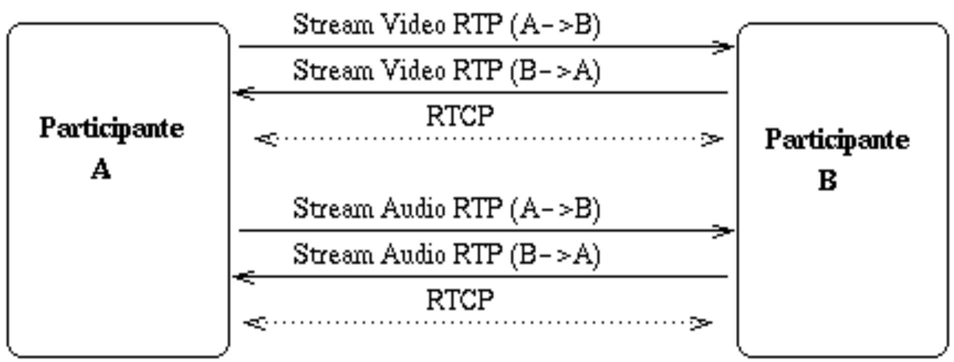
RTP

- RTP Session = data channel + control channel
- Session address = network address (unicast,multicast), pair of ports
- RTCP channel = RTP port +1
- In presence of multiple media streams, each one is transmitted in a separate session or mixed in a single session in each direction, depending on the codecs (e.g. MPEG-x mix audio and vídeo / session).



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- RTP issues
  - No reservation, No Quality of Service (QoS) guarantees
  - No guarantee of packet delivery

  - Note that RTP does not provide any mechanism to ensure timely data delivery or other QoS guarantees.
    - routers may provide differentiated service delivery, but there is no guarantees that RTP packets arrive at destination in a timely manner.

  - RTP encapsulation is only seen at end-systems, not by intermediate routers.

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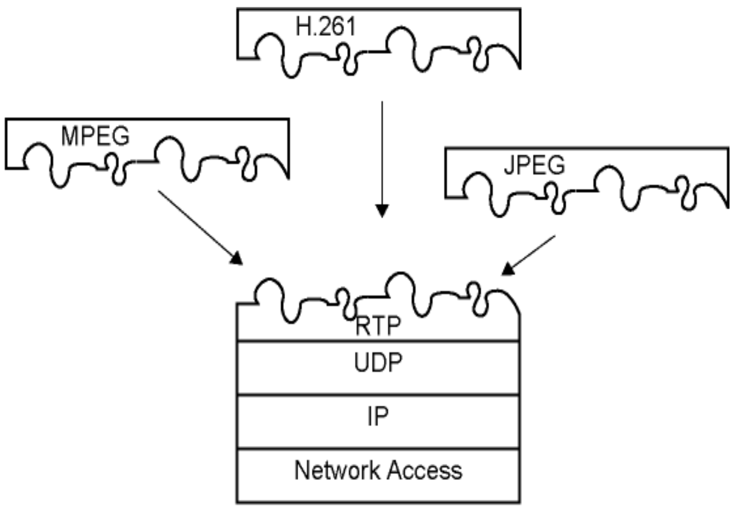
- RTP issues
  * Allows timing reconstruction, loss detection and content identification



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- RTP specifies a packet structure for packets carrying audio, video data
- RTP packet provides
  - time stamping
  - packet sequence numbering
  - payload type identification
- RTP packets usually encapsulated in UDP segments (also in DCCP)
- Interoperability: if two Internet phone applications run RTP, then they may be able to work together (session control at RTP level)

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RTP example

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- Consider sending 64 kbps PCM-encoded (8 bits) voice over RTP.
- application collects encoded data in chunks, e.g., every 20 msec = 160 samples in a chunk (160 bytes).
- an audio chunk + RTP header form an RTP packet, which is encapsulated in e.g. an UDP segment
- RTP header indicates type of audio encoding in each packet
  - sender can change encoding during conference.
- As mentioned, RTP header also contains sequence number and timestamp.

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- RTP Timestamp
  - primary goal: represent the inherent notion of real-time associated with media (sampling units)
  - used to place packets in correct timing order
  - may be used to evaluate jitter
- Sequence Number
  - used to detect RTP packet loss

- If a video frame spans multiple RTP packets, then timestamp is unchanged and sequence number is incremented

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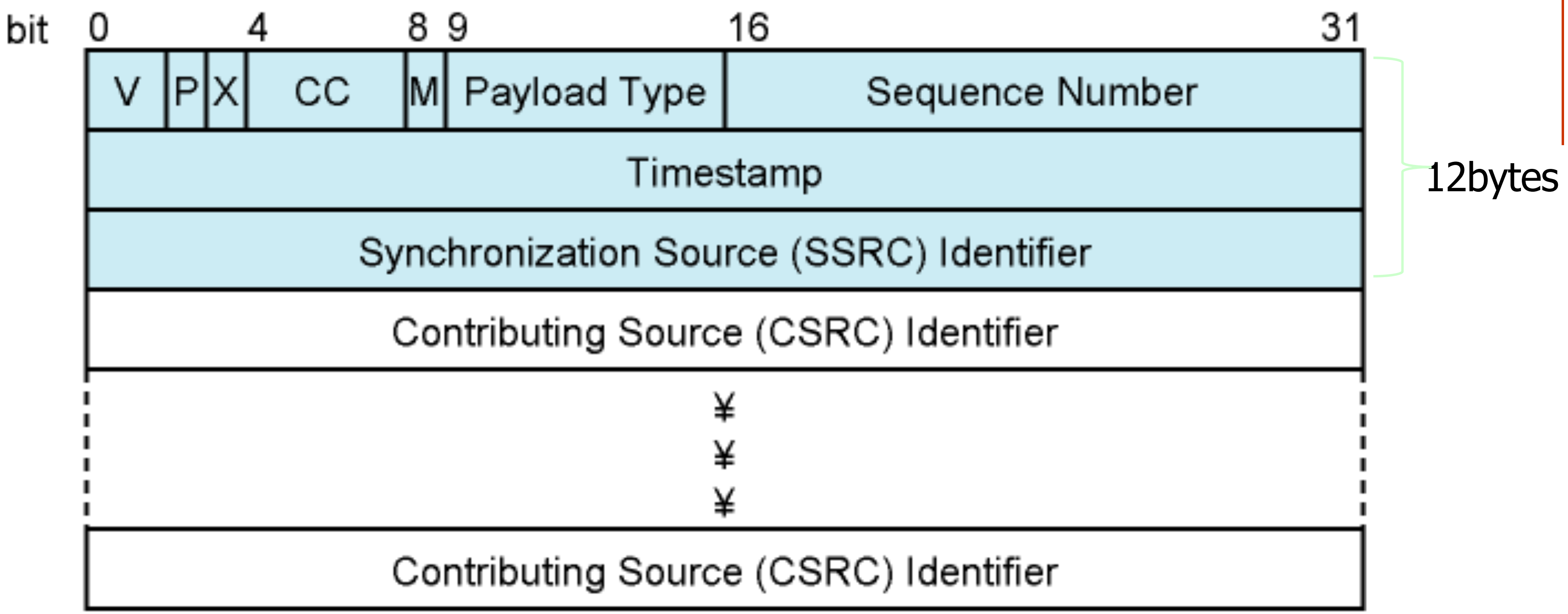
RTP header



V = Version
P = Padding
X = Extension
CC = CSRC count
M = Marker

[W.Stallings, Computer Networks w/ Internet Prot. & Tech., 2004]

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RTP header

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Payload Type (7 bits): Indicates the type of encoding currently being used. If sender changes encoding in middle of a conference, sender informs receiver via payload type field.

- Payload type 0: PCM mu-law, 64 kbps
- Payload type 3, GSM, 13 kbps
- Payload type 7, LPC, 2.4 kbps
- Payload type 26, Motion JPEG
- Payload type 31. H.261
- Payload type 33, MPEG2 video
- see RFC 3551…

Seq. Number (16 bits): Increments by one for each RTP packet sent; may be used to detect packet loss and to restore packet sequence at receiver.

SSRC field (32 bits): Identifies the source of the RTP stream (allows independence of the network address), provides an ID for playout.

Each stream in RTP session should have distinct SSRC (random init).

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RTP header – SSRCs vs. CSRCs

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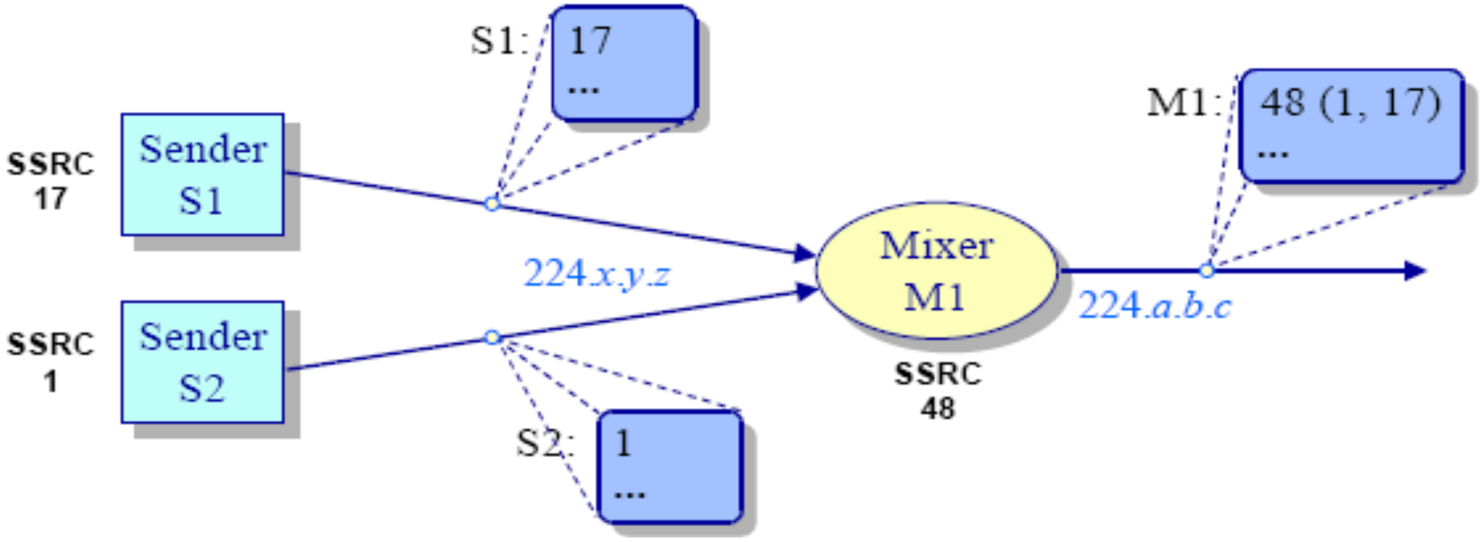
RTP mixer - an intermediate system that receives and combines RTP PDUs of one or more RTP sessions into a new RTP PDU



Synchronization Sources(SSRC) become Contributing Sources (CSRC)

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RTP header - timestamp

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- Timestamp field (32 bits): reflects the sampling instant of first octet in the RTP data packet
  - initial value random for each RTP stream
  - for audio, timestamp clock typically increments by one for each sampling period (e.g., each 125 usecs for 8 kHz sampling clock)
  - if application generates chunks of 160 encoded samples each 20ms, then timestamp increases by 160 units for each RTP packet when source is active.
  - timestamp clock continues to increase at constant rate when source is inactive; it increments monotonically and linearly
  - if the sampling rate is variable (e.g. software codecs), the system clock may be used (e.g. gettimeofday())

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RTCP – Real-Time Control Protocol

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- RTCP works in conjunction with RTP.
- each participant in RTP session periodically transmits RTCP control packets to all other participants.
- each RTCP packet contains sender and/or receiver reports
- minimal overhead (e.g. 1 RTCP pkt each 5 sec)
- feedback can be used to control application performance
  - sender may modify its transmissions based on feedback
  - report statistics useful to application: #packets sent, #packets lost, interarrival jitter, etc.
- allows to identify and keep track of participants

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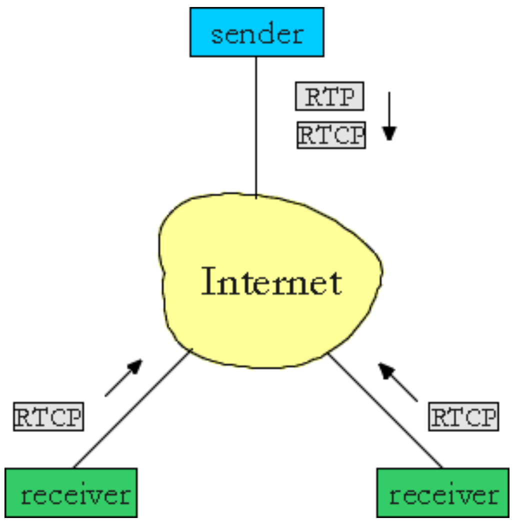

- each RTP session: typically uses a single multicast address; all RTP/RTCP packets belonging to that session use multicast address (IP class D).
- RTP, RTCP packets distinguished from each other via distinct port numbers.
- to limit traffic, each participant reduces RTCP traffic as the number of conference participants increases

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RTCP packets

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Sender Report (SR) packets:

- SSRC of RTP stream, current time, number of packets sent, number of octets sent

- allows receiver to:
  - estimate mean data rate;
  - estimate mean packet size;
  - distinguish reception breaks (pauses in transmission vs. network problems)

Receiver Report (RR) packets:

- fraction of packets lost, last sequence number, average interarrival jitter

Source Description packets:

- e-mail address of sender, sender's name, SSRC of associated RTP stream
- provide mapping between the SSRC and the user/host name

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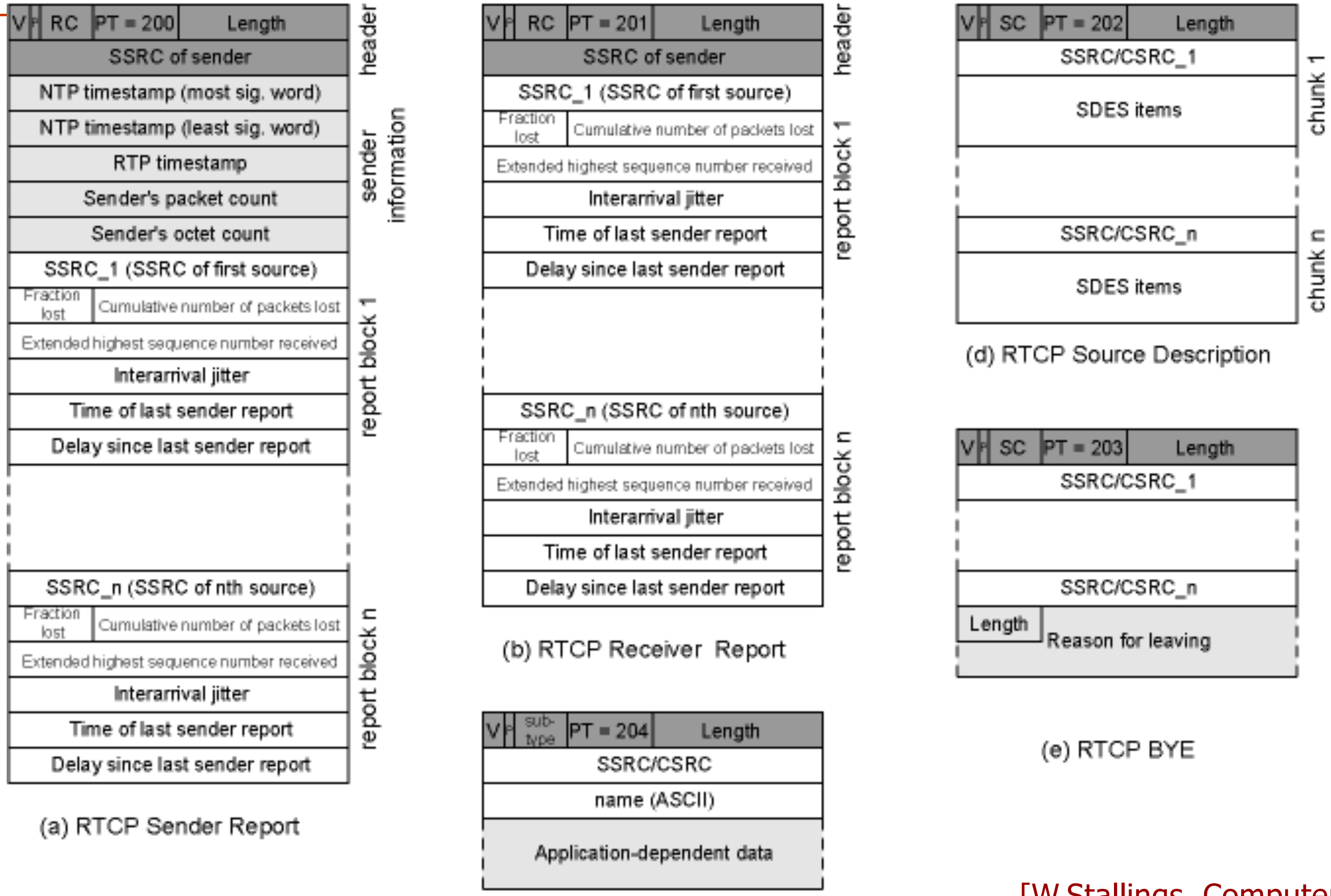
RTCP packet formats



(a) RTCP Sender Report

(b) RTCP Receiver Report

(c) RTCP Application-defined packet

(d) RTCP Source Description

(e) RTCP BYE

[W.Stallings, Computer Networks w/ Internet Prot. & Tech., 2004]

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RTCP Reports

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- RRs and SRs indicate reception statistics from each receiver and for every sender. These statistics include:
  - packet loss ratio since the last SR or RR was sent;
  - total # packets lost since the beginning of the session (may be broken down per reporting period);
  - highest sequence number received so far, which allows to estimate how much data is in flight (when used together with the SR and RR timestamps);
  - moving average of the interarrival jitter of media packets; which gives the sender an indirect view of the size of any adaptive playout buffer used at the receiver ([RFC3611]), e.g. useful for VoIP sessions.

[RFC 5968., 2010]

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RTCP synchronization of streams

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- RTCP can synchronize different media streams within a RTP session
- consider videoconferencing app for which each sender generates one RTP stream for video, one for audio.
- timestamps in RTP packets are tied to the video, audio sampling clocks
  - not tied to wall-clock time (NTP gives an absolute time)
- each RTCP sender-report packet contains (for most recently generated packet in associated RTP stream):
  - timestamp of RTP packet
  - wall-clock time for when packet was created.
- receivers uses association to synchronize playout of audio, video

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RTCP bandwidth scaling

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- RTCP has a potential bandwidth scaling problem as the number of receivers R increases
- RTCP aggregated bandwidth consumption increases linearly with R
- To minimize that, RTCP receivers adjusts the sending rate to the multicast tree according to R

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RTCP bandwidth scaling

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- RTCP attempts to limit its traffic to 5% of session bandwidth.

Example

- Suppose one sender, sending video at 2 Mbps. Then RTCP attempts to limit its traffic to 100 kbps.
- RTCP gives 75% of rate to receivers; remaining 25% to sender
- 75 kbps is equally shared among receivers:
  - with R receivers, each receiver gets to send RTCP traffic at 75/R kbps.
- sender gets to send RTCP traffic at 25 kbps.
- participant determines RTCP packet transmission period by calculating avg RTCP packet size (across entire session) and dividing by its allocated rate

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RTP/RTCP references

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- Lots of good material at Henning Schulzrinne's RTP page
  http://www.cs.columbia.edu/~hgs/rtp/
- RFCs on RTP/RTCP
  - G.Ott, C.Perkins, *Guidelines for Extending the RTP Control Protocol* (RTCP), RFC 5968, Sep. 2010.
  - C.Perkins, T. Schierl, *Rapid Synchronisation of RTP Flows*, RFC 6051, Nov. 2010
  - A.Begen, *RTP Control Protocol (RTCP) Port for Source-Specific Multicast (SSM) Sessions*, RFC 6128, Feb. 2011.
  - A. Began, C. Perkins, D.Wing, *Guidelines for Choosing RTP Control Protocol (RTCP) Canonical Names (CNAMEs)*, RFC 6222, Apr. 2011.
  - ...