

CSc 461/561

Multimedia Systems

RTP/RTCP

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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

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What RTP offers?

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

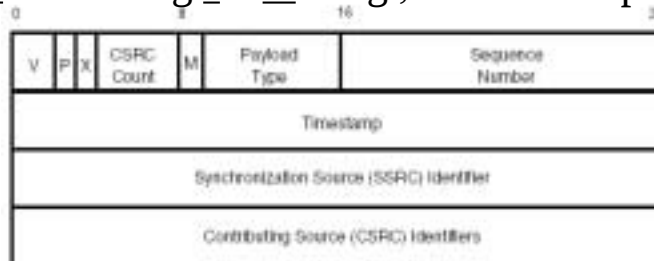
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RTP header

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



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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

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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

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RTP profiles

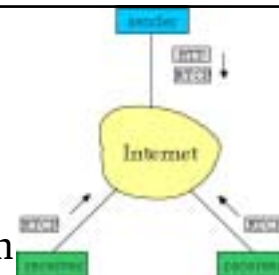
- Media specific (e.g., audio)
 - Marker: e.g., the start of a talk spurt
 - Payload Type: e.g., specific audio codec
 - PT=0: uPCM 64Kbpz; PT=3: GSM 13Kbps
 - timestamp: e.g., sampling rate, 8KHz PCM
 - packet size: e.g., about 20ms samples in PCM
 - packets independent as much as possible: ALF
 - other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

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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)

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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

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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
 - extended highest sequence # received (32-bit)
 - inter-arrival jitter (32-bit)
 - LSR timestamp (32-bit): last received SR
 - delay since last SR (32-bit): in NTP ticks

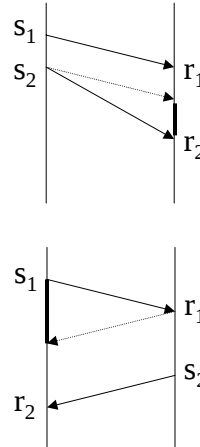
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Jitter and delay estimation

- 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



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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

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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

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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”

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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/>

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Next lecture

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

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