CSc 461/561
Multimedia Systems
RTP/RTCP

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Spring 2015

Why RTP?

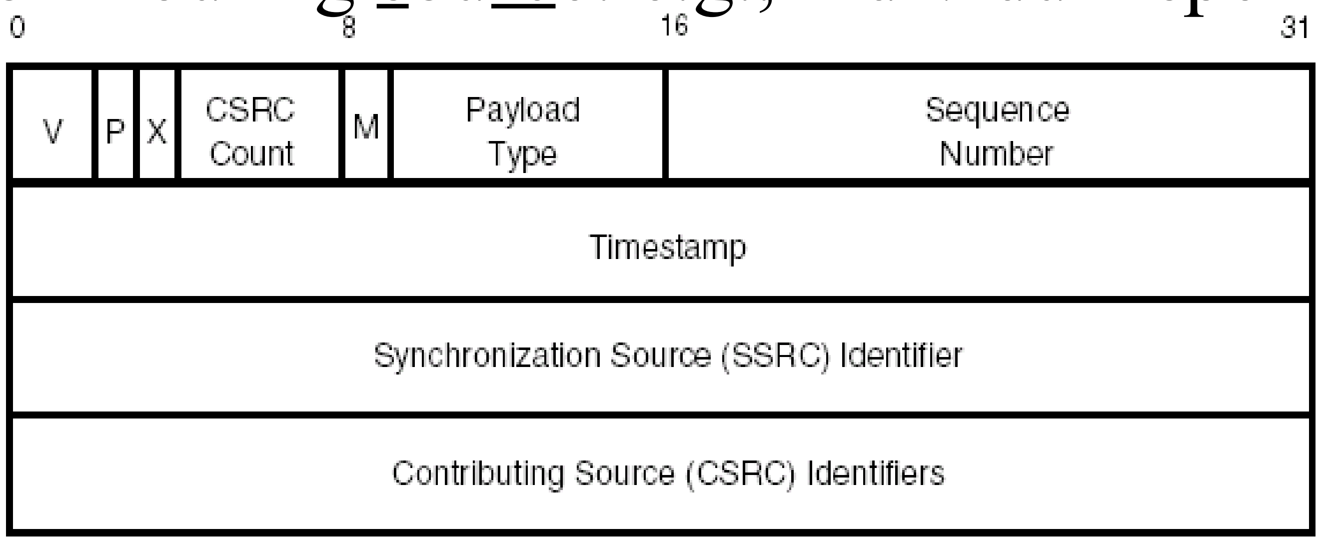
- Why not TCP?
  - TCP offers reliable, in-sequence data transfer
  - TCP embeds flow/error/congestion control
- Why not UDP?
  - UDP offers datagram-like service
- RTP: transport protocol for multimedia traffic
  - application level framing; integrated layer processing
  - usually RTP/UDP/IP, or RTP/IP

What RTP offers?

- RTP: “real-time” transport protocol
  - does NOT guarantee real-time itself
  - but does provide <u>mechanisms</u> to achieve so
- RTP header fields
  - payload type (media type and coding scheme)
  - sequence number (packet count)
  - timestamp (sample count)
  - synchronization/contributing source identifier

RTP header

- Version=2; Padding; extension; Marker
- Synchronization source: e.g., source, mixer
- Contributing source: e.g., individual speaker



Source, translator and mixer

- Translator
  - one-in-and-one-out, media transcoding
  - protocol translation
  - outgoing packets source IP: translator’s IP
- Mixer
  - multiple-in-and-(usually)-one-out
  - outgoing packets SSRC: mixer’s ID
  - outgoing CSRC: IDs from input streams

Sequence # and timestamp

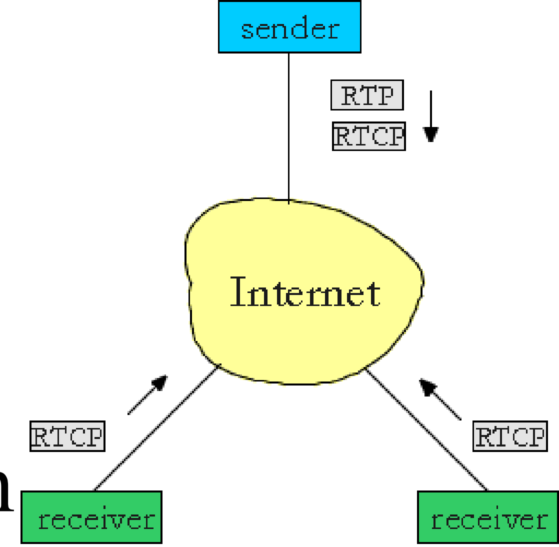
- Sequence number (packet count)
  – gap in sequence #: packet loss
- Timestamp (sample count)
  – e.g., audio@8KHz, 20ms samples/packet
    - timestamp increment per packet: 160
  – 90KHz used for video
    - a video frame may be encapsulated in a few packets
  – gap in timestamp: silence

RTP profiles

- Media specific (e.g., audio)
  - Marker: e.g., the start of a talk spurt
  - Payload Type: e.g., specific audio codec
    - PT=0: uPCM 64Kbps; PT=3: GSM 13Kbps
  - timestamp: e.g., sampling rate, 8KHz PCM
  - packet size: e.g., about 20ms samples in PCM
    - packets independent as much as possible: ALF
  - other issues: e.g., mixed audio channels

* ALF: application-layer framing

RTCP



- RTCP: RTP's control companion
  - purpose: feedback control information
    - for flow/error/congestion/quality control
  - two consecutive UDP ports for RTP and RTCP
  - sender report: offer sending/reception statistics
  - receiver report: offer reception statistics
- RTCP common header: Version (2-bit); Padding (1); RR count (5); Type (8); Length (16)

RTCP: SR

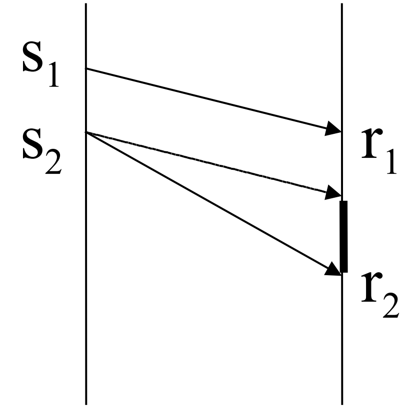
- Sender report (SR)
  - SSRC of sender (32-bit): source media
  - NTP timestamp (64-bit): wall clock
  - RTP timestamp (32-bit): sample count
    - for synchronization among media streams
  - packet count (32-bit): total # of packets sent
  - byte count (32-bit): total amount of data sent
    - receiver: rate/loss estimation

RTCP: RR

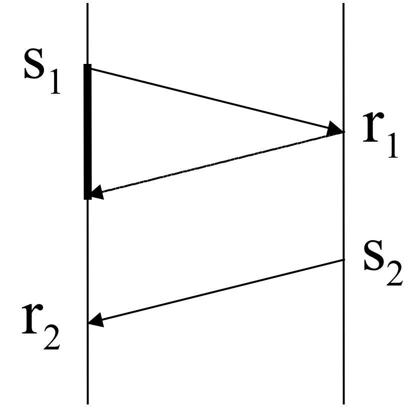
- Receiver report (RR)
  - SSRC of source (32-bit): source media
  - fraction lost (8-bit): binary fraction, short-term
  - cumulative lost (24-bit): # of packets, long-term
  - <u>extended</u> highest sequence # received (<u>32</u>-bit)
  - inter-arrival jitter (32-bit)
  - LSR timestamp (<u>32</u>-bit): last received SR
  - delay since last SR (32-bit): in NTP ticks

Jitter and delay estimation

- One-way jitter
  - $V = (r_2 - r_1) - (s_2 - s_1) = (r_2 - s_2) - (r_1 - s_1)$
  - EWMA: $J = J + (V-J)/16$



- Round-trip delay
  - $R = (r_2 - s_1) - (s_2 - r_1)$
  - $(s_2 - r_1)$: DLSR
  - s_1: LSR



RTCP packets: more

- Source description (SDES)
  - CNAME (canonical name, user@host), NAME (real name), EMAIL, PHONE, LOC (location), TOOL, NOTE, PRIV (private extension)
- Explicit leave (BYE)
  - optional string: reason for leaving
- Application-specific (APP)
  - application-specific extensions

RTCP bandwidth scaling

- Goal: limited control overhead
  - RTCP counts for 5% session bandwidth
    - sender RTCP counts for 25%
    - receiver RTCP counts for 75%
- Approach: adjust RTCP interval adaptively
  - scale RTCP inter-packet time according to
    - amount of data traffic
    - # of senders and receivers

RTP header compression

- IP+UDP+RTP: 40 bytes
  - G.729 (8Kbps) 30ms packet: 30 bytes audio
- Compressed RTP header: as few as 2 bytes
  - derived from Jacobson's TCP/IP header compression
  - IP header: only IP ID changes between packets
  - UDP: only checksum changes
  - RTP: timestamp and sequence # often change regularly
- Approach: send "full" first,then only "delta"

This lecture

- RTP/RTCP
  - Why RTP
  - What RTP offers
    - RTP header fields, entities and profiles
  - What RTCP offers
    - sender report and receiver report
- Probe further
  - http://www.cs.columbia.edu/~hgs/rtp/

Next lecture

- SIP
  - http://www.cs.columbia.edu/~hgs/sip/