

CS 414 – Multimedia Systems Design
Lecture 24 -
Case Studies for Multimedia
Network Support (Layer 4-5)

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Administrative

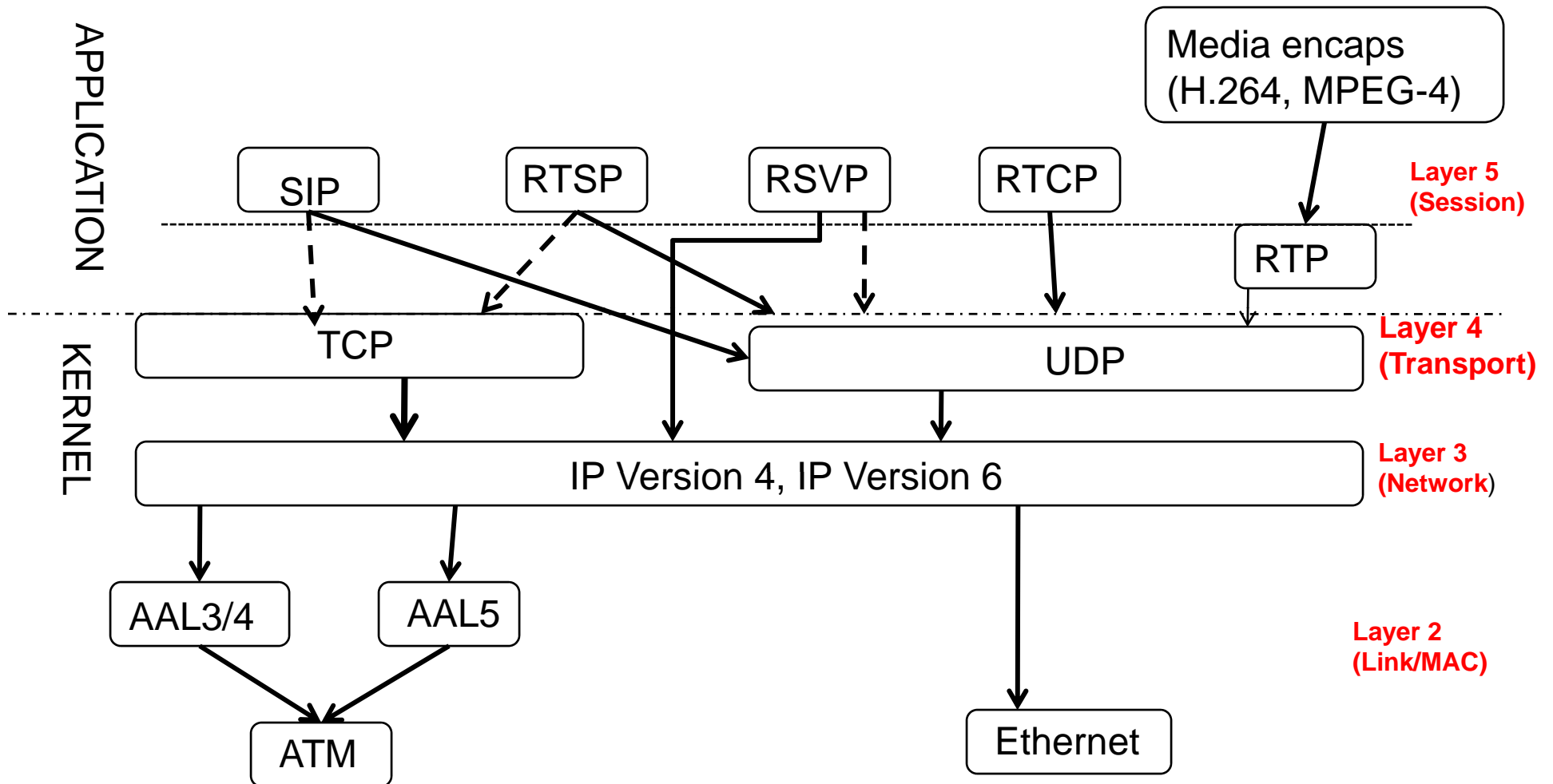
- Re-grading of Midterm and HW1 until March 14
- MP3 is out – deadline April 4
 - Discussion Section, Wednesday, March 12 at 6pm in 3405



User Datagram Protocol

- UDP is extension of IP
- UDP supports multiplexing of datagrams
- Check-summing
- Higher level protocols using UDP must provide
 - Retransmission
 - Segmentation and reassembly
 - Flow control, congestion avoidance

Internet Multimedia Protocol Stack



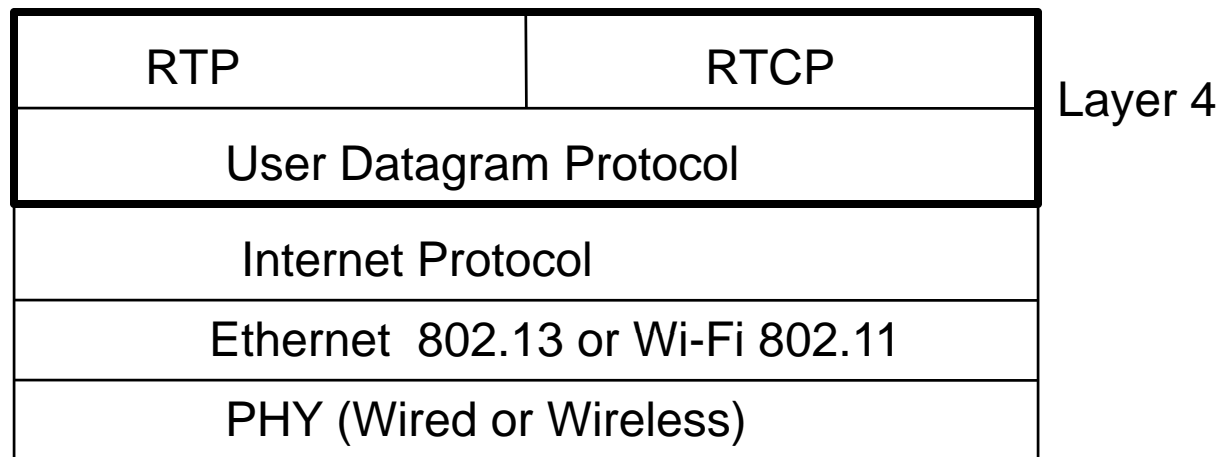


Service Requirements for Real-time Flows (Voice/Video)

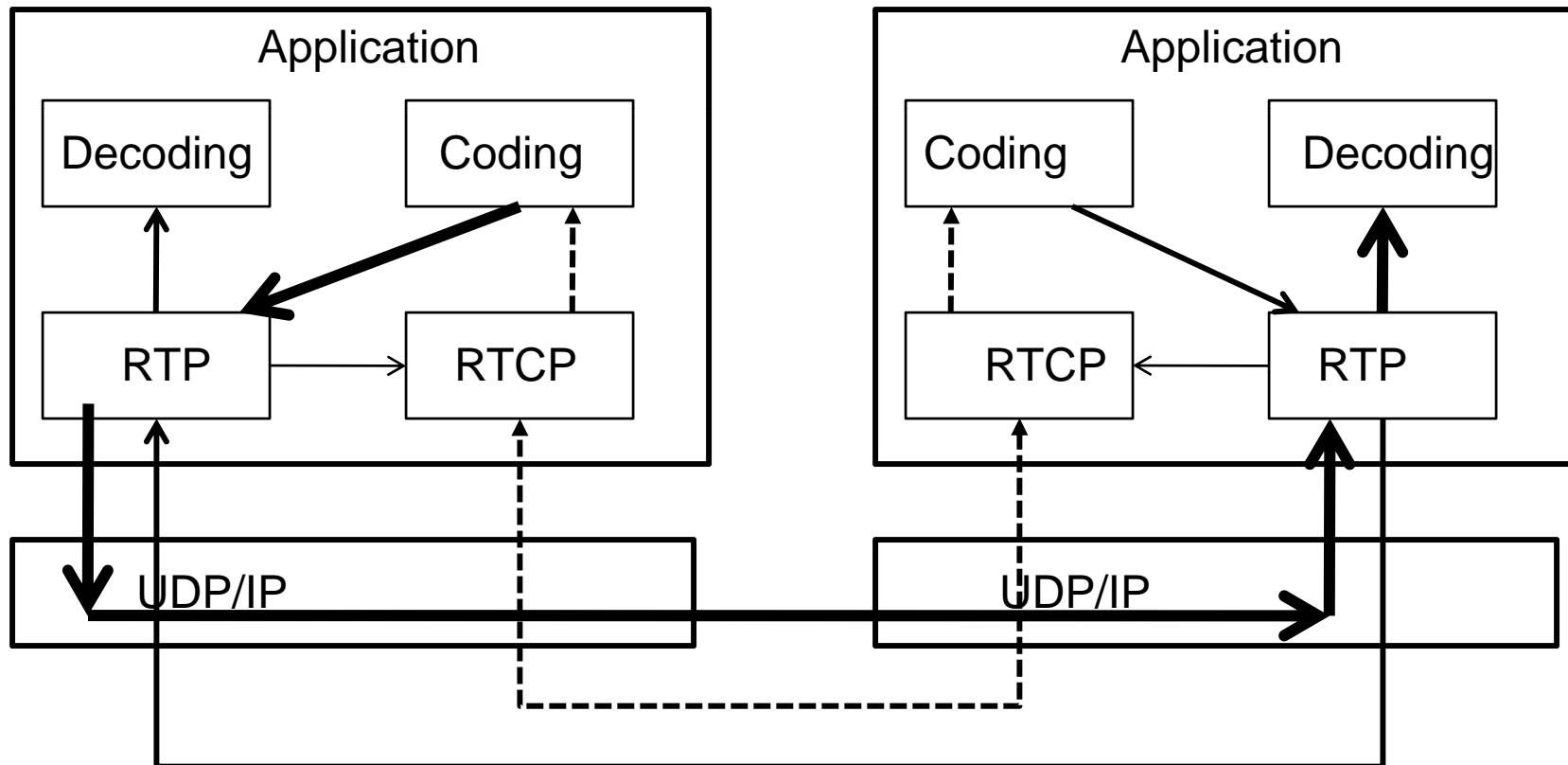
- Sequencing
- Intra-media synchronization
- Inter-media synchronization
- Payload identification
- Frame indication

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

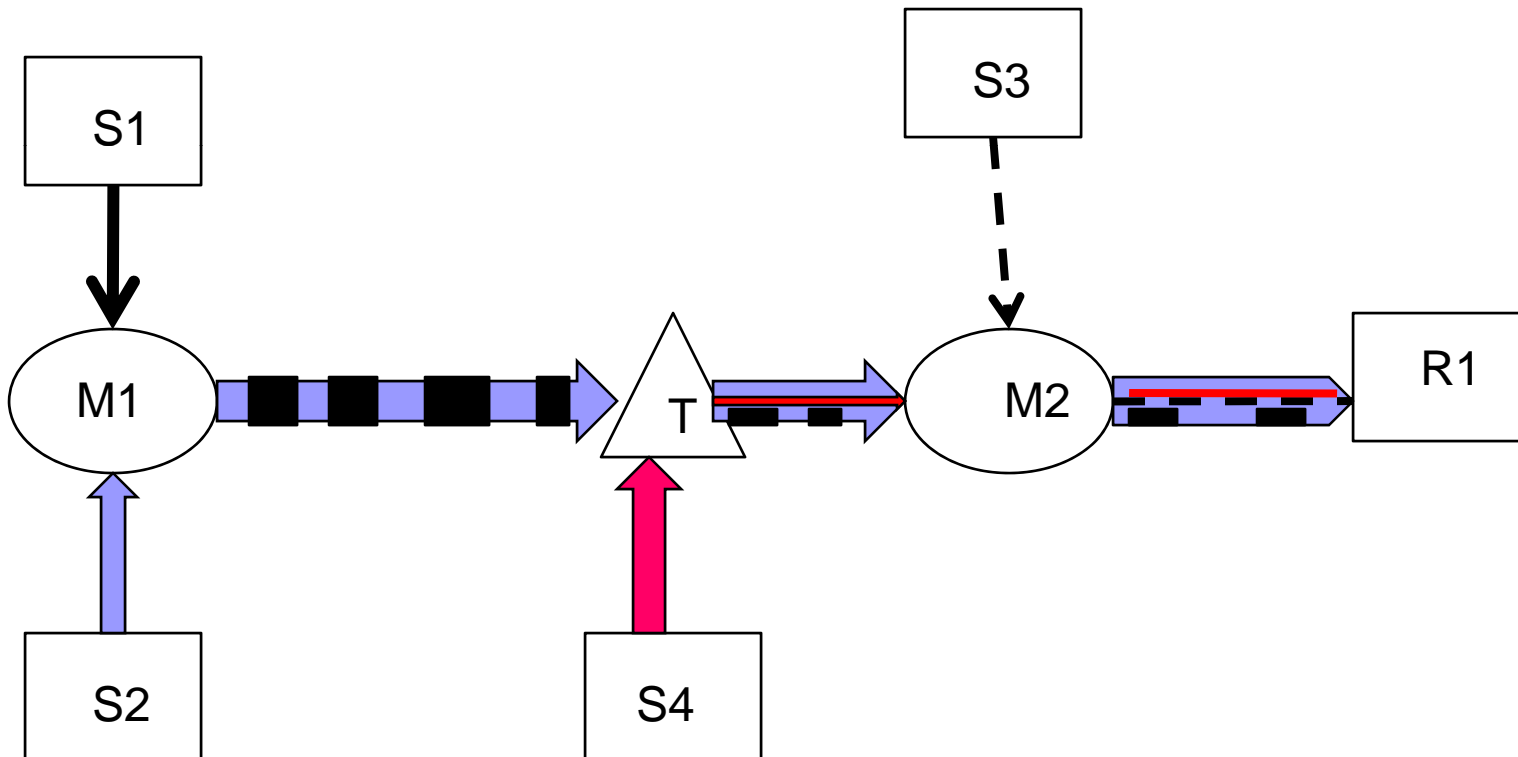
■ Mixer service

- 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

Difference between Mixers and Translators





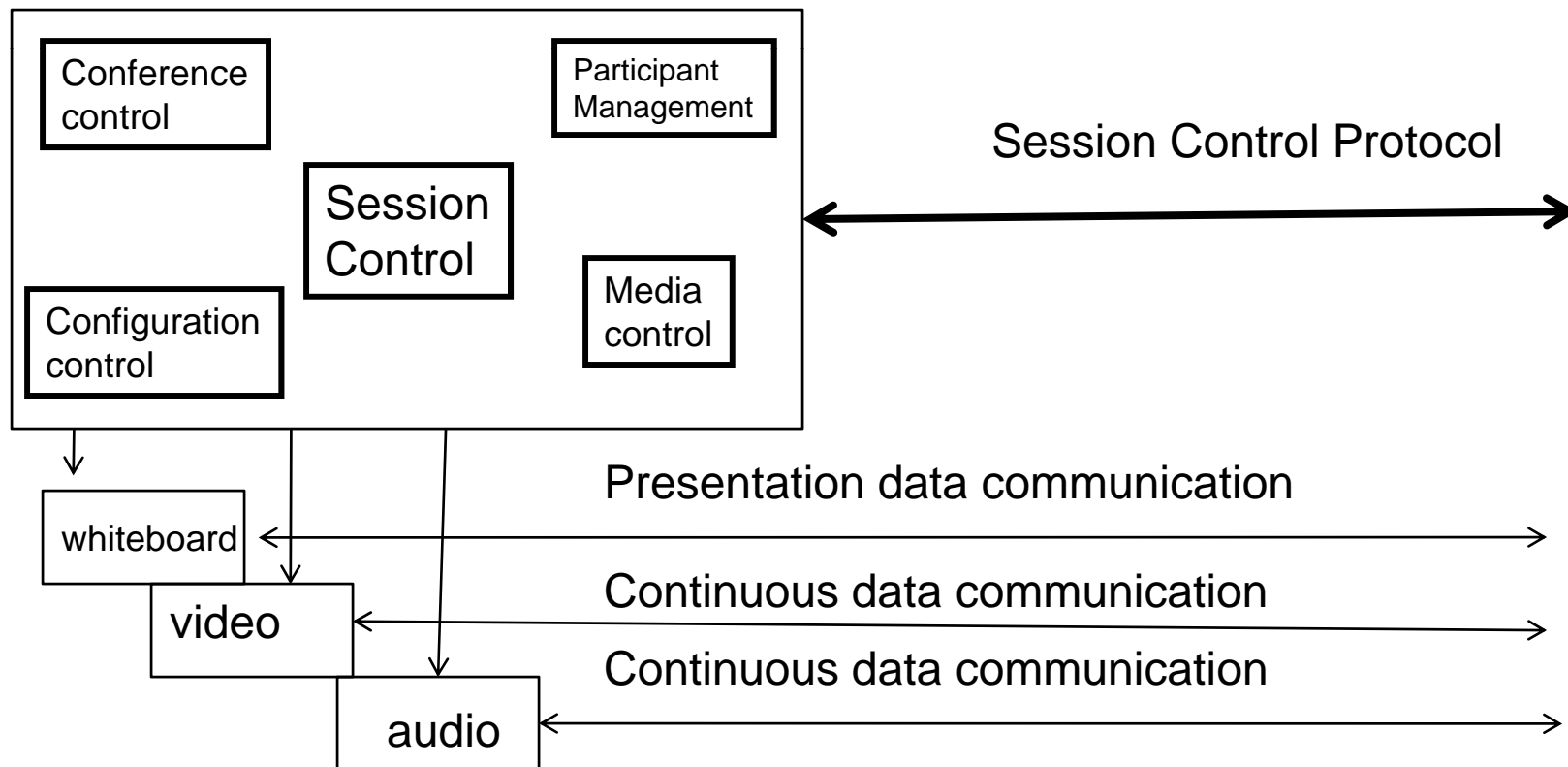
Payload Formats

- 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





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

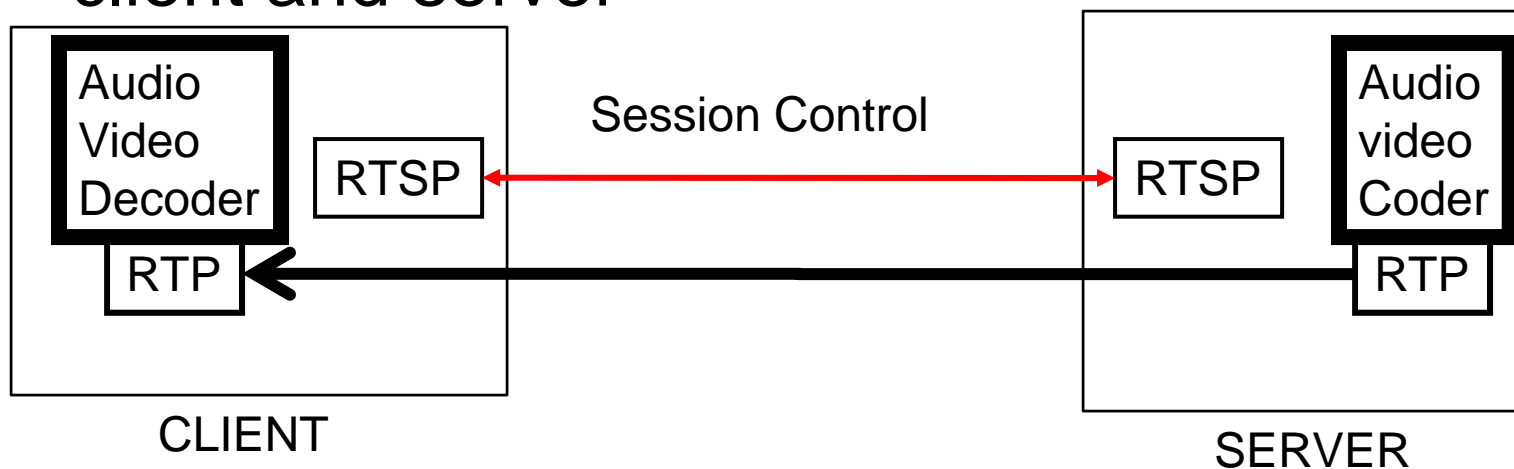
- Negotiation of QoS parameters, admission control and reservation/allocation of resources

- Membership Control

- Invitation of users; registration of users, change of membership

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

- Approved as Internet Draft, February 2, 1998, authors H. Schulzrinne, A. Rao, R. Lanphier
- 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

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 hours, minutes, seconds and fractions of second
- Absolute time
 - Wall clock



Conclusion

- 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