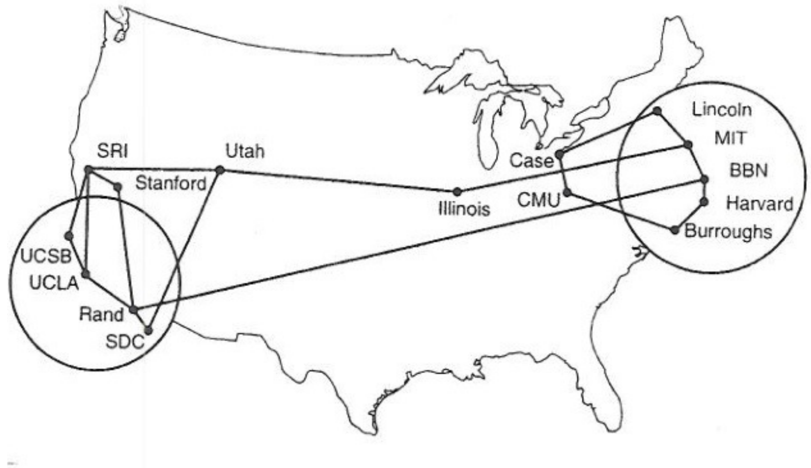


Real-time Protocol

Outline

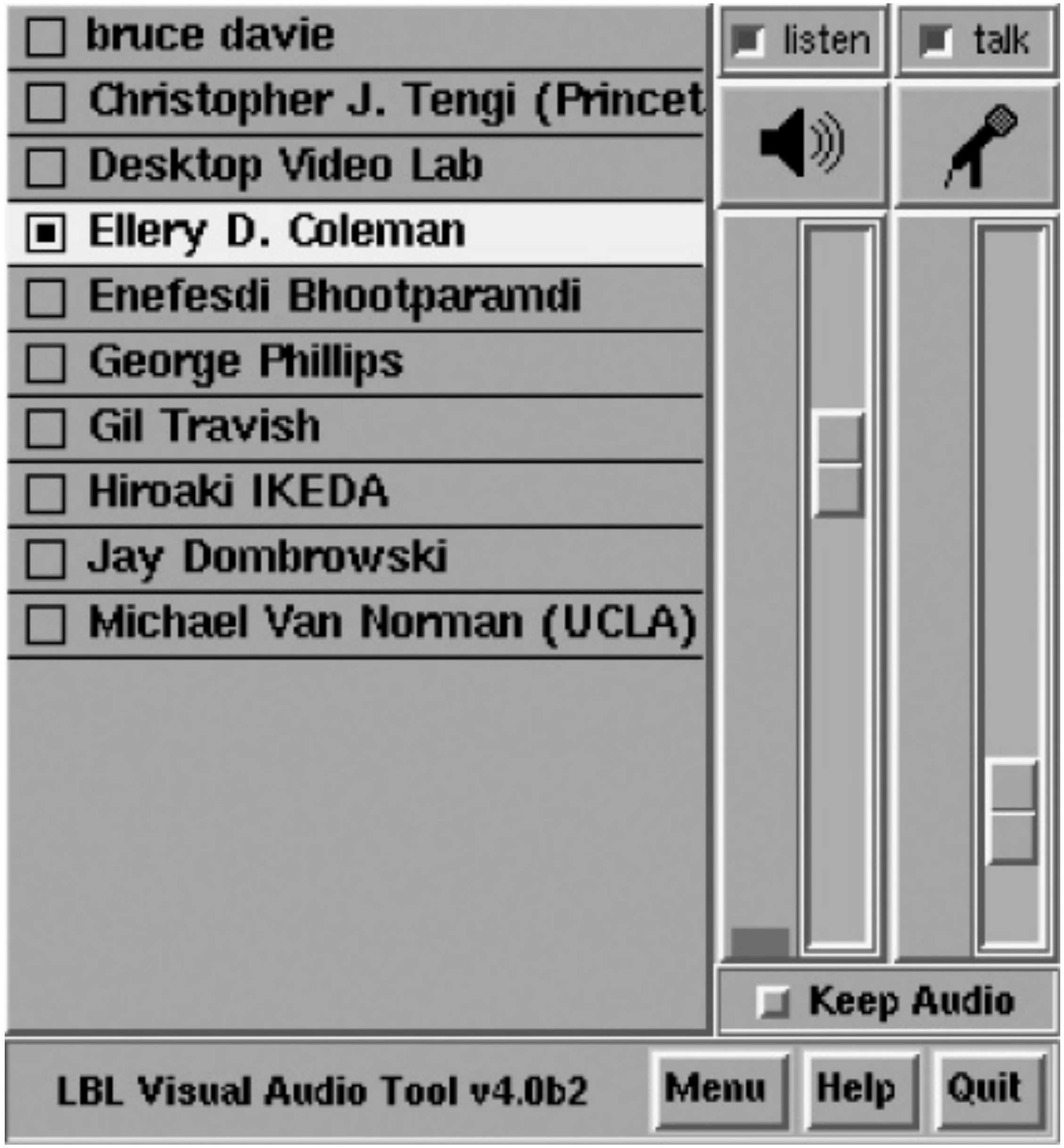
- RTP
- RTCP
- Readings 5.4
- Why? - New Apps, New Protocols

Why RTP ?

- Real-time means strong requirements for ‘timely’ delivery
- Internet - Not originally designed for Real-time
- Experiments as early as 1981

Real-time Requirements

- Synchronization of multimedia audio/video
- Allow similar apps to interoperate, e.g., 2 different conferencing apps
  - Need same encoding, compression, etc.
- Provide way for sender to tell receiver what scheme to use for voice, video, etc.
- Determine timing relationship among received data: jitter, timestamping
- Indicate packet loss - no recovery as in TCP
  - Allows sender to adjust to packet loss
- Indicate frame boundary
  - Application specific, e.g., talkspurt in audio
- Identity of senders beyond IP address
- Be efficient - short header
  - allows fast audio packet processing


bruce davie
Christopher J. Tengi (Princet
Desktop Video Lab
Ellery D. Coleman
Enefesdi Bhootparamdi
George Phillips
Gil Travish
Hiroaki IKEDA
Jay Dombrowski
Michael Van Norman (UCLA)
listen
talk
Keep Audio
LBL Visual Audio Tool v4.0b2
Menu
Help
Quit

Multimedia Apps

- Video, Audio, Data
- Interactive or Streaming
- Interactive
  - Vat - audio conferencing
  - IP Telephone
- Steaming
  - Real Audio
  - Video

RTP Details

RTP - Real-time Transport Protocol

multimedia data

-maintain the quality of traffic for real-time apps

RTCP - Real-time Transport Control Protocol

-periodically send control info between end points about data flow behaviors

-Not used to set up or manage connections

-Not used to directly control the way data is exchanged

Use consecutive ports, RTP on even port

-server ports assigned, 5004 and 5005

ALF - Application Level Framing

New Applications, New Requirements, & Old Protocols

Application ‘best’ understands its needs

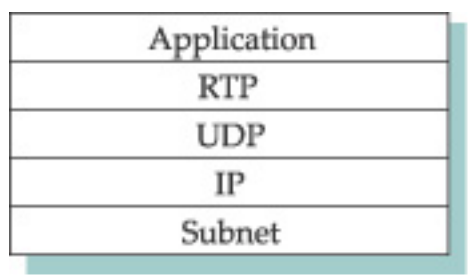
- e.g., how to recover frames
- e.g., how to segment data for transmission, etc.

RTP Stack

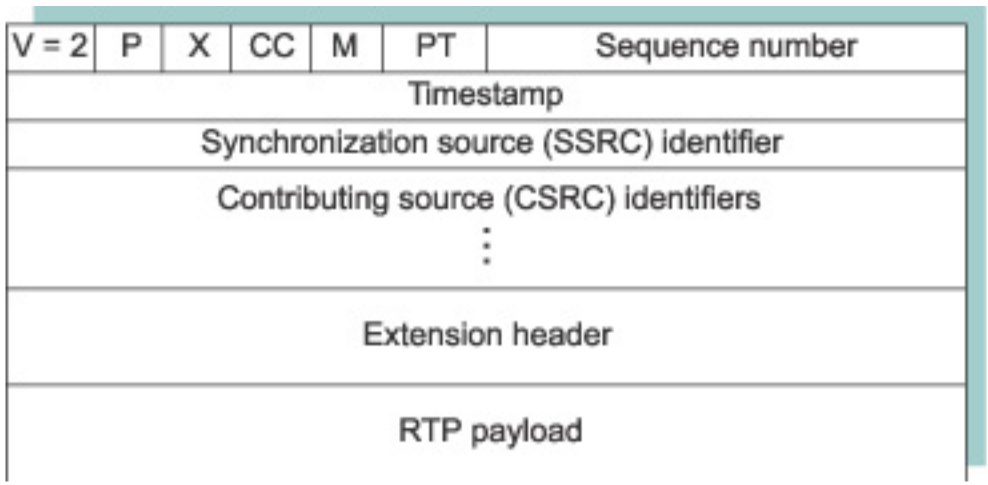
- Rides on UDP
- UDP provides minimal functionality
  - Ports & port demultiplexing
- Transport on a Transport
  - Each provides support for other
  - Think of transport 'extension headers'

Application on RTP

- Changing applications
- Application Classes
  - Profiles: provides info on understanding of RTP fields
  - Formats: how data following RTP header is to be interpreted, e.g., sequence of bytes or MPEG video



RTP Header



V = 2
P
X
CC
M
PT
Sequence number
Timestamp
Synchronization source (SSRC) identifier
Contributing source (CSRC) identifiers
Extension header
RTP payload

RTP Header

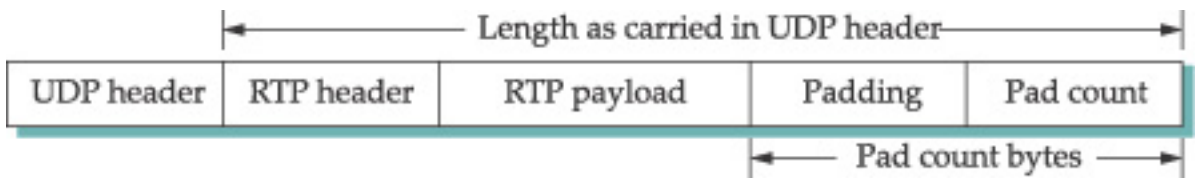
- First 12 bytes ALWAYS there
  - General app fields
- Optional header extensions
  - Specific app fields
- Version - 2 bits
- Padding - 1 bit
  - Padding count last byte of message
  - Message size known through UDP
  - No length in header
- Extensions - 1 bit
  - Yes or no

- Contributing sources - 4 bits
  Count of the number of contributing sources
- Frame Mark - 1 bit
  Yes/no, app specific
  Provide synchronization events such as end of frame markers.
- Payload Field - 2 bits
  App specific
  Controls how the RTP header is used and therefore RTCP messages

RTP Header

- Sequence # - 16 bits
  - Detect missing or misordered packets
  - Left to app to decide what to do
- Timestamp - 32 bits
  - Timestamp of 1st byte of payload data
  - Ticks, monotonically increasing
  - Difference between timestamps important, granularity for jitter issues
  - Left to app to use
- SSRC - Synchronization source - 32 bits
  - Identifies source of an RTP stream
  - E.g., Allows multiple camera streams, from same IP Address
- CSRC - Contributing Source Identifiers
  - Contributing source Ids
  - RTP stream mixer
  - Can take many streams & send as a single stream

UDP/RTP Messages



RTCP

- Control stream associated with data stream for a multimedia app.
- Feedback on performance of app and networ
  – Rate-adaptive apps
  – Diagnose network
- Synchronize different media streams from a sender
  – Different streams may have different clocks
- Convey identity of a sender for display

RTCP Messages

- Sender Reports
  - Enable active senders to report transmission and reception stats
- Receiver Reports
  - Receivers that are not senders use to report reception stats
- Source Descriptions
  - Carry CNAMEs and other sender description info
  - Mikeatcs.hmc.edu would be included with the SSRC
- Application Specific control packets
- Transmission
  - Several RTCP packets can be packed into UDP packet
  - Actually, at least 2 are always sent:
    - Report packeet
    - Source description packet
- Traffic control
  - Mechanisms for participants to scale back reporting frequency
  - Lots of rules, but basically, keep RTCP at about 5% of RTP traffic

RTCP Report Messages

- Sender Reports
  - Timestamp of time of day
  - RTP timestamp of report
  - Cumulative counts of packets and bytes sent since beginning
- Sender and Receiver Reports
  - Its SSRC
  - Fraction of data packets from this source lost since last report
  - Total number of packets lost since 1st transmission
  - Highest sequence number received from this source
  - Estimated interarrival jitter for the source
  - Last actual timestamp received via RTCP for this source
  - Delay since last sencder report received via RTCP for this source
- Lots of catch all info, probably added for certain apps

RTP & RTCP Summary

Trying to add transport functionality that is directed at a set of applications

At the same time trying to include all the specific info that any app within the set of applications may desire

BEEP is another such protocol

SCTP - Stream Control Transmission Protocol is another