VoIP and SIP
(Session Initiation Protocol)

Par :

Arnaud Morin – Orange

Mail : arnaud.morin@gmail.com

Tel : +33 2 96 07 22 97

Summary

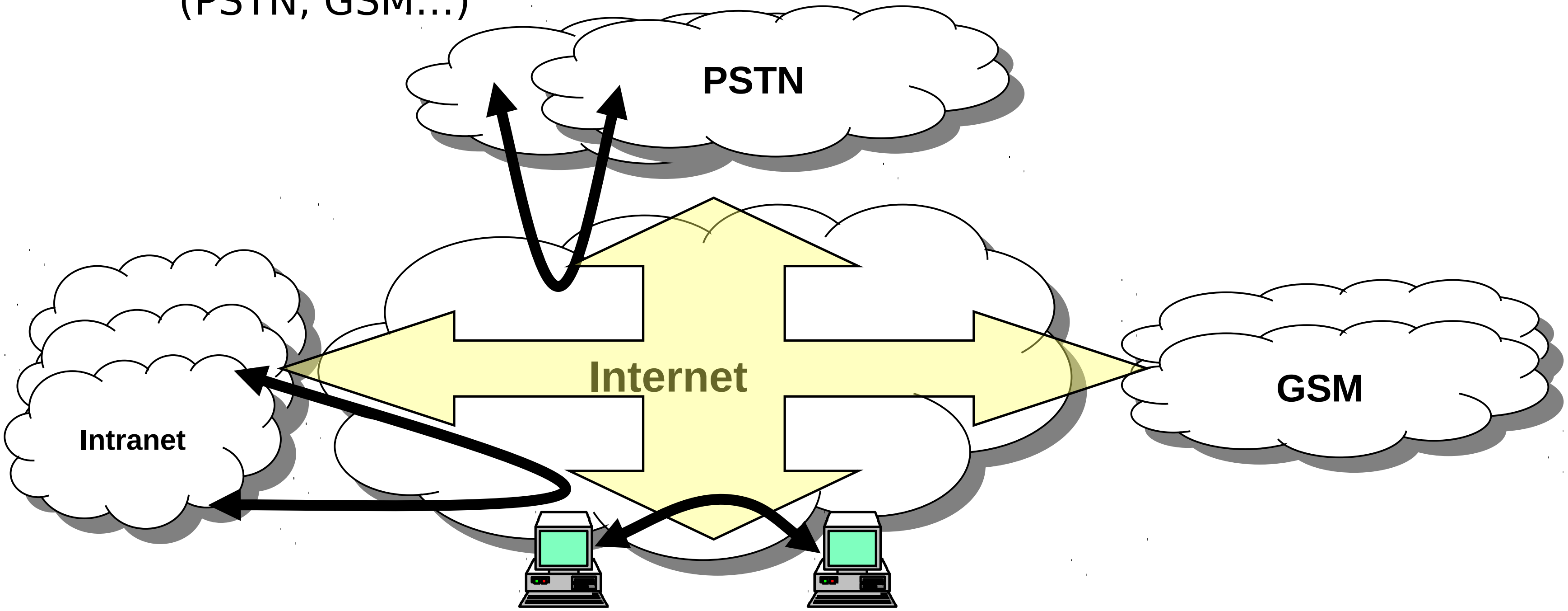
- **What is VoIP ?**
- **Focus on SIP protocol**
- **Focus on media component (RTP/RTCP)**
- **Examples**

Summary

- **What is VoIP ?**
- **Focus on SIP protocol**
- **Focus on media component (RTP/RTCP)**
- **Examples**

What does VoIP stand for ?

- VoIP : Voice over Internet Protocol
  - also called IP Telephony
- VoIP defines mechanisms to
  - Route voice conversations over the Internet or through any other IP-based network (intranet)
  - Interconnect other disparate telecommunication networks (PSTN, GSM…)



Before VoIP

- One line per call

Before VoIP

- Manual circuit switching

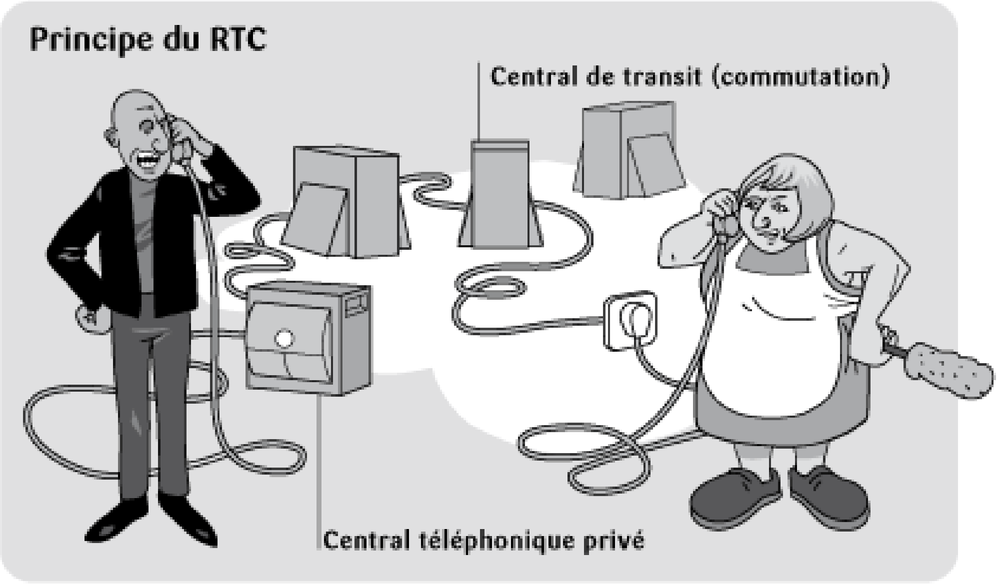
Before VoIP

- Automatic circuit switching

From PSTN to VoIP

Circuit Switching → Packet Switching (data)

Dedicated line → All channels over Internet connection



VoIP needs

- **Signalling protocol** to establish presence, locate user, setup, modify and tear down sessions

  SIP

- **Media Transport protocols** for transmission of packetized audio/video

  RTP + RTCP

- **Media Codecs**

  G711, G729, etc.

- Other supporting protocols like
  - IPv4/IPv6, TCP/UDP…
  - DNS (Domain Name System)
  - RSVP (Resource Reservation Setup Protocol)
  - DIAMETER (Authentication, Accounting, Authorization)

VoIP market

- Telcos
  - VoIP service with broadband offer
    - XXX Box
  - VoIP trunkings to minimize PSTN infrastructure
  - **Target is IMS** (IP Multimedia Subsystem) based on SIP. IMS is a unique infrastructure for fixed and mobile phones
- Service providers
  - Microsoft (Netmeeting, Windows Live Messenger)
  - FaceTime, Skype, Google Talk, Yahoo Messenger…
- Enterprises
  - IP PBX
  - Cisco, Nortel, Alcatel…

Summary

- **What is VoIP ?**
- **Focus on SIP protocol**
  - **History**
  - Basis
  - Basic SIP dialog dissection
  - Registrar function
  - Proxy and Redirect Servers
  - Advanced functions
  - SIP and security
  - Retransmission
  - SIP and NAT/FW
  - Presence and Instant Messaging
- **Focus on media component (RTP/RTCP)**

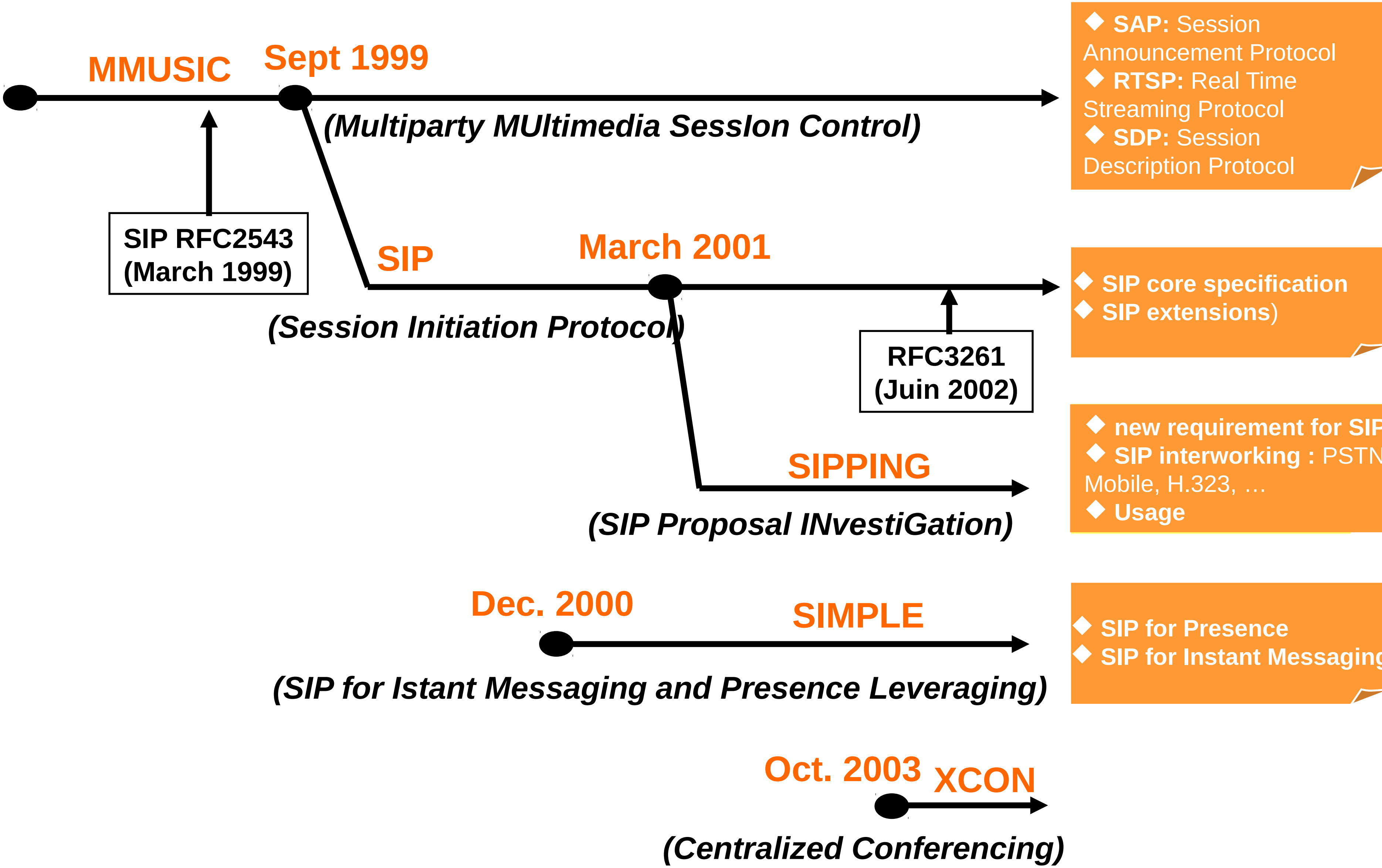
SIP

- SIP = Session Initiation Protocol
- SIP protocol is defined in IETF (Internet Engineering Task Force) under **RFC 3261**
  - **http://tools.ietf.org/html/rfc3261**
- IETF is a large open international community of network designers, operators, vendors, and researchers concerned with the evolution of the Internet architecture and the smooth operation of the Internet. It is open to any interested individual.

  IETF is composed by 8 Areas:
  - Applications Area
  - General Area
  - Internet Area (IPv4, IPv6…)
  - Operations and Management Area
  - Real-time Applications and Infrastructure Area
    - Avt   Audio/Video Transport
    - Mmusic Multiparty Multimedia Session Control
    - Simple SIP for Instant Messaging and Presence Leveraging Extensions
    - Sip Session Initiation Protocol
    - Sipping Session Initiation Proposal Investigation
    - Xcon Centralized Conferencing
  - Routing Area
  - Security Area
  - Transport Area (TCP, UDP…)

  (source : http://www.ietf.org/iesg/area.html)

SIP history

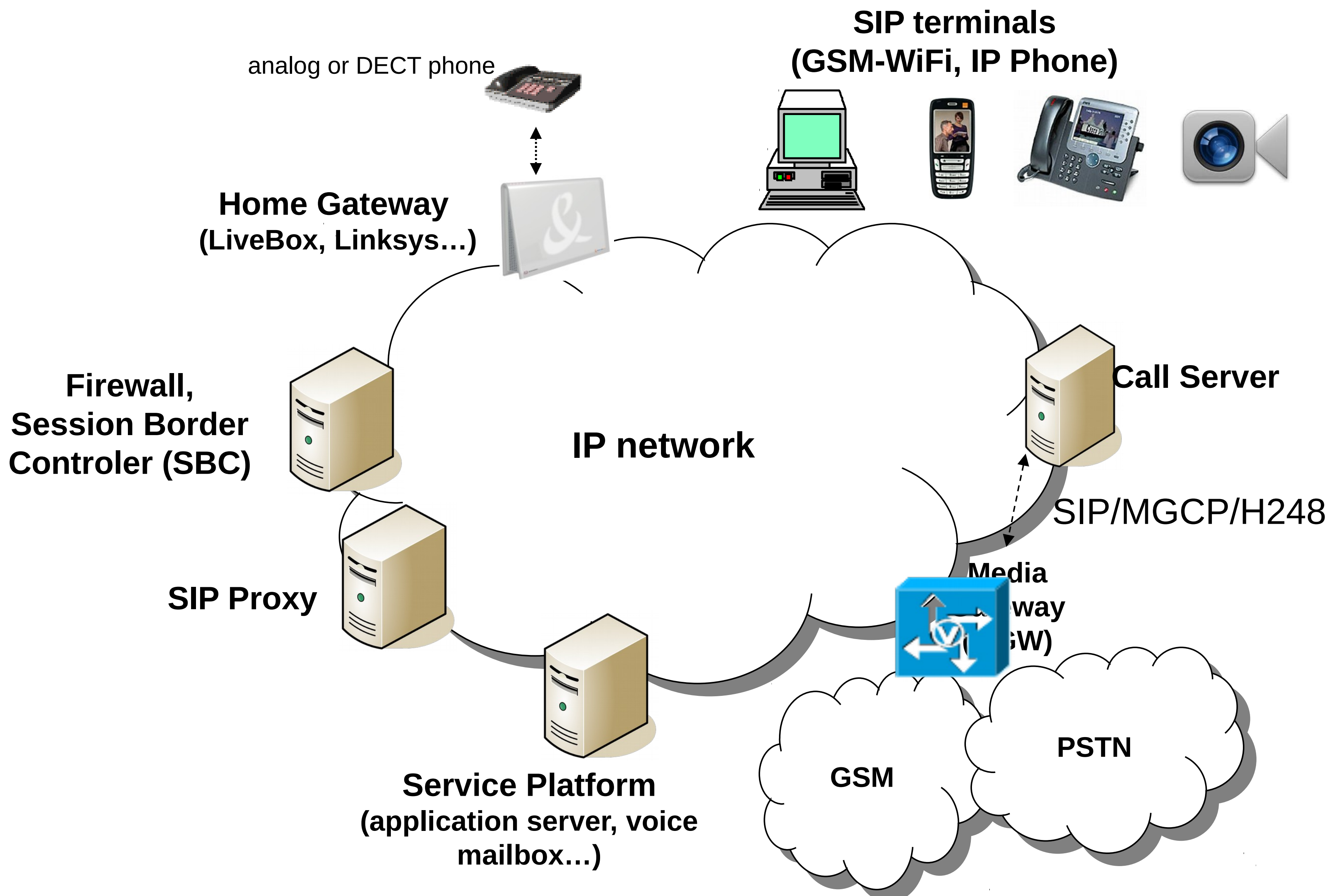


MMUSIC
Sept 1999
(Multiparty MUltimedia SessIon Control)
SIP RFC2543
(March 1999)
◆ SAP: Session Announcement Protocol
◆ RTSP: Real Time Streaming Protocol
◆ SDP: Session Description Protocol
SIP
March 2001
(Session Initiation Protocol)
RFC3261
(Juin 2002)
◆ SIP core specification
◆ SIP extensions)
SIPPING
(SIP Proposal INvestiGation)
◆ new requirement for SIP
◆ SIP interworking : PSTN, Mobile, H.323, …
◆ Usage
Dec. 2000
SIMPLE
(SIP for Istant Messaging and Presence Leveraging)
◆ SIP for Presence
◆ SIP for Instant Messaging
Oct. 2003
XCON
(Centralized Conferencing)

SIP references

- http://www.ietf.org/html.charters/sip-charter.html
- http://www.ietf.org/html.charters/sipping-charter.html
- http://www.cs.columbia.edu/~hgs/sip/
- http://www.sipcenter.com
- http://www.sipforum.org
- http://www.sipit.net
- http://voip-info.org/wiki-SIP

Where could I find SIP ?



SIP applications

- Audio / Video
  - FaceTime
- Media Streaming
  - Live555 library
- Share application
  - Desktop sharing applications
- Messaging
  - Chat with Rich Communication Suite system
- Conferencing

Summary

- **What is VoIP ?**
- **Focus on SIP protocol**
  - History
  - **SIP Basis**
  - Basic SIP dialog dissection
  - Registrar function
  - Proxy and Redirect Servers
  - Advanced functions
  - SIP and security
  - Retransmission
  - SIP and NAT/FW
  - Presence and Instant Messaging
- **Focus on media component (RTP/RTCP)**

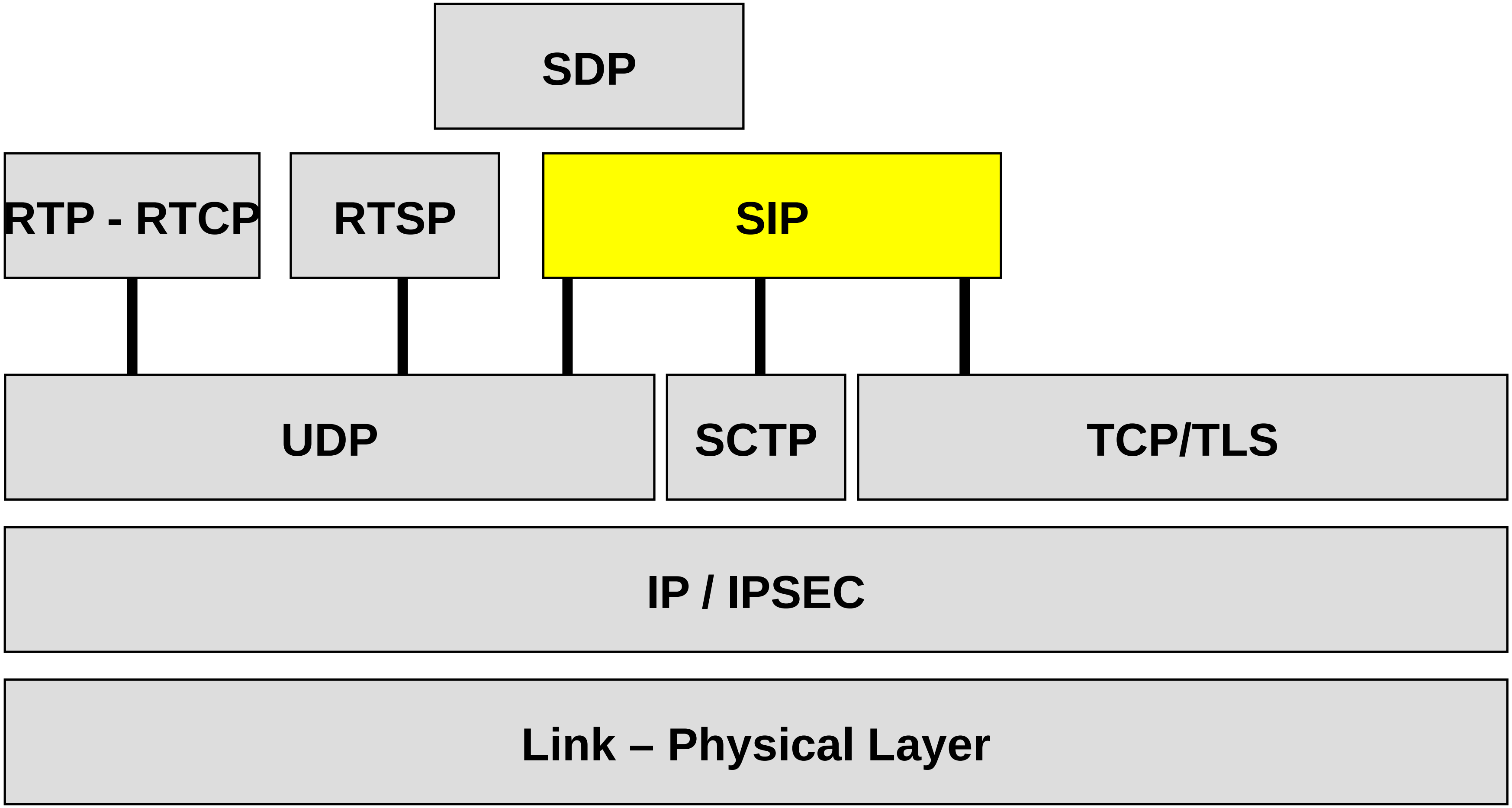
SIP provides ...

- **User location**: determination of the end system to be used for communication
- **User availability**: determination of the willingness of the called party to engage in communications
- **User capabilities**: determination of the media and media parameters to be used
- **Session setup**: "ringing", establishment of session parameters at both called and calling party
- **Session management**: including transfer and termination of sessions, modifying session parameters, and invoking services

… but SIP is not …

- A transport protocol
  - Real-time Transport Protocol (RTP) is a transport protocol
- A quality of service protocol
  - Resource Reservation Protocol (RSVP) is a QoS protocol
- A media control protocol
  - RTP Control Protocol (RTCP) is a media control protocol

SIP – OSI model



Some other protocols around SIP in IETF Real Time Multimedia Architecture

RTP

Real Time Transport Protocol RFC 3550

- Packet Sequencing
- Packet Intra-media Synchronization
- Payload Type Identification
- Frame Indication

RTCP

Real Time Control Protocol

- QoS Feedback
- Audio-Video Intra-media Synchronization : lip-sync
- Participant Identification : address, name…
- Session Control

RTSP

Real Time Streaming Protocol RFC 2326

- Stored Media
- Client-Server
- Stateful
- URL rtsp://server.com/movies/twister:5080
- Methods : DESCRIBE, SETUP, PLAY, RECORD, PAUSE, TEARDOW

SDP

Session Description Protocol RFC 4566

- Media Stream, Addresses, Ports, RTP Payload…

SIP devices types (Terminology)

- **User Agent Client** (UAC)

  Endpoint that sends a request
- **User Agent Server** (UAS)

  Endpoint that receives a request and sends a response
- **Registrar**

  Server that registers clients and stores user addresses in location server
- **Redirect Server**

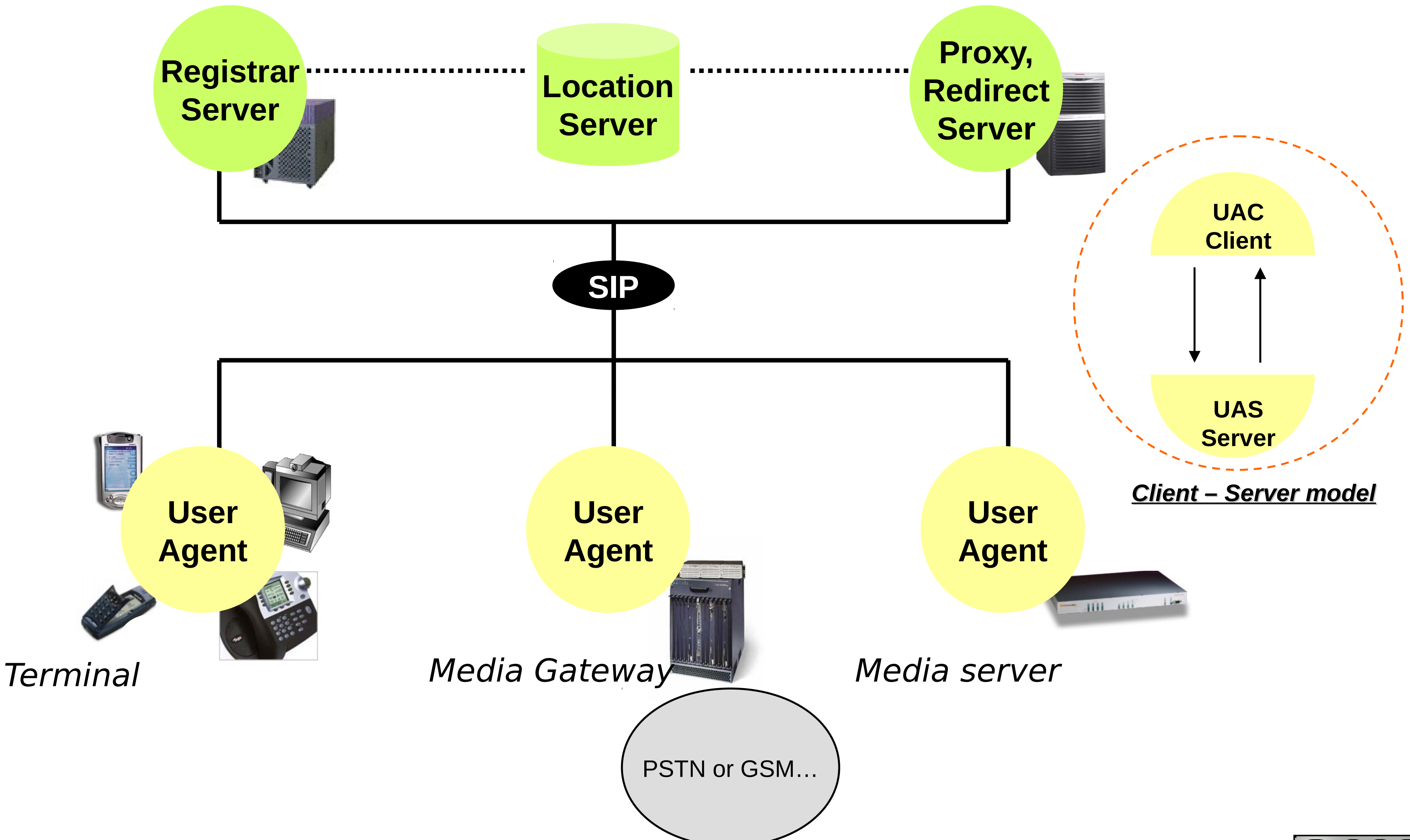
  Server that returns specific response (3xx) in order to redirect client to another destination
- **Proxy Server**

  Both UAC/UAS. It interprets, and rewrites specific parts of a request message before forwarding it
- **Back To Back User Agent** (B2B)

  Unlike a proxy server, it maintains dialog state and must participate in all requests sent on the dialogs it has established
- **Media Gateway** (MGW)

  Translation unit between disparate telecommunications networks (Ex: VoIP<>PSTN ; VoIP<>GSM…)

SIP reference architecture



SIP main characteristics

- SIP re-uses HTTP 1.1
  - Text-based protocol (UTF-8 charset)
- SIP is independent from the type of session to establish
  - Audio, video, text, game…
- SIP is based on an HTTP-like request/response transaction model
- SIP uses 2 kinds of message
  - Request from a client to a server
  - Response from a server to a client
- SIP is independent from transport layer protocol
  - UDP (by default), TCP, SCTP, TLS, IPsec…
  - Default port is 5060
- SIP Request and Response messages use the same format

Message structure

startLine : RequestLine | status Line

Header (general / request / response / entity)

From: …
To: …
…

Blank

Body

SDP (Multimedia Session Description)
/ **ISUP** (encapsulated ISUP – ie SIP-T)
/ **XML** (ex. Presence document)
…

Request-Line = Method Request-URI SIP-Version

REQUEST

Status-Line = SIP-Version Status-Code Reason-Phrase

RESPONSE

INVITE sip:Rob@orange.com SIP/2.0

Via: SIP/2.0/UDP host1.francetelecom.com:5060
Date: Wed, 04 Oct 2000 07:14:34 GMT
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>

…

v=0
o=Tin 562413 562413 IN IP4 194.240.47.217
s=phone call
...

SIP/2.0 200 OK

Via: SIP/2.0/UDP host1.francetelecom.com:5060
Date: Wed, 04 Oct 2000 07:14:34 GMT
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>

…

v=0
o=Tin 562413 562413 IN IP4 194.240.47.200
s=phone call
...

SIP Requests (Methods)

REGISTER	Registration of UA location	RFC 3261
INVITE	Request a party to participate in a service session	RFC 3261
ACK	Acknowledgement of the reception of the final response	RFC 3261
PRACK	Acknowledgement of reception of the provisional response	RFC 3262
OPTIONS	Request server capacities	RFC 3261
BYE	Terminatation of a session	RFC 3261
CANCEL	Cancellation of a pending request	RFC 3261
UPDATE	Modify characteristics of an active session	RFC 3311
INFO	Request for session related control information that is generated during a session	RFC 2976
REFER	Call Transfer	RFC 3515
SUBSCRIBE / NOTIFY	Subscription to specific event - Notification of event	RFC 3265
PUBLISH	Request to publish presence information	
MESSAGE	Instant Messaging	RFC 3428

SIP Responses status code

Class	Description	Type
1xx	Informational	Provisional
2xx	Success	Final
3xx	Redirection	Final
4xx	Client Error	Final
5xx	Server Failure	Final
6xx	Global Failure	Final

- Provisional Response indicate progress, but that does not terminate a SIP transaction
- Final Response terminates a SIP transaction
- Examples :
  - *100 Trying, 180 Ringing, 182 Queued*
  - *200 OK,*
  - *301 Moved Permanently,*
  - *401 Unauthorized, 486 Busy Here,*
  - *500 Server Internal Error, 505 Version Not Supported,*
  - *603 Decline, 606 Not Acceptable*

SIP headers

Header field	where	proxy	ACK	BYE	CAN	INV	OPT	REG
Accept	R		–	o	–	o	m*	o
Accept	2xx		–	–	–	o	m*	o
Accept	415		–	c	–	c	c	c
Accept-Encoding	R		–	o	–	o	o	o
Accept-Encoding	2xx		–	–	–	o	m*	o
Accept-Encoding	415		–	c	–	c	c	c
Accept-Language	R		–	o	–	o	o	o
Accept-Language	2xx		–	–	–	o	m*	o
Accept-Language	415		–	c	–	c	c	c
Alert-Info	R	ar	–	–	–	o	–	–
Alert-Info	180	ar	–	–	–	o	–	–
Allow	R		–	o	–	o	o	o
Allow	2xx		–	o	–	m*	m*	o
Allow	r		–	o	–	o	o	o
Allow	405		–	m	–	m	m	m
Authentication-Info	2xx		–	o	–	o	o	o
Authorization	R		o	o	o	o	o	o
Call-ID	c	r	m	m	m	m	m	m
Call-Info		ar	–	–	–	o	o	o
Contact	R		o	–	–	m	o	o
Contact	1xx		–	–	–	o	–	–
Contact	2xx		–	–	–	m	o	o
Contact	3xx	d	–	o	–	o	o	o
Contact	485		–	o	–	o	o	o
Content-Disposition			o	o	–	o	o	o
Content-Encoding			o	o	–	o	o	o
Content-Language			o	o	–	o	o	o
Content-Length		ar	t	t	t	t	t	t
Content-Type			*	*	–	*	*	*
CSeq	c	r	m	m	m	m	m	m
Date		a	o	o	o	o	o	o
Error-Info	300-699	a	–	o	o	o	o	o
Expires			–	–	–	o	–	o
From	c	r	m	m	m	m	m	m
In-Reply-To	R		–	–	–	o	–	–
Max-Forwards	R	amr	m	m	m	m	m	m
Min-Expires	423		–	–	–	–	–	m
MIME-Version			o	o	–	o	o	o
Organization		ar	–	–	–	o	o	o

Header field	where	proxy	ACK	BYE	CAN	INV	OPT	REG
Priority	R	ar	–	–	–	o	–	–
Proxy-Authenticate	407	ar	–	m	–	m	m	m
Proxy-Authenticate	401	ar	–	o	o	o	o	o
Proxy-Authorization	R	dr	o	o	–	o	o	o
Proxy-Require	R	ar	–	o	–	o	o	o
Record-Route	R	ar	o	o	o	o	o	–
Record-Route	2xx,18x	mr	–	o	o	o	o	–
Reply-To			–	–	–	o	–	–
Require		ar	–	c	–	c	c	c
Retry-After	404,413,480,486		–	o	o	o	o	o
	500,503		–	o	o	o	o	o
	600,603		–	o	o	o	o	o
Route	R	adr	c	c	c	c	c	c
Server	r		–	o	o	o	o	o
Subject	R		–	–	–	o	–	–
Supported	R		–	o	o	m*	o	o
Supported	2xx		–	o	o	m*	m*	o
Timestamp			o	o	o	o	o	o
To	c(1)	r	m	m	m	m	m	m
Unsupported	420		–	m	–	m	m	m
User-Agent			o	o	o	o	o	o
Via	R	amr	m	m	m	m	m	m
Via	rc	dr	m	m	m	m	m	m
Warning	r		–	o	o	o	o	o
WWW-Authenticate	401	ar	–	m	–	m	m	m
WWW-Authenticate	407	ar	–	o	–	o	o	o

More info on:
http://tools.ietf.org/html/rfc3261#section-20

SIP headers

- The "where" column describes the request and response types in which the header field can be used. Values in this column are:
  - R: header field may only appear in requests;
  - r: header field may only appear in responses;
  - 2xx, 4xx, etc.: A numerical value or range indicates response
  - codes with which the header field can be used;
  - c: header field is copied from the request to the response.
  - An empty entry in the "where" column indicates that the header field may be present in all requests and responses.

- For others colums, see http://tools.ietf.org/html/rfc3261#section-20

SIP address

- Based on URI (Uniform Resource Identifier) and defined in RFC 1630

  RFC 1630 defines the syntax used by the World-Wide Web initiative to encode the names and addresses of objects on the Internet

- URI defines a generic format

  <scheme name> : <hierarchical part> [? <query>] [# <fragment>]

  Ex: http://example.com/articles/recent/

sip URI (or tel URI)

sip:chantal.martin@francetelecom.com

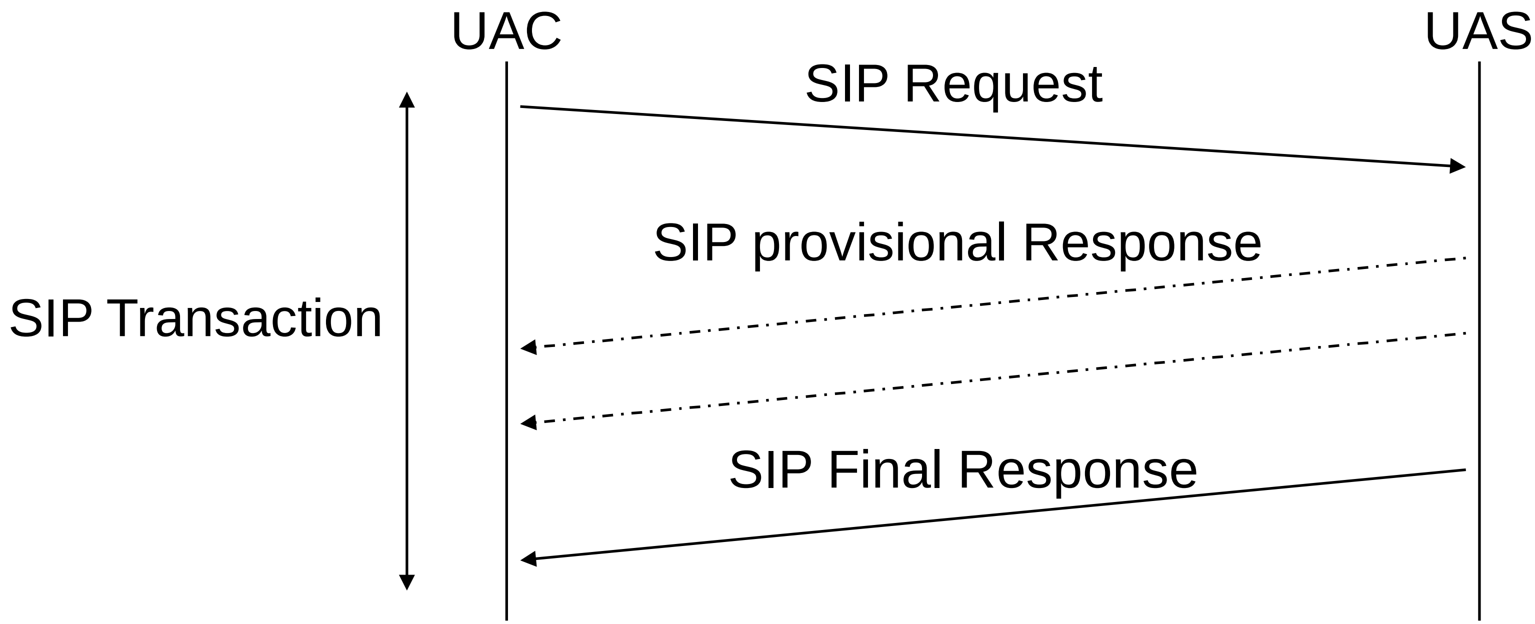
sip:chantal.martin@162.23.21.24

sip:+33296053017@orange.com; user=phone

tel:+33296053017

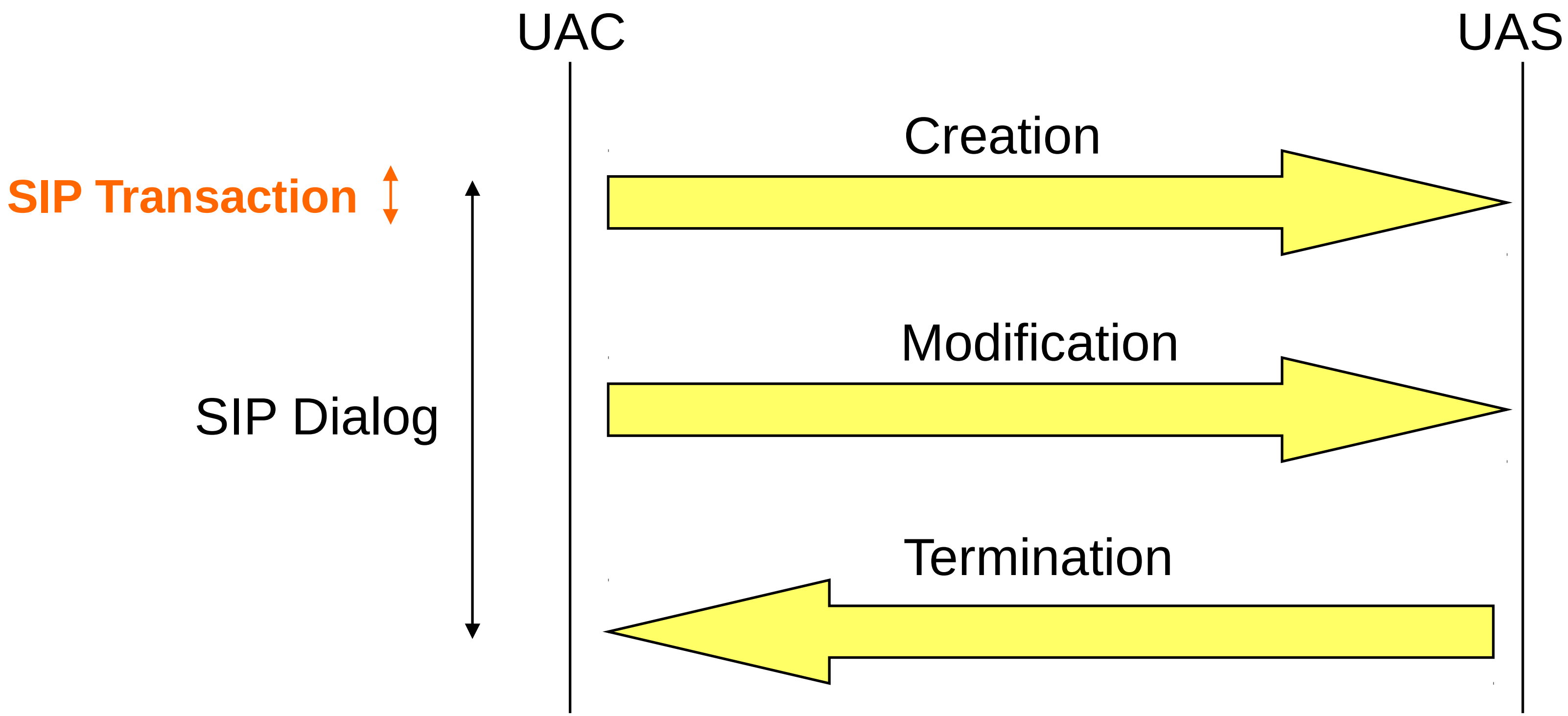
SIP transaction

- SIP is a transactional protocol
- A transaction consists of a single request and any responses to that request
- A transaction is identified by
  - The **branch** parameter of the Via
    ex : Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bK776asdhds
  - The **CSeq** parameter
    ex : CSeq: 4711 INVITE

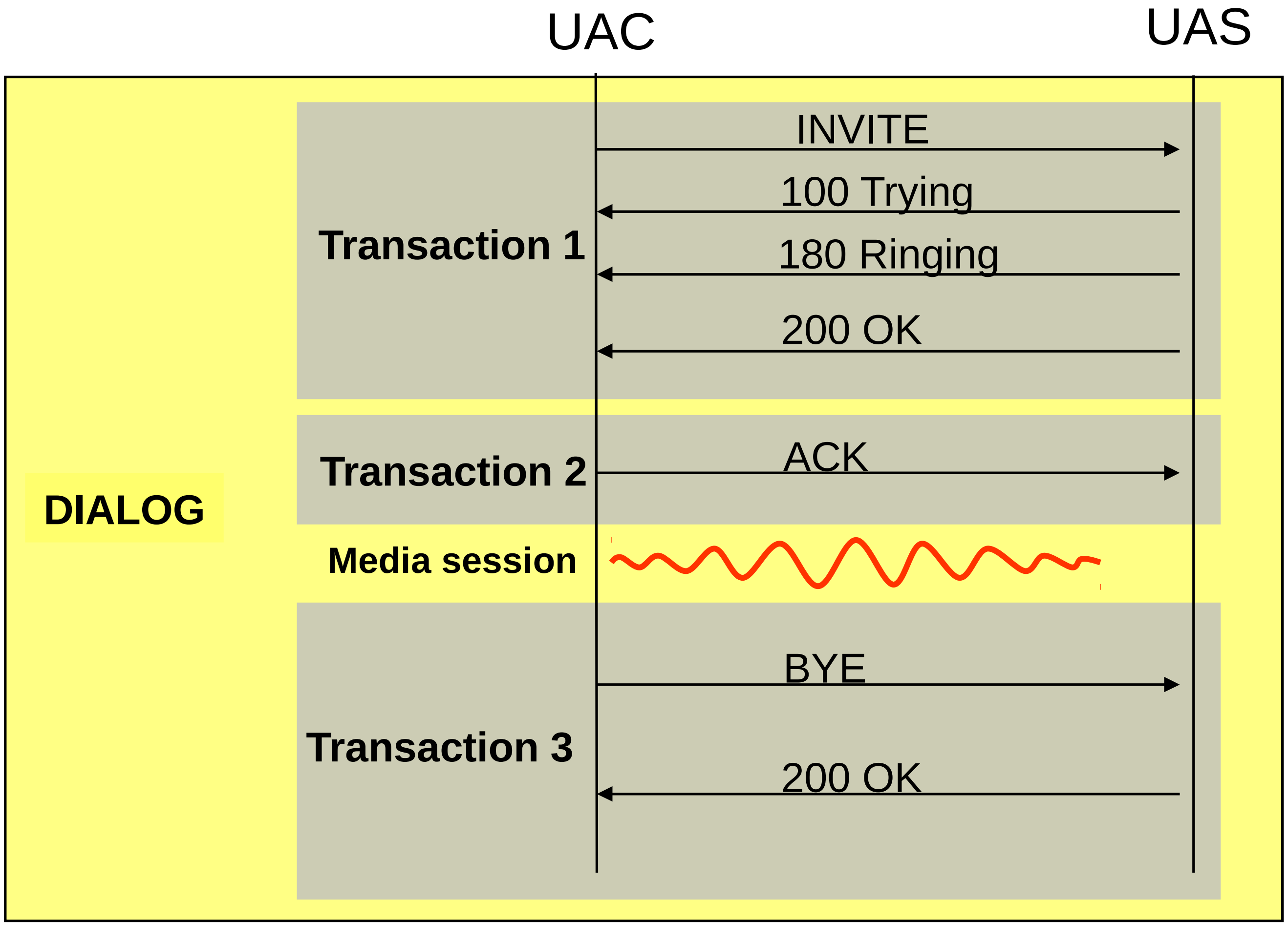


SIP dialog

- SIP dialog is a peer-to-peer relationship between 2 UA that persists for some time
- Dialog state can change only after a new transaction
- A dialog is identified by the triplet: { **Call-ID** ; **From tag** ; **To tag** }
- Cseq is incremented after each transaction



Basic Call Flow



Finite State Machine (FSM)

- 4 FSM are defined in RFC 3261 for transaction behavior
  - Invite client transaction
  - Non Invite client transaction
  - Invite server transaction
  - Non Invite server transaction
- Each FSM consists of 4 states
  - Calling
  - Proceeding
  - Completed
  - Terminated

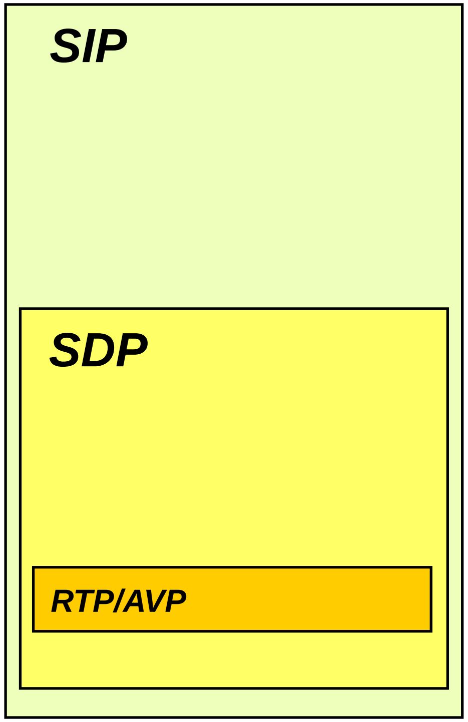
FSM Invite Client Transaction

```
                               |INVITE from TU
             Timer A fires     |INVITE sent
             Reset A,          V                      Timer B fires
             INVITE sent +-----------+                or Transport Err.
               +---------|           |---------------+inform TU
               |         |  Calling  |               |
               +-------->|           |-------------->|
                         +-----------+ 2xx           |
                            |  |       2xx to TU     |
                            |  |1xx                  |
    300-699 +---------------+  |1xx to TU            |
   ACK sent |                  |                     |
resp. to TU |  1xx             V                     |
            |  1xx to TU  -----------+               |
            |  +---------|           |               |
            |  |         |Proceeding |-------------->|
            |  +-------->|           | 2xx           |
            |            +-----------+ 2xx to TU     |
            |       300-699    |                     |
            |       ACK sent,  |                     |
            |       resp. to TU|                     |
            |                  |                     |      NOTE:
            |  300-699         V                     |
            |  ACK sent  +-----------+Transport Err. |  transitions
            |  +---------|           |Inform TU      |  labeled with
            |  |         | Completed |-------------->|  the event
            |  +-------->|           |               |  over the action
            |            +-----------+               |  to take
            |              ^   |                     |
            |              |   | Timer D fires       |
            +--------------+   | -                   |
                               |                     |
                               V                     |
                         +-----------+               |
                         |           |               |
                         | Terminated|<--------------+
                         |           |
                         +-----------+
```

Figure 5: INVITE client transaction

Protocol(s)

- 3 independant protocols
  - Level 1 : Session control
    - SIP
  - Level 2 : Service session description
    - SDP
  - Level 3 : Media description
    - RTP/AVP



SDP: Session Description Protocol

- **SDP is defined in RFC 2327**
- Description of the service session
- Description format : { <type> = <value> }
- 3 subsets of description:
  - **General description (session level)**
    - Protocol version, session owner, session name, …
  - **Date and periodicity (time level)**
    - start, stop time, …
  - **Media Description (media level)**
    - Media type, payload type, connection address, …

SDP: Session Description Protocol

- Example :
  - v[ersion]=0
  - o[wner]=Tin ‘session_id‘ IN IP4 194.240.47.217
  - s[ession]=Session VoIP
  - c[onnection]=IN IP4 194.240.47.217
  - b[bandwidth] =CT:128[kb/sec]
  - m[edia]=audio 49170 RTP/AVP 0 4
  - a=rtpmap:0 PCMU
  - a=rtpmap:4 G723
  - m=video 49190 RTP/AVP 31
  - a=rtpmap:31 H261
  - a=recvonly

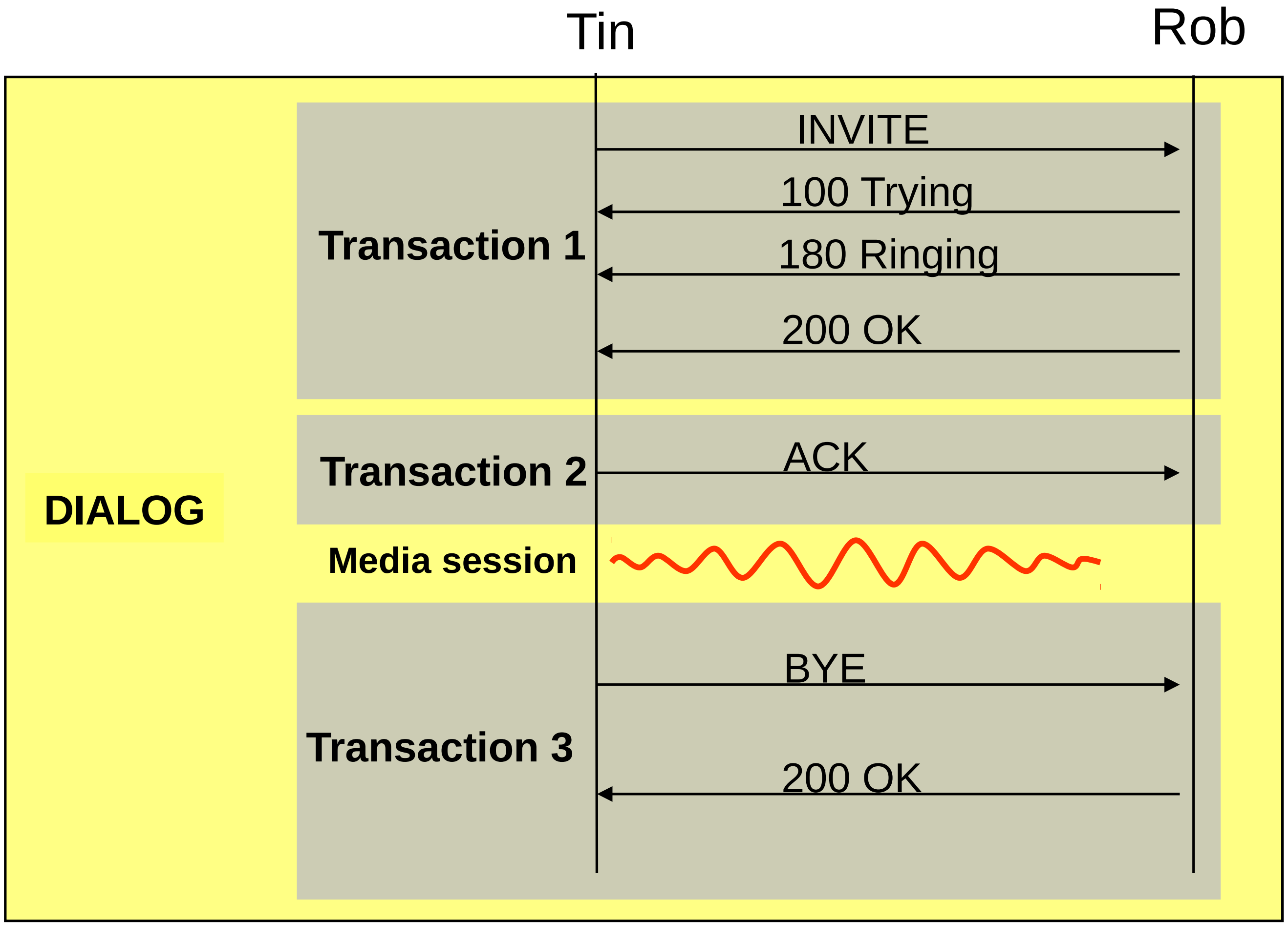
RTP/AVP

- **RFC 1890**
- RTP/AVP : RTP Profile for Audio and Video Conferences with minimal Control
- Principe :
  - Profile of audio and video
  - Definition of payload types
    - 0 PCMU (G711Ulaw)
    - 3 GSM
    - 8 PCMA (G711Alaw) => used in europe
    - 9 G722 => used in orange HD offer
    - 26 JPEG
    - 31 H261 (video)
    - 96--127 dynamic

Summary

- **What is VoIP ?**
- **Focus on SIP protocol**
  - History
  - SIP Basis
  - **Basic SIP dialog dissection**
  - Registrar function
  - Proxy and Redirect Servers
  - Advanced functions
  - SIP and security
  - Retransmission
  - SIP and NAT/FW
  - Presence and Instant Messaging
- **Focus on media component (RTP/RTCP)**

Basic Call Flow dissection



Basic Call Flow dissection – INVITE

Tin
10.0.0.2
UDP 5060

Rob
10.0.0.3
UDP 5060

INVITE sip:rob@10.0.0.3 SIP/2.0
Via: SIP/2.0/UDP **10.0.0.2:5060**;branch=**z9hG4bK74bf9**
Max-Forwards: 70
From: Tin <sip:tin@10.0.0.2>;tag=**9fxced76sl**
To: Rob <sip:rob@10.0.0.3>
Call-ID: 64367476413647@orange.com
Cseq: 1 INVITE
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, REFER, SUBSCRIBE, NOTIFY, MESSAGE, INFO, UPDATE, PRACK
Contact: <sip:tin@10.0.0.2:5060;transport=udp>
Content-Type: application/sdp
Content-Length: 150

v=0
o=tin 26533456 26533456 IN IP4 10.0.0.2
s=-
c=IN IP4 10.0.0.2
t=0 0
m=**audio 49172** RTP/AVP **8**
a=rtpmap:8 PCMA/8000

Tin asks Rob to establish an audio session:
RTP port: 49172
RTCP port : 49173
Codec: G711A

Basic Call Flow dissection – 180 Ringing

Tin
10.0.0.2
UDP 5060

Rob
10.0.0.3
UDP 5060

SIP/2.0 **180 Ringing**
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK74bf9
From: Tin <sip:tin@10.0.0.2>;tag=9fxced76sl
To: Rob <sip:rob@10.0.0.3>;**tag=7553a2yr5s**
Call-ID: 64367476413647@orange.com
Cseq: 1 INVITE
Contact: <sip:bob@10.0.0.3:5060;transport=udp>
Content-Length: 0

Basic Call Flow dissection – 200 OK

Tin
10.0.0.2

Rob
10.0.0.3

UDP 5060

UDP 5060

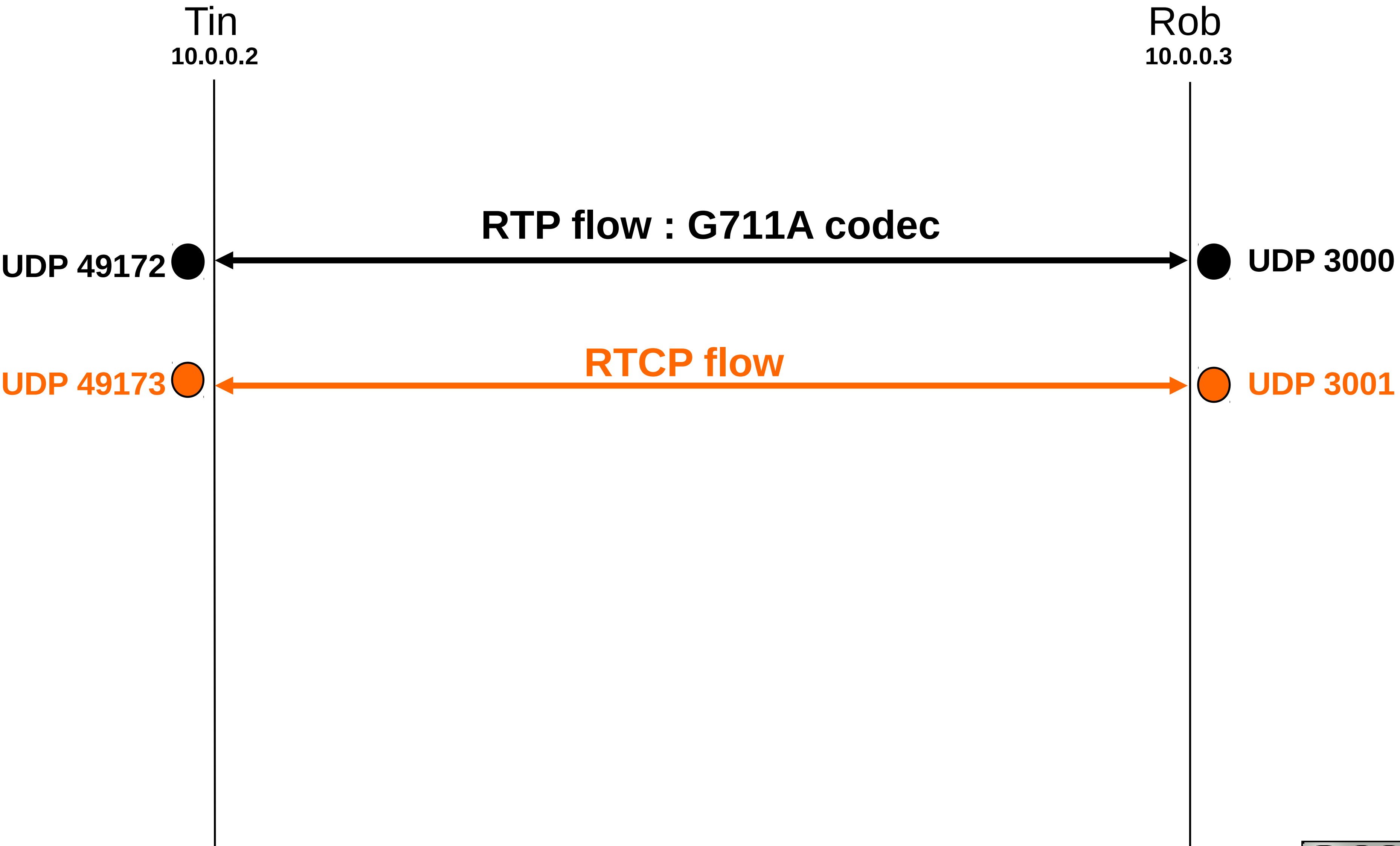
SIP/2.0 **200 OK**
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK74bf9
From: Tin <sip:tin@10.0.0.2>;tag=9fxced76sl
To: Rob <sip:rob@10.0.0.3>;tag=7553a2yr5s
Call-ID: 64367476413647@orange.com
Cseq: 1 INVITE
Contact: <sip:bob@10.0.0.3:5060;transport=udp>
Content-Type: application/sdp
Content-Length: **137**

v=0
o=Rob 1692182 1 IN IP4 **10.0.0.3**
s=-
c=IN IP4 **10.0.0.3**
t=0 0
m=audio **3000** RTP/AVP **8**
a=rtpmap:8 PCMA/8000

Rob accepts the establishment of the audio session:
RTP port : 3000
RTCP port : 3001
Codec: G711A

Basic Call Flow dissection – ACK

Basic Call Flow dissection – Media Session



Basic Call Flow dissection – BYE

Tin

10.0.0.2

Rob

10.0.0.3

UDP 5060 ← **UDP 5060**

BYE sip:tin@10.0.0.2 SIP/2.0
Via: SIP/2.0/UDP **10.0.0.3**:5060;branch=**z9hG4bKaj7G5e567u982**
Max-Forwards: 70
From: Rob <sip:rob@10.0.0.3>;tag=7553a2yr5s
To: Tin <sip:tin@10.0.0.2>;tag=9fxced76sl
Call-ID: 64367476413647@orange.com
Cseq: **2 BYE**
Content-Length: 0

Basic Call Flow dissection – 200 OK

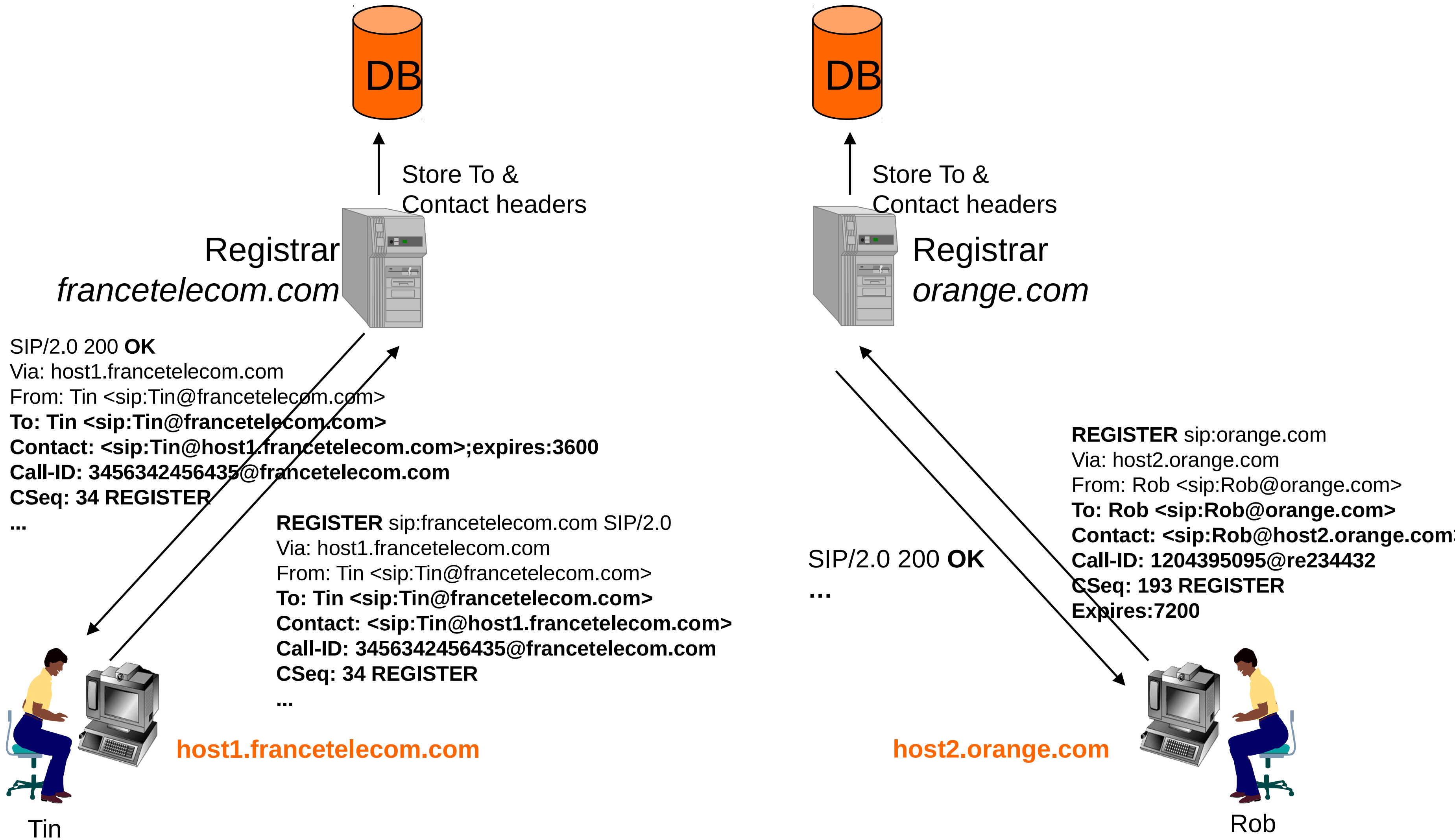
Summary

- **What is VoIP ?**
- **Focus on SIP protocol**
  - History
  - SIP Basis
  - Basic SIP dialog dissection
  - **Registrar function**
  - Proxy and Redirect Servers
  - Advanced functions
  - SIP and security
  - Retransmission
  - SIP and NAT/FW
  - Presence and Instant Messaging
- **Focus on media component (RTP/RTCP)**

Registration

- The Registrar is designed to keep track of the current location of the user (dynamic @IP, mobility...)
- The Registrar stores in the location service database the information of each UA received in REGISTER request:
  - Address of Record (AoR) (**To** header, eg. sip:bob@orange.com)
  - Physical Address(es) of user terminal(s) (**Contact** header, eg. sip:bob@10.0.0.3:5060)
- It also store two other information in memory to keep trace of this REGISTER request:
  - Dialog identification (**Call-ID** header, eg. 64367476413647@orange.com)
  - Transaction identification (**CSeq** header, eg. 1826 REGISTER)
- The registered state is not permanent. If not refreshed by subsequent REGISTER sent by the UA, it will 'time out' after 1h (3600s) by default (can be more or less)

Registration



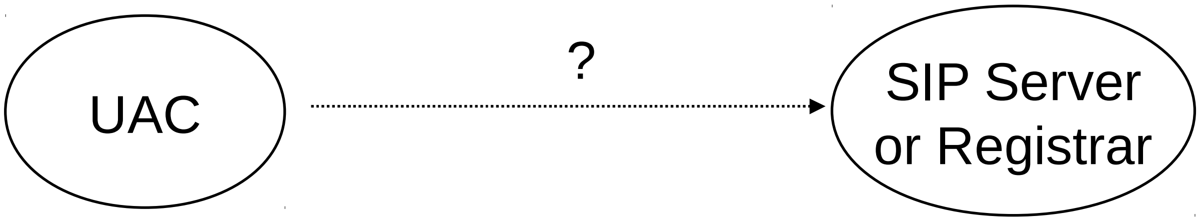
Summary

- **What is VoIP ?**

- **Focus on SIP protocol**
  - History
  - SIP Basis
  - Basic SIP dialog dissection
  - Registrar function
  - **Proxy and Redirect Servers**
  - Advanced functions
  - SIP and security
  - Retransmission
  - SIP and NAT/FW
  - Presence and Instant Messaging

- **Focus on media component (RTP/RTCP)**

Locating SIP Server from UA Client



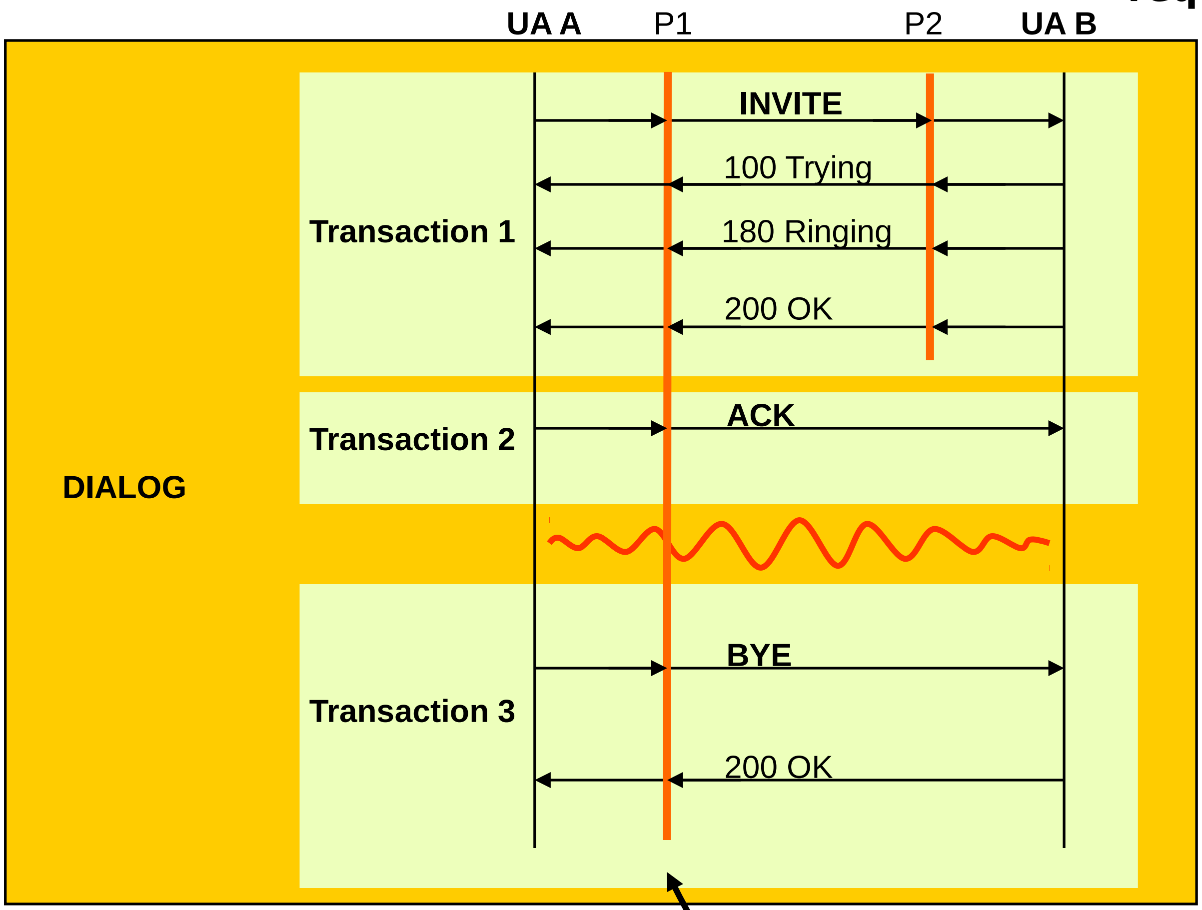
- Client configuration : Outbound proxy
  - Local configuration
  - DHCP option 120
  - DNS SRV lookup

    SIP SRV francetelecom.com => 172.20.35.21

- Outbound Proxy is the equipment that receive all the SIP trafic of an UA

Messages traversal in SIP network

Proxies P1 & P2 in the path of the first transaction : request + responses *[Via]*

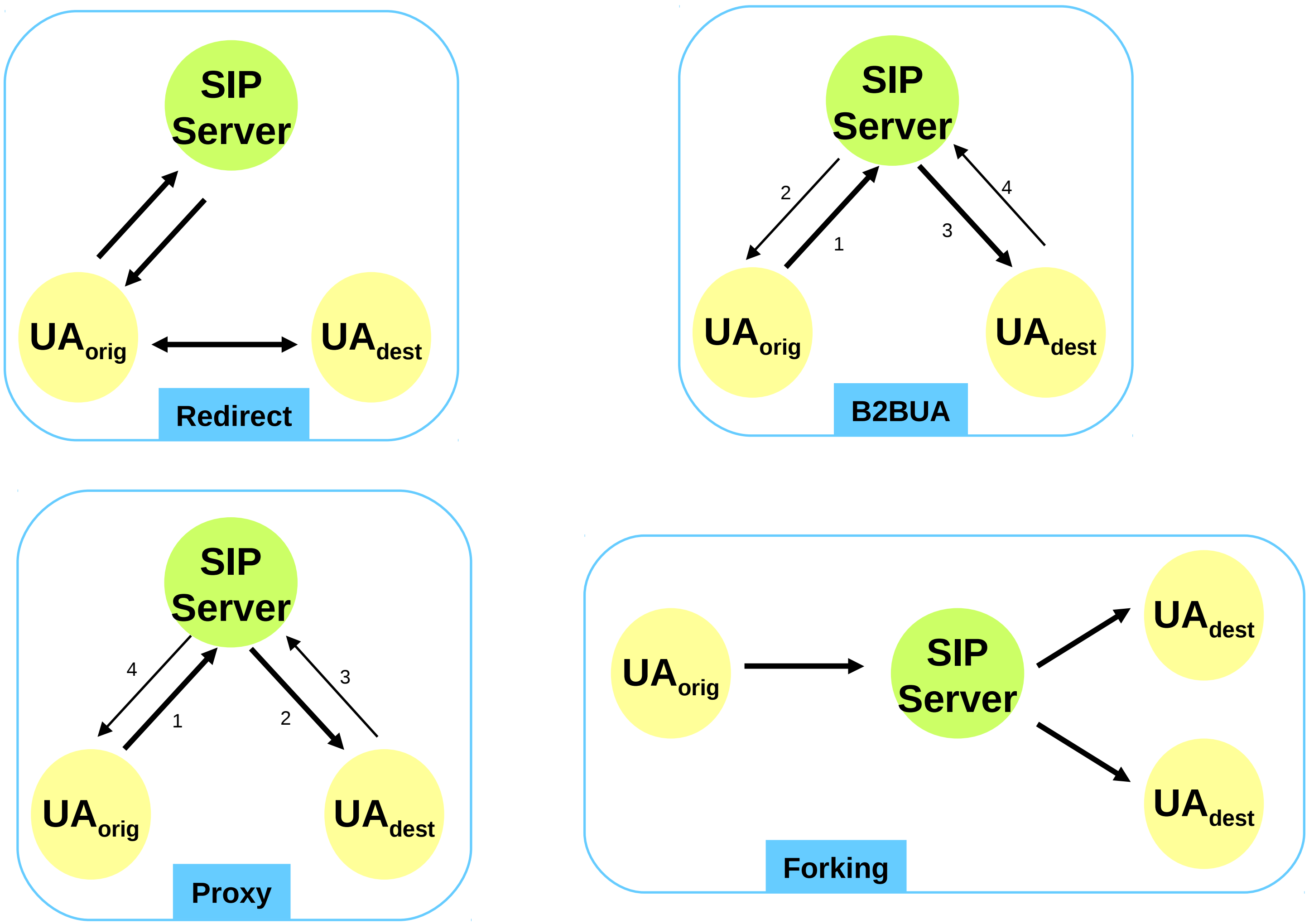


Behavior of SIP node can be :

- ***Stateless***
- ***Transaction Stateful***
- ***Call (or dialog) stateful***

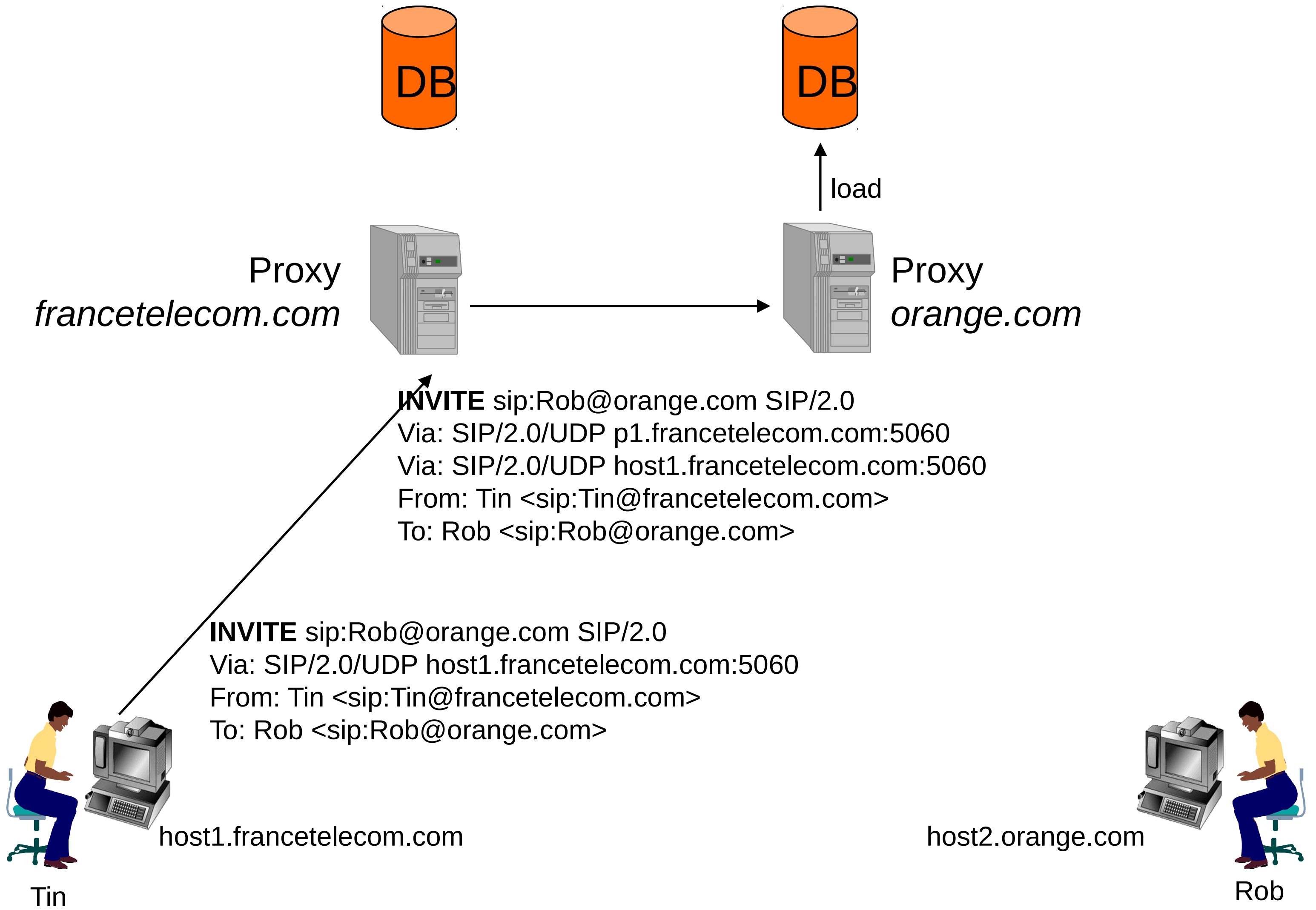
Proxy P1 in the path of all transactions *[Record-Route]*

Call model

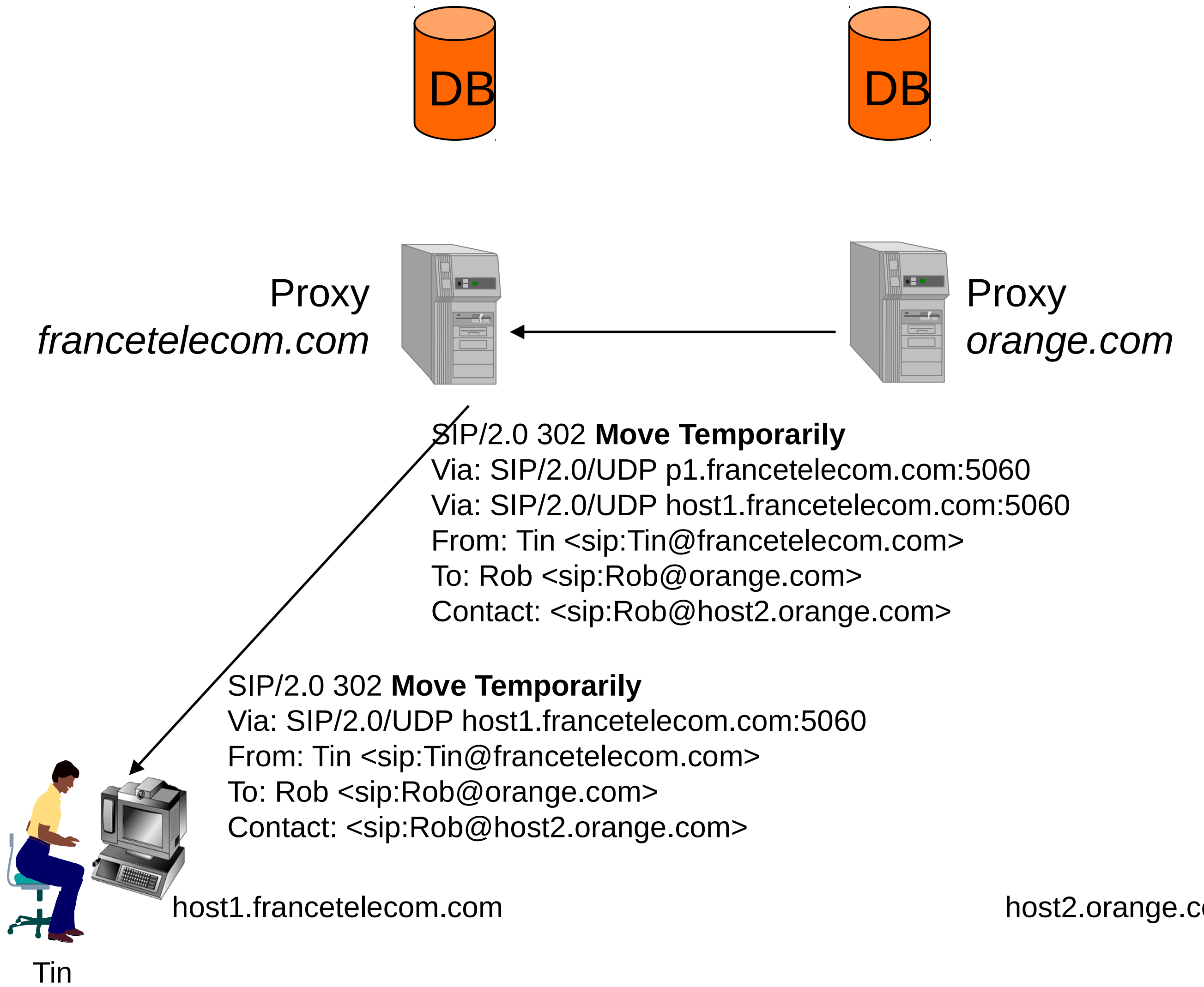


SIP Server
UA_{orig}
UA_{dest}
Redirect
SIP Server
2
1
3
4
UA_{orig}
UA_{dest}
B2BUA
SIP Server
4
1
2
3
UA_{orig}
UA_{dest}
Proxy
UA_{orig}
SIP Server
UA_{dest}
UA_{dest}
Forking

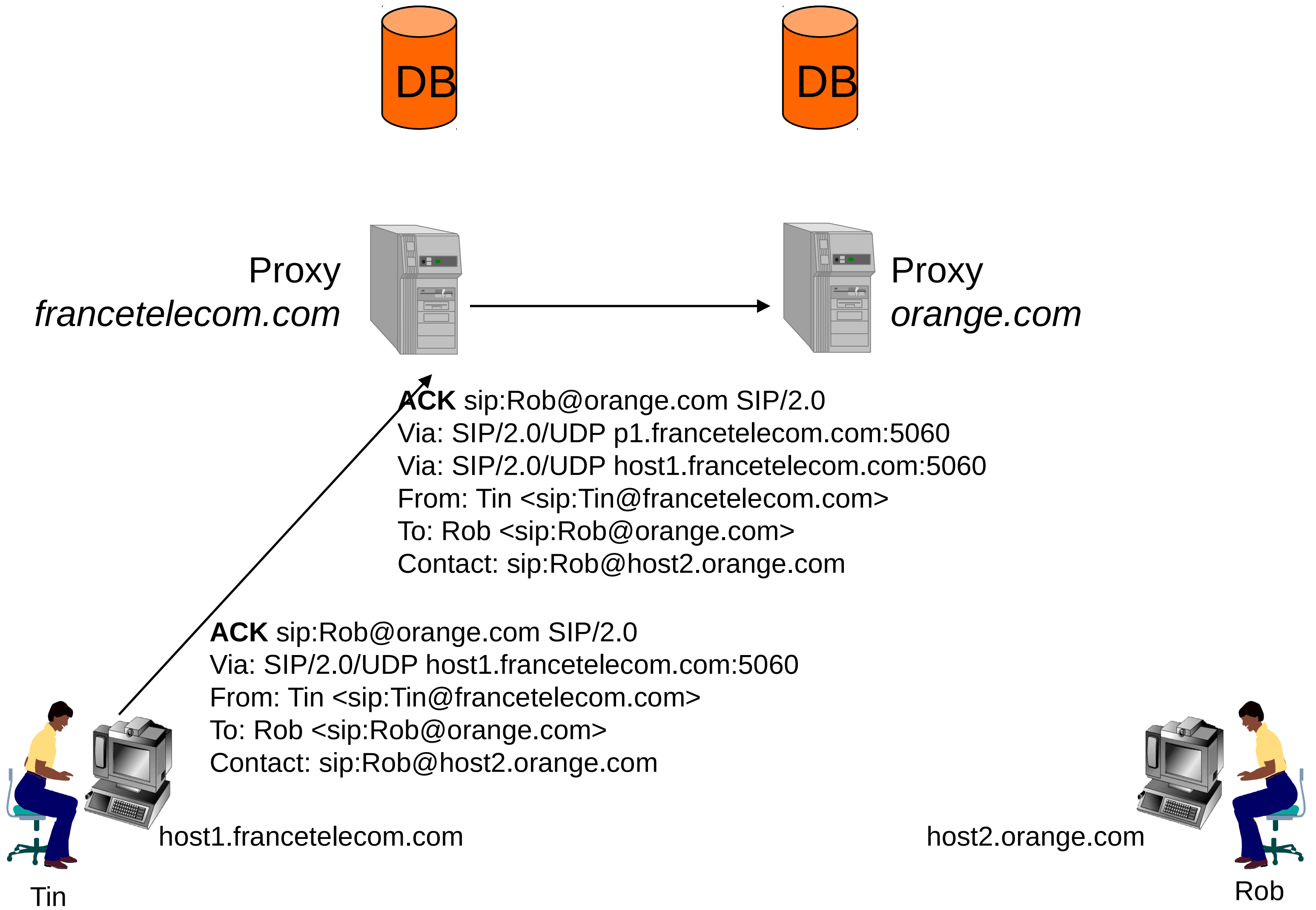
Redirect mode



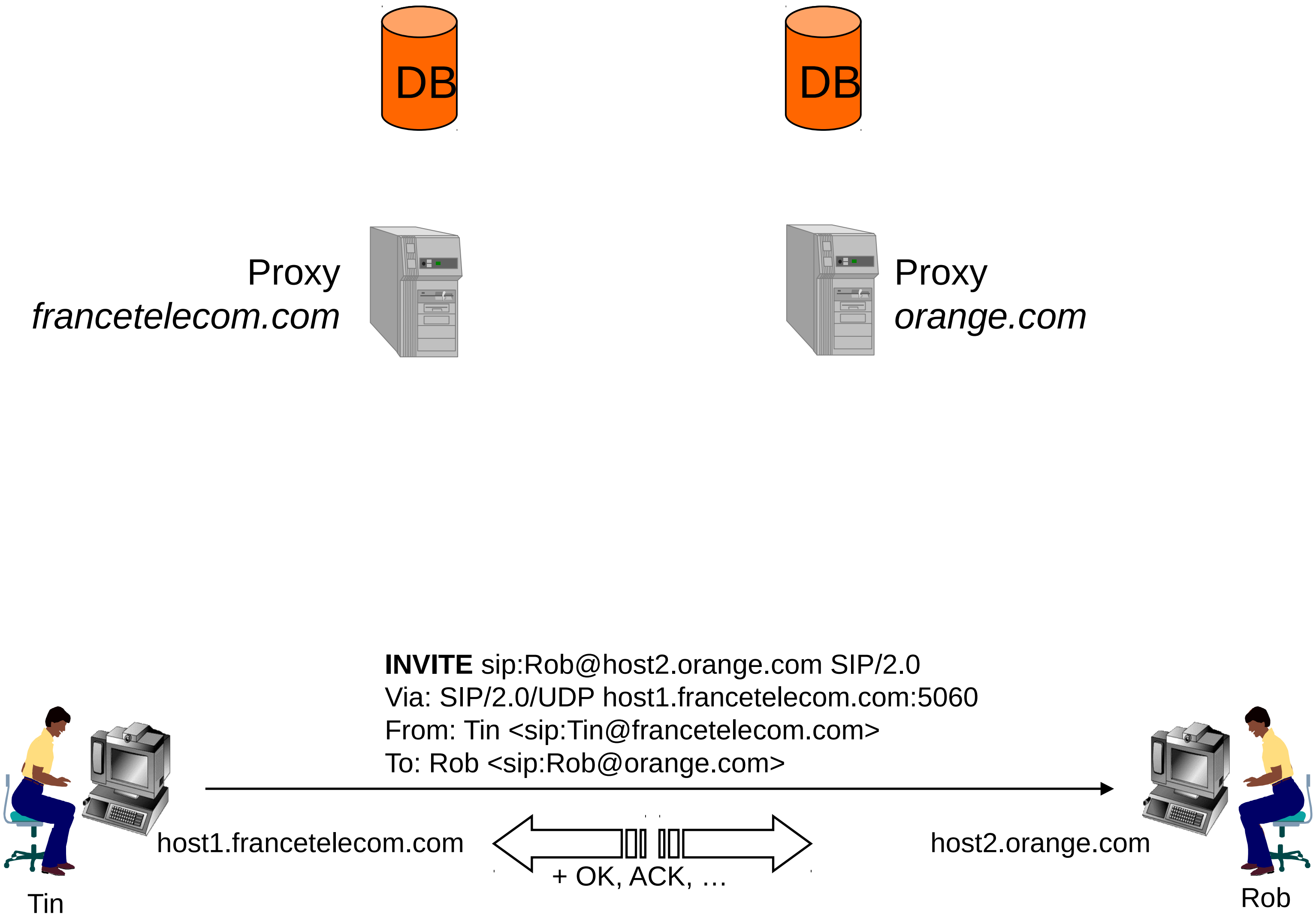
Redirect mode



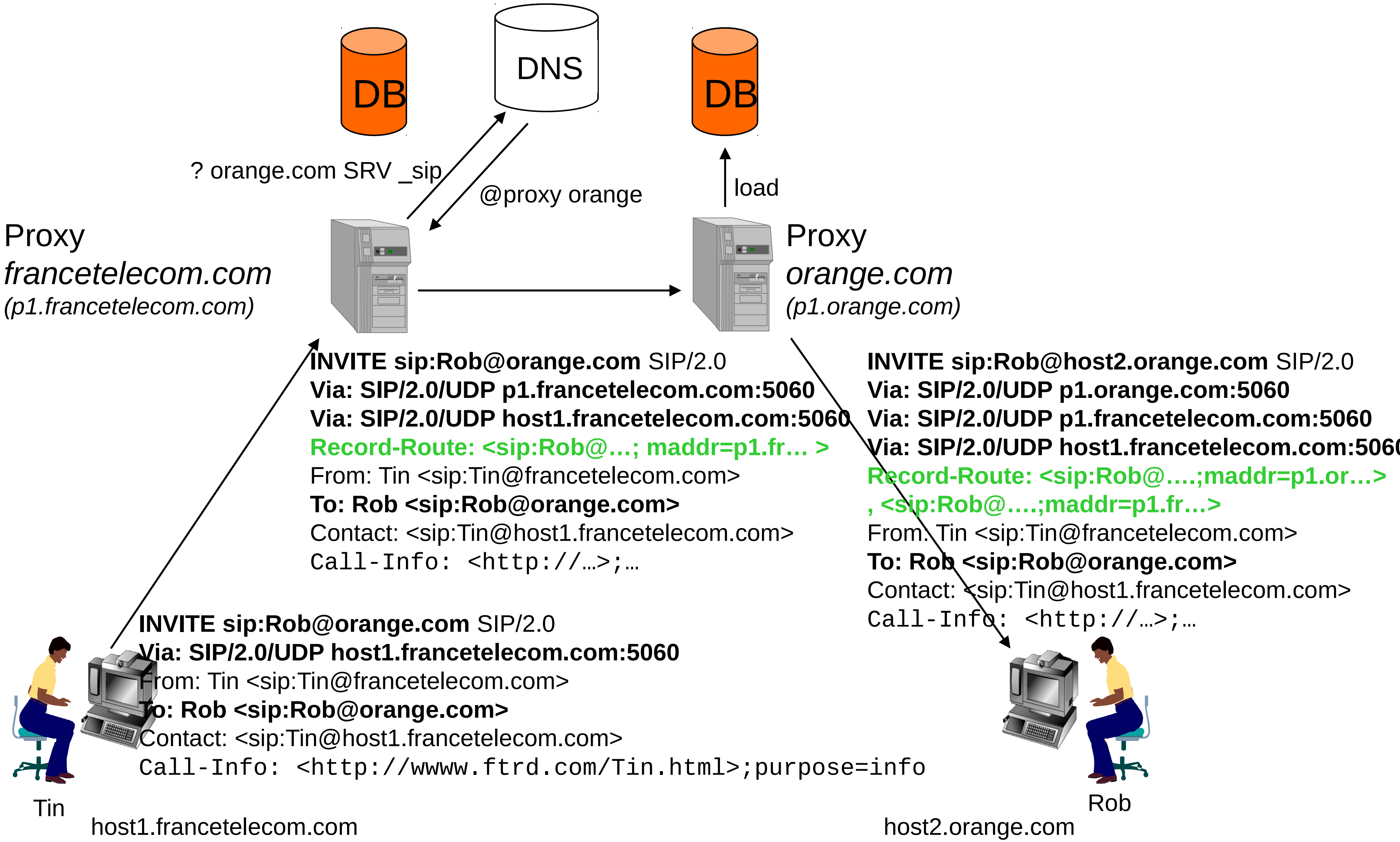
Redirect mode



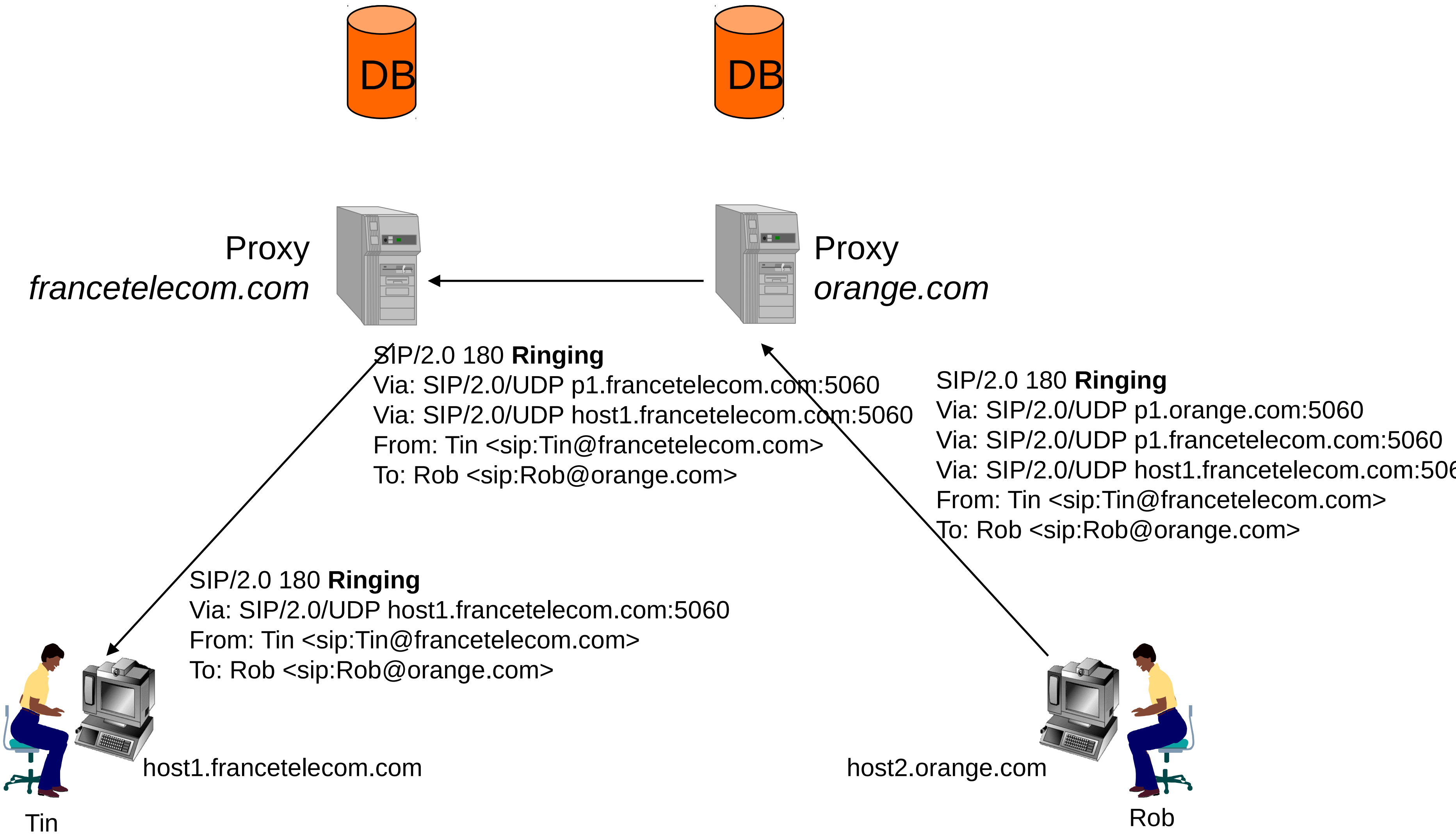
Redirect mode



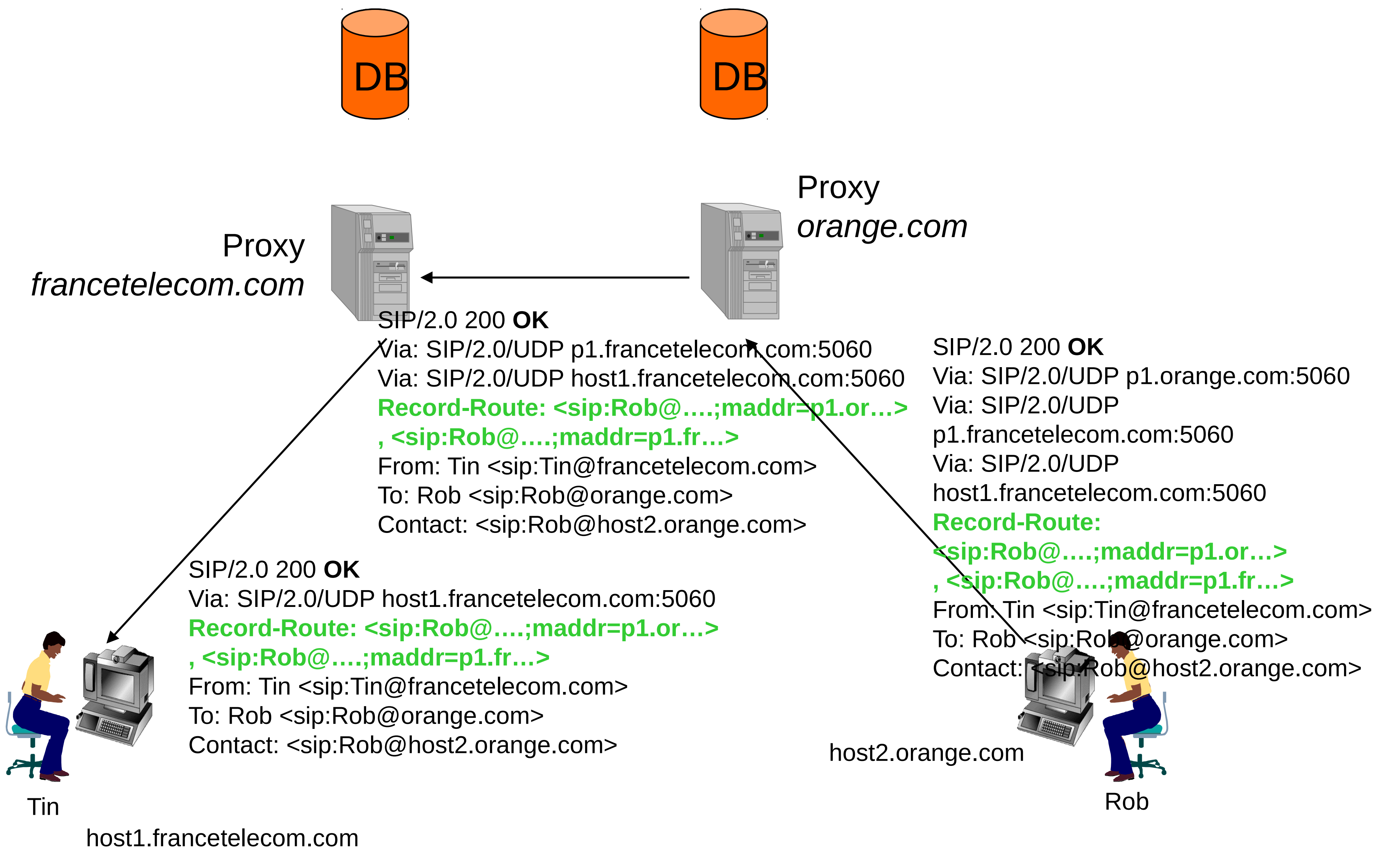
Proxy mode


DB
DNS
DB
? orange.com SRV _sip
@proxy orange
load
Proxy
francetelecom.com
(p1.francetelecom.com)
Proxy
orange.com
(p1.orange.com)
INVITE sip:Rob@orange.com SIP/2.0
Via: SIP/2.0/UDP p1.francetelecom.com:5060
Via: SIP/2.0/UDP host1.francetelecom.com:5060
Record-Route: <sip:Rob@…; maddr=p1.fr… >
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>
Contact: <sip:Tin@host1.francetelecom.com>
Call-Info: <http://…>;…
INVITE sip:Rob@host2.orange.com SIP/2.0
Via: SIP/2.0/UDP p1.orange.com:5060
Via: SIP/2.0/UDP p1.francetelecom.com:5060
Via: SIP/2.0/UDP host1.francetelecom.com:506
Record-Route: <sip:Rob@….;maddr=p1.or…>
, <sip:Rob@….;maddr=p1.fr…>
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>
Contact: <sip:Tin@host1.francetelecom.com>
Call-Info: <http://…>;…
INVITE sip:Rob@orange.com SIP/2.0
Via: SIP/2.0/UDP host1.francetelecom.com:5060
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>
Contact: <sip:Tin@host1.francetelecom.com>
Call-Info: <http://wwww.ftrd.com/Tin.html>;purpose=info
Tin
host1.francetelecom.com
Rob
host2.orange.com

Proxy mode


DB
DB
Proxy
francetelecom.com
Proxy
orange.com
SIP/2.0 180 **Ringing**
Via: SIP/2.0/UDP p1.francetelecom.com:5060
Via: SIP/2.0/UDP host1.francetelecom.com:5060
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>
SIP/2.0 180 **Ringing**
Via: SIP/2.0/UDP p1.orange.com:5060
Via: SIP/2.0/UDP p1.francetelecom.com:5060
Via: SIP/2.0/UDP host1.francetelecom.com:506
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>
SIP/2.0 180 **Ringing**
Via: SIP/2.0/UDP host1.francetelecom.com:5060
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>
host1.francetelecom.com
host2.orange.com
Tin
Rob

Proxy mode


DB
DB
Proxy
francetelecom.com
Proxy
orange.com
SIP/2.0 200 **OK**
Via: SIP/2.0/UDP p1.francetelecom.com:5060
Via: SIP/2.0/UDP host1.francetelecom.com:5060
Record-Route: <sip:Rob@….;maddr=p1.or…>
, <sip:Rob@….;maddr=p1.fr…>
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>
Contact: <sip:Rob@host2.orange.com>
SIP/2.0 200 **OK**
Via: SIP/2.0/UDP p1.orange.com:5060
Via: SIP/2.0/UDP
p1.francetelecom.com:5060
Via: SIP/2.0/UDP
host1.francetelecom.com:5060
Record-Route:
<sip:Rob@….;maddr=p1.or…>
, <sip:Rob@….;maddr=p1.fr…>
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>
Contact: <sip:Rob@host2.orange.com>
SIP/2.0 200 **OK**
Via: SIP/2.0/UDP host1.francetelecom.com:5060
Record-Route: <sip:Rob@….;maddr=p1.or…>
, <sip:Rob@….;maddr=p1.fr…>
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>
Contact: <sip:Rob@host2.orange.com>
host2.orange.com
Tin
Rob
host1.francetelecom.com

Proxy mode (offer/answer model)

INVITE sip:Rob@orange.com SIP/2.0

......

Subject: phone call
Content-Type: application/SDP
Content-Length: 148

OFFER

v=0
o=Tin 562413 562413 IN IP4 194.240.47.217
s=phone call

c=IN IP4 194.240.47.217
m=audio 4710 RTP/AVP 0 4
a=rtpmap:0 PCMU/8000
a=rtpmap:4 G723/8000
m=video 8524 RTP/AVP 31
a=rtpmap:31 H261/90000

Stream description

SIP/2.0 200 OK

......

Subject: phone call
Content-Type: application/SDP
Content-Length: 148

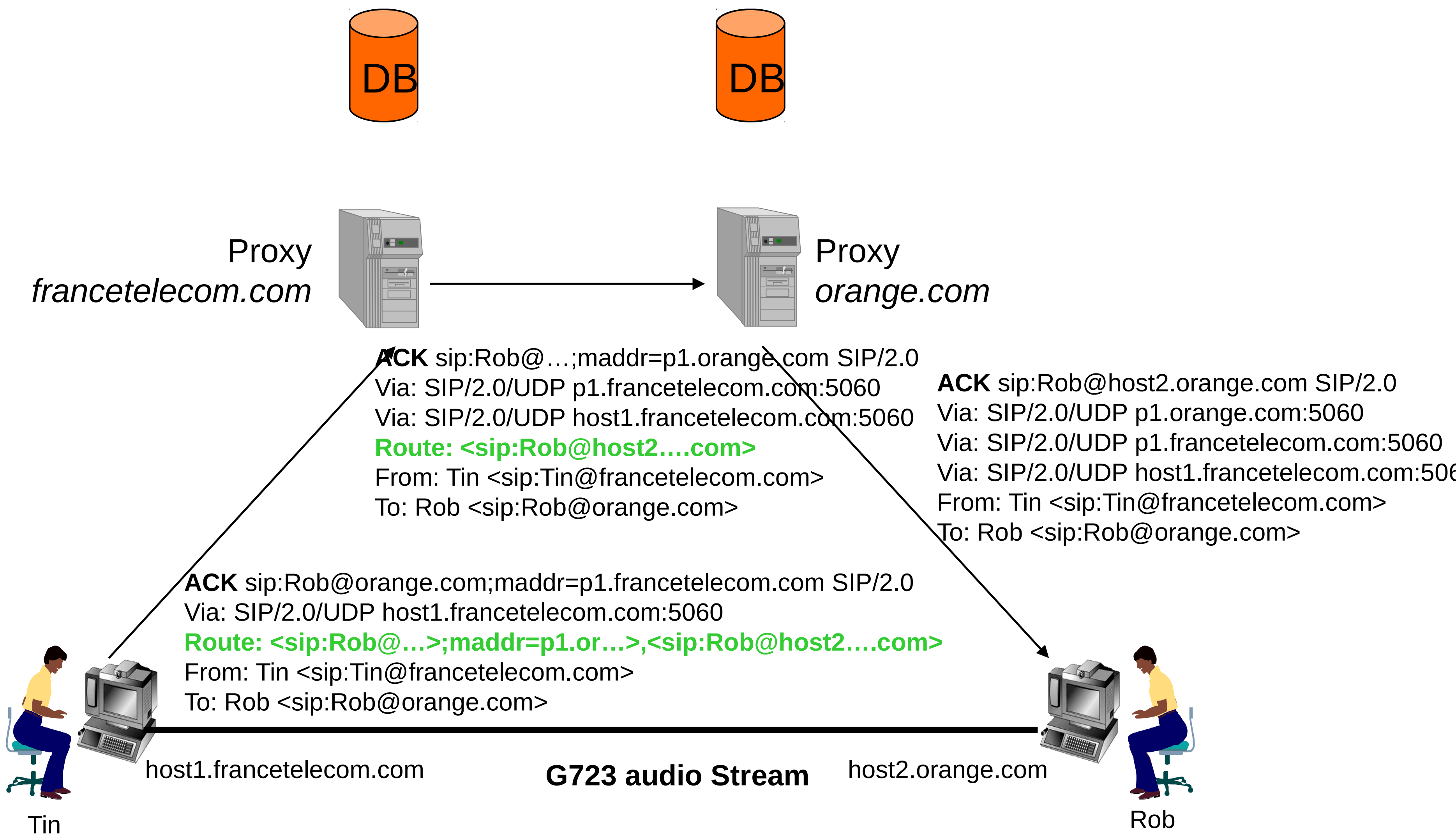
ANSWER

v=0
o=Rob 562414 562414 IN IP4 172.32.12.32
s=phone call

c=IN IP4 172.32.12.32
m=audio 3456 RTP/AVP 4
a=rtpmap:4 G723/8000
m=video **0** RTP/AVP 31

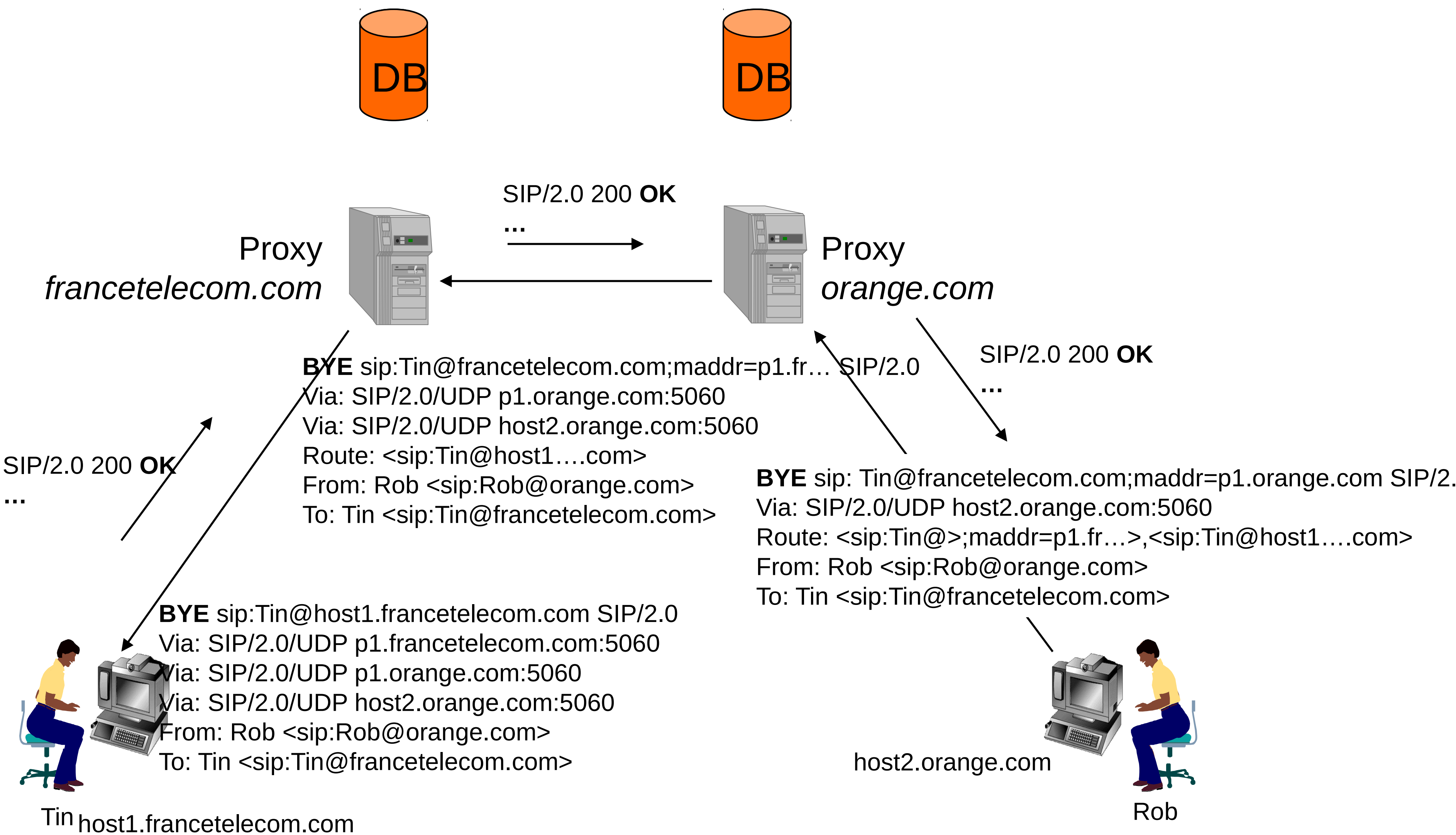
If session description not acceptable (ex. incompatible media, Insufficient bandwidth) : 606 (Not Acceptable) response code with reasons in Warning header

Proxy mode



DB
DB
Proxy
francetelecom.com
Proxy
orange.com
ACK sip:Rob@…;maddr=p1.orange.com SIP/2.0
Via: SIP/2.0/UDP p1.francetelecom.com:5060
Via: SIP/2.0/UDP host1.francetelecom.com:5060
Route: <sip:Rob@host2….com>
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>
ACK sip:Rob@host2.orange.com SIP/2.0
Via: SIP/2.0/UDP p1.orange.com:5060
Via: SIP/2.0/UDP p1.francetelecom.com:5060
Via: SIP/2.0/UDP host1.francetelecom.com:506
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>
ACK sip:Rob@orange.com;maddr=p1.francetelecom.com SIP/2.0
Via: SIP/2.0/UDP host1.francetelecom.com:5060
Route: <sip:Rob@…>;maddr=p1.or…>,<sip:Rob@host2….com>
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>
host1.francetelecom.com
G723 audio Stream
host2.orange.com
Tin
Rob

Proxy mode


DB
DB
SIP/2.0 200 **OK**
...
Proxy
francetelecom.com
Proxy
orange.com
BYE sip:Tin@francetelecom.com;maddr=p1.fr… SIP/2.0
Via: SIP/2.0/UDP p1.orange.com:5060
Via: SIP/2.0/UDP host2.orange.com:5060
Route: <sip:Tin@host1….com>
From: Rob <sip:Rob@orange.com>
To: Tin <sip:Tin@francetelecom.com>
SIP/2.0 200 **OK**
...
SIP/2.0 200 **OK**
...
BYE sip: Tin@francetelecom.com;maddr=p1.orange.com SIP/2.
Via: SIP/2.0/UDP host2.orange.com:5060
Route: <sip:Tin@>;maddr=p1.fr…>,<sip:Tin@host1….com>
From: Rob <sip:Rob@orange.com>
To: Tin <sip:Tin@francetelecom.com>
BYE sip:Tin@host1.francetelecom.com SIP/2.0
Via: SIP/2.0/UDP p1.francetelecom.com:5060
Via: SIP/2.0/UDP p1.orange.com:5060
Via: SIP/2.0/UDP host2.orange.com:5060
From: Rob <sip:Rob@orange.com>
To: Tin <sip:Tin@francetelecom.com>
host2.orange.com
Rob
Tin
host1.francetelecom.com

Proxy mode – info to remember

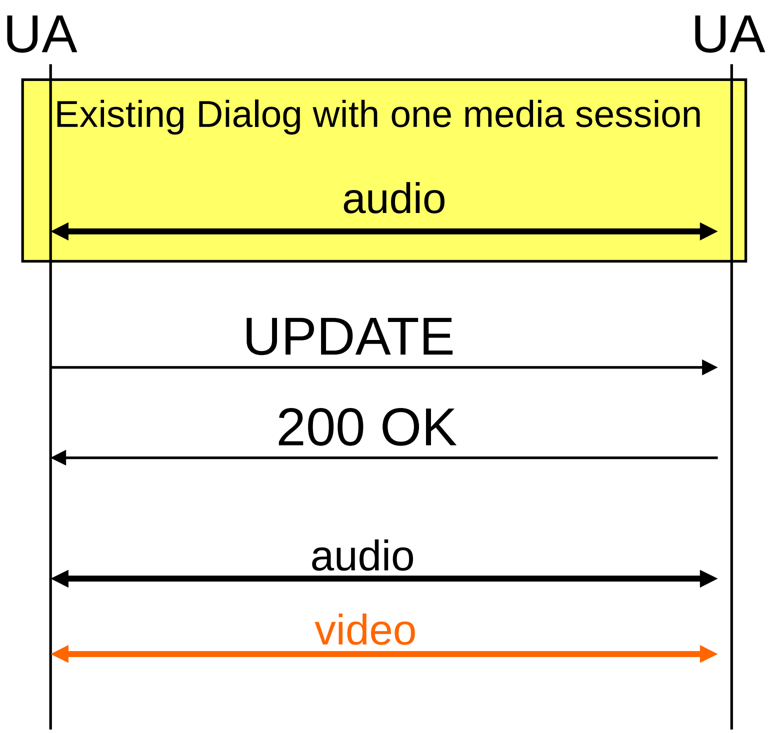
- Via are used in transactions to record the SIP route taken by a Request and are used to route Response back to the originator
  - Added with Requests
  - Removed with Responses
- Route is used in dialogs
  - Record-route built with INVITE Request
  - Used in any subsequent Request

Summary

- **What is VoIP ?**
- **Focus on SIP protocol**
  - History
  - SIP Basis
  - Basic SIP dialog dissection
  - Registrar function
  - Proxy and Redirect Servers
  - **Advanced functions**
  - SIP and security
  - Retransmission
  - SIP and NAT/FW
  - Presence and Instant Messaging
- **Focus on media component (RTP/RTCP)**

Session update

- Re-INVITE or UPDATE: request within the same dialog
  - same From,To (+tags) and Call-ID headers as initial Invite
  - Update or modify session description



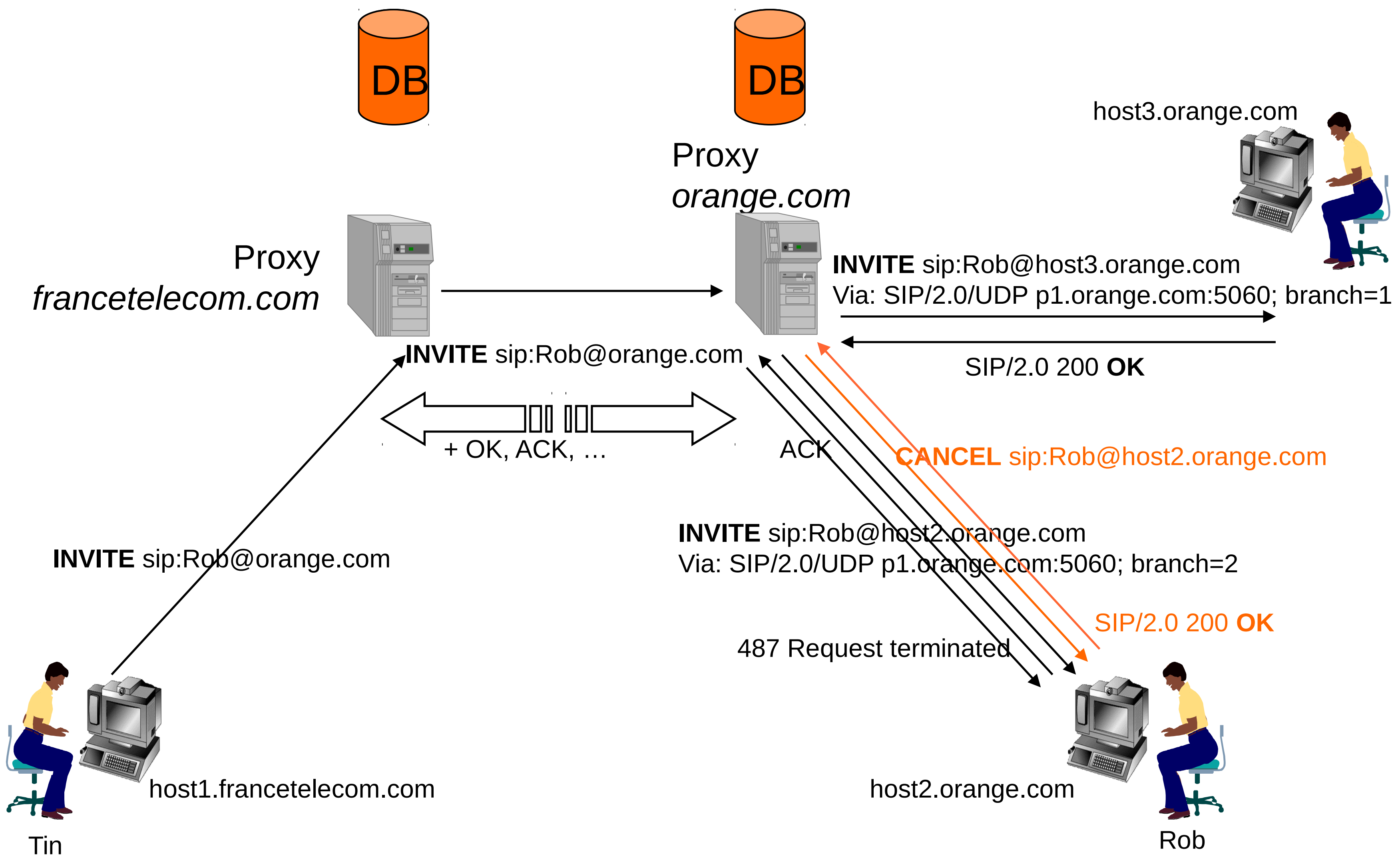
Example : Add video media in the session

UPDATE sip:Rob@orange.com SIP/2.0
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>
Cseq: 3 UPDATE
Call-ID: 124325617@host1.francetelecom.com
…

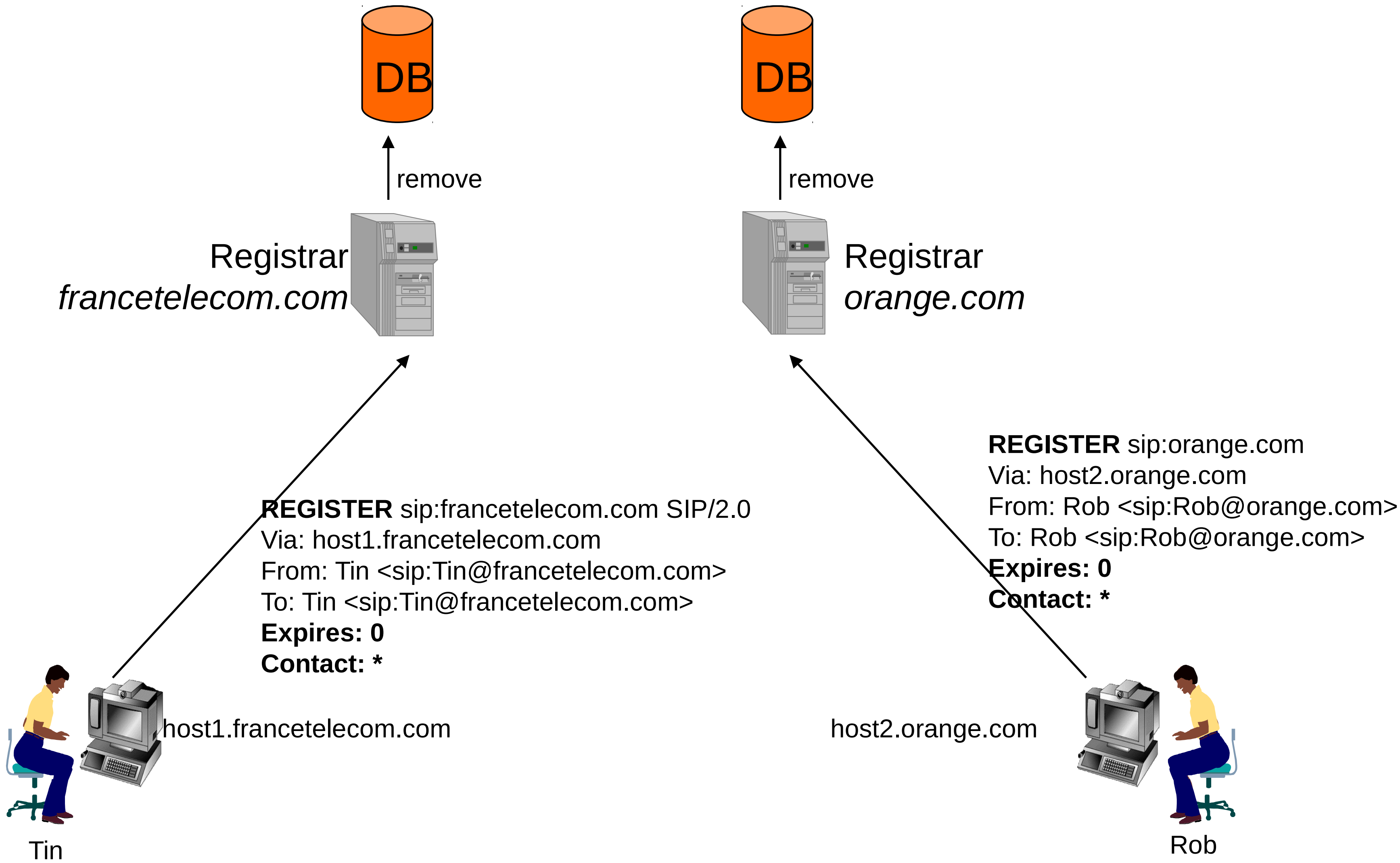
v=0
o=Tin 562413 562414 IN IP4 194.240.47.217
s=phone call
c=IN IP4 194.240.47.217

m=audio 4710 RTP/AVP 4
a=rtpmap:4 G723/8000
m=video 5643 RTP/AVP 31
a=rtpmap:31 H261/90000

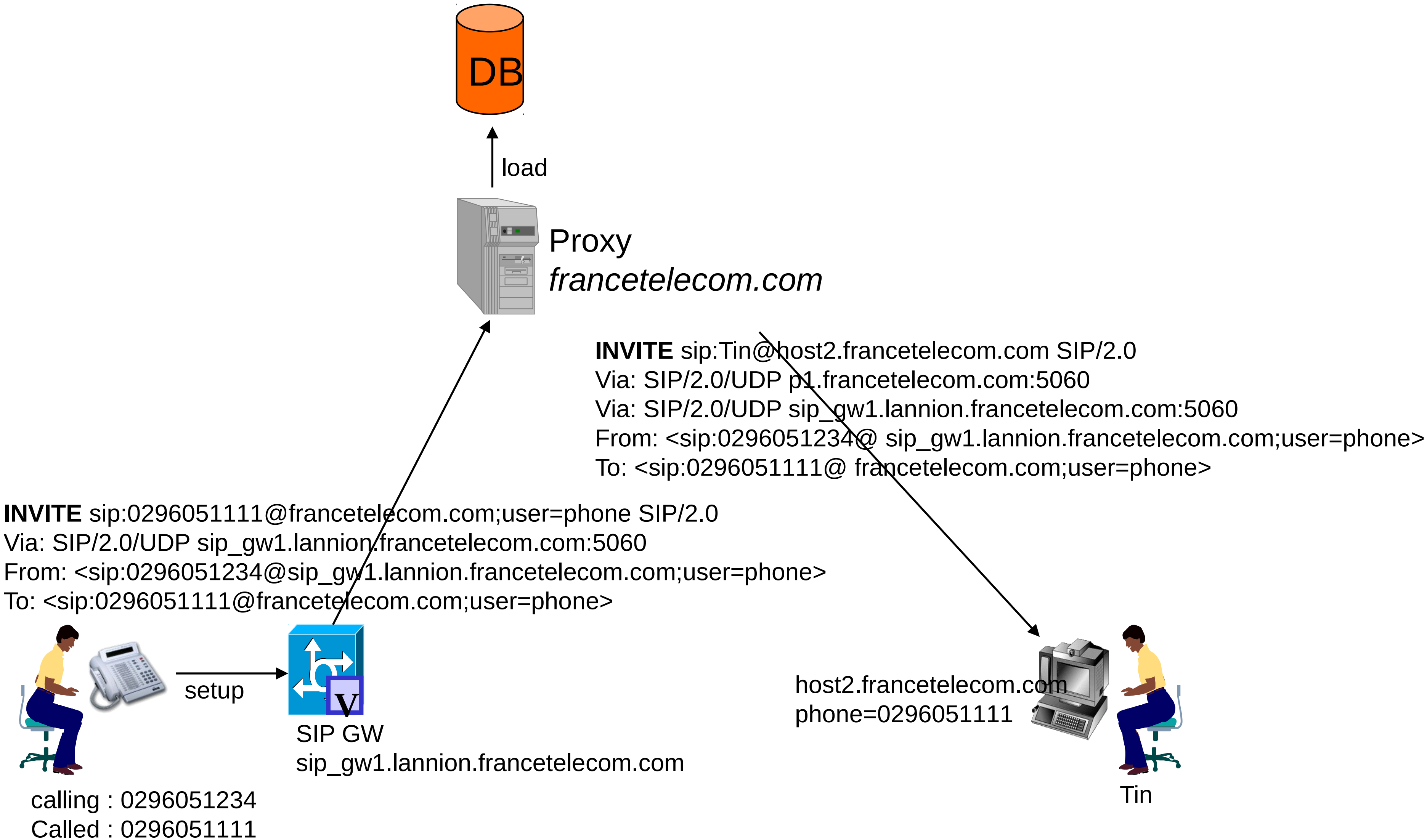
Forking mode (in parallel)



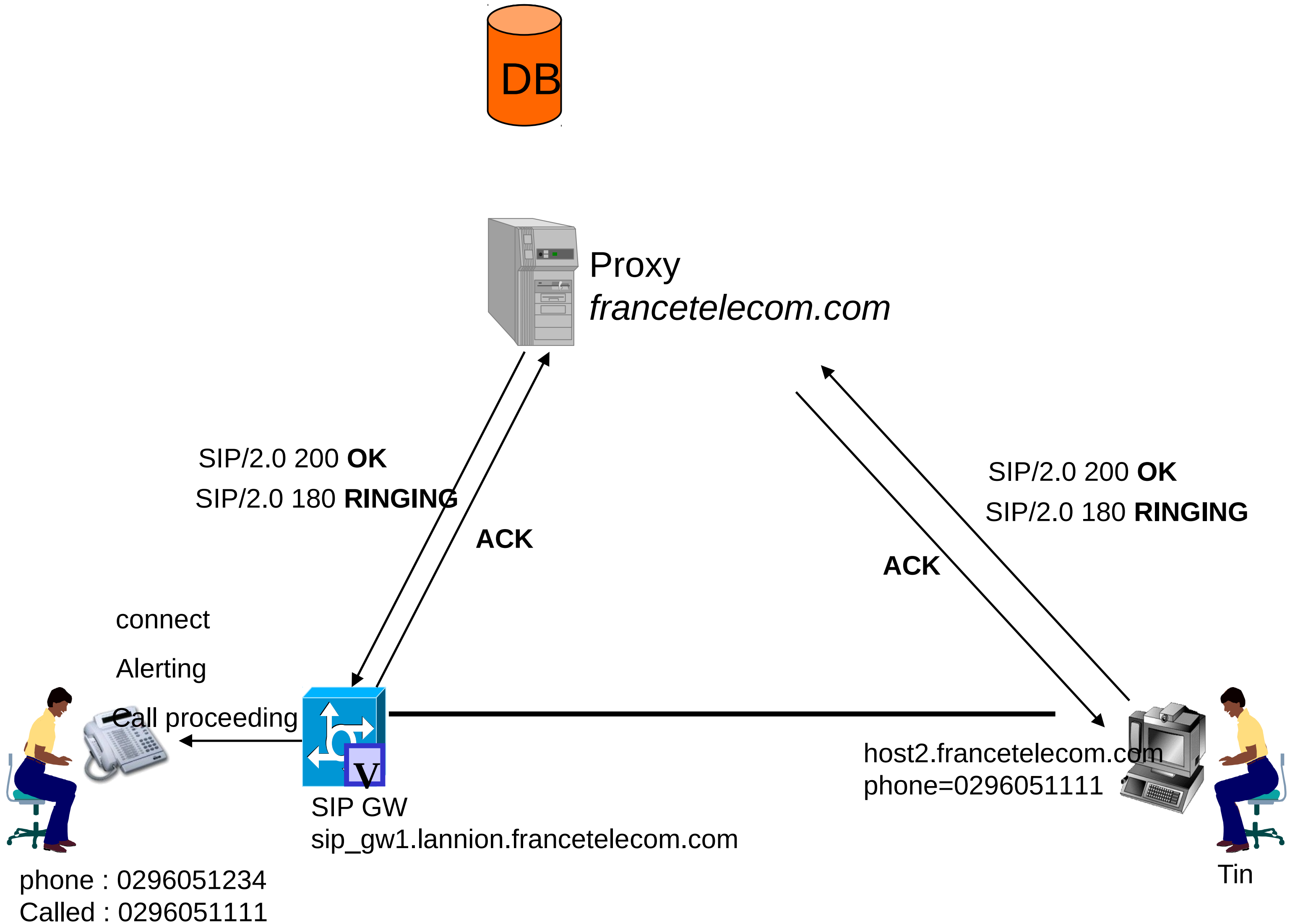
Remove registration



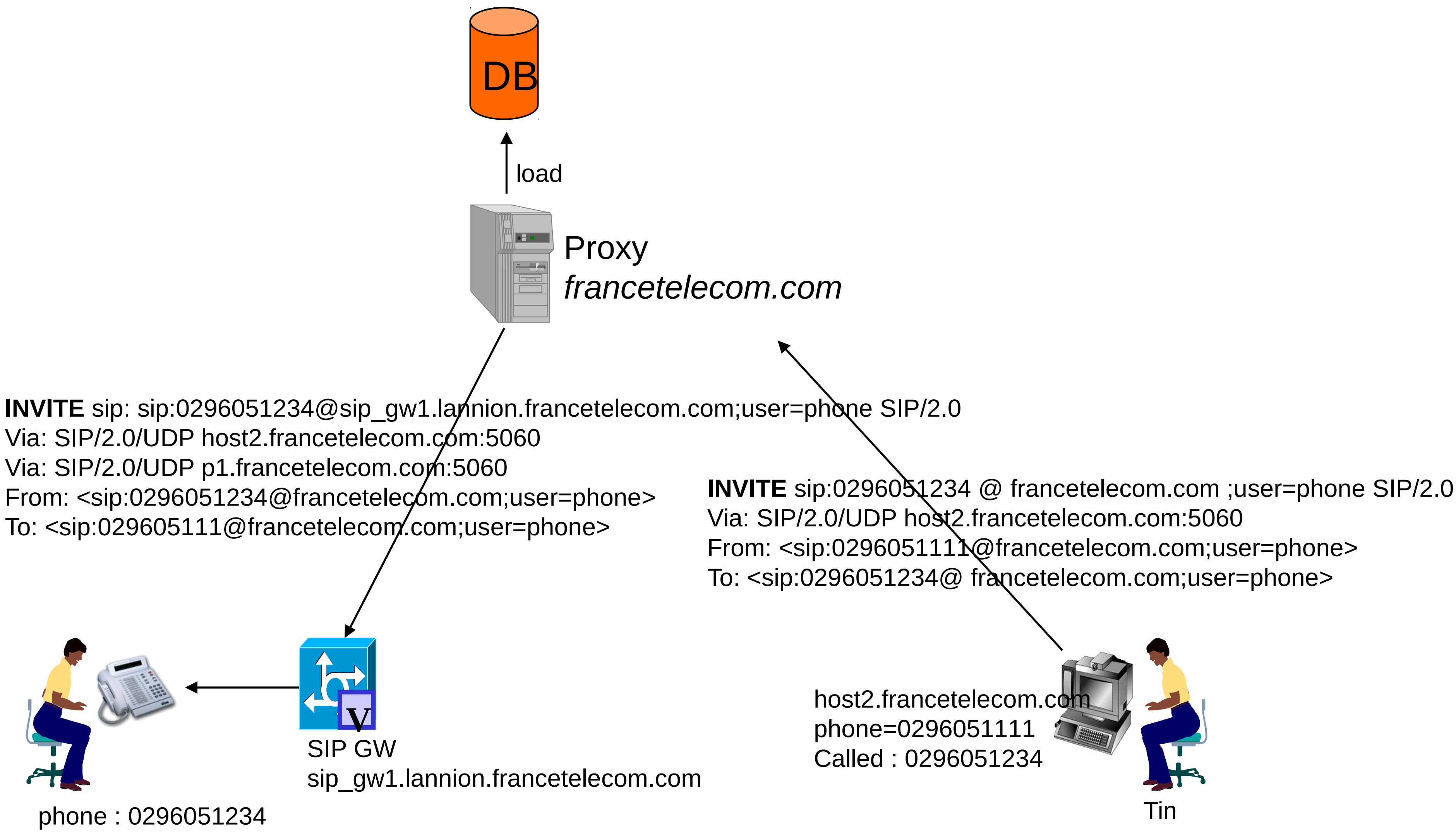
Phone-to-pc


DB
load
Proxy
francetelecom.com
INVITE sip:Tin@host2.francetelecom.com SIP/2.0
Via: SIP/2.0/UDP p1.francetelecom.com:5060
Via: SIP/2.0/UDP sip_gw1.lannion.francetelecom.com:5060
From: <sip:0296051234@ sip_gw1.lannion.francetelecom.com;user=phone>
To: <sip:0296051111@ francetelecom.com;user=phone>
INVITE sip:0296051111@francetelecom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP sip_gw1.lannion.francetelecom.com:5060
From: <sip:0296051234@sip_gw1.lannion.francetelecom.com;user=phone>
To: <sip:0296051111@francetelecom.com;user=phone>
setup
V
SIP GW
sip_gw1.lannion.francetelecom.com
host2.francetelecom.com
phone=0296051111
Tin
calling : 0296051234
Called : 0296051111

Phone-to-pc


DB
Proxy
francetelecom.com
SIP/2.0 200 **OK**
SIP/2.0 180 **RINGING**
ACK
SIP/2.0 200 **OK**
SIP/2.0 180 **RINGING**
ACK
connect
Alerting
Call proceeding
SIP GW
sip_gw1.lannion.francetelecom.com
host2.francetelecom.com
phone=0296051111
phone : 0296051234
Called : 0296051111
Tin

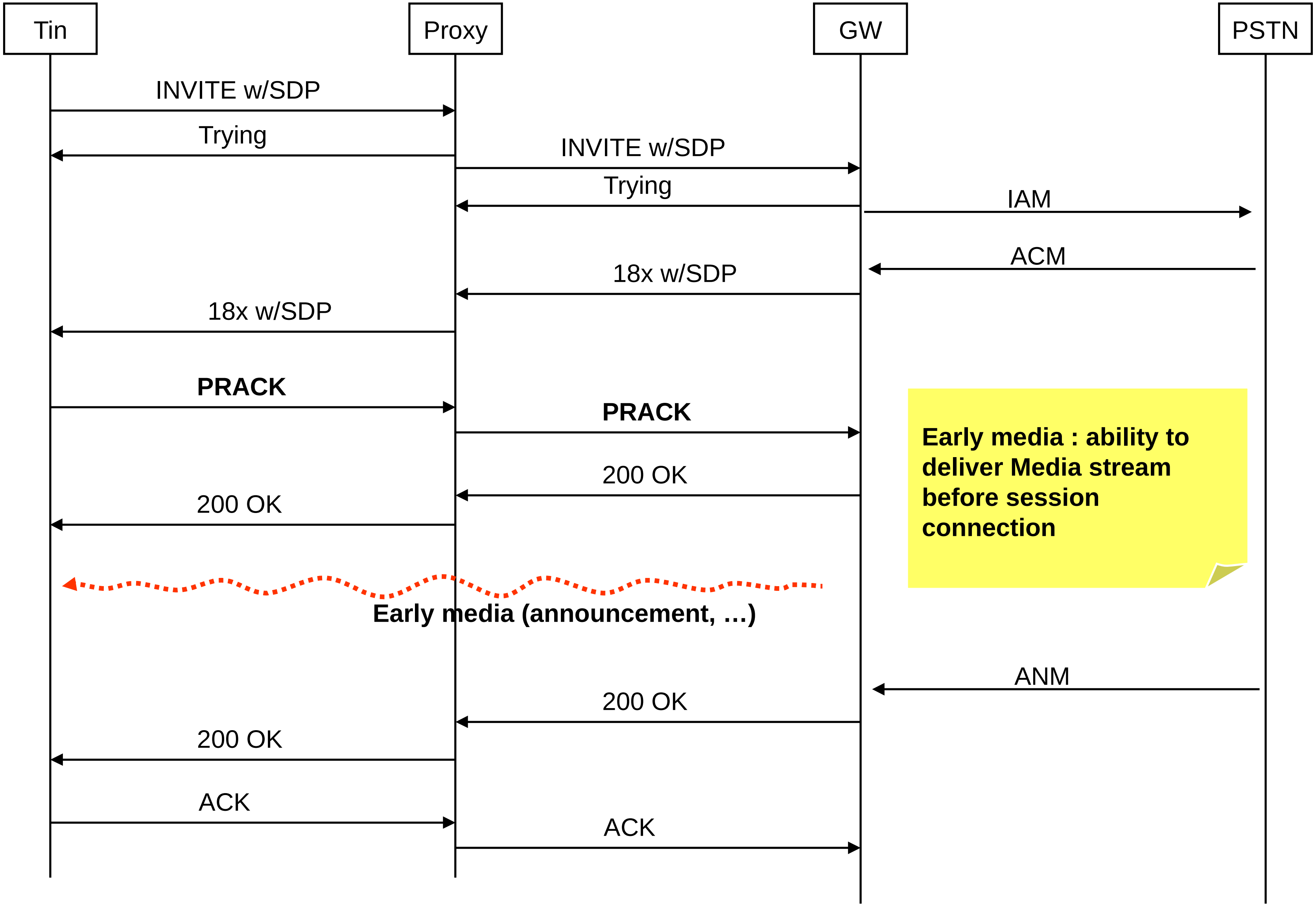
Pc-to-phone


DB
load
Proxy
francetelecom.com
INVITE sip: sip:0296051234@sip_gw1.lannion.francetelecom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP host2.francetelecom.com:5060
Via: SIP/2.0/UDP p1.francetelecom.com:5060
From: <sip:0296051234@francetelecom.com;user=phone>
To: <sip:029605111@francetelecom.com;user=phone>
INVITE sip:0296051234 @ francetelecom.com ;user=phone SIP/2.0
Via: SIP/2.0/UDP host2.francetelecom.com:5060
From: <sip:0296051111@francetelecom.com;user=phone>
To: <sip:0296051234@ francetelecom.com;user=phone>
SIP GW
sip_gw1.lannion.francetelecom.com
host2.francetelecom.com
phone=0296051111
Called : 0296051234
phone : 0296051234
Tin

Advanced functions – To remember – 1

- Subsequent INVITE or UPDATE to update a in-call dialog
- Forking mode create a new transaction (two branches)
- Removing Registration consists on setting:
  - Expires: 0
  - Contact: *
- PSTN interworking is possible thanks to SIP-PSTN gateways

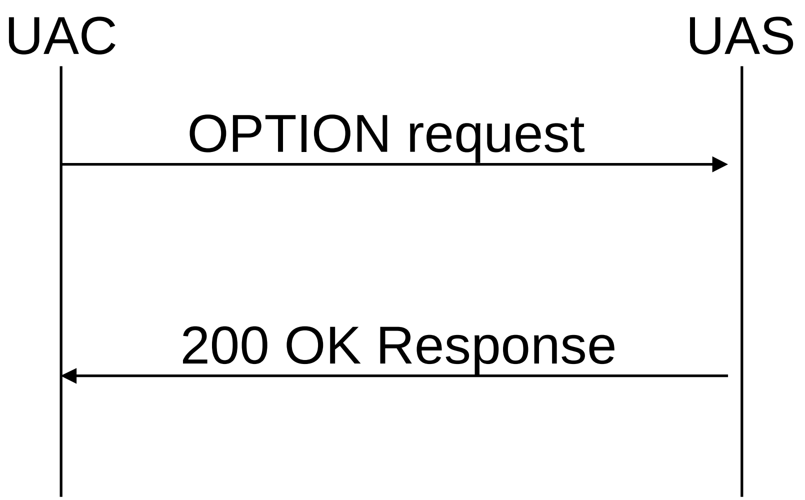
Early media (PSTN interworking) – PRACK – RFC 3262



Early media : ability to deliver Media stream before session connection

Query capabilities

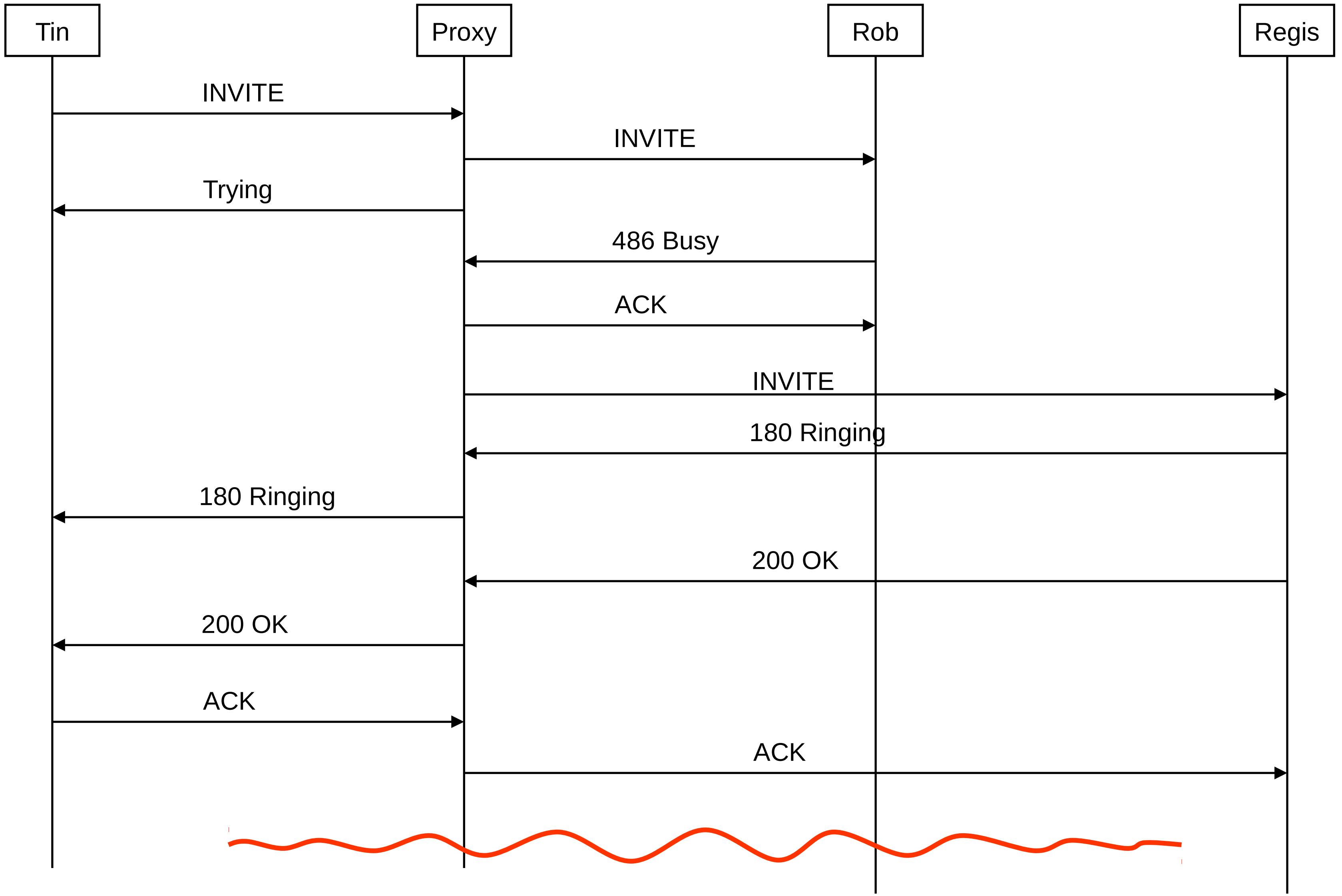
- After OPTION reception, UAS returns in 200 OK response its capabilities
  - Codec, SIP methods, languages…



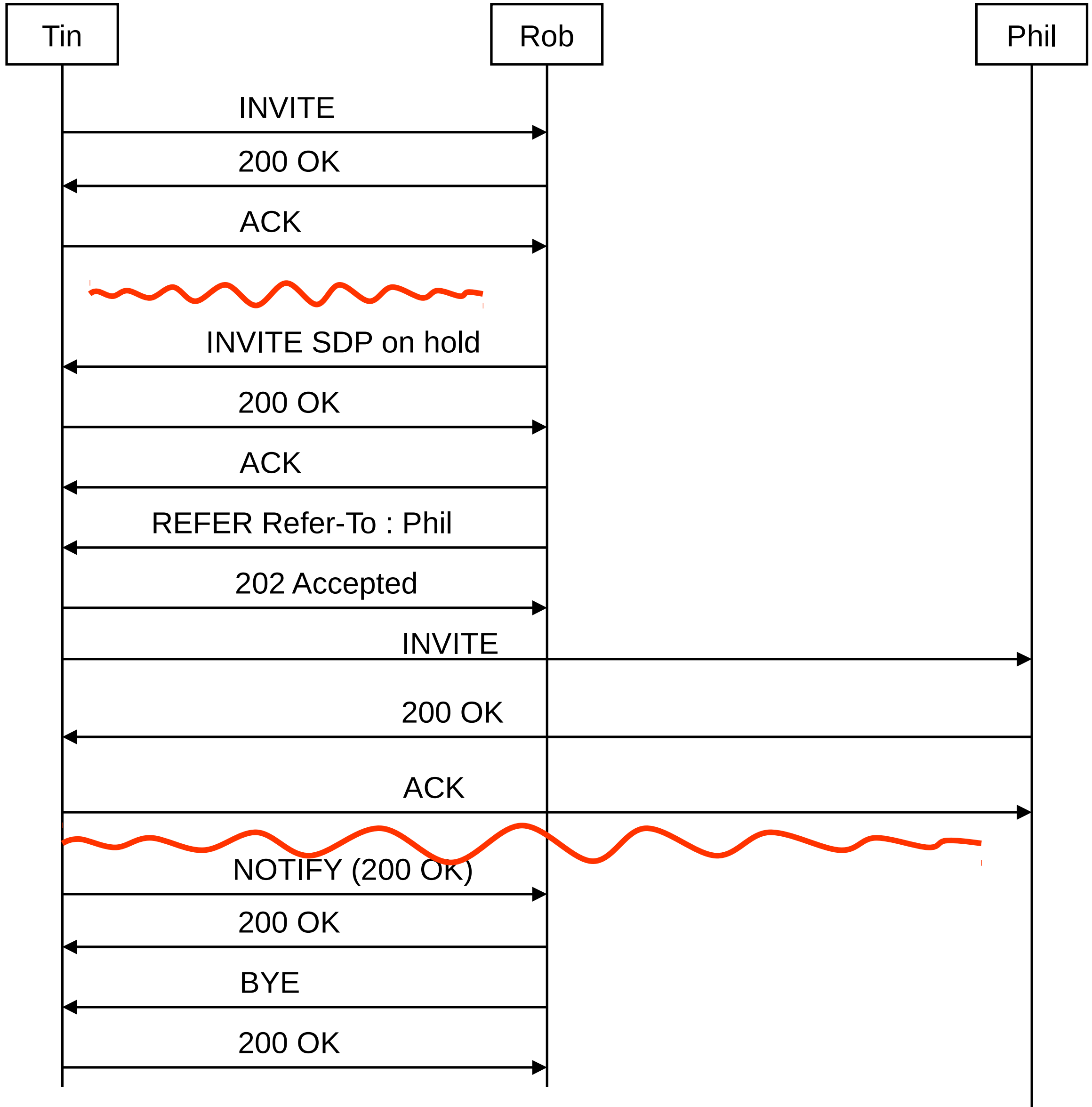
SIP/2.0 200 OK
From …
To …
Allow: INVITE, ACK, CANCEL, OPTIONS, BY
Accept: application/SDP
Accept-Language: en
Content-Type: application/SDP
Contact-Length: 232

v=0
o= - 562414 562414 IN IP4 172.32.12.32
s=-
c= IN IP4 172.32.12.32
m=audio 0 RTP/AVP 0 3
a=rtpmap:0 PCMU/8000
a=rtpmap:3 GSM/8000
m=video 0 RTP/AVP 31 34
a=rtpmap:31 H261/90000
a=rtpmap:34 H263/90000

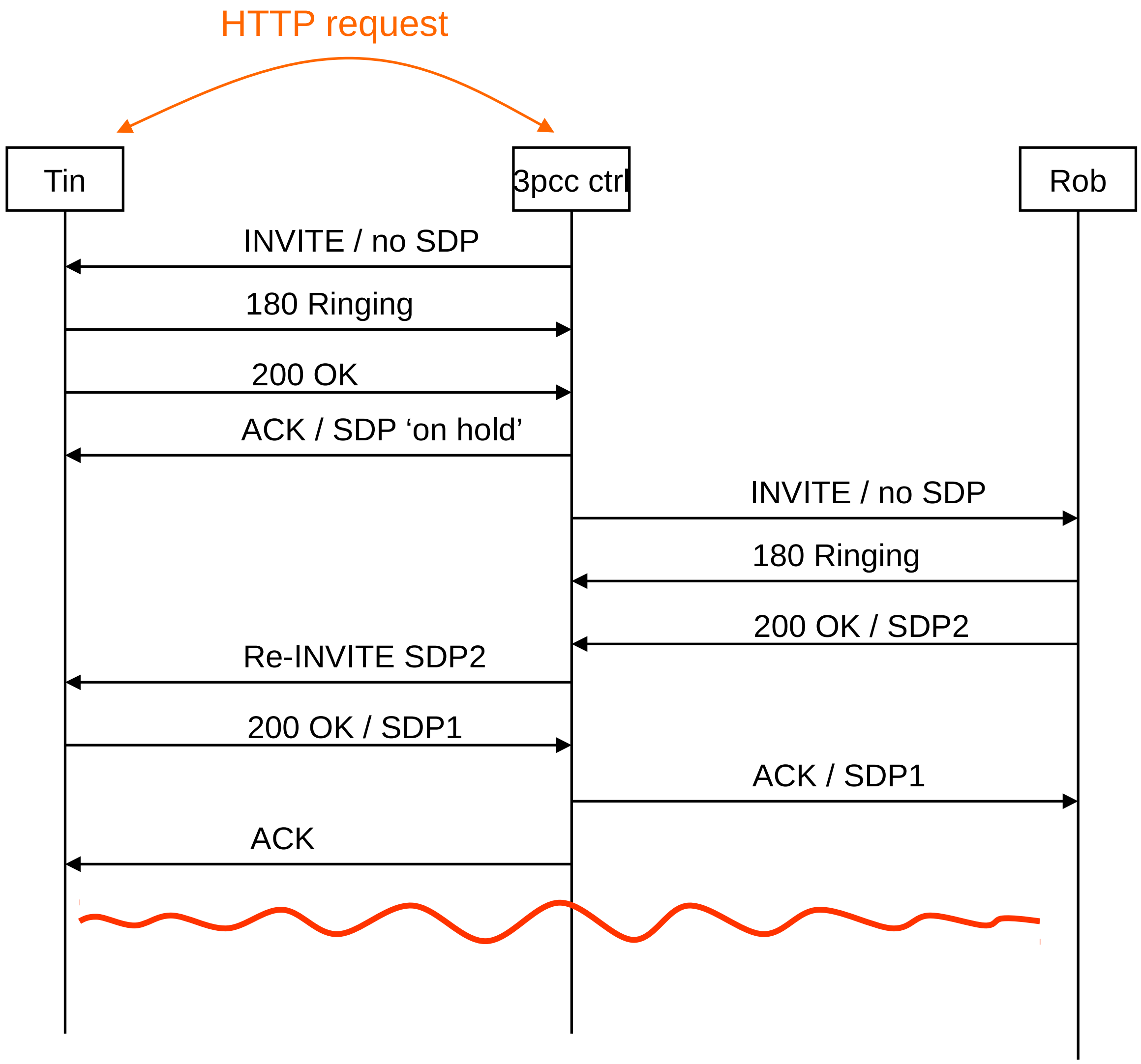
Supplementary Service : Forward



Supplementary Service : Transfer



3rd party call control (ex. Click to dial)



Advanced functions – To remember – 2

- Early Media and Provisional Ack (PRACK) are important to deliver inband call progress messages when interworking with PSTN
- OPTIONS Requests are used to query capabilities
- Call forward on Busy (CFB) can be performed by a SIP Server when receiving a 486 Busy Response
- Call transfer can be performed by a SIP Server when receiving a REFER Request

Summary

- **What is VoIP ?**
- **Focus on SIP protocol**
  - History
  - SIP Basis
  - Basic SIP dialog dissection
  - Registrar function
  - Proxy and Redirect Servers
  - Advanced functions
  - **SIP and security**
  - Retransmission
  - SIP and NAT/FW
  - Presence and Instant Messaging
- **Focus on media component (RTP/RTCP)**

Digest Authentication

- Based on HTTP authentication (RFC 2617)
- It does NOT guaranty message integrity in the default usage
- Authentication is applied to a SIP domain (a realm)
- This mechanism can be used for all request: INVITE, REGISTER…

- Mechanism:
  - UAC sends a request to its proxy/registrar
  - UAC receives a 401/407 response with a specific header (xxx-Authenticate). This header contains a "challenge"
  - UAC stores "challenge" info ("nonce", "opaque"…)
  - Based on this challenge and its password, the UAC calculates the response (with MD5 or SHA1 algorithm)
  - UAC re-sends its request with an "Authorization" header that contains the response. Cseq value is incremented
  - Proxy/registrar checks the validity of the response. If it's ok, request is acknowledged. Else, request is rejected

REGISTER Authentication

REGISTER sip:orange.com

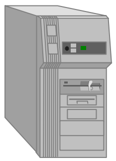
Cseq:1

SIP/2.0 401 Unauthorized

…

WWW-Authenticate: Digest realm="orange.com", qop="auth",
nonce="f84f1cec41e6cbe5aea9c8e88d359 ", opaque="", stale=FALSE, algorithm=MD5

Tin



Registrar

REGISTER sip:orange.com
Cseq: 2
Authorization: Digest **username="tin@orange.com"**, realm="orange.com",
nonce="f84f1cec41e6cbe5aea9c8e88d359", opaque="", uri="sip:orange.com" ,
response="d91jstdy65867dtydq32dsdzv628"

SIP/2.0 200 OK

INVITE Authentication

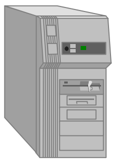
INVITE

Cseq: 1

SIP/2.0 **407 Proxy authentication required**
Proxy-Authenticate:Digest …

Tin

INVITE
Cseq: 2
Proxy-Authorization: Digest …



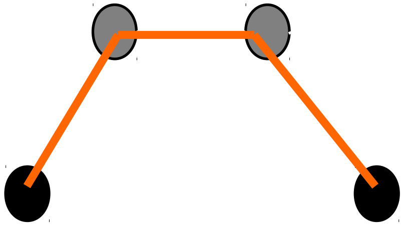
Proxy

SIP/2.0 200 OK

Confidentiality - Integrity

➨ **End to end encryption** :

difficult issue, no mechanism currently specified



Header fields :
used by proxy for routing

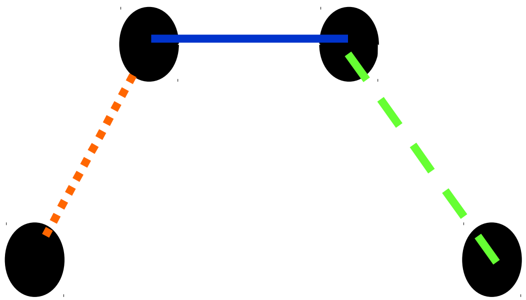
Body : MIME/SDP manipulate by particular proxies like certains Firewall Proxy

invite sip:Rob@orange.com SIP/2.0
Via: SIP/2.0/UDP host1.francetelecom.com:5060
Date: Wed, 04 Oct 2000 07:14:34 GMT
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>
Cseq: 1 INVITE
Call-ID: 124325617@host1.francetelecom.com
Contact: <sip:Tin@host1.francetelecom.com>
Call-Info: <http://wwww.ftrd.fr/Tin.html>;purpose=info
Subject: phone call
Content-Type: application/SDP
Content-Length: 148

v=0
o=Tin 562413 562413 IN IP4 194.240.47.217
s=phone call
c=IN IP4 194.240.47.217
m=audio 4710 RTP/AVP 4
a=rtpmap:4 G723/8000

Confidentiality - Integrity

- **Hop by hop encryption**
  - Network layer :
    - IPSec
  - Transport layer :
    - TLS (Transport Layer Security)
      - over TCP
      - SSL

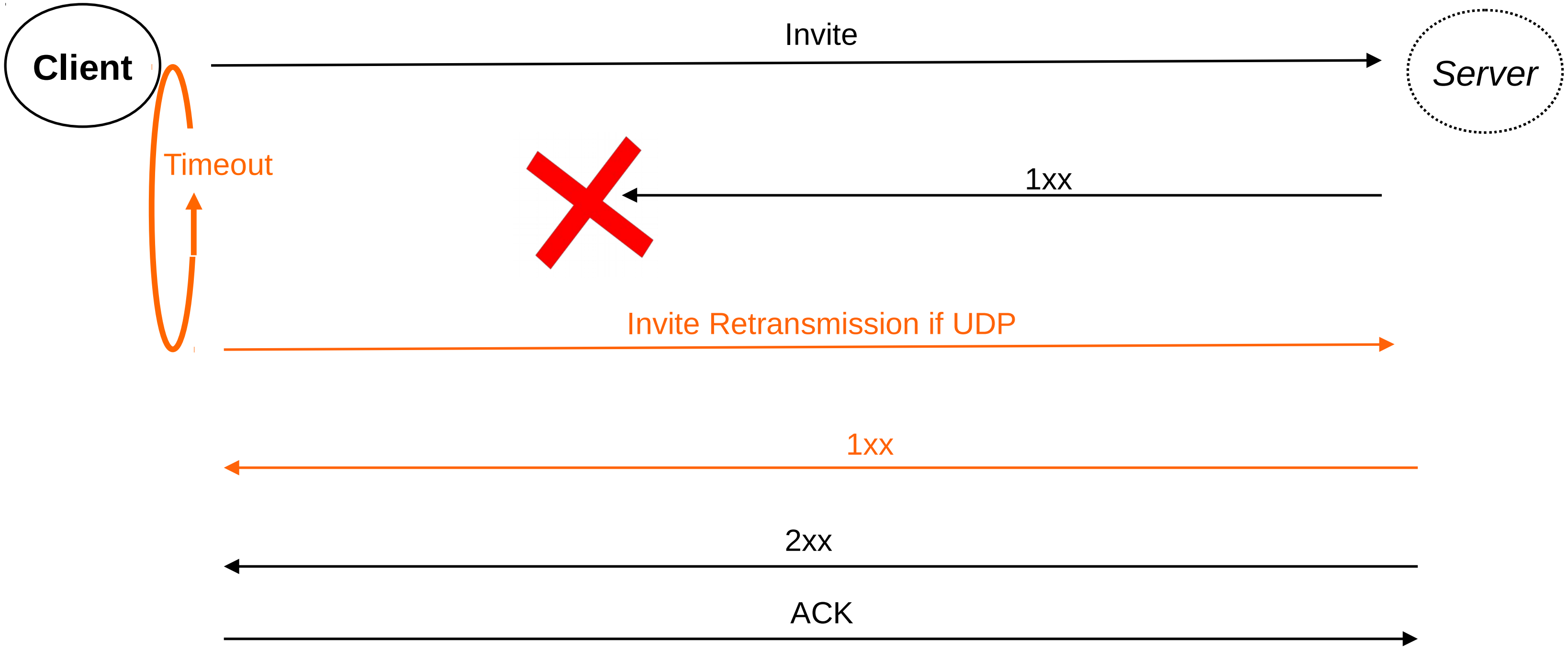


Summary

- **What is VoIP ?**
- **Focus on SIP protocol**
  - History
  - SIP Basis
  - Basic SIP dialog dissection
  - Registrar function
  - Proxy and Redirect Servers
  - Advanced functions
  - SIP and security
  - **Retransmission**
  - SIP and NAT/FW
  - Presence and Instant Messaging
- **Focus on media component (RTP/RTCP)**

Timers : Invite

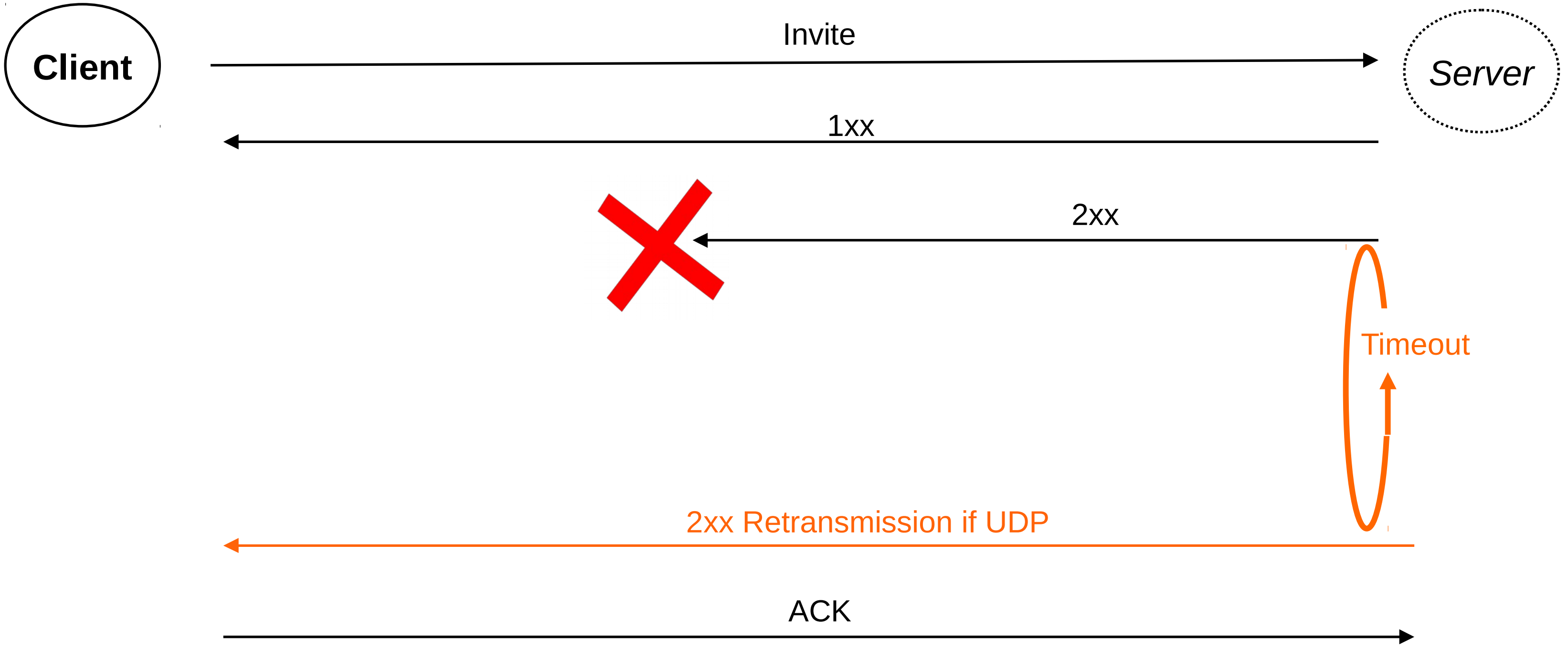
- Error while transmitting 1xx response from Server to Client



See RFC 3261 section 17.1.1.1

Timers : Invite

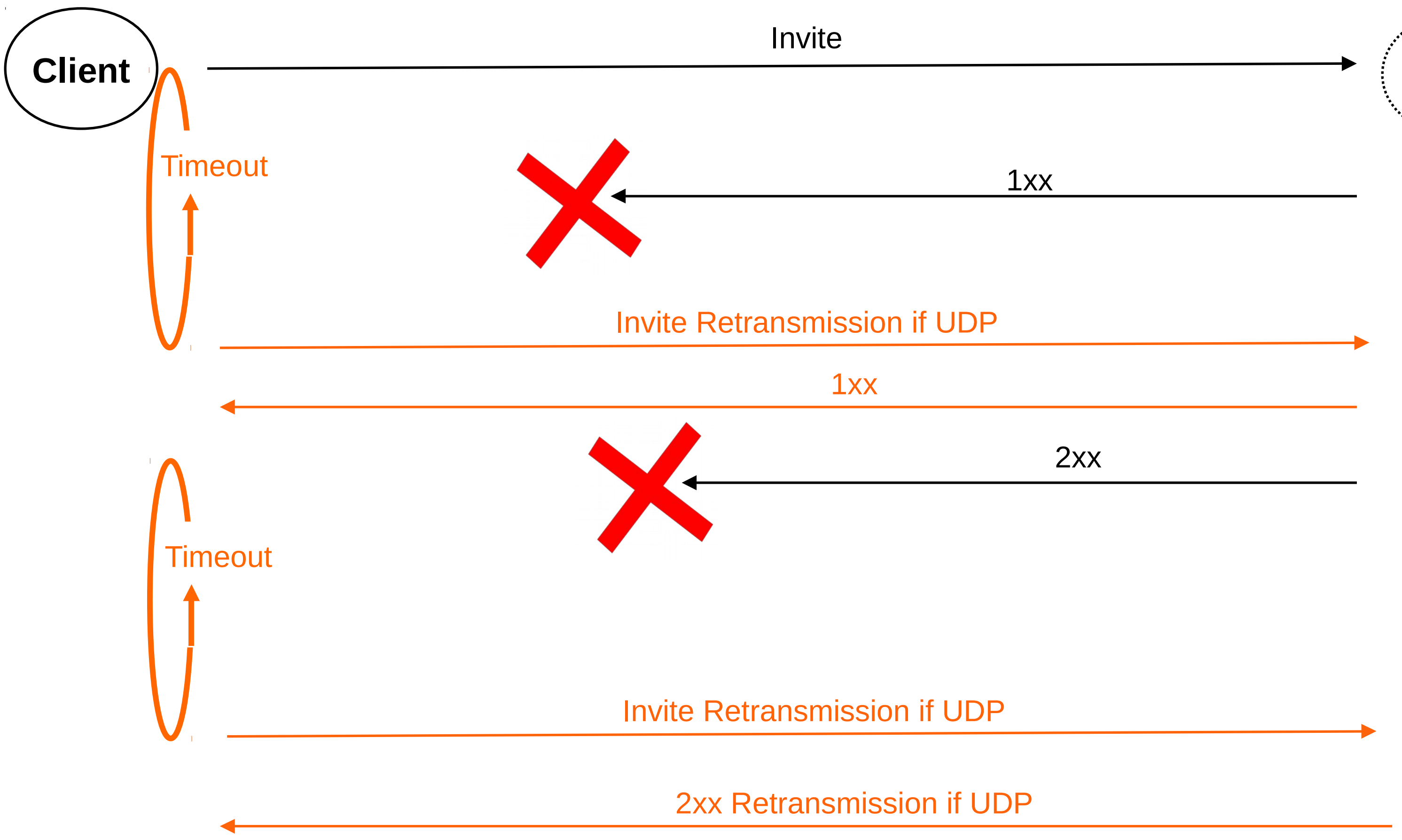
- Error while transmitting 2xx response from Server to Client



See RFC 3261 section 17.2.1

Timers : Non Invite

- Error while transmitting any response from Server to Client

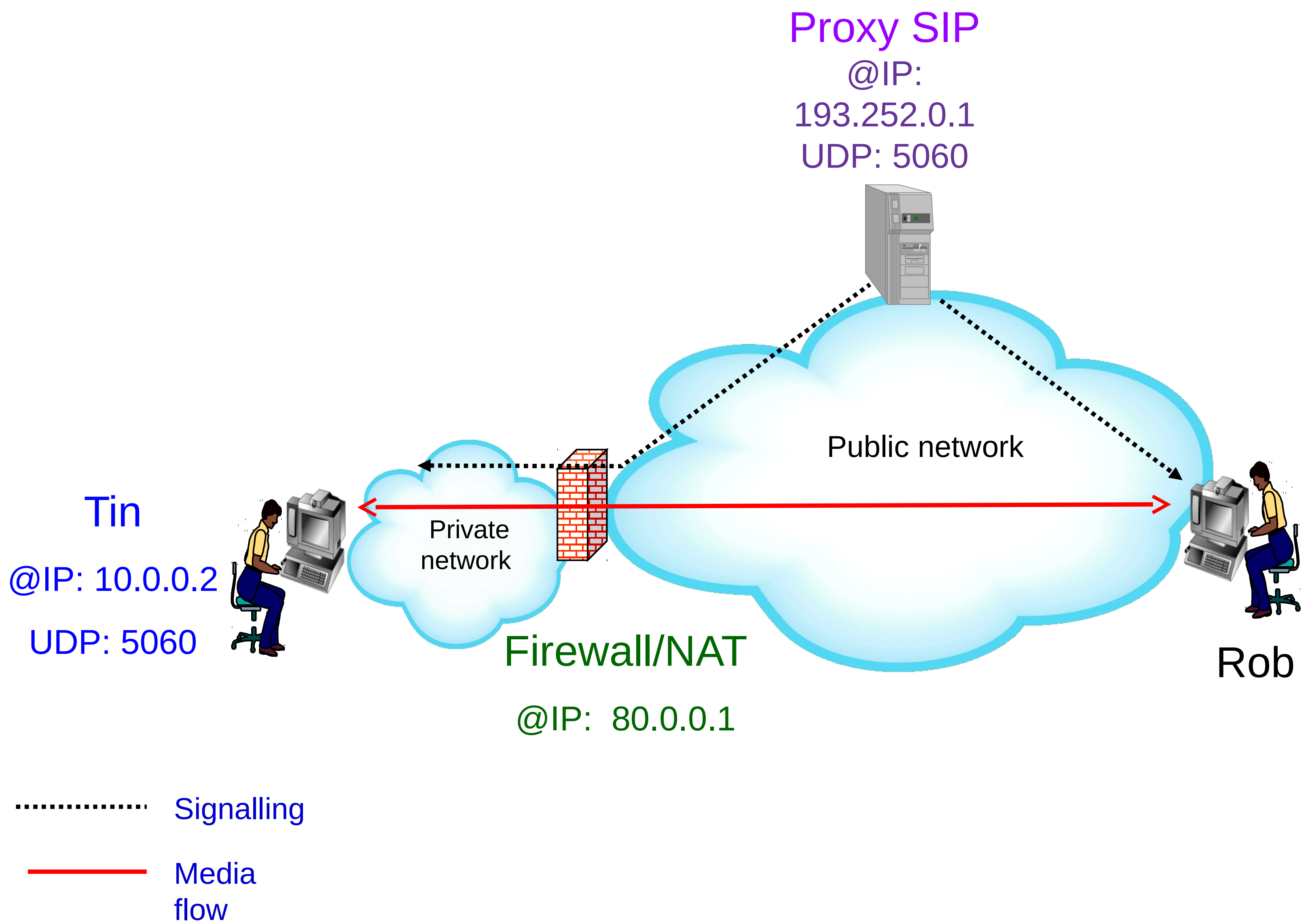


See RFC 3261 section 17.1.2.1 and 17.2.1

Summary

- **What is VoIP ?**
- **Focus on SIP protocol**
  - History
  - SIP Basis
  - Basic SIP dialog dissection
  - Registrar function
  - Proxy and Redirect Servers
  - Advanced functions
  - SIP and security
  - Retransmission
  - **SIP and NAT/FW**
  - Presence and Instant Messaging
- **Focus on media component (RTP/RTCP)**

NAPT/Firewall traversal



NAPT/FW traversal

NAT

Binding: 10.0.0.2:5060 <> 80.0.0.1:7000 ; expire=120s

INVITE sip:Rob@orange.com SIP/2.0
Via: SIP/2.0/UDP 10.0.0.2:5060
Date: Wed, 04 Oct 2000 07:14:34 GMT
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>
Cseq: 1 INVITE
Call-ID: 124325617@host1.francetelecom.com
Contact: <sip:Tin@10.0.0.2>
Call-Info:
<http://wwww.ftrd.fr/Tin.html>;purpose=info
Subject: phone call
Content-Type: application/SDP
Content-Length: 148

v=0
o=Tin 562413 562413 IN IP4 10.0.0.2
s=phone call
c=IN IP4 10.0.0.2
m=audio **4710** RTP/AVP 0 4
a=rtpmap:0 PCMU/8000
a=rtpmap:4 G723/8000
m=video 8524 RTP/AVP 31
a=rtpmap:31 H261/90000

UDP scr port : 5060	UDP dest port : 5060
@IP scr : 10.0.0.2	@IP scr : 193.252.0.1

Tin →

INVITE sip:Rob@orange.com SIP/2.0
Via: SIP/2.0/UDP 10.0.0.2:5060
Date: Wed, 04 Oct 2000 07:14:34 GMT
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>
Cseq: 1 INVITE
Call-ID: 124325617@host1.francetelecom.com
Contact: <sip:Tin@10.0.0.2>
Call-Info:
<http://wwww.ftrd.fr/Tin.html>;purpose=info
Subject: phone call
Content-Type: application/SDP
Content-Length: 148

v=0
o=Tin 562413 562413 IN IP4 10.0.0.2
s=phone call
c=IN IP4 10.0.0.2
m=audio **4710** RTP/AVP 0 4
a=rtpmap:0 PCMU/8000
a=rtpmap:4 G723/8000
m=video 8524 RTP/AVP 31
a=rtpmap:31 H261/90000

UDP scr port : 7000	UDP dest port : 5060
@IP scr : 80.0.0.1	@IP scr : 193.252.0.1

→ Rob

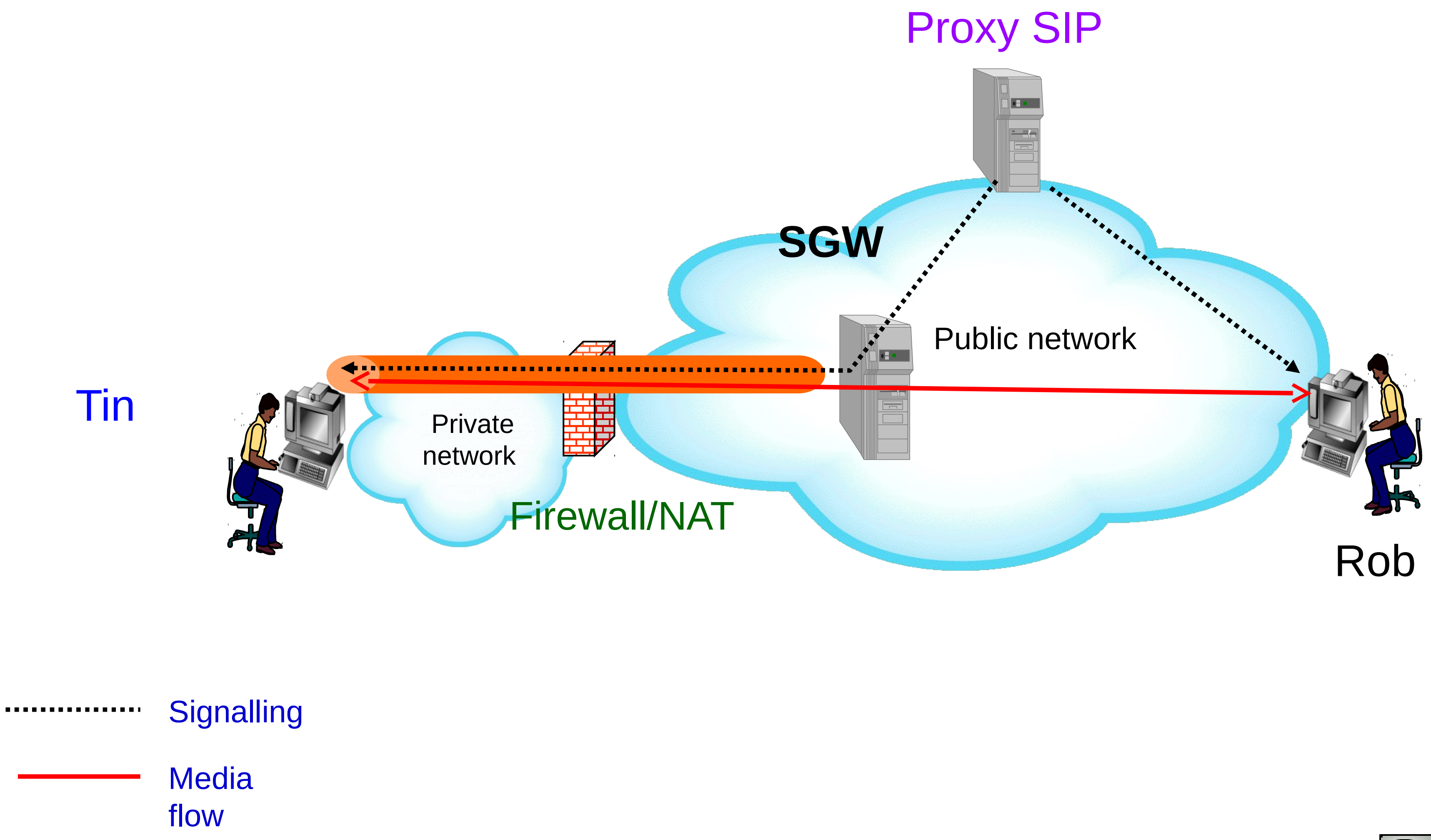
NAPT/FW traversal

- Problem to be solved:
  - NAPT doesn't modify @IP and ports at SIP/SDP layers
  - Private @IP is not-routable from the public network
  - Local NAPT bindings will 'time-out' if no packet refreshes NAPT rules
  - Same problem for RTP/RTCP streams (dynamic ports)

Current solutions

- The current solutions for solving NAPT and FW problems are:
  - To configure static port in terminal and static NAPT
  - To use UPnP (Universal Plug and Play)
  - To establish one tunnel (IPsec) between the UA and a Secure Gateway (SGW). All SIP and RTP flows are embedded in this tunnel
  - To implement an Application Layer Gateway (ALG). ALG modifies SIP/SDP layers in coherence with NAPT (layers 2 and 3)
  - To implement STUN (Simple Traversal of UDP Trough Network Address Translators – RFC 3489) or ICE (Interactive Connectivity Establishment)

IPsec tunnel



ALG SIP

10.0.0.2:5060 <> 80.0.0.1:7000
10.0.0.2:4710 <> 80.0.0.1:9400
10.0.0.2:4711 <> 80.0.0.1:9401

```
INVITE sip:Rob@orange.com SIP/2.0
Via: SIP/2.0/UDP 10.0.0.2:5060
Date: Wed, 04 Oct 2000 07:14:34 GMT
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>
Cseq: 1 INVITE
Call-ID: 124325617@host1.francetelecom.com
Contact: <sip:Tin@10.0.0.2>
Call-Info:
<http://wwww.ftrd.fr/Tin.html>;purpose=info
Subject: phone call
Content-Type: application/SDP
Content-Length: 148

v=0
o=Tin  562413 562413 IN IP4  10.0.0.2
s=phone call
c=IN IP4 10.0.0.2
m=audio 4710 RTP/AVP 0 4
a=rtpmap:0 PCMU/8000
a=rtpmap:4 G723/8000
m=video 8524 RTP/AVP 31
a=rtpmap:31 H261/90000
```

UDP scr port : 5060	UDP dest port : 5060
@IP scr : 10.0.0.2	@IP scr : 193.252.0.1

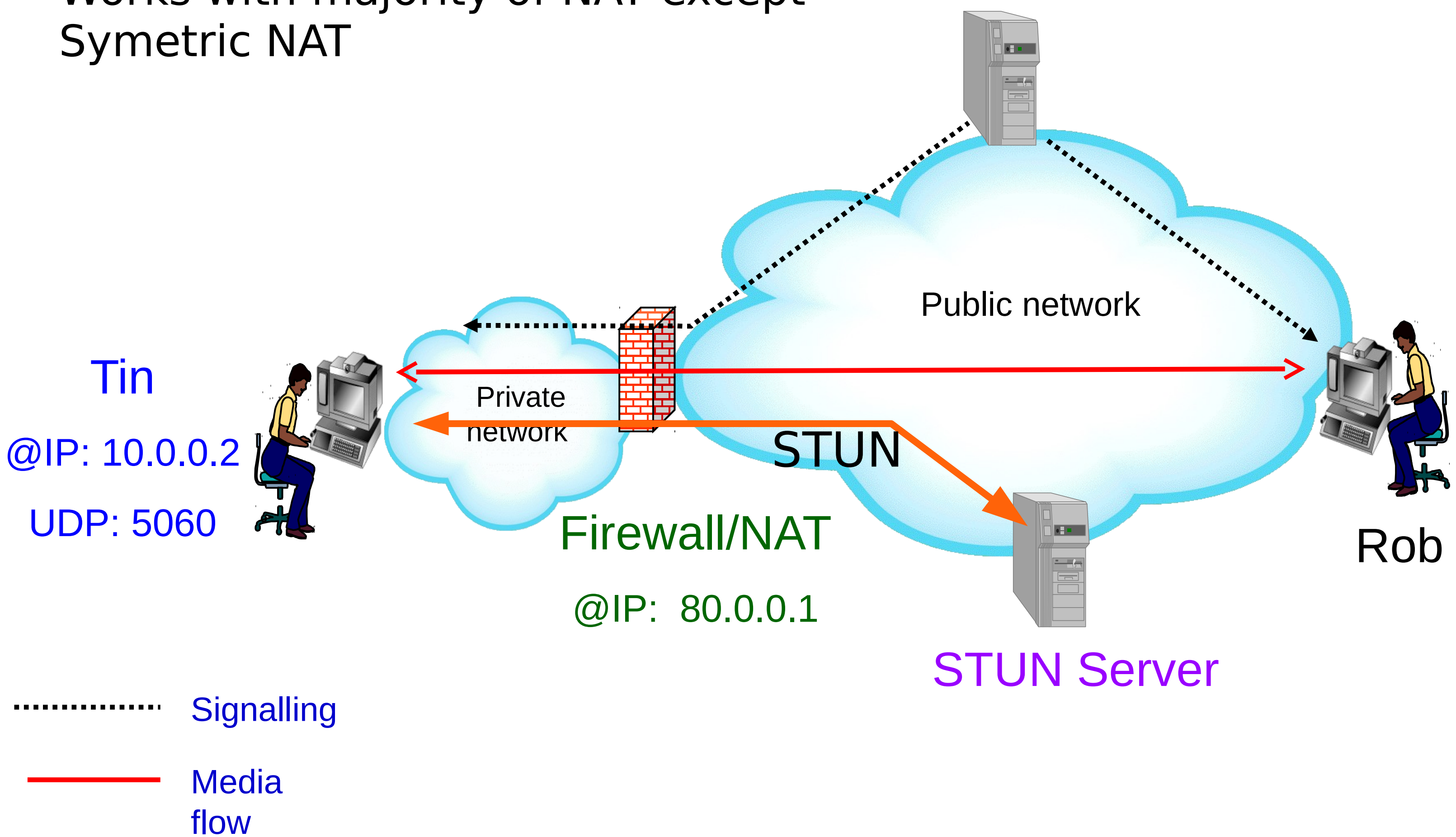
```
INVITE sip:Rob@orange.com SIP/2.0
Via: SIP/2.0/UDP 80.0.0.1:7000
Date: Wed, 04 Oct 2000 07:14:34 GMT
From: Tin <sip:Tin@francetelecom.com>
To: Rob <sip:Rob@orange.com>
Cseq: 1 INVITE
Call-ID: 124325617@host1.francetelecom.com
Contact: <sip:Tin@80.0.0.1:7000>
Call-Info:
<http://wwww.ftrd.fr/Tin.html>;purpose=info
Subject: phone call
Content-Type: application/SDP
Content-Length: 148

v=0
o=Tin  562413 562413 IN IP4  80.0.0.1
s=phone call
c=IN IP4 80.0.0.1
m=audio 9400 RTP/AVP 0 4
a=rtpmap:0 PCMU/8000
a=rtpmap:4 G723/8000
m=video 8524 RTP/AVP 31
a=rtpmap:31 H261/90000
```

UDP scr port : 7000	UDP dest port : 5060
@IP scr : 80.0.0.1	@IP scr : 193.252.0.1

Tin → → Rob

STUN

- Allow an end host to discover its public IP address & port
- Works with majority of NAT except Symetric NAT



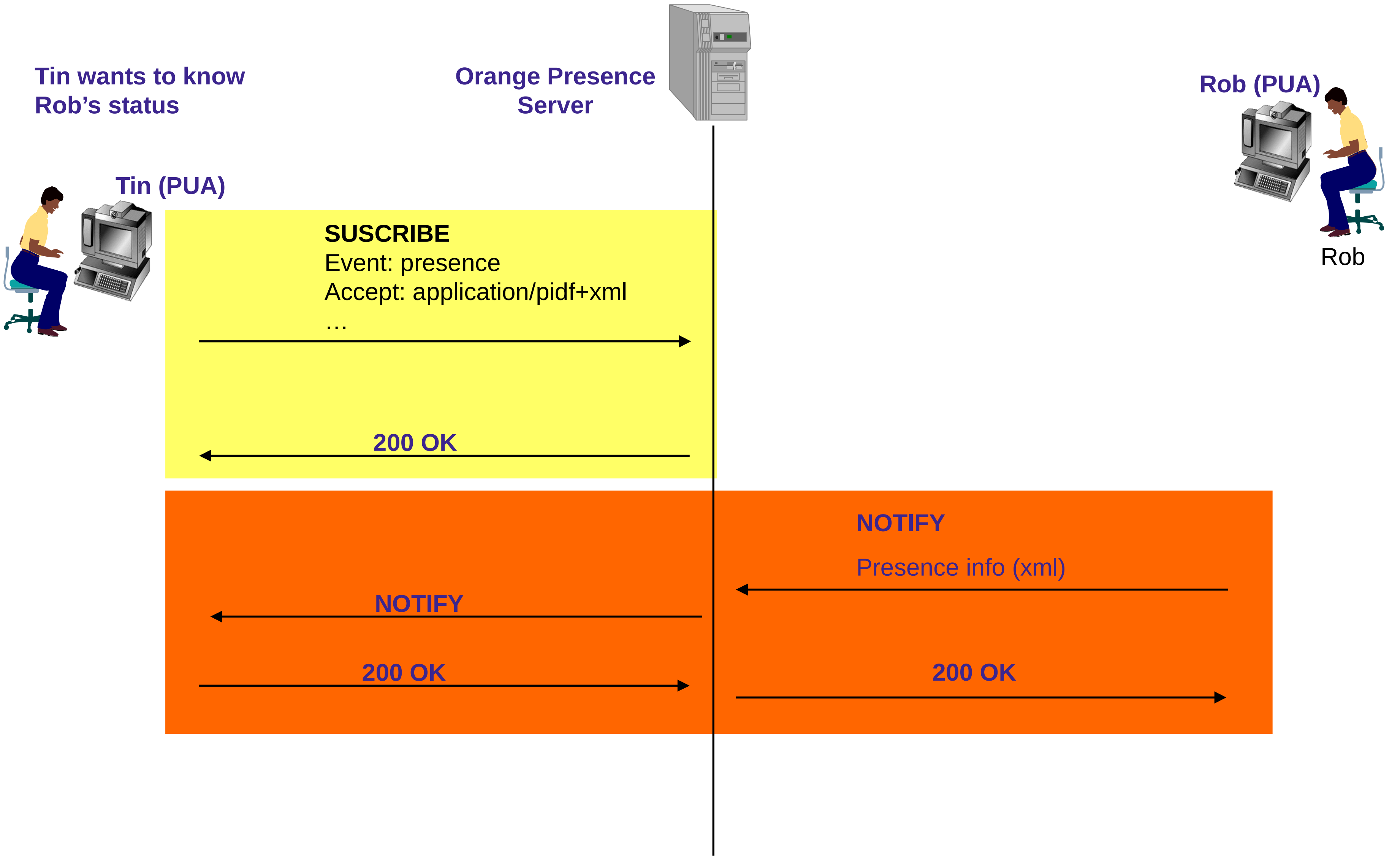
Summary

- **What is VoIP ?**
- **Focus on SIP protocol**
  - History
  - SIP Basis
  - Basic SIP dialog dissection
  - Registrar function
  - Proxy and Redirect Servers
  - Advanced functions
  - SIP and security
  - Retransmission
  - SIP and NAT/FW
  - **Presence and Instant Messaging**
- **Focus on media component (RTP/RTCP)**

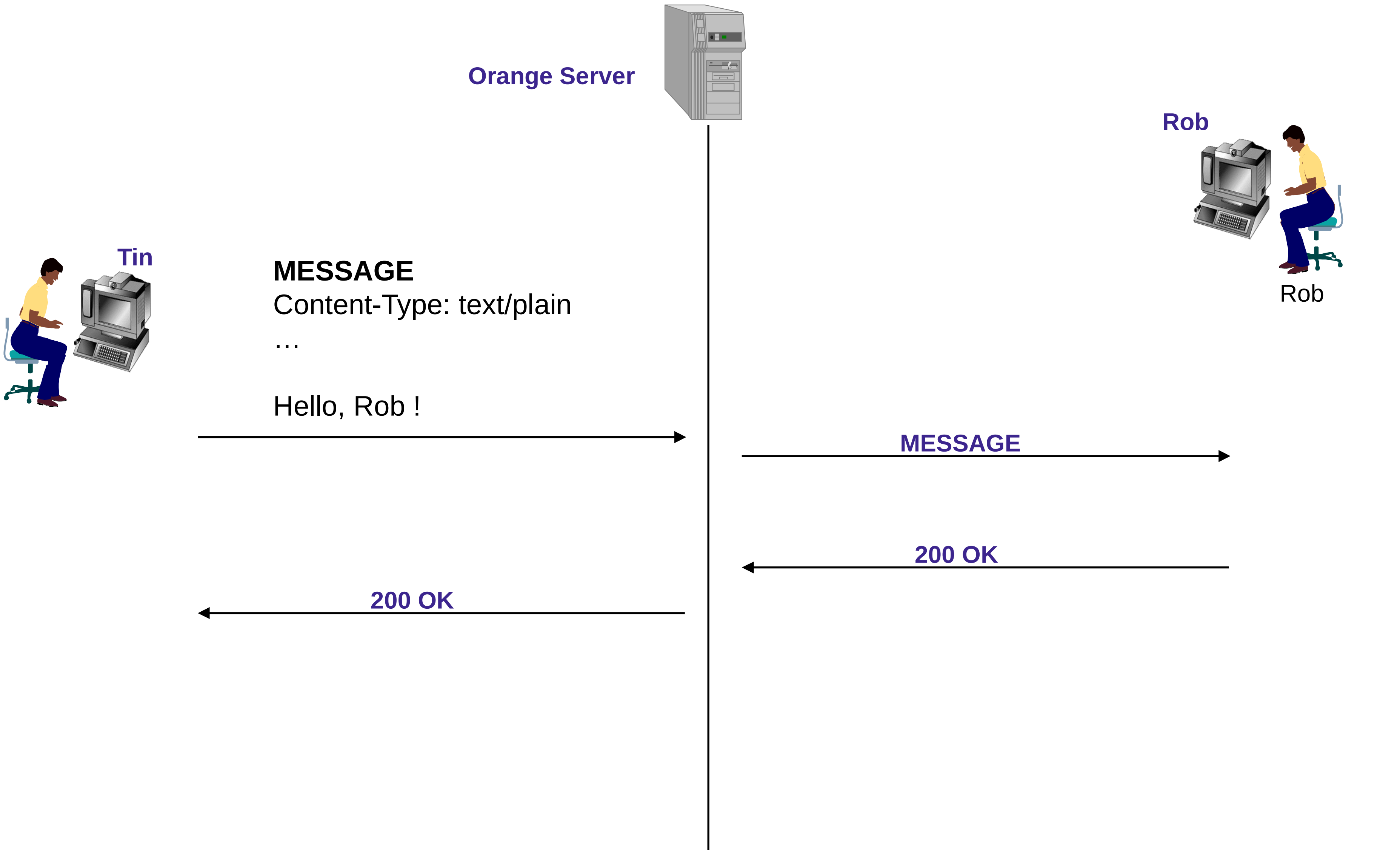
SIP for Presence – Instant Messaging

- Presence is the subscription and notification of changes in the communications state of a user.
  - Presence information can include :
    - Willingness to accept communication
    - Preferred medium for communication
    - Physical location
    - Call state
  - Presentity : publish and distribute presence information to watcher
  - Watcher : subscribe to know presentity information
- SIMPLE WG define Presence based on the SIP Event Notification model and CPIM framework (Common framework for Presence and IM)

SIMPLE – Presence



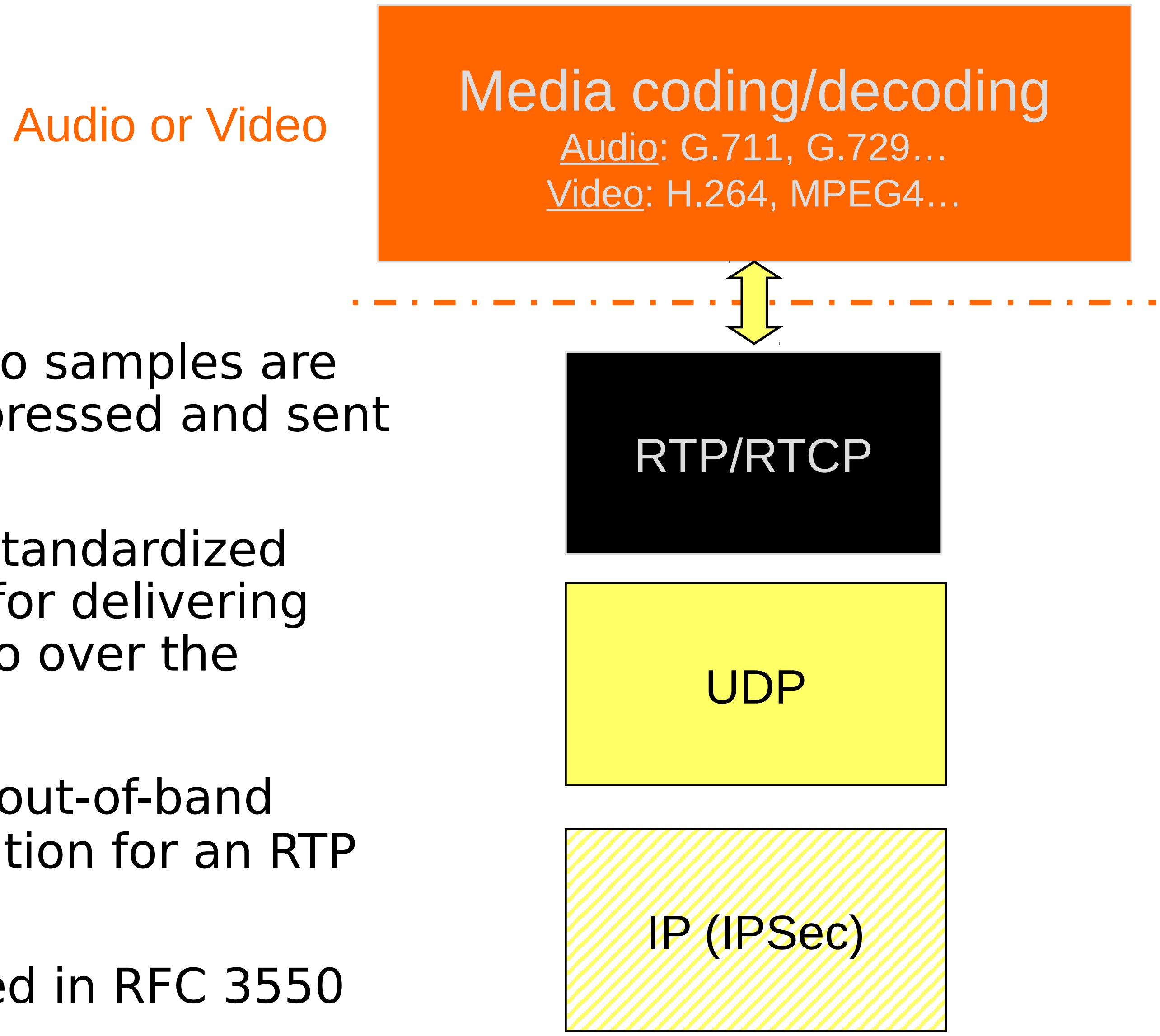
SIMPLE – Instant Messaging



Summary

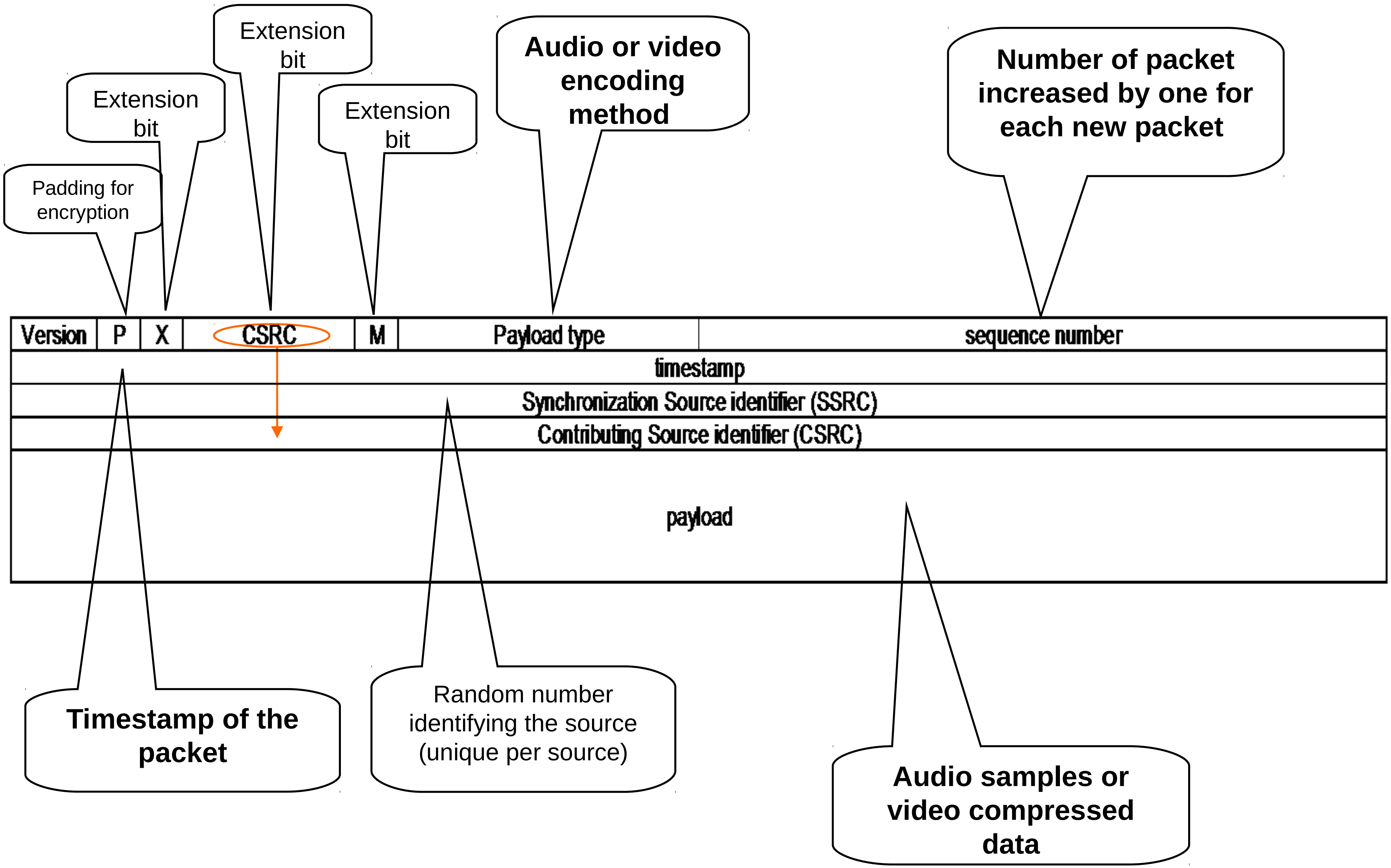
- **What is VoIP ?**
- **Focus on SIP protocol**
- **Focus on media component** (RTP/RTCP)

Protocol Layers

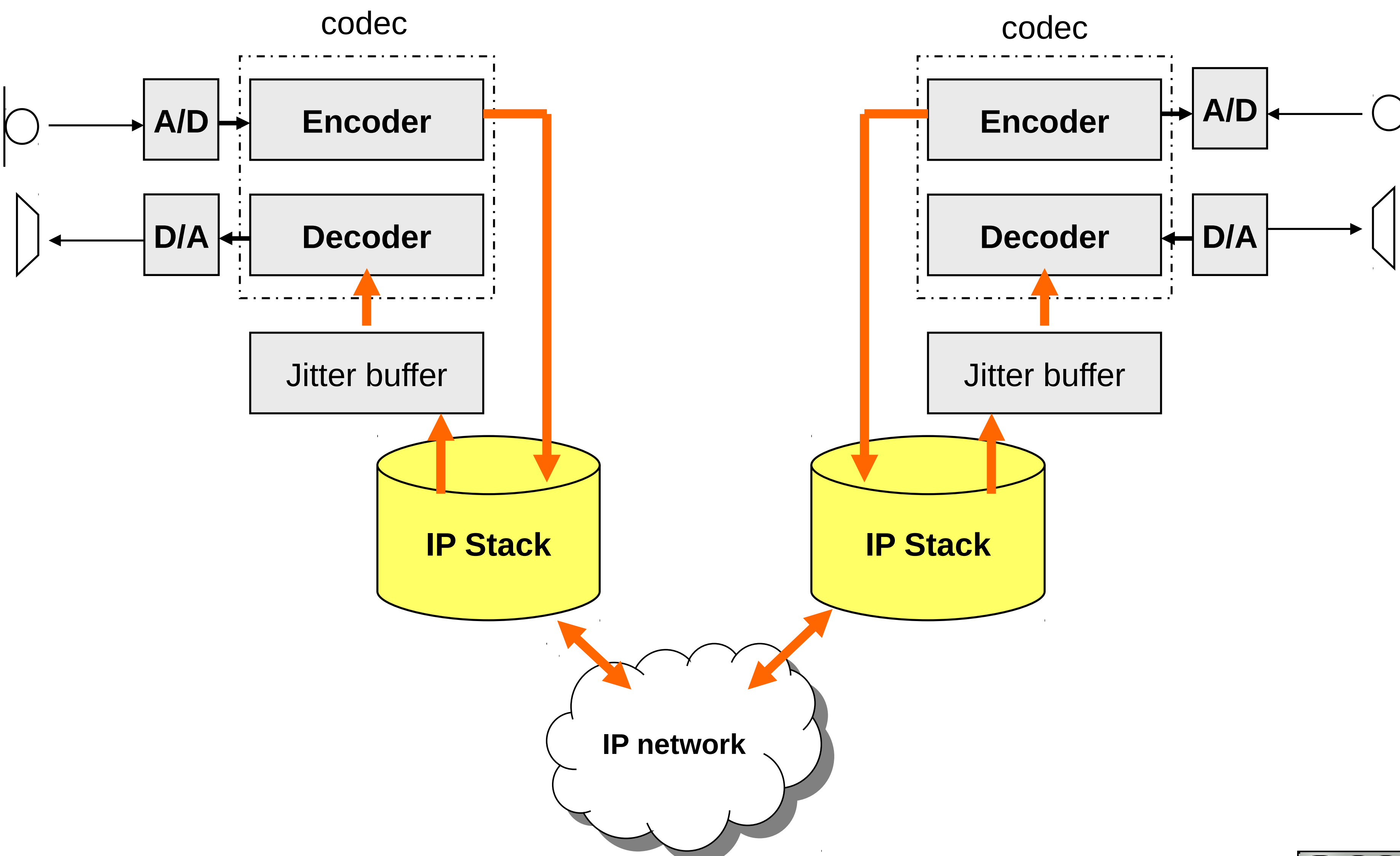


- Audio and Video samples are digitized, compressed and sent in UDP packets
- RTP defines a standardized packet format for delivering audio and video over the Internet
- RTCP provides out-of-band control information for an RTP flow
- Both are defined in RFC 3550

RTP packet structure

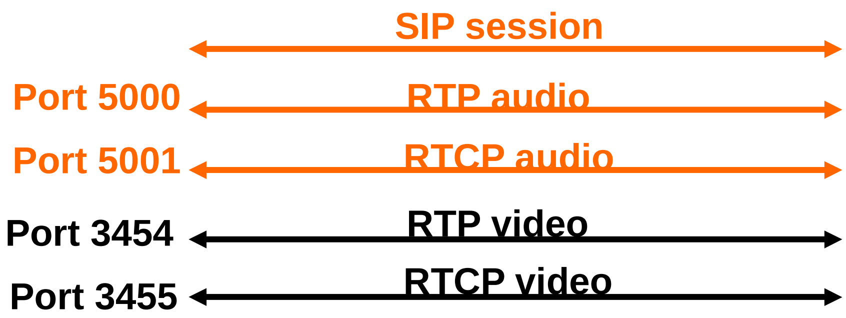


Media Path


codec
codec
A/D
Encoder
D/A
Decoder
Jitter buffer
IP Stack
Encoder
A/D
Decoder
D/A
Jitter buffer
IP Stack
IP network

RTCP

- For each RTP session, one RTCP session can be established
- RTCP port = RTP port + 1
  - RTP uses even port number
  - RTCP uses odd port number



- RTCP defines several type of packets
  - **Sender Report (SR)**: information about sent data, synchronization timestamp
  - **Receiver Report (RR)**: information about received data, jitter, loss, delay
  - **Source Description (SDES)**: name, email, phone…
  - **Bye:** end of participation

Questions ?