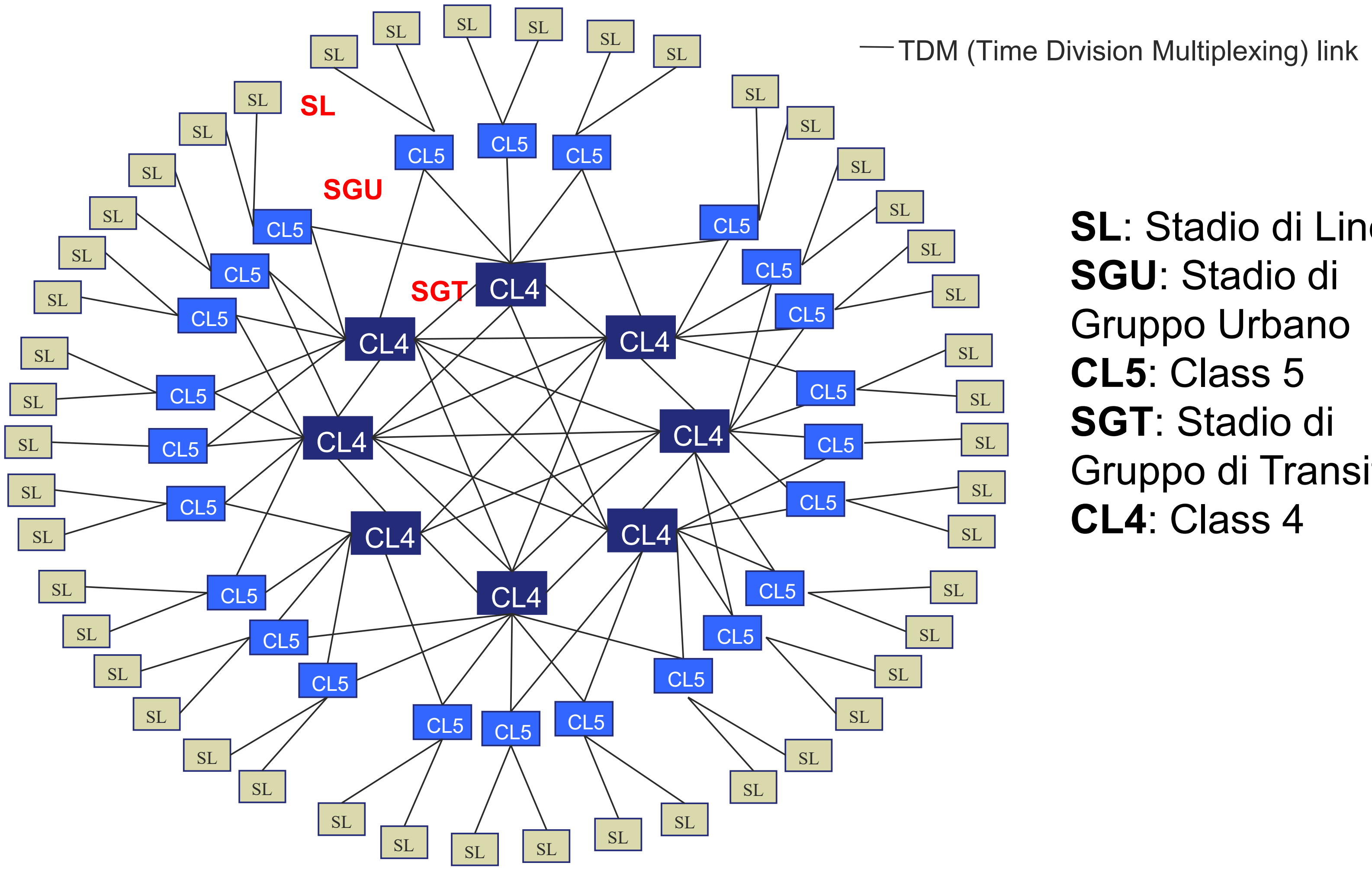
VoIP Architectures

Some notes on legacy architectures: PSTN

- The traditional telephone network (PSTN (Public Switched Telephone Network) was born in the 1960s from studies and projects carried out by dominant operators (incumbents).
- The PSTN provided top-level performance in terms of reliability, security and ubiquity.
- The obsolescence of the PSTN is due to technological aging and conceptual design that does not allow for rapid and complete evolution towards the broadband services required by new applications.
- The convergence towards network infrastructures based on the IP protocol is now complete.

Legacy Network topology



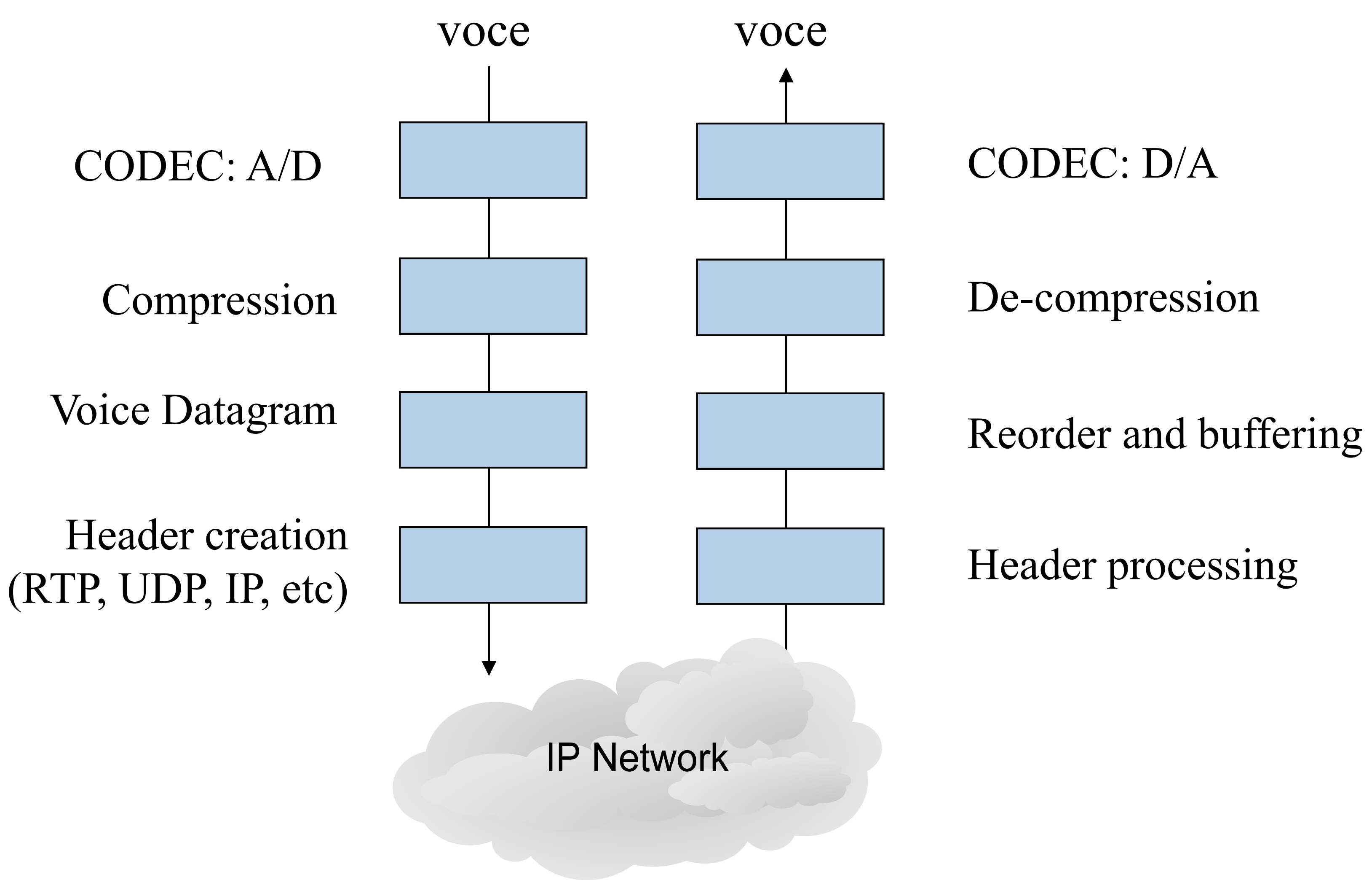
SL: Stadio di Linea
SGU: Stadio di Gruppo Urbano
CL5: Class 5
SGT: Stadio di Gruppo di Transito
CL4: Class 4

Voice over IP (VoIP)

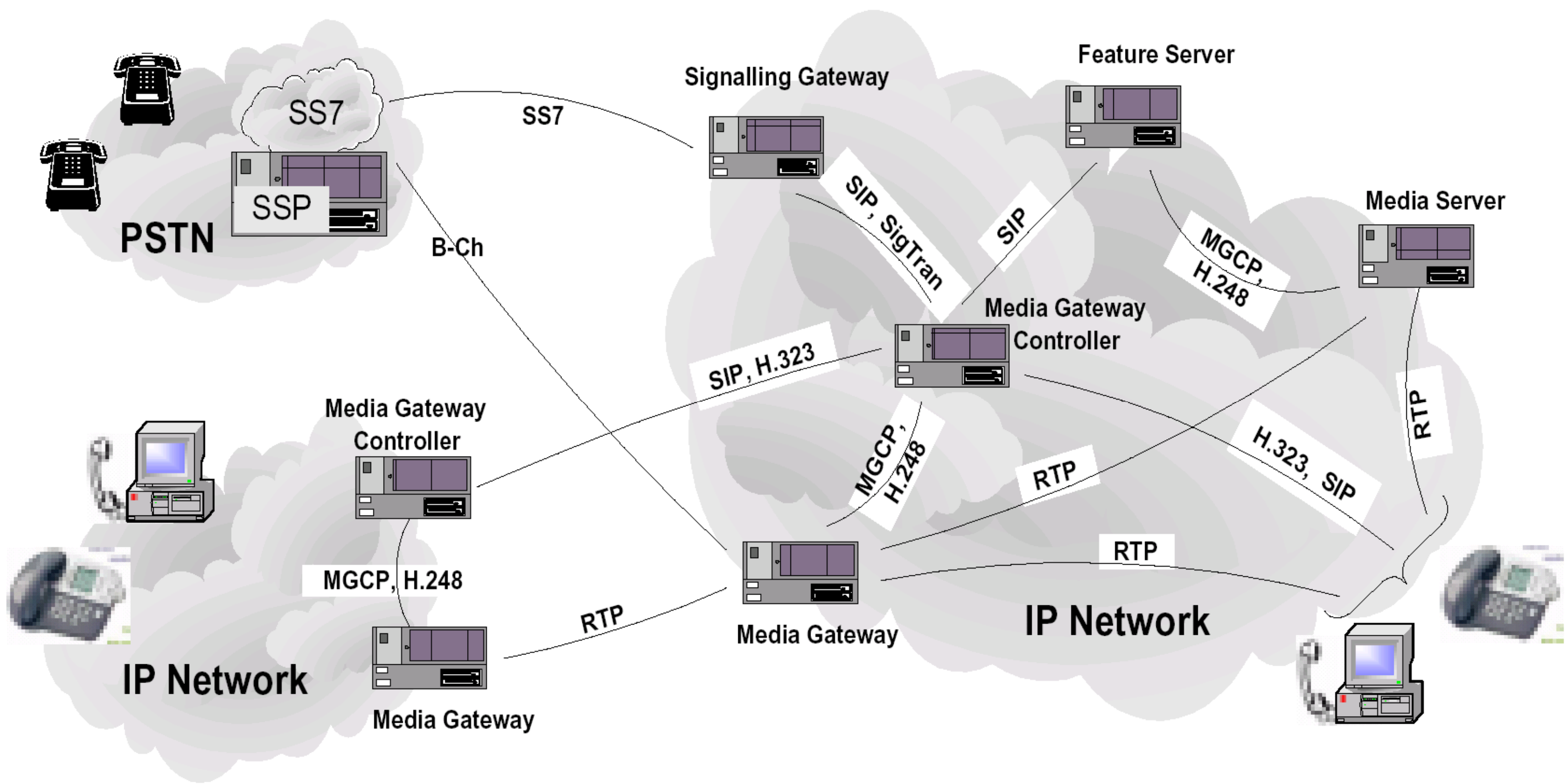
- Definition: VoIP is the ability to make telephone calls over IP networks ensuring an appropriate Quality of Service (QoS) and a better benefit/cost ratio.

- More generally, we speak of Multimedia over IP (MoIP), or the convergence of multiple types of data (voice, video, fax, etc.) on a common network protocol (IP).

VoIP


voce
voce
CODEC: A/D
Compression
Voice Datagram
Header creation
(RTP, UDP, IP, etc)
CODEC: D/A
De-compression
Reorder and buffering
Header processing
IP Network

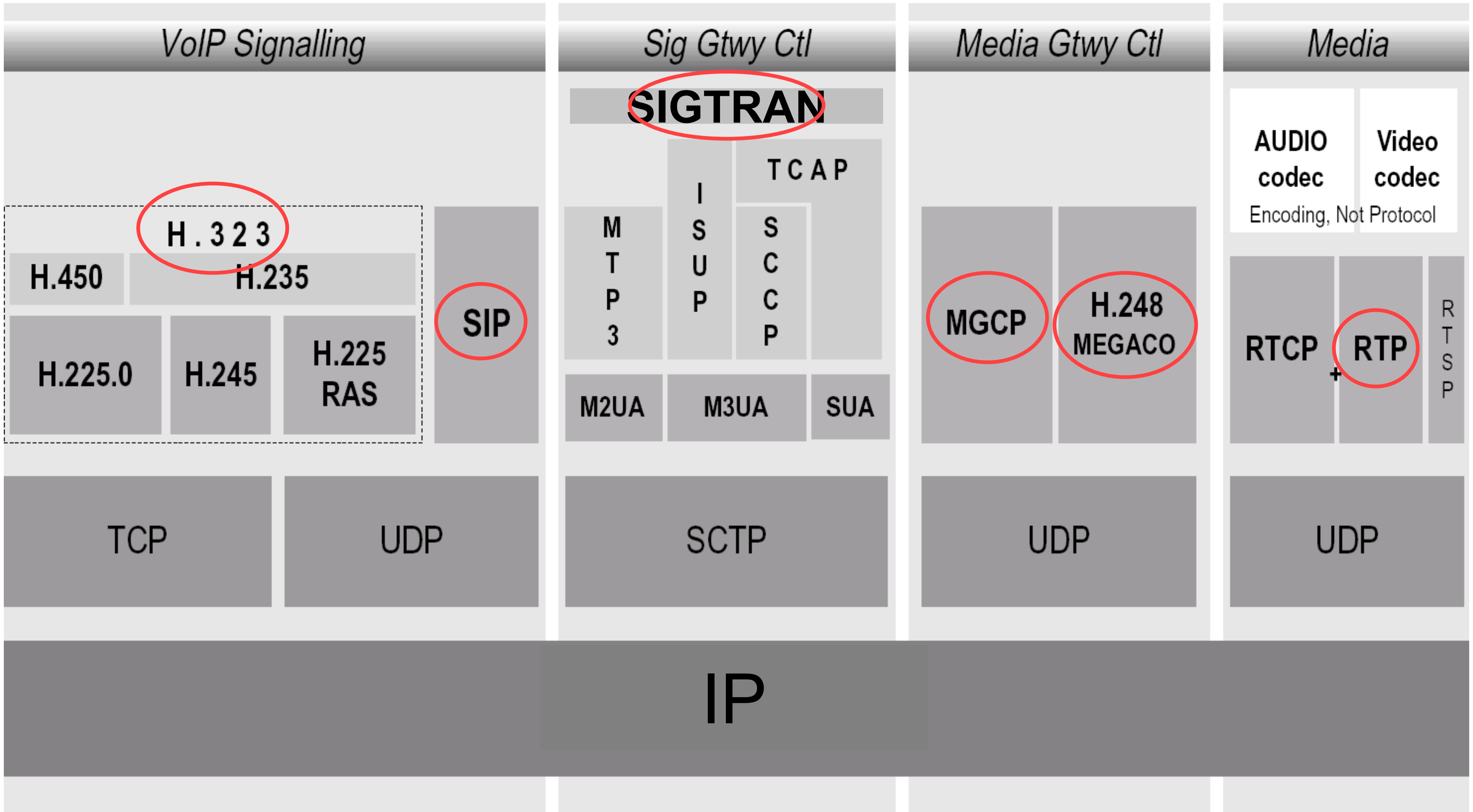
Network architecture



Components

- **Media Gateway Controller (Call Agent, Softswitch)**
  - **Provides call control and call state maintenance logic**
- **Signaling Gateway**
  - **Enables signaling between PSTN and VoIP**
  - **Encapsulates and transports PSTN signaling over IP**
- **Media Gateway (Trunking Gateway)**
  - **Act as a voice gateway between PSTN and VoIP or between different packet networks**
  - **Transforms voice from one transmission format to another**
- **Media Server**
  - **Manipulates and processes voice streams on request from some application (Digital Signal Processing)**
- **Application Server (Feature Server)**
  - **Enables the execution of applications for the provision of services**

VoIP protocols



VoIP protocols: RTP

- The RTP (Realtime Transport Protocol) protocol allows real-time transmission of voice, but does not provide quality of service.
- The voice is digitized, compressed, packetized and sent over the IP infrastructure using the UDP protocol (faster than TCP because it does not provide ACK and retransmissions).
- RTP does not manage QoS but provides additional information to the upper layers so that applications can determine how to treat the packets.

RTP Header

V=2	P	X	CC	M	PT	Sequence Number
Timestamp						
Synchronization source (SSRC) identifier						
Contributing source (SSRC) identifiers						

- Marker Bit (M): indicates the start of a talkspurt.
- Payload Type (PT): specifies the format of the payload; it is equal to 4 for G.723.
- Sequence Number: is used to reorder packets and detect lost packets.
- Timestamp: represents the sampling instant of the first byte of the payload. It is important for synchronization and for jitter calculation.
- SSRC: identifies the source of the RTP packet stream.

RTCP Sender and Receiver Reports

<table>
<tr><td>V=2</td><td>P</td><td>RC</td><td>PT</td><td>Length</td><td rowspan="2">header</td></tr>
<tr><td colspan="5">SSRC of sender</td></tr>
<tr><td colspan="5">NTP Timestamp, most significant word</td><td rowspan="5">sender info</td></tr>
<tr><td colspan="5">NTP Timestamp, least significant word</td></tr>
<tr><td colspan="5">RTP Timestamp</td></tr>
<tr><td colspan="5">Sender’s packet count</td></tr>
<tr><td colspan="5">Sender’s octet count</td></tr>
<tr><td colspan="5">SSRC_1 (SSRC of first source)</td><td rowspan="6">report block</td></tr>
<tr><td colspan="3">Fraction lost</td><td colspan="2">Cumulative number of packets lost</td></tr>
<tr><td colspan="5">Extended highest sequence number received</td></tr>
<tr><td colspan="5">Interarrival jitter</td></tr>
<tr><td colspan="5">Last SR (LSR)</td></tr>
<tr><td colspan="5">Delay since last SR (DLSR)</td></tr>
</table>

VoIP signaling protocols : SIP vs H.323

- H.323 is an ITU (International Telecommunications Unit) standard that specifies components, protocols, and mechanisms for transmitting multimedia data over packet-switched networks without quality-of-service guarantees.
- SIP is an IETF (Internet Engineering Task Force) signaling protocol that is responsible for creating, modifying, and terminating sessions, such as a voice call or a multimedia conference.
- They differ in scalability, simplicity, extensibility, and transmission format.

SIP: Session Initiation Protocol

"[…] an application-layer control (signaling) protocol for creating, modifying, and terminating sessions with one or more participants. These sessions include Internet telephone calls, multimedia distribution, and multimedia conferences."

(RFC 3261)

SIP – Architectural Model

- SIP is similar to HTTP:
  - Client/Server achitecture
  - Request/Response model
  - BNF (Backus-Naur Form) textual coding
  - Codes for response messages:
    - 1xx: Informational → Provisional Response
    - 2xx: success
    - 3xx: Redirection
    - 4xx: Client error
    - 5xx: server Failure
    - 6xx: Global Failure

2xx–6xx → Final Response

in https client server sono entita ben distinti e diverse, invece sip c'è una sorta di bidirezionalita quindi tutti sono sia client che server

SIP entities

- User Agent Client (UAC)
- User Agent server (UAS)
- Registrar
- Location Service
- Redirect server
- Proxy (Proxy server)

They are ***logical*** entities: a particular implementation can combine several of them into a single application.

User Agent (Client e Server)

- User Agent Client: generates a new request (e.g. INVITE)

- User Agent Server: generates a response to a request. The request can be accepted, rejected or assigned to another entity

Registrar and Location Service

- A Registrar
  - Accepts REGISTER requests from UAC
  - Updates the Location Service data accordingly
- A Location Service
  - Contains records of each user's contacts
  - Is consulted by proxies or redirect servers to find out where to contact a usere

Redirect e Proxy Server

- A Redirect Server
  - is a UAS that generates responses in order to direct a UAC to the desired user
- A Proxy acts as an intermediary
  - It covers both the roles of server and client in order to make requests on behalf of other clients
  - It can be stateless or stateful

utile per networking problem

Indirizzi SIP

- Similar to e-mail adresses :
  sip:user@domain

  sip:Giuseppe.Podesta@unige.it
- They follows the rules specified for URI (Uniform Resource Identifier) within RFC 2396

Messaggi SIP - tipologie

- ***Request – from client to server, contains:***
  - a method type, identifies the type of request
  - a Request-URI, indicates the user or service to which the request is addressed

- ***Response – from server to client, contains:***
  - a status code, identifies the type of response
  - a Request-URI, indicates the user or service to which the request is addressed
  - a reason phrase, intended for a human user

Components of SIP messages

- ***Header***
  - headers that appear at the top of the message, mandatory
  - There can be multiple headers of the same type
  - The order of the headers can be significant
- ***Body***
  - the body of the message, optional
  - It can contain the description of a session
  - JAIN SIP allows you to use a string, an array of bytes or any object defined in the SDP specification

SIP Transactions

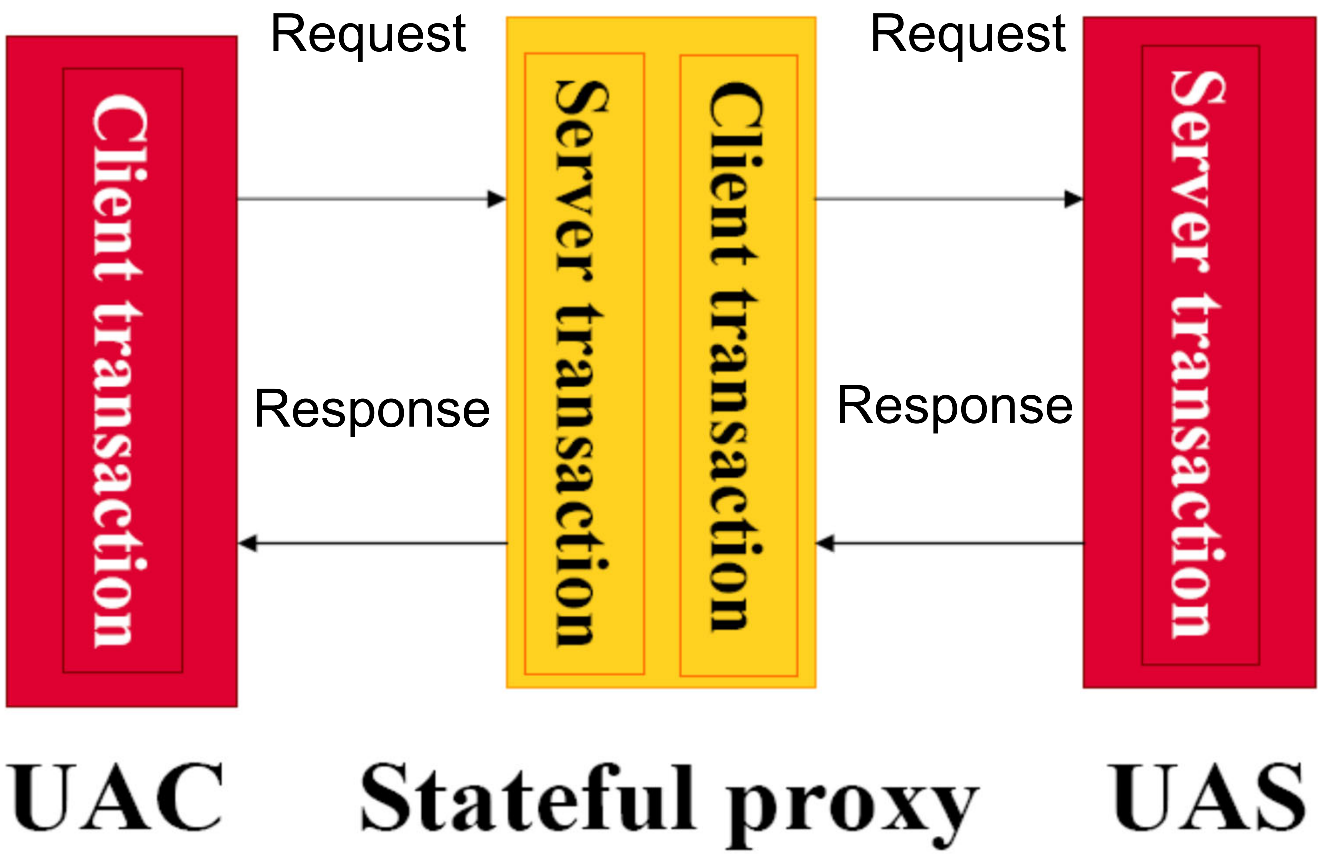
- Message exchange can occur within a transactional context
- SIP defines transactions as the set of a Request and one or more Responses related to the Request
- In the case of multiple Responses, the last one is final, while the others are called provisional
- The protocol associates transactions with finite state machines, which describe and regulate the progress of the state

Client Transaction, Server Transaction

- ***Client e Server are logical roles***
- ***Client Transaction***
  - Sending a Request, Receiving a Response

- ***Server Transaction***
  - – Receiving a Request, sending a Response
- A single application can be characterized by both roles
  - – Es. A softphone acts as a client when making calls and as a server when receiving them.

Another SIP entity with both roles



SIP messagges– an INVITE Request

INVITE sip:bob@biloxi.com SIP/2.0

Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bK776asdhds

Max-Forwards: 70

To: Bob <sip:bob@biloxi.com>

From: Alice <sip:alice@atlanta.com>;tag=1928301774

Call-ID: a84b4c76e66710@pc33.atlanta.com

CSeq: 314159 INVITE

Contact: <sip:alice@pc33.atlanta.com>

Content-Type: application/sdp

Content-Length: 142

<body ……..>

il body è scritto secondo un certo linguaggio non proprio un protocollo

SIP+SDP: request

Request Message line	Description
INVITE sip:bob@acme.com SIP/2.0	Request line: Method type, request URI (SIP address of called party), SIP version.
Via: SIP/2.0/UDP alice_ws.radvision.com	Address of previous hop.
From: Alice A. <sip:alice@radvision.com>	User originating this request.
To: Bob B. <sip:bob@acme.com>	User being invited, as specified originally.
Call-ID: 2388990012@alice_ws.radvision.com	Globally unique ID of this call.
CSeq: 1 INVITE	Command sequence. Identifies transaction.
Subject: Lunch today.	Call subject and/or nature.
Content-Type: application/SDP	Type of body—in this case SDP.
Content-Length: 182	Number of bytes in the body.
	Blank line marks end of SIP headers and beginning of body.
v=0	Version of SDP.
o=Alice 53655765 2353687637 IN IP4 128.3.4.5	Owner/creator and session identifier, session version address type and address.
s=Call from Alice.	Session subject.
c=IN IP4 alice_ws.radvision.com	Connection information.
m=audio 3456 RTP/AVP 0 3 4 5	Media description: type, port, possible formats caller is willing to receive and send.

SDP: Session Description Protocol

linguaggio per scrivere il body non da sapere

v= (protocol version)
o= (owner/creator and session identifier).
s= (session name)
i=* (session information)
u=* (URI of description)
e=* (email address)
p=* (phone number)
c=* (connection information)
b=* (bandwidth information)
One or more <u>time descriptions</u>

Time description

t= (time the session is active)
r=* (zero or more repeat times)

Media description

m= (media name and transport address)
i=* (media title)
c=* (connection information - optional if included at session-level)
b=* (bandwidth information)
k=* (encryption key)
a=* (zero or more media attribute lines)

Messaggi SIP : a Response 200 OK

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP server10.biloxi.com;
    branch=z9hG4bKnashds8;received=192.0.2.3
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com;
    branch=z9hG4bK77ef4c2312983.1;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com;
    branch=z9hG4bK776asdhds ;received=192.0.2.1
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131
<body non mostrato>
```

SIP+SDP: Response

Response Message line	Description
SIP/2.0 200 OK	Status line: SIP version, response code, reason phrase.
Via: SIP/2.0/UDP alice_ws.radvision.com	Copied from request.
From: Alice A. <sip:alice@radvision.com>	Copied from request.
To: Bob B. <sip:bob@acme.com>;tag=17462311	Copied from request. Includes unique tag to identify call-leg.
Call-ID: 2388990012@alice_ws.radvision.com	Copied from request.
CSeq: 1 INVITE	Copied from request.
Content-Type: application/SDP	
Content-Length: 200	
	Blank line marks end of SIP headers and beginning of the body.
v=0	Version of SDP.
o=Bob 4858949 4858949 IN IP4 192.1.2.3	Owner/creator and session identifier, session version address type and address.
s=Lunch	Session subject.
c=IN IP4 machine1.acme.com	Connection information.
m=audio 5004 RTP/AVP 0 3	Description of media streams the receiver of the call is willing to receive and send.

"RTP / AVP"

- RTP / Audio Video Profile, specified in RFC 3551
- Describes how to transmit audio and video in RTP
- Describes the RTP header "payload types" for different encodings

z=* (time zone adjustments)

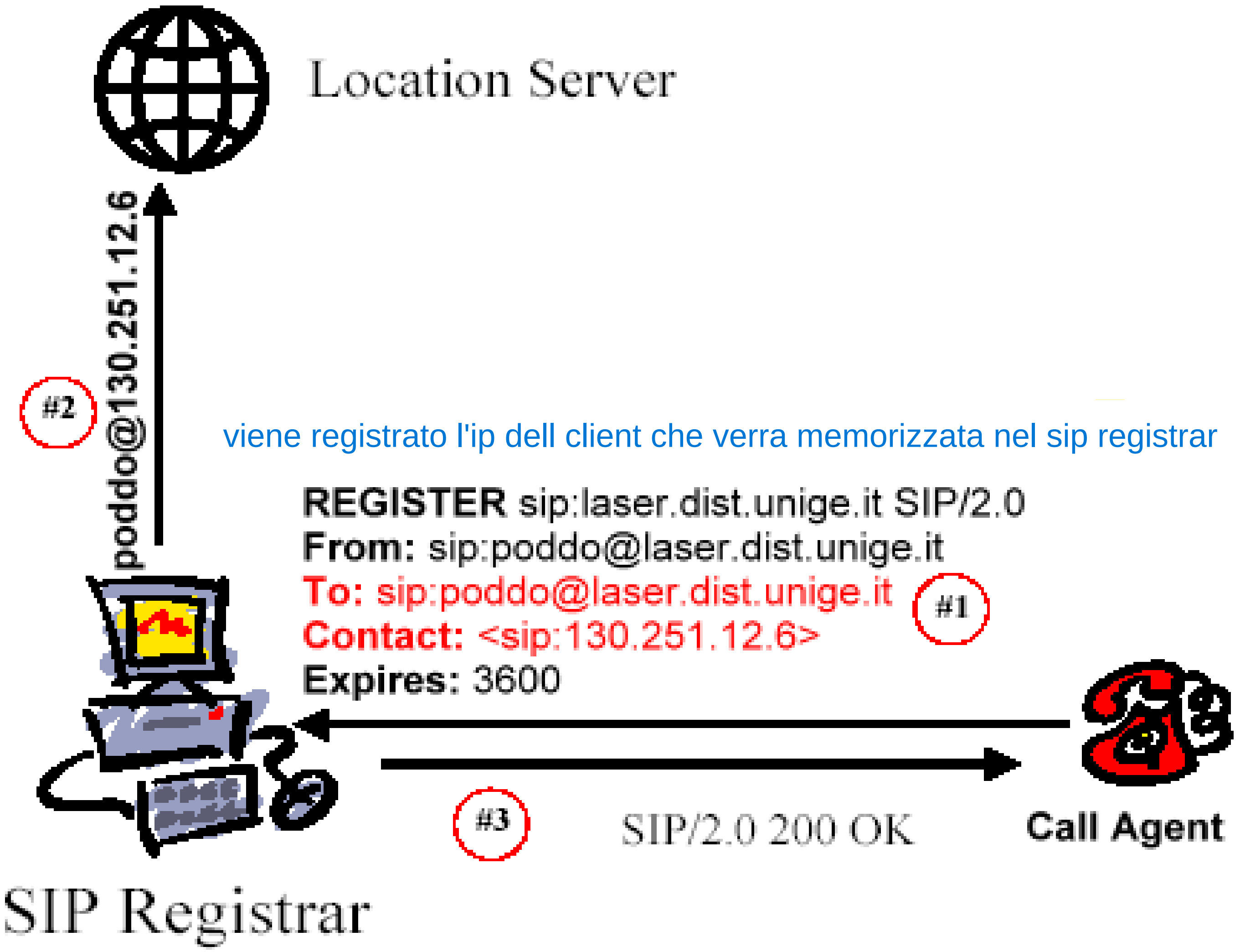
k=* (encryption key)

a=* (zero or more session attribute lines)

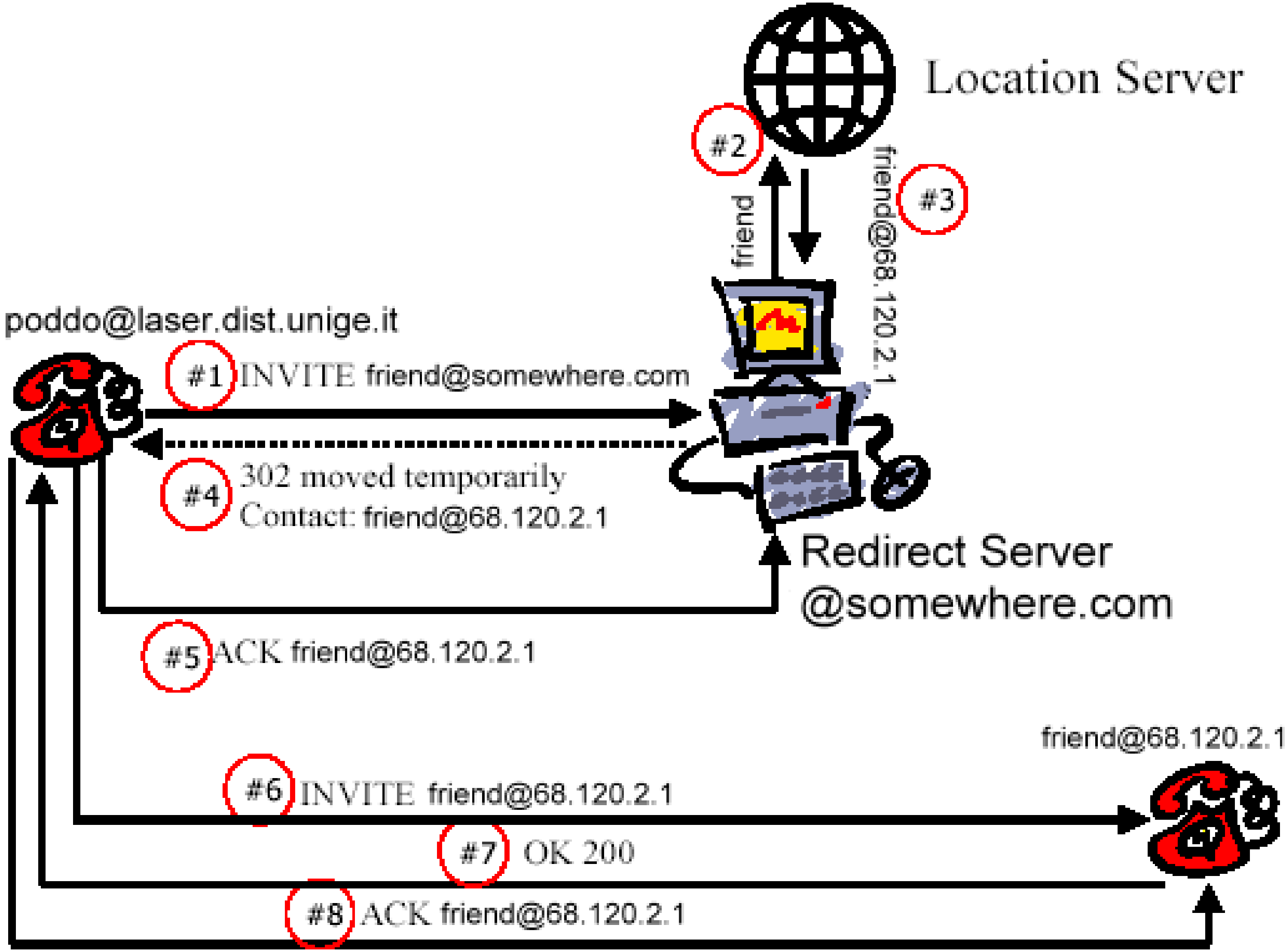
Zero or more <u>media descriptions</u>

PT	encoding name	media type	clock rate (Hz)	channels
0	PCMU	A	8,000	1
1	reserved	A		
2	reserved	A		
3	GSM	A	8,000	1
4	G723	A	8,000	1
5	DVI4	A	8,000	1
6	DVI4	A	16,000	1
7	LPC	A	8,000	1
8	PCMA	A	8,000	1
9	G722	A	8,000	1
10	L16	A	44,100	2
11	L16	A	44,100	1
12	QCELP	A	8,000	1
13	CN	A	8,000	1
14	MPA	A	90,000	(see text)
15	G728	A	8,000	1
16	DVI4	A	11,025	1
17	DVI4	A	22,050	1
18	G729	A	8,000	1
19	reserved	A		
20	unassigned	A		
21	unassigned	A		
22	unassigned	A		
23	unassigned	A		
dyn	G726-40	A	8,000	1
dyn	G726-32	A	8,000	1
dyn	G726-24	A	8,000	1
dyn	G726-16	A	8,000	1
dyn	G729D	A	8,000	1
dyn	G729E	A	8,000	1
dyn	GSM-EFR	A	8,000	1
dyn	L8	A	var.	var.
dyn	RED	A		(see text)
dyn	VDVI	A	var.	1

Registration (REGISTER)



Call (Redirect Mode)



Call (Proxy Mode)