CS 414 – Multimedia Systems Design
Lecture 25 - Case Studies for Multimedia Network Support (Layer 5-7)

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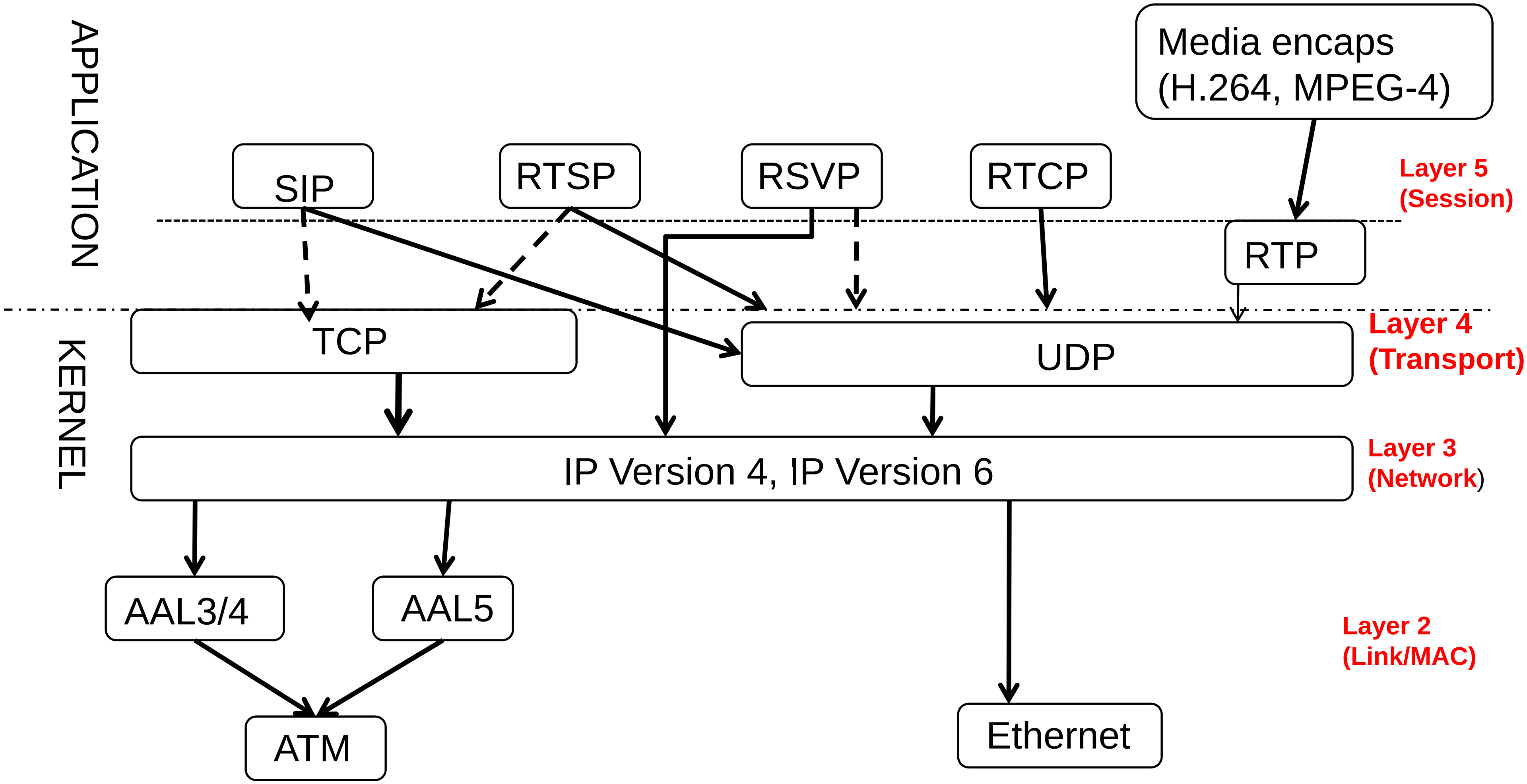
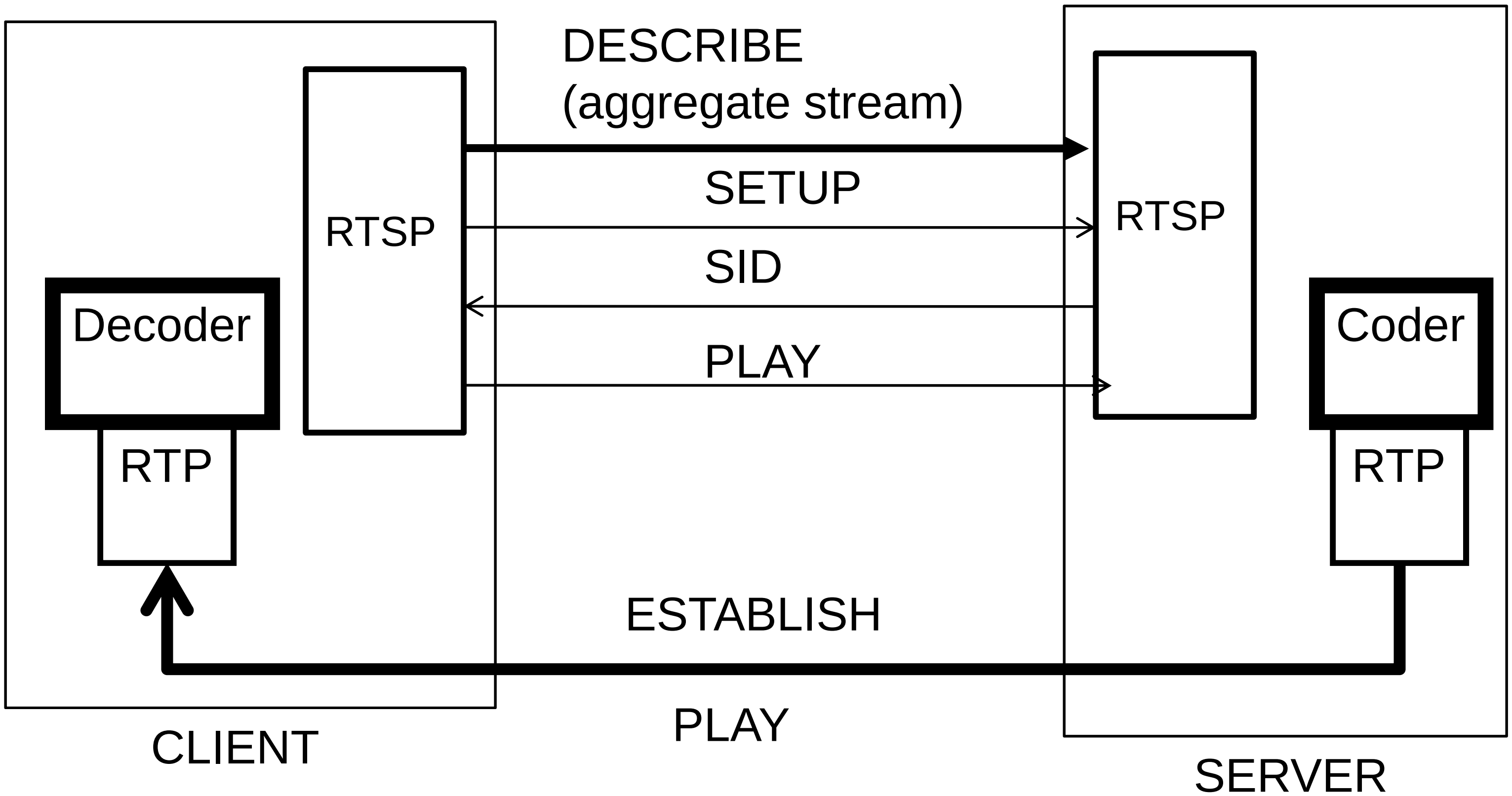
Administrative

- Re-grading of Midterm and HW1 until March 14
- MP3 is out – deadline April 4

Internet Multimedia Protocol Stack



RTSP – Aggregate and Stream Control



RTSP Extension - Caching

- Offers cache control
- Indicates whether intermediate caching proxies are allowed to deliver cached material rather than forwarding RTSP request to origin server
- Indicates via responses whether proxies are allowed to cache media streams

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.

SDP Information

- **Session description**
  - v= (protocol version) ; o= (originator and session identifier)
  - s= (session name) ; i=* (session information)
  - u=* (URI of description) ; e=* (email address) p
  - =* (phone number) ; c=* (connection information -- not required if included in all media) ; b=* (zero or more bandwidth information lines)
  - **One or more time descriptions**
    - ("t=" and "r=" lines; see below) z=* (time zone adjustments) k=* (encryption key) a=* (zero or more session attribute lines)
- **Time description**
  - t= (time the session is active) ; r=* (zero or more repeat times)
- **Media description, if present**
  - m= (media name and transport address) ; i=* (media title)
  - c=* (connection information -- optional if included at session level)
  - b=* (zero or more bandwidth information lines)
  - k=* (encryption key) a=* (zero or more media attribute lines)

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

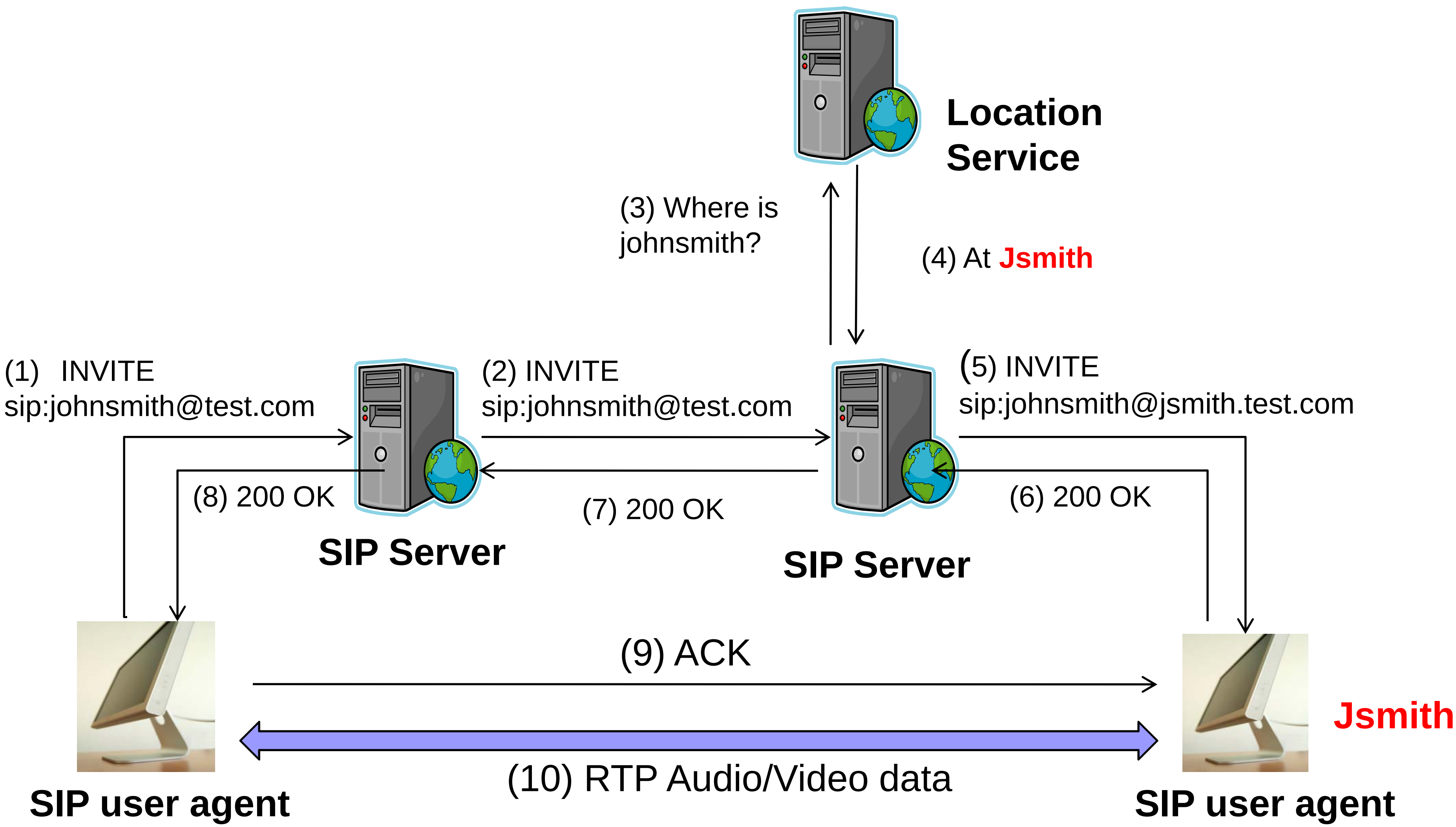
SIP - Message

- 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)

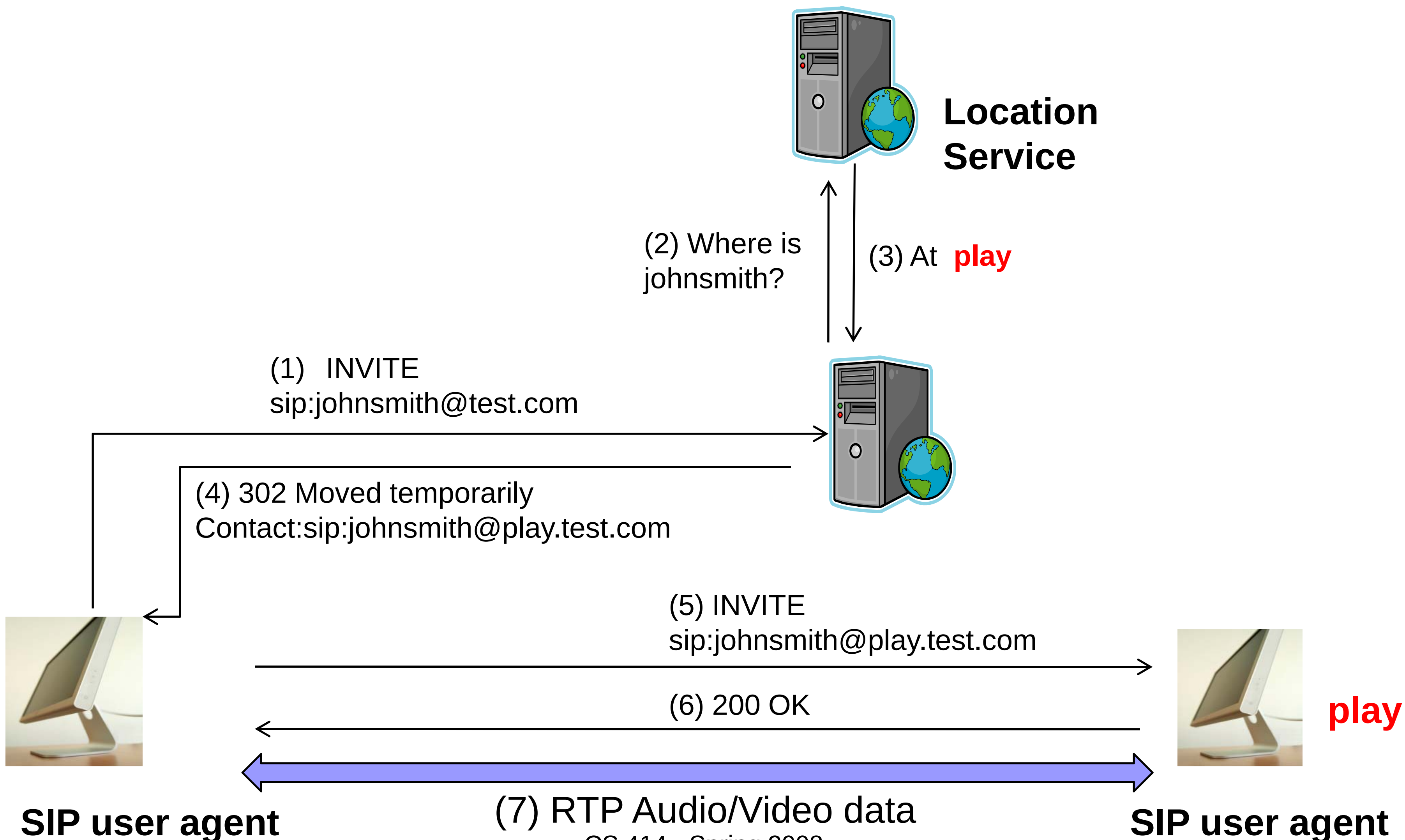
SIP – Addressing and Naming

- 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.

SIP Responses

- **1xx—Informational Responses**
  - 100 Trying (extended search being performed may take a significant time so a forking proxy must send a 100 Trying response)
  - 180 Ringing
  - 181 Call Is Being Forwarded
  - 182 Queued
  - 183 Session Progress
- **2xx—Successful Responses**
  - 200 OK
  - 202 accepted: It Indicates that the request has been understood but actually can't be processed
- **3xx—Redirection Responses**
  - 300 Multiple Choices
  - 301 Moved Permanently
  - 302 Moved Temporarily

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

Conclusion

- Internet protocol suite has now basic ingredients to support streaming audio and video
  - Both for distribution and communication applications
- Challenges:
  - **No session control protocol that can be used to perform floor control in distributed multimedia conferences**
  - **Network reliability and deployment multicast of services with predictable quality-of-service are major hurdles beyond need for continuous upgrades in network capacity**