VoIP and SIP
(Session Initiation Protocol)

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Summary

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

Summary

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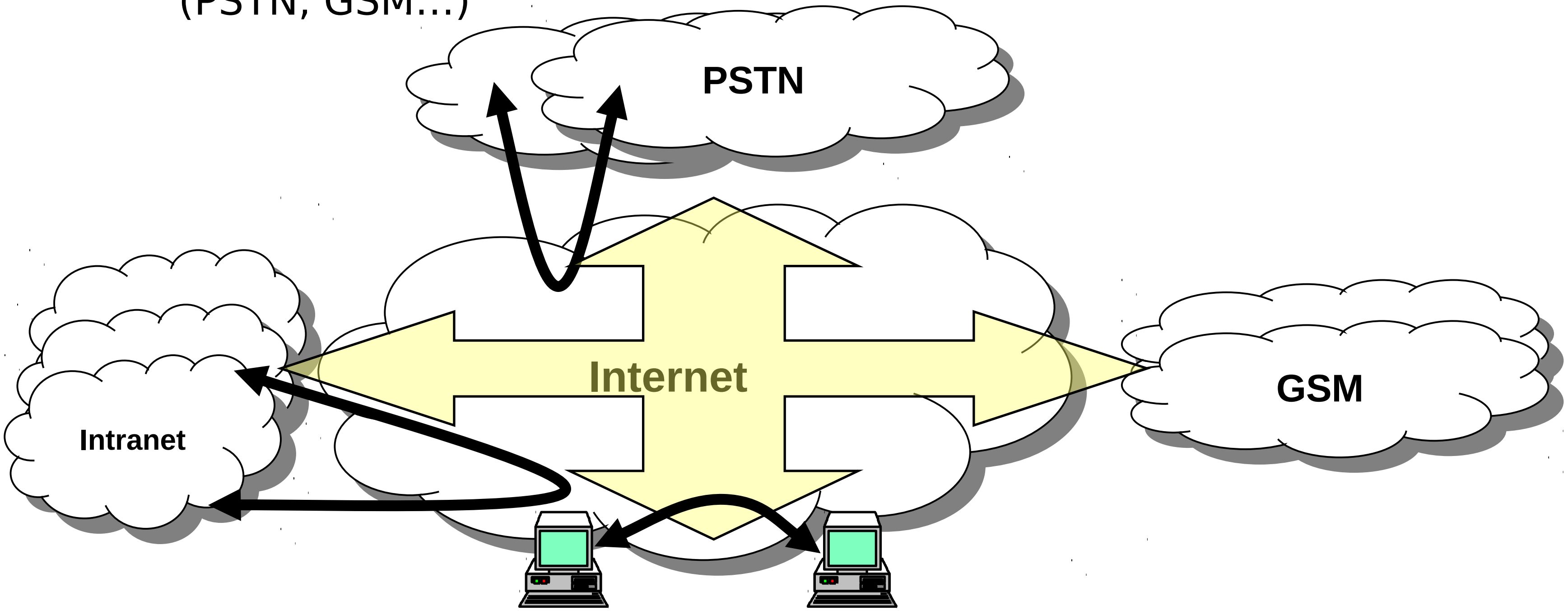
- **Focus on SIP protocol**
- **Focus on media component (RTP/RTCP)**
- **Examples**

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 (still exists but to handle data now)

Before VoIP

- Manual circuit switching

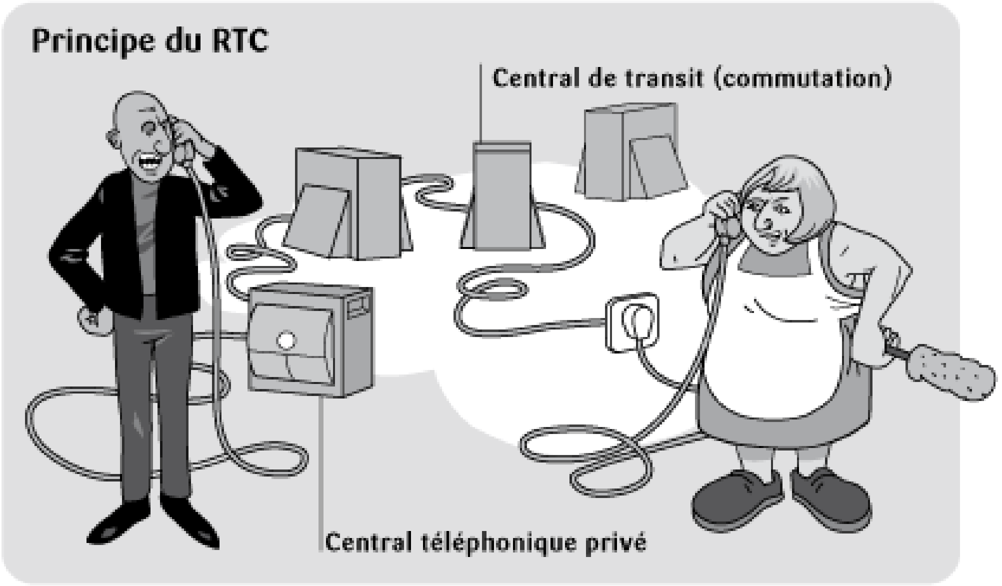
Before VoIP

- Automatic circuit switching

From PSTN (RTC in french) to VoIP

Circuit Switching → Packet Switching (data)

Dedicated line → All channels over Internet connection



VoIP needs

- **Signalling protocol**
- **Media Transport protocols which includes :**
  - **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

- Telecom providers
  - VoIP service with broadband offer
    - XXX Box
  - VoIP trunkings (mostly for professionals)
- Service providers
  - Skype
  - FaceTime
  - WhatApps
  - etc
- Other things
  - IP PBX
  - Cisco, Nortel, Alcatel…

Summary

- **What is VoIP ?**
- **Focus on SIP protocol**
  - **SIP Basis**
  - Basic SIP dialog dissection
  - Registrar
  - Proxy
  - Advanced functions
  - SIP and security
  - SIP and NAT/FW
- **Focus on media component (RTP/RTCP)**

SIP

- SIP = Session Initiation Protocol
- SIP protocol is defined within IETF (Internet Engineering Task Force) under **RFC 3261**

→ **http://tools.ietf.org/html/rfc3261**

- Created in 2002

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

SIP is ...

- A signalling protocol

... 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

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

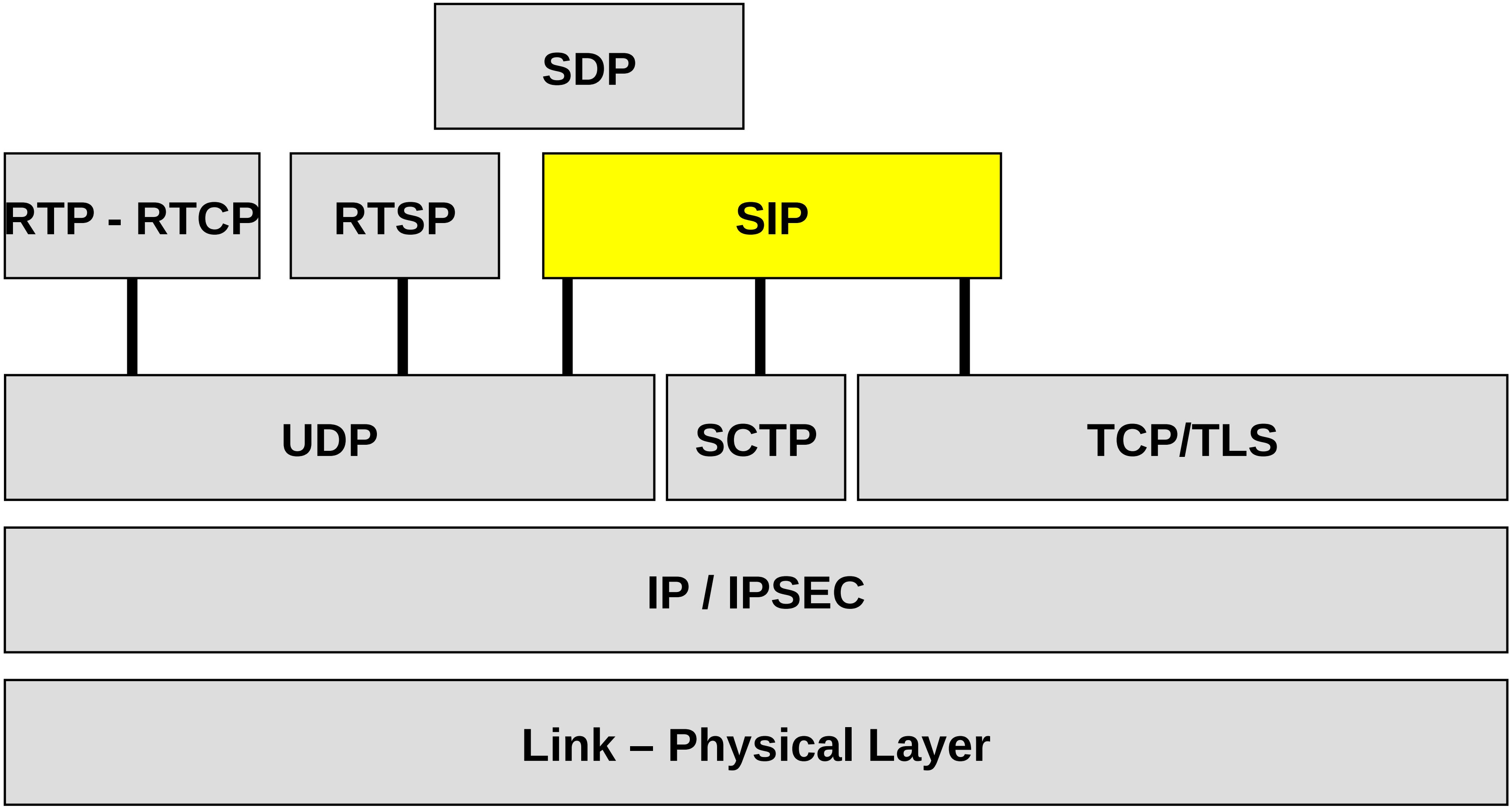
Stored Media
Client-Server
Stateful
URL rtsp://server.com/movies/twister:5080
Methods : DESCRIBE, SETUP, PLAY, RECORD, PAUSE, TEARDOW

SDP

Session Description Protocol RFC 4566

Media Stream, Addresses, Ports, RTP Payload…

SIP in OSI model



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**

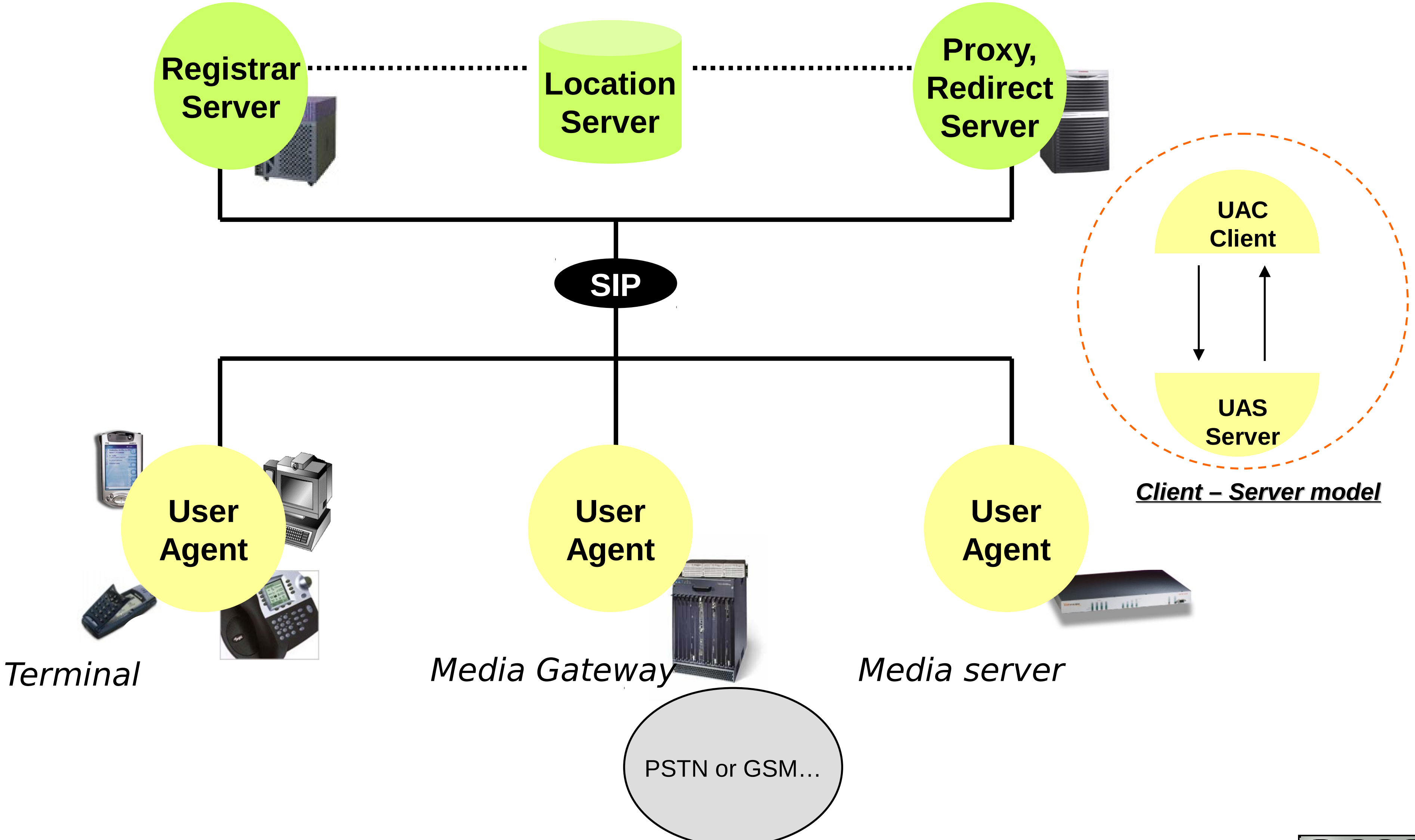
  Server that returns specific response (3xx) in order to redirect client to another destination
- **Proxy Server**

  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 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:

sip URI (or tel URI)

sip:chantal.martin@francetelecom.com

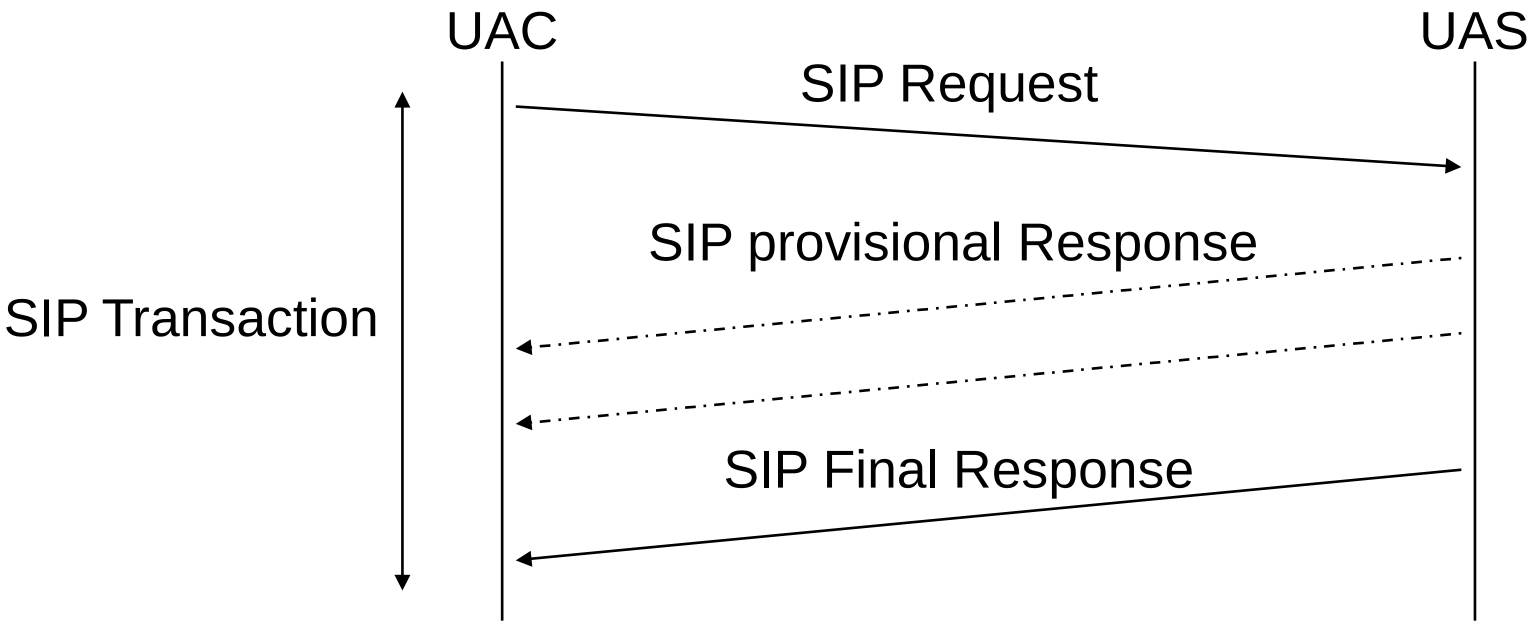
sip:chantal.martin@162.23.21.24

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

tel:+33296053017

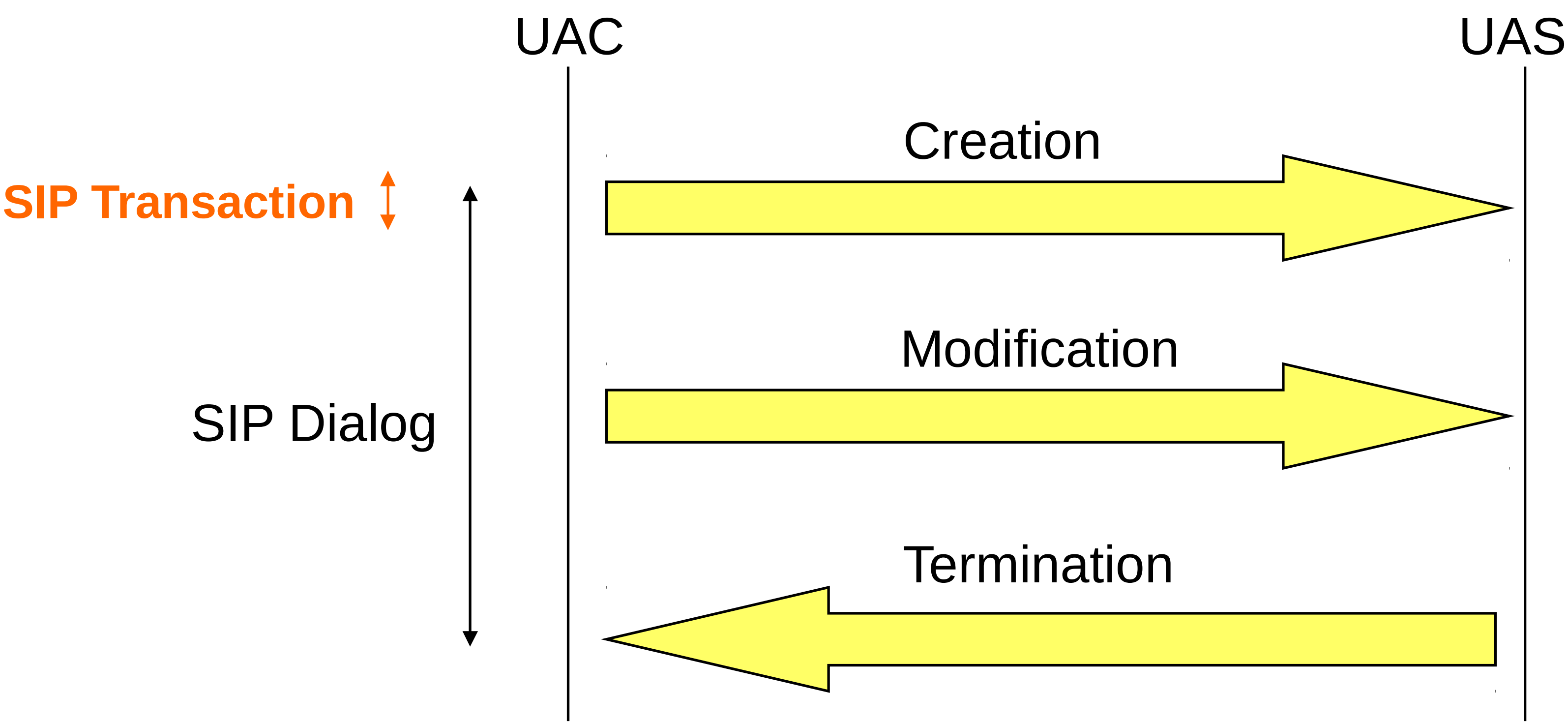
SIP transaction

- 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

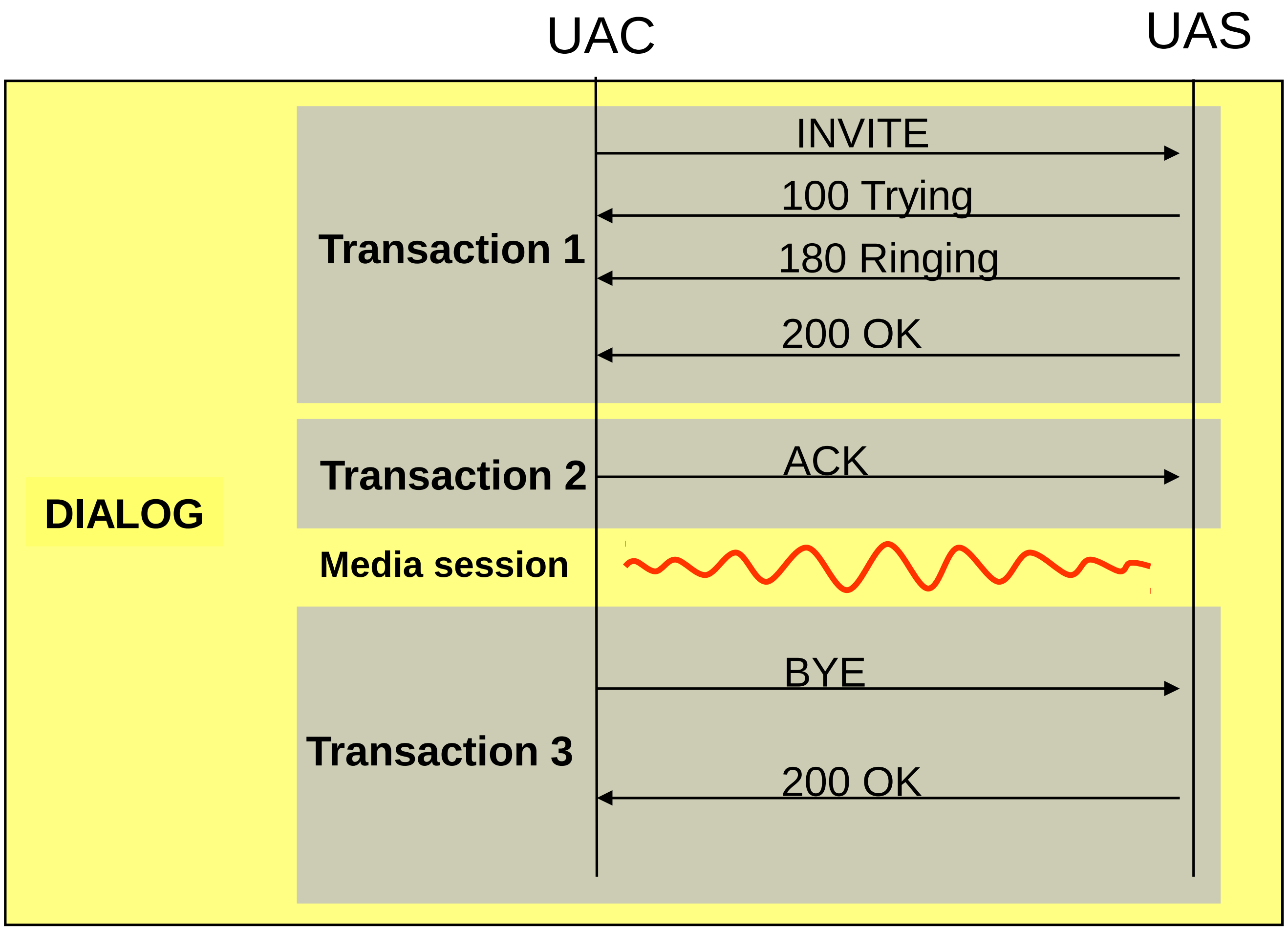


SIP dialog

- 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

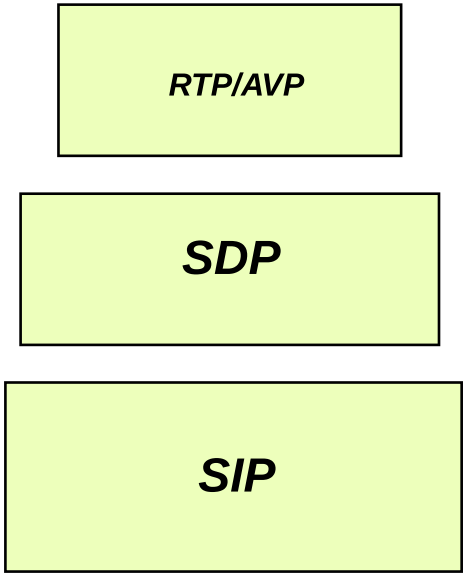
FSM Invite Client Transaction

```
                               |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

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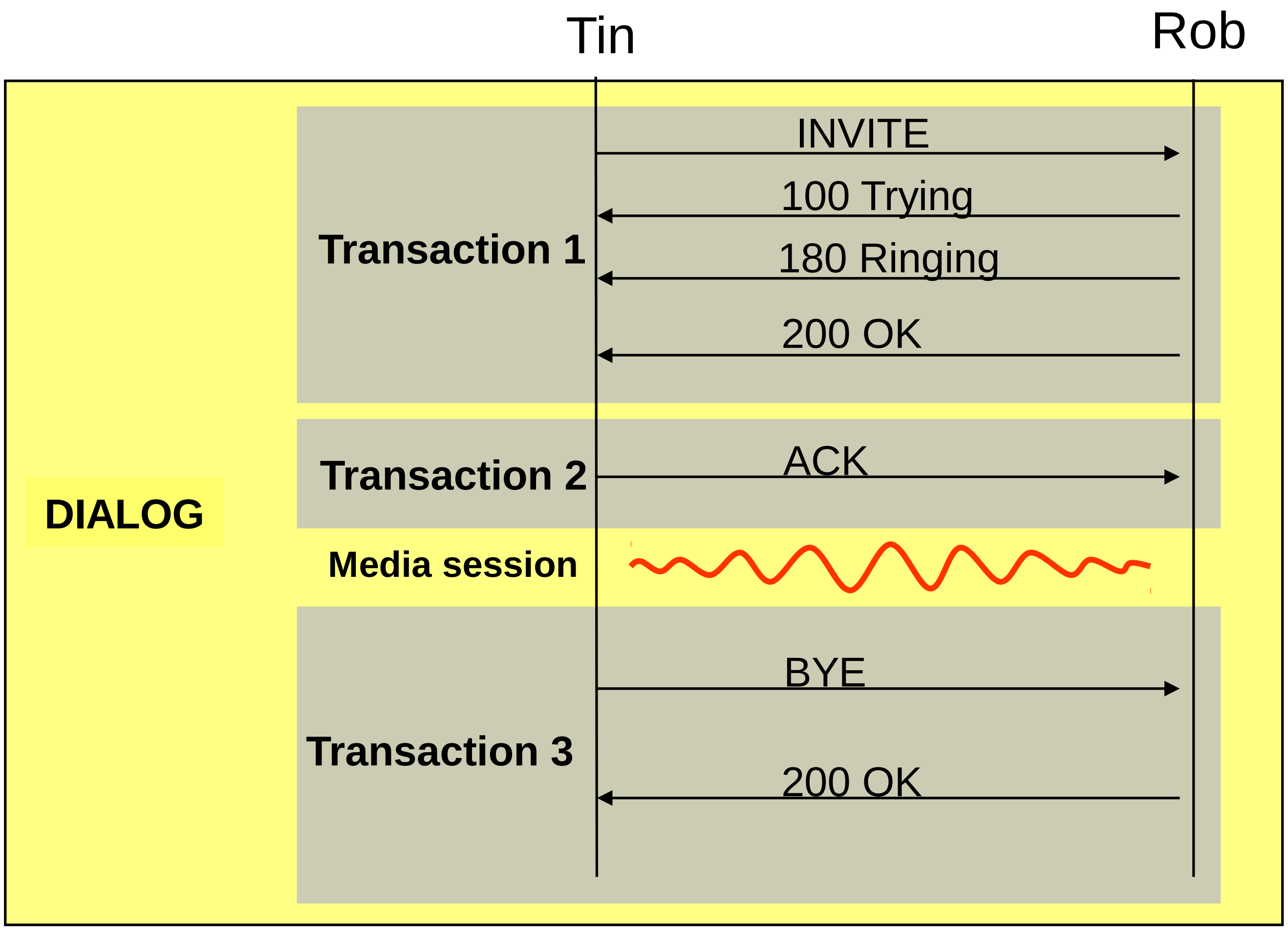
- **Focus on SIP protocol**
  - SIP Basis
  - **Basic SIP dialog dissection**
  - Registrar
  - Proxy
  - Advanced functions
  - SIP and security
  - SIP and NAT/FW

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

Basic Call Flow dissection



Basic Call Flow dissection – INVITE

Tin
10.0.0.2
UDP 5060

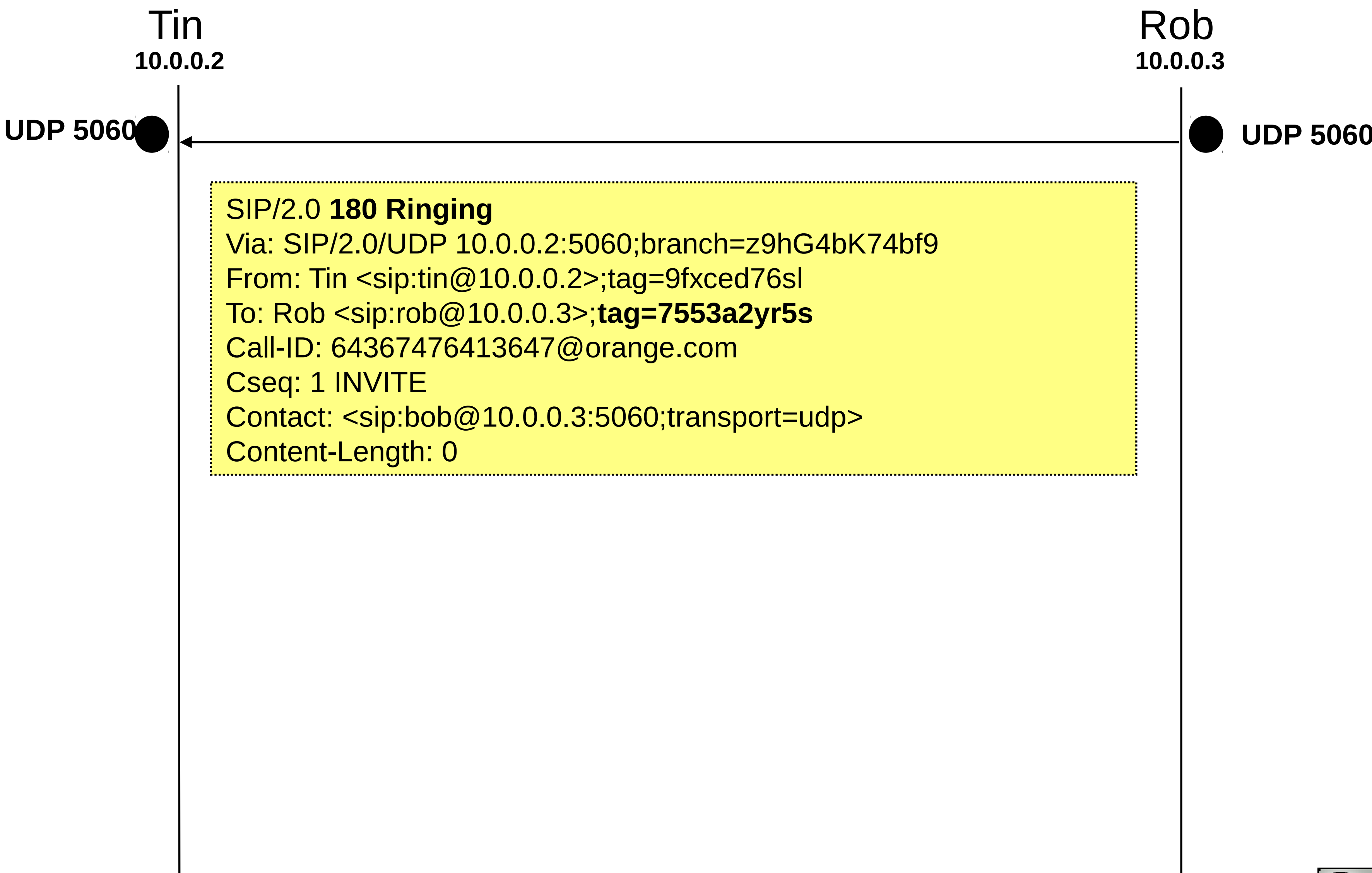
Rob
10.0.0.3
UDP 5060

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



Basic Call Flow dissection – 200 OK

Tin
10.0.0.2

Rob
10.0.0.3

UDP 5060

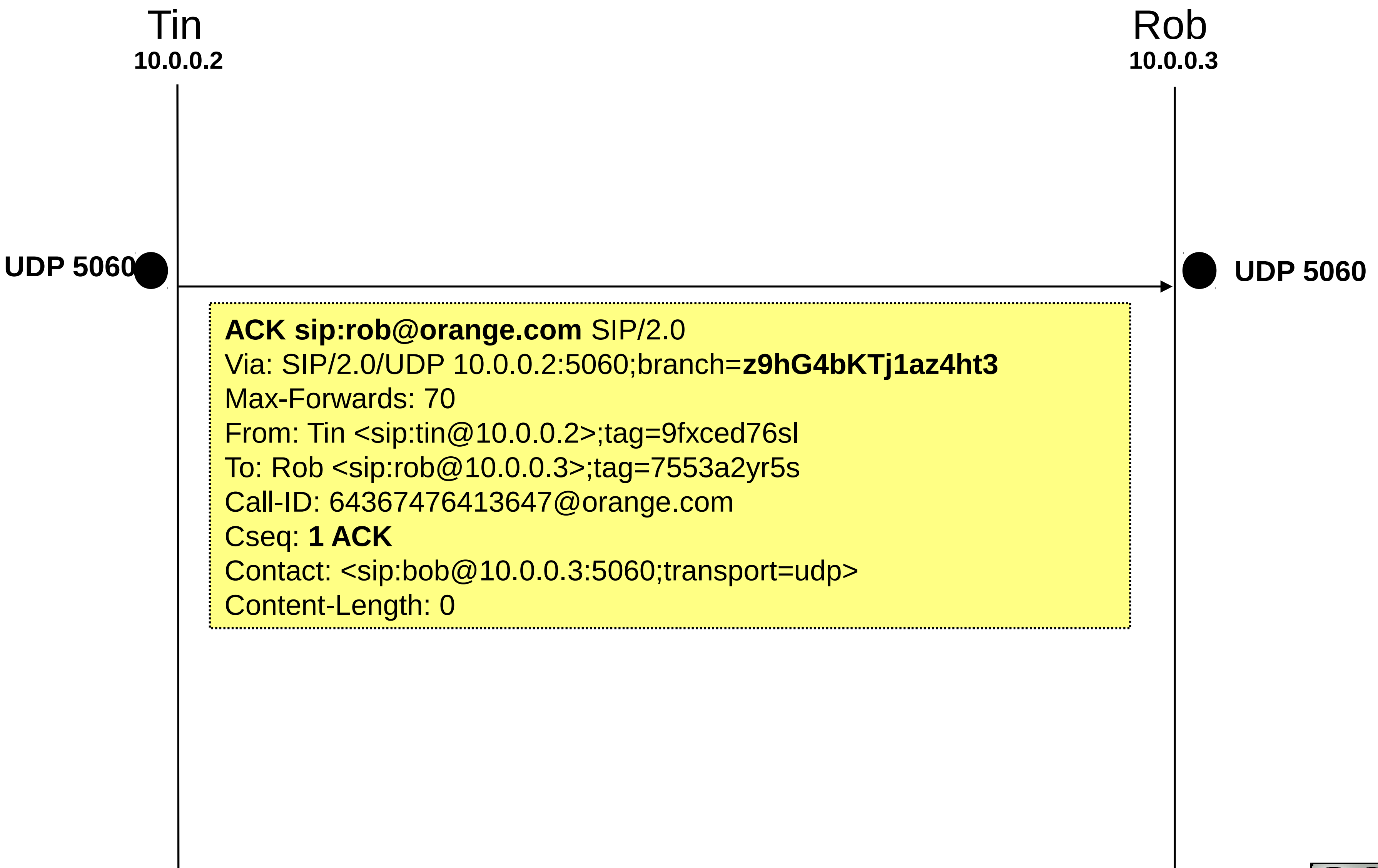
UDP 5060

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**

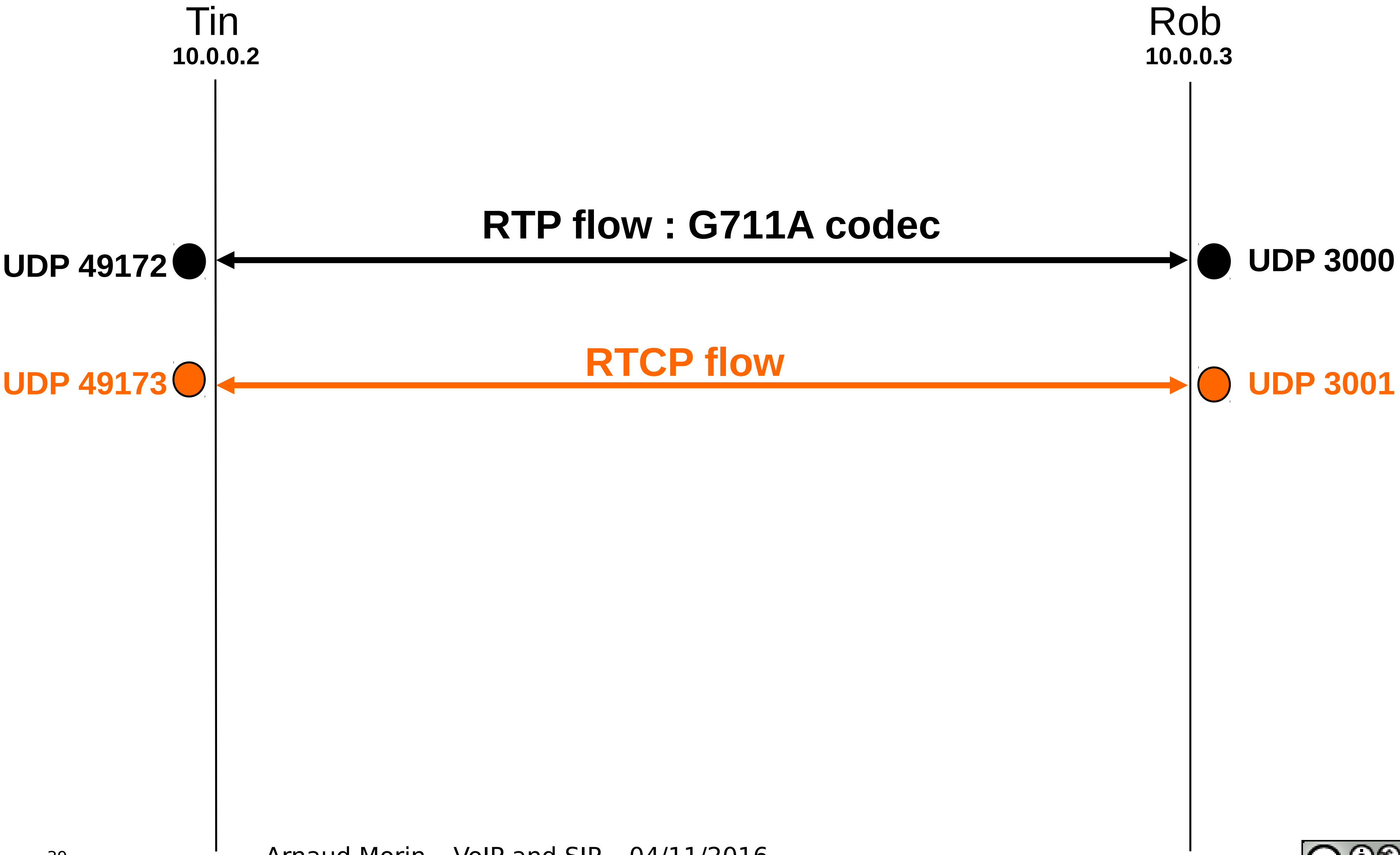
v=0
o=Rob 1692182 1 IN IP4 **10.0.0.3**
s=-
c=IN IP4 **10.0.0.3**
t=0 0
m=audio **3000** RTP/AVP **8**
a=rtpmap:8 PCMA/8000

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

Tin

10.0.0.2

Rob

10.0.0.3

UDP 5060 → **UDP 5060**

SIP/2.0 **200 OK**
Via: SIP/2.0/UDP 10.0.0.3:5060;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

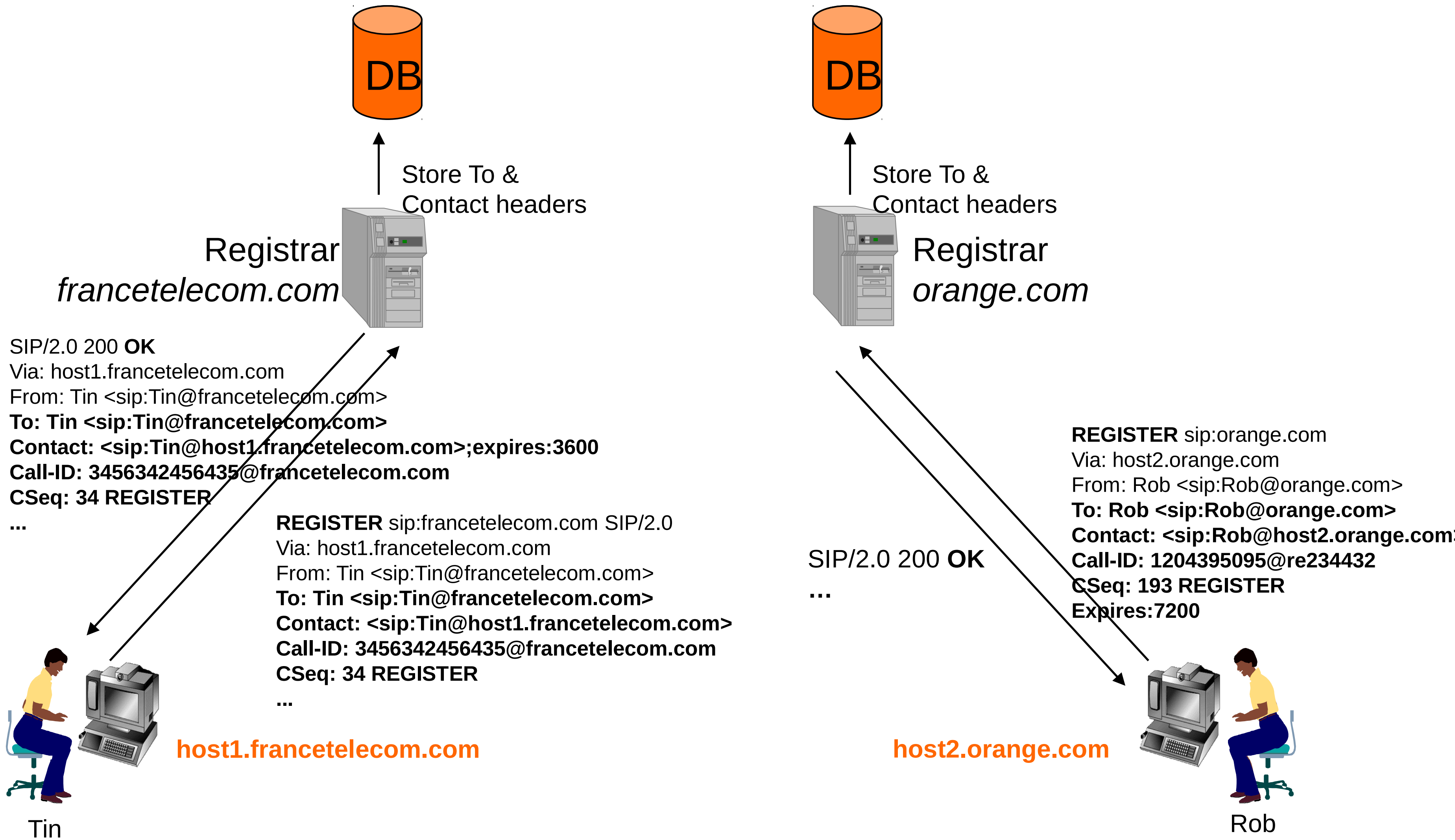
Summary

- **What is VoIP ?**
- **Focus on SIP protocol**
  - SIP Basis
  - Basic SIP dialog dissection
  - **Registrar**
  - Proxy
  - Advanced functions
  - SIP and security
  - SIP and NAT/FW
- **Focus on media component (RTP/RTCP)**

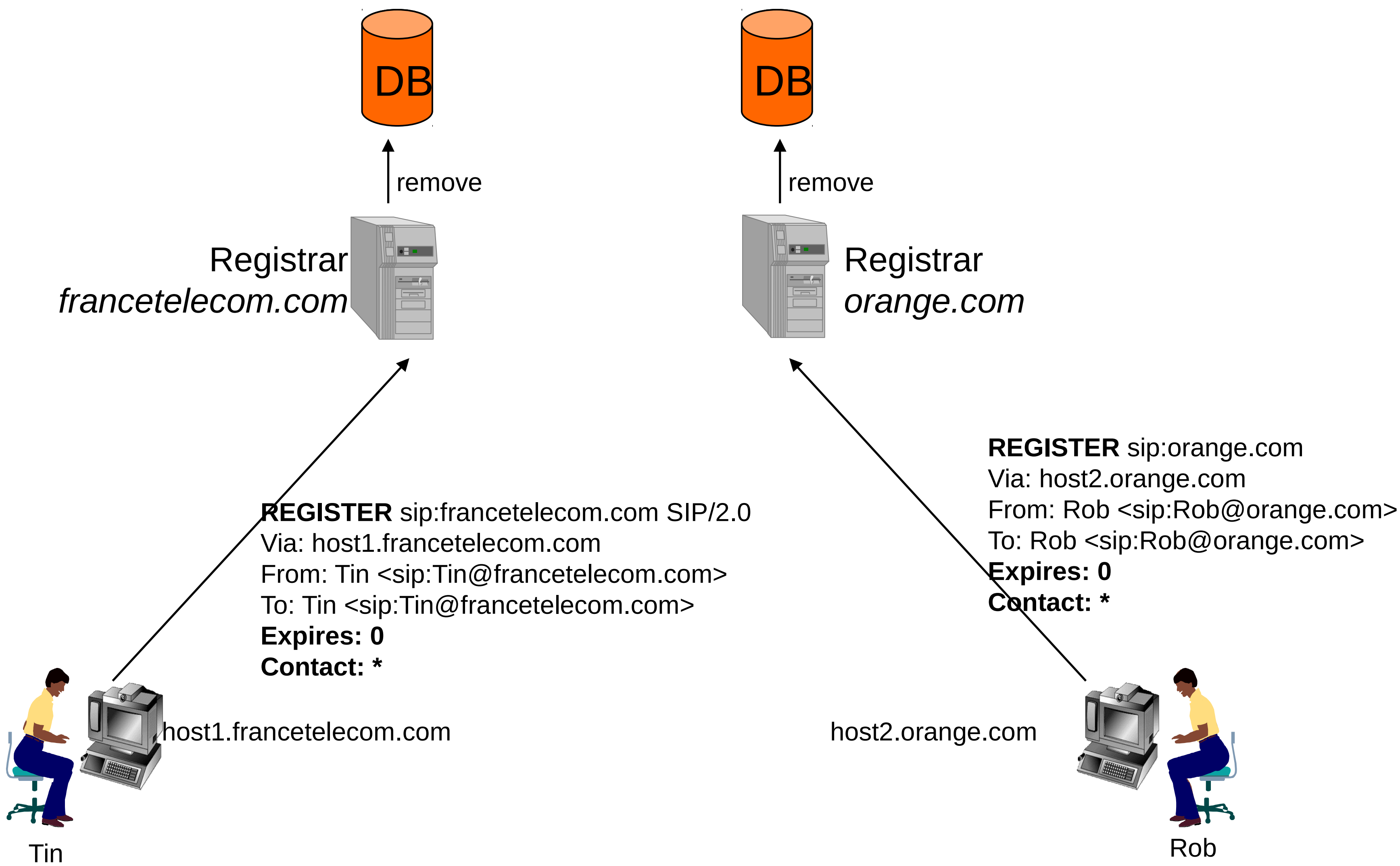
Registration

- 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



Remove registration



Registrar – To remember – 1

- Registrar server is usually used to keep track of User Agent location
- Removing registration consists on setting:
  - Expires: 0
  - Contact: *

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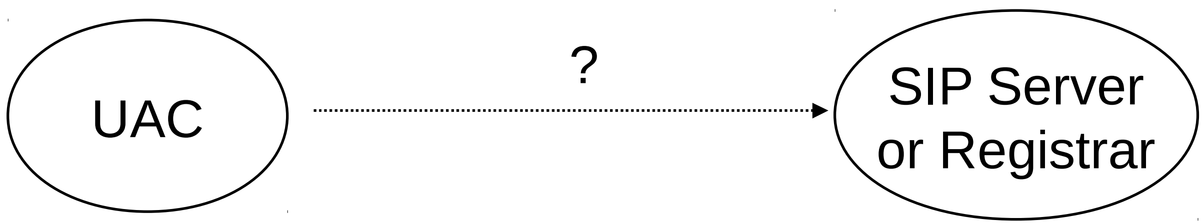
Locating SIP Server from UA Client



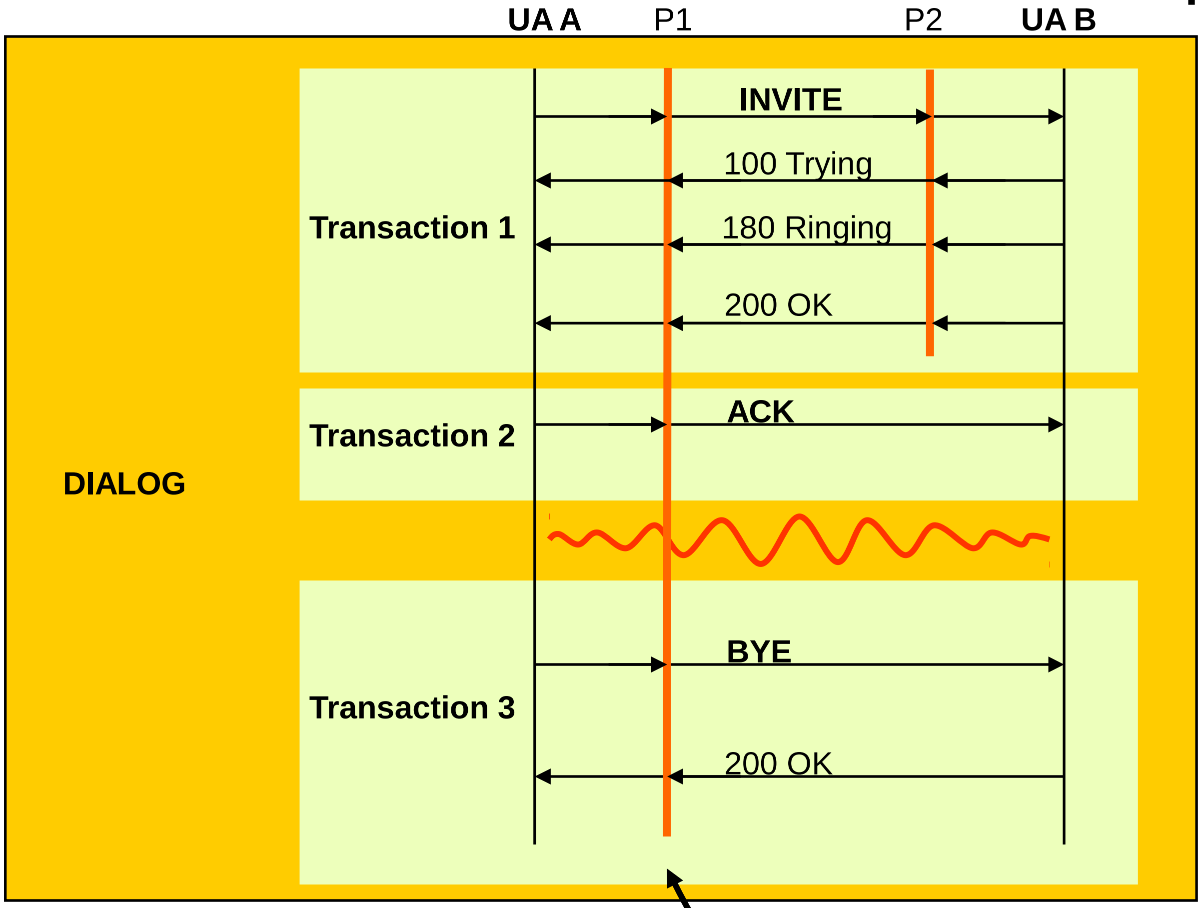
- Client configuration : Outbound proxy
  - Local configuration
  - DHCP option 120
  - DNS SRV lookup

    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]*

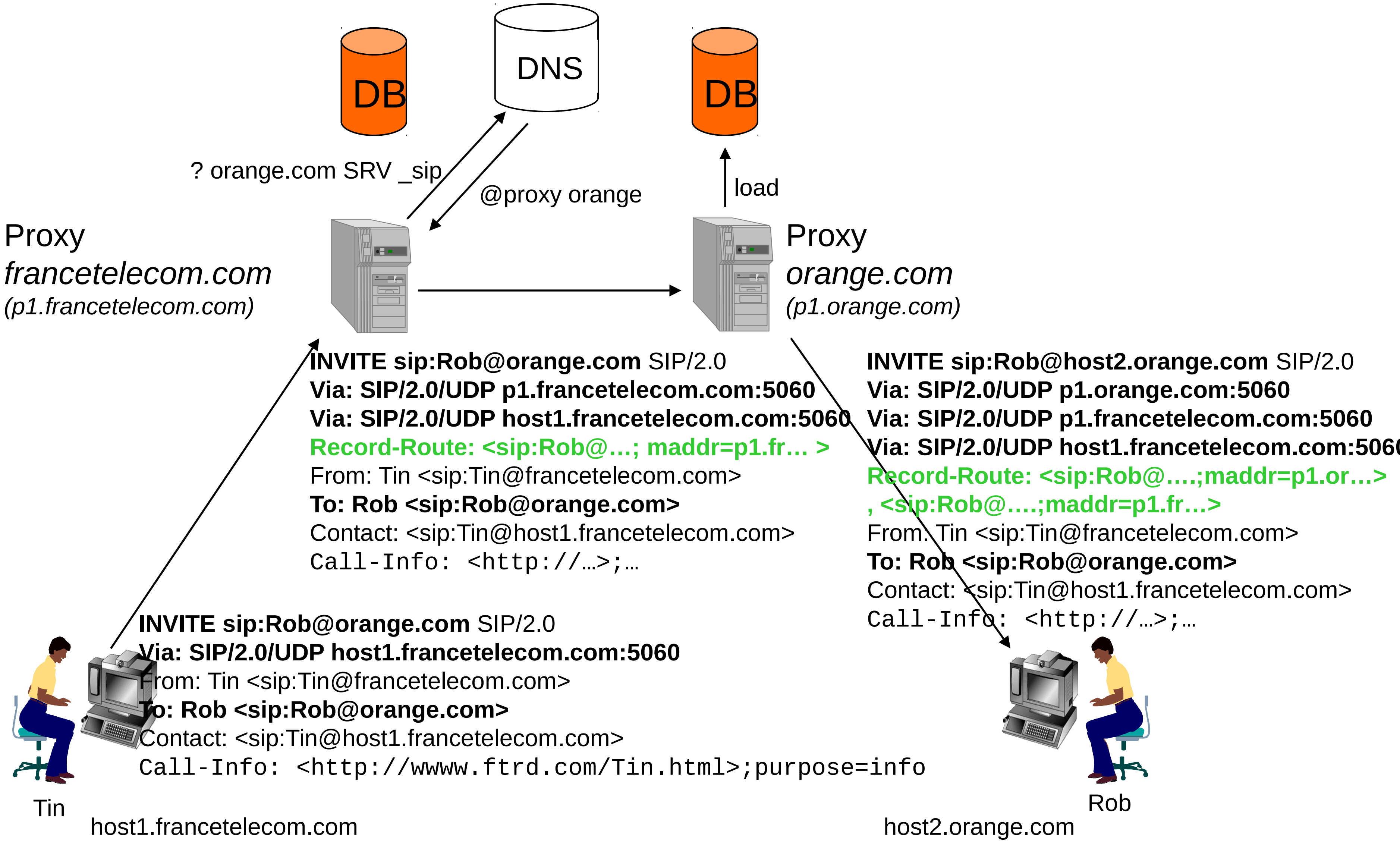


Proxy P1 in the path of all transactions *[Record-Route]*

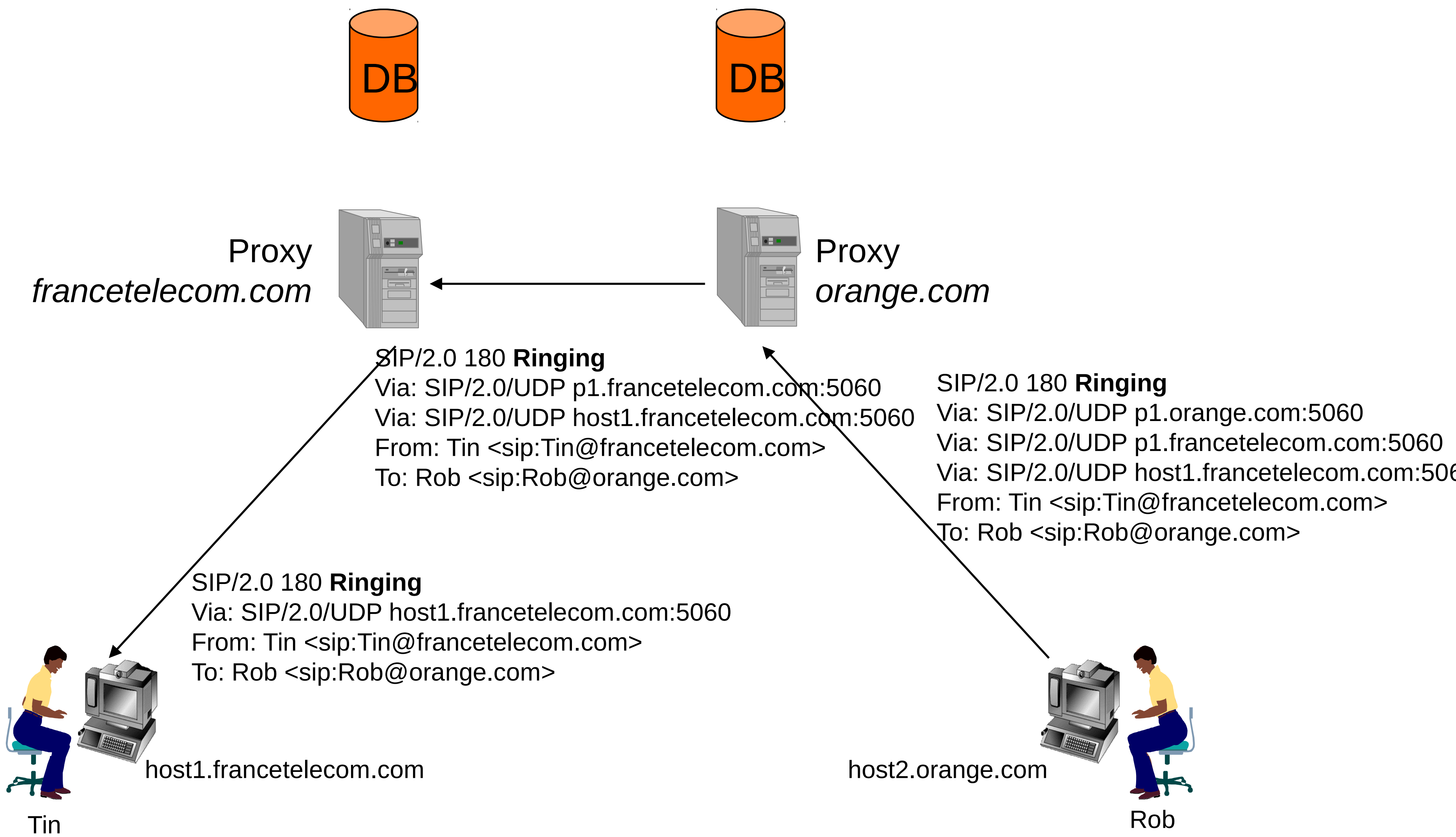
Behavior of SIP node can be :

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

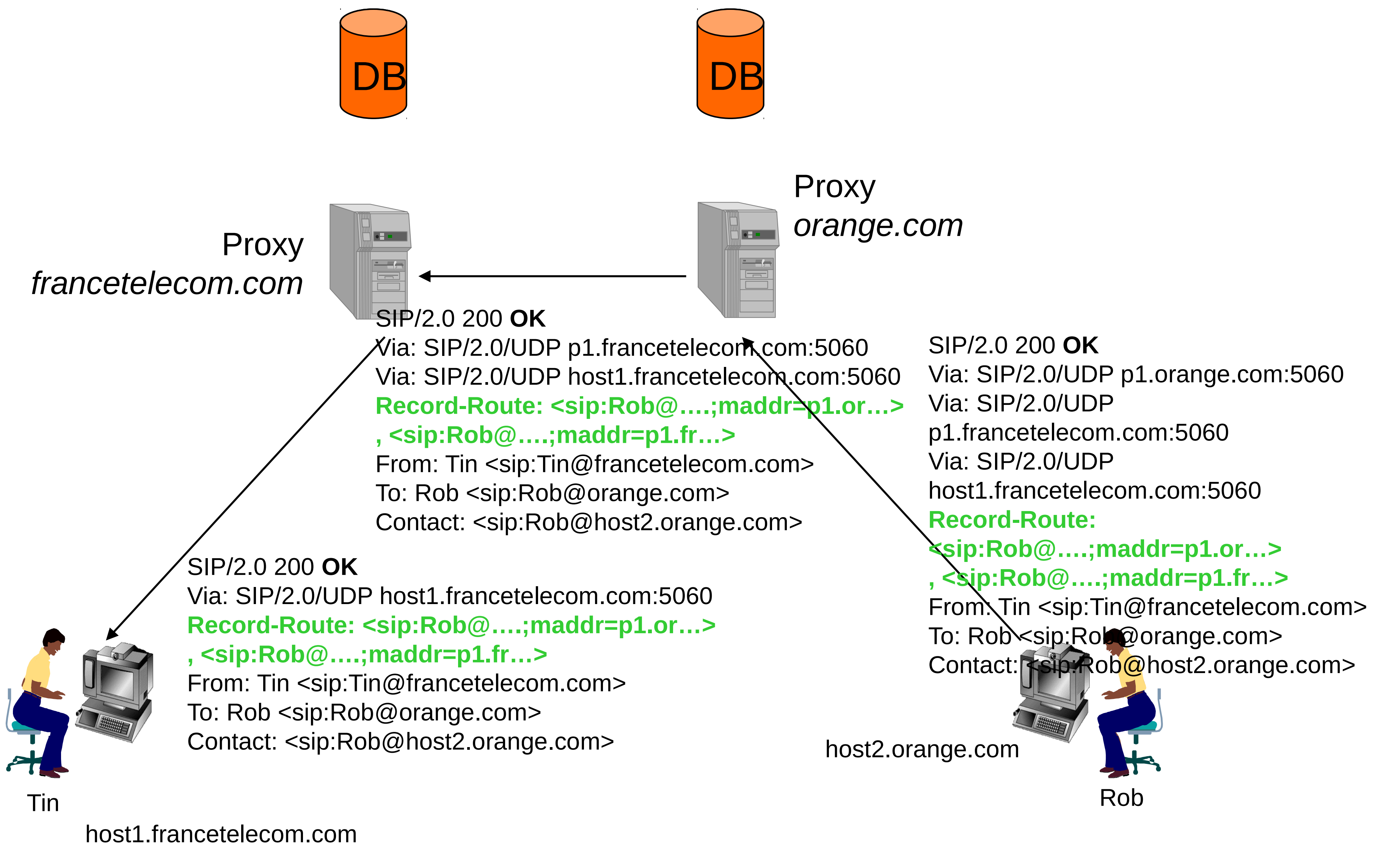
Proxy dialog stateful



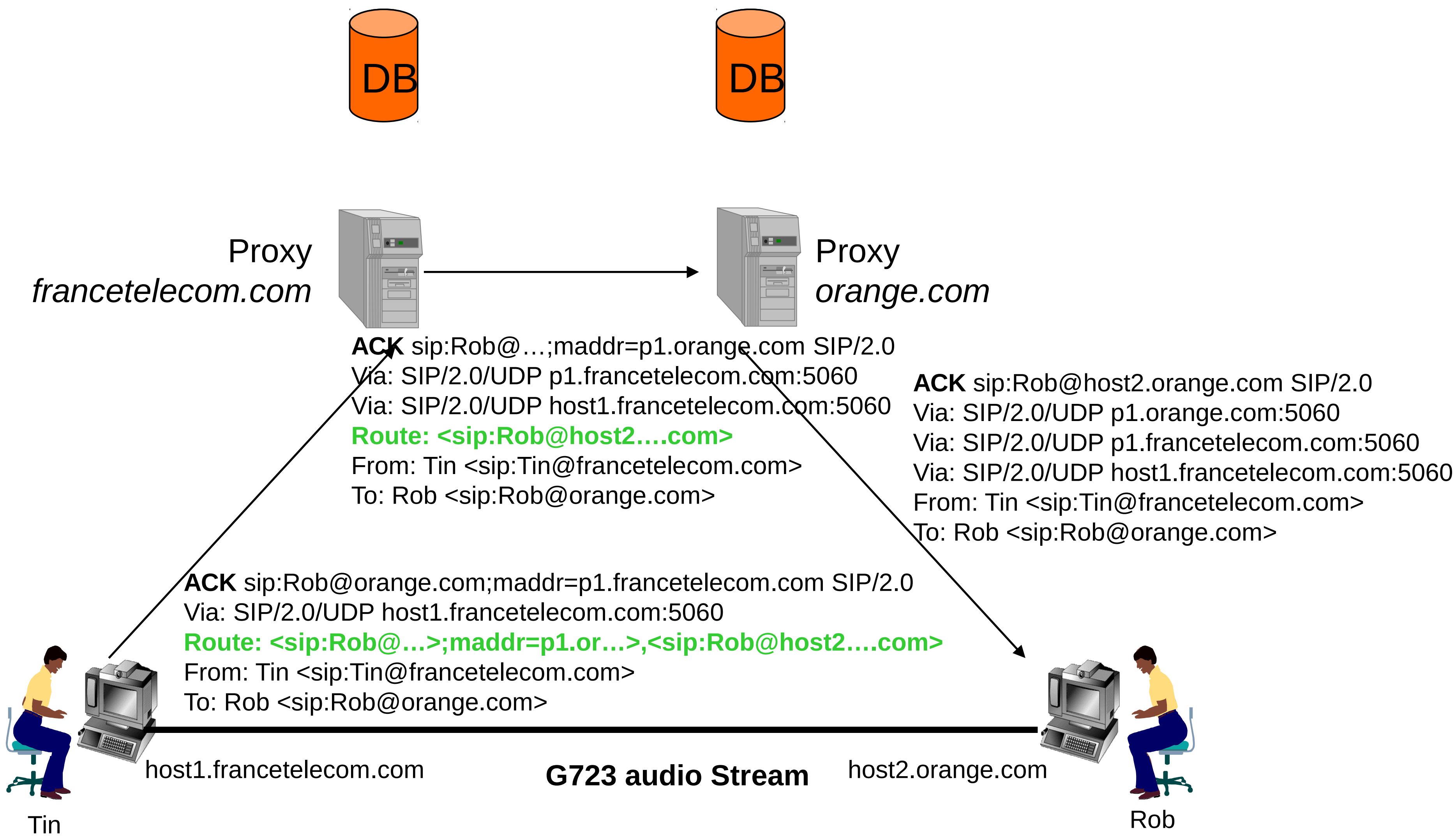
Proxy dialog stateful



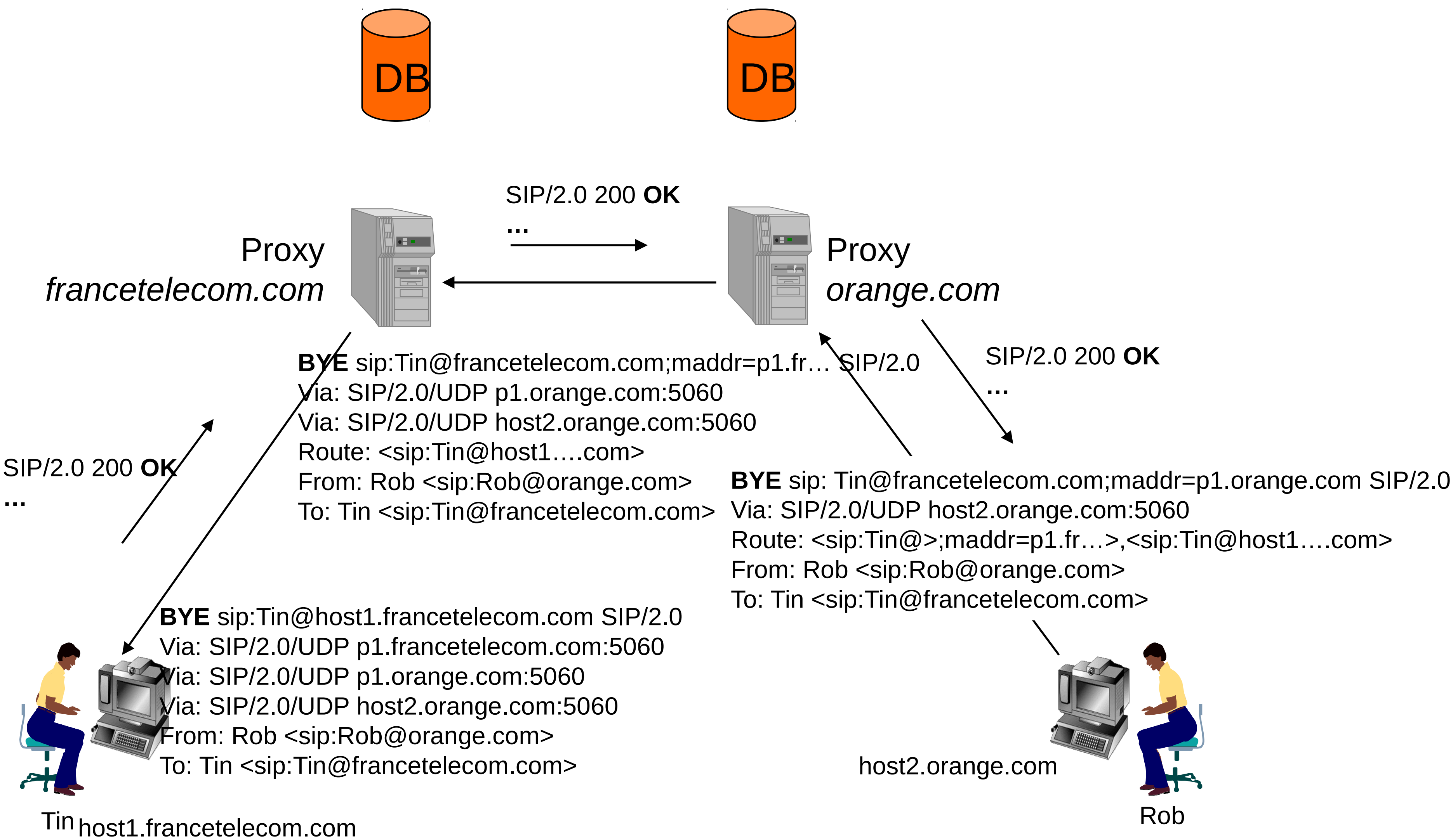
Proxy dialog stateful



Proxy dialog stateful



Proxy dialog stateful



Proxy dialog stateful – 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

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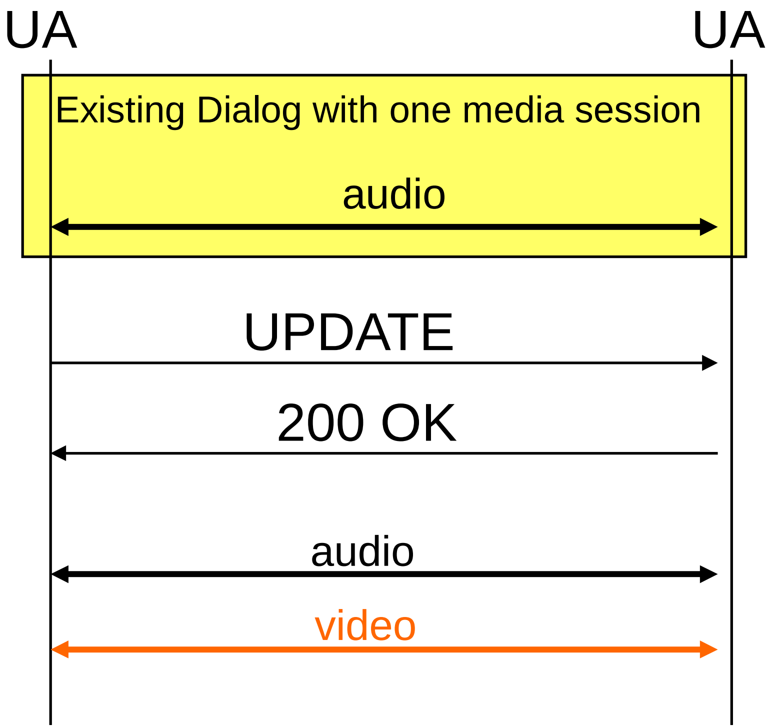
- **Focus on media component (RTP/RTCP)**

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



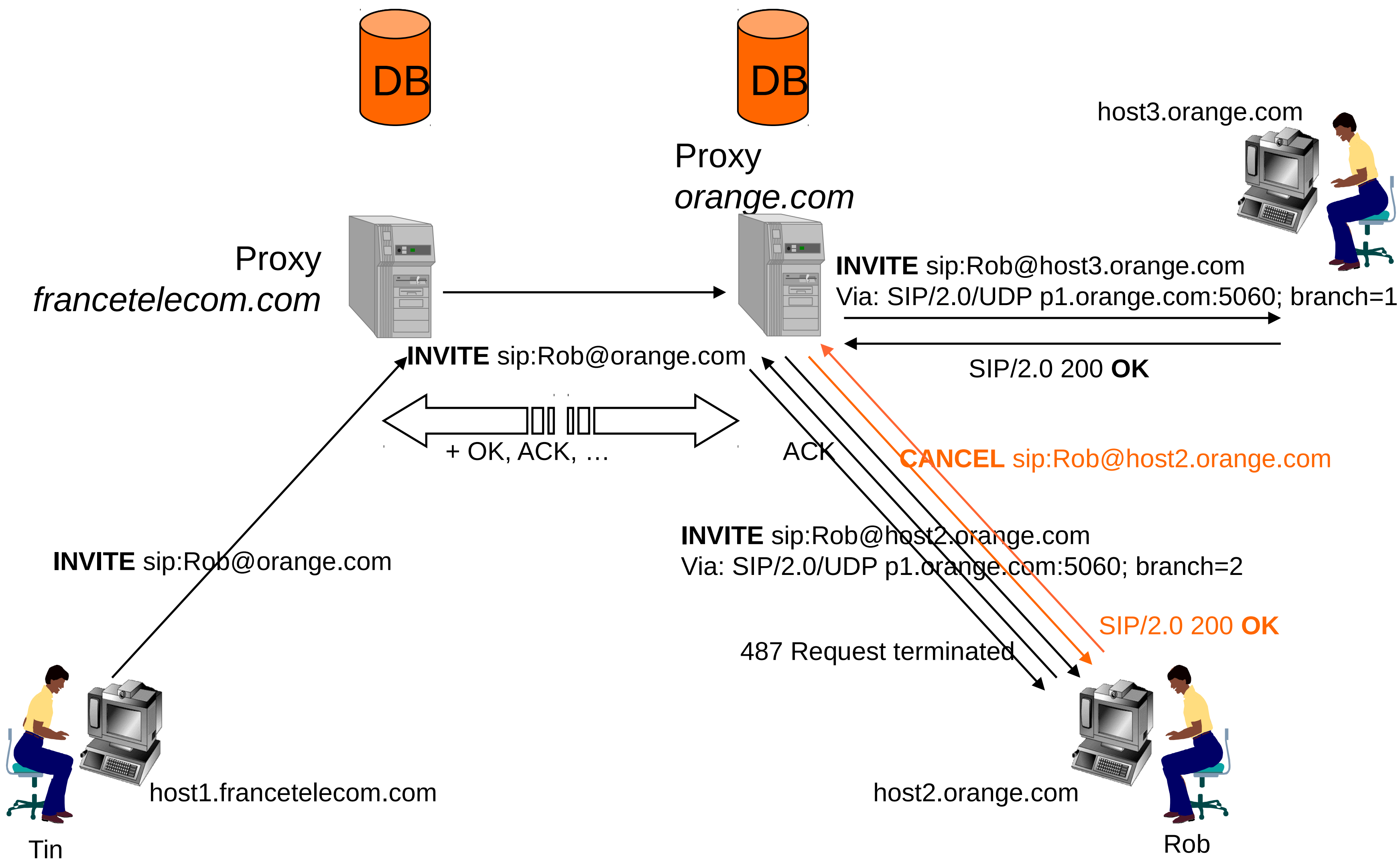
Example : Add video media in the session

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
…

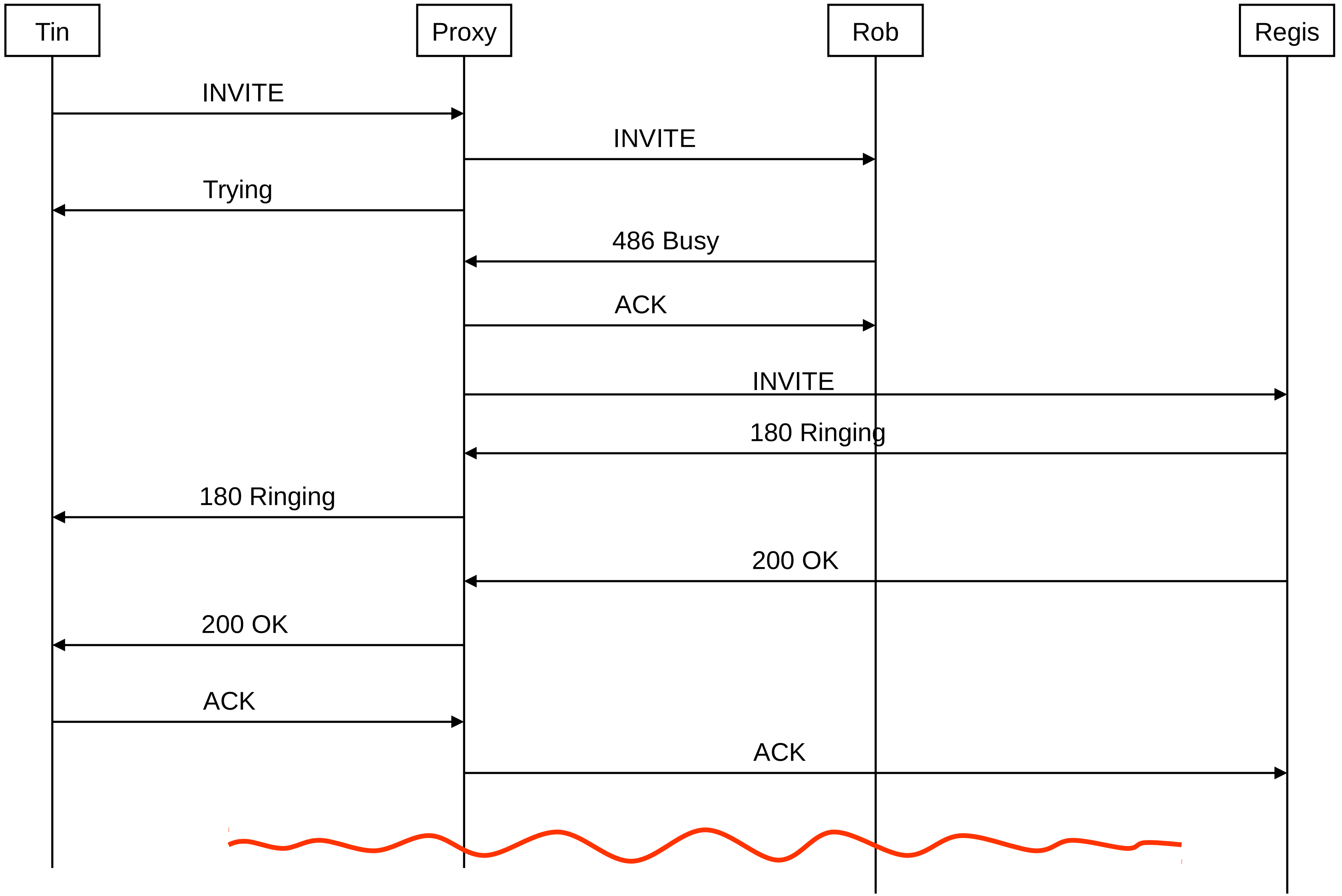
v=0
o=Tin 562413 562414 IN IP4 194.240.47.217
s=phone call
c=IN IP4 194.240.47.217

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

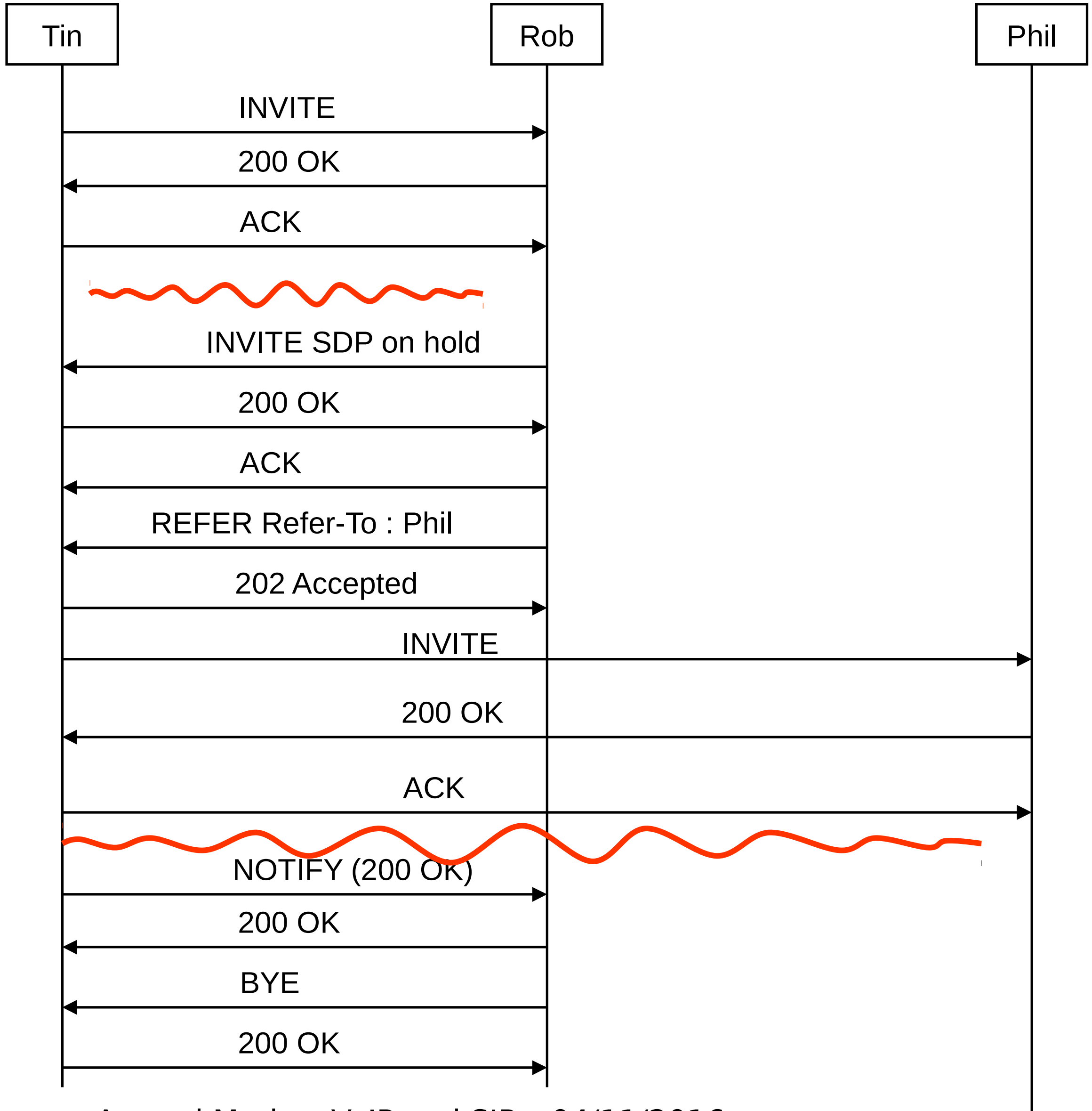
Forking mode (in parallel)



Supplementary Service : Forward



Supplementary Service : Transfer



Advanced functions – To remember

- Subsequent INVITE or UPDATE to update a in-call dialog
- Forking mode create a new transaction (two branches)
- 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**
  - SIP Basis
  - Basic SIP dialog dissection
  - Registrar
  - Proxy
  - Advanced functions
  - **SIP and security**
  - SIP and NAT/FW
- **Focus on media component (RTP/RTCP)**

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

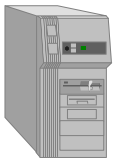
Cseq:1

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

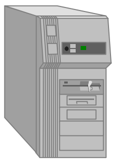
INVITE

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

Summary

- **What is VoIP ?**
- **Focus on SIP protocol**
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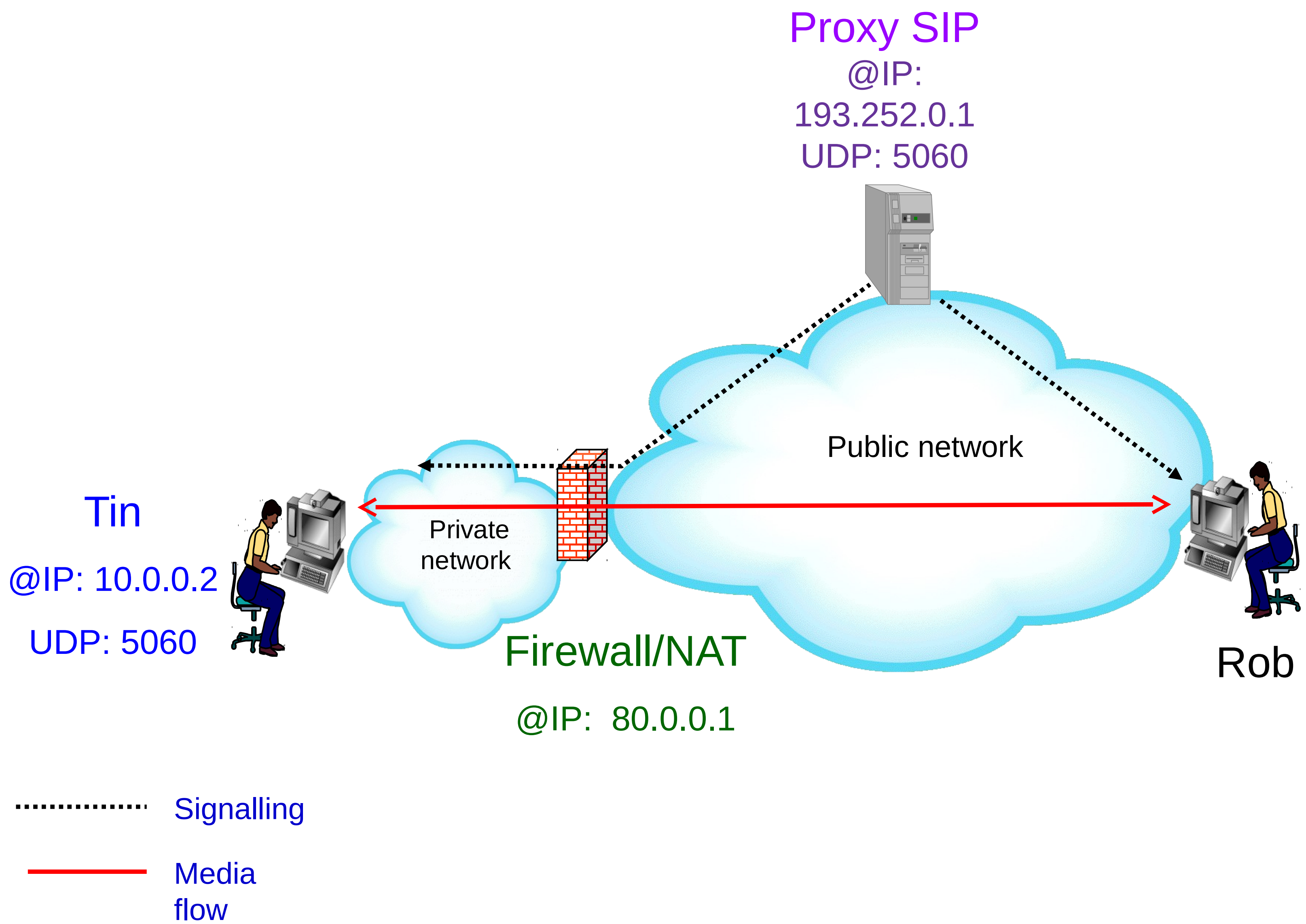
  - Registrar
  - Proxy
  - Advanced functions
  - SIP and security
  - **SIP and NAT/FW**
- **Focus on media component (RTP/RTCP)**

NAPT / Firewall traversal



NAPT / Firewall traversal

NAT

<u>Binding</u>: 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

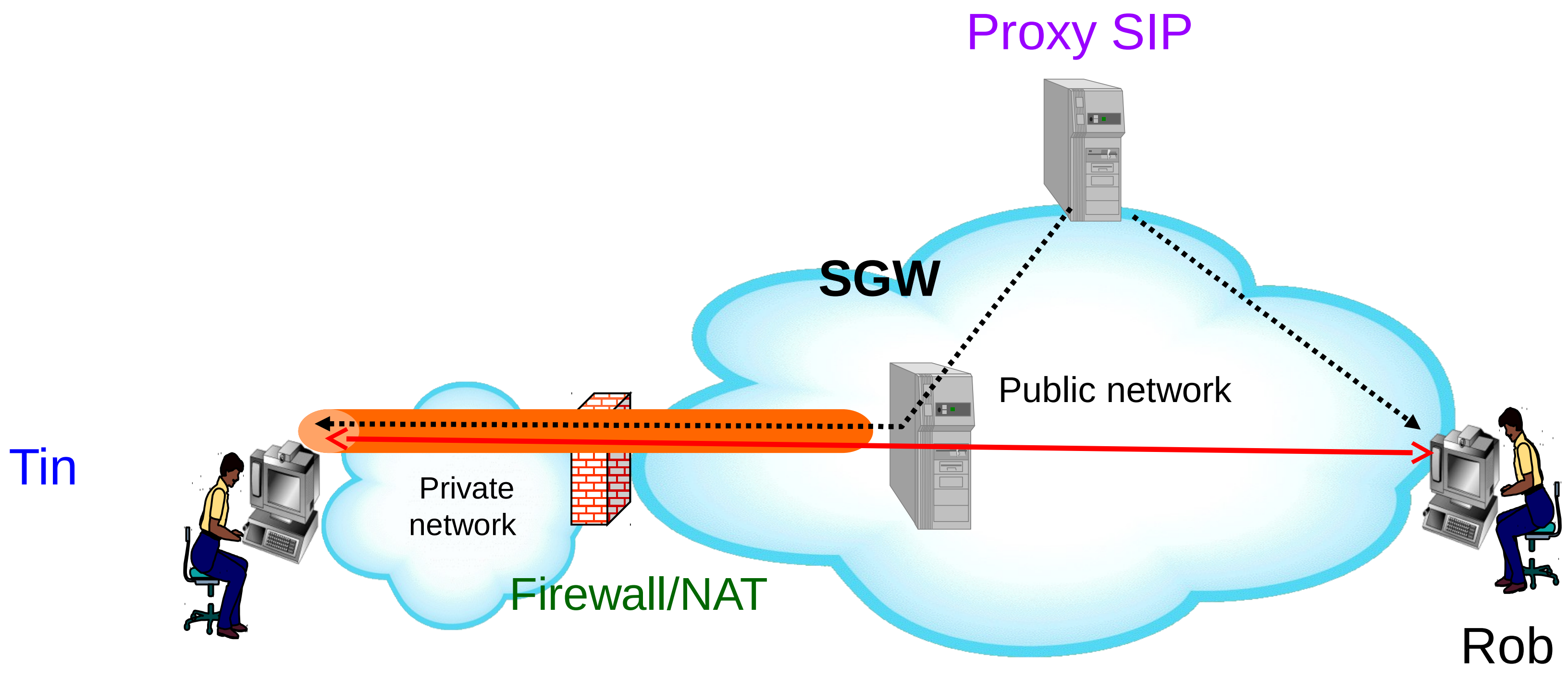
NAPT / Fifrewall traversal

- 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



Signalling

Media
flow

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

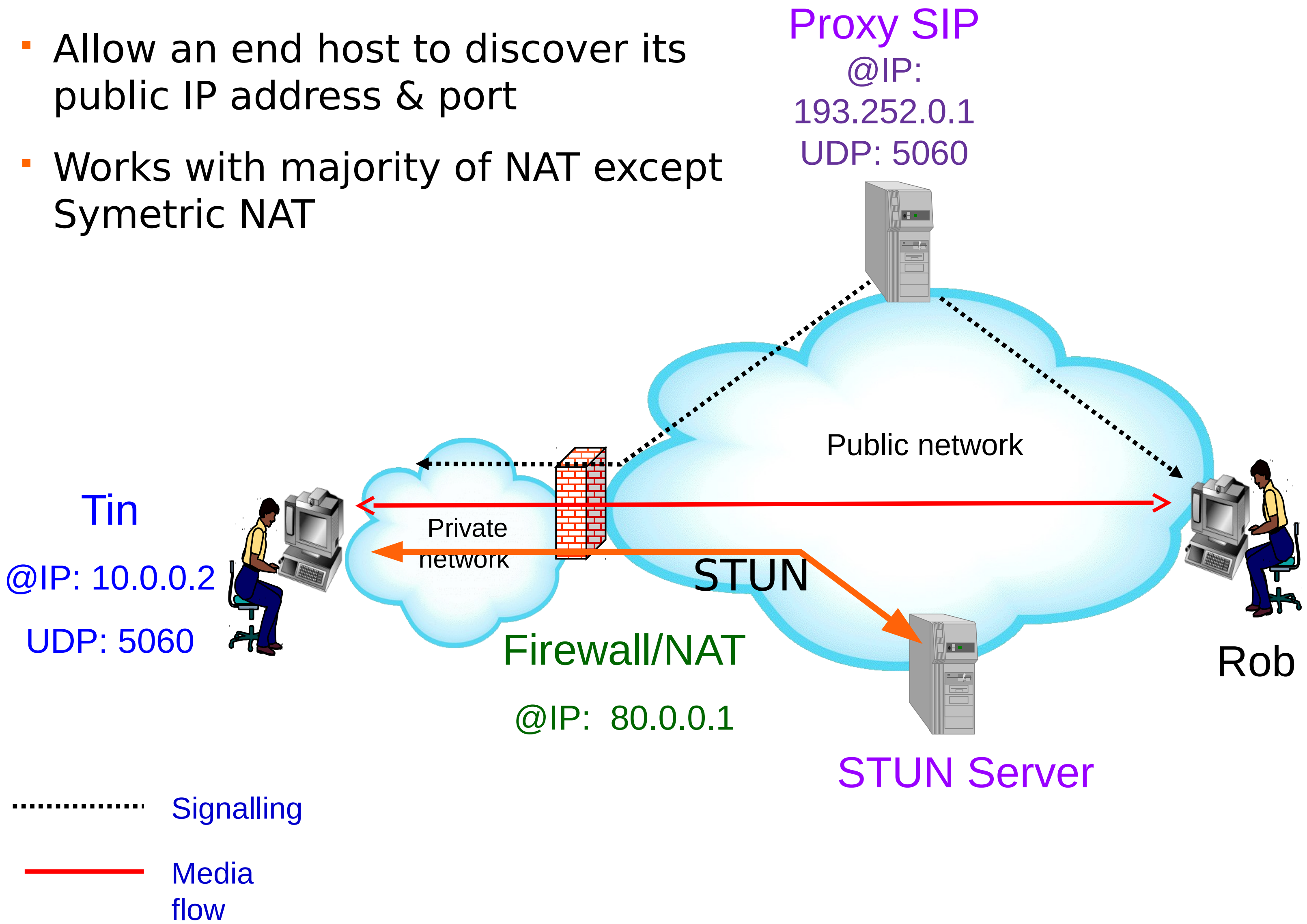
v=0
o=Tin 562413 562413 IN IP4 **80.0.0.1**
s=phone call
c=IN IP4 **80.0.0.1**
m=audio **9400** RTP/AVP 0 4
a=rtpmap:0 PCMU/8000
a=rtpmap:4 G723/8000
m=video 8524 RTP/AVP 31
a=rtpmap:31 H261/90000

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

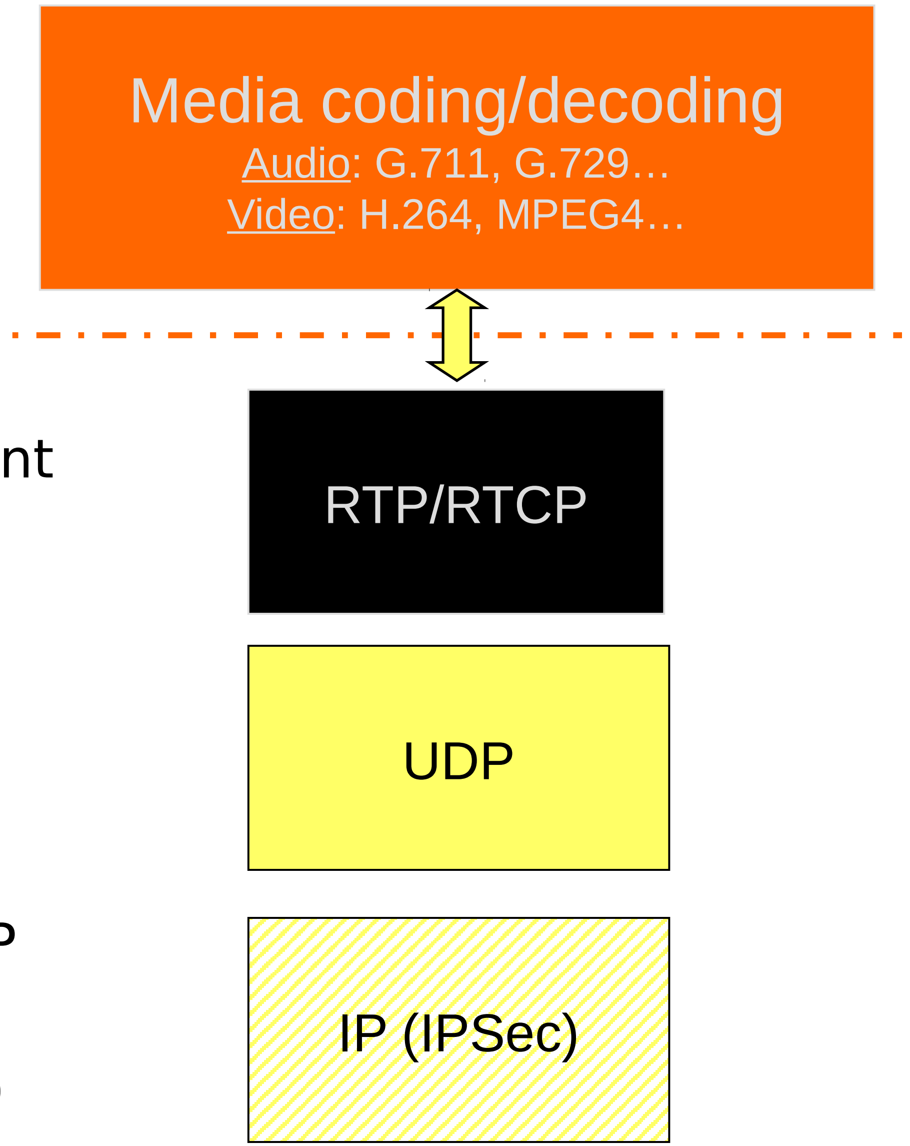


Summary

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

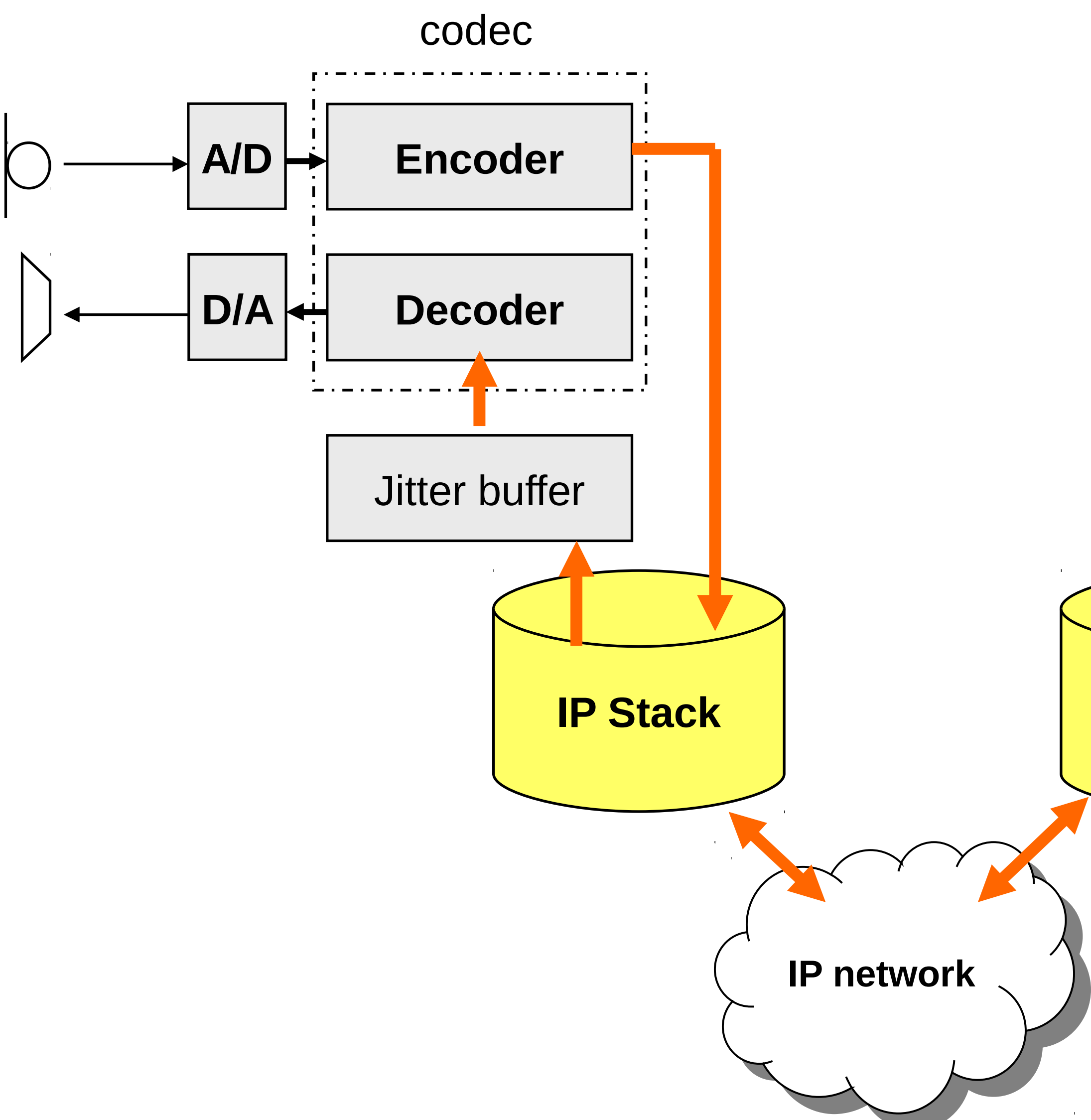
Protocol Layers

Audio or Video



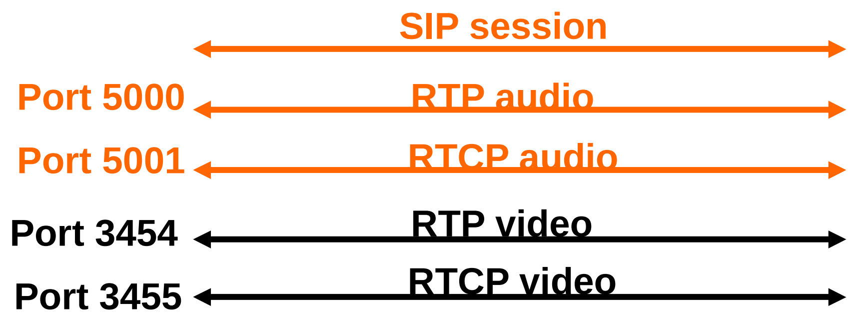
- Audio and Video samples are digitized, compressed and sent in UDP packets
- RTP defines a standardized packet format for delivering audio and video over the Internet
- RTCP provides out-of-band control information for an RTP flow
- Both are defined in RFC 3550

Media Path


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