# RTP / RTCP

**Internet Protocols**

*CSC / ECE 573*

*Fall, 2005*

*N. C. State University*

## *Announcements*

*I. Final Exam study guide online*

*II. Signup for project demos*

*III. Teaching evaluations at end today*



## Today's Lecture

I. RTP

II. RTCP

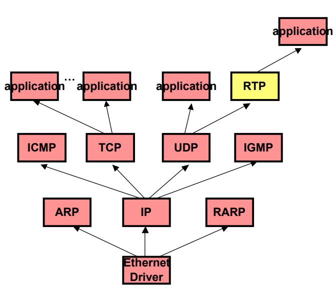
III. Header Compression



## RTP (Real-Time Protocol)

## RTP Info

- RTP adds a new header to each packet






## RTP (RFC 3550)

- TCP unacceptable for voice and video
  - why?
- Preferable: use UDP for transport
  - but UDP is connectionless, no in-order delivery, no detection of losses
  - RTP adds: Payload Type, Sequence #, Timestamp
- RTP does not…
  - guarantee reliable delivery of packets
  - guarantee QoS

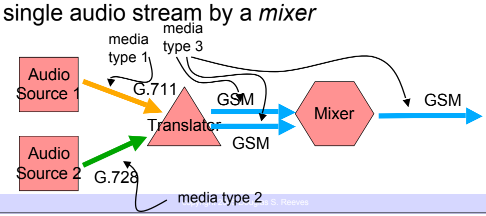
## RTP Functions

1. Payload Type: identification of media stream
2. Timestamp
   - timing reconstruction (for control of playback)
   - synchronization of different media types (e.g., audio + video)
3. Sequence #
   - packet sequencing
   - loss detection (and rate adaptation)




## Translators and Mixers

- Participants in a multimedia session may use different media formats or compression standards
  – a *translator* (also called a *media gateway*) converts from one media format to another
- Audio from multiple senders can be mixed into a single audio stream by a *mixer*






## RTP Header Contents

- Appears in every data packet, minimum of 12 bytes
  – Byte 1: Version, Padding Flag, Extension Flag, # of Sources (max of 15)
  – Byte 2: Application Marker Flag, Payload Type (7 bits)
  – Bytes 3-4: Sequence Number
  – Bytes 5-8: Timestamp (format is application-specific)
  – Bytes 9-12: SSRC (Synchronization Source Identifier) = random #, not IP address, identifies a single media stream




## RTP Ports and Profiles

- RFC 1990 specifies *A/V Profiles*
  – default Payload Types for common audio and video compression standards (currently: 24 types)
- Contains
  – mapping of media encoding to payload format
  – default packet rate
- Allocate a port pair for RTP and RTCP
  – e.g., 5004 for RTP, 5005 for RTCP
  – each media stream (voice, video, etc.) normally has own RTP connection




## RTP Timestamps

- Initial value is random number (i.e., absolute timing not used)
- Timing rate or resolution is payload-dependent, specified as part of the A/V profile
  – resolution must be ≥ sampling rate of the media type (e.g., 8KHz for compressed voice)
- Timestamp in an RTP packet is time of the first sample in the frame




## RTCP (Real-Time Control Protocol)

## RTCP Purposes

- Used for negotiation between senders and receivers
- Reports the quality of the connection between sender and receivers



## Types of RTCP Messages (Partial)

1. Source Description – useful information about the source data
2. Sender Report – report current time and amount of data sent so far
3. Receiver Report – feedback to sender about what has been received so far
4. Bye – source is disconnecting
5. Application-specific



## #1: Source Description Message

- Contains up to 31 Source Descriptors
- Each Source Descriptor contains
  - Synchronization Source ID (32 bits)
  - some number of TLV (tag, length, value)-encoded fields
  - Examples of standardized fields
    - Name of originator
    - Email address
    - phone number
    - Location
    - …



## #2: Receiver Report Message

- There is one Source Report Block for each sender about which the receiver is reporting. Contents as follows:

1. SSRC (32 bits)
2. Fraction of blocks lost since last report (out of 255)
   - 8 bits, i.e., 13 = 13/255 = 5% lost
3. Cumulative number of packets lost (24 bits)

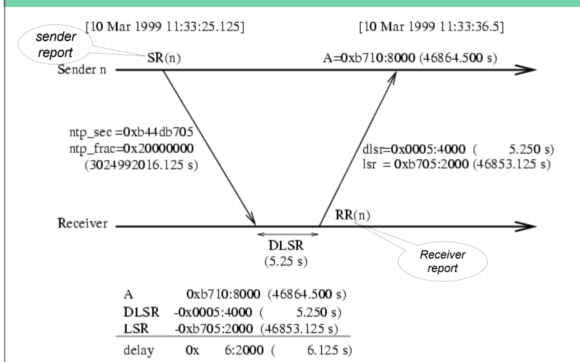


## Receiver Report Message (con'td)

4. Highest sequence number received (32 bits)
   - RTP sequence numbers wrap around after $2^{16}$-1
   - this is "extended sequence number"
5. Interarrival Jitter Estimate (32 bits)
6. LSR: Time of last Sender Report (32 bits)
   - in NTP format, just the middle 4 bytes
7. DLSR: Delay since last Sender Report (32 bits)
   - in units of 1/65536 seconds



## Computing Round-Trip Delay



## RTP Jitter Calculations

- *Jitter*: statistical variance of the RTP data interarrival times
- Symbols for calculations
  - **s** is the source
  - **r** is the receiver
  - **rr** is receiver report
  - jitter estimate is a floating point number
  - **rr->jitter** field of the receiver report is integer approximation




## RTP Jitter Calculations (cont'd)

- Inputs
  - **r->ts** = timestamp from the incoming packet
  - **arrival** = the current time (in the same units)
- Outputs
  - **s->transit** = the transit time for the previous packet
  - **s->jitter** = the estimated jitter

```
int transit = arrival - r->ts;
int d = | transit - s->transit |;
s->transit = transit;
s->jitter += (1./16.) * ((double)d - s->jitter);
rr->jitter = (u_int32) s->jitter;
```




## Example

- Initially, s->transit = 0, s->jitter = 0)

arrival = 1000, r->ts = 700
transit = 300
d = 300 – 0 = 300
s->transit = 300
s->jitter = 0 + 1/16*(300-0) = 18.75
rr->jitter = 18

← *first packet*

*next packet* →

arrival=1800,r->ts=1700
transit = 100
d = 200
s->transit = 100
s->jitter = 18.75 + 1/16*(200 – 18.75) = 30.01
rr->jitter = 30

## Sender Report Message

- Sender Source ID (SSRC)
- NTP Timestamp
  - standard representation of time of day, 8 bytes long
  - (better than) nanosecond resolution
- RTP Timestamp (application specific) (32 bits)
- Number of packets sent so far (32 bits)
- Number of bytes sent so far (32 bits)




## Additional RTCP Messages…

- MsgType 4: Bye message
  - SSRCs of sources leaving
  - optionally: "reason for leaving"
- MsgType 5: Application-Specific message
  - makes RTCP easily extensible




## Moderating RTCP Traffic Volume

- RTCP traffic volume is carefully controlled to prevent excessive overhead
- RTCP traffic designed to be no more than 5% of the media traffic (RTP)
  - 1.25% allocated to senders, and 3.75% allocated to receivers
  - as number of receivers increases, frequency of response per receiver decreases
- Minimum packet transmission frequency is 5 seconds

## HEADER COMPRESSION

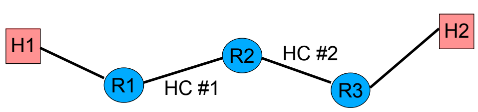
## Header Compression (RFC 2507)

- Every voice packet may have following overhead
  - IP = 20 bytes
  - UDP = 8 bytes
  - RTP = 12 bytes
  - = 40 bytes per packet!
- With 20 bytes of payload, 66% of the packet is overhead!
  - this is particularly a problem with slow links (e.g., access network)




## Header Compression (cont'd)

- Q. Why not put more payload per packet?
  - A: larger payload = more packetization delay
- Goal: reduce packet overhead, but make it transparent to the endpoints
  - *solution*: compress at the link layer







## Header Compression Principles

- Many of the fields in the headers do not change (immutable)
  - same values in packets *i*, *i*+1, *i*+2, …
  - transmit value only with first packet
  - e.g., "value always = 4"
- Many of the fields change by a constant increment between packets, and are therefore predictable
  - transmit with the first packet, + increment value to use
  - e.g., "first value = 3000, add 80 for each additional packet"




## Header Compression Principles (cont'd)

- Many field values change by only small amount
  - use differential coding on these fields
  - e.g., "use value from previous packet + 2"
- Result: substitute for the normal (IP+UDP+RTP) header…
  - session index #
  - differences, for differentially-coded fields




## HC: Identifying Each RTP Session

- Identifying a "session context"
  - IP source and destination addresses
  - UDP source and destination ports
  - RTP synchronization source ID (SSRC)
- Hash this session context to a unique session index #

## Mutable or Predictable Fields?

**IPv4**

- **Version (4 bits)**
- **Header Length x4 (4)**
- **Type of Service (8)**
- **Total Length (16)**
- **Identification (16)**
- **Flags (3)**
- **Fragment Offset x8 (13)**
- **Time-to-Live (8)**
- **Protocol Identifier (8)**
- **Header Checksum (16)**
- **Source IP Address (32)**
- **Destination IP Address (32)**
- **IP Options (≤ 320)**

**UDP**

- Source Port
- Destination Port
- Payload Length
- Checksum

**RTP**

- Version, Flags, #Sources
- Marker Flag, Payload Type
- Sequence #
- Timestamp
- SSRC




## HC: Results

- Amount of compression (typical): from 40 down to 2-4 bytes
- Success rate of prediction (typical): 95-98%




## Summary

- RTP and RTCP are widely used for voice and video traffic
- Provide a number of capabilities, mainly synchronization, QoS adaptation, and playback control
- Overhead per packet may justify header compression




## Next Lecture

- QoS