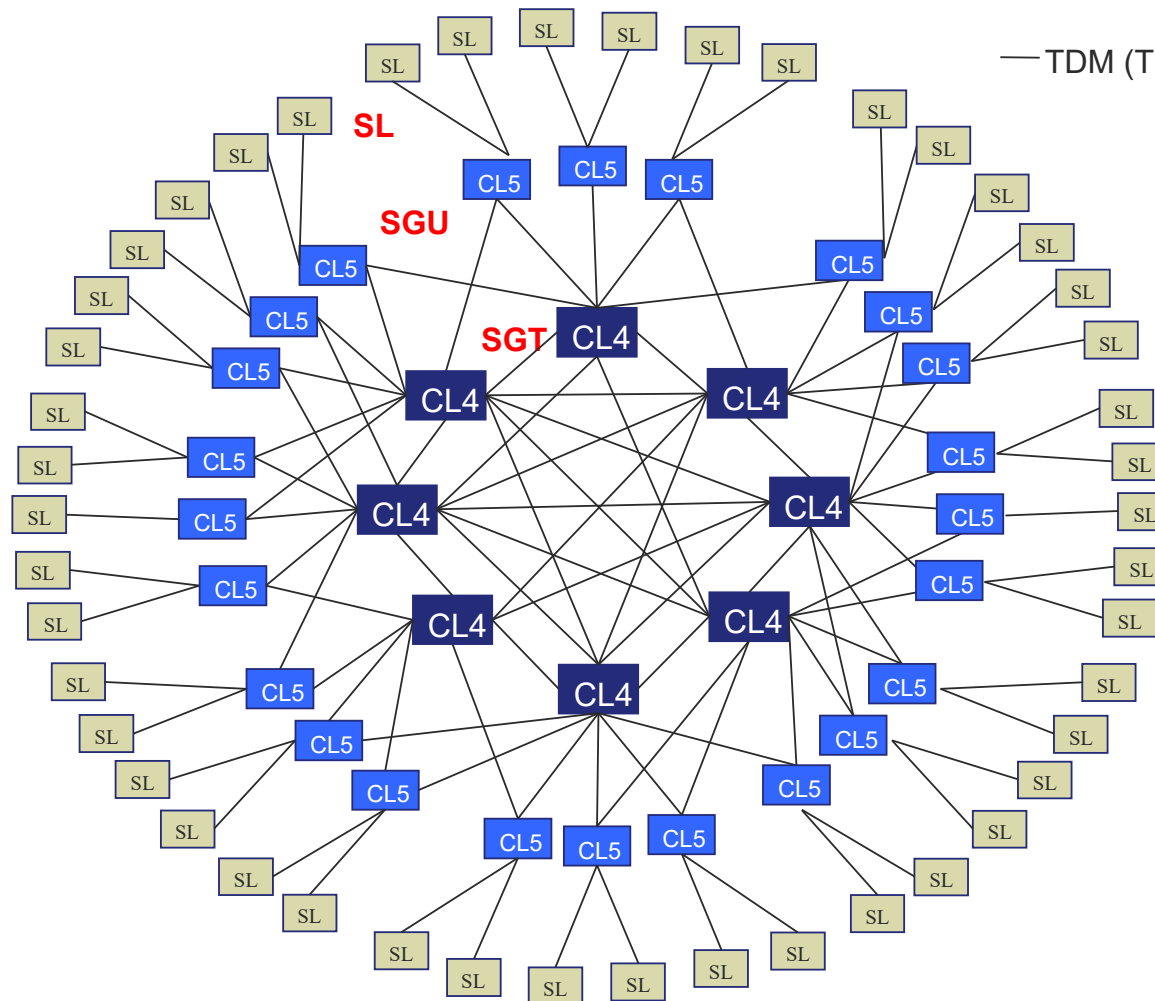


VoIP Architectures

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

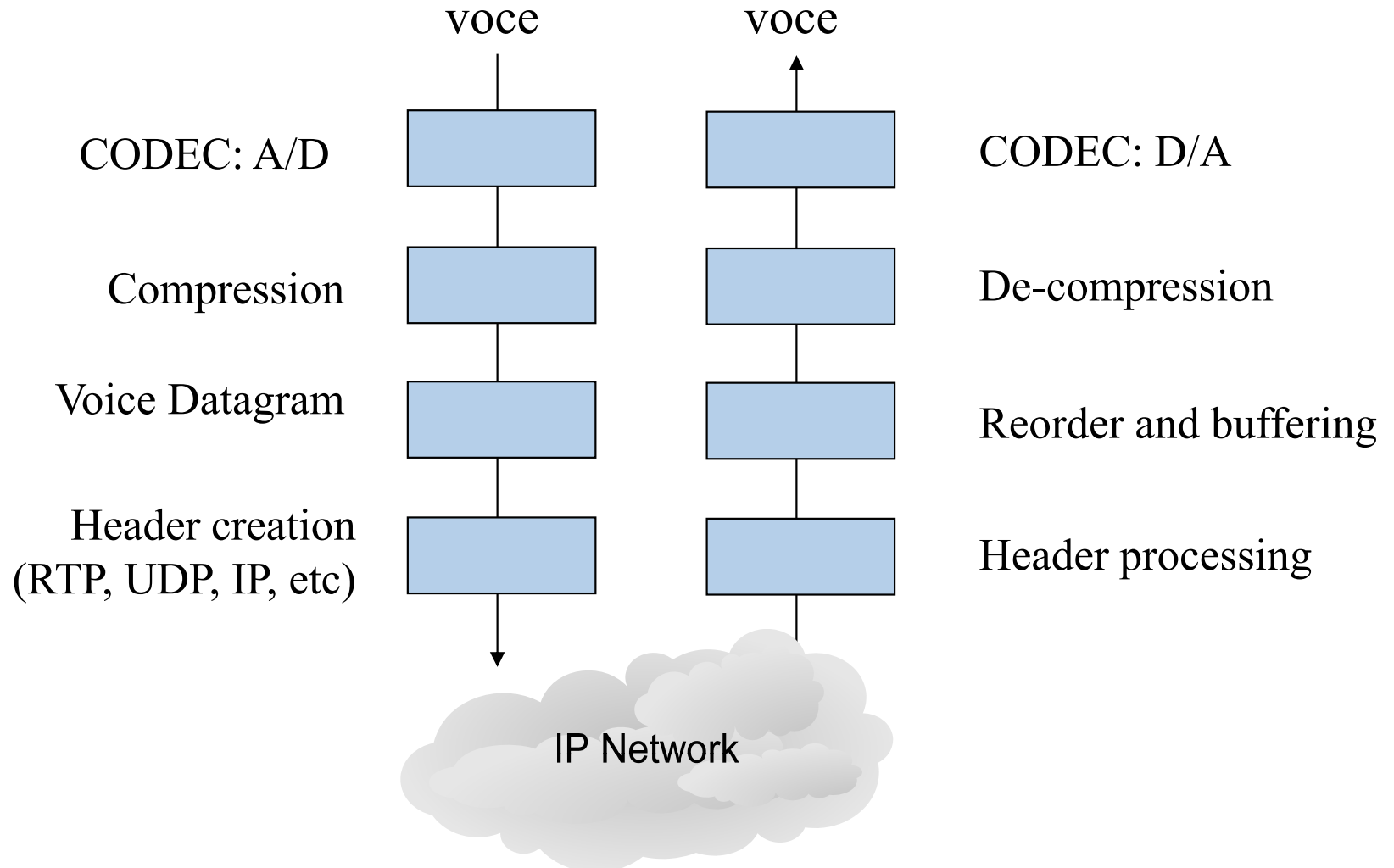


SL: Stadio di Linea
SGU: Stadio di Gruppo Urbano
CL5: Class 5
SGT: Stadio di Gruppo di Transito
CL4: Class 4

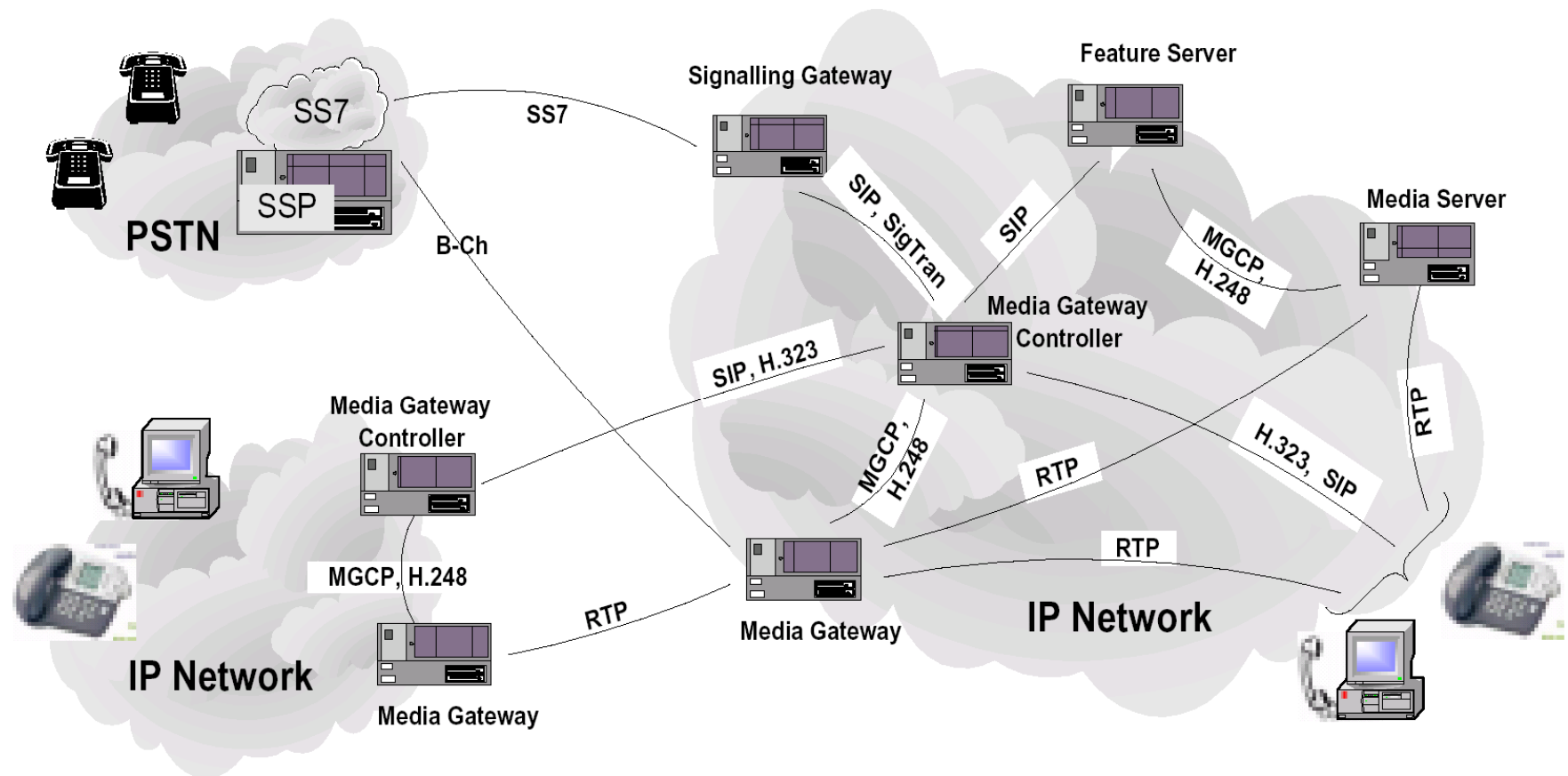
Voice over IP (VoIP)

- 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



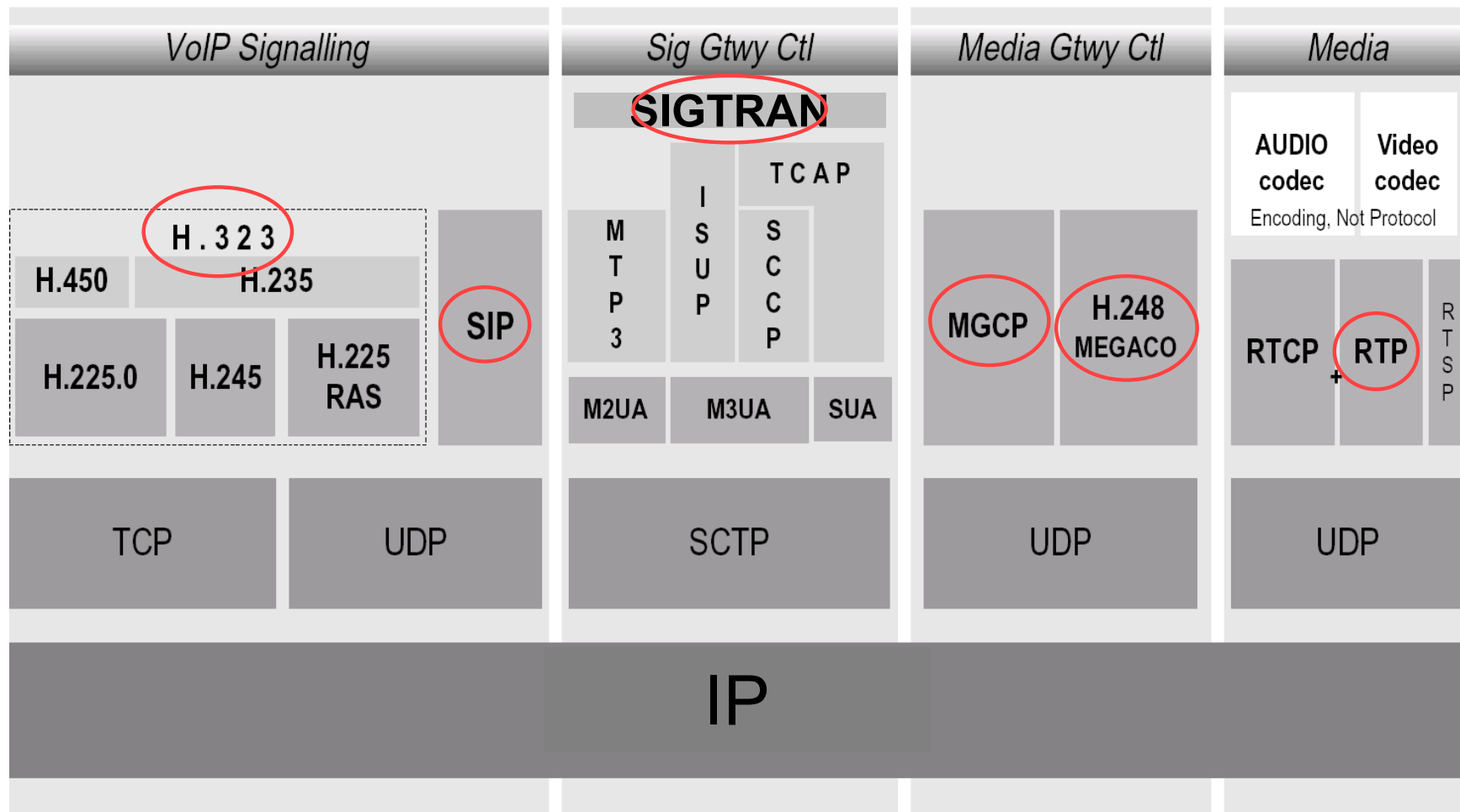
Network architecture



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

V=2	P	RC	PT	Length	header
SSRC of sender					
NTP Timestamp, most significant word					
NTP Timestamp, least significant word					
RTP Timestamp					
Sender's packet count					sender info
Sender's octet count					
SSRC_1 (SSRC of first source)					
Fraction lost	Cumulative number of packets lost				report block
Extended highest sequence number received					
Interarrival jitter					
Last SR (LSR)					
Delay since last SR (DLSR)					

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 architecture
- 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

→ 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 user

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 addresses :
sip:user@domain
sip:Giuseppe.Podesta@unige.it
- They follow 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

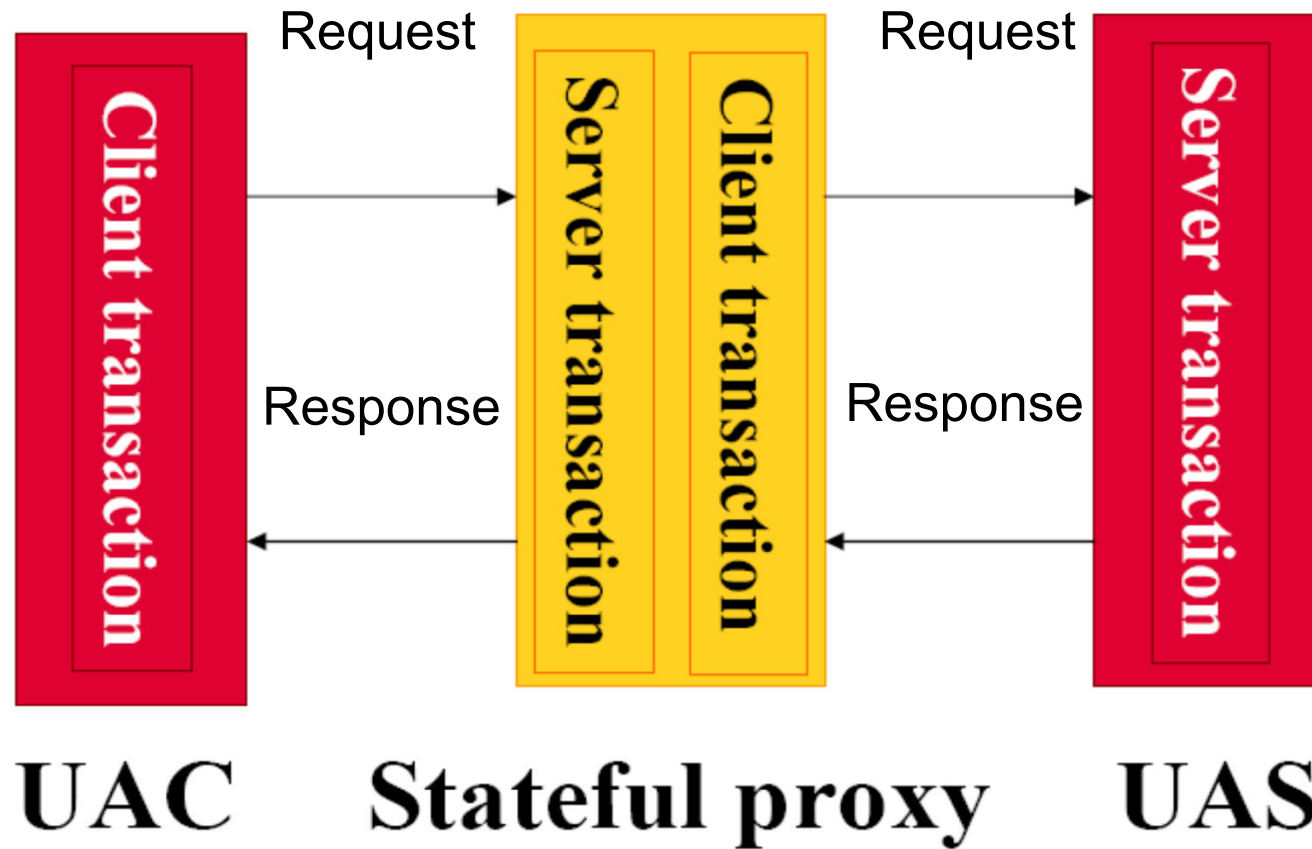
SIP Transactions

- 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 messages– 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 time descriptions

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)

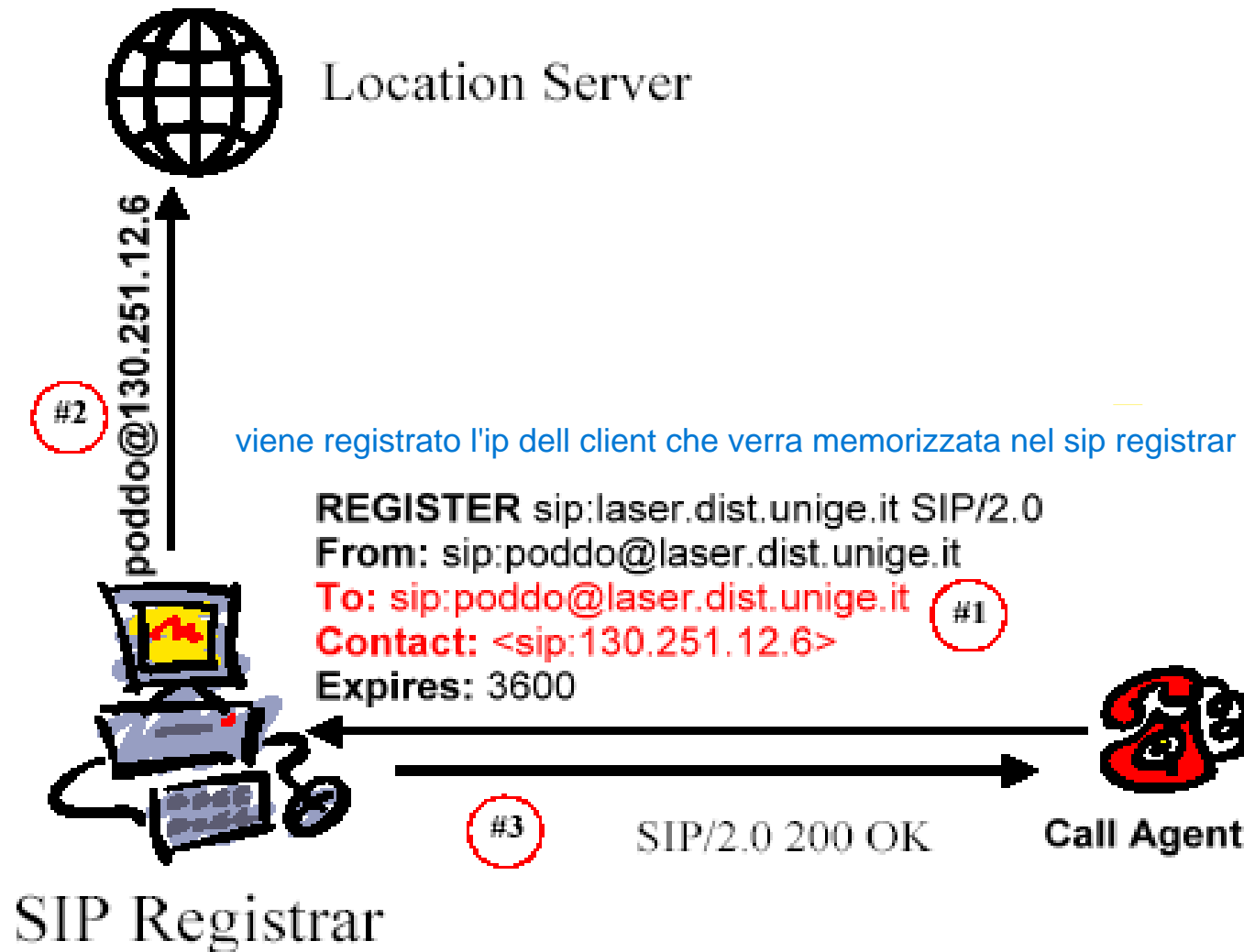
k=* (encryption key)

a=* (zero or more session attribute lines)

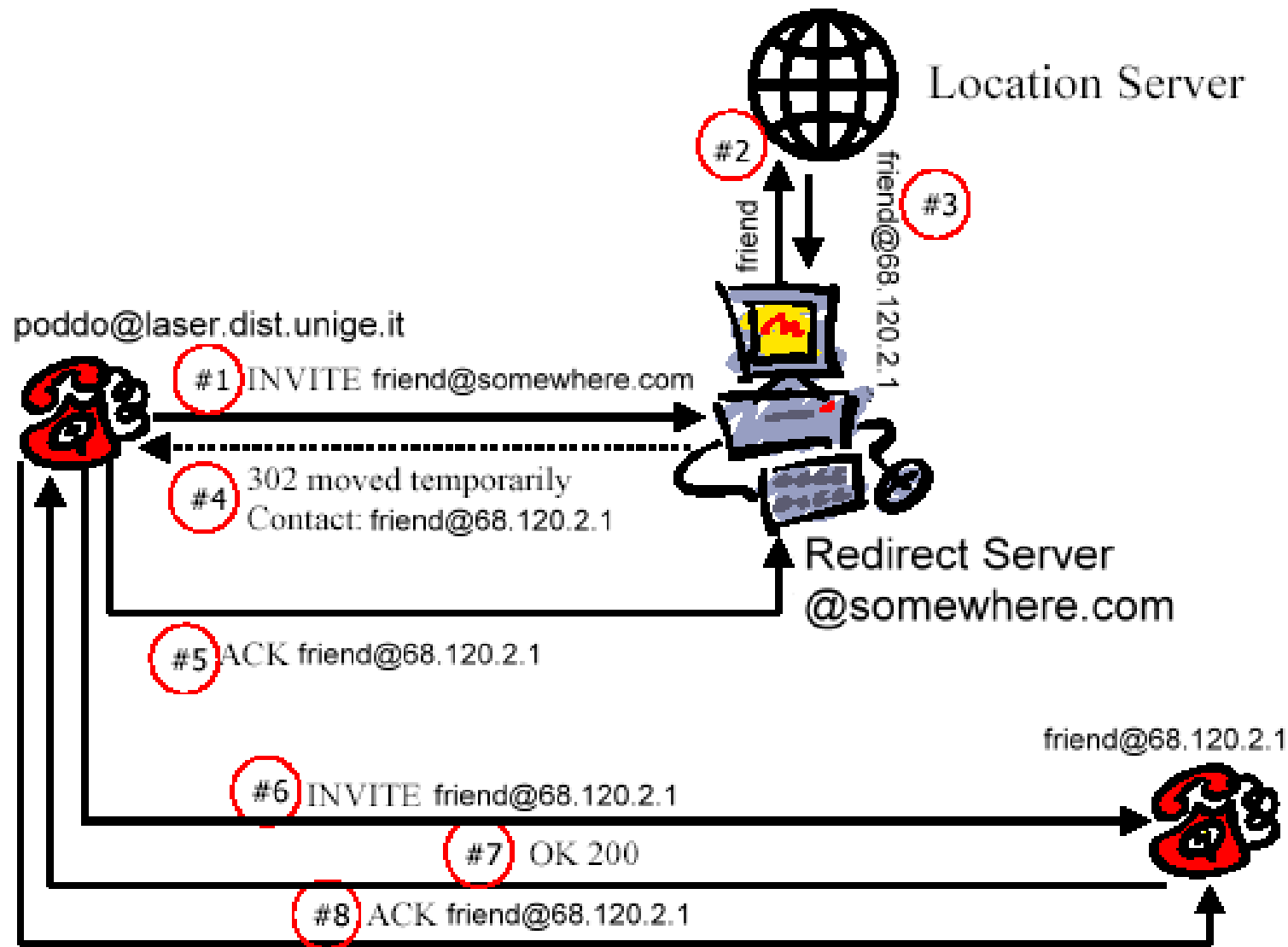
Zero or more media descriptions

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)

