SIP: Session Initiation Protocol

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SIP
Session Initiation Protocol

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Session Initiation Protocol

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Short History

- **Developments of SIP fall under MMUSIC within IETF**
  - Multiparty Multimedia Session Control (MMUSIC)
- **February 1996**
  - Session Invitation Protocol (SIPv1) Internet Draft
    - Mark Handley & Eve Schooler
    - Purpose was to invite registered users to conference sessions
    - Specified SDP and UDP
  - Simple Conferencing Invitation Protocol (SCIP) Internet Draft
    - Henning Schulzrinne
    - Purpose was to invite users to point to point and multicast sessions
    - Used email identifiers, TCP, but defined its own format for session description
- **December 1996**
  - Session Initiation Protocol (SIPv2) Internet Draft
    - Handley, Schooler & Schulzrinne
    - HTTP based, could use UDP or TCP, and SDP for session description
    - Jonathan Rosenberg became co-author in 1998
- **February 1999**
  - SIP became a proposed standard, published as RFC 2543

Short History

- **March 2001: SIP Working group split:**
  SIP Fundamental specification and its extensions
  SIPPING Applications that use SIP
- **Notification services added later:**
  - **SIMPLE** IETF WG for Instant Messaging and Presence using SIP
- **June 2002 new version published: RFC3261 obsoletes RFC 2543**
- **Today, there are many WGs, including:**
  - mmusic - Multiparty Multimedia Session Control
  - p2psip - Peer-to-Peer Session Initiation Protocol
  - simple - SIP for Instant Messaging and Presence Leveraging Extensions
  - sipclf - SIP Common Log Format
  - sipcore - Session Initiation Protocol Core

See: http://www.ietf.org/html.charters/sip-charter.html
http://www.ietf.org/html.charters/sipping-charter.html
http://www.ietf.org/html.charters/simple-charter.html

Standardisation

- **SIP @ IETF (ietf.org) - several RFC**
  - RFC 3261 : SIP: Session Initiation Protocol
  - RFC 3262: Reliability of Provisional Responses in Session Initiation Protocol (SIP)
  - RFC 3263: Session Initiation Protocol (SIP): Locating SIP Servers
  - RFC 3264: An Offer/Answer Model with Session Description Protocol (SDP)
  - RFC 3265: Session Initiation Protocol (SIP)-Specific Event Notification
  - RFC 3266: Support for IPv6 in Session Description Protocol (SDP)
  - RFC 3428: SIP Message Extension

  - RFC 5411: A Hitchhiker's Guide to the Session Initiation Protocol (SIP)
  - RFC 3665 - BCP 75: Session Initiation Protocol (SIP) Basic Call Flow Examples
  - RFC 5359 - BCP 144: Session Initiation Protocol Service Examples

What is SIP?

- **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 relies on**
  - RTP / RTCP to transport the media
  - SDP for describing multimedia session
  - MEGACO for controlling gateways to the PSTN

The Session Initiation Protocol

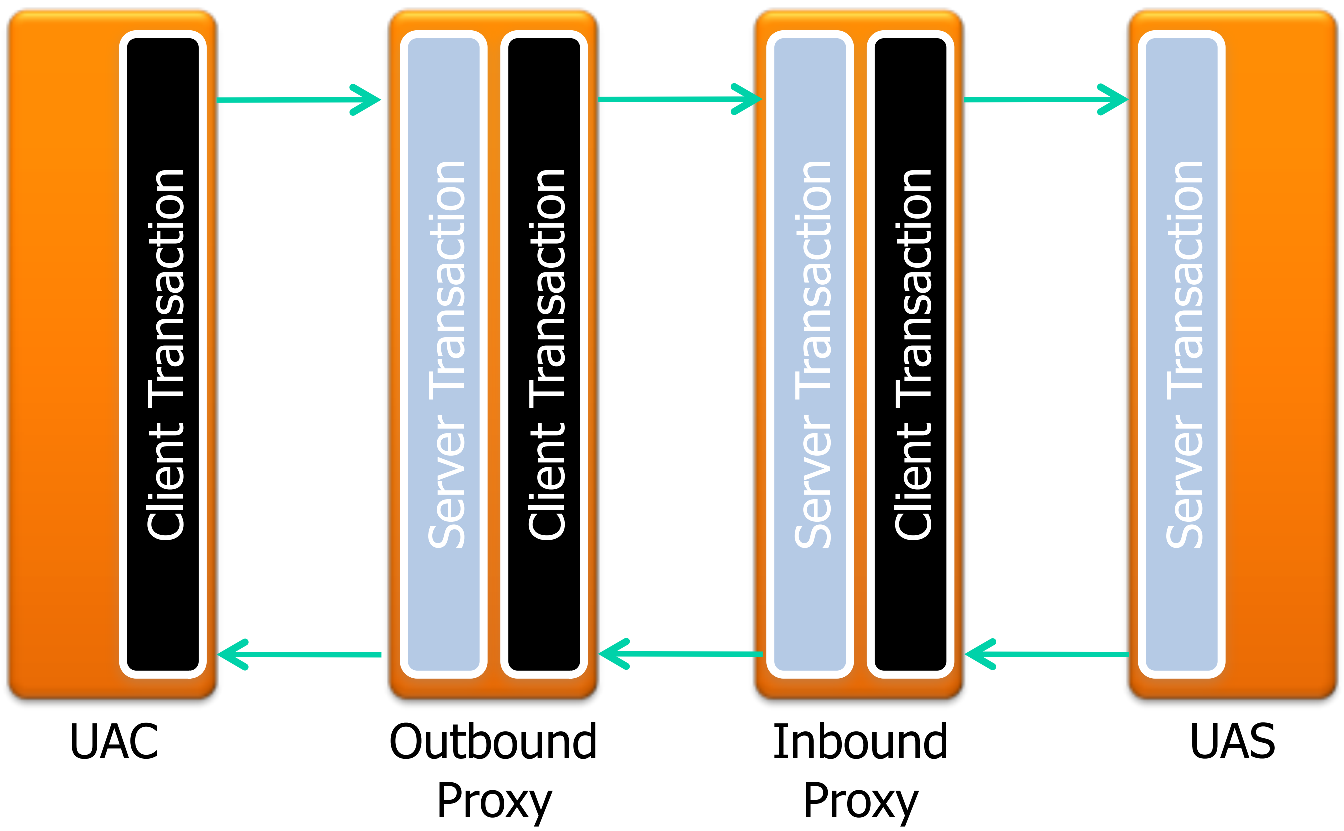
- **The Session Initiation Protocol (SIP) is an application-layer control protocol that can establish, modify, and terminate different kinds of sessions such as Internet telephony calls**
  - Request/response protocol (like HTTP)
  - Uses a <header:value> format (like SMTP)
  - Simple and extensible
  - Designed for mobility (proxy/redirect servers)
  - Authentication
  - Capability negotiation
  - Works on any transport: UDP, TCP, SCTP, ATM
- **SIP is used for signaling:**
  - Instant Messaging sessions
  - Phone calls over the Internet
  - Gaming servers

Key Elements

- **Client /server model**
  - Determined by the initiator of the requests
- **Client / server Exchange**
  - Transaction: request - response
  - Dialog: SIP relation SIP between 2 UA which lasts, indicate the context on how to interpret SIP messages
- **User Agent : endpoint**
  - Commands issued by user (human or gateway) and act as an agent to set up and clear down sessions
- **URI**
  - Identification of the users
  - sip:user@host
- **Proxy**
  - May play the role of the client or the server
  - Rendez-vous point

SIP User Agent

- **User Agent**
  - Commands issued by user (human or gateway) and act as an agent to set up and clear down sessions
  - Act as a UA Client
    - Generate requests
  - Act as a UA server
    - Respond to request



SIP servers

- **Servers - Applications that accept SIP request and respond to them.**
  - **Proxy server** - receive request from UA or another proxy and act on behalf of the UA in forwarding or responding to the request
    - Help in routing SIP messages
    - Can be used to enforce policies
    - Transaction stateless / stateful
    - Call stateless / stateful
    - Preserve the end-to-end transparency
  - Forking proxy: routes call requests
    - Duplicates (“forks”) requests
    - Forward only one final answer back to the UAC
  - B2B UA / ALG: Application Layer Gateway
    - Reformulate requests
    - (See next slide)
- **Redirect server**
  - respond to but do not forward request
  - Return new locations for servers
- **Registration server** - aka Registrar, bind a SIP URI to an address of Record (IP address)

Application Layer Gateway

- A back-to-back user agent (B2BUA) is a logical entity that receives a request and processes it as a **user agent server** (UAS). In order to determine how the request should be answered, it acts as a **user agent client** (UAC) and generates requests. Unlike a proxy server, it **maintains dialog state** and must participate in all requests sent on the dialogs it has established. Since it is a concatenation of a UAC and UAS, no explicit definitions are needed for its behavior.

- **Can be used as anonymizer**
- **(Break the end-to-end nature of SIP, constitutes a single point of failure)**
- **Difference between a proxy server and an ALG**
  - A proxy server does not issue Requests; it only responds to requests from a user agent (except for CANCEL)
  - A proxy server has no media capabilities
  - A proxy server does not parse message bodies; it relies exclusively on header fields

SIP URIs

User identification

- **Why IP addresses are not enough?**
  - IP addresses are allocated dynamically
  - An IP address identifies (locates) a device in the topology
  - Communication are user-to-user not device-to-device
- **SIP uses URI (Unique Ressource Identifier)**
  - Email-like names for addresses
  - Can handle phone number, transport parameters, etc
  - An URI is a name that can be translated into an IP address using proxy server and DNS lookups
- **2 URI categories**
  - For user: known as **Address of Record**
  - For a device or end-point: temporarily allocated to a user, indicated in the contact field

Quelques types d'URI

- SIP URI with username:
  - `sip:andrea@sip-communicator.org`
  - `sip:andrea@starsip.tilab.com`
  - `sip:andrea@163.162.3.19`
- SIP URI without a username:
  - `sip:example.com`
  - `sip:x.example.com`
  - `sip:163.162.3.19`
- SIP URI with parameters:
  - `sip:abc@example.com;transport=tcp;user=phone`
- IPv6 SIP URI:
  - `sip:andrea@[fe80::5445:5245:444f]:5560`

Messages

Message Structure: First Line

- The first line, determines the semantical type of the message:
  - Request
  - Response
- Request line contains:
  - method: determines the type of the request
  - SIP URI: determines the destination of the request
  - SIP protocol version

```
<METHOD> <Request-URI> SIP/2.0
```

- Response line contains
  - SIP protocol version
  - status-code: digital response code
  - reason phrase

```
SIP/2.0 <Status-Code> <Reason-Phrase>
```

*1xx informational responses are not retransmitted if lost **
2xx success responses are delivered to the end with reliability
3xx - 6xx non-successful responses delivered hop-by-hop

The Session Initiation Protocol

- Methods are the "verbs" of the protocol
- Original six methods in version 2.0 of SIP
  - INVITE
  - REGISTER
  - BYE
  - ACK
  - CANCEL
  - OPTIONS

Request Format

Request line
Several Headers
Empty Line
Message Body

- Both Requests and Responses can carry SIP bodies
  - usually SDP, but could be a JPEG or JAVA script
- SIP Responses carry a status code and a reason phrase - human readable

Response Format

Status line
Several Headers
Empty Line
Message Body

SIP Response Codes

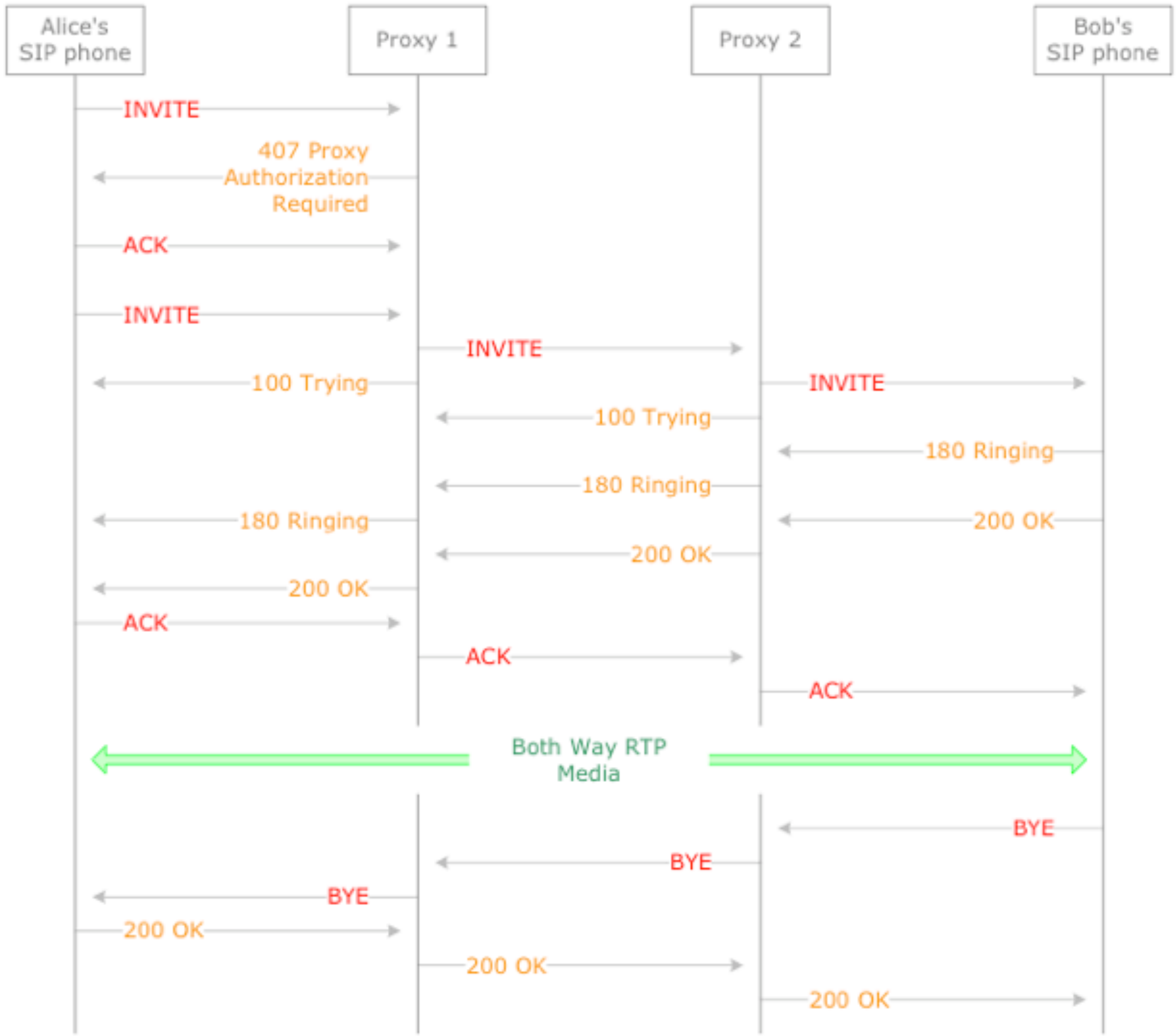
CODE RANGE	RESPONSE CLASS		EXAMPLES
1XX	Informational	Provisional	-Queued, Ringing, Being Forwarded
2XX	Success	Final	-OK, Accepted
3XX	Redirection	Final	-Moved Temporarily, Moved Permanently
4XX	Client error	Final	-Payment Required, Method Not Allowed
5XX	Server error	Final	-Not Implemented, Service Unavailable
6XX	Global failure	Final	-Busy Everywhere, Decline

TELECOM Bretagne

SIP Header

```
Request message :

  <METHOD> <Request-URI> SIP/2.0 CRLF
  <Header1>: <Value1> CRLF
  <Header2>: <Value2> CRLF
  <HeaderN>: <ValueN> CRLF
   CRLF
  <Message Body>

Response message:
  SIP/2.0 <Status-Code> <Reason-Phrase> CRLF
  <Header1>: <Value1> CRLF
  <Header2>: <Value2> CRLF
  <HeaderN>: <ValueN> CRLF
  CRLF
  <Message Body>
```

SIP call


Alice's SIP phone
Proxy 1
Proxy 2
Bob's SIP phone
INVITE
407 Proxy Authorization Required
ACK
INVITE
INVITE
100 Trying
INVITE
100 Trying
180 Ringing
180 Ringing
180 Ringing
200 OK
200 OK
200 OK
ACK
ACK
ACK
Both Way RTP Media
BYE
BYE
BYE
200 OK
200 OK
200 OK

Requests

- **INVITE:** Start / modify sessions
- **ACK :** Acknowledge the reception of a final response to an INVITE
- **CANCEL:** cancel a pending INVITE
- **UPDATE**: Update the parameters of a pending session
- **BYE:** ends a session
- **OPTIONS:** Request the supported features
- **REGISTER:** Attach an IP address to a SIP URI
- **REFER:** request a UA to access a URI or URL
- **SUBSCRIBE:** establish a subscription to receive a notification about an event
- **NOTIFY:** Convey information about the occurence of a particular event
- **PRACK:** acknowledge receipt of reliably transported provisional response
- **MESSAGE:** Transport instant message using SIP
- **INFO:** Send call signalling information to another user agent with which it has an established media session

Responses

- **Responses**
  - 1xx : information
  - 200 : succès
  - 3xx : redirection
  - 4xx : erreur client
  - 5xx : erreur serveur
  - 6xx : échec

Error Codes

Note:
Many codes are same as HTTP
SIP specific codes start x80

Informational

100	Trying
180	Ringing
181	Call Is Being Forwarded
182	Queued
183	Session Progress

Success

200	OK
202	Accepted

Redirection

300	Multiple Choices
301	Moved Permanently
302	Moved Temporarily
303	See Other
305	Use Proxy
380	Alternative Service

Client error

400	Bad Request
401	Unauthorized
402	Payment Required
403	Forbidden
404	Not Found
405	Method Not Allowed
406	Not Acceptable
407	Proxy Authentication Required
408	Request Timeout
409	Conflict
410	Gone
411	Length Required
413	Request Entity Too Large
414	Request-URI Too Large
415	Unsupported Media Type
420	Bad Extension
480	Temporarily not available
481	Call Leg/Transaction Does Not Exist
482	Loop Detected
483	Too Many Hops
484	Address Incomplete
485	Ambiguous
486	Busy Here
487	Request Cancelled
488	Not Acceptable Here

Server error

500	Internal Server Error
501	Not Implemented
502	Bad Gateway
503	Service Unavailable
504	Gateway Time-out
505	SIP Version not supported

Global failure

600	Busy Everywhere
603	Decline
604	Does not exist anywhere
606	Not Acceptable

SIP Headers

Examples of Headers Used in Requests and Responses

HEADER	FUNCTION
Call-ID	-Used to uniquely identify a call between two user agents
Contact	-Used to convey URL of original resource requested or request originator
CSeq	-Command Sequence identifies out of sequence requests & retransmissions
From	-Identifies originator of request
To	-Indicates recipient of request
Subject	-Optional header indicating subject of media session
Content-Length	-Number of octets in the message body
Content-Type	-Indicates Internet media type. If not present application/SDP is assumed
User Agent	-Provides additional information about the user agent e.g. manufacturer
Server	-Provides additional information about the User Agent Server
Via	-Records the route taken by a request and used to route response
Record-Route	-Used to force all requests between UAs to be routed through a Proxy
Route	-Forces routing through a path extracted from a Record-Route header
Max-forwards	-limit the number of hops a request can make on the way to its destination (70)
Authorization	-Carries credentials of user agent to a server
Encryption	-Used to specify the portion of a SIP message that has been encrypted
Hide	-Requests next hop proxy to encrypt the Via headers
Priority	-Allow the user agent to set the priority of a request: e.g. urgent, emergency
Supported	-List one more options implemented in a user agent or server
Unsupported	-Indicates features that are not supported by the server

Dialog identification: call-id, local tag (after the from), remote tag (after the to)

Transaction identification: branch parameter in the via header

Minimum required fields

Additional required field for an INVITE

SIP Request

```
INVITE sip:barbara@b.com SIP/2.0
Via: SIP/2.0/UDP 10.43.122.3;branch=1
From: sip:alice@a.com;tag=4ad340f
To: sip:barbara@b.com
Contact: <sip:alice@10.43.122.3>
Call-ID: 1874630@10.43.122.3
Cseq: 12442 INVITE


v=0
o=user 14341433 14341433 IP4 10.43.122.3
s=.
t=0 0
c=IN IP4 10.43.122.3
m=audio 13222 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

Methods

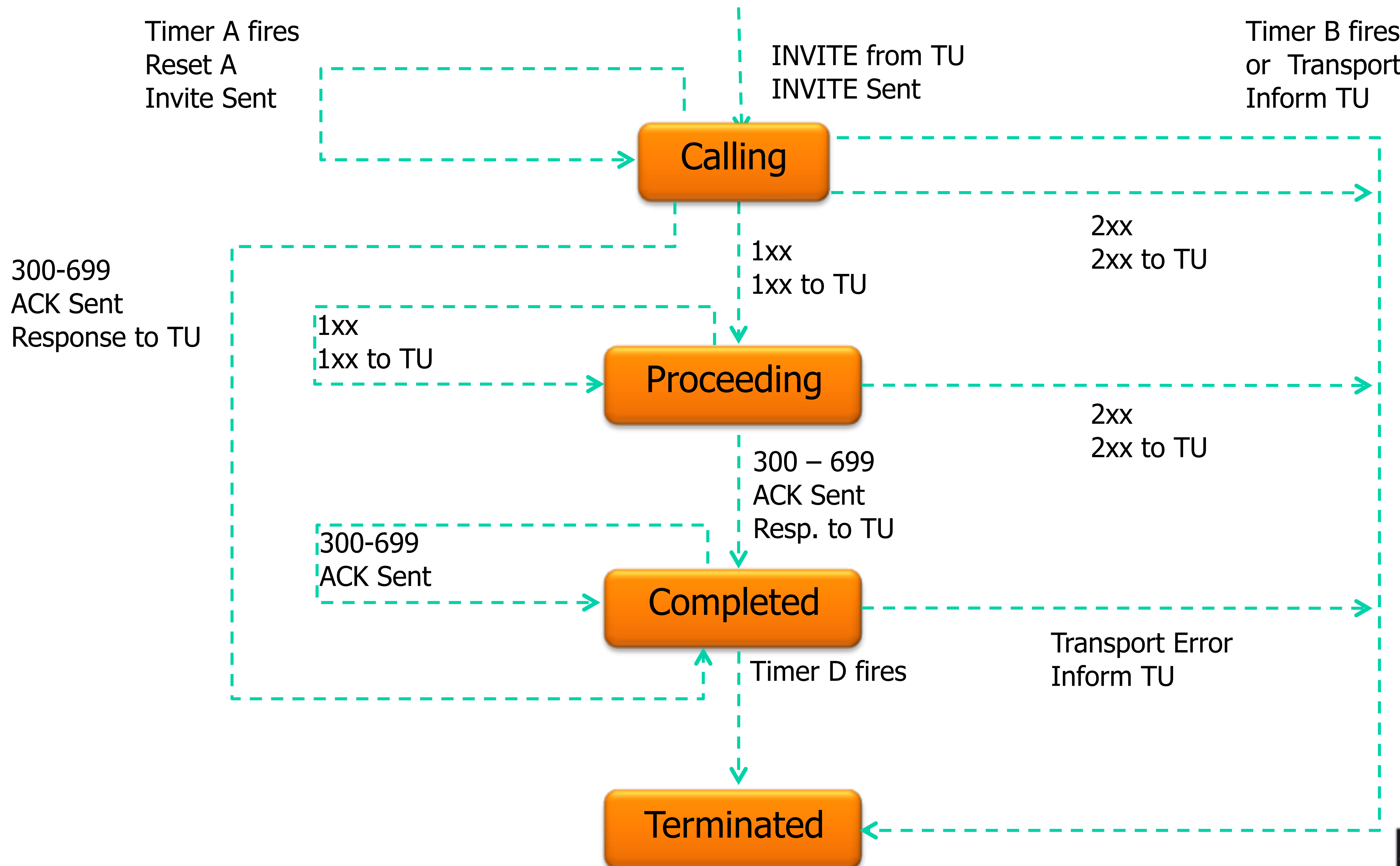
➡ *INVITE*

Methods

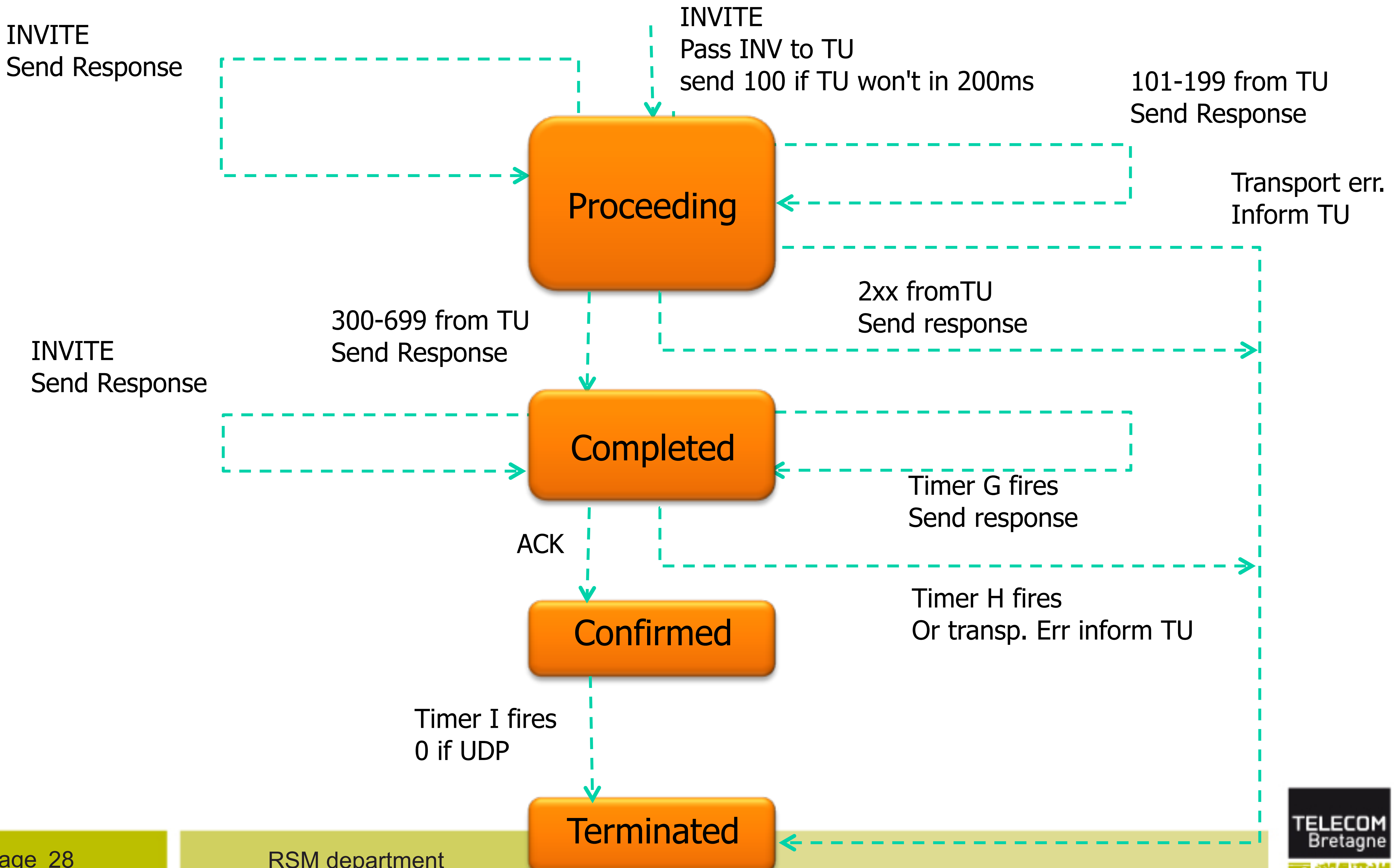
- **INVITE - establish media session between user agents**
  - Equivalent to Set Up message of Q.931
  - Always acknowledged with an ACK method
  - Usually contain a message body with the session description
  - A session is established only when the INVITE, 200 OK and ACK have been exchanged
  - A session (opened with an INVITE) is closed with a BYE method
  - An INVITE establishes a **dialog** identified with Call-ID, to and from tags

- **BYE - Terminate an established media session**
  - Equivalent to the Release message of Q.931
  - Only sent by user agents participating in the session (never by a proxy)

INVITE Client transaction



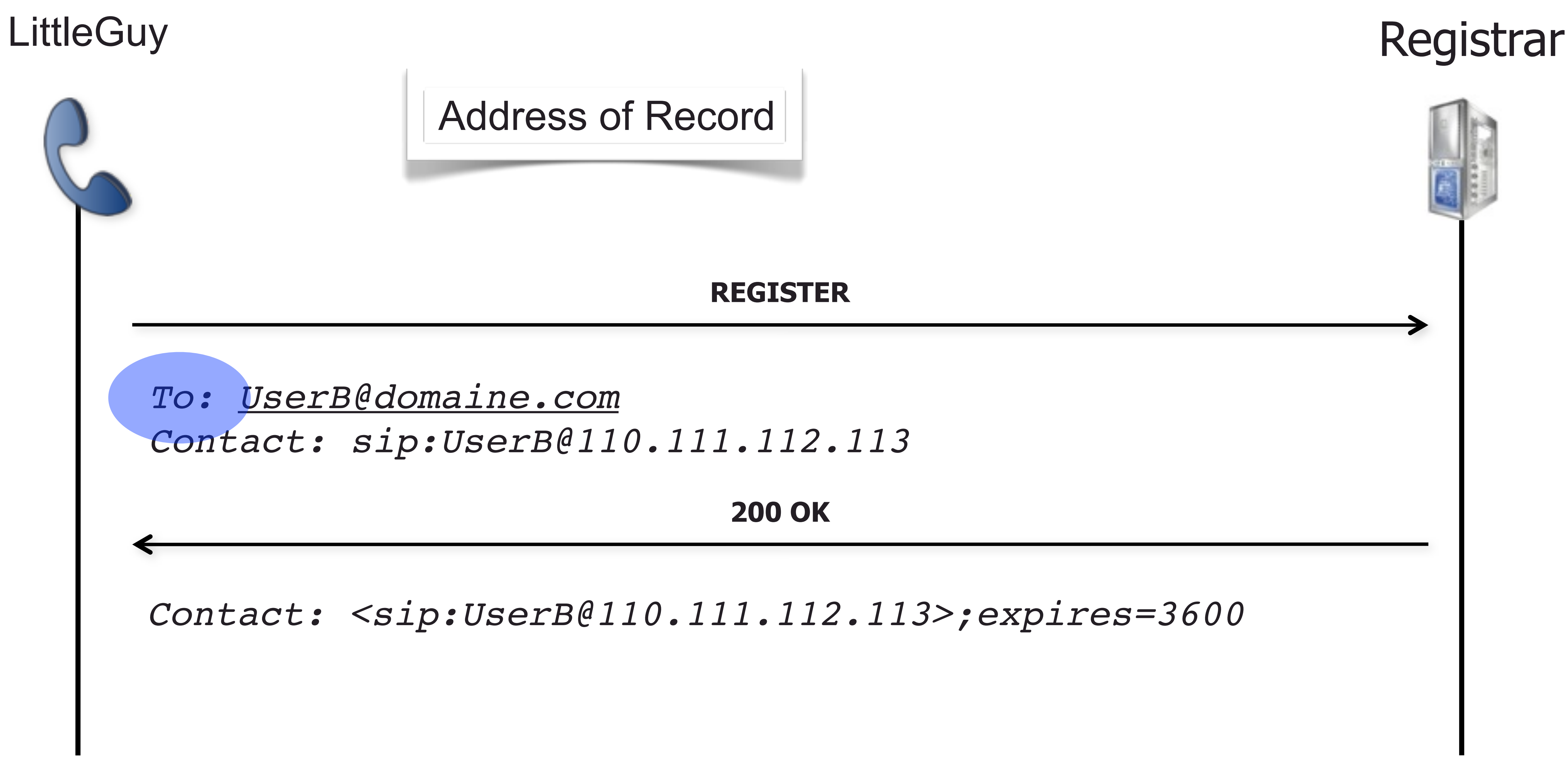
INVITE server transaction



Methods

➡ *Non-INVITE*

Registration



Authenticated registration

- **REGISTER - Notify a SIP network about the current contact URI (IP address) of the user agent**

An example REGISTER request

```
REGISTER sip:b.com SIP/2.0
Via: SIP/2.0/UDP 192.168.15.2
From: sip:barbara@b.com;tag=199257
To: sip:barbara@b.com
Contact: <sip:b@192.168.15.2>
Expires: 45
Call-ID: 950398549@192.168.15.2
CSeq: 1 REGISTER
```

An example REGISTER request

Request-URI registration domain	REGISTER sip:b.com SIP/2.0
	Via: SIP/2.0/UDP 192.168.15.2
Who’s registering	From: sip:barbara@b.com;tag=199257
AOR	To: sip:barbara@b.com
Contact	Contact: <sip:barbara@192.168.15.2>
Duration in minutes	Expires: 45
	Call-ID: 950398549@192.168.15.2 CSeq: 1 REGISTER
Empty line	

An example REGISTER response

```
SIP/2.0 200 Ok
Via: SIP/2.0/UDP 192.168.15.2
From: sip:barbara@b.com;tag=199257
To: sip:barbara@b.com;tag=jjf223
Contact:<sip:barbara@192.168.15.2>;expires=
2700
Contact:<sip:10.0.0.1>;expires=345
Contact:<sip:10.0.0.2>;expires=1000
Call-ID: 950398549@192.168.15.2
CSeq: 1 REGISTER
```

An example REGISTER response

	SIP/2.0 200 Ok
	Via: SIP/2.0/UDP 192.168.15.2
Who's regisering	From: sip:barbara@b.com;tag=199257
AOR	To: sip:barbara@b.com;tag=jjf223
List of all Contact headers for know AORs	Contact:<sip:barbara@192.168.15.2>;expires= 2700 Contact:<sip:10.0.0.1>;expires=345 Contact:<sip:10.0.0.2>;expires=1000
	Call-ID: 950398549@192.168.15.2 CSeq: 1 REGISTER
Empty line	

Register: lifetime of the registration

- **Either use**
  - ***expires*** parameter

    ```
    Contact:<sip:barbara@192.168.15.2>;expires=2700
    ```

    - In seconds
    - only concerns that contact
  - ***Expires*** header

    ```
    Contact:<sip:barbara@[2001:660:200::1]>
    Contact:<sip:barbara@192.168.15.2>
    Expires:45
    ```

    - In minutes
    - Concerns all contacts that do not have an *expires* parameter
  - **No indication**
    - Default is 1 hour

REGISTER: refresh, cancel, query

- It is up to the user agent to refresh registrations of Contact addresses. In order to do so, a UA has to resend its initial REGISTER request.
- In order to cancel a Contact registration, a user agent has to set its "Expires" time to zero

```
To: sip:barbara@b.com
Contact: <sip:barbara@192.168.15.2>
Expires: 0
```

- In order to cancel all contact address of records, a UA could use an asterisk (*)

```
To: sip:barbara@b.com
Contact: *
Expires: 0
```

- Omitting the `Contact` header would not modify any AOR and the corresponding response would contain all existing registrations.

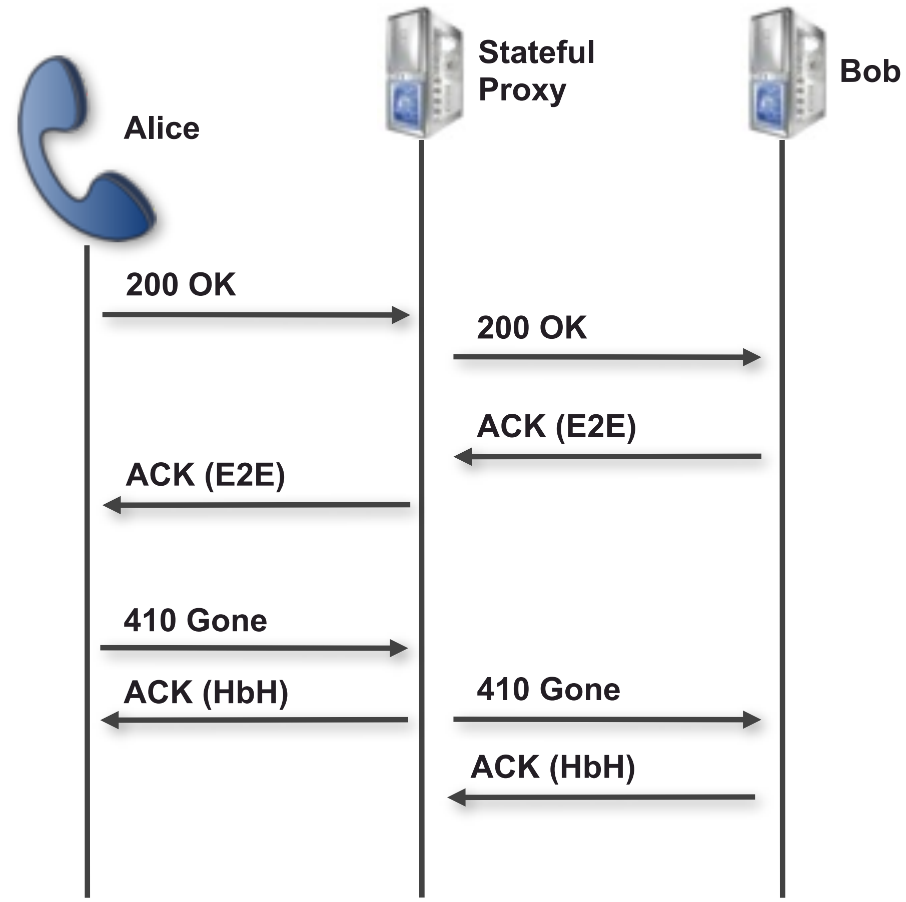
Register: multiple contacts registration

- **It is possible to associate more than one device URI to an AoR**
  - Use multiple `contact` headers
  - Use the parameter `q` to give preferences on the contact addresses
    - q varies between 0 and 1, the highest the preferred

```
Contact:&lt;sip:barbara@[2001:660:200::1]&gt;;q=0.4
Contact:&lt;sip:barbara@192.168.15.2&gt;;q=0.1
Expires:45
```

Methods

- **ACK - Acknowledge final responses to INVITE requests**
  - Final responses are 2xx, 3xx, 4xx, 5xx, 6xx
  - An Ack is end-to-end for 2xx responses, otherwise hop-by-hop (for stateful proxies)
  - CSeq is not incremented (same as the request), but the CSeq request method is Ack
  - Branch ID
    - Hop-by-hop ACK reuses the same branch ID as the INVITE
    - End-to-end ACK uses a different branch ID



Ack method

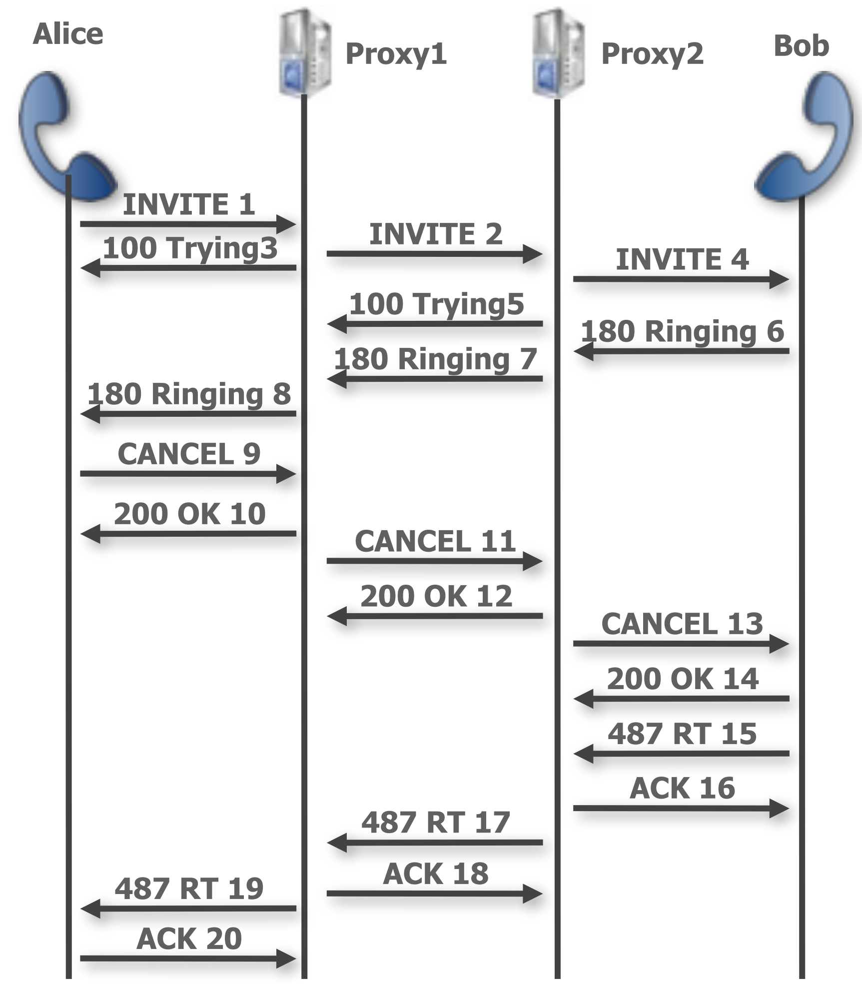
```
INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com>
Route: <sip:alg1.atlanta.example.com;lr>
Content-Type: application/sdp
Content-Length: 151
```

...

```
ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bhh
Max-Forwards: 70
Route: <sip:alg1.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0
```

Methods

- **CANCEL - terminate pending searches or call attempt**
  - Can be generated by user agents or proxy (provided that a 1xx response was received, and no final response)
  - Hop-by-hop request - receive a response by the next stateful element
  - Cseq and Branch ID are the same as the INVITE
  - A proxy receiving a CANCEL forwards the CANCEL and generates a response (200 OK). The INVITE is answered with a 487 Request Terminated



CANCEL method

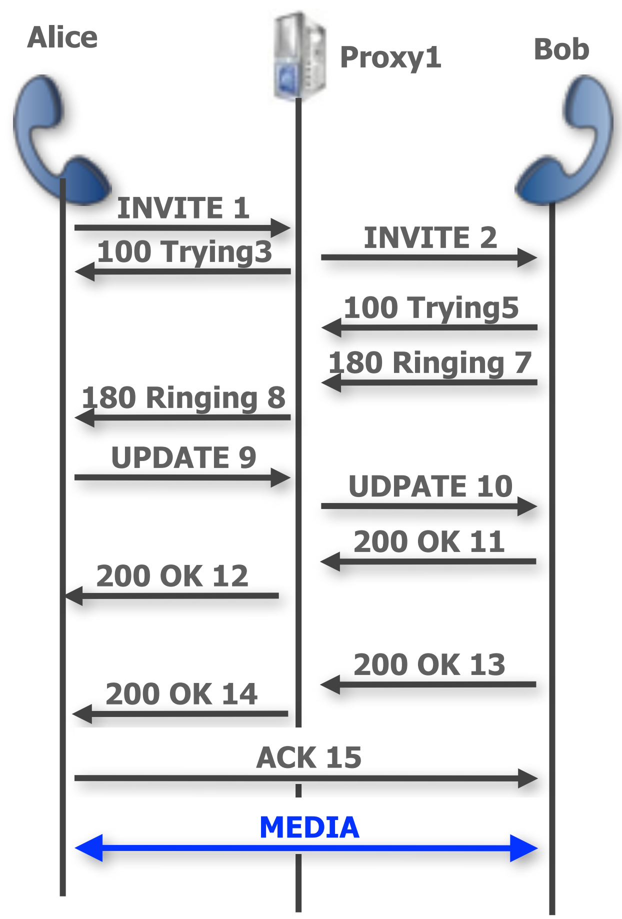
```
INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
Route: <sip:ss1.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com>
Content-Type: application/sdp
Content-Length: 151
```

...

```
CANCEL sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Route: <sip:ss1.atlanta.example.com;lr>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 CANCEL
Content-Length: 0
```

Methods

- **UPDATE - modify the state of a session without changing the state of the dialog**
  - Used instead of a re-INVITE in a pending session



Methods

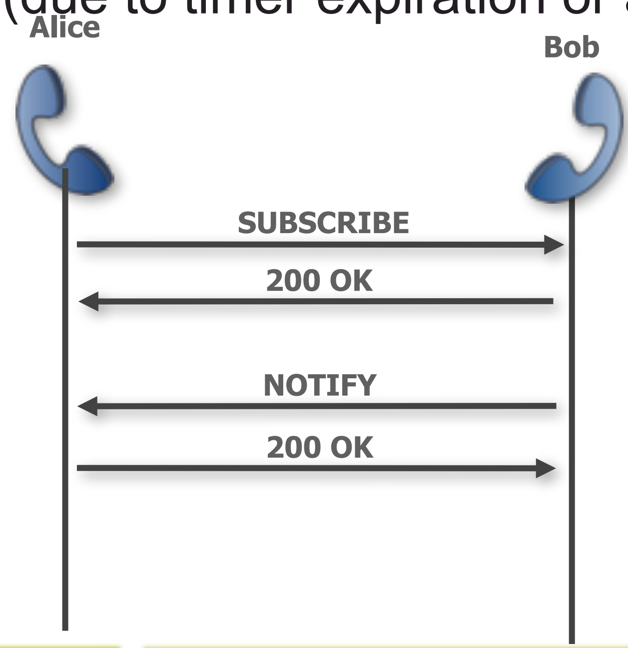
- **OPTIONS - query a user agent or server about its capabilities and discover its current availability**
  - Only generated by server or user agent
  - Responses are 4xx, 6xx for negative answers and 2xx for positive answers with
    - Allow: specify the requests it accepts
    - Accept: type of accepted Internet media types (e.g., Application/SDP)
    - Accept-Encoding: used to specify acceptable message body encoding schemes (e.g., Accept-Encoding: text/plain)
    - Accept-Language: preferences for language (such as the one used for reason phrase, or subject)

OPTIONS: request and response

```
OPTIONS sip:norton@savons.com SIP/2.0
via: SIP/2.0/UDP client1.telecom-bretagne.eu;branch=z9hG4bK1834
Max-Forwards: 70
To: <sip:edouard.norton@savons.com>
From: B. Pitt <sip:brad.pitt@savons.com>;tag=34
Call-ID: 9352812@client1.telecom-bretagne.eu
CSeq : 1 OPTIONS
Content-Length: 0


SIP/2.0 200 OK
via: SIP/2.0/UDP client1.telecom-bretagne.eu;branch=z9hG4bK1834;
received=192.168.0.2
Max-Forwards: 70
To: <sip:norton@savons.com>;tag=68
From: B. Pitt <sip:brad.pitt@savons.com>;tag=34
Call-ID: 9352812@client1.telecom-bretagne.eu
CSeq : 1 OPTIONS
Allow: INVITE, OPTIONS, ACK, BYE, CANCEL, REFER
Accept-Language: en, de, fr
Content-Length: ...
Content-Type: application/sdp

v=0
...
```

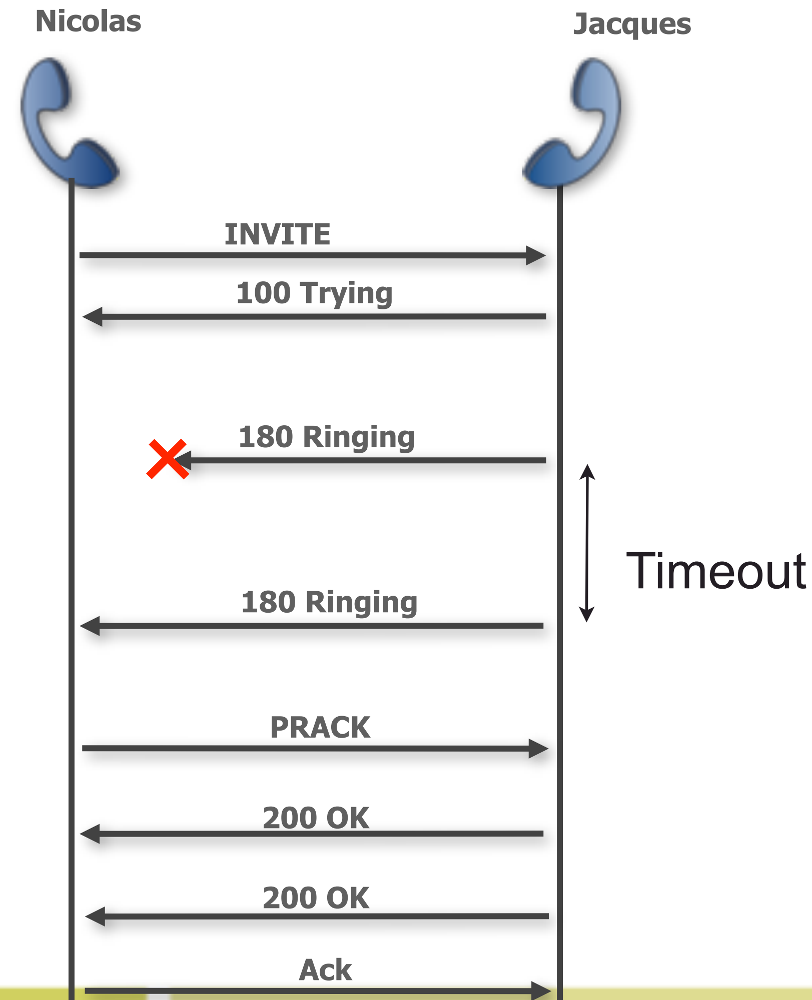
Methods

- **SUBSCRIBE - establish a subscription for the purpose of receiving notifications about a particular event**
  - Request the state and state updates from a remote node
  - Establish a dialog during the time indicated in the `Expires` header. A server accepting the request responds with a 200 OK
  - Identification of events is provided by three pieces of information:
    - Request URI
    - Event Type (*Event* header)
    - and (optionally) message body.
- **NOTIFY - Convey information about the occurrence of a particular event**
  - Always sent within a dialog
  - Is also used to notify the unsubscribe (due to timer expiration or an explicit unsuscribe (with `Expires: 0`)

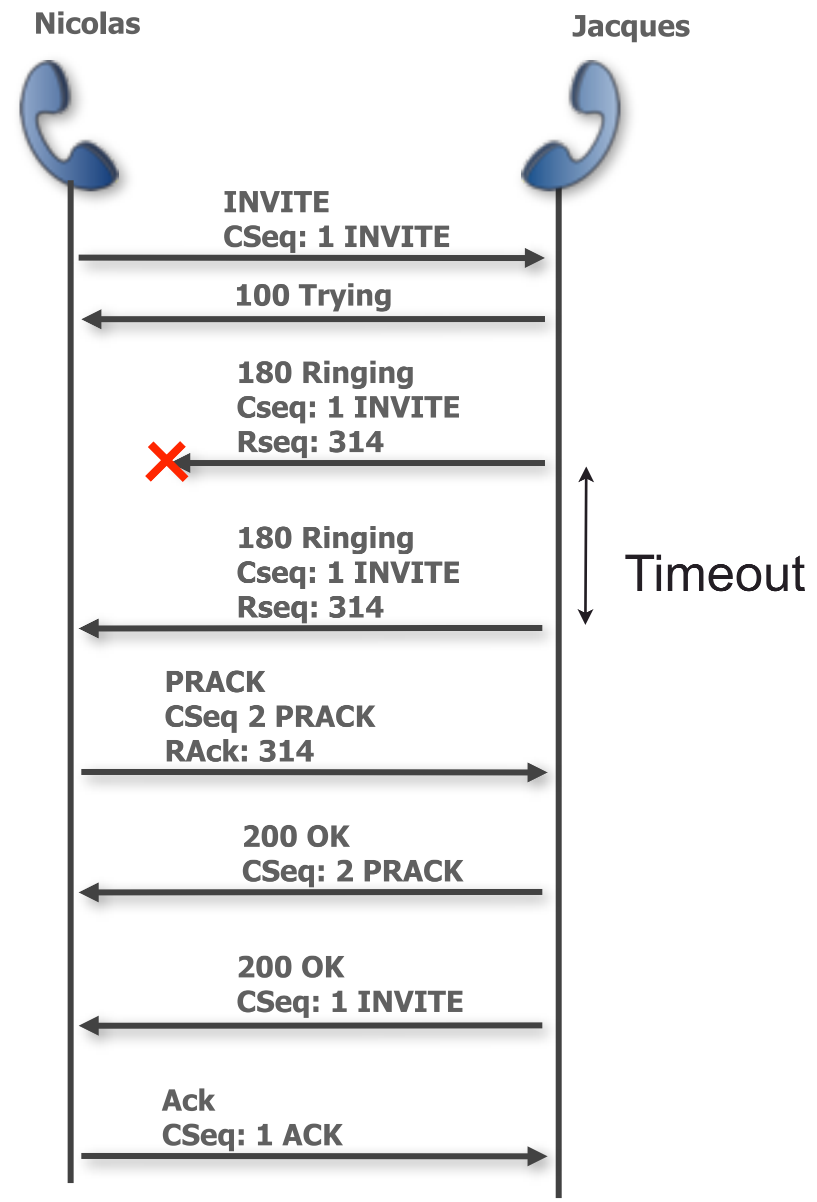


Methods

- **PRACK (RFC 3262)**
  - **Used to acknowledge receipt of reliability transported provisional responses (1xx - exept the 100 which is never reliably transported)**
  - **Reliably transported provisional responses contain a *RSeq* header (with a sequence number) and a *Supported: 100rel* header**
  - **A timer on the UAS triggers the retransmission of the provisional response**
  - **A `RAck` header field is used within a response to a PRACK request to reliably acknowledge a provisional response that contained a `RSeq` header field. The RAck echoes the `CSeq` and `RSeq` from the provisional response**

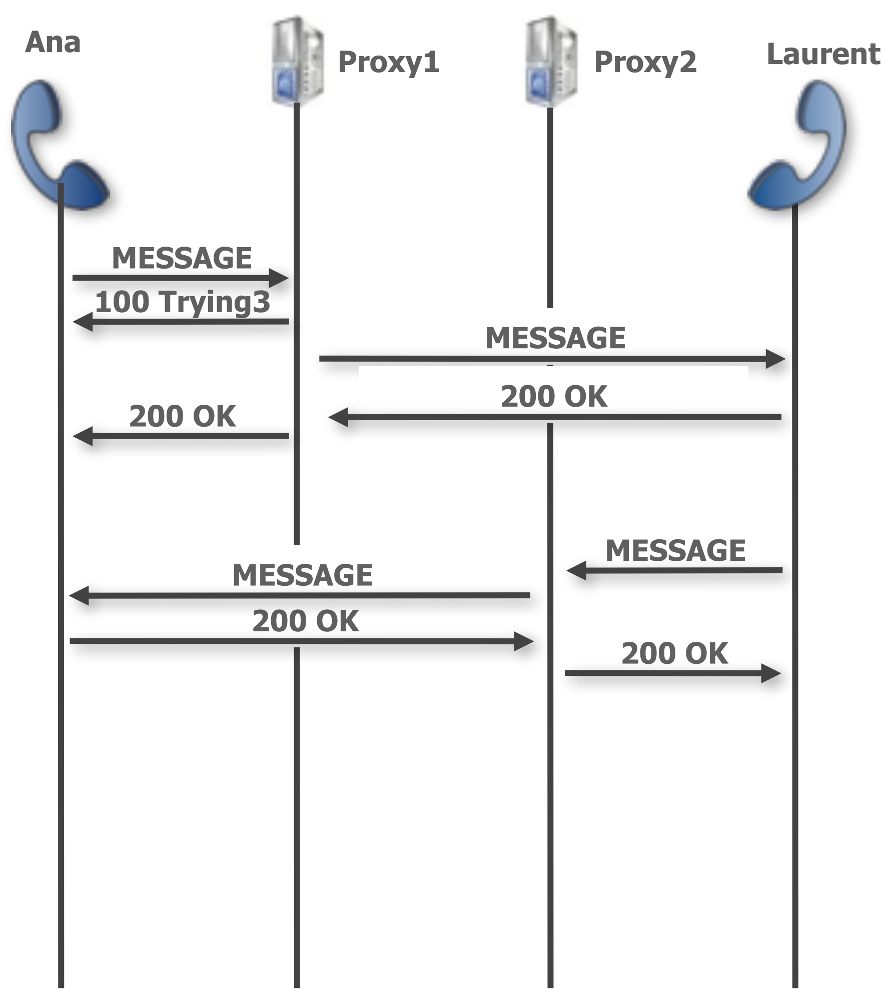


PRACK - usage of CSeq and RSeq



Methods

- **MESSAGE - Transport instant message using SIP**
  - Can be exchanged within a dialog or outside a dialog
  - Acknowledged by a 200 OK message



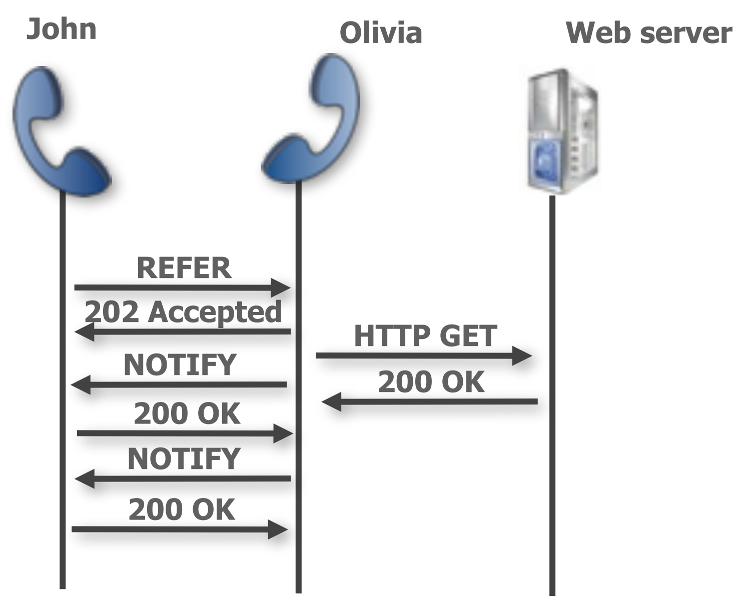
Methods

- **INFO - Send call signaling information to another user agent with which it has an established media session**
  - Different from a re-INVITE as it does not change the characteristics of the call
  - Can be used to transport midcall signaling information

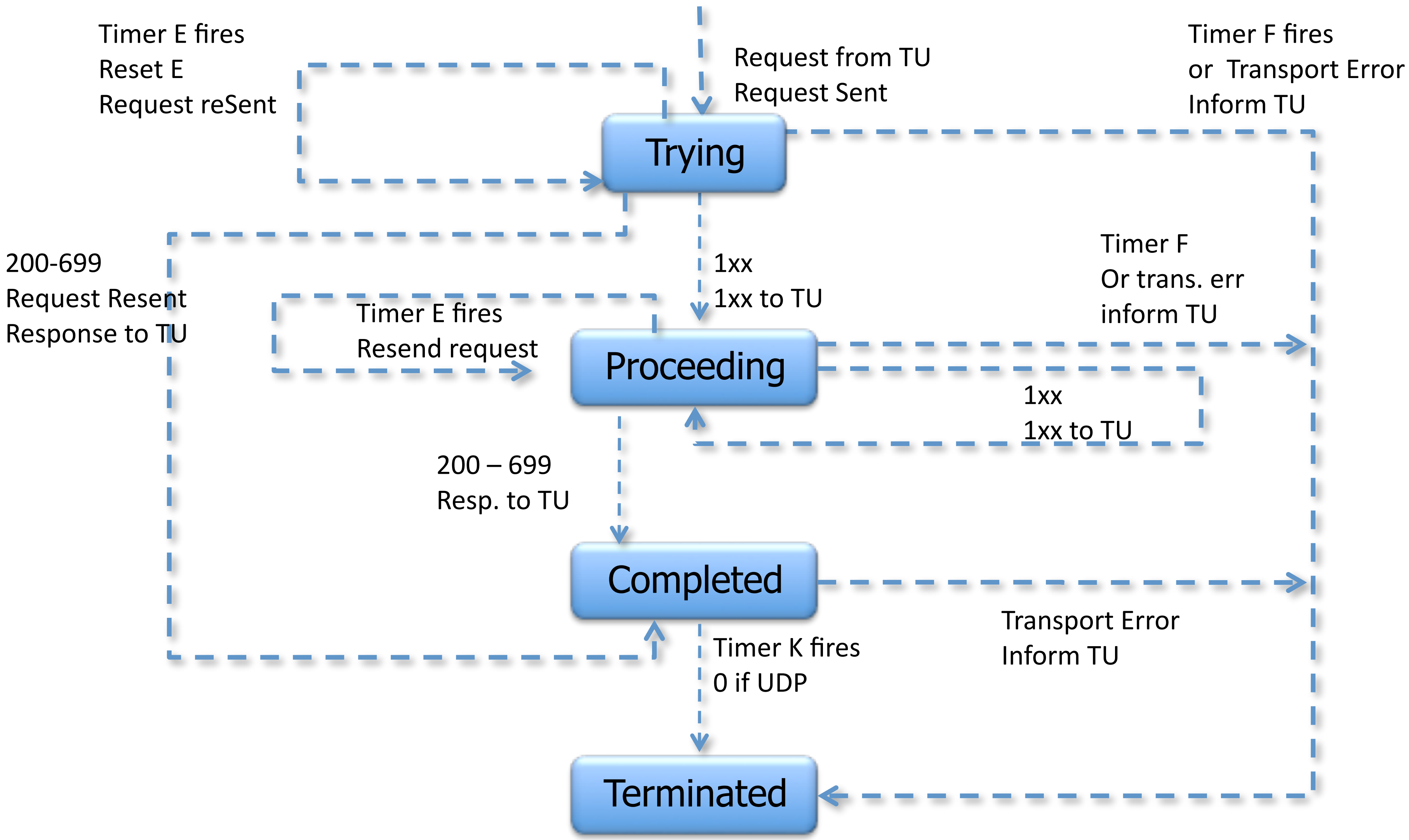
```
INFO sip:poynting@mason.edu.uk SIP/2.0
Via: SIP/2.0/UDP cavendish.kings/cambridge.edu.uk;
banch=z9hG4bK24555
Max-Forwards: 70
To: sip:poynting@mason.edu.uk SIP/2.0; tag=12390
From: sip:quelqun@kings.cambridge.edu.uk; tag=5289
Call-ID: 18379@cavendish.kings.cambridge.edu.uk
CSeq: 6 INFO
Content-Type message/ISUP
Content-Length: 16

51a6324134527
```

Methods

- **REFER - used by a user agent to request another user agent to access a URI or URL ressource**
  - May be used to a call transfer service

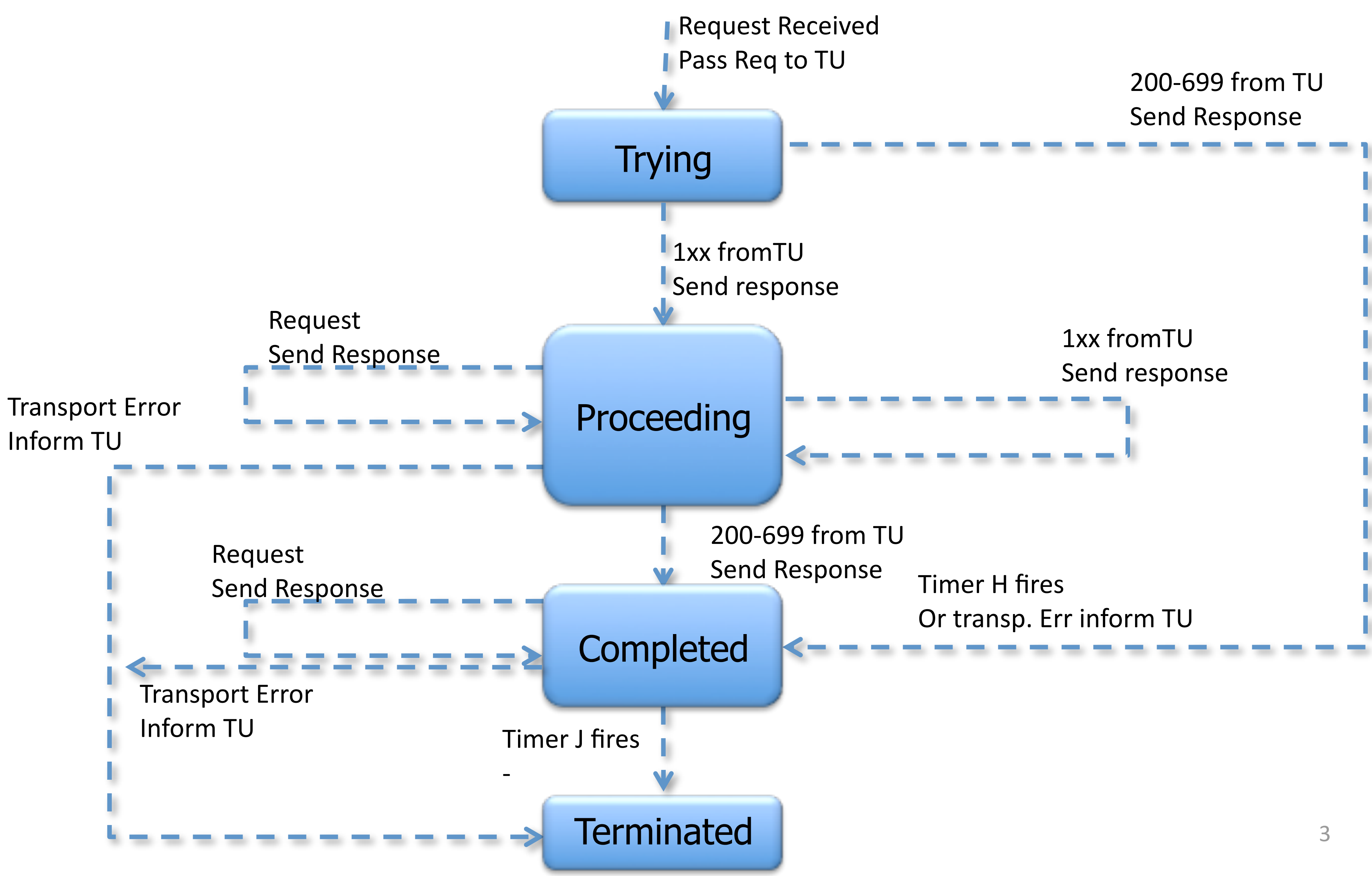


Non-INVITE client transactions



Monday, February 20, 12

Non-INVITE server transaction



SIP models

- **Client / Server**
  - Determined by the initiator of the message
    - Request sender: client
    - Request receiver: server
- **Transaction : Messages exchange from a request to the final response (between 200 and 699)**
- **Dialog: SIP relation between two user agents that last for a while.**
  - Can be understood as a session
  - Created by a reponse to an INVITE (like a *200 Ok*) or to SUBSCRIBE
  - Ease messages sequencing
  - Allow routing SIP messaging that belong to the same dialog

Transaction and Dialog

Transaction

- Identified by the *branch* field of the *via* header
- Determined and unique on each traversed proxy / UA

```
REGISTER sip:registrar.biloxi.com SIP/2.0
Via: SIP/2.0/UDP bobspc.biloxi.com:5060;branch=z9hG4bKnashds7
```

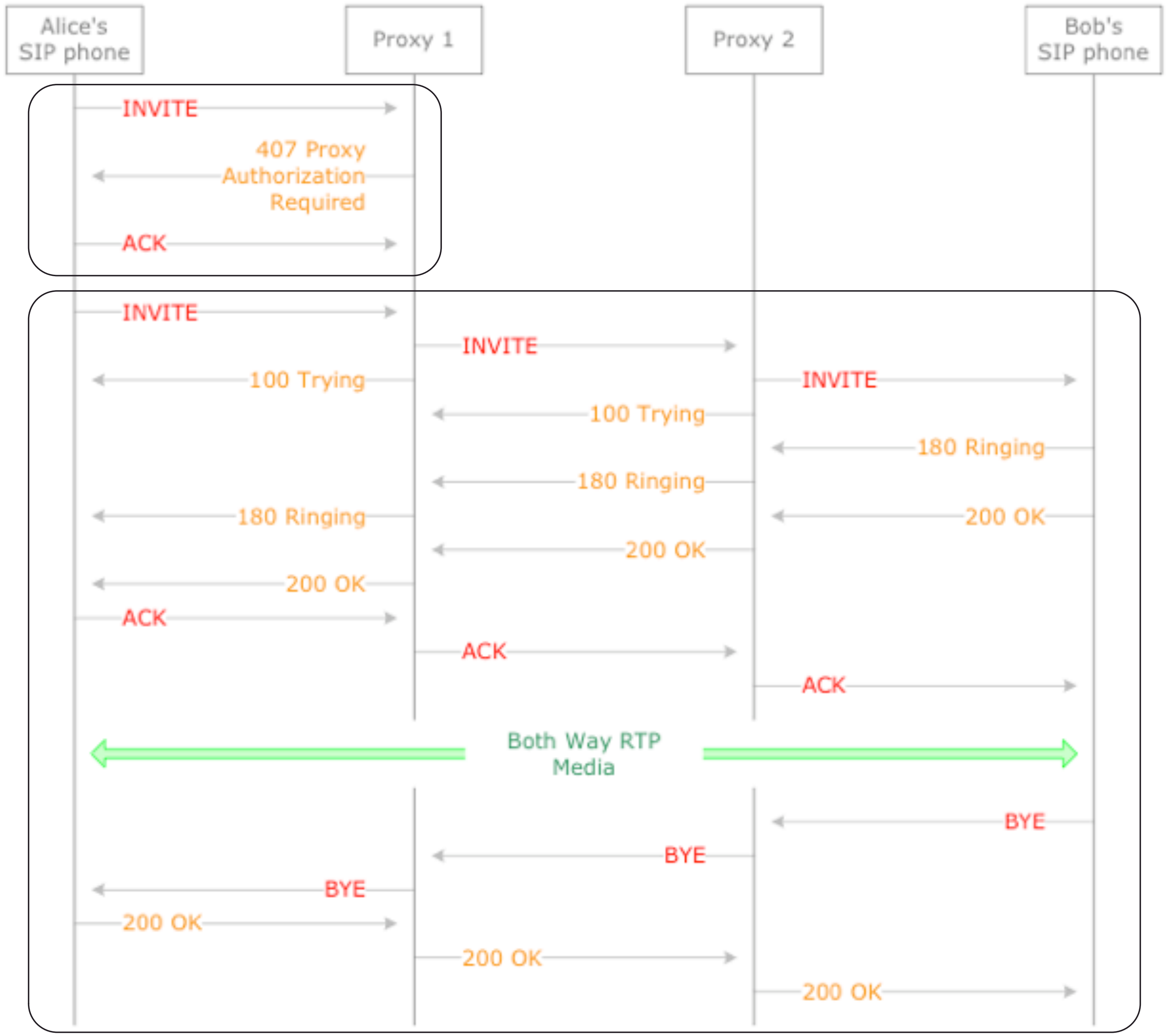
Dialog

- Identified by the *call-id*, the *to tag* and the *from tag*

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bobspc.biloxi.com:5060;branch=z9hG4bKnashds7;received=192.0.2.4
To: Bob <sip:bob@biloxi.com>;tag=2493k59kd
From: Bob <sip:bob@biloxi.com>;tag=456248
Call-ID: 843817637684230@998sdasdh09
```

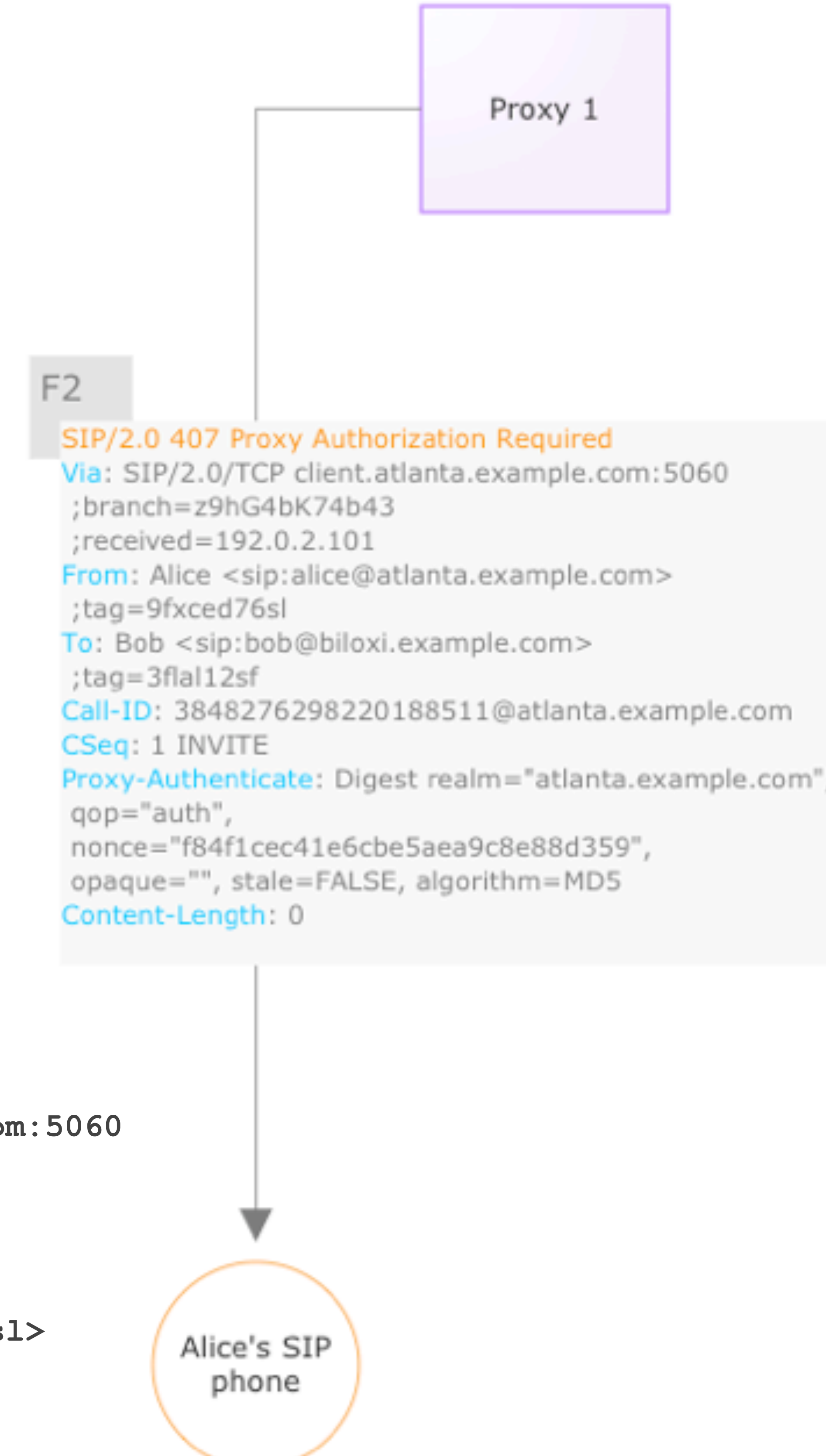
SIP call flows - session establishment through two proxies (RFC 3665)

Transaction



Dialog


Proxy 1
F1
INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TCP client.atlanta.example.com:5060
;branch=z9hG4bK74b43
Max-Forwards: 70
Route: <sip:ss1.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>
;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com
;transport=tcp>
Content-Type: application/sdp
Content-Length: 151
v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
Alice's SIP phone




```
INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060
;branch=z9hG4bk74b43
Max-Forwards: 70
Route: <sip:ss1.altanta.example.com;lr>
From: Alice <sip:alice@a.com;tag=9fxced76sl>
To: Bob <sip:bob@biloxi.example.com>
Contact: <sip:alice@10.43.122.3>
Call-ID: 3838276298220188511@atlanta.example.com
Cseq: 1 INVITE
```


```
INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060
;branch=z9hG4bk74b43
Max-Forwards: 70
Route: <sip:ss1.altanta.example.com;lr>
From: Alice <sip:alice@a.com;tag=9fxced76sl>
To: Bob <sip:bob@biloxi.example.com>
Contact: <sip:alice@10.43.122.3>
Call-ID: 3838276298220188511@atlanta.example.com
Cseq: 1 INVITE
```

Proxy 1

F1

```
INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TCP client.atlanta.example.com:5060
 ;branch=z9hG4bK74b43
Max-Forwards: 70
Route: <sip:ss1.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>
 ;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com
 ;transport=tcp>
Content-Type: application/sdp
Content-Length: 151

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

Alice's SIP phone

Proxy 1

F4

```
INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TCP client.atlanta.example.com:5060
 ;branch=z9hG4bK74bf9
Max-Forwards: 70
Route: <sip:ss1.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>
 ;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 INVITE
Contact: <sip:alice@client.atlanta.example.com
 ;transport=tcp>
Proxy-Authorization: Digest username="alice",
 realm="atlanta.example.com",
 nonce="wf84f1ceczx41ae6cbe5aea9c8e88d359", opaque="",
 uri="sip:bob@biloxi.example.com",
 response="42ce3cef44b22f50c6a6071bc8"
Content-Type: application/sdp
Content-Length: 151

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

Alice's SIP phone

F5

```
INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TCP ss1.atlanta.example.com:5060
 ;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP client.atlanta.example.com:5060
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>
 ;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 INVITE
Contact: <sip:alice@client.atlanta.example.com
 ;transport=tcp>
Content-Type: application/sdp
Content-Length: 151

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F6

```
SIP/2.0 100 Trying
Via: SIP/2.0/TCP client.atlanta.example.com:5060
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>
 ;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 INVITE
Content-Length: 0
```

```
INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060
;branch=z9hG4bk74bf9
Max-Forwards: 70
Route: <sip:ss1.altanta.example.com;lr>
From: Alice <sip:alice@a.com;tag=9fxced76sl>
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
Cseq: 2 INVITE
Contact: <sip:alice@client.atlanta.example.com ;transport:tcp>
Proxy-Authorization: Digest username="alice",
realm="atkanta.example.com",
nonce="wf84f1ceczx41ae6cbe5aea9c8e88d359", opaque="",
uri="sip:bob@biloxi.example.com",
response="42ce3cef44b22f50c6a6071bc8"
Content-Type: application/sdp
Content-Length: 151
```

Proxy 1

F8

SIP/2.0 100 Trying
Via: SIP/2.0/TCP ss1.atlanta.example.com:5060
;branch=z9hG4bK2d4790.1
;received=192.0.2.111
Via: SIP/2.0/TCP client.atlanta.example.com:5060
;branch=z9hG4bK74bf9
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>
;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 INVITE
Content-Length: 0

Proxy 2

```
INVITE sip:bob@biloxi.example.com SIP/2.0

Via: SIP/2.0/UDP client.atlanta.example.com:5060
;branch=z9hG4bk74bf9
Max-Forwards: 70
Route: <sip:ss1.altanta.example.com;lr>
From: Alice <sip:alice@a.com;tag=9fxced76sl>
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
Cseq: 2 INVITE
Contact: <sip:alice@client.atlanta.example.com ;transport:tcp>
Proxy-Authorization: Digest username="alice",
realm="atkanta.example.com",
nonce="wf84f1ceczx41ae6cbe5aea9c8e88d359", opaque="",
uri="sip:bob@biloxi.example.com",
response="42ce3cef44b22f50c6a6071bc8"
Content-Type: application/sdp
Content-Length: 151
```

F7

INVITE sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TCP ss2.biloxi.example.com:5060
;branch=z9hG4bK721e4.1
Via: SIP/2.0/TCP ss1.atlanta.example.com:5060
;branch=z9hG4bK2d4790.1
;received=192.0.2.111
Via: SIP/2.0/TCP client.atlanta.example.com:5060
;branch=z9hG4bK74bf9
;received=192.0.2.101
Max-Forwards: 68
Record-Route: <sip:ss2.biloxi.example.com;lr>,
<sip:ss1.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>
;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 INVITE
Contact: <sip:alice@client.atlanta.example.com
;transport=tcp>
Content-Type: application/sdp
Content-Length: 151

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

Bob's SIP phone

Proxy 1

F10

SIP/2.0 180 Ringing
Via: SIP/2.0/TCP ss1.atlanta.example.com:5060
;branch=z9hG4bK2d4790.1
;received=192.0.2.111
Via: SIP/2.0/TCP client.atlanta.example.com:5060
;branch=z9hG4bK74bf9
;received=192.0.2.101
Record-Route: <sip:ss2.biloxi.example.com;lr>,
<sip:ss1.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>
;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
;tag=314159
Call-ID: 3848276298220188511@atlanta.example.com
Contact: <sip:bob@client.biloxi.example.com;transport=tcp>
CSeq: 2 INVITE
Content-Length: 0

Proxy 2

F9

SIP/2.0 180 Ringing
Via: SIP/2.0/TCP ss2.biloxi.example.com:5060
;branch=z9hG4bK721e4.1
;received=192.0.2.222
Via: SIP/2.0/TCP ss1.atlanta.example.com:5060
;branch=z9hG4bK2d4790.1

F11

SIP/2.0 180 Ringing
Via: SIP/2.0/TCP client.atlanta.example.com:5
;branch=z9hG4bK74bf9
;received=192.0.2.101
Record-Route: <sip:ss2.biloxi.example.com;lr
<sip:ss1.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>
;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
;tag=314159
Call-ID: 3848276298220188511@atlanta.exan
Contact: <sip:bob@client.biloxi.example.com;
CSeq: 2 INVITE
Content-Length: 0

Alice's SIP phone

```
INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060
;branch=z9hG4bk74bf9
Max-Forwards: 70
Route: <sip:ss1.altanta.example.com;lr>
From: Alice <sip:alice@a.com;tag=9fxced76sl>
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
Cseq: 2 INVITE
Contact: <sip:alice@client.atlanta.example.com ;transport:tcp>
Proxy-Authorization: Digest username="alice",
realm="atkanta.example.com",
nonce="wf84f1ceczx41ae6cbe5aea9c8e88d359", opaque="",
uri="sip:bob@biloxi.example.com",
response="42ce3cef44b22f50c6a6071bc8"
Content-Type: application/sdp
Content-Length: 151
```

Proxy 2

F12

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP ss2.biloxi.example.com:5060
 ;branch=z9hG4bK721e4.1
 ;received=192.0.2.222
Via: SIP/2.0/TCP ss1.atlanta.example.com:5060
 ;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP client.atlanta.example.com:5060
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss2.biloxi.example.com;lr>,
 <sip:ss1.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>
 ;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
 ;tag=314159
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=tcp>
 Content-Type: application/sdp
 Content-Length: 147
```

```
INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060
;branch=z9hG4bk74bf9
Max-Forwards: 70
Route: <sip:ss1.altanta.example.com;lr>
From: Alice <sip:alice@a.com;tag=9fxced76sl>
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
Cseq: 2 INVITE
Contact: <sip:alice@client.atlanta.example.com ;transport:tcp>
Proxy-Authorization: Digest username="alice",
realm="atkanta.example.com",
nonce="wf84f1ceczx41ae6cbe5aea9c8e88d359", opaque="",
uri="sip:bob@biloxi.example.com",
response="42ce3cef44b22f50c6a6071bc8"
Content-Type: application/sdp
Content-Length: 151
```

```
v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=-
c=IN IP4 192.0.2.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

Bob's SIP phone

F13

Proxy 1 ← Proxy 2

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP ss1.atlanta.example.com:5060
 ;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP client.atlanta.example.com:5060
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss2.biloxi.example.com;lr>,
 <sip:ss1.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=314159
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 147

v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=-
c=IN IP4 192.0.2.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

Proxy 1

F14

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP client.atlanta.example.com:5060
 ;branch=z9hG4bK74bf9;received=192.0.2.101
Record-Route: <sip:ss2.biloxi.example.com;lr>,
 <sip:ss1.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=314159
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 147

v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=-
c=IN IP4 192.0.2.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

Alice's SIP phone

```
INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060
;branch=z9hG4bk74bf9
Max-Forwards: 70
Route: <sip:ss1.altanta.example.com;lr>
From: Alice <sip:alice@a.com;tag=9fxced76sl>
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
Cseq: 2 INVITE
Contact: <sip:alice@client.atlanta.example.com ;transport:tcp>
Proxy-Authorization: Digest username="alice",
realm="atkanta.example.com",
nonce="wf84f1ceczx41ae6cbe5aea9c8e88d359", opaque="",
uri="sip:bob@biloxi.example.com",
response="42ce3cef44b22f50c6a6071bc8"
Content-Type: application/sdp
Content-Length: 151
```

SIP call flows - session establishment through two proxies (RFC 3665)

F16

Proxy 1 → Proxy 2

```
ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TCP ss1.atlanta.example.com:5060
 ;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP client.atlanta.example.com:5060
 ;branch=z9hG4bK74b76
 ;received=192.0.2.101
Max-Forwards: 69
Route: <sip:ss2.biloxi.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>
 ;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
 ;tag=314159
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 ACK
Content-Length: 0
```

Proxy 2

F17

```
ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TCP ss2.biloxi.example.com:5060
 ;branch=z9hG4bK721e4.1
Via: SIP/2.0/TCP ss1.atlanta.example.com:5060
 ;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP client.atlanta.example.com:5060
 ;branch=z9hG4bK74b76
 ;received=192.0.2.101
Max-Forwards: 68
From: Alice <sip:alice@atlanta.example.com>
 ;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
 ;tag=314159
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 ACK
Content-Length: 0
```

Bob's SIP phone

```
INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060
;branch=z9hG4bk74bf9
Max-Forwards: 70
Route: <sip:ss1.altanta.example.com;lr>
From: Alice <sip:alice@a.com;tag=9fxced76sl>
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
Cseq: 2 INVITE
Contact: <sip:alice@client.atlanta.example.com ;transport:tcp>
Proxy-Authorization: Digest username=”alice”,
realm=”atkanta.example.com”,
nonce=”wf84f1ceczx41ae6cbe5aea9c8e88d359”, opaque=””,
uri=”sip:bob@biloxi.example.com”,
response=”42ce3cef44b22f50c6a6071bc8”
Content-Type: application/sdp
Content-Length: 151
```

SIP call flows - session establishment through two proxies (RFC 3665)

Proxy 1 ← Proxy 2

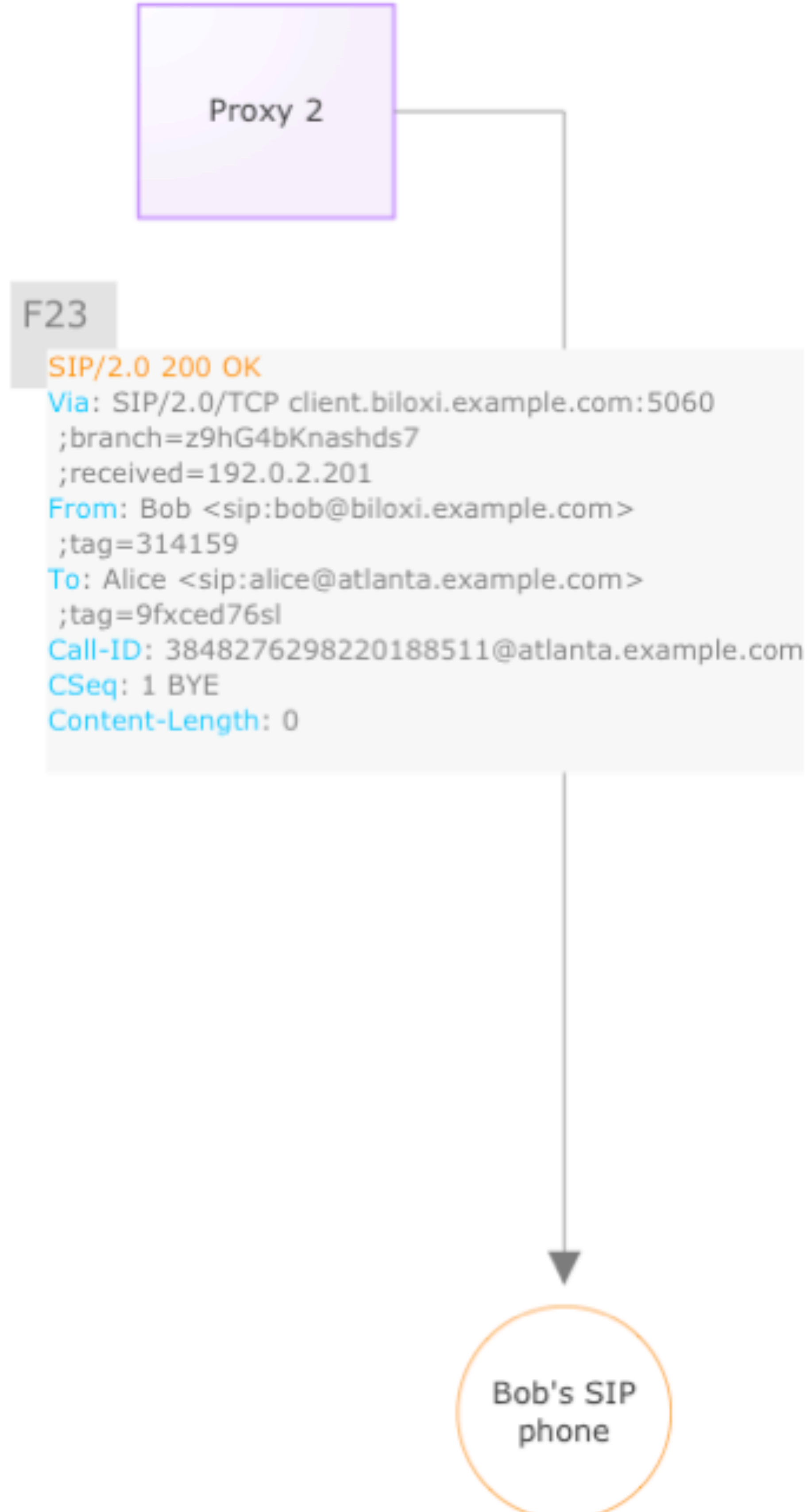
F19

```
BYE sip:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TCP ss2.biloxi.example.com:5060
 ;branch=z9hG4bK721e4.1
Via: SIP/2.0/TCP client.biloxi.example.com:5060
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.201
Max-Forwards: 69
Route: <sip:ss1.atlanta.example.com;lr>
From: Bob <sip:bob@biloxi.example.com>
 ;tag=314159
To: Alice <sip:alice@atlanta.example.com>
 ;tag=9fxced76sl
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 BYE
Content-Length: 0
```


Proxy 1
F20
BYE sip:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TCP ss1.atlanta.example.com:5060
;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP ss2.biloxi.example.com:5060
;branch=z9hG4bK721e4.1
;received=192.0.2.222
Via: SIP/2.0/TCP client.biloxi.example.com:5060
;branch=z9hG4bKnashds7
;received=192.0.2.201
Max-Forwards: 68
From: Bob <sip:bob@biloxi.example.com>
;tag=314159
To: Alice <sip:alice@atlanta.example.com>
;tag=9fxced76sl
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 BYE
Content-Length: 0
Alice's SIP phone


Proxy 1
F21
SIP/2.0 200 OK
Via: SIP/2.0/TCP ss1.atlanta.example.com:5060
;branch=z9hG4bK2d4790.1
;received=192.0.2.111
Via: SIP/2.0/TCP ss2.biloxi.example.com:5060
;branch=z9hG4bK721e4.1
;received=192.0.2.222
Via: SIP/2.0/TCP client.biloxi.example.com:5060
;branch=z9hG4bKnashds7
;received=192.0.2.201
From: Bob <sip:bob@biloxi.example.com>;tag=314159
To: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 BYE
Content-Length: 0
Alice's SIP phone

SIP call flows - session establishment through two proxies (RFC 3665)


Proxy 2
F23
SIP/2.0 200 OK
Via: SIP/2.0/TCP client.biloxi.example.com:5060
;branch=z9hG4bKnashds7
;received=192.0.2.201
From: Bob <sip:bob@biloxi.example.com>
;tag=314159
To: Alice <sip:alice@atlanta.example.com>
;tag=9fxced76sl
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 BYE
Content-Length: 0
Bob's SIP phone