Popular protocols for serving media

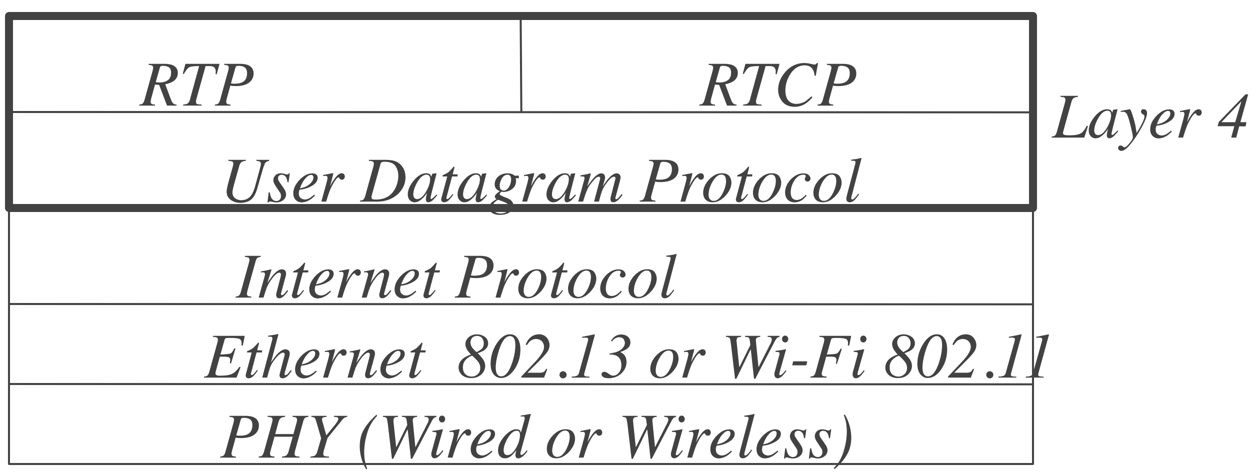
- Network transmission control
  - RTP – Realtime Transmission Protocol
    - RTCP – Realtime Transmission Control Protocol
- Session control
  - Real-Time Streaming Protocol (RTSP)
  - Session Description Protocol (SDP) – textual representation of sesion
- VOIP – SIP – Session Initiation Protocol
  - Signaling for IP Telephony
- SAP – Session announcement protocol for multicast sessions

RTP and RTSP

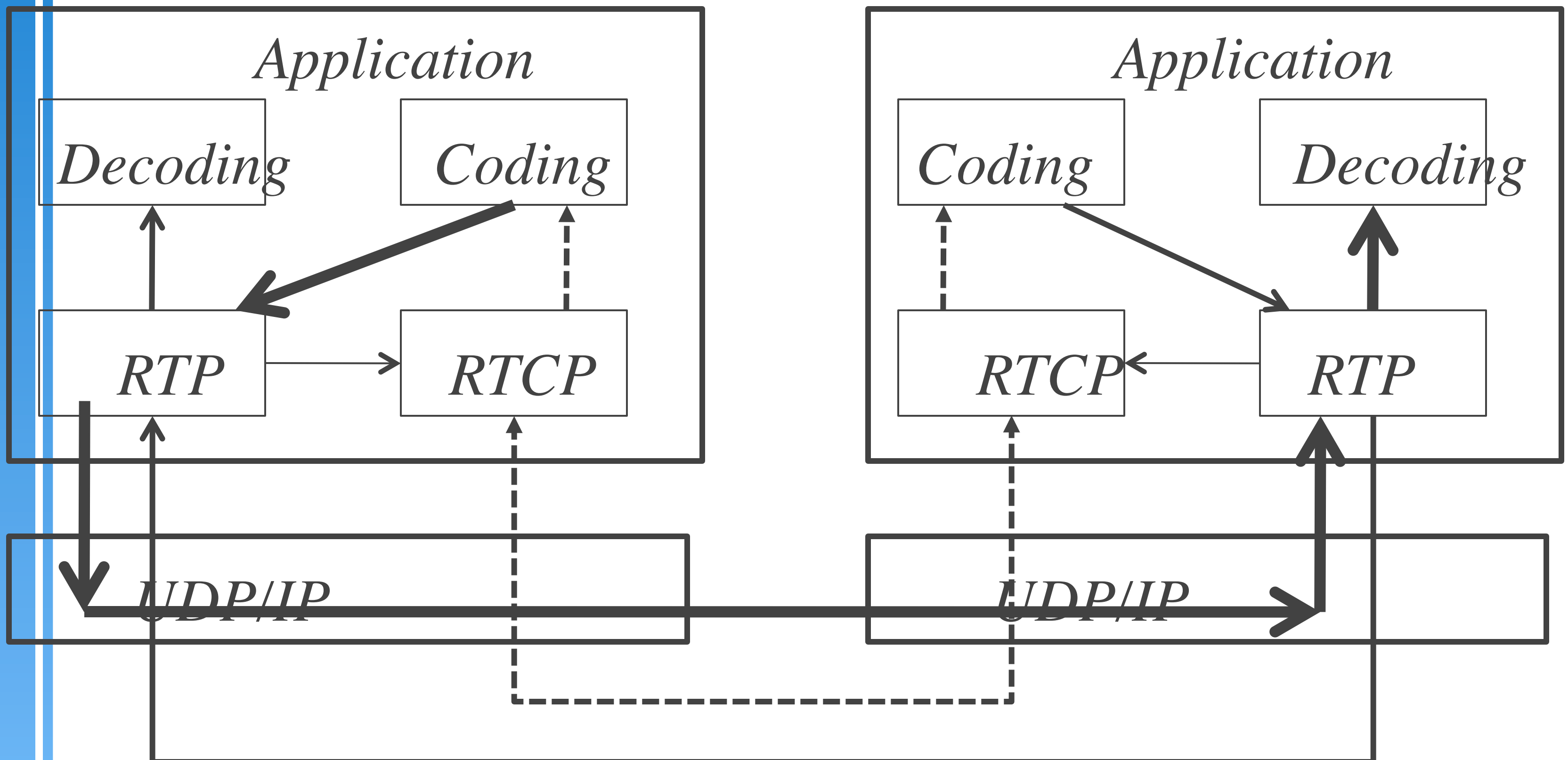
- RTP usage – in several application audio and video tools (vat, vic)
- RTP follows the principle of application level framing and integrated layer processing
- RTP/UDP/IP is being used by the current streaming session protocols such as RTSP
- Session protocols are actually negotiation/session establishment protocols that assist multimedia applications
- Multimedia applications such as QuickTime, Real Player and others use them

Real-time Transmission Protocol (RTP)

- RTP provides end-to-end transport functions suitable for real-time audio/video applications over multicast and unicast network services
- RTP companion protocol – Real-time Transport Control Protocol (RTCP)



Relation between RTP and RTCP



RTCP: Control and Management

- Out-of-band control information for RTP flow.
  - Monitors QoS for RTP in the delivery and packaging of multimedia data
  - Used periodically to transmit control packets to participants in a streaming multimedia session.
  - Provides feedback on the quality of service being provided by RTP
  - Gathers statistics on media connection
    - Bytes sent, packets sent, lost packets, jitter, feedback and round trip delay
    - Application may use this information to increase the quality of service, perhaps by limiting flow or using a different codec

RTCP Functions

- There are several type of RTCP packets:
  - Sender report packet,
  - Receiver report packet,
  - Source Description RTCP Packet,
  - Goodbye RTCP Packet and
  - Application Specific RTCP packets.
- RTCP itself does not provide any flow encryption or authentication means. SRTCP protocol can be used for that purpose.

RTP Services

- Payload Type Identification
  - Determination of media coding
  - Source identification
  - RTP works with Profiles
    - Profile defines a set of payload type codes and their mappings to payload formats
- Sequence numbering
  - Error detection
- Time-stamping
  - Time monitoring, synchronization, jitter calculation
- Delivery monitoring

RTP Services – Support of heterogeneity

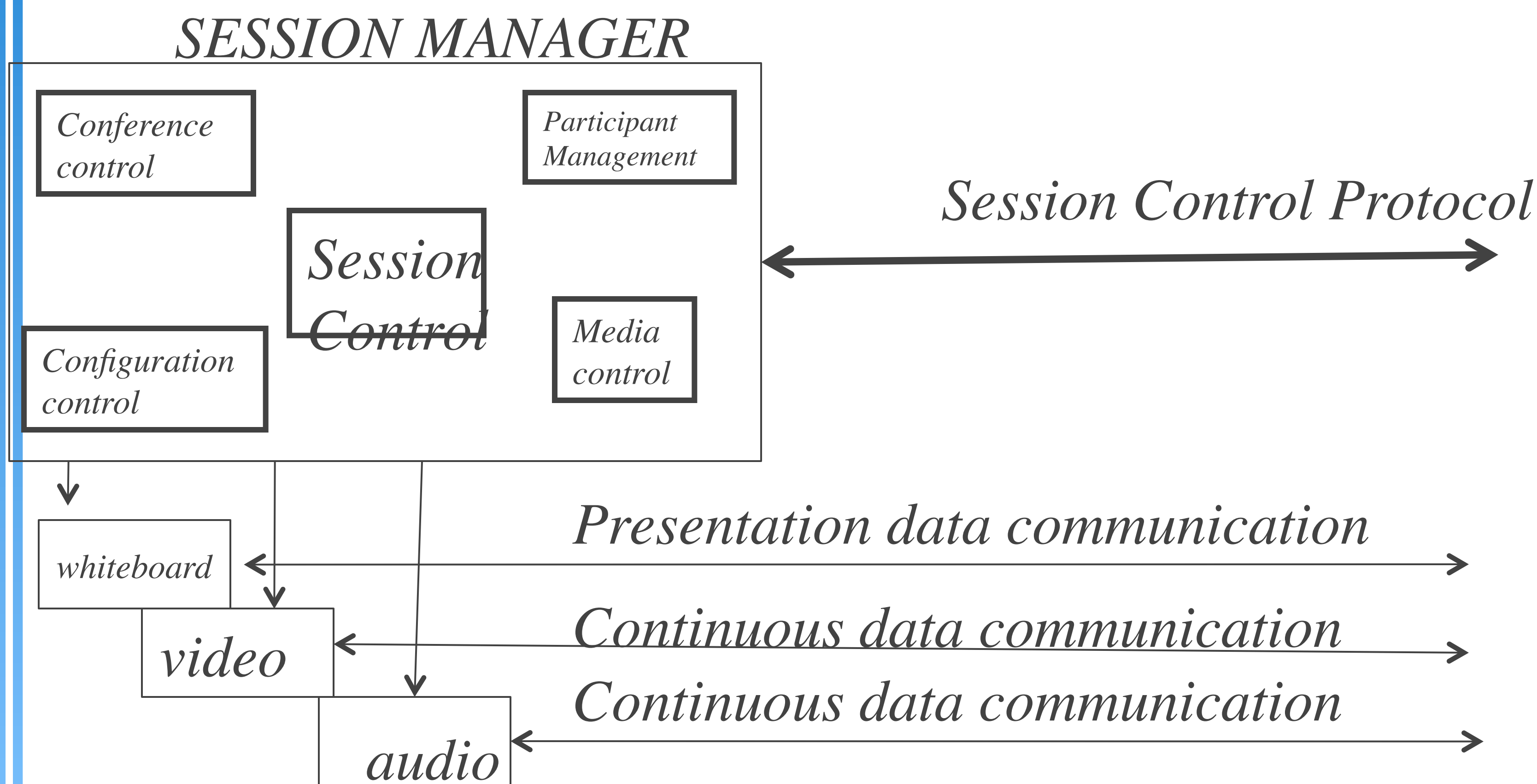
- Mixer service
  - Allows for resynchronization of incoming audio packets
  - Reconstructs constant 20 ms spacing generated by sender
  - Mixes reconstructed audio streams into single stream
  - Translated audio encoding to lower bandwidth
  - Forwards lower bandwidth packet streams
- Translator service
  - Allows for translation between IP and other high speed protocols
  - May change encoding data

Payload Formats

- Static Payload formats
  - Established in RTP Profile
  - Payload type 0 := μ-law audio codec
- Dynamic Payload formats
  - Applications agree per session on payload format
  - H.263, JPEG, MPEG

Session Management (Layer 5)

- Important part of multimedia communication
- Separates control aspects from transport aspects



Session Manager

- Tasks:
  - Membership control
  - Monitoring of shared workspace
  - Coordination of Media control management
  - Exchange of QoS parameters
  - Conference control management – establishment, modification, termination

Session Control

- Session Described by
  - Session state
    - Name of session, start, valid policies
- Session management – two steps for state processing
  - Establishment of session
  - Modification of session

Session Control

- Conference Control
  - Centralized or distributed approach
- Media Control
  - Synchronization
- Configuration Control
  - Negotiation of QoS parameters, admission control and reservation/allocation of resources
- Membership Control
  - Invitation of users; registration of users, change of membership

RTSP

- Enables controlled, on-demand delivery of real-time data such as audio and video
- Intends to control multiple data delivery sessions
- Provides means for choosing delivery channels
  - UDP
  - Multicast UDP,
  - TCP

Real-Time Streaming Protocol (RTSP)

- Application Protocol for control of multimedia streams
- This is not an application data transmission protocol, just remote control protocol between client and server

RTSP Methods

Request	Direction	Description
OPTIONS	S <-> C	Determine capabilities of server (S) or client (C)
DESCRIBE	C -> S	Get description of media stream
ANNOUNCE	S <-> C	Announce new session description
SETUP	C -> S	Create media session
RECORD	C -> S	Start media recording
PLAY	C -> S	Start media delivery
PAUSE	C -> S	Pause media delivery
REDIRECT	S -> C	Use other server
TEARDOWN	C -> S	Destroy media session
SET_PARAMETER	S <-> C	Set server or client parameter
GET_PARAMETER	S <-> C	Read server or client parameter

RTSP Extensions

- Timing
  - RTSP needs to hide latency variations
  - PLAY request may contain information about when request is to be executed
- Three types of timestamps
  - SMPTE (the same as in TV production)
    - Format: hours:minutes:seconds:frames
  - Normal play time
    - Measured relative to beginning of stream and expressed in ours, minutes, seconds and fractions of second
  - Absolute time
    - Wall clock

Session Description Protocol (SDP)

- Text format for describing multimedia sessions
- Not really a protocol (similar to markup language like HTML)
- Can be carried in any protocol, e.g., RTSP or SIP
- Describes unicast and multicast sessions

SDP

- There are five terms related to multimedia session description:
  - Conference: It is a set of two or more communicating users along with the software they are using.
  - Session : Session is the multimedia sender and receiver and the flowing stream of data.
  - Session Announcement: A session announcement is a mechanism by which a session description is conveyed to users in a proactive fashion, i.e., the session description was not explicitly requested by the user.
  - Session Advertisement : same as session announcement
  - Session Description : A well defined format for conveying sufficient information to discover and participate in a multimedia session.

Sample SDP file

```
v=0
o=- 19 1077294547 IN IP4 127.0.0.0
s=QuickTime
t=0 0
a=range:npt=now-
a=control:rtsp://127.0.0.1/mystream.sdp
a=isma-compliance:2,2.0,2
m=audio 0 RTP/AVP 96
c=IN IP4 0.0.0.0
b=AS:8
a=rtpmap:96 mpeg4-generic/8000/1
a=fmtp:96 profile-level-id=15;mode=AAC-
    hbr;sizelength=13;indexlength=3;indexdeltalength=3;config=1588
a=mpeg4-esid:101
m=video 0 RTP/AVP 97
c=IN IP4 0.0.0.0
b=AS:30
a=rtpmap:97 H264/90000
a=fmtp:97 packetization-mode=1;profile-level-id=4D400A;sprop-parameter-
    sets=J01ACqkYUI/LgDUGAQa2wrXvfAQ=,KN4JF6A=
a=mpeg4-esid:201
a=cliprect:0,0,120,160
a=framesize:97 160-120
```

VOIP


CISCO IP PHONE
7960 SERIES
21:32 08/09/29
105
Sonus
NETWORKS
Your current options
Redial
NewCall
CFwdALL
messages
directories
services
help
settings
VOLUME
HEADSET
MUTE
SPEAKER

Signaling for IP Telephony

- Internet Telephone – needs ability of one party to signal to other party to initiate a new call
- Call – association between a number of participants
  - Note: there is no physical channel or network resources associated with the session layer connection, the connection exists only as signaling state at two end points

IP Telephony Signaling Protocol (Requirements)

- Name translations and user location
  - Mapping between names of different levels of abstraction
    - Email address to IP address of host
- Feature negotiation
  - Group of end systems must agree on what media to exchange ad their respective parameters
    - Different encodings, rates
- Call Participant Management
    - Invite participants to existing call, transfer call and hold other users

IP Telephony Signaling (Requirements)

- Feature change
  - Adjust composition of media sessions during the course of call
    - Add or reduce functionality
    - Impose or remove constraints due to addition or removal of participants
- Two signaling protocols:
  - **SIP** (IETF Standard)
  - **H.323** (ITU Standard)

SIP (Session Initiation Protocol)

- SIP Goal: invite new participants to call
- Client-Server protocol at the application level
- Protocol:
  - User/Client creates requests and sends to server;
  - User agent server responds;
- SIP requests can traverse many proxy servers
- Server may act as redirect server
- Proxies or redirect servers cannot accept/reject requests, only user agent server can
- Requests/Responses are textual

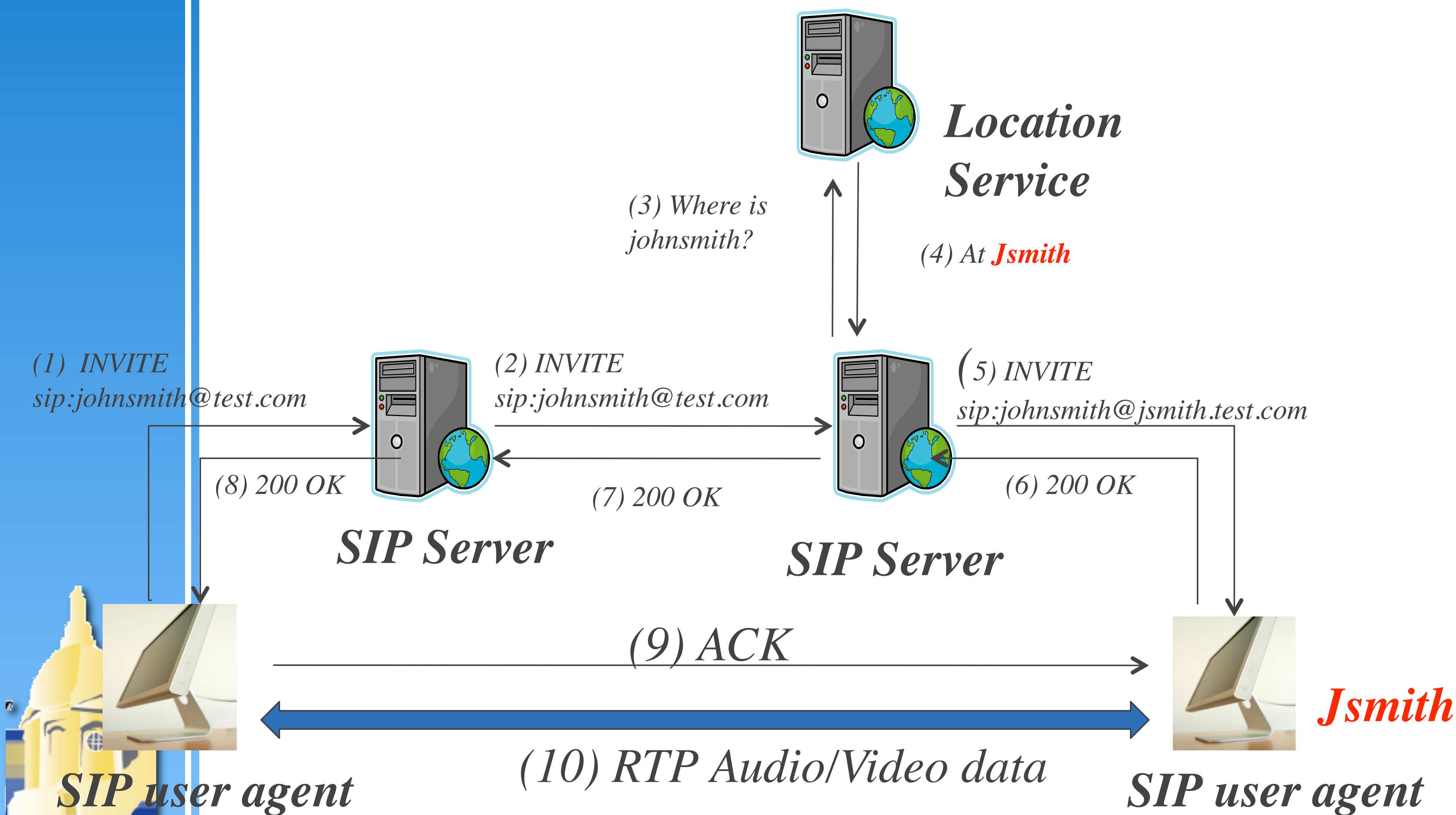
SIP - Message

- Calls in SIP – have unique call ID (carried in Call-ID header field of SIP message)
- Call identifier is created by the caller and used by all participants
- SIP messages have information
  - Logical connection source
  - Logical connection destination
  - Media destination
  - Media capabilities (use SDP)

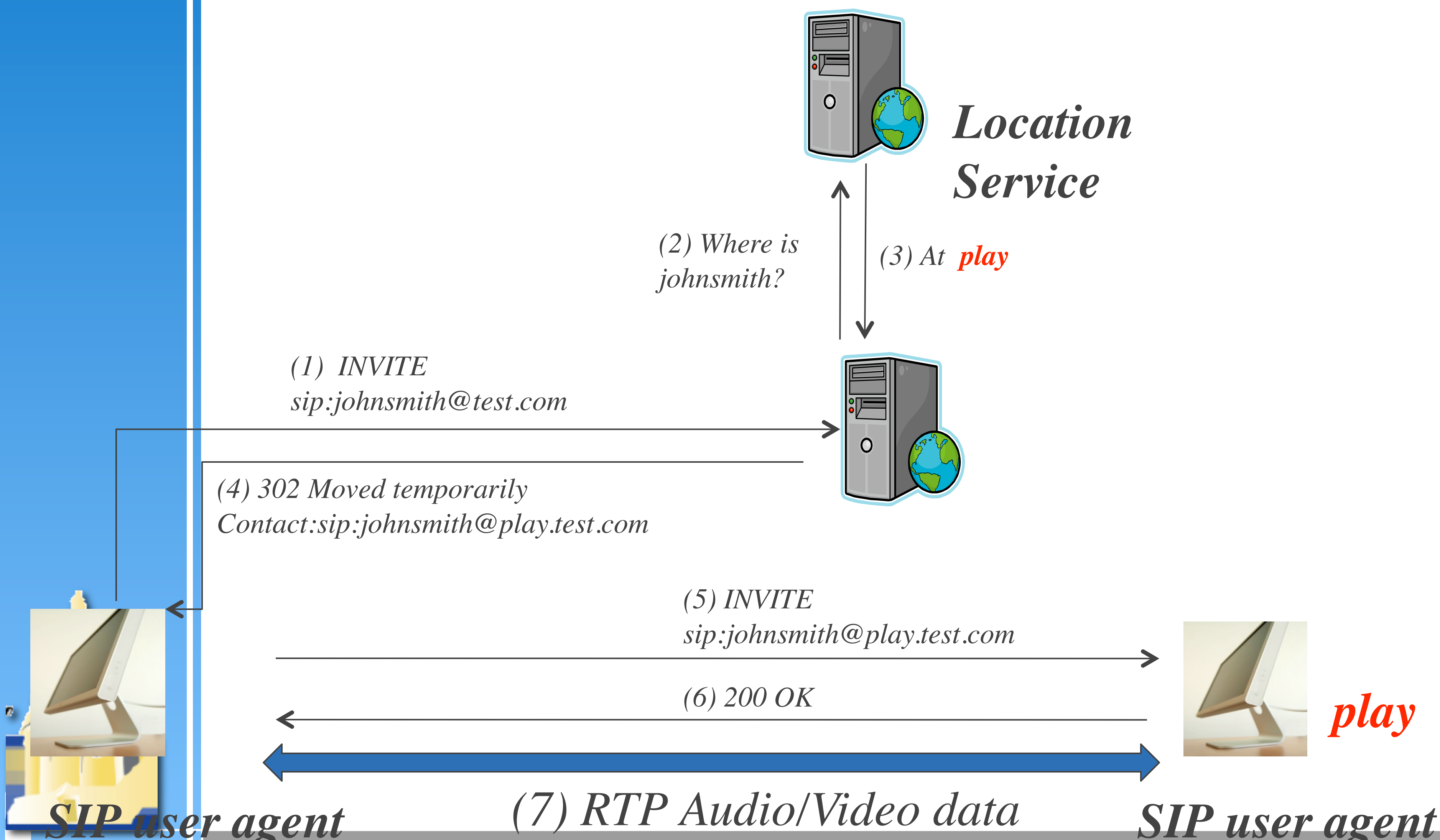
SIP – Addressing and Naming

- To be invited and identified, called party must be named
- SIP chooses email-like identifier
  - user@domain
  - user@host
  - user@IPaddress
  - phone-number@gateway
- SIP's address: part of SIP URL
  - sip:j.doe@example.com
  - URL can be placed on web page
- Interactive audio/video requests translation
  - name@domain to host@host

Call Setup Process using SIP



SIP Redirect Server Operation



SIP Requests/Methods

- INVITE—Indicates a client is being invited to participate in a call session.
- ACK—Confirms that the client has received a final response to an INVITE request.
- BYE—Terminates a call and can be sent by either the caller or the callee.
- CANCEL—Cancels any pending searches but does not terminate a call that has already been accepted.
- OPTIONS—Queries the capabilities of servers.
- REGISTER—Registers the address listed in the To header field with a SIP server.

SAP – Session Announcement Protocol

- RTSP and SIP are designed for one-on-one session
- SAP is multicast announcement protocol
- Protocol
  - Distributed servers periodically send multicast packets (advertisements) containing descriptions of sessions generated by local sources
  - Advertisements are received by multicast receivers on well-known , static multicast address/port
- Advertisement contains SDP information to start media tools needed in the session