Real-time Services

BUPT/QMUL

2019-05-27

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Electronic Engineering

Agenda

- Real-time services over Internet
- Real-time transport protocols
  - RTP (Real-time Transport Protocol)
  - RTCP (RTP Control Protocol)
- Multimedia signaling protocols
  - H.323
  - SIP (Session Initiation Protocol)

Real-time services over Internet

Real-time Services Over Internet

- What is a real-time application?
- Multimedia application types
- Isochronous feature
- QoS requirements

What is a Real-time Application?

- A real-time application (RTA) is an application program that functions within a time frame that the user senses as immediate or current. (*whatis.com*)
  - The latency must be less than a defined value, usually measured in seconds.
- Examples of RTA:
  - Videoconference applications
  - VoIP (Voice over Internet Protocol)
  - Online gaming
  - Some e-commerce transactions
  - Chatting
  - IM (Instant Messaging)

Multimedia Application Types

- Streaming
  - Clients request audio/video files from servers and pipeline reception over the network and display
  - User can control the operation (similar to VCR: pause, resume, fast forward, rewind, etc.)
  - Delay: typically 1 to 10 seconds from client request until display start
- Unidirectional real-time
  - Push service, similar to existing TV and radio stations, but delivery over the network
  - Non-interactive, just listen/view
- Interactive real-time
  - Phone conversation or video conference
  - More stringent delay requirement than Streaming and Unidirectional because of real-time nature

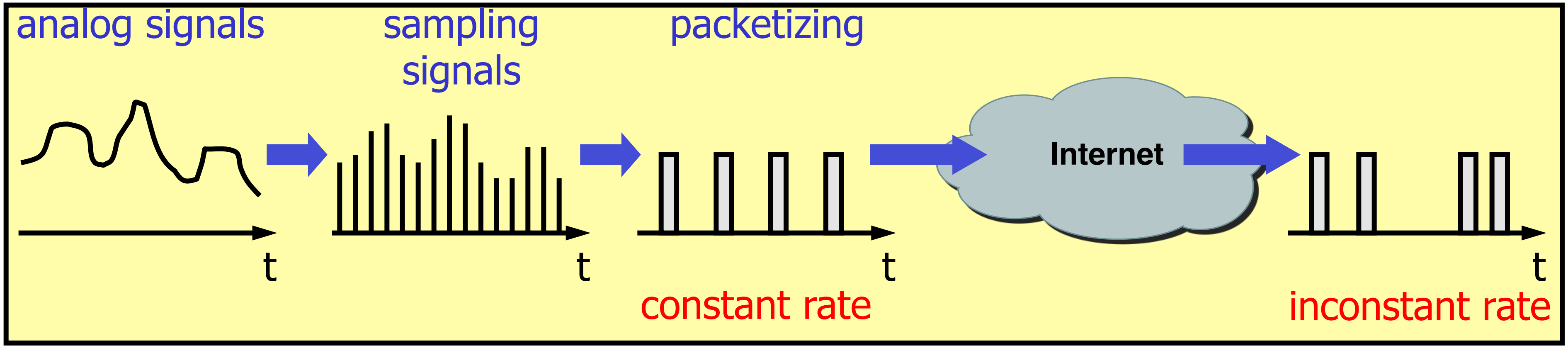
Isochronous Services

- Isochronous: From Greek iso, *equal* + chronos, *time*.
  - literally means to occur at the same time or at equal time intervals. *(wikipedia)*
- In information technology, isochronous pertains to processes that require timing coordination to be successful, such as voice and digital video transmission *(whatis.com)*
  - A coder~decoder (codec) device coverts between an analogue signal and an equivalent digital representation
  - The conventional telephone system uses the Pulse Code Modulation (PCM) standard that specifies taking an 8-bit sample every 125 μs (i.e., 8000 times per second). As a result, a digitized telephone call produces data at a rate of 64 Kbps.

Data flow should be continuously and at a steady rate

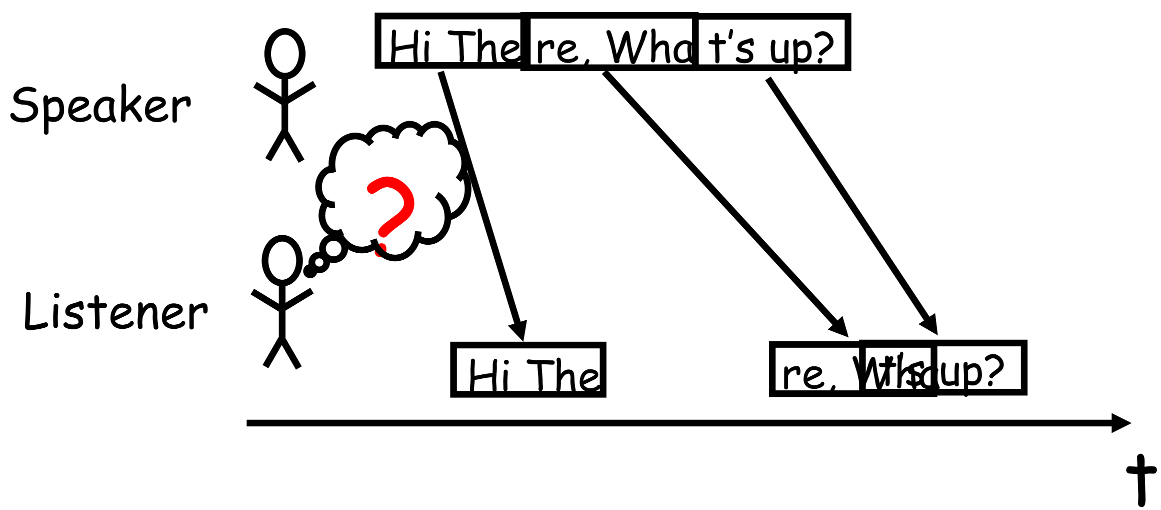
Isochronous Services over Internet

- Internet is not isochronous
  - Connectionless
  - Best effort
  - Adaptive routing



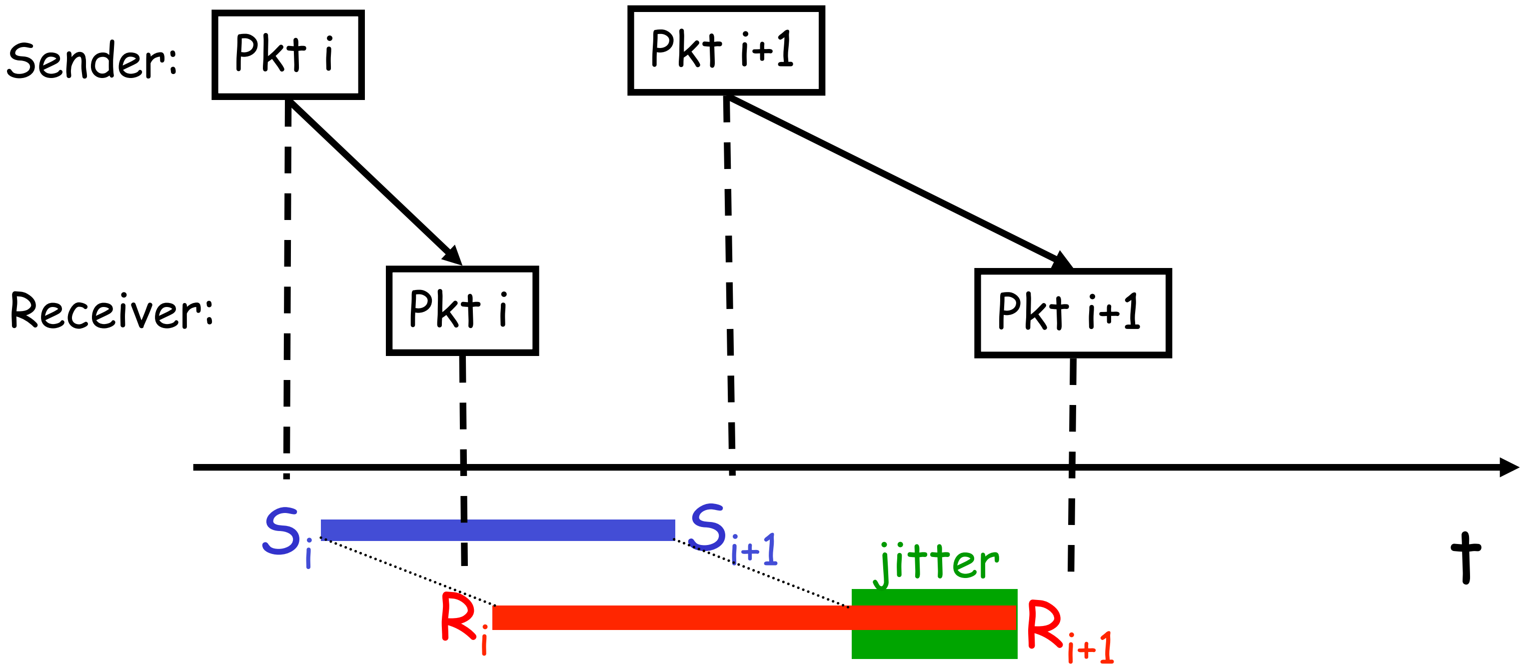
QoS Requirements of Real-time Services (1)

- Typically sensitive to delay (both end-to-end delay and jitter), but can tolerate packet loss
  - E.g., delay < 150msec , jitter<10msec
- Jitter
  - The Internet makes no guarantees about time of delivery of a packet
  - Consider an IP telephony session:



What is Jitter?

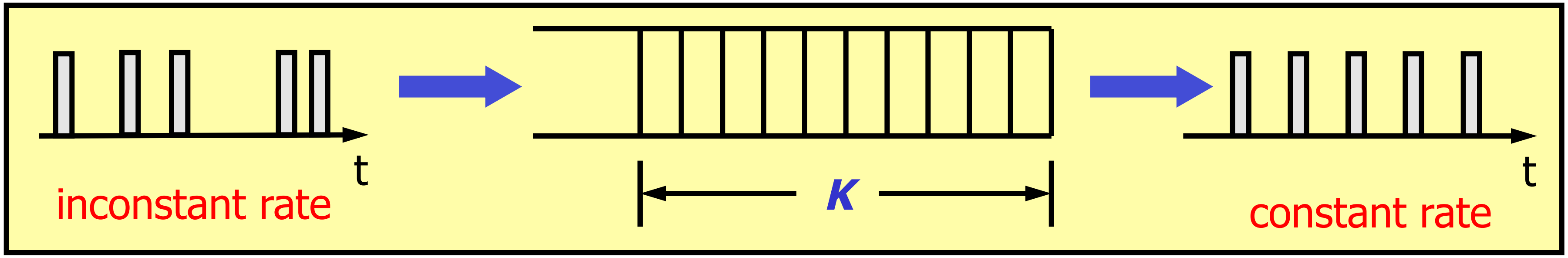
- A packet pair's jitter is the difference between the transmission time gap and the receive time gap



- Desired time-gap: $S_{i+1} - S_i$ Received time-gap: $R_{i+1} - R_i$
- Jitter between packets i and i+1: $(R_{i+1} - R_i) - (S_{i+1} - S_i)$

Jitter Compensation and Playback Delay

- Jitter compensation: playback buffer

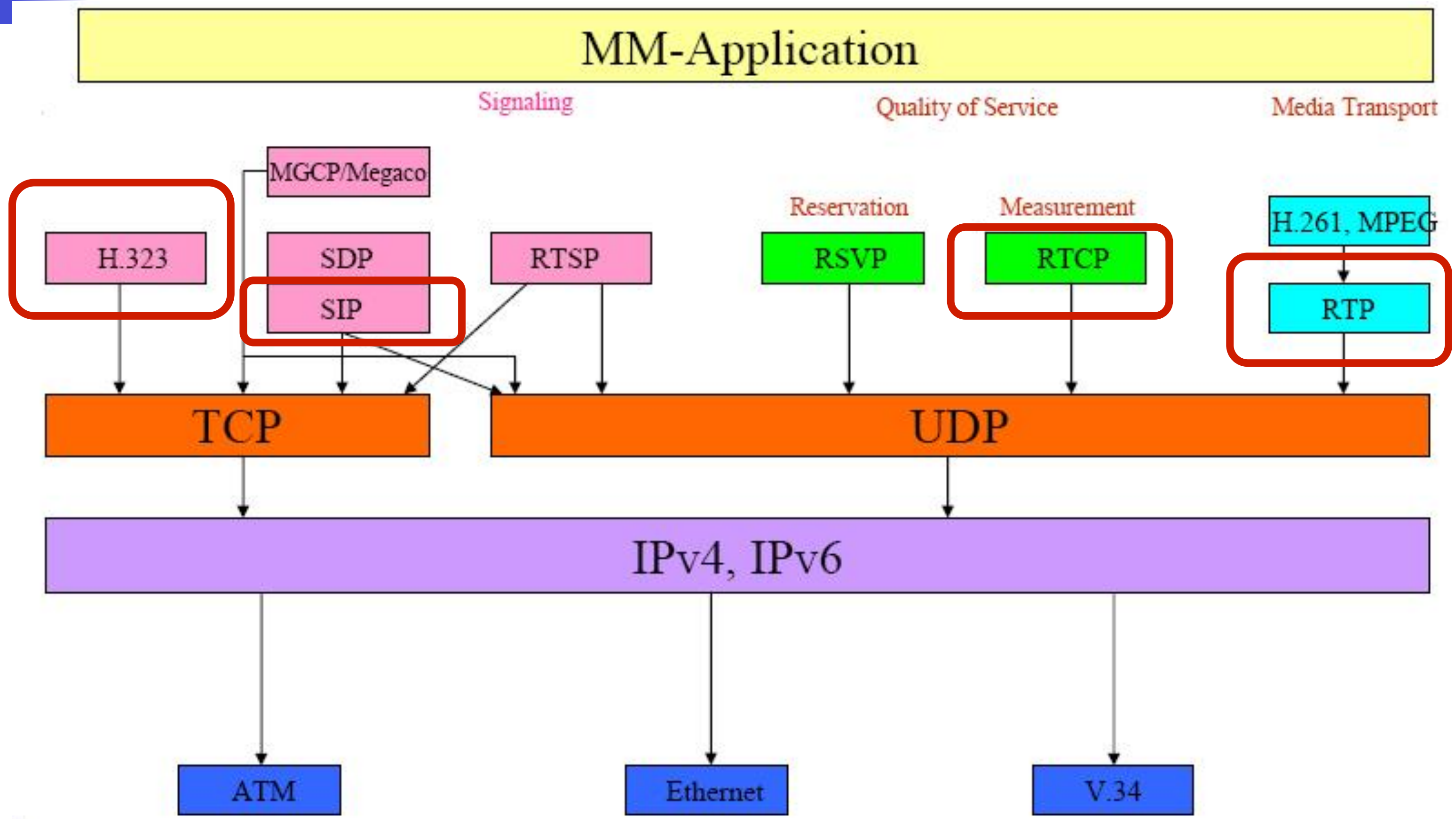


- Playback point
  - Labeled *K* in the figure
  - Is measured in time units of data to be played
- Incurs playback delay
- Choice of *K*
  - Large K: playback delay is increased
  - Small K: Jitter compensation effect is weakened

Key Technologies

- **Signaling technology**
  - ITU-T H.323, IETF SIP for multimedia signaling control
  - RTSP (Real-Time Streaming Protocol) for streaming media application control
- **Media Coding technology**
  - G.723.1、G.729，G.729A etc. for voice compress coding
  - MPEG-X for media compress coding
- **Real-time transport technology**
  - RTP (Real-time Transport Protocol)
- **QoS guarantee technology**
  - RTCP (RTP Control Protocol)
  - RSVP (Resource ReSerVation Protocol)/IntServ
  - DiffServ
- **Others**
  - Synchronization technology
  - Gateway interconnection
  - Network management technology
  - Security
  - Billing

Multimedia Technologies over Internet



RTP / RTCP
Real-time Transport Protocol
RTP Control Protocol

RTP/RTCP Overview

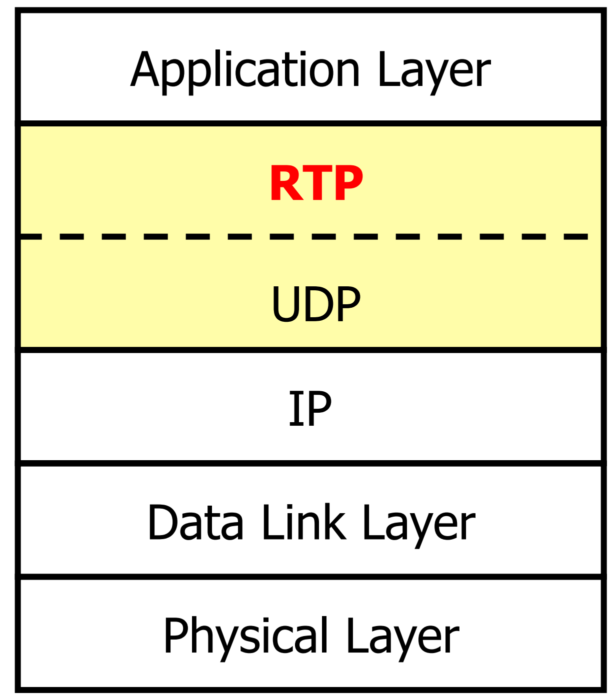
- Defined in RFC 3550 etc.
- RTP
  - Provides end-to-end transmission for real-time applications
  - Does NOT define any QoS mechanism for real-time delivery
- RTCP
  - RTP's companion control protocol
  - Provides flow control and congestion control for RTP

RTP

- Because an Internet is not isochronous, additional protocol support is required when sending digitized real-time data
  - The basic sequence information that allows detection of lost, duplicate or reordered packets
  - Each packet must carry a separate timestamp that tells the receiver the exact time at which the data in the packet should be played
- RTP provides these two facilities and is used to transmit digitized audio or video signals over an Internet
- RTP is suitable for applications transmitting real-time data, such as audio or video, over multicast or unicast network services

RTP and UDP

- RTP is a transport-level protocol
- RTP runs over UDP
  - Each RTP message is encapsulated in a UDP datagram
- RTP does not use a reserved UDP port number
  - A port is allocated for use with each session and the remote application must be informed about the port number
  - Generally, RTP chooses an even numbered UDP port

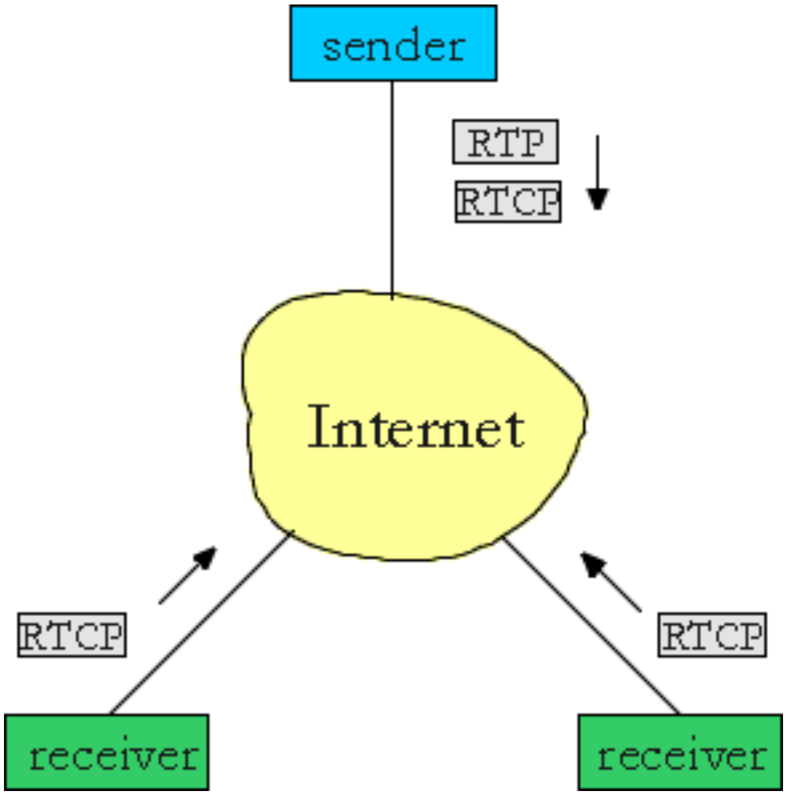


RTP and QoS

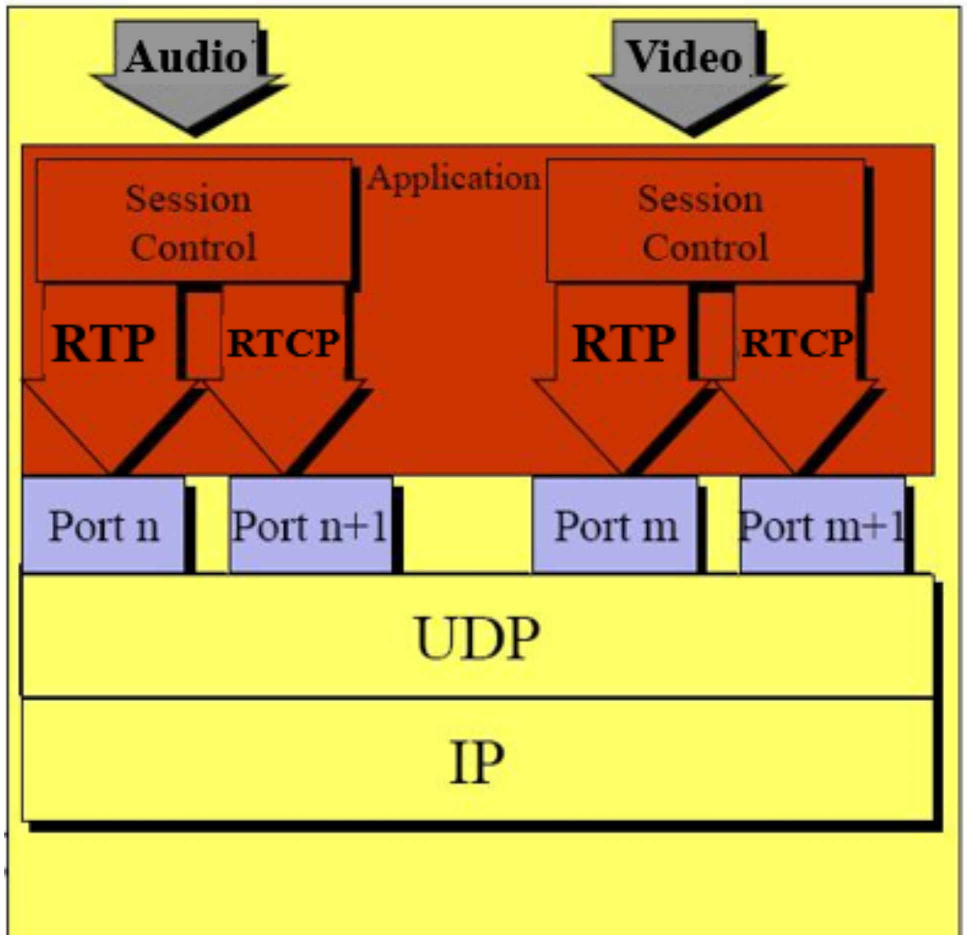
- RTP does not address resource reservation and does not guarantee QoS for real-time services
  - RSVP/IntServ
  - DiffServ
- RTP does not guarantee reliable and sequenced delivery
  - It assume that the underlying network is reliable and delivers packets in sequence
  - Relies on RTCP to provide flow control and congestion control

RTCP

- RTCP is a companion protocol and integral part of RTP. It provides control functionality
- Primary function
  - provide feedback on the quality of the data distribution exchanged between sources and destinations of multimedia information
- Usage of the feedback report
  - Be directly useful for control of adaptive encodings
  - Allows receivers /monitors to diagnose faults in the distribution
- RTCP messages are encapsulated in UDP for transmission
  - RTCP-port = RTP-port+1



RTP and RTCP Comparison



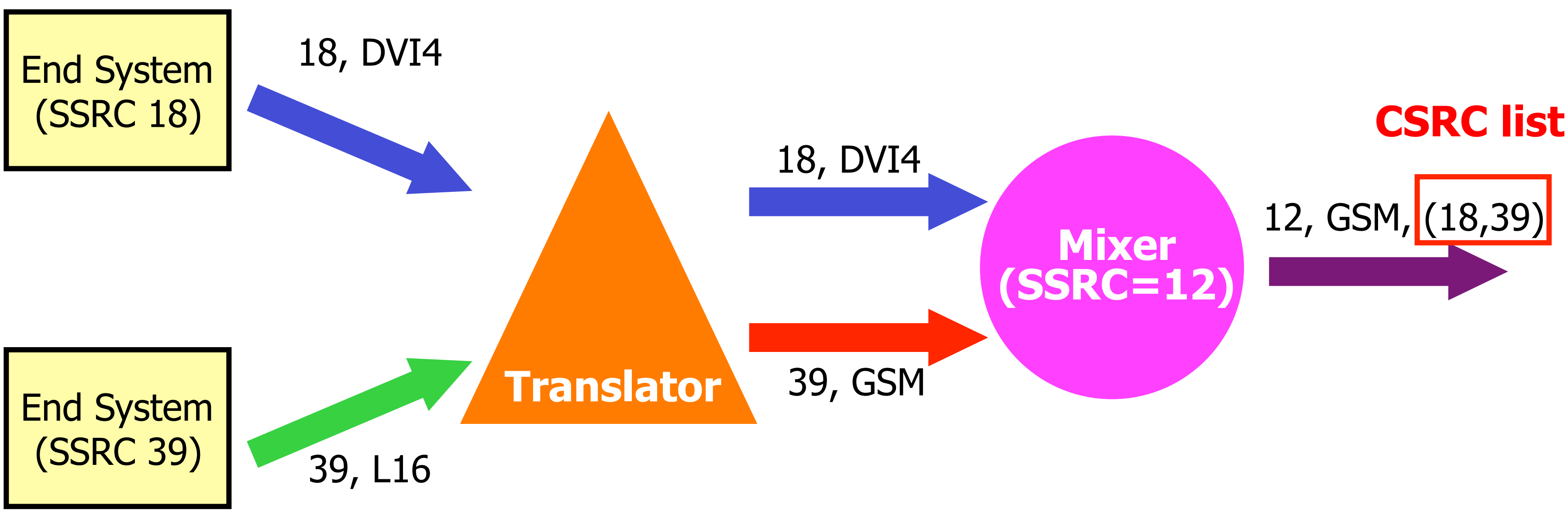
RTP

- For data
- Important fields in header
  - sequence number
  - timestamp
  - synchronization ID

RTCP

- For QoS monitoring and control
- Provides feedback on the quality of the data distribution

RTP/RTCP System (1)



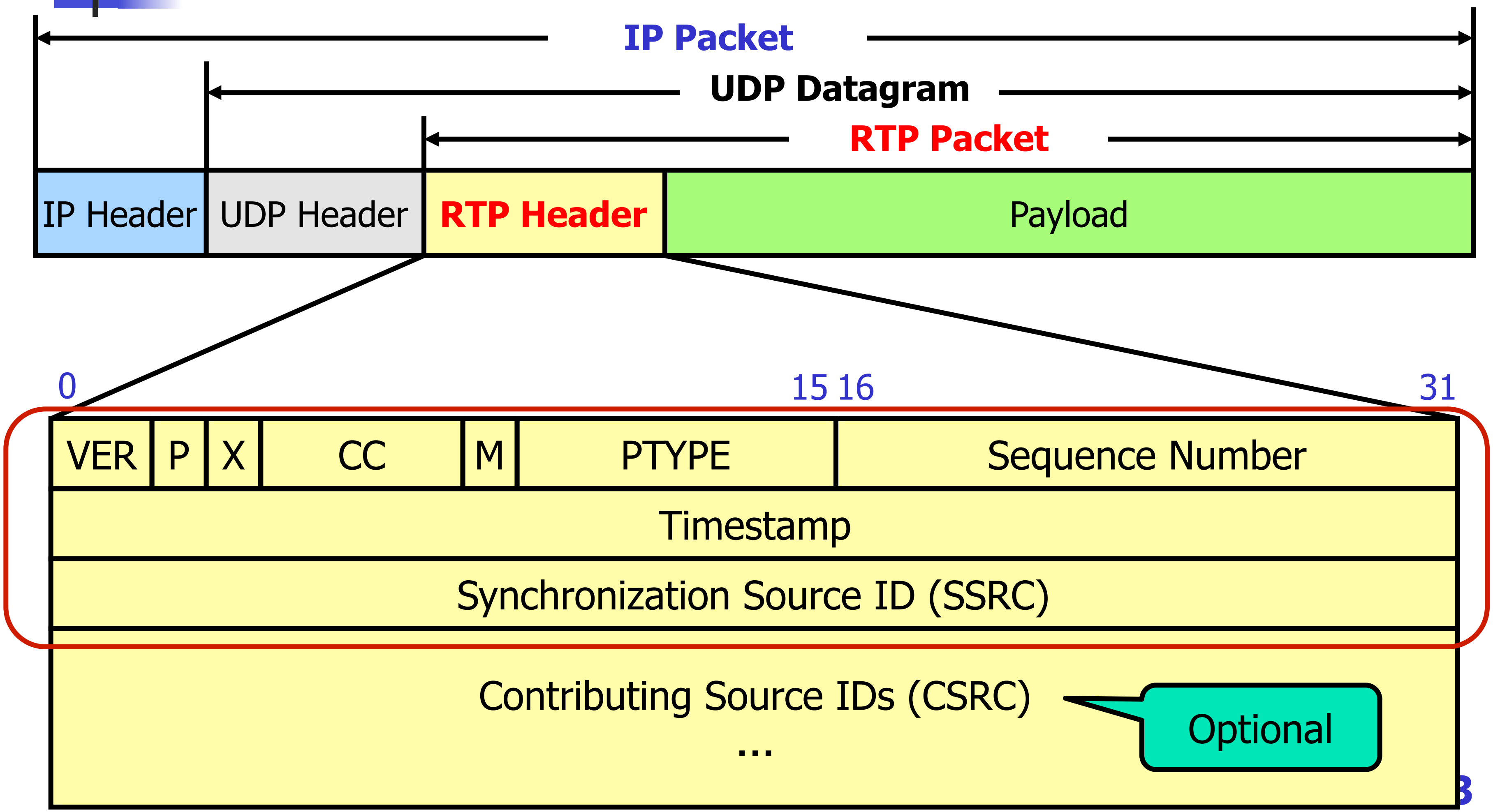
SSRC: Synchronization SouRCe Identifier

CSRC: Contributing SouRCe Identifier

RTP/RTCP System (2)

- ***End system***: An application that generates the content to be sent in RTP packets and/or consumes the content of received RTP packets.
- ***Mixer***: An intermediate system that receives RTP packets from one or more sources, possibly changes the data format, combines the packets in some manner and then forwards a new RTP packet.
- ***Translator***: An intermediate system that forwards RTP packets with their synchronization source identifier(SSRC) unchanged. E.g., devices that convert encoding without mixing
- ***Monitor***: An application that receives RTCP packets sent by participants in an RTP session, in particular the reception reports, and estimates the current quality of service for distribution monitoring, fault diagnosis and long-term statistics.

RTP Packet Format (1)



Version 2 bits	Padding 1 bit	Extension 1 bit	CC 4 bits	Marker 1 bit	Payload Type 7 bits	Sequence Number 2 bytes	Time stamp 4 bytes	SSRC 4 bytes	CSRCs 0-60 bytes

RTP Packet Format (2)

- The first 12 octets are present in every RTP packet, while the list of CSRC identifiers is present only when inserted by a mixer.
- The fields have the following meaning:
  - **Version (V)**: 2 bits - This field identifies the version of RTP. The current version is two (2).
  - **Padding (P)**: 1 bit - If the padding bit is set, the packet contains one or more additional padding octets at the end which are not part of the payload. The last octet of the padding contains a count of how many padding octets should be ignored, including itself
  - **Extension (X)**: 1 bit - If the extension bit is set, the fixed header MUST be followed by exactly one header extension, with a defined format

Version 2 bits	Padding 1 bit	Extension 1 bit	CC 4 bits	Marker 1 bit	Payload Type 7 bits	Sequence Number 2 bytes	Time stamp 4 bytes	SSRC 4 bytes	CSRCs 0-60 bytes

RTP Packet Format (3)

- **CSRC count (CC)**: 4 bits - The CSRC count contains the number of CSRC identifiers that follow the fixed header
- **Marker (M)**: 1 bit - The interpretation of the marker is defined by a profile
- **Payload type (PTYPE)**: 7 bits - This field identifies the format of the RTP payload and determines its interpretation by the application. (RFC3551)
- **Sequence number**: 16 bits - The sequence number increments by one for each RTP data packet sent, and may be used by the receiver to detect packet loss and to restore packet sequence

Payload Types

Type	*Application*	*Type*	*Application*	*Type*	*Application*
0	PCMμ Audio	7	LPC audio	15	G728 audio
1	1016	8	PCMA audio	26	Motion JPEG
2	G721 audio	9	G722 audio	31	H.261
3	GSM audio	10-11	L16 audio	32	MPEG1 video
5-6	DV14 audio	14	MPEG audio	33	MPEG2 video

Version 2 bits	Padding 1 bit	Extension 1 bit	CC 4 bits	Marker 1 bit	Payload Type 7 bits	Sequence Number 2 bytes	Time stamp 4 bytes	SSRC 4 bytes	CSRCs 0-60 bytes

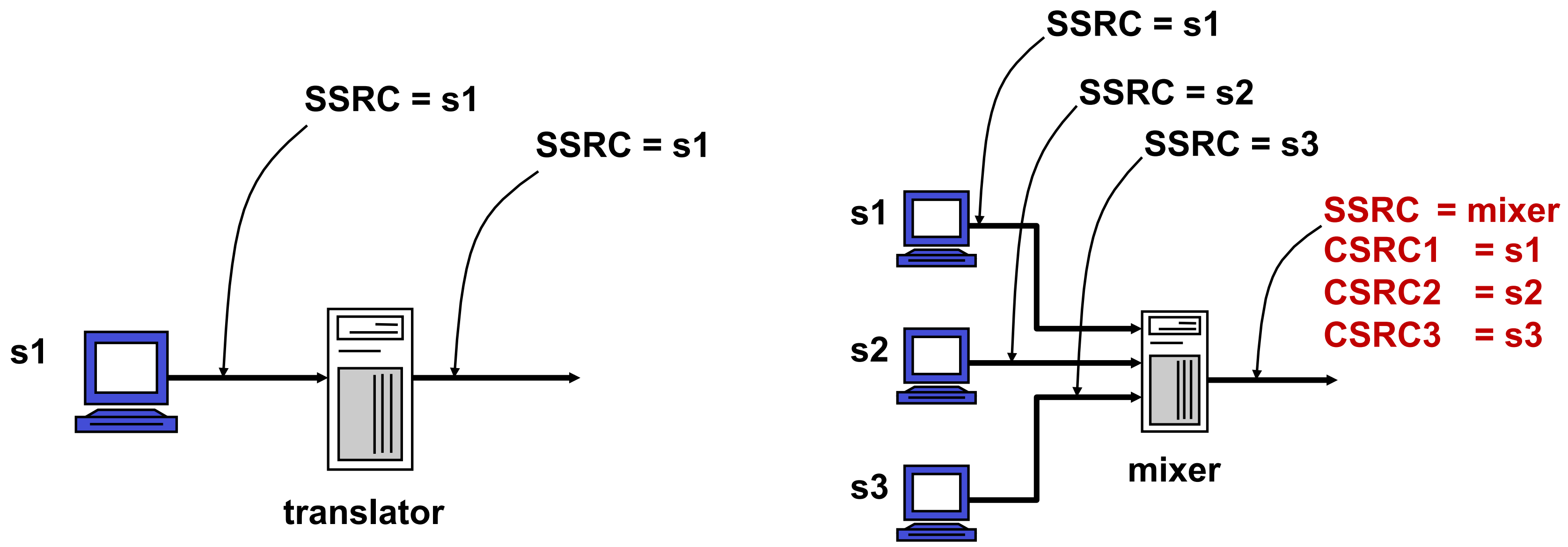
RTP Packet Format (4)

- **Timestamp**: 32 bits - The timestamp reflects the sampling instant of the first octet in the RTP data packet. The sampling instant MUST be derived from a clock that increments monotonically and linearly in time to allow synchronization and jitter calculations

- **SSRC**: 32 bits - The SSRC field identifies the synchronization source (the source of a RTP packet stream). i.e. mixer.

- **CSRC list**: 0 to 15 items, 32 bits each - The CSRC list identifies the contributing sources (sources before mixing) for the payload contained in this packet. The number of identifiers is given by the CC field.

SSRC vs. CSRC



SSRC = s1
SSRC = s1
s1
translator
SSRC = s1
SSRC = s2
SSRC = s3
s1
s2
s3
mixer
SSRC = mixer
CSRC1 = s1
CSRC2 = s2
CSRC3 = s3

RTCP Packet Types

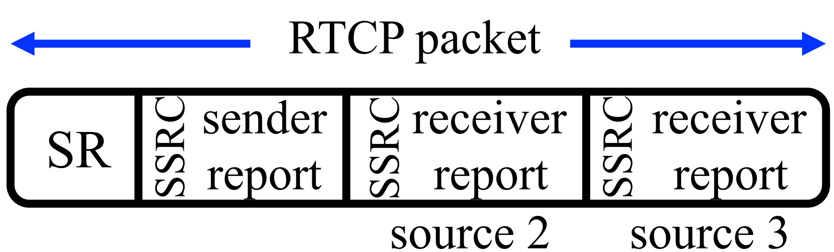
Type	Name	Description
200	SR	Sender Report – transmission & reception statistics
201	RR	Receiver Report – reception statistics
202	SDES	Source DEScription – Source description items
203	BYE	Indicates end-of-participation
204	APP	application-specific functions

- Typically, several RTCP packets of different types are transmitted in a single UDP packet

RTCP Packets-SR

- includes
  - SSRC of sender — identify source of data
  - NTP timestamp — when report was sent
  - RTP timestamp — corresponding RTP time
  - packet count — total number sent
  - octet count — total number sent
  - followed by zero or more receiver reports…
  - example:

    source 1 reports, there are 2 other sources



RTCP Packets-RR

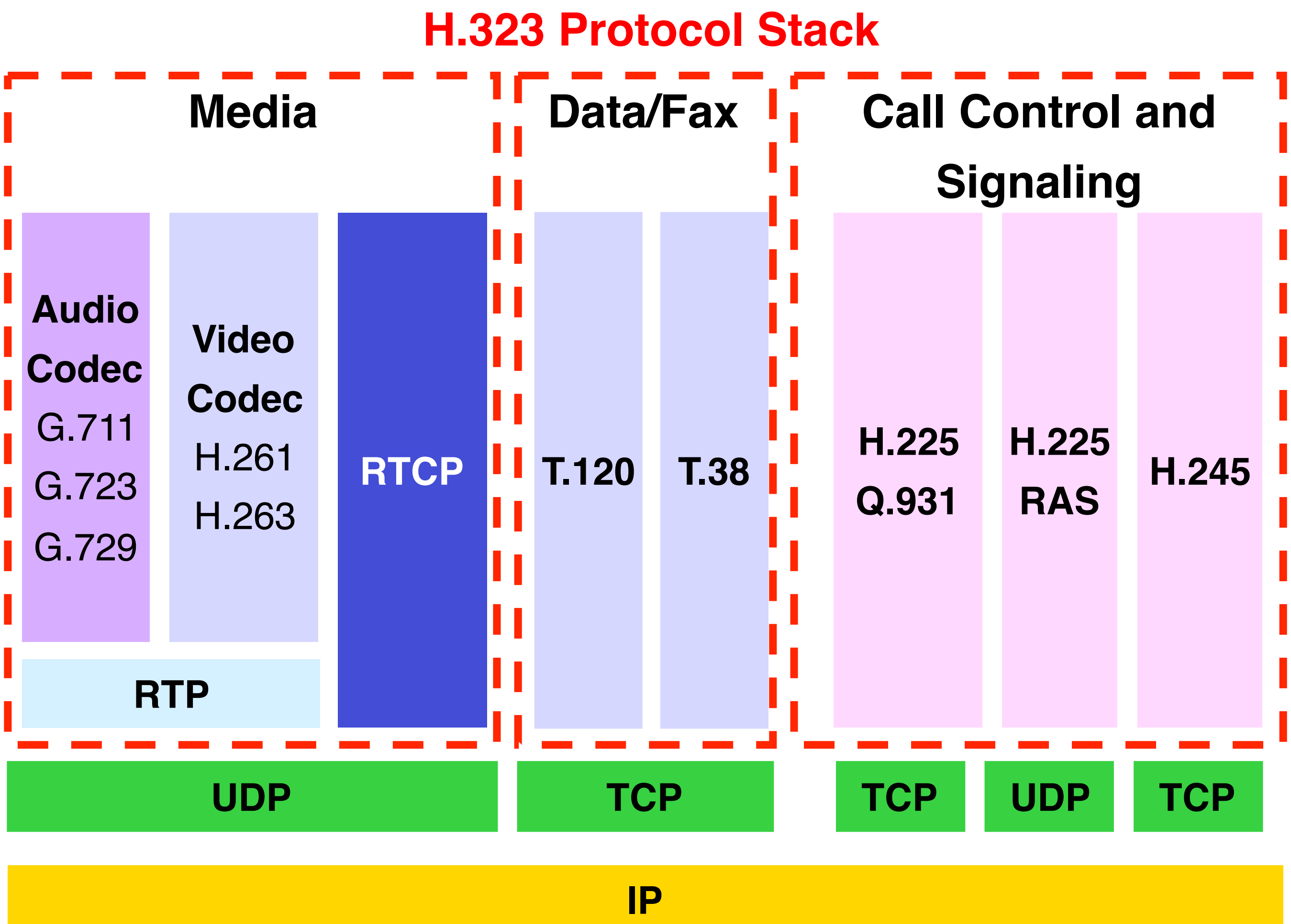
- includes
  - SSRC of source — identify the source to which this RR block pertains
  - fraction lost — since previous RR (SR) sent
  - cumul # of packets lost — long term loss
  - highest seq # received — compare losses
  - interarrival jitter — smoothed interpacket distortion
  - LSR — time when last SR heard
  - DLSR — delay since last SR

Multimedia Signaling Protocols

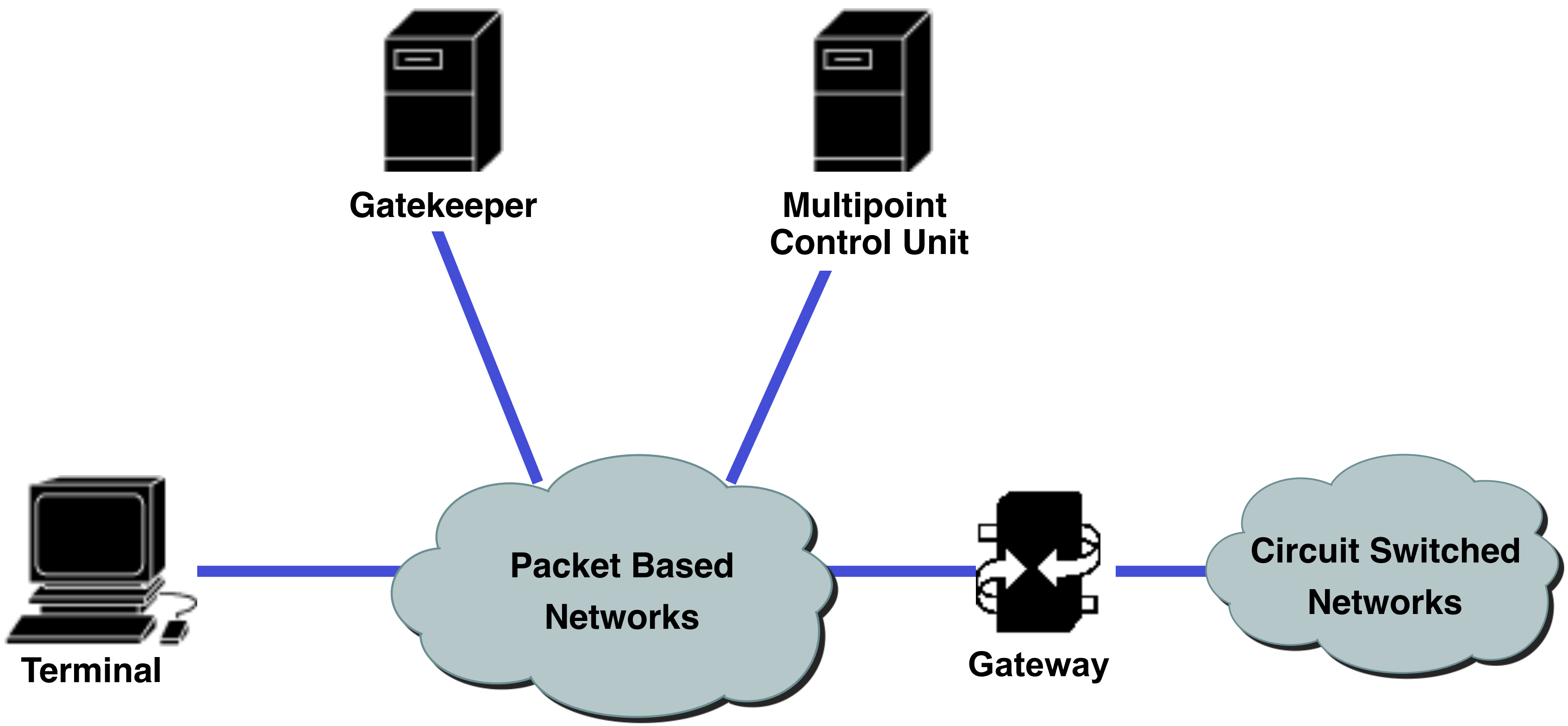
H.323 – Overview

- H.323 – Describes terminals and other entities that provide multimedia communications services over Packet Based Networks (PBN) which may not provide a guaranteed Quality of Service. H.323 entities may provide real-time audio, video and/or data communications
  - *(from ITU-T Recommendation H.323 Version 4)*
- Defined by ITU-T
- Actually is composed of a set of protocols

H.323 Protocol Stack



H.323 – Components (1)



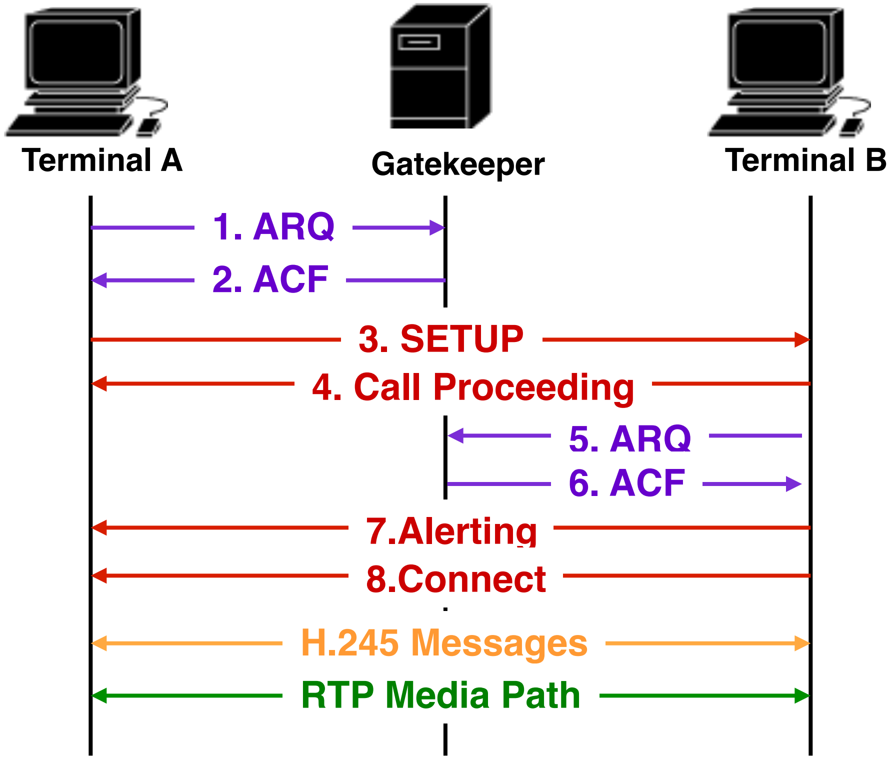
H.323 – Components (2)

- **H.323 Terminals**: client endpoints that must support:
  - H.225 call control signaling
  - H.245 channel control signaling
  - RTP/RTCP protocols for media packets
  - Audio codecs
  - Video codecs support is optional
- **H.323 Gateway**: provides translation
  - Provide translation between entities in a packet switched network (e.g. IP network) and circuit switched network (e.g., PSTN network).
  - Provide transmission formats translation, communication procedures translation, H.323 and non-H.323 endpoints translations or codec translation.

H.323 – Components (3)

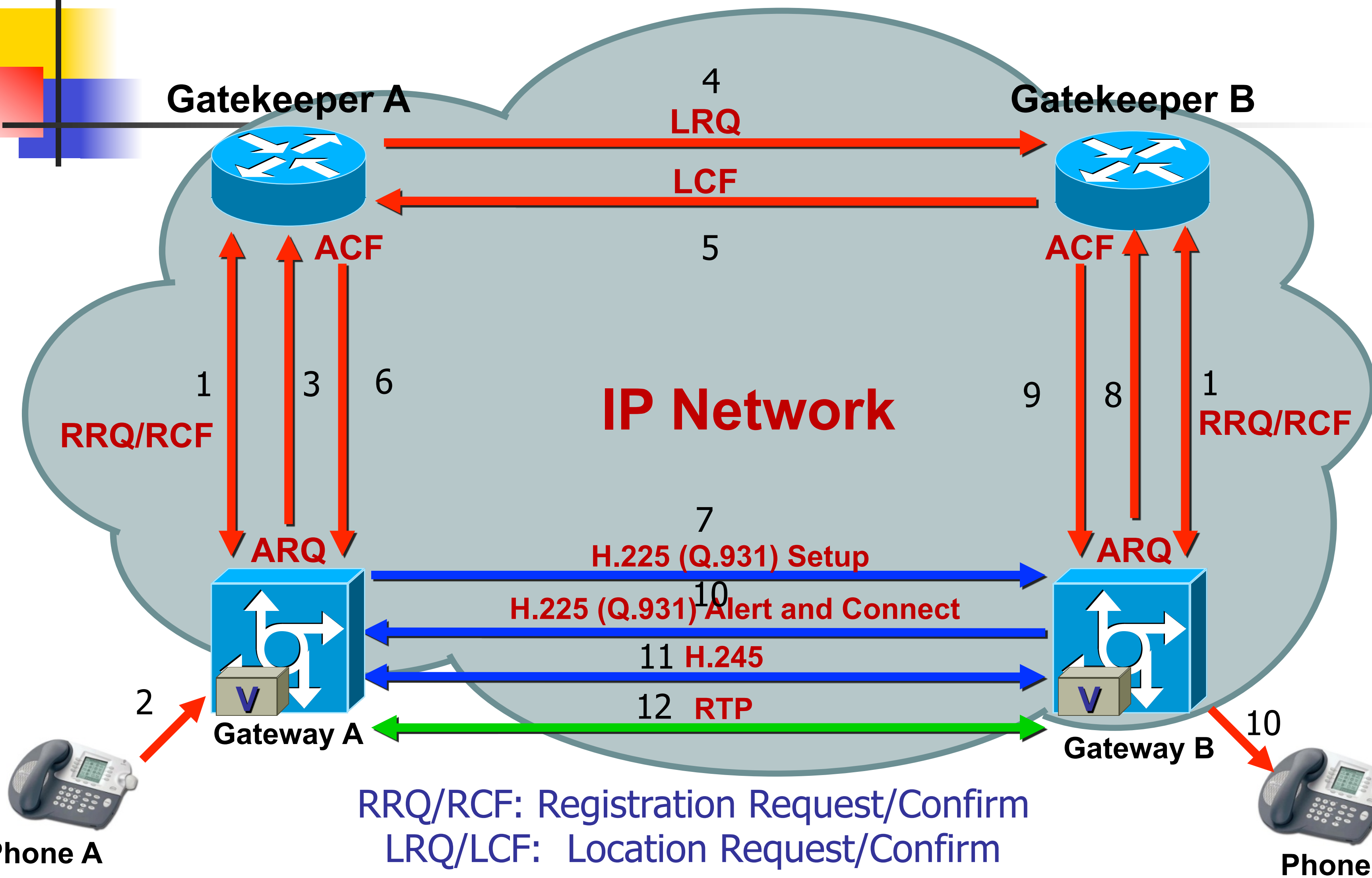
- **H.323 Gatekeepers**
  - Provide following functions
    - Address translation
    - Admission control
    - Bandwidth control
    - Zone management
    - Optional: Call control signaling, Call authorization, Bandwidth management, Call management
  - Gatekeepers are optional but if present in a H.323 system, all H.323 endpoints must register with the gatekeeper and receive permission before making a call
- **H.323 MCU (Multipoint Control Unit)**
  - Providing support for conferences of three or more endpoints
  - A MCU consists of:
    - Multipoint Controller (MC) – provides control functions
    - Multipoint Processor (MP) – receives and processes audio, video and/or data streams

Simplified H.323 Call Setup Procedure (with 1 Gatekeeper)



- **RAS messages**
  - **ARQ – Admission Request**
  - **ACF – Admission Confirm**

Basic H.323 Call Procedure (with 2 Gatekeepers)



RRQ/RCF: Registration Request/Confirm
LRQ/LCF: Location Request/Confirm
ARQ/ACF: Admission Request/Confirm

SIP – Overview

- Session Initiation Protocol - An application layer signaling protocol that defines initiation, modification and termination of interactive, multimedia communication sessions between users
  - *(from RFC 3261)*
- Defined in RFC 3261 etc.
- Based on TCP or UDP

Instant Messaging

Personal Mobility

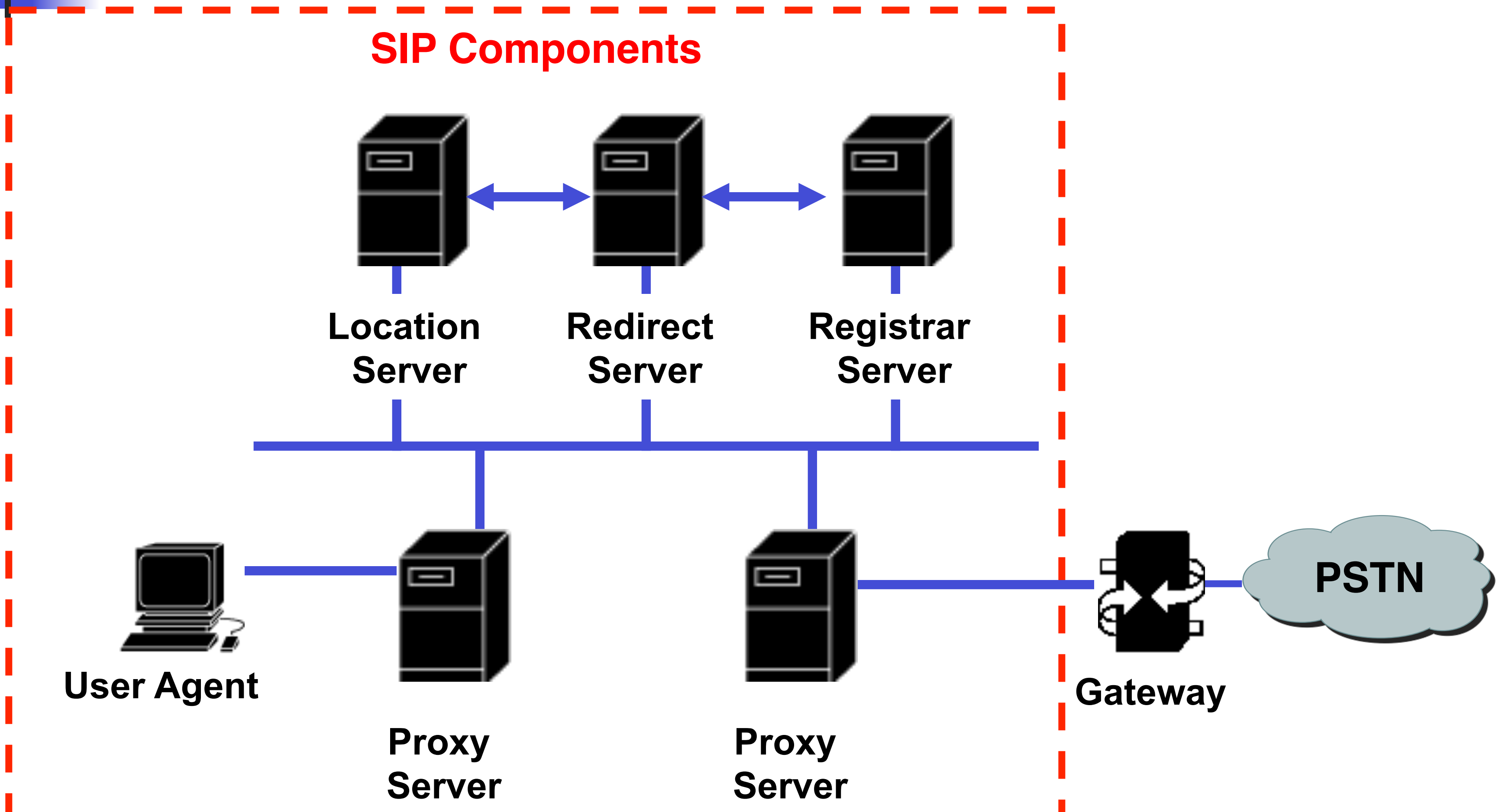
Voice Calls

Video Conferencing

Distance Learning

MPEG, MP3, Audio, HTML, XML

SIP – Architecture



SIP – Components (1)

- UA (User Agent): An application that initiates, receives and terminates calls
  - UAC (User Agent Clients) – An entity that initiates a call
  - UAS (User Agent Server) – An entity that receives a call
  - Both UAC and UAS can terminate a call
- Proxy Server
  - An intermediary program that acts as both a server and a client to make requests on behalf of other clients.
  - Requests are serviced internally or by passing them on, possibly after translation, to other servers.
  - Interprets, rewrites or translates a request message before forwarding it

SIP – Components (2)

- Redirect Server
  - A server that accepts a SIP request, maps the address into zero or more new addresses and returns these addresses to the client.

- Registrar Server
  - A server that accepts REGISTER requests.
  - The register server may support authentication.
  - A registrar server is typically co-located with a proxy or redirect server and may offer location services.

- Location Server
  - A location server is used by a SIP redirect or proxy server to obtain information about a called party's possible location(s).

SIP Messages – Methods and Responses

- SIP components communicate by exchanging SIP messages

- **SIP Methods**
  - INVITE – Initiates a call by inviting user to participate in session.
  - ACK - Confirms that the client has received a final response to an INVITE request.
  - BYE - Indicates termination of the call.
  - CANCEL - Cancels a pending request.
  - REGISTER – Registers the user agent.
  - OPTIONS – Used to query the capabilities of a server.

- **SIP Responses**
  - 1xx - Informational Messages.
  - 2xx - Successful Responses.
  - 3xx - Redirection Responses.
  - 4xx - Request Failure Responses.
  - 5xx - Server Failure Responses.
  - 6xx - Global Failures Responses.

SIP Headers

- SIP borrows much of the syntax and semantics from HTTP.
- A SIP messages looks like an HTTP message – message formatting, header and MIME support.
- An example SIP header:

Startline

route

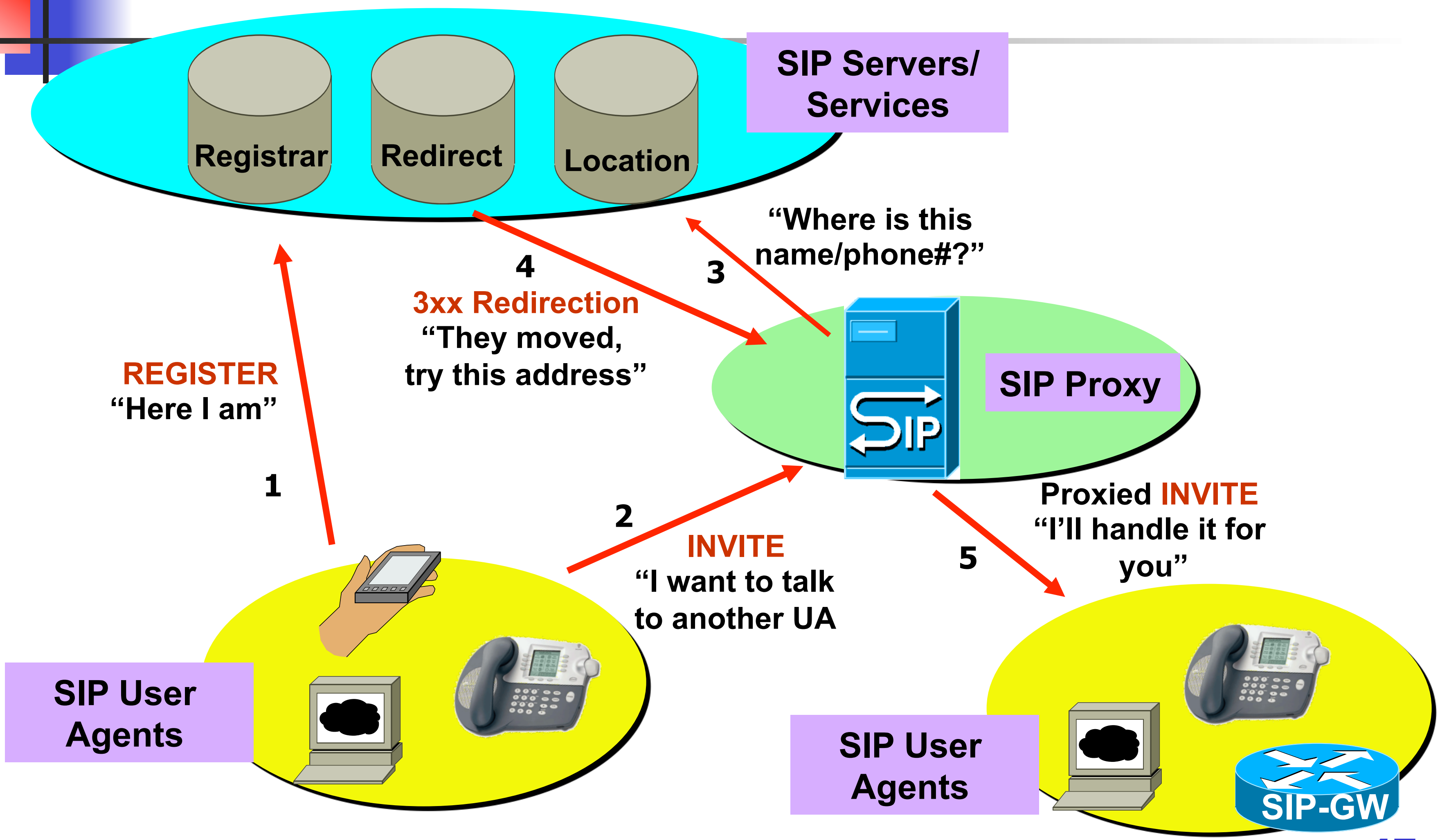
dialog

```
------------------------------------------------------------
                        SIP Header
------------------------------------------------------------
INVITE sip:5120@192.168.36.180 SIP/2.0
Via: SIP/2.0/UDP 192.168.6.21:5060
From: sip:5121@192.168.6.21
To: <sip:5120@192.168.36.180>
Call-ID: c2943000-e0563-2a1ce-2e323931@192.168.6.21
CSeq: 100 INVITE
Expires: 180
User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled
Accept: application/sdp
Contact: sip:5121@192.168.6.21:5060
Content-Type: application/sdp
```

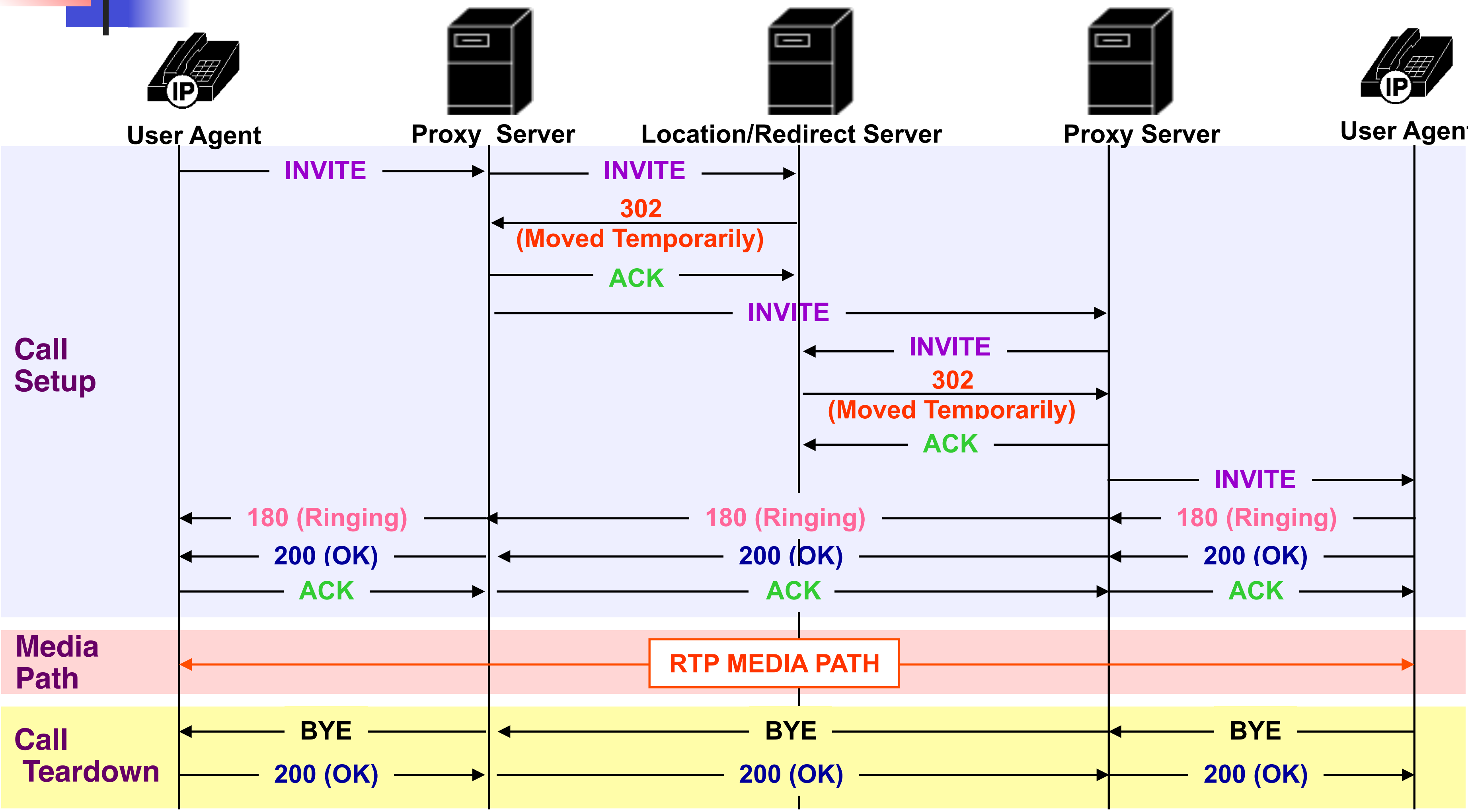
SIP Addressing

- The SIP address is identified by a SIP URL, in the format: user@host
- Examples of SIP URLs:
  - sip:hostname@vovida.org
  - sip:hostname@192.168.10.1
  - sip:14083831088@vovida.org

SIP Service Example (with 1 proxy server)



Example SIP Call Procedure (with 2 proxy servers)



SIP – Integration With IETF Protocols

- SIP can works with existing IETF protocols, for example:
  - RSVP – to reserve network resources.
  - RTP / RTCP – to transport real time data and provide QOS feedback.
  - RTSP – for controlling delivery of streaming media.
  - SAP (Session Advertisement Protocol) – for advertising multimedia session via multicast
  - SDP (Session Description Protocol) – for describing multimedia sessions

H.323 vs. SIP

	H.323	SIP
Philosophy	Designed for multimedia communication over different types of networks	Designed to session between two points
Reliability	Designed to handle failure of network entities	No defined procedures for handling device failure
Message Encoding	Encodes in compact binary format	Encodes in ASCII text format. Hence easy to debug and process
Addressing	Flexible addressing scheme using URLs and E.164 numbers	Understands only URLs style addresses
IM Support	No	Yes
Architecture	Monolithic	Modular

Summary

Summary

- Real-time services
  - Isochronous services and QoS requirements
  - Jitter compensation via playback buffer
- RTP/RTCP
  - Functions of RTP/RTCP
  - Definitions: end system, translator, mixer
  - RTP packet format
  - RTCP packet types
- Multimedia signaling protocols
  - H.323
    - Function, H.323 protocol stack
    - Architecture and components
  - SIP
    - Function
    - Architecture and components

Useful URLs

- RFCs for RTP/RTCP/SIP
  - http://www.ietf.org/
- SIP
  - http://www.siptutorial.net/SIP/index.html
- H.323
  - http://www.itu.int/rec/T-REC-H.323/en
  - http://www.telecomspace.com/vop-h323.html
- Useful tools downloading
  - http://www.pjsip.org/
  - http://sourceforge.net/projects/rtpmonitor/

Abbreviations

CSRC	Contributing SouRCe Identifier
IM	Instant Messaging
RTCP	RTP Control Protocol
RTP	Real-time Transport Protocol
SIP	Session Initiation Protocol
SSRC	Synchronization SouRCe Indentifier
VoIP	Voice over IP