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VVT-2001

Intermediate SIP for Voice, Video and Instant Messaging

James M. Polk

Recuerde siempre:

- Apagar su teléfono móvil/pager, o usar el modo "silencioso".
- Completar la evaluación de esta sesión y entregarla a los asistentes de sala.
- Ser puntual para asistir a todas las actividades de entrenamiento, almuerzos y eventos sociales para un desarrollo óptimo de la agenda.
- Completar la evaluación general incluida en su mochila y entregarla el miércoles 8 de Junio en los mostradores de registración. Al entregarla recibirá un regalo recordatorio del evento.

Understanding SIP: Intermediate

Learning Objectives

- Learn the Session Initiation Protocol as a protocol, not as a product
- Learn how SIP signals Voice, Video and Instant Messaging applications in IP networks
- Learn how SIP interworks with H.323 and MGCP

SIP Signaling Architecture

- **Elements of SIP**
- **SIP Signaling**
- **SIP in a Network**
- **SIP Summary**

Session Initiation Protocol

- **The Session Initiation Protocol (SIP) is an application layer control (signaling) protocol for:**
  - **creating**
  - **modifying and**
  - **terminating**

multimedia sessions with one or more participants

SIP History

- SIP was originally a multicast session set-up protocol for the I2 (mid-late 90s)
  - then someone figured out it was good for unicast
- First Standardized in March 1999 in RFC 2543
- Revised Standard in May 2002 in RFC 3261, with
  - 34 Standards Track Extension RFCs
  - 15 Working Group Internet Drafts, and
    - > 50 individual IDs that are not WG items (yet)

SIP – What is it?

SIP is a Session Set-up Protocol

SIP sessions include:

- Internet multimedia conferences
- Internet telephone calls
- Internet Video sessions and
- multimedia distribution

SIP – What is it? (cont'd)

SIP entities can communicate via:

- unicast
- multicast
- via a mesh of unicast relations or
- a combination of these

- in IPv4 and IPv6 environments using:
  - UDP
  - TCP
  - SCTP or
  - TLS over TCP

Generic SIP Packet format

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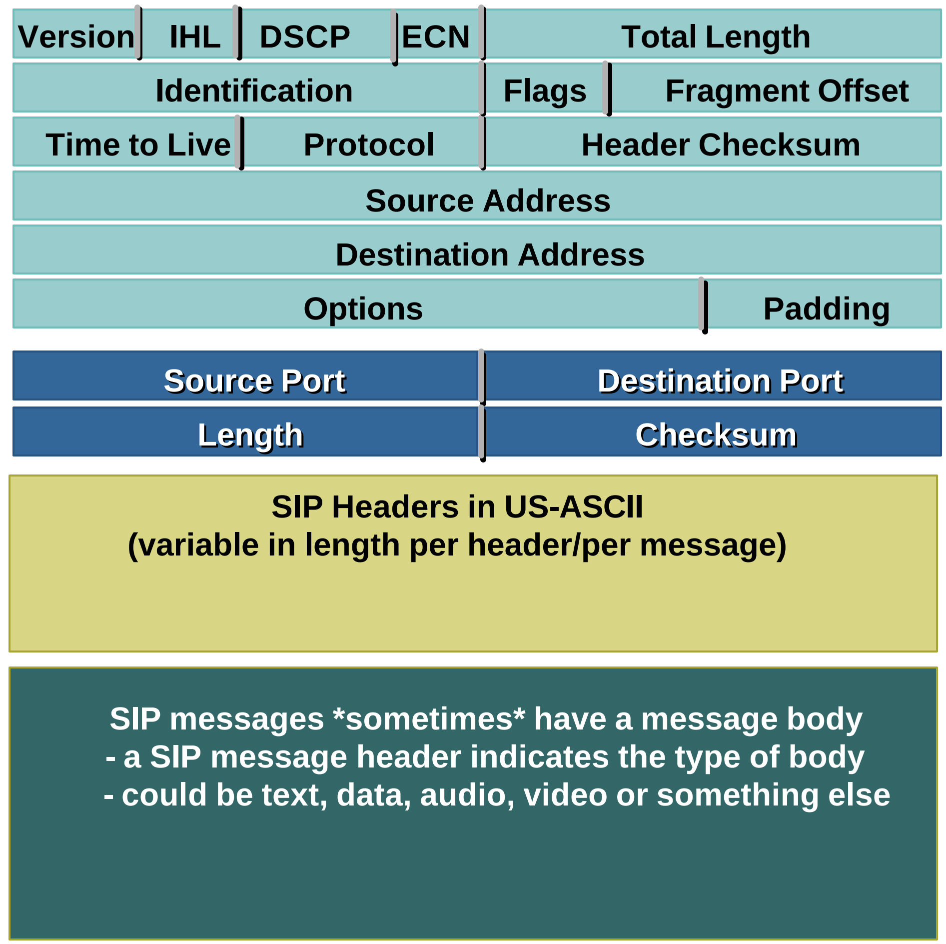
IPv4 Header is 20 Bytes and Binary

UDP Header is 8 Bytes and Binary (Layer 4 here could also be TCP or SCTP)

SIP Header is Text-based and variable in length

SIP message body is also variable, but not always present (depending on the Message-type)



SIP – What is it? (cont'd)

SIP is a simple extensible protocol

- Methods - Define the transaction
- Headers - Describe the transaction
- Message Body - SDP, Text, XML

The Power of SIP

- **SIP uses several existing IETF protocols to provide:**
  - Message formatting (HTTP 1.1) RFC 2616
  - Media Description (SDP) RFC 2327
  - Media (RTP) RFC 3550 and (RTSP) RFC 2326
  - Addressing (URL) RFC 1738 and (URI) RFC 2396
  - Name resolution and mobility (DHCP) RFC 2131 and (DNS) RFCs 1034&1035
  - Application encoding (MIME) RFC 2045
  - Security (TLS) RFC 2246 and (IPsec) RFC 2401&2406

SIP Tidbits

- **SIP chosen as signaling protocol for 3GPP/3GPP2 wireless networks (version 6 and later)**
- **PDAs will embrace 802.11a/b/g & Cellular connectivity**
- **SIP is text-based for easy implementation and debugging**

5 Facets for Establishing and Terminating Multimedia Communications using SIP

- **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 calling and called party
- **Session management:** Including transfer and termination of sessions, modifying session parameters, and invoking services

5 Components of SIP

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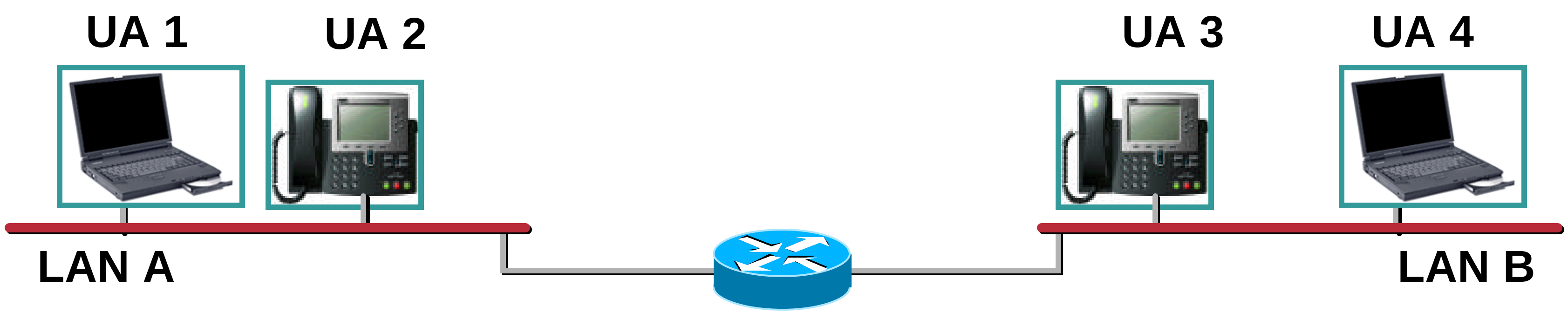
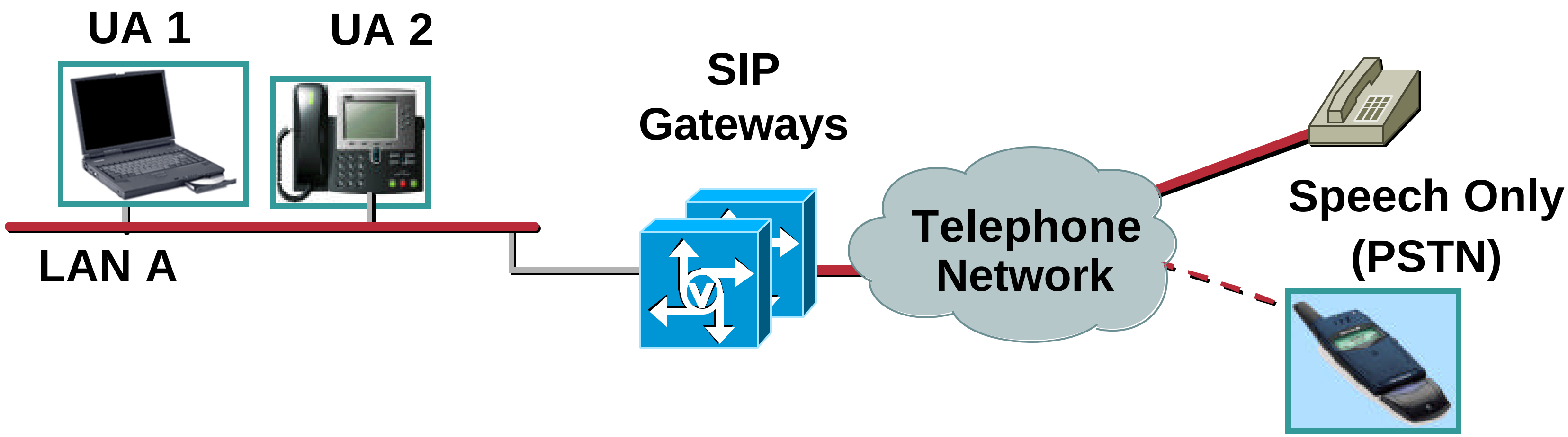
- User Agent [Mandatory]
- SIP Gateway [*optional*]
- Registrar Server [*optional*]
- Proxy Server [*optional*]
- Redirect Server [*optional*]

SIP User Agents



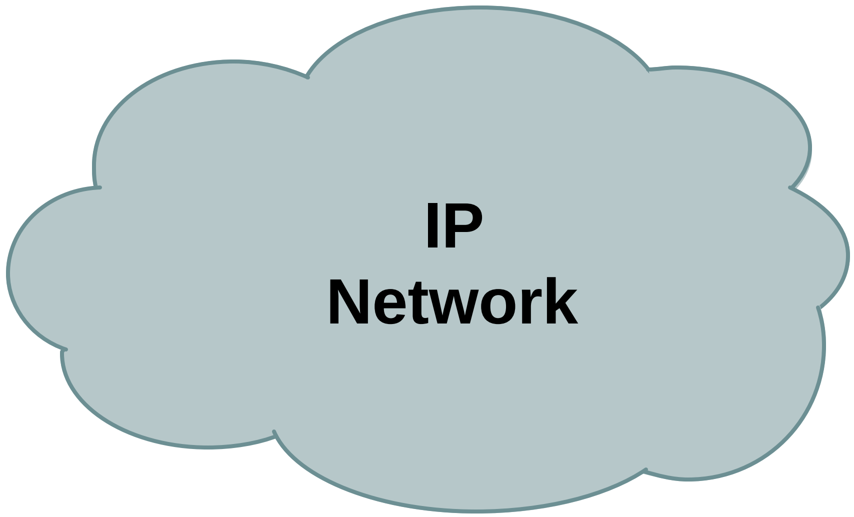
- Client—Server model
- User Agent Client (UAC)—Initiates sessions
- User Agent Server (UAS)—Responds to session requests
- User Agent = UAC + UAS

SIP Gateways



- **Translation between SIP protocol format to and from non-SIP protocol format**

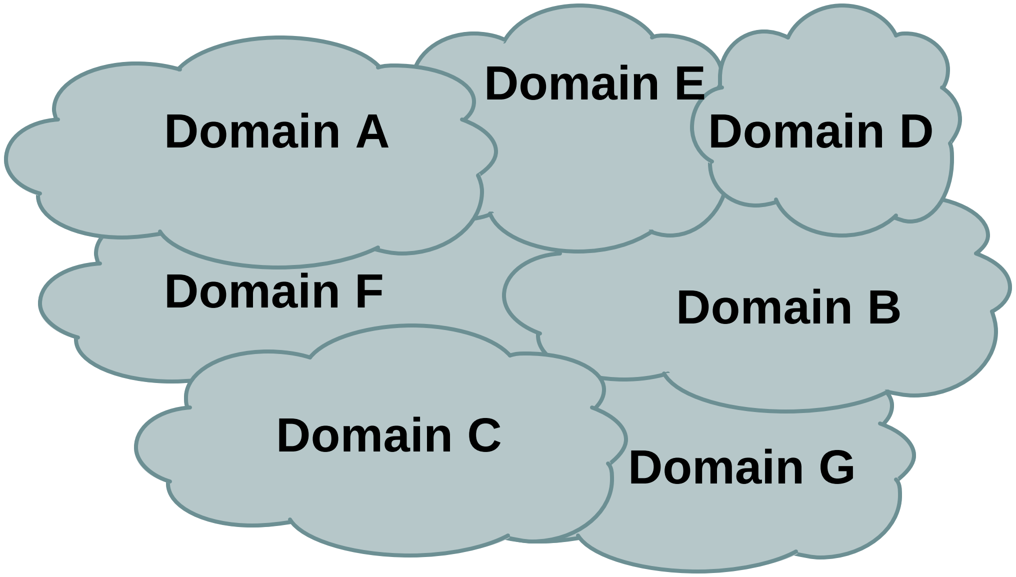
A “cloud” is a “cloud”... or is it?



- **When referring to an “IP Network”**

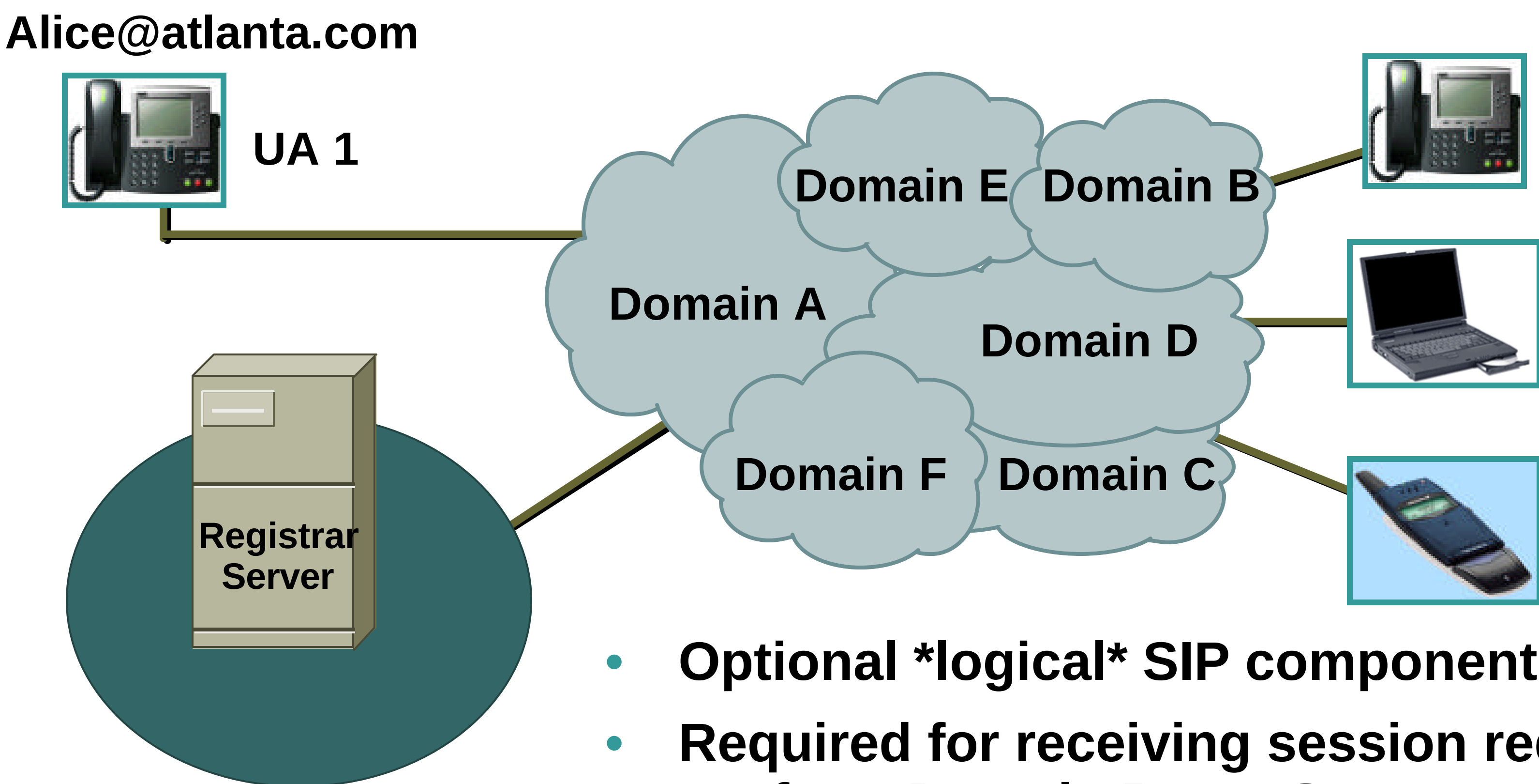
A “cloud” is a “cloud”... or is it?

IP Network =



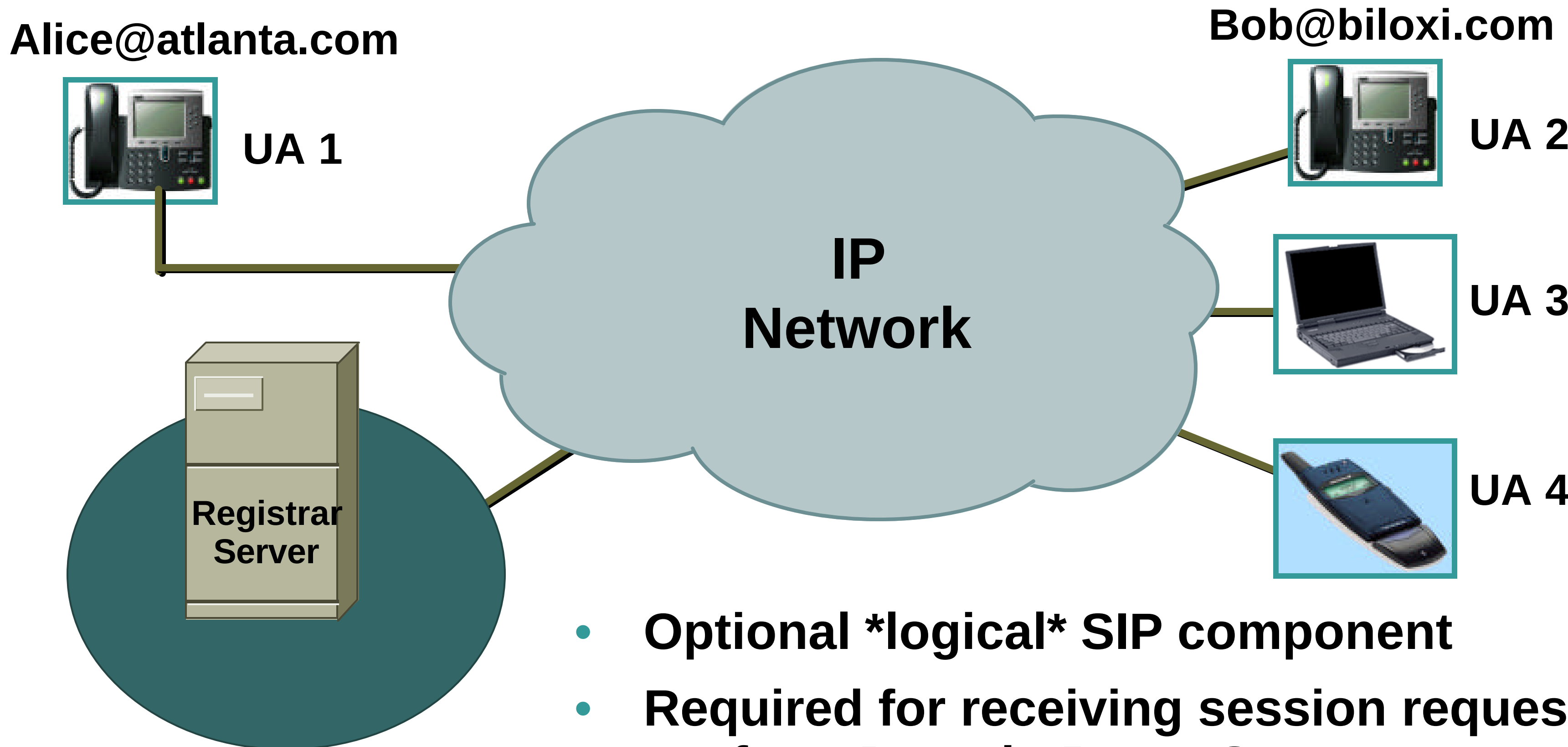
- **When referring to an “IP Network”**
- **This is what it will look more like**
- **SLAs could/will determine packet paths**

SIP Registrar Server



- Optional *logical* SIP component
- Required for receiving session requests from Domain Proxy Server
- Does not route SIP messages
- Also a mechanism for “Presence”

SIP Registrar Server



- Optional *logical* SIP component
- Required for receiving session requests from Domain Proxy Server
- Does **not route** SIP messages
- Also a mechanism for "Presence"

SIP Servers—Registrar Server

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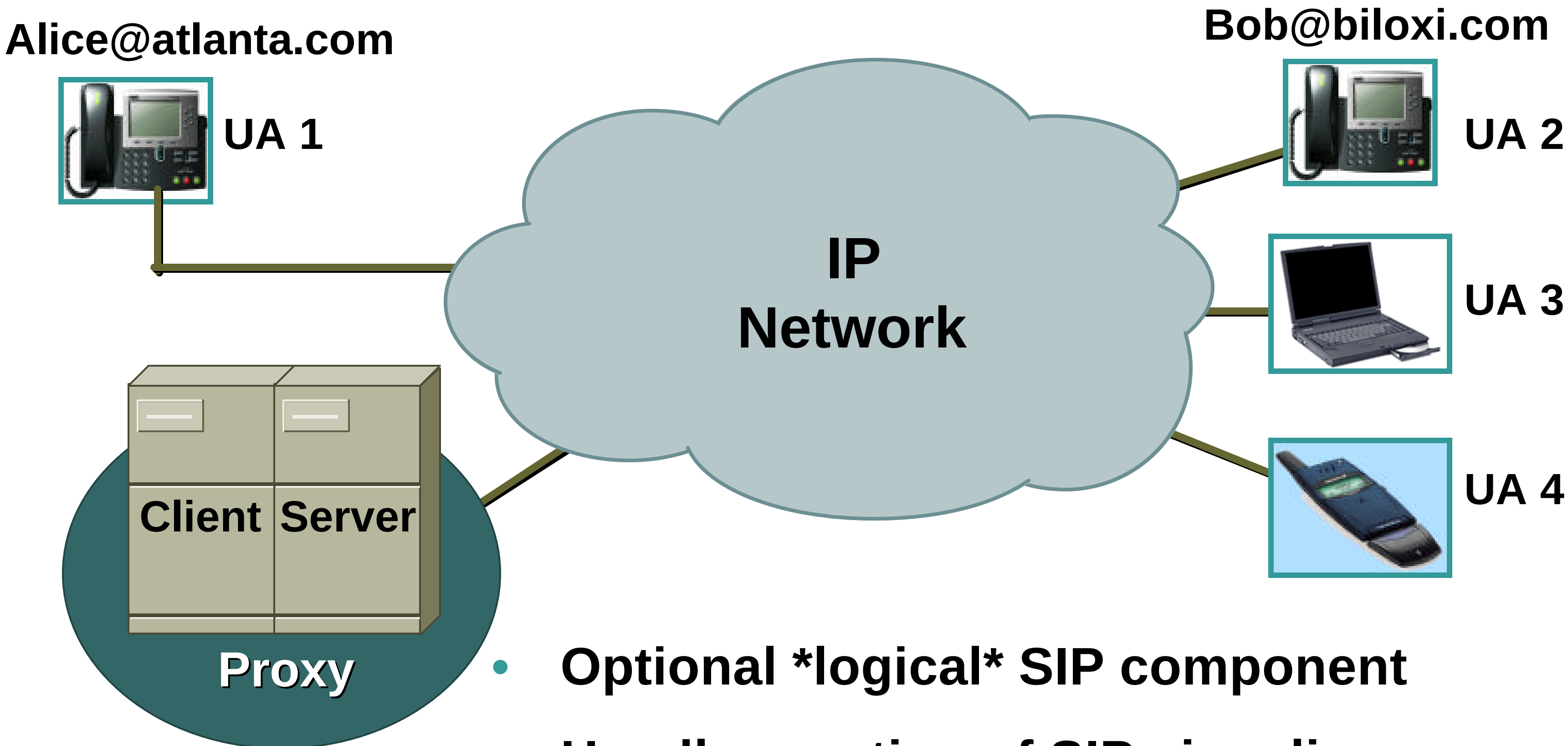
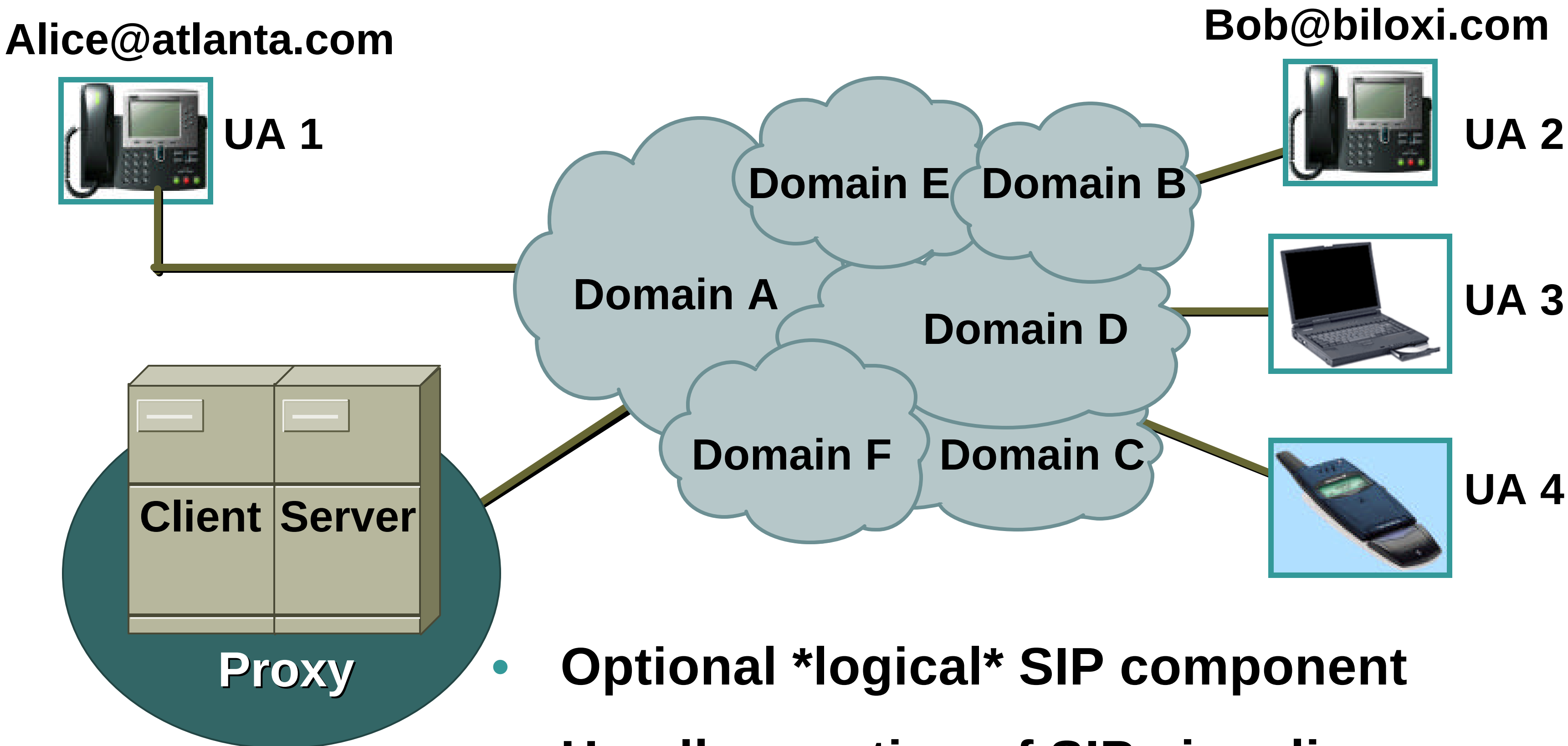
- An optional SIP *logical* component [can coexist with redirect and proxy server on same physical Computer]
- Binds the SIP URI of a user to the device known to that SIP domain
- Once a SIP UA is registered within a domain, the domain Proxy Server is able to route session requests to that user (agent) properly
- Not required to make a session request
- Registrar server is the device that handles SIP REGISTER messages from non-gateway SIP user agents
- Registrar server stores the values from a user agent REGISTER messages for location services

SIP Proxy Server



- Optional *logical* SIP component
- Handles routing of SIP signaling messages
- Can add/modify/remove Headers from SIP messages

SIP Proxy Server



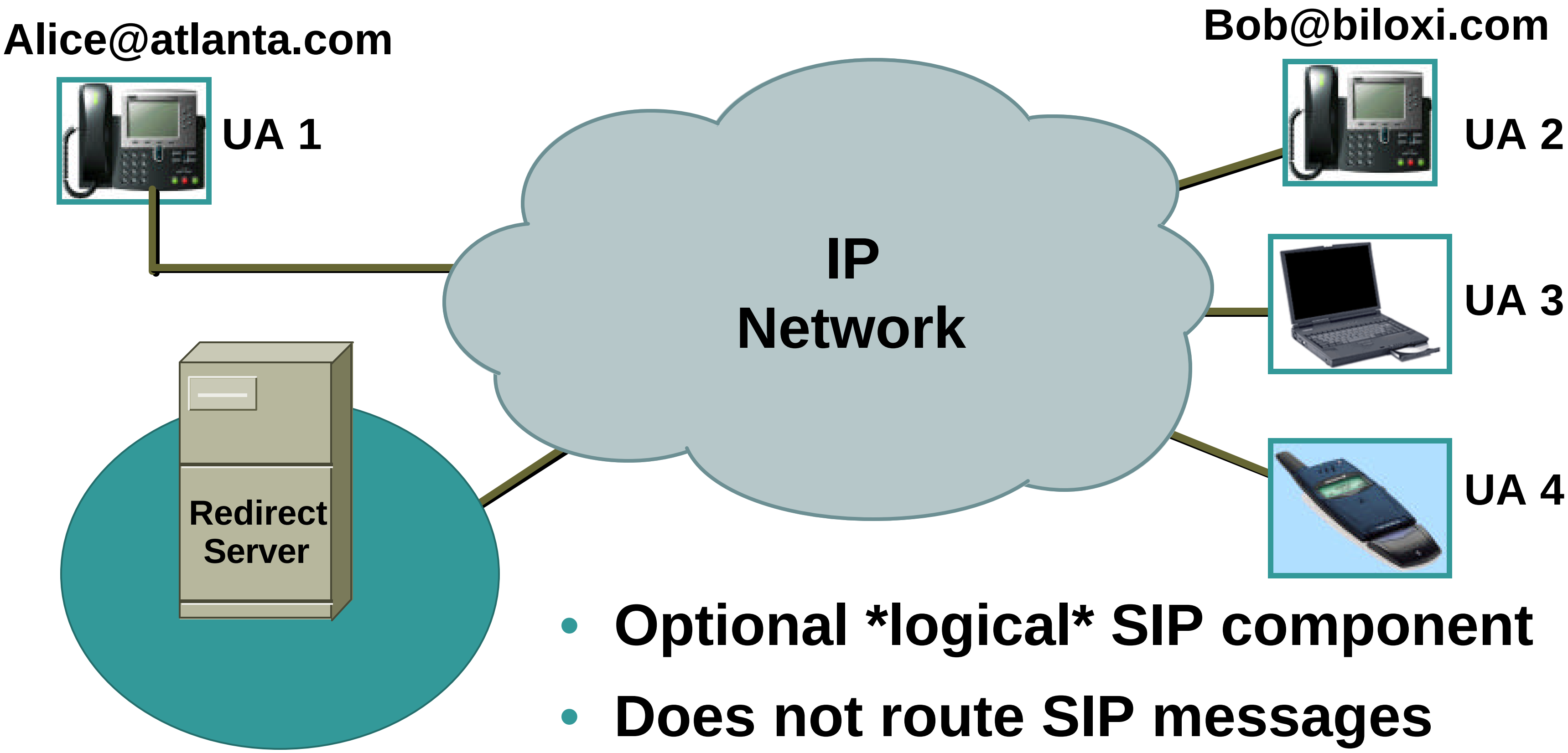
- Optional *logical* SIP component
- Handles routing of SIP signaling messages
- Can add/modify/remove Headers from SIP messages

SIP Servers—Proxy Server

- **An optional SIP *logical* component**
  - **can coexist with registrar and proxy server on same physical computer**
- **Handles the routing of SIP messages**
- **SIP proxies can insert and/or remove one or more headers from SIP messages;**

  “Record Route” and “Via” Headers, for example

SIP Redirect Server



- Optional *logical* SIP component
- Does not route SIP messages
- Returns a redirect to UAC for directed routing to the given “new” destination

SIP Servers—Redirect Server

- An optional SIP component
- A redirect server does not route messages
- The redirect server determines next destination of the now moved UA and returns a 3xx redirect message for where that new location is with the translated addresses in the Contact: header
- The originating UA initiates a new session using the information supplied from the redirect server

SIP Signaling Architecture

- Elements of SIP
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- SIP Summary

General SIP Packet format

IPv4 Header is 20 Bytes and Binary

Version	IHL	DSCP	ECN	Total Length
Identification			Flags	Fragment Offset
Time to Live	Protocol			Header Checksum
Source Address				
Destination Address				
Options				Padding

UDP Header is 8 Bytes and Binary (Layer 4 here could also be TCP or SCTP)

Source Port	Destination Port
Length	Checksum

SIP Header is Text-based and variable in length

SIP Headers in US-ASCII
(variable in length per header/per message)

SIP message body is also variable, but not always present (depending on the Message-type)

SIP messages *sometimes* have a message body
- a SIP message header indicates the type of body
- could be text, data, audio, video or something else

SIP— Sample Header

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;
branch=z9hG4bK776asdhds
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

(Alice’s SDP not shown – and is part of the message body)

SIP Addressing

- **Fully-Qualified Domain Names**

  sip:jdoe.cisco.com

- **SMTP-style Domain Names [RFC 2368]**

  sip:jdoe@cisco.com

- **E.164 style addresses [RFC 2806]**

  sip:14085551234@gateway.com; user=phone

  user=phone means this is a gateway

  (gateway.com is the FQDN of the egress IP gateway)

- **Mixed addresses**

  sip:14085551234@10.1.1.1; user=phone

  sip:jdoe@10.1.1.1

Session Description Protocol (SDP) RFC2327

- A session description protocol for multimedia connections
- Presents a set of parameters for a multimedia session

  Similar to H.245 in functionality
- Developed by IETF MMUSIC WG
- Simple/Flexible

  Text-based

  Extensible
- SIP Offer/Answer Model is

  RFC 3264

"Lines" below are in order

- v = protocol version
- o = owner/creator and session identifier
- s = session name
- c = connection information – not required if included in all media
- k = encryption keys
- t = time the session is active
- m = media description and transport address
- a = (zero or more) media attributes lines

SIP Message Body for multimedia

- An SDP message body for voice only

- Audio
- UDP port # 49172
- Codecs supported: G.711

```
v=0
o=alice 2890844526 2890844526 IN IP4 atlanta.com
c=IN IP4 10.1.3.33
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

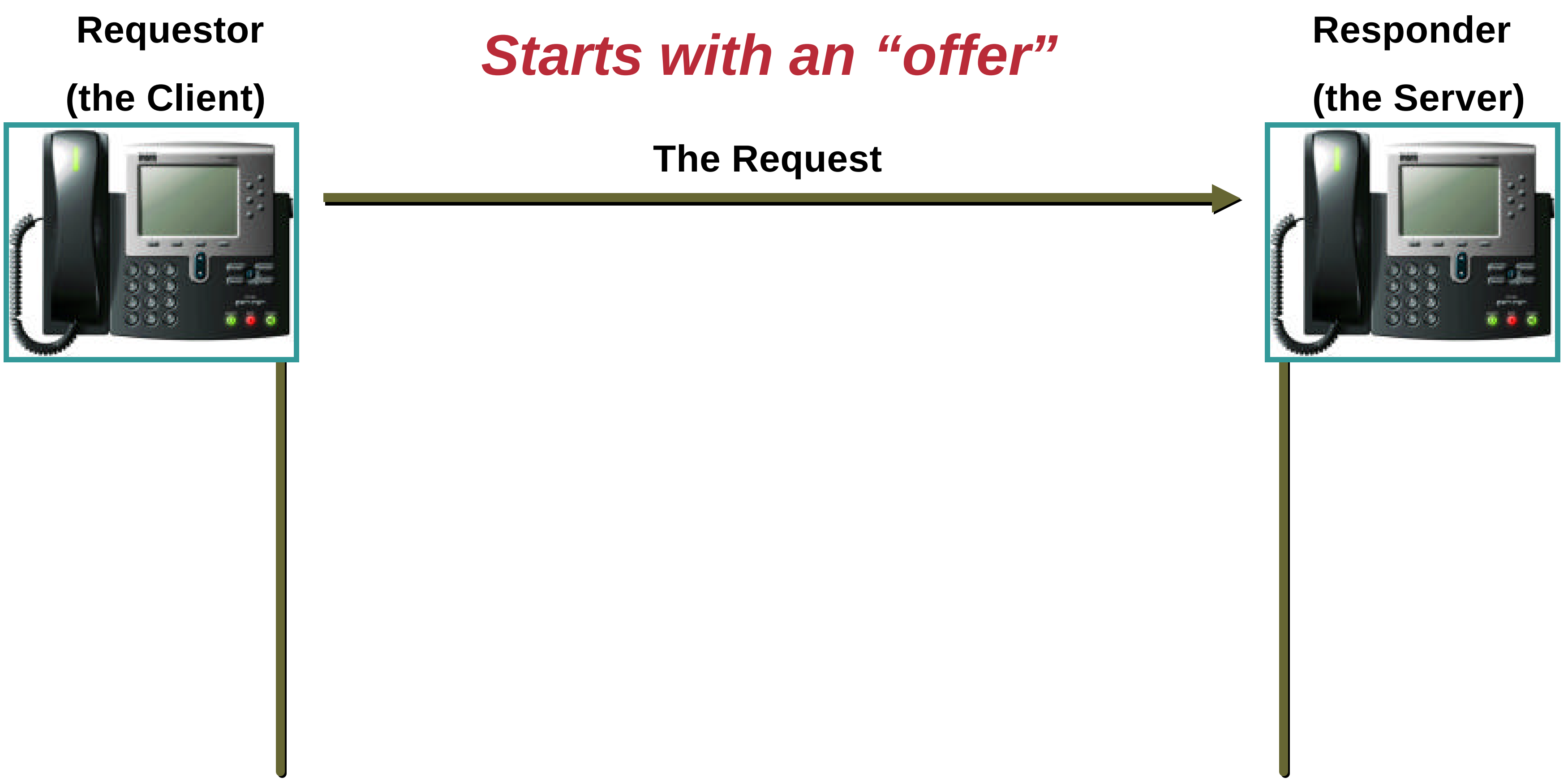
- An SDP message body for voice and video

- Audio
- UDP port # 49172
- Codecs supported: G.711

- Video
- UDP port # 51172
- Codecs supported: H.261, H.263

```
v=0
o=alice 2890844526 2890844526 IN IP4 atlanta.com
c=IN IP4 10.1.3.33
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
m=video 51172 RTP/AVP 31 34
a=rtpmap:31 H.261/90000
a=rtpmap:34 H.263/90000
```

SIP is a Request/Response Protocol



SIP Methods (which are Requests) from RFC 3261

- INVITE—A user or service is being invited to participate in a multimedia session
- ACK—Confirms that a client has received a final response to an INVITE request
- BYE—Terminates an existing session; can be sent by any user agent (in a multiparty session)
- CANCEL—Cancels pending requests; does not terminate sessions that have been accepted
- OPTIONS—Queries the capabilities of servers
- REGISTER—Registers the user agent with the registrar server of a domain

Additional SIP Request Methods

- INFO (RFC 2976)
- PRACK (RFC 3262)
- SUBSCRIBE and NOTIFY (RFC 3265)
- UPDATE (RFC 3311)
- MESSAGE (RFC 3428)
- REFER (RFC 3515)
- PUBLISH (RFC 3903)

SIP is a Request/Response Protocol

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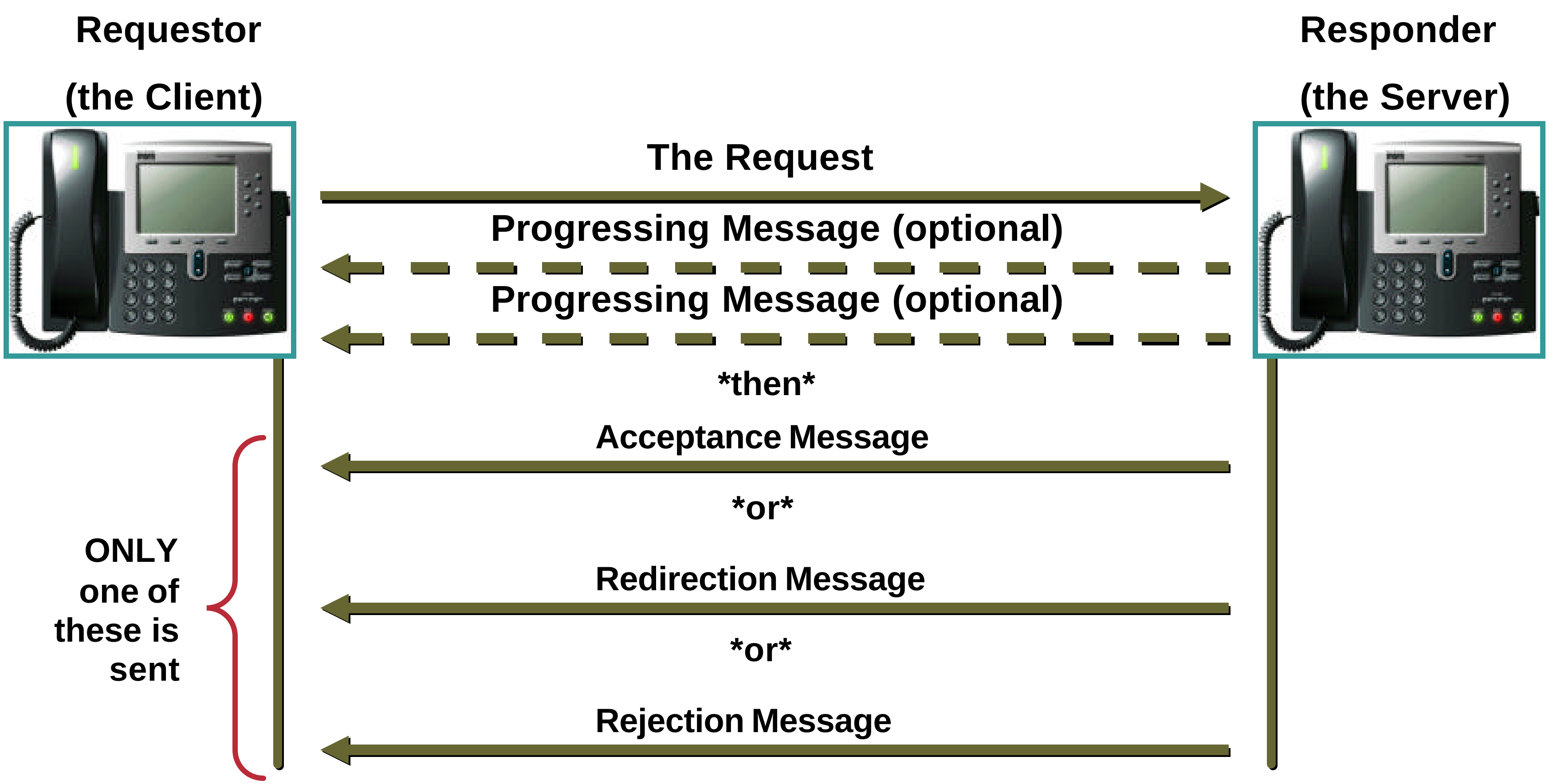
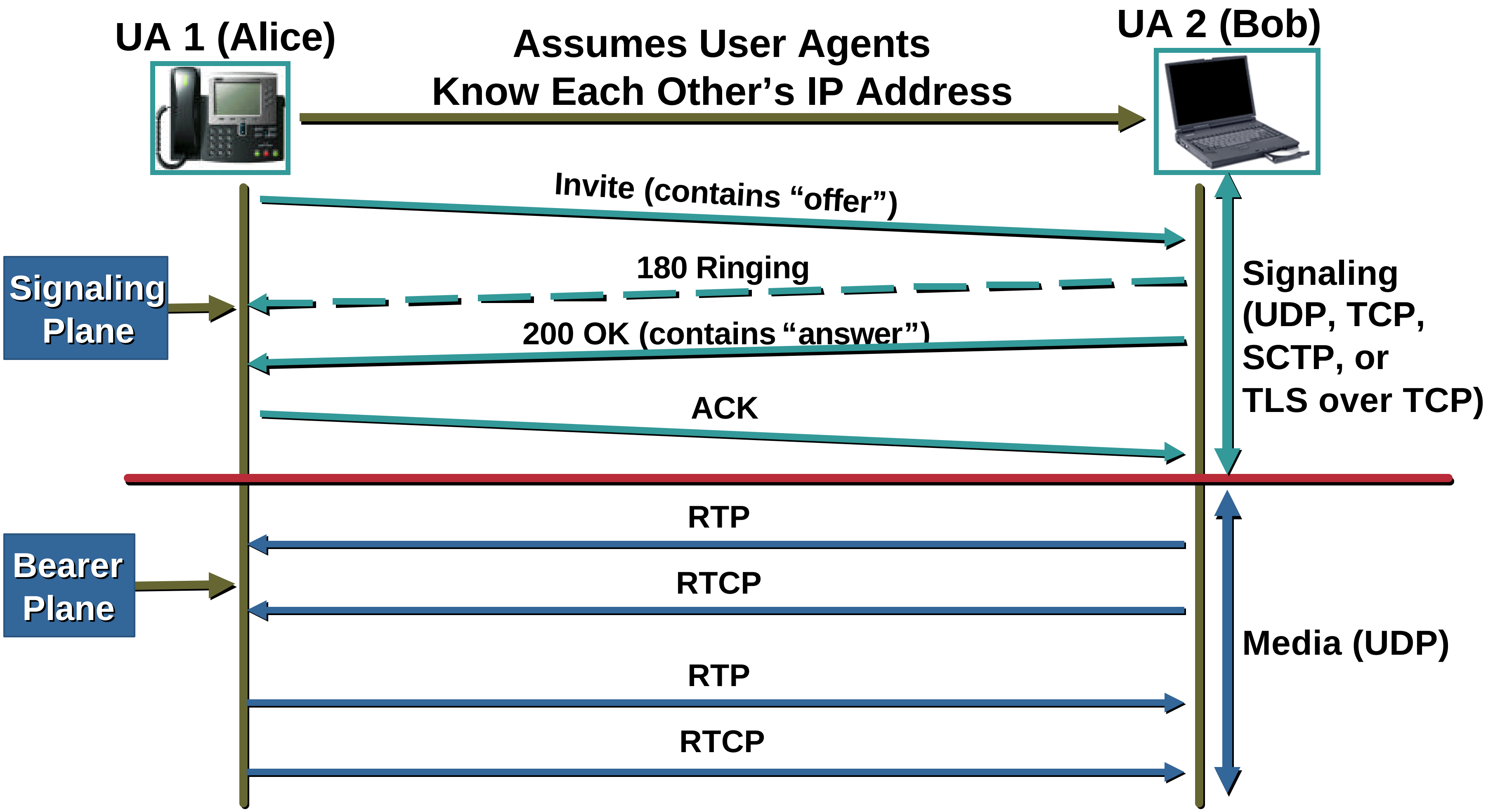


Expects an "answer"

SIP Responses (which are Replies)

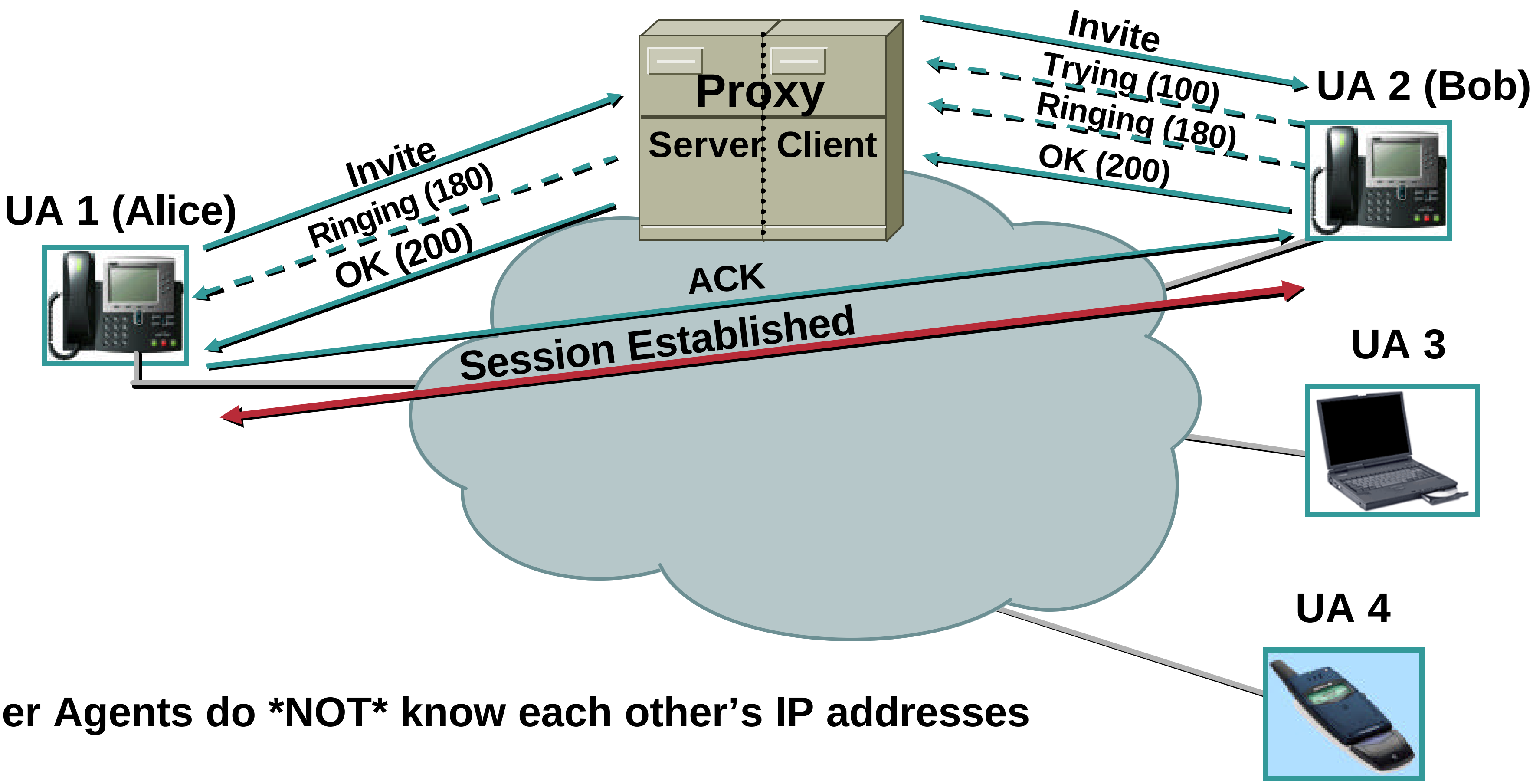
	Description	Examples
1xx	Informational – Request received, continuing to process request.	100 Trying 180 Ringing 181 Call is Being Forwarded 183 Session Progressing
2xx	Success – Action was successfully received, understood and accepted.	200 OK 202 Acceptable
3xx	Redirection – Another SIP Element needs to be contacted in order to complete the request.	300 Multiple Choices 301 Moved Permanently 302 Moved Temporarily
4xx	Client Error – Request contains bad syntax or cannot be fulfilled at this server.	401 Unauthorized 406 Not Acceptable 407 Proxy Authentication Required 486 Busy Here 487 Request Terminated 488 Not Acceptable Here
5xx	Server Error – Server failed to fulfill an apparently valid request.	502 Bad Gateway 503 Service Unavailable
6xx	Global Failure – Request is invalid at any server.	600 Busy Everywhere 603 Decline

SIP Endpoint-to-Endpoint Signaling without a Server



3 Mandatory Packets for Establishment Handshake INVITE/200 OK/ACK

SIP Proxy Server



User Agents do *NOT* know each other's IP addresses

SIP Call Flow with Proxy

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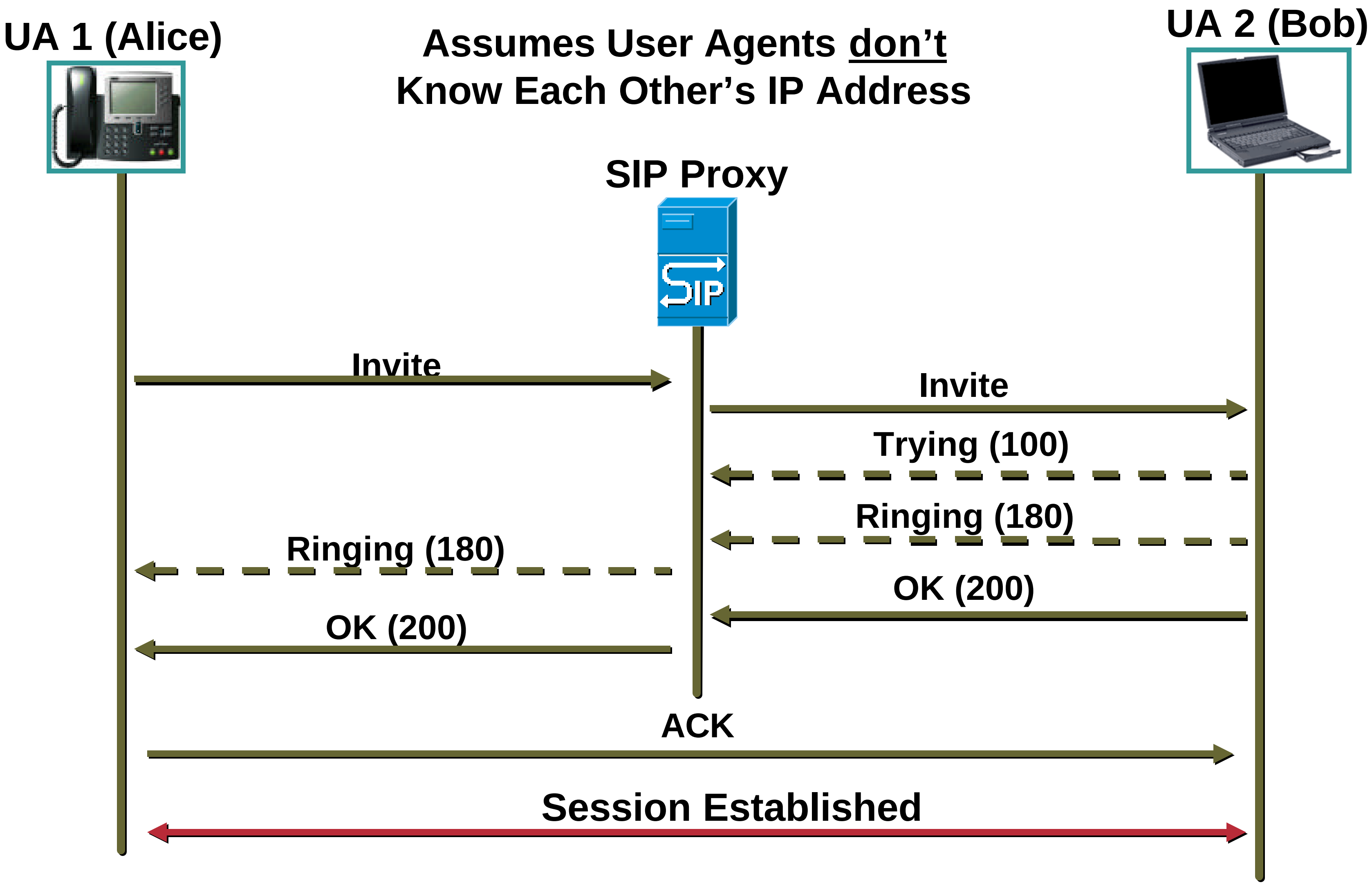

Stateless, Stateful & "Really" Stateful Proxy

Transaction Stateless

- The proxy server forwards all messages and responses without maintaining any state

Transaction Stateful

- A Proxy Server that receives a SIP Request retains state of that transaction until that Server receives a Final Response (meaning a 2XX, 3XX, 4XX, 5XX or 6XX Response). Transaction Stateful has no knowledge of a session Update Request (UPDATE), a Transfer Request (REFER) or of a Termination Request (BYE)

Dialog Stateful

- When Record Route Header is utilized by a Proxy during the first SIP Request to ensure all remaining messages traverse that Proxy; this applies to each proxy that is in the signaling path between UAs

Record-Route Header

- Optional SIP message header
- *Can* be inserted by any and all Proxies viewing a SIP message
- Routes all SIP requests and responses for that Call-ID through that proxy
  - a new Call-ID Request is treated independently
- Initial Response Message uses a <u>Route</u> Header taken from the <u>Record-Route</u> header values
- Useful for billing, call control, CALEA

Call Flow with Record Route (and Route)

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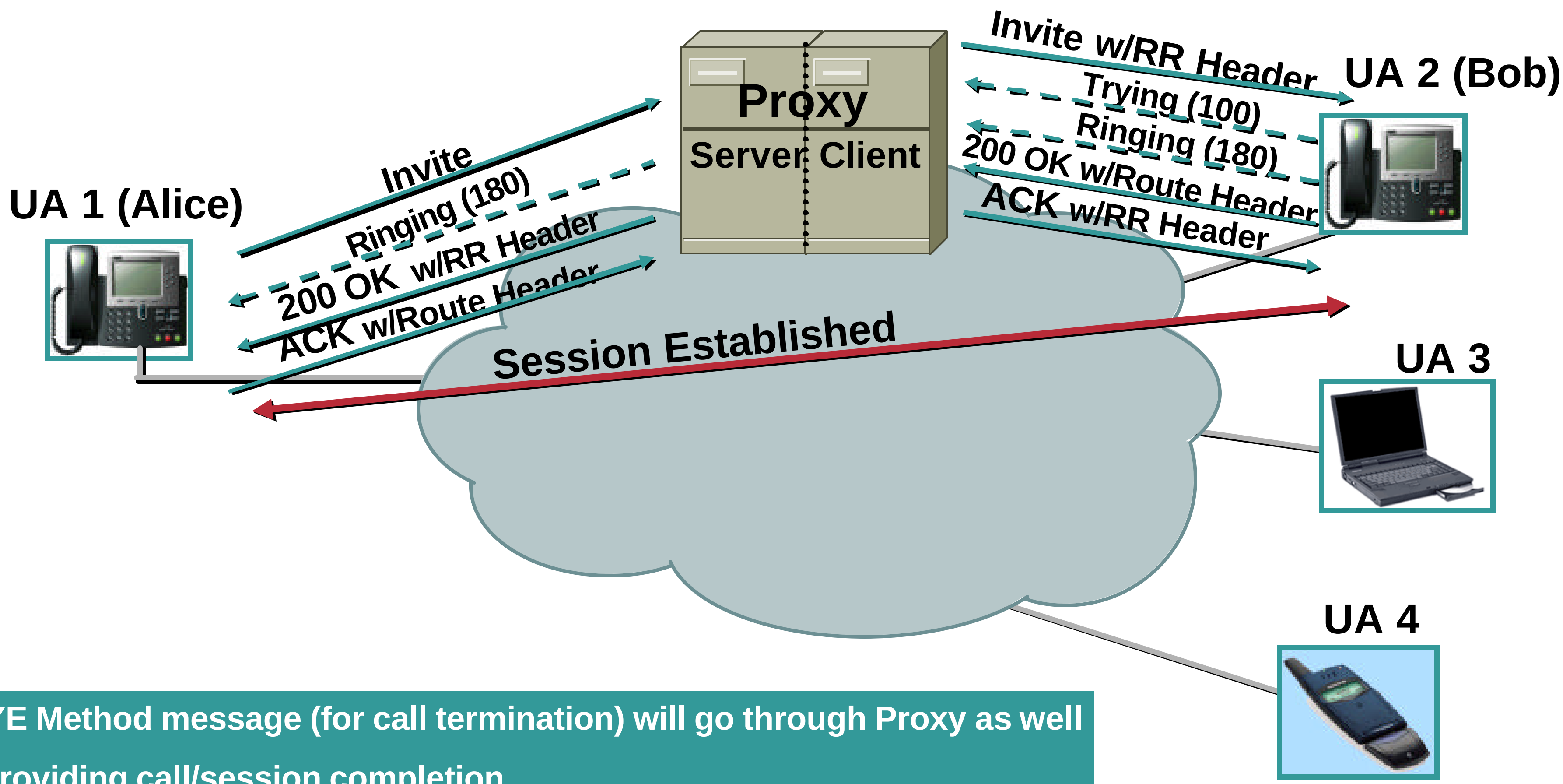
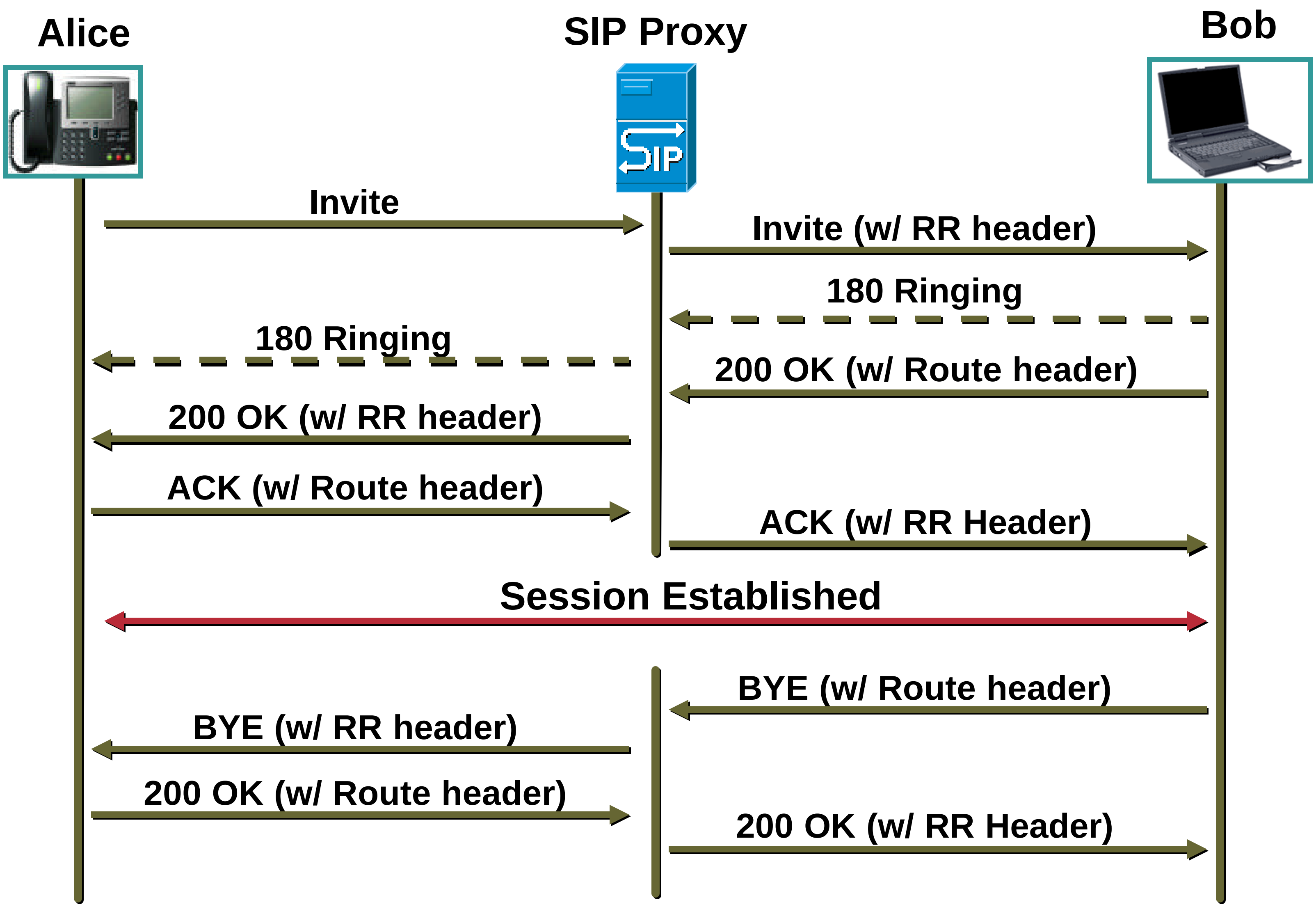


BYE Method message (for call termination) will go through Proxy as well

- providing call/session completion

SIP Call Flow with Proxy and Record Route



SIP Call Flow w/ 2 Proxies and Record Route

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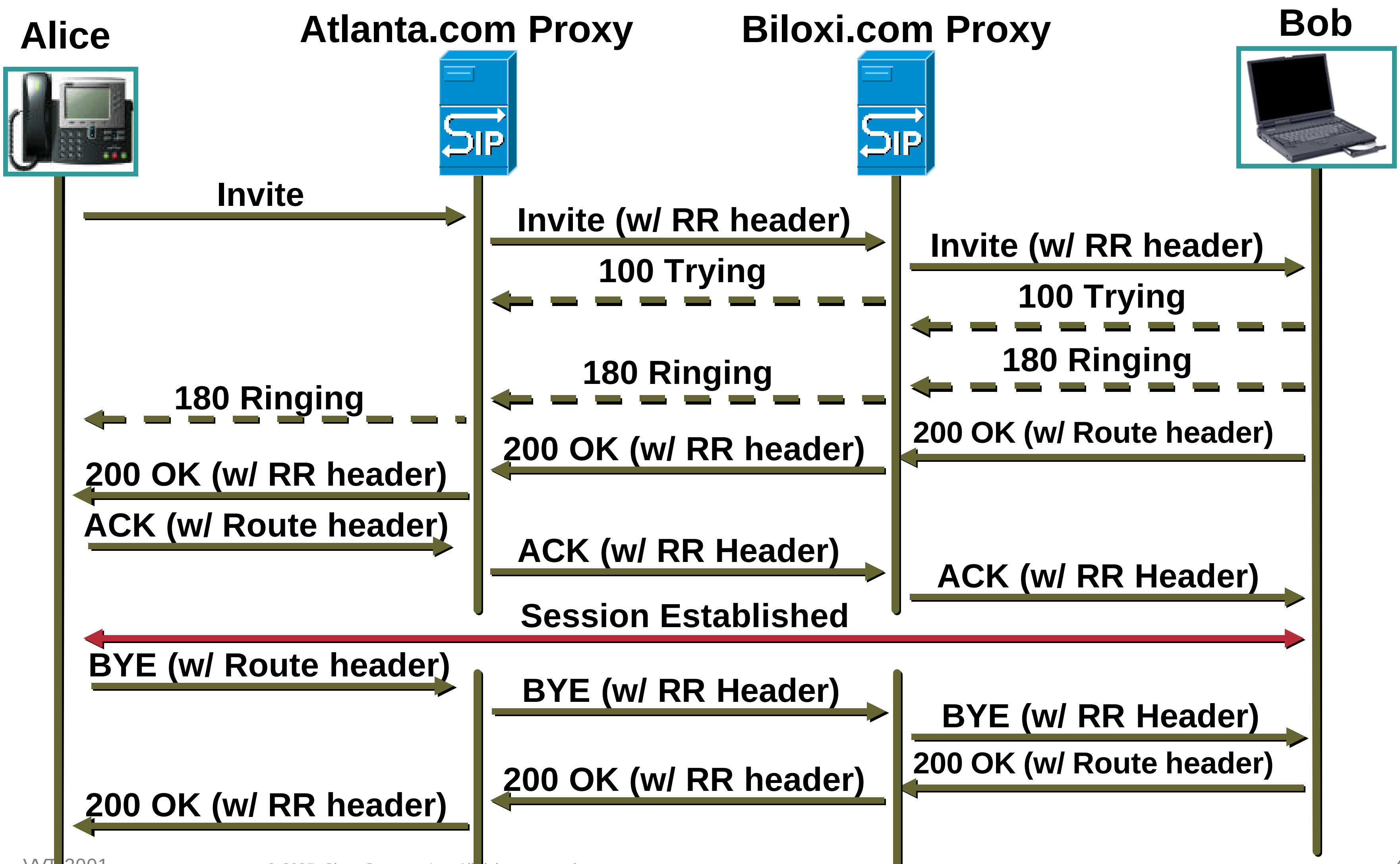
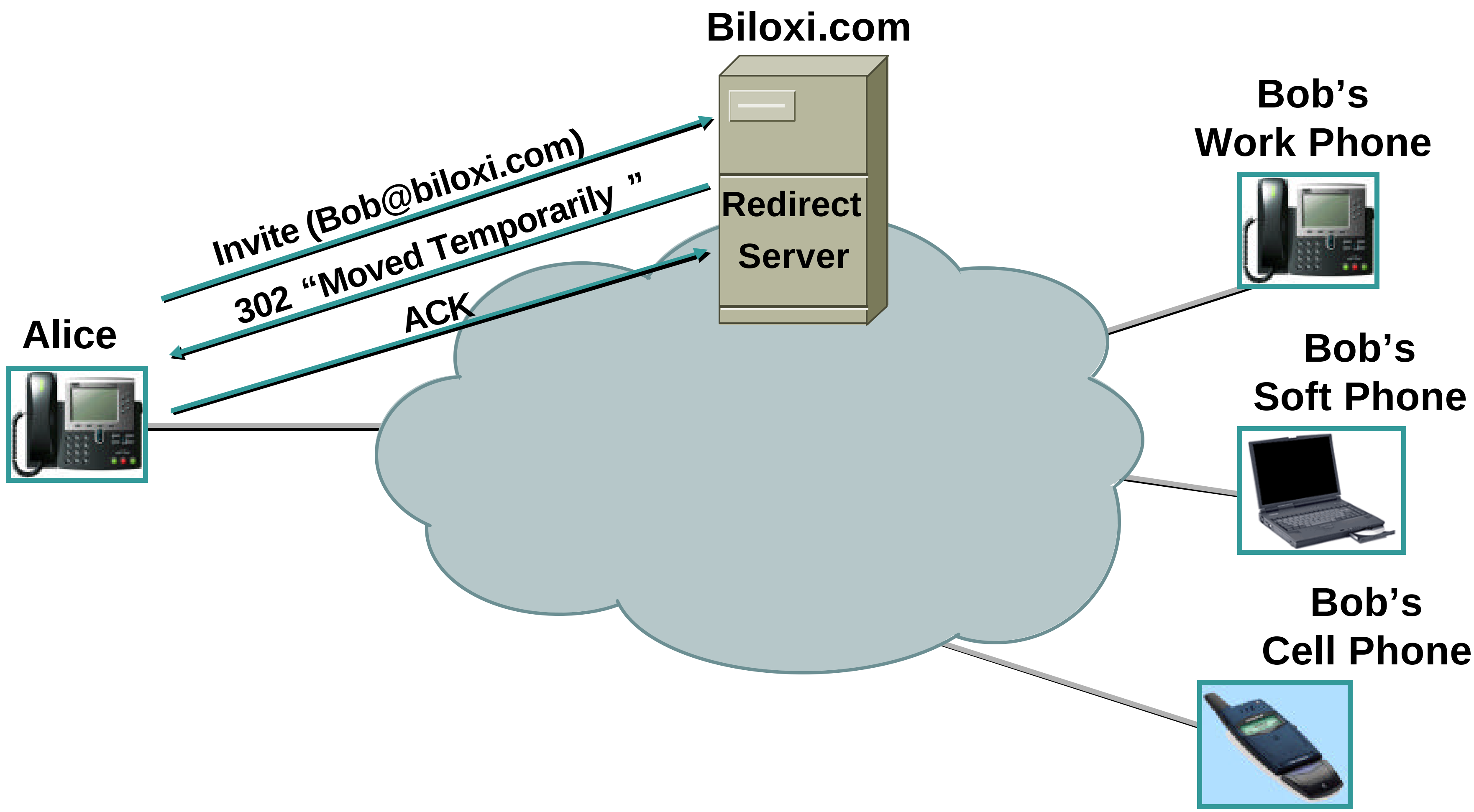


SIP Redirect Server



SIP Redirect Server

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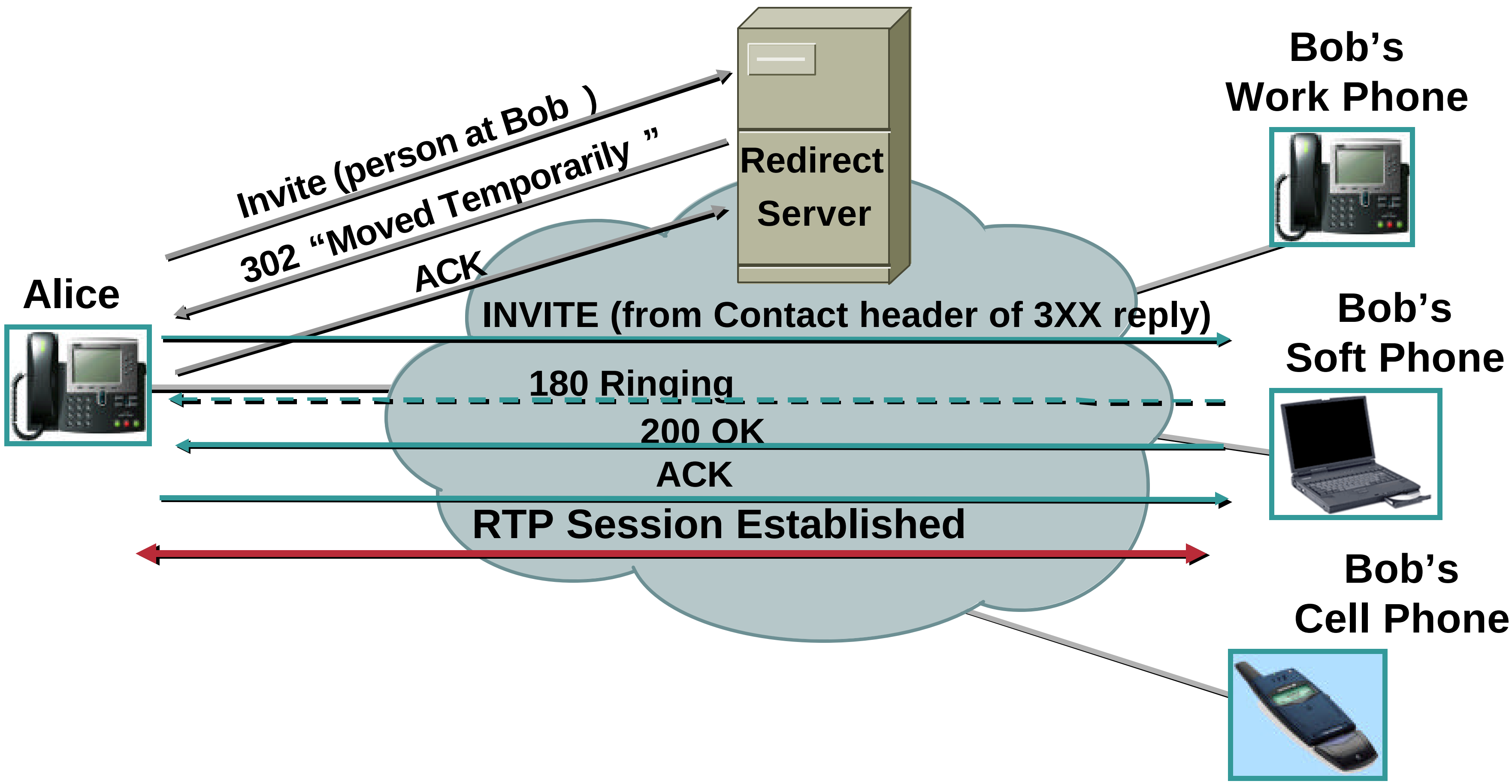
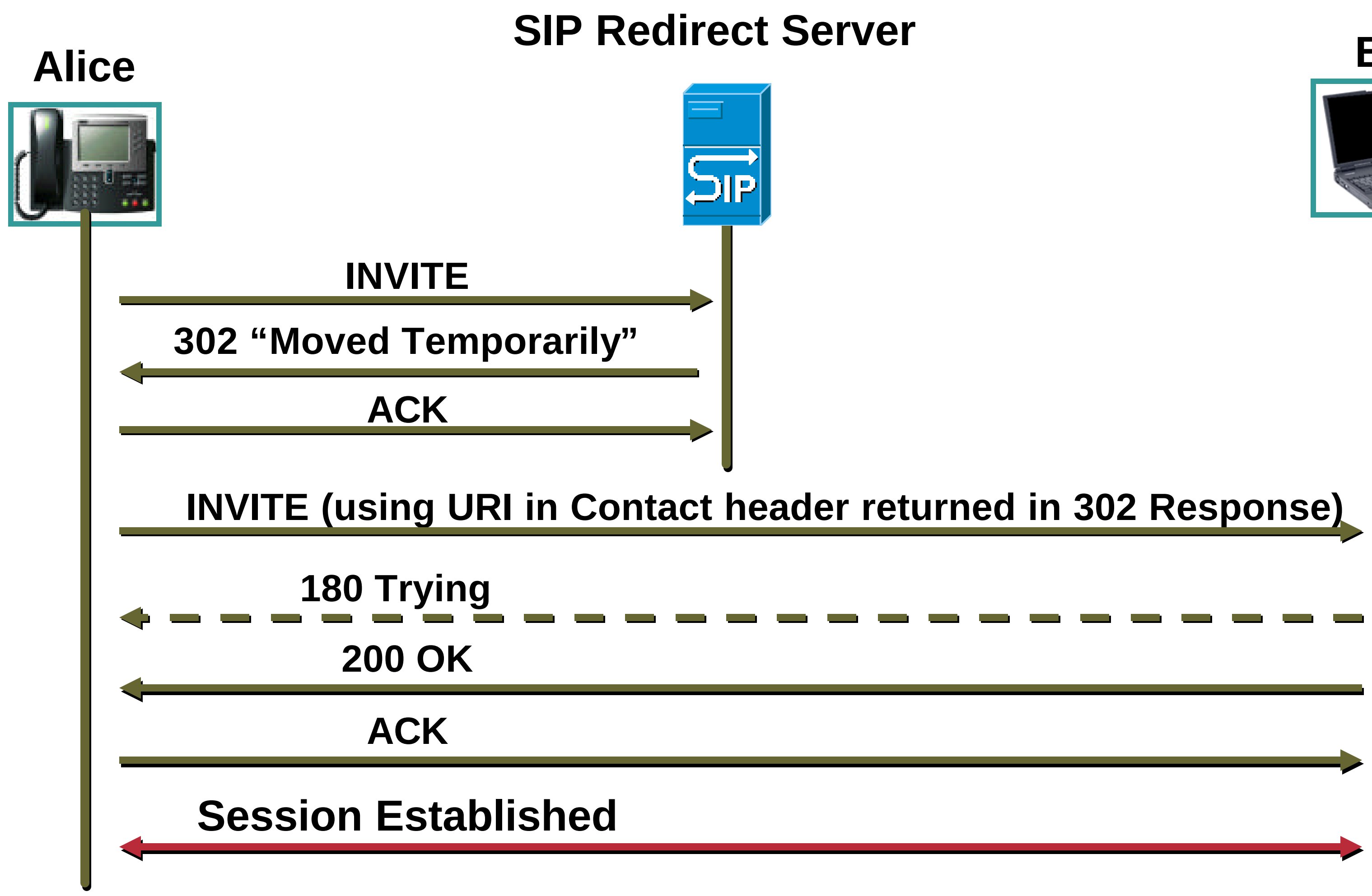


SIP Call Flow with Redirect



SIP Redirecting at the User Agent Server

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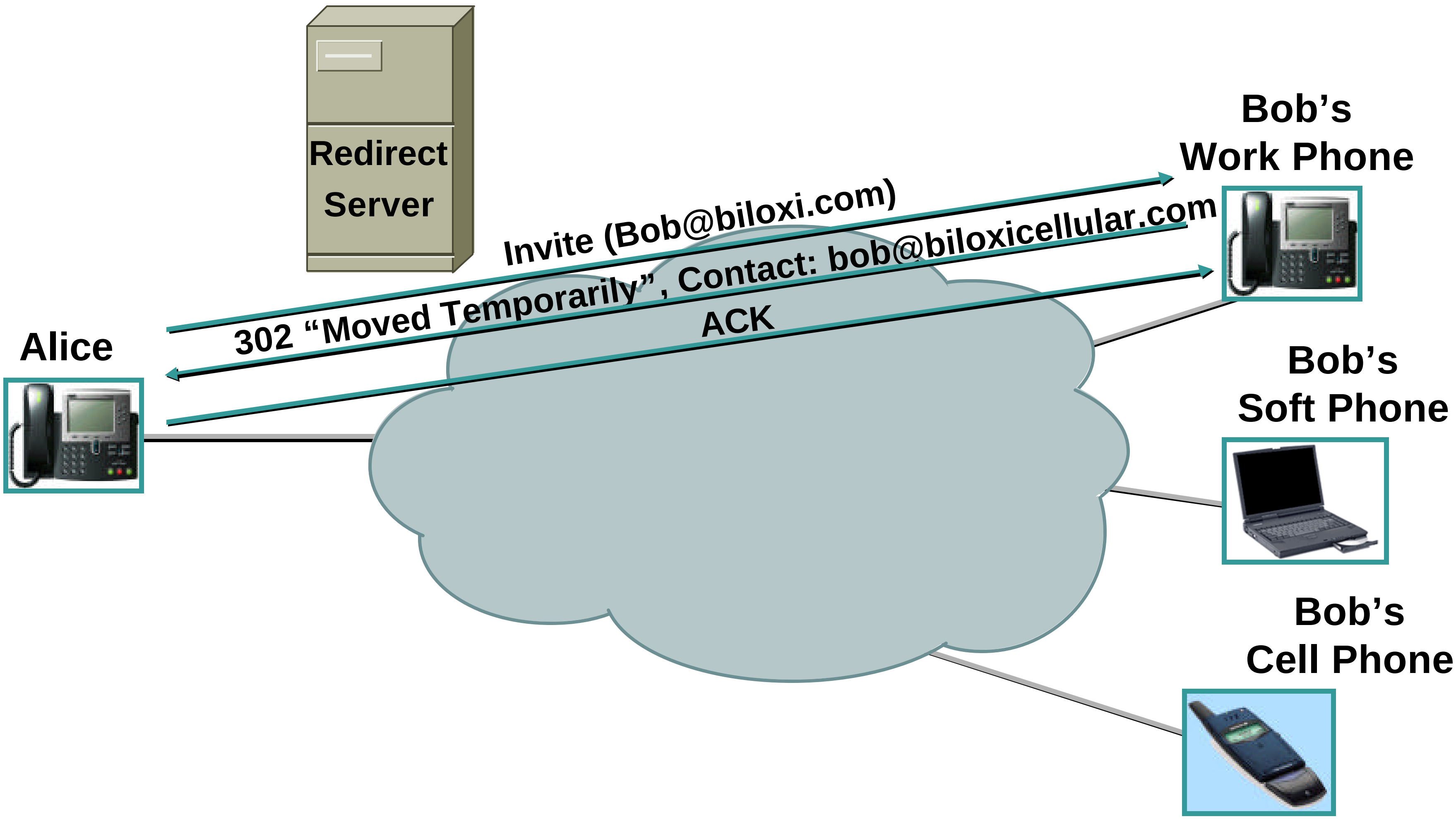
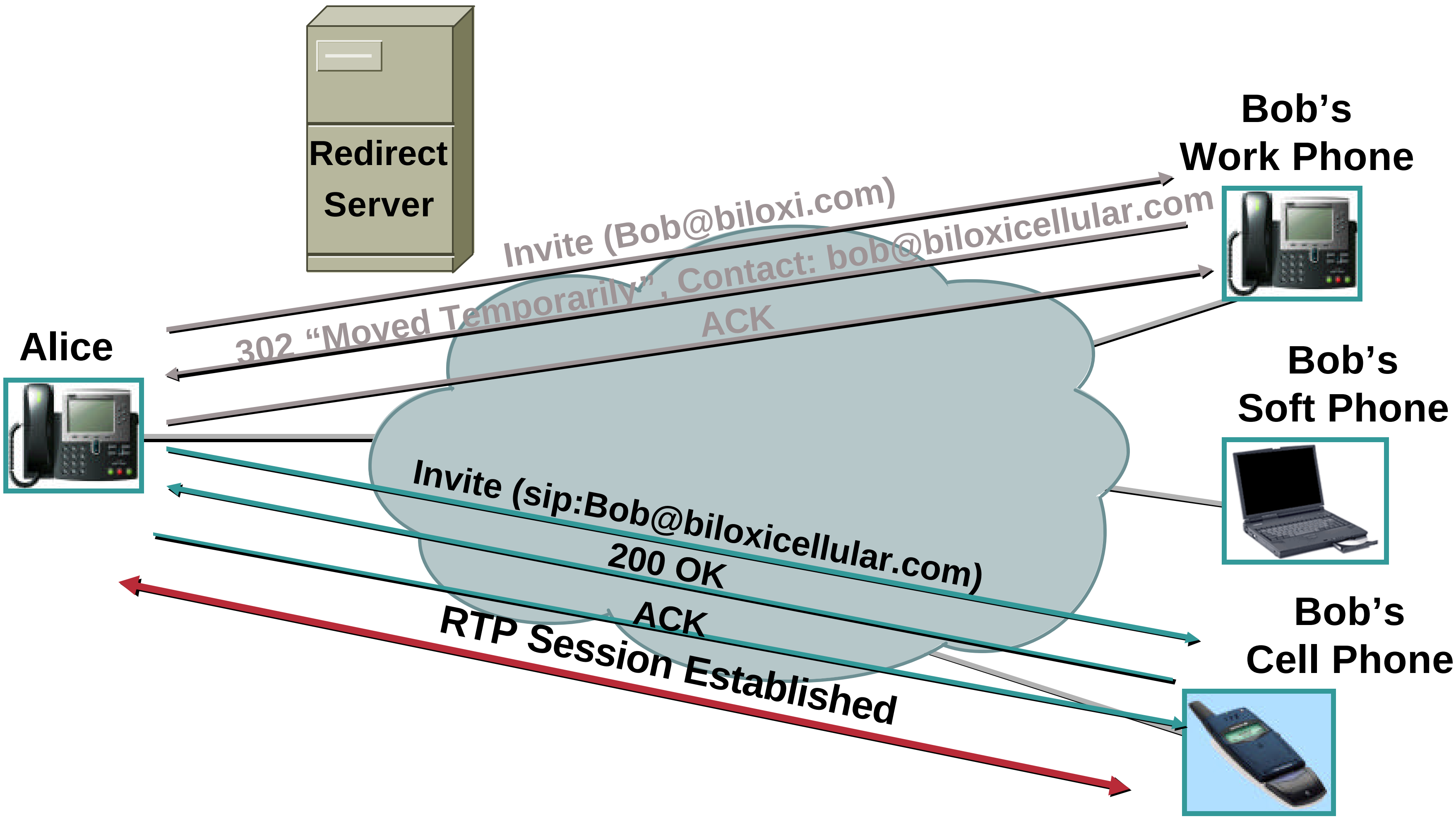


SIP Redirecting at the User Agent Server



SIP Signaling Architecture

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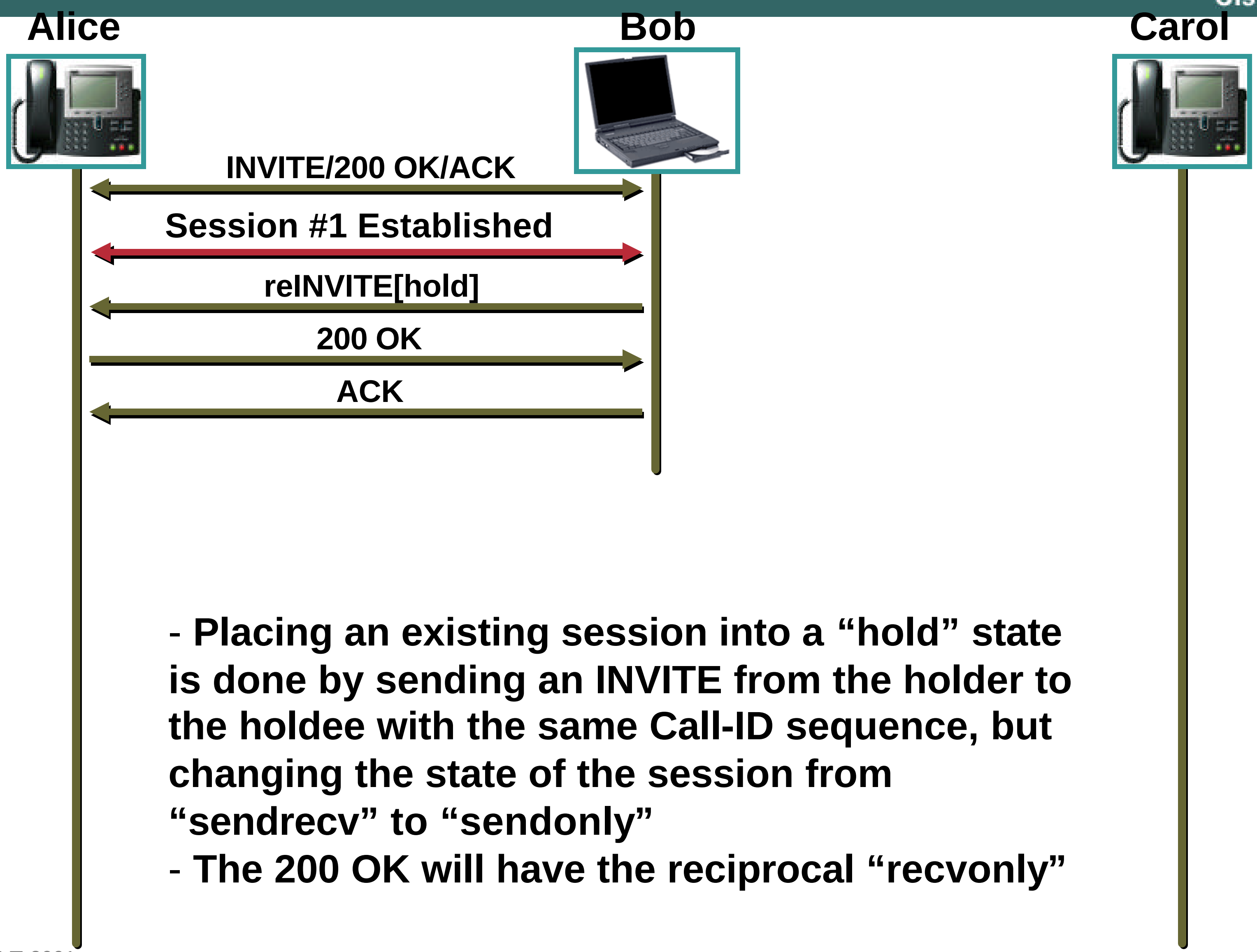
SIP Example: Call Hold



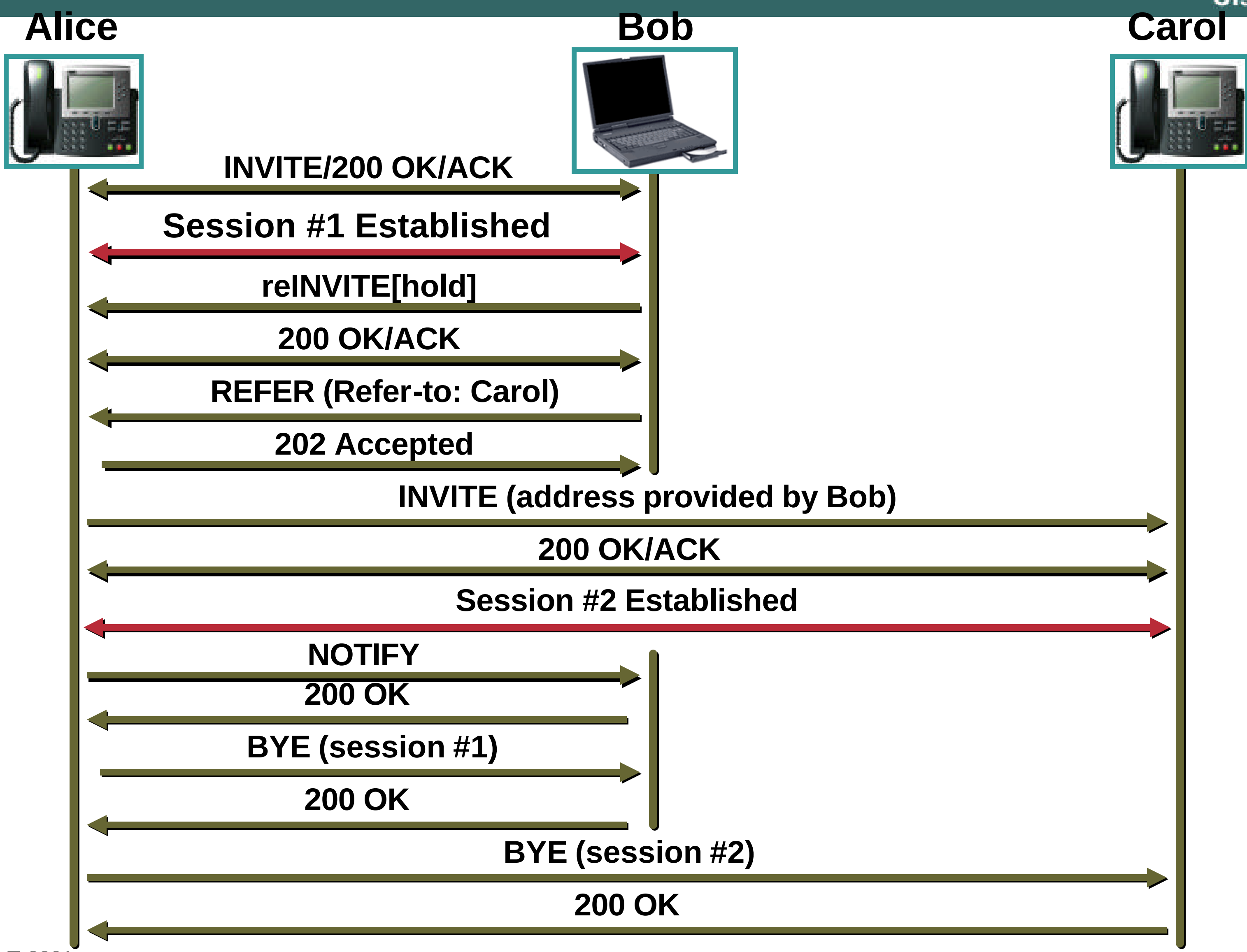
- Placing an existing session into a “hold” state is done by sending an INVITE from the holder to the holdee with the same Call-ID sequence, but changing the state of the session from “sendrecv” to “sendonly”
- The 200 OK will have the reciprocal “recvonly”

Call Transfer and 3-way Conferencing

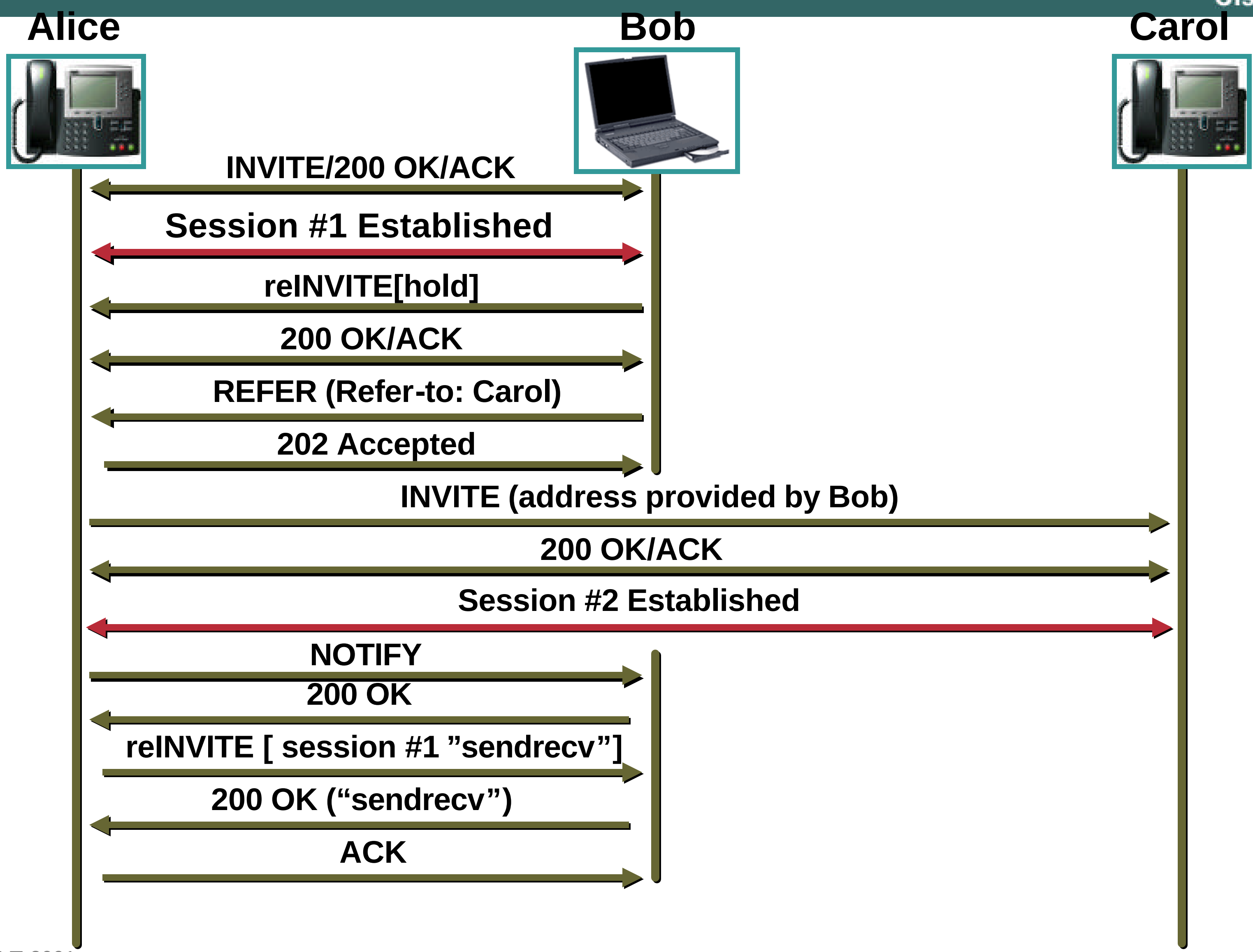
Several Ways to Perform Call Transfer to another UA:

- 3rd Party Call Control - reINVITE the transferee with the address of the transfer target
- REFER method - Refer the transferee to contact the transfer target
- Mixing media streams within a UA
- Using conferencing extensions

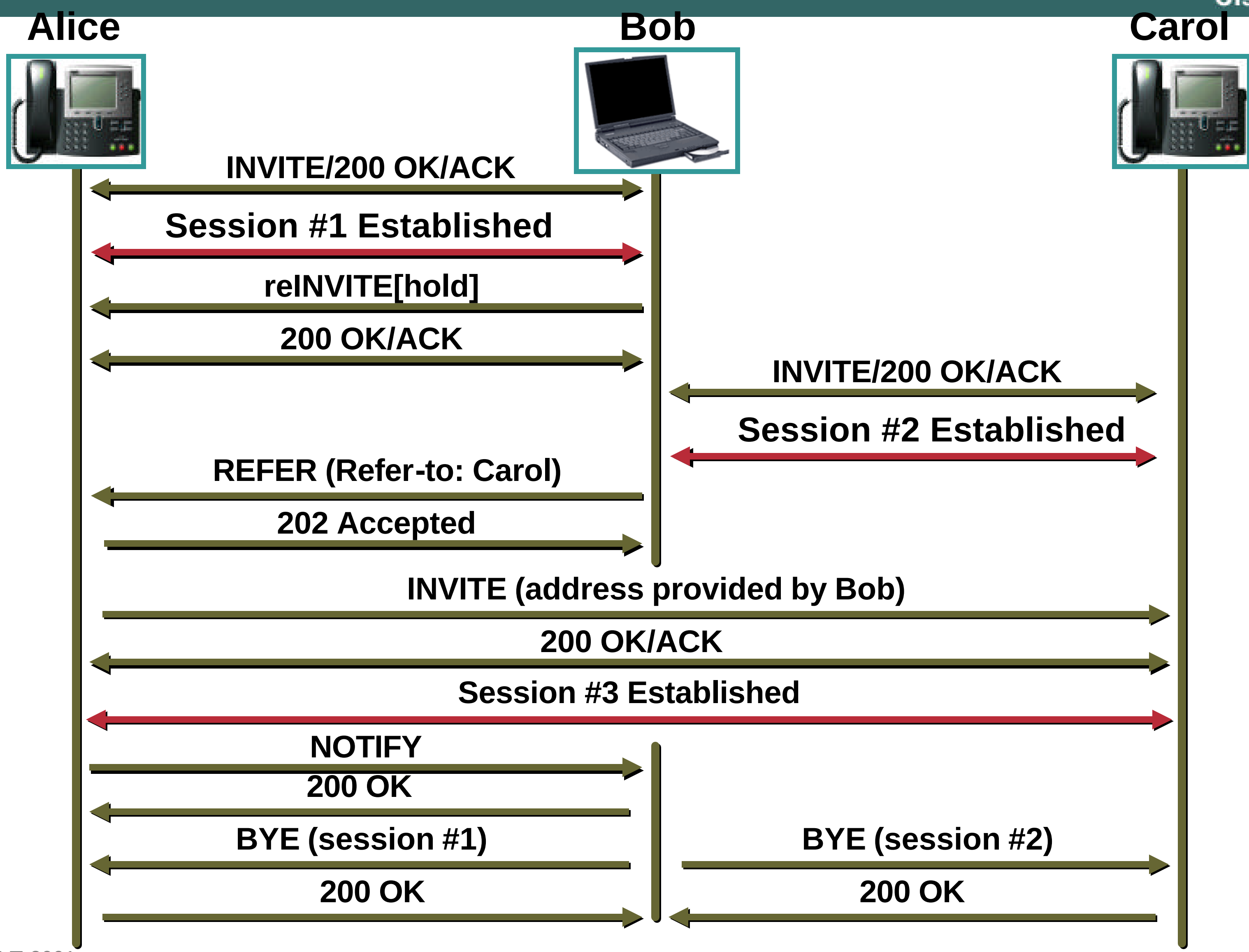
SIP Example: Call Transfer Example



SIP Example: 3-Way Conference via Alice



SIP Example: Call Transfer II



SIP Example: 3-Way Conference via Bob

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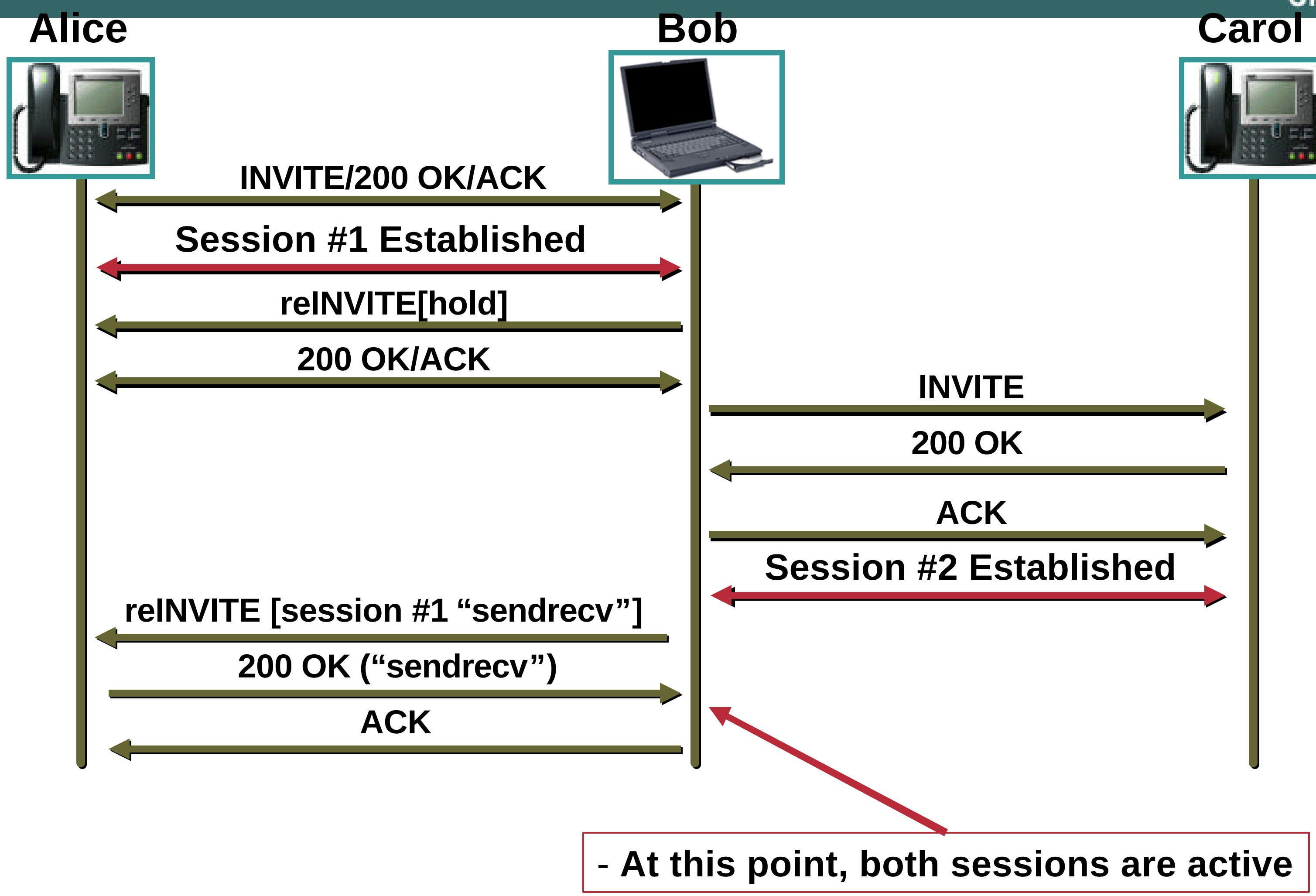
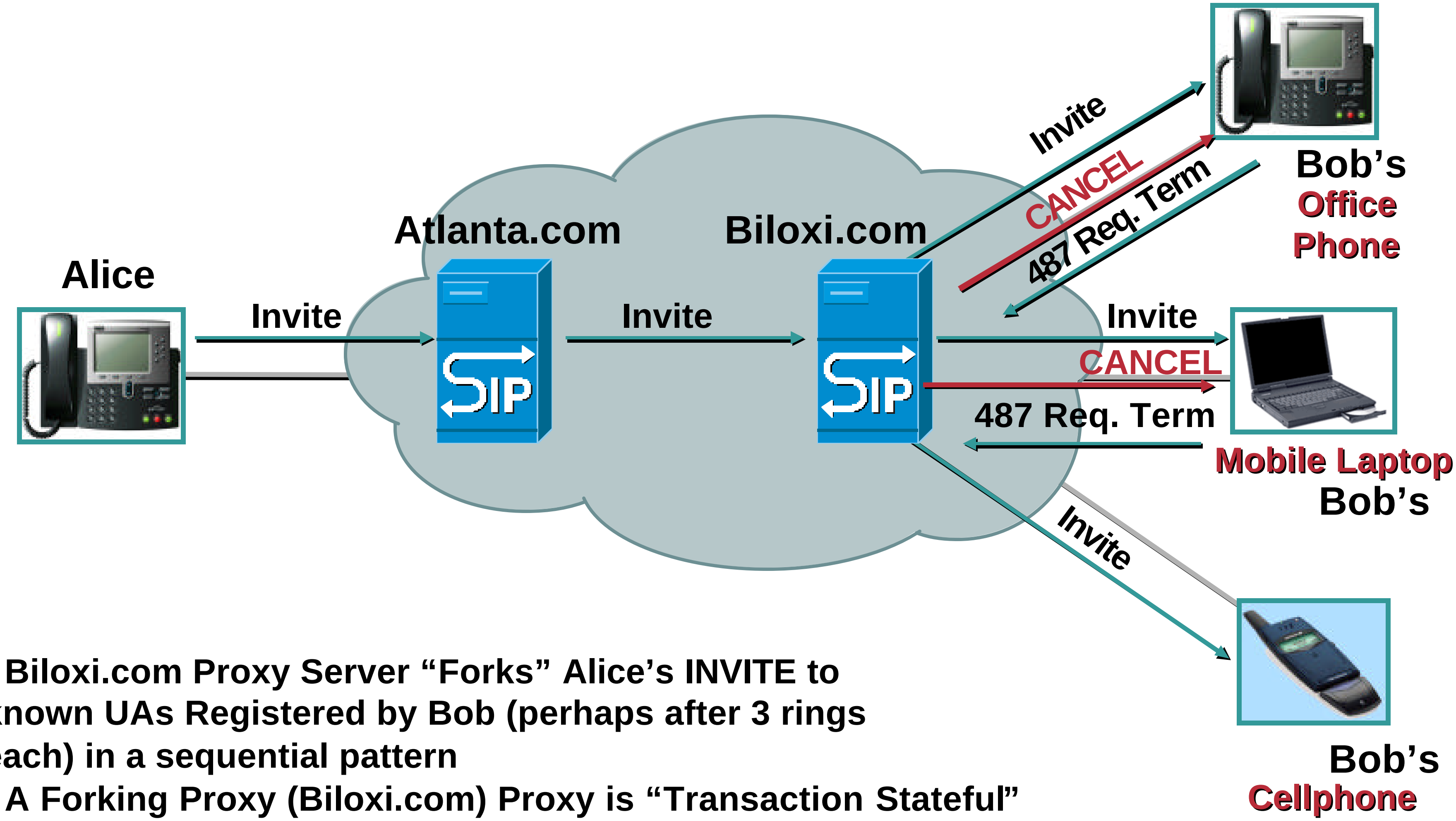
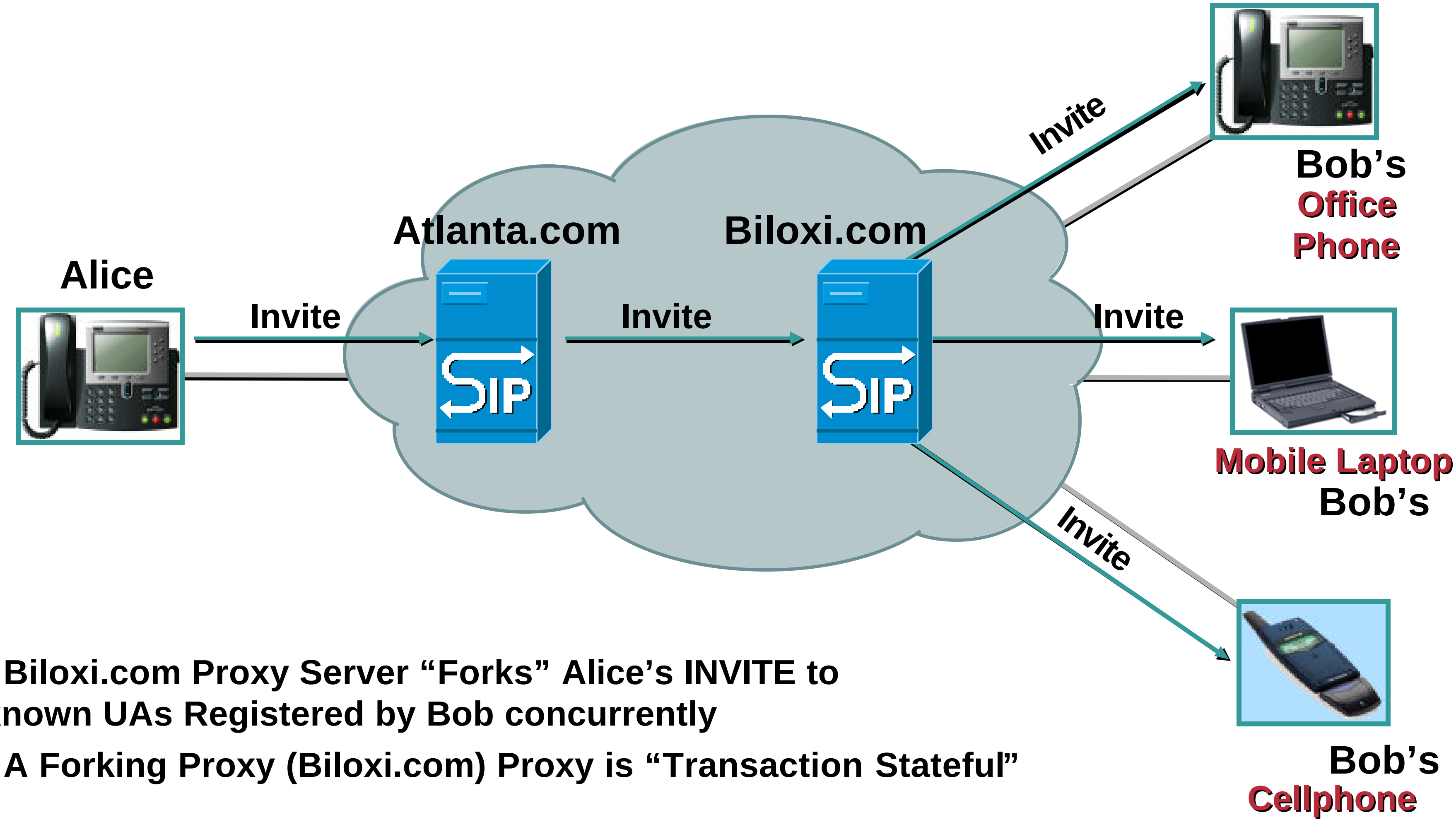


Call Forking (Sequentially)



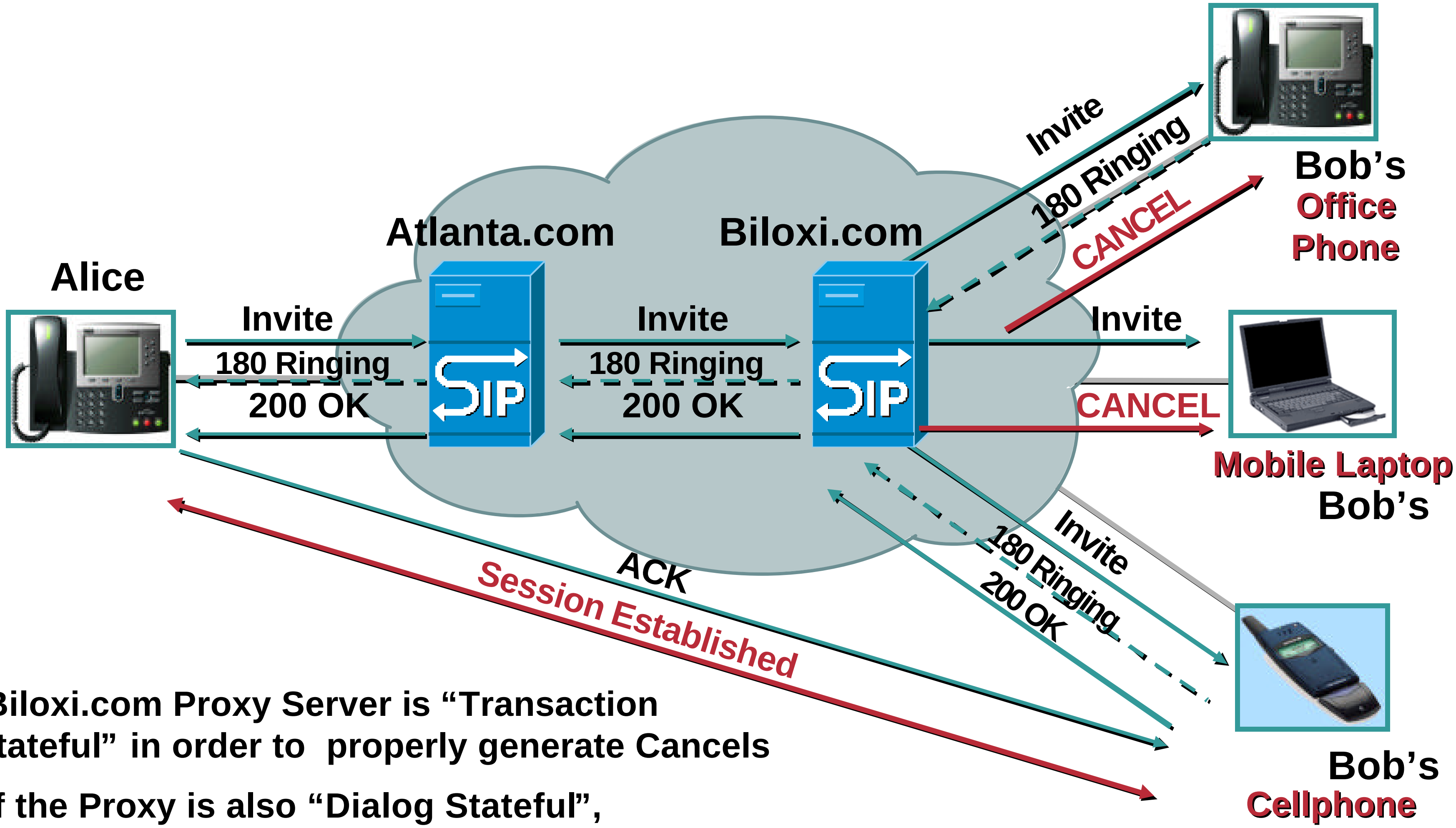
- Biloxi.com Proxy Server "Forks" Alice's INVITE to known UAs Registered by Bob (perhaps after 3 rings each) in a sequential pattern
- A Forking Proxy (Biloxi.com) Proxy is "Transaction Stateful"
- "branch" values are different per forked INVITE

Call Forking (Concurrently)



- Biloxi.com Proxy Server "Forks" Alice's INVITE to known UAs Registered by Bob concurrently
- A Forking Proxy (Biloxi.com) Proxy is "Transaction Stateful"

Call Forking Flow



-Biloxi.com Proxy Server is "Transaction Stateful" in order to properly generate Cancels

-If the Proxy is also "Dialog Stateful", it will receive BYE Request from Alice or Bob

Call Forking

- Proxy forks original INVITE to multiple user agents
- Forks can be sequential, or concurrent
- “Branch” values within the Via header are different for each forked INVITE
- Session established to first user agent to respond with 200 OK
- CANCEL sent to non-respondent user agents within forking procedure
- Proxy MUST be at least “Transaction Stateful” for Forking
- Proxy could be controlled by Callee’s User Profile

Third Party Call Control

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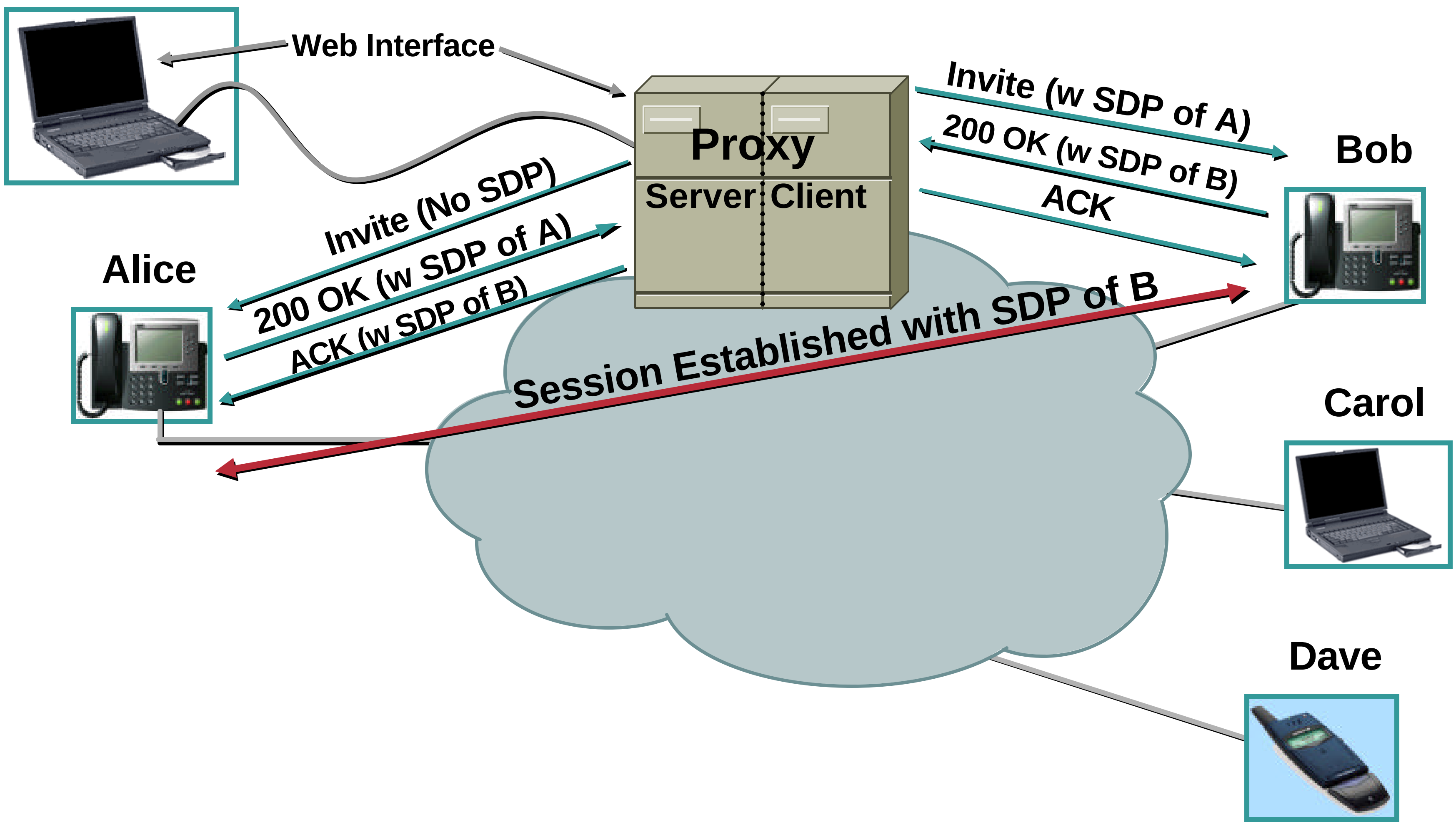

Third Party Call Control

- Allows calls to be set-up by an independent third party
- Uses INVITE method, without SDP in initial INVITE
- Facilitates additional capabilities, i.e. scheduling conference calls

Session Set-up Example Between Domains

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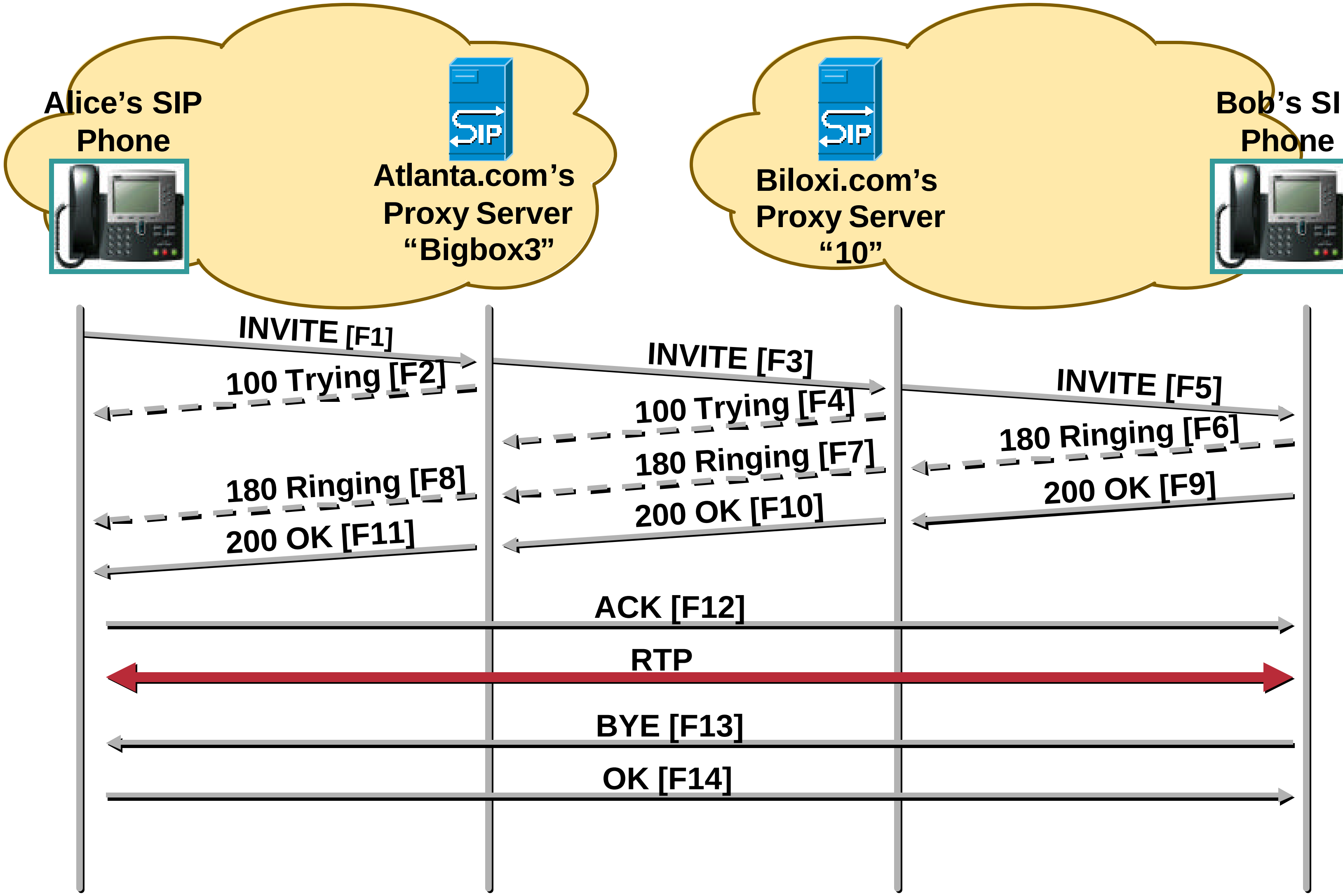


SIP—Headers Explained

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bK776asdhds
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

- Message body goes down here
- Content-Length Header indicates one is present

SIP—Headers Explained

Method = Command   Request URI   SIP Protocol Version

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bK776asdhds
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

- Message body goes down here
- Content-Length Header indicates one is present

SIP—Headers Explained

List of all SIP devices to be included in the path

Mandatory header in all SIP Requests

'branch' is for identifying forking

Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bK776asdhds
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

- Message body goes down here
- Content-Length Header indicates one is present

SIP—Headers Explained

Mandatory header in all SIP Requests except INFO

Maximum Number of SIP Server hops permissible in signal path

Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

- Message body goes down here
- Content-Length Header indicates one is present

SIP—Headers Explained

Mandatory header in all SIP Requests

Destination for the SIP Message (but isn’t used for routing message)

To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

- Message body goes down here
- Content-Length Header indicates one is present

SIP—Headers Explained

Mandatory header in all SIP Requests

The Originator of the SIP Message

Pseudo Random String (used for id)

From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

- Message body goes down here
- Content-Length Header indicates one is present

SIP—Headers Explained

Mandatory header in all SIP Requests

Globally Unique Call Identifier within the Domain for session

Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

- Message body goes down here
- Content-Length Header indicates one is present

SIP—Headers Explained

Mandatory header in all SIP Requests

Incremental Sequence number of Method used from this UA

CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

- Message body goes down here
- Content-Length Header indicates one is present

SIP—Headers Explained

Direct route to Originator with Fully Qualified Domain Name or IP address

Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

- Message body goes down here
- Content-Length Header indicates one is present

SIP—Headers Explained

Description of the message body (if and when there is one)

Content-Type: application/sdp
Content-Length: 142

- Message body goes down here
- Content-Length Header indicates one is present

SIP—Headers Explained

Expected header in all SIP Requests

Content Length in Octets of Message Body

Content-Length: 142

- Message body goes down here
- Content-Length Header indicates one is present

SIP—Headers Explained

Cisco.com

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bK776asdhds
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

- Message body goes down here
- Content-Length Header indicates one is present

Various SIP Headers

INVITE sip:23198@10.17.207.91:5060 SIP/2.0 method name: Request URI line starts

SIP/2.0 200 OK status-line for initial Response line

Expires: when this SIP message will expire (in seconds)

Via: a list of all SIP devices that *are* to be in the signaling path; also includes Proxy "branch" parameter which identifies this transaction

Max-Forwards: serves to limit the number of hops, decremented by one per hop (server)

To: The destination of the SIP message

From: the originator of the SIP message; also contains a "tag" parameter which is a pseudorandom string for identification purposes

Call-ID: the globally unique Call Identifier

CSeq: a sequence of messages in that method

Contact: specifies a direct route to the originator with a FQDN (preferred) or an IP address

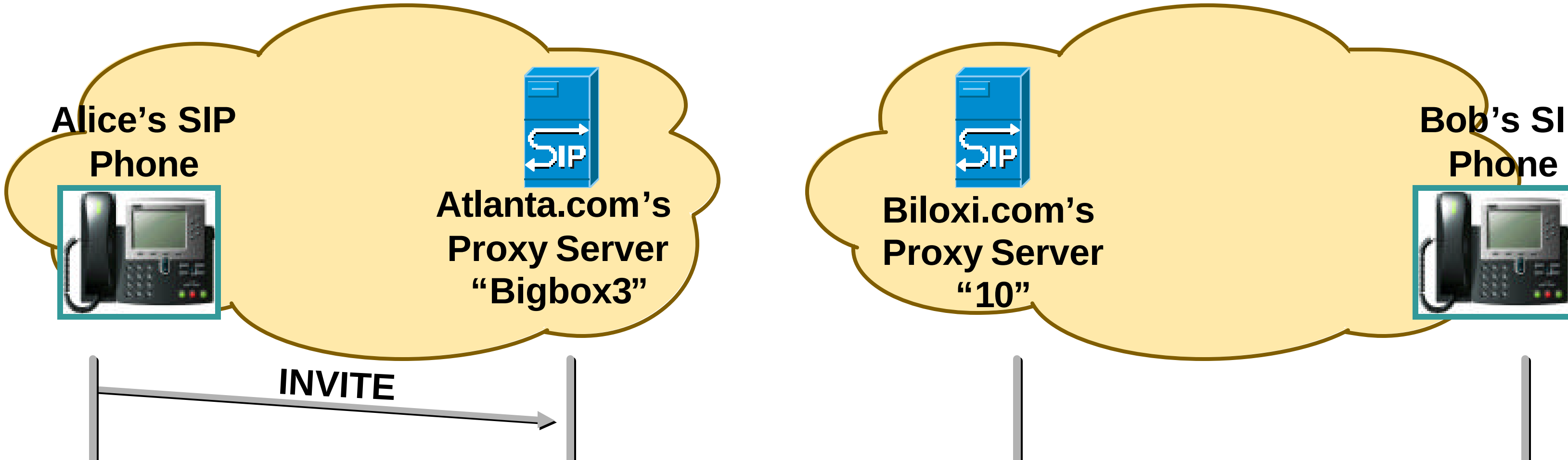
Content-Type: contains a description of the message body

Content-Length: the octet count of the message body

Refer-to: the URI address one UA is directing another UA to INVITE (i.e. Call Transfer)

Record-Route: the list of SIP Proxies that must be in future signaling path of established session

SIP Call Set-up [F1]



INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com
;branch=z9hG4bK776asdhds
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

(Alice's SDP not shown)

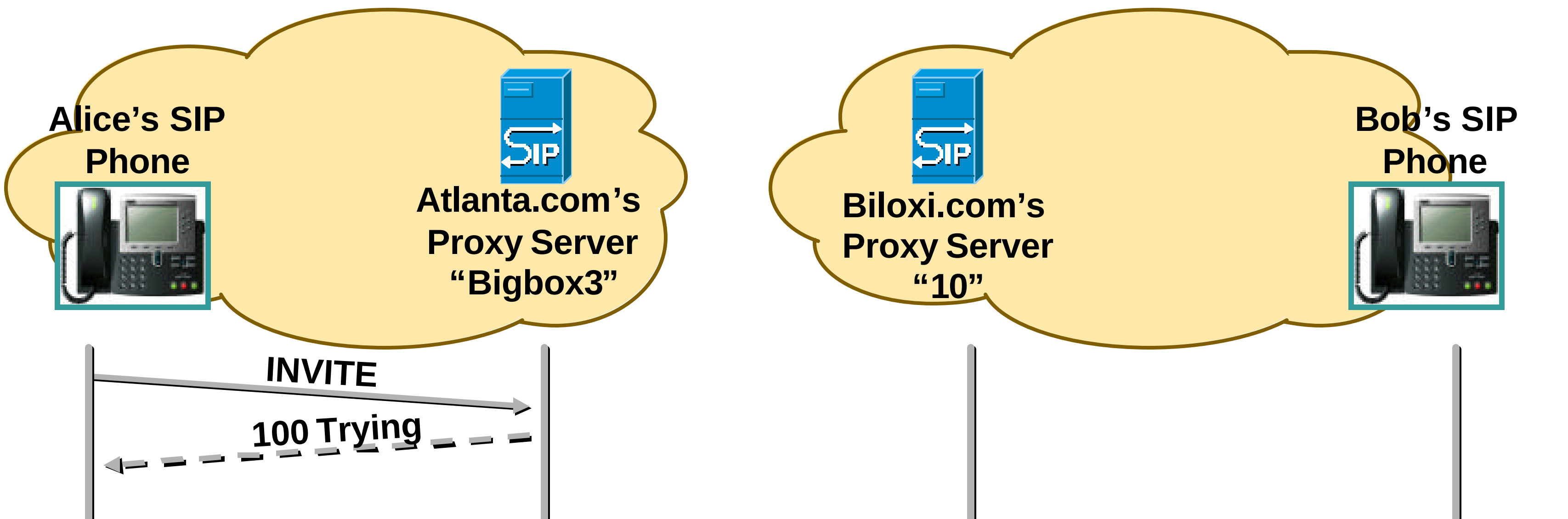
[F1] Alice's INVITE to Bob, through Atlanta.com Proxy Server

Cisco.com

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com
;branch=z9hG4bK776asdhds
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

(Alice's SDP not shown)

SIP Call Set-up [F2]



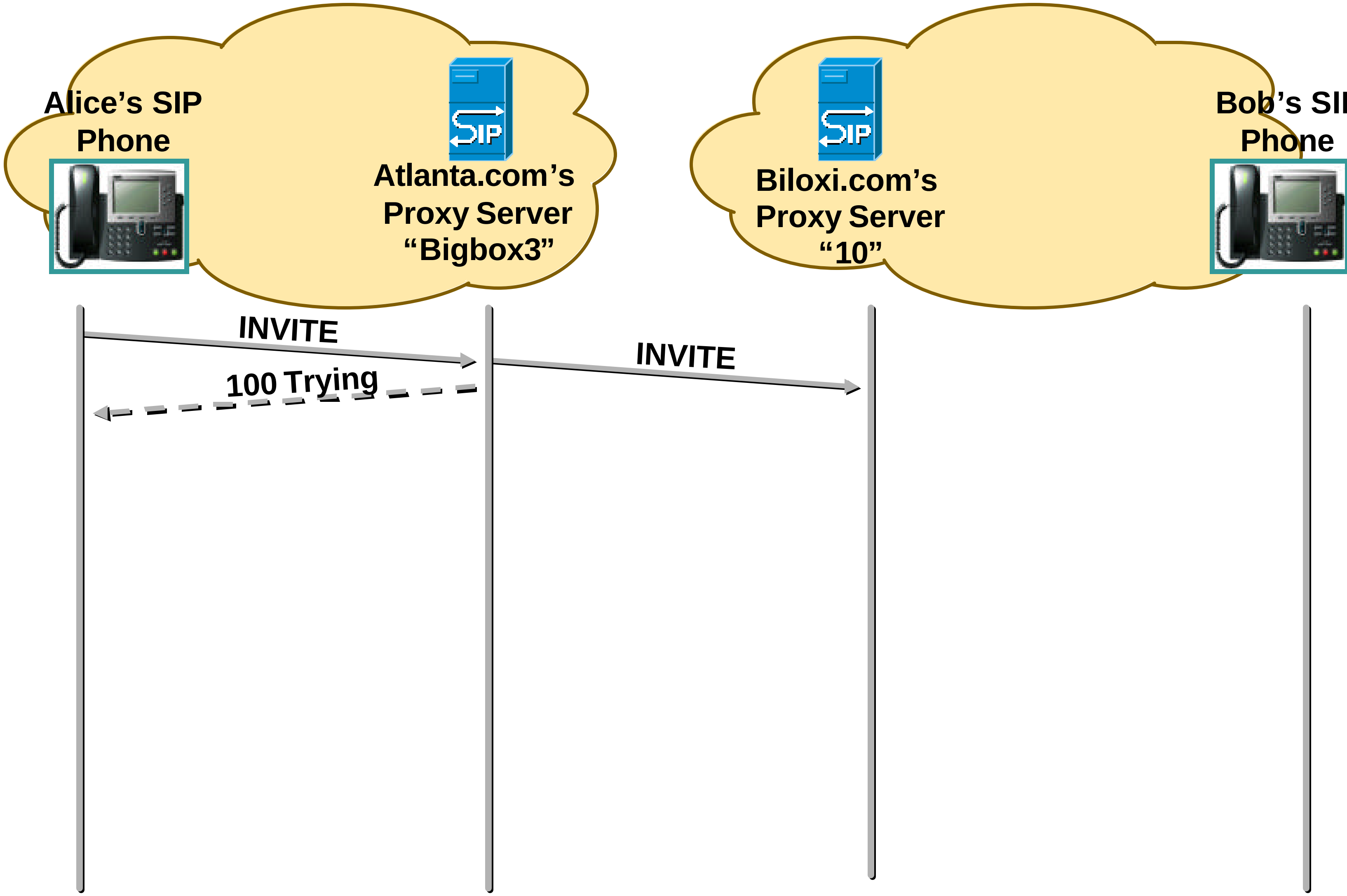
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
;received=10.1.3.33
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Content-Length: 0

[F2] 100 Trying atlanta.com Proxy -> Alice

SIP/2.0 100 Trying
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
;received=10.1.3.33
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Content-Length: 0

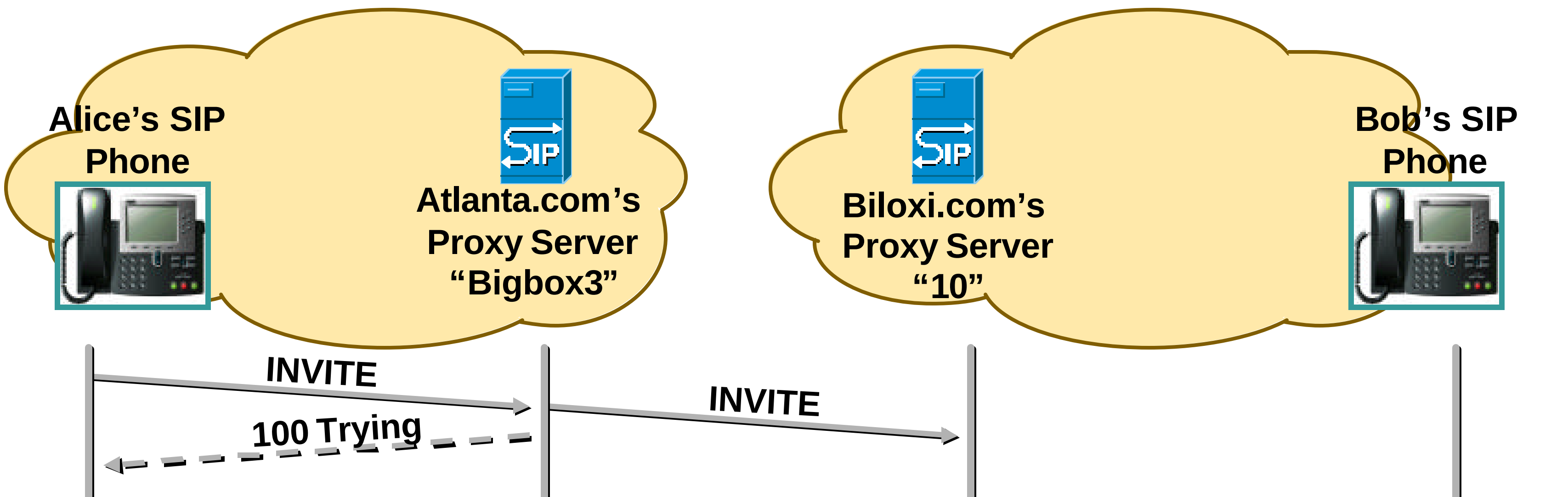
SIP Call Set-up [F3]

Cisco.com



How does Atlanta.com find Biloxi.com?

Cisco.com



Remembering SIP is an *Applications Layer* Protocol that uses other complementary protocols; one of these is DNS

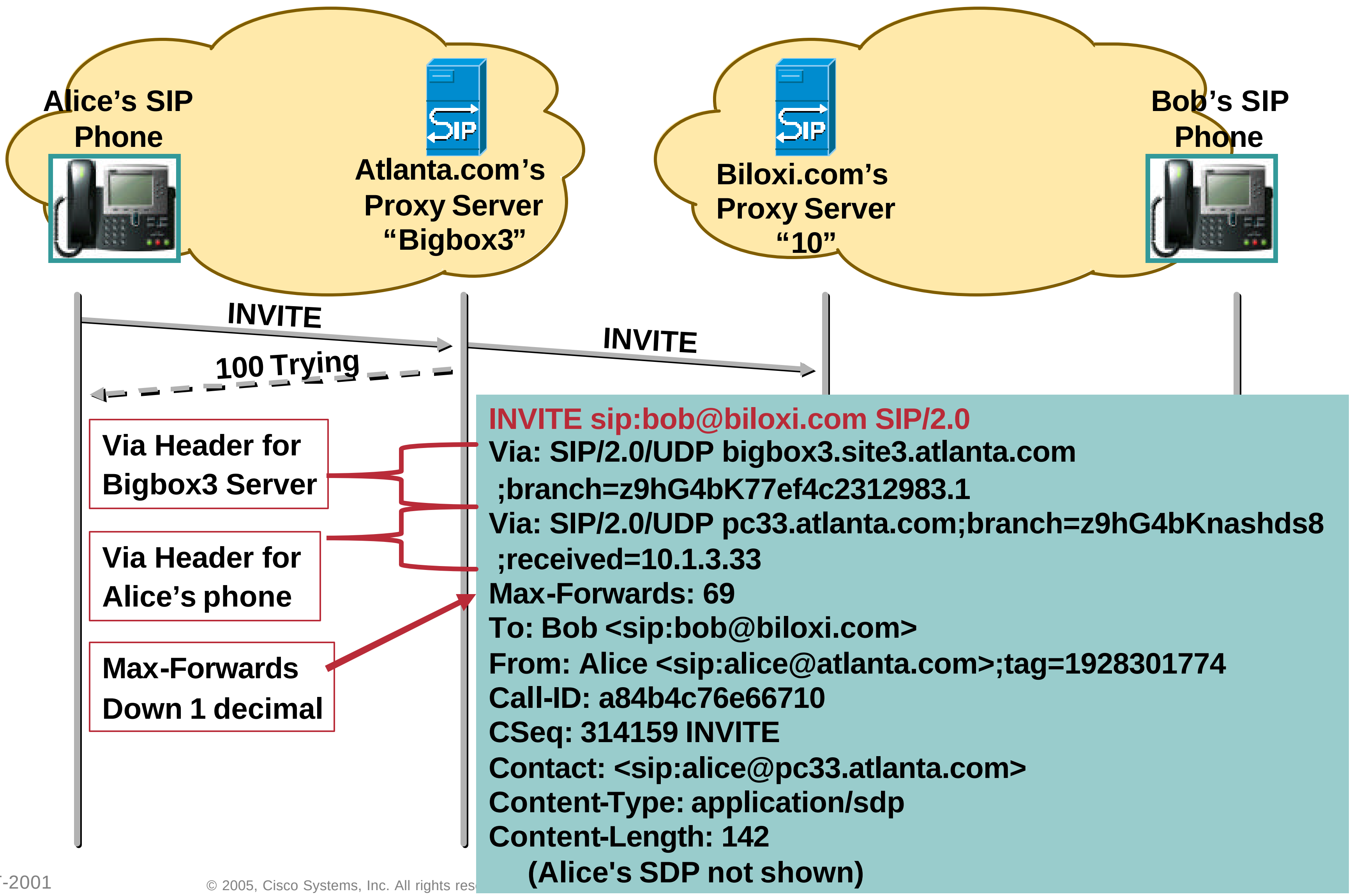
Bigbox3 uses DNS NAPTR and DNS SRV to resolve the appropriate transport Protocol (TCP, UDP, SCTP or TLS over TCP), IP address and port number for the targeted SIP Server within Biloxi.com's network (here this is to Server "10")

See the following for more details on this interaction:

- **RFC 3263 "Locating SIP Servers"**
- **RFC 2915 "The Naming Authority Pointer (NAPTR) DNS Resource Record"**
- **RFC 2782 "A DNS RR for Specifying the Location of Services (DNS SRV)"**

SIP Call Set-up [F3]

Cisco.com



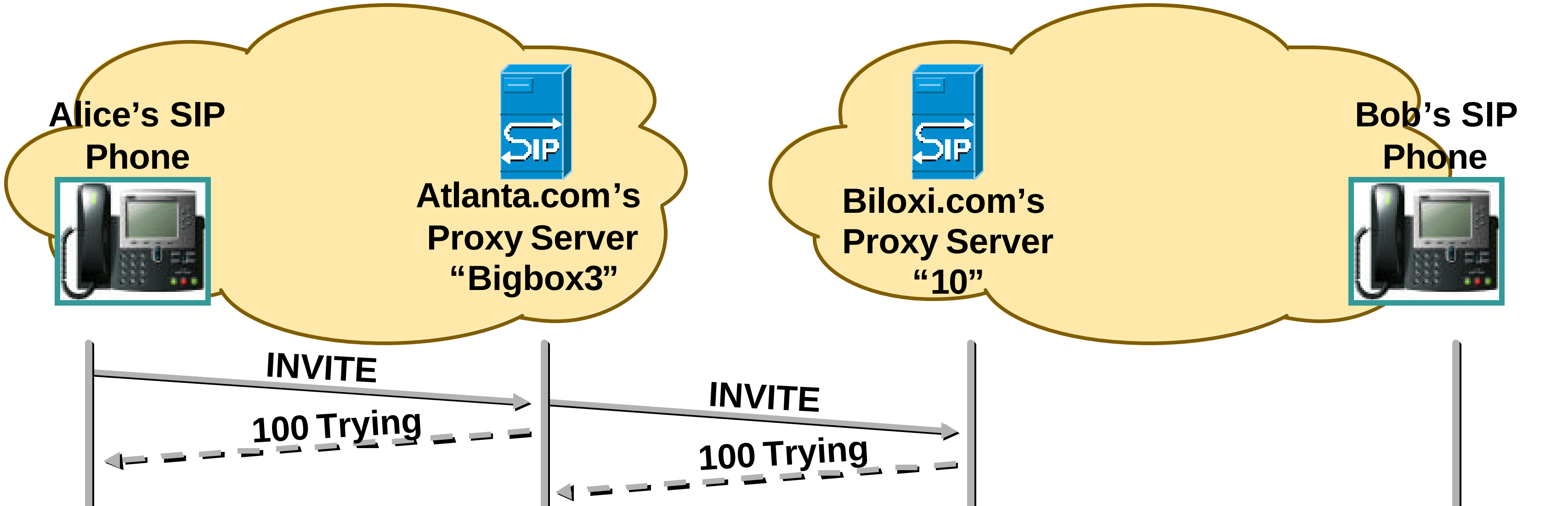
[F3] INVITE atlanta.com Proxy -> biloxi.com Proxy

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
;branch=z9hG4bK77ef4c2312983.1
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
;received=10.1.3.33
Max-Forwards: 69
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

(Alice's SDP not shown)

SIP Call Set-up [F4]

Cisco.com



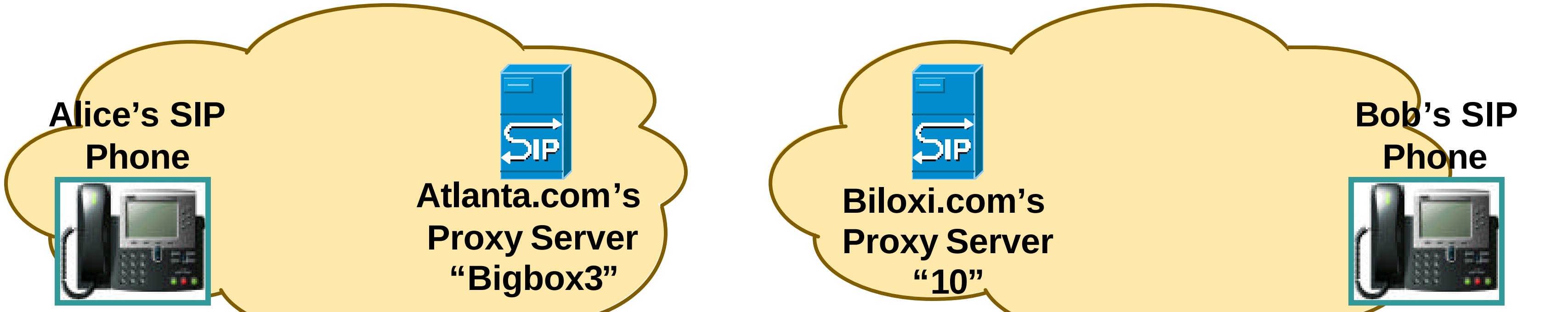
SIP/2.0 100 Trying
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
;branch=z9hG4bK77ef4c2312983.1
;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
;received=10.1.3.33
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Content-Length: 0

[F4] 100 Trying biloxi.com Proxy -> atlanta.com Proxy

SIP/2.0 100 Trying
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
;branch=z9hG4bK77ef4c2312983.1
;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
;received=10.1.3.33
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Content-Length: 0

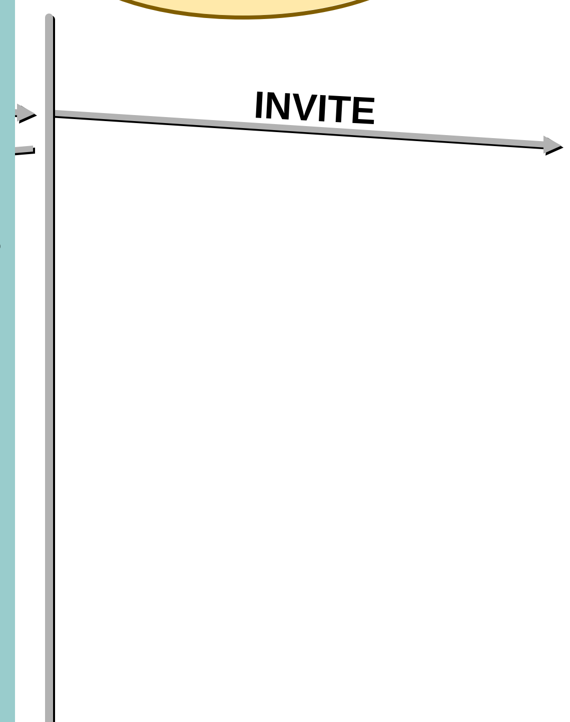
SIP Call Set-up [F5]

Cisco.com



INVITE sip:bob@192.168.10.20 SIP/2.0
Via: SIP/2.0/UDP server10.biloxi.com
;branch=z9hG4bK4b43c2ff8.1
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
;branch=z9hG4bK77ef4c2312983.1;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
;received=10.1.3.33
Max-Forwards: 68
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

(Alice's SDP not shown)



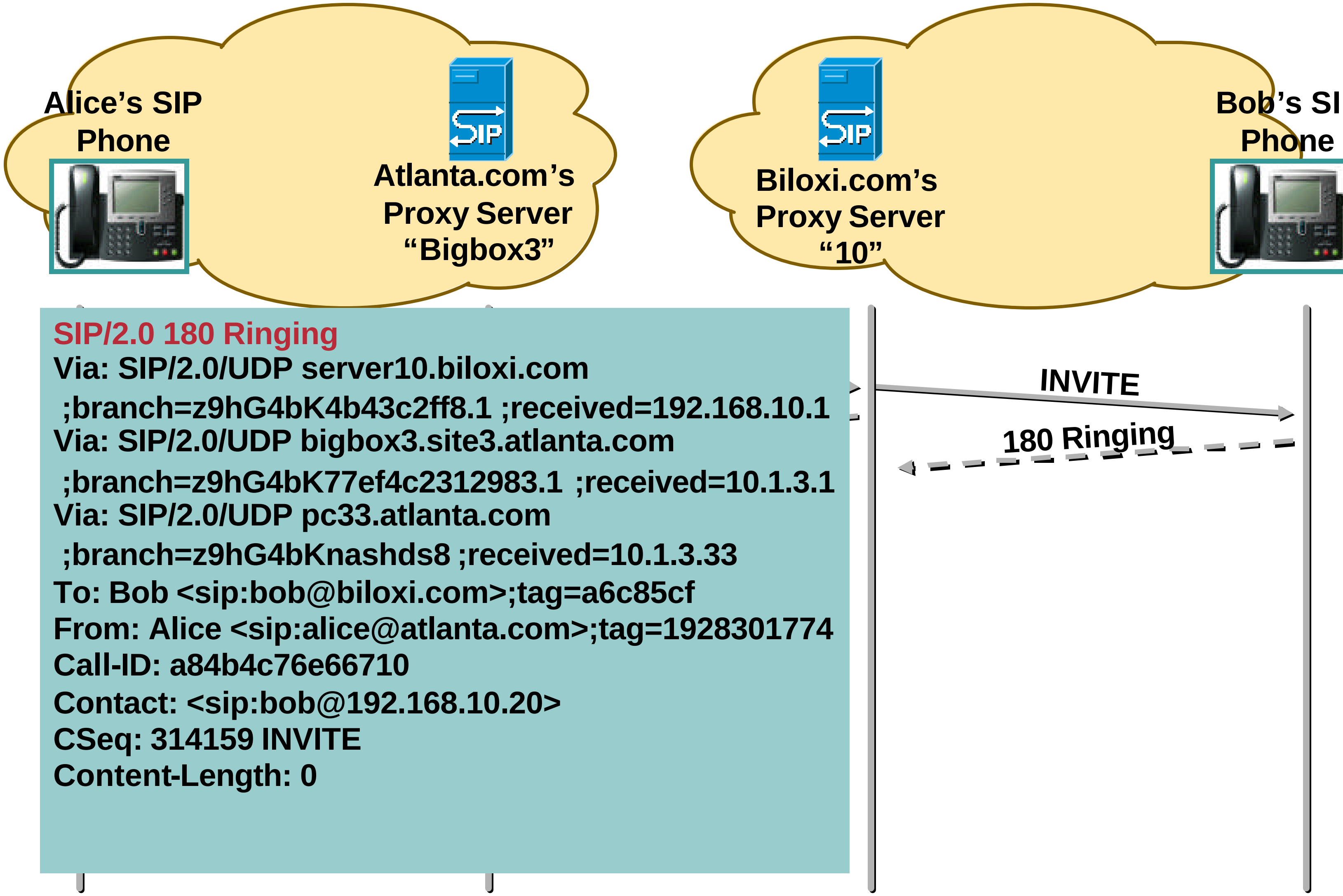
[F5] INVITE biloxi.com Proxy -> Bob

INVITE sip:bob@192.168.10.20 SIP/2.0
Via: SIP/2.0/UDP server10.biloxi.com
;branch=z9hG4bK4b43c2ff8.1
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
;branch=z9hG4bK77ef4c2312983.1;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
;received=10.1.3.33
Max-Forwards: 68
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

(Alice's SDP not shown)

SIP Call Set-up [F6]

Cisco.com




```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP server10.biloxi.com
 ;branch=z9hG4bK4b43c2ff8.1 ;received=192.168.10.1
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
 ;branch=z9hG4bK77ef4c2312983.1 ;received=10.1.3.1
Via: SIP/2.0/UDP pc33.atlanta.com
 ;branch=z9hG4bKnashds8 ;received=10.1.3.33
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
Contact: <sip:bob@192.168.10.20>
CSeq: 314159 INVITE
Content-Length: 0
```

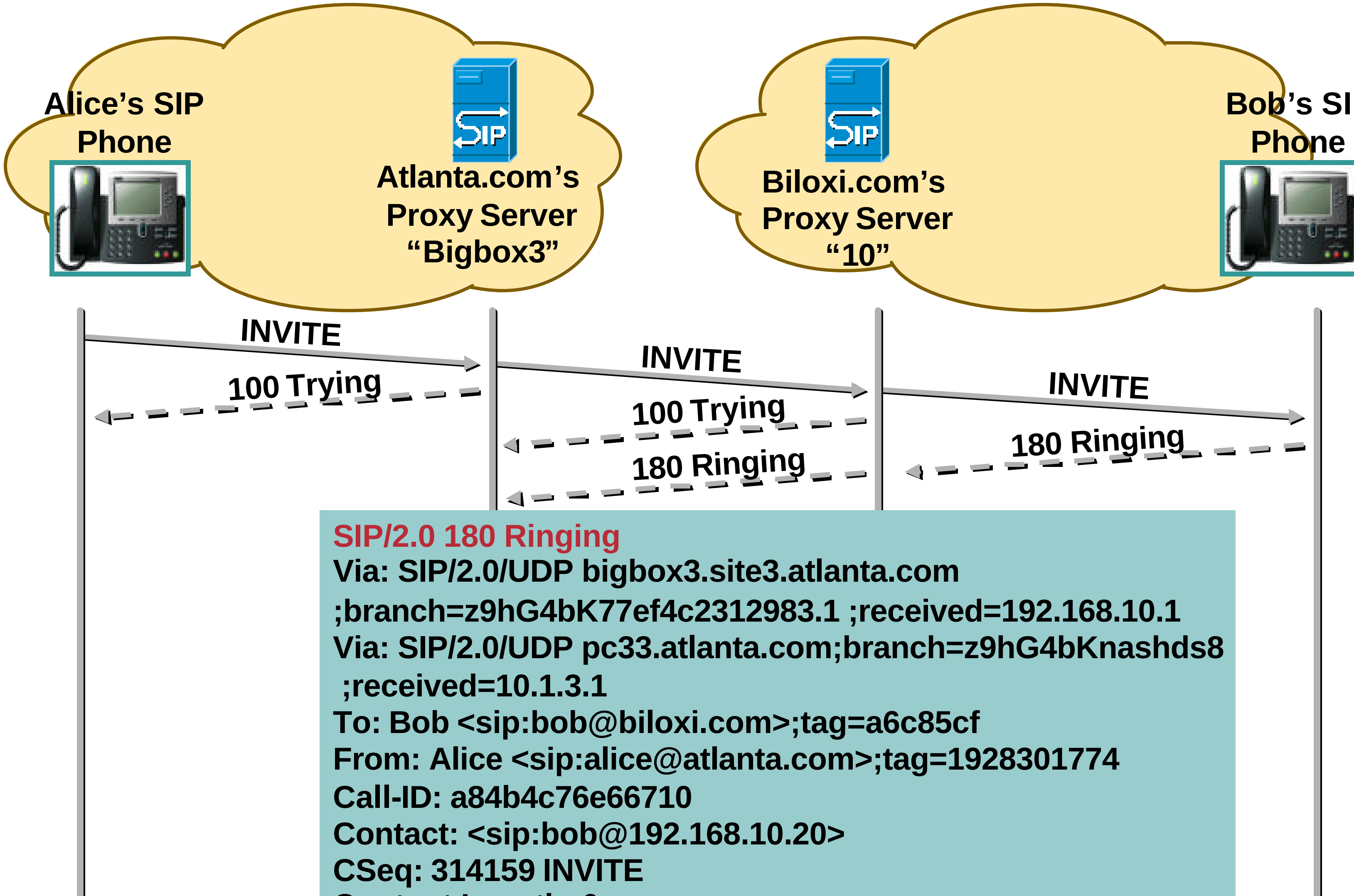
[F6] 180 Ringing Bob -> biloxi.com Proxy

Cisco.com

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP server10.biloxi.com
;branch=z9hG4bK4b43c2ff8.1 ;received=192.168.10.1
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
;branch=z9hG4bK77ef4c2312983.1 ;received=10.1.3.1
Via: SIP/2.0/UDP pc33.atlanta.com
;branch=z9hG4bKnashds8 ;received=10.1.3.33
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
Contact: <sip:bob@192.168.10.20>
CSeq: 314159 INVITE
Content-Length: 0

SIP Call Set-up [F7]

Cisco.com



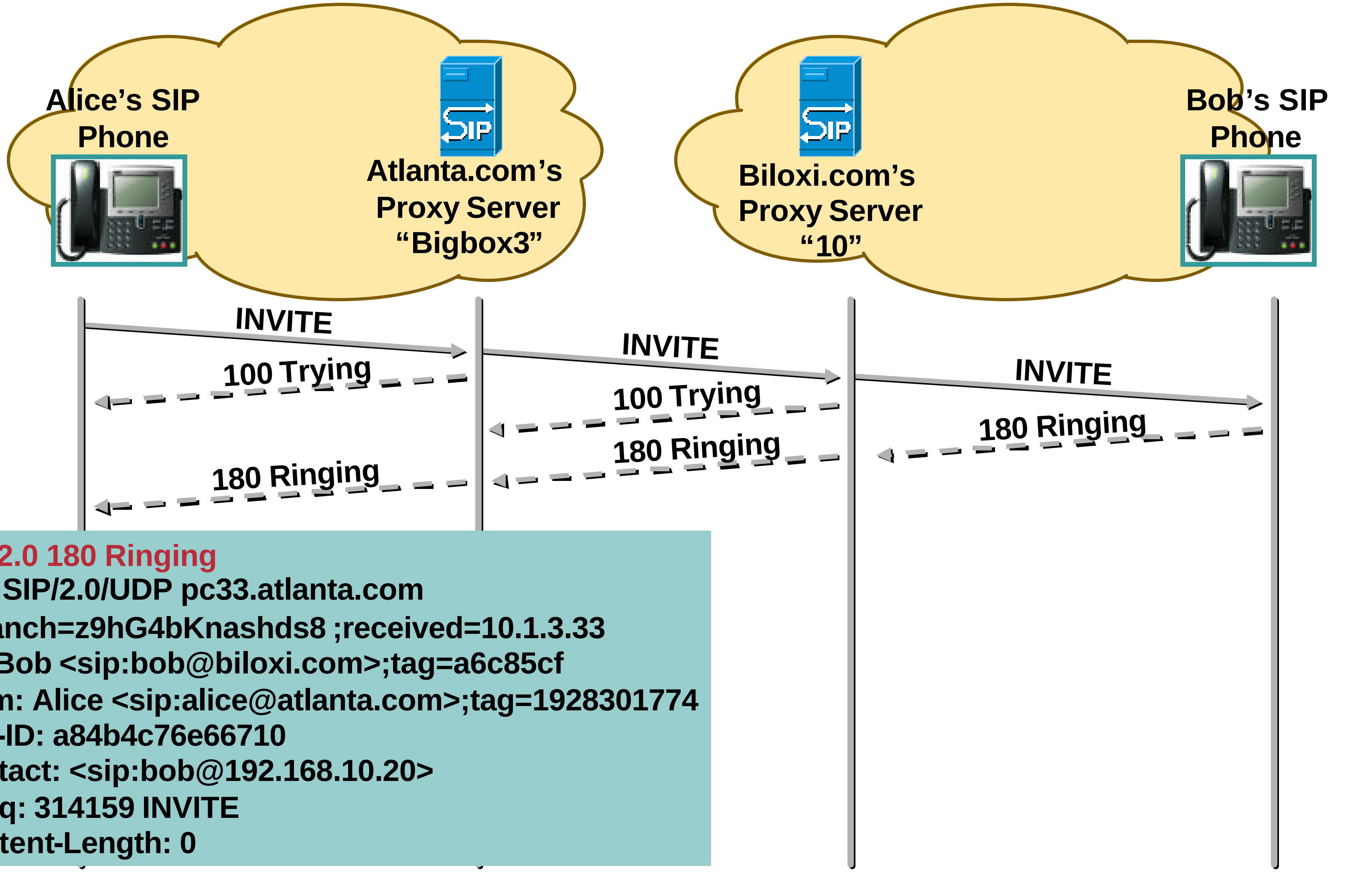
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
;branch=z9hG4bK77ef4c2312983.1 ;received=192.168.10.1
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
;received=10.1.3.1
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
Contact: <sip:bob@192.168.10.20>
CSeq: 314159 INVITE
Content-Length: 0

[F7] 180 Ringing biloxi.com proxy -> atlanta.com Proxy

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
;branch=z9hG4bK77ef4c2312983.1 ;received=192.168.10.1
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
;received=10.1.3.1
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
Contact: <sip:bob@192.168.10.20>
CSeq: 314159 INVITE
Content-Length: 0

SIP Call Set-up [F8]

Cisco.com



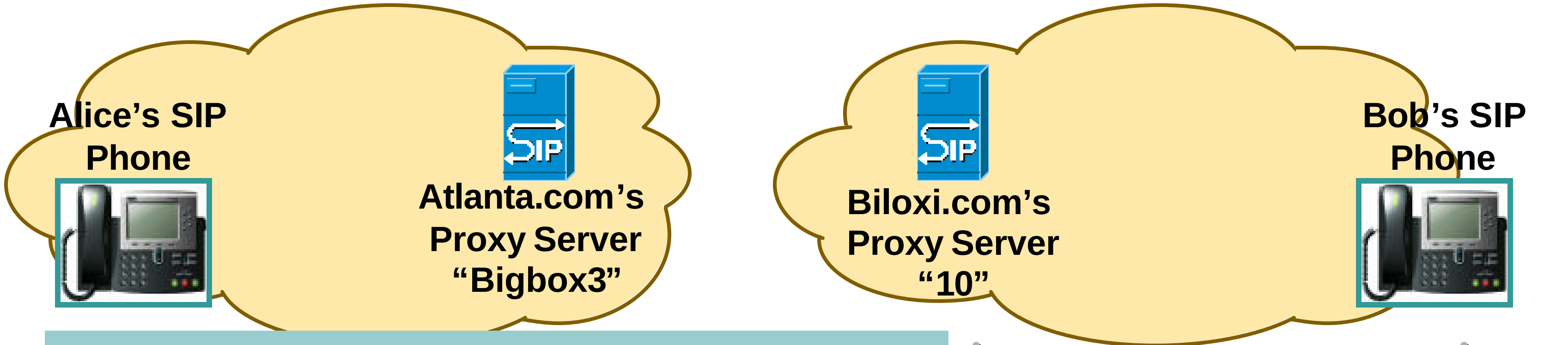
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pc33.atlanta.com
;branch=z9hG4bKnashds8 ;received=10.1.3.33
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
Contact: <sip:bob@192.168.10.20>
CSeq: 314159 INVITE
Content-Length: 0

[F8] 180 Ringing atlanta.com Proxy -> Alice

Cisco.com

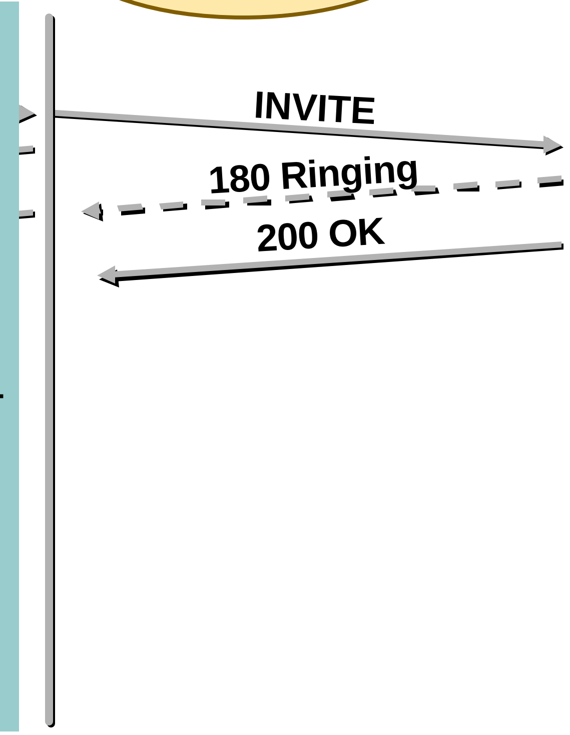
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pc33.atlanta.com
;branch=z9hG4bKnashds8 ;received=10.1.3.33
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
Contact: <sip:bob@192.168.10.20>
CSeq: 314159 INVITE
Content-Length: 0

SIP Call Set-up [F9]



SIP/2.0 200 OK
Via: SIP/2.0/UDP server10.biloxi.com
;branch=z9hG4bK4b43c2ff8.1 ;received=192.168.10.1
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
;branch=z9hG4bK77ef4c2312983.1 ;received=10.1.3.1
Via: SIP/2.0/UDP pc33.atlanta.com
;branch=z9hG4bKnashds8 ;received=10.1.3.33
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.168.10.20>
Content-Type: application/sdp
Content-Length: 131

(Bob's SDP not shown)



[F9] 200 OK Bob -> biloxi.com Proxy

Cisco.com

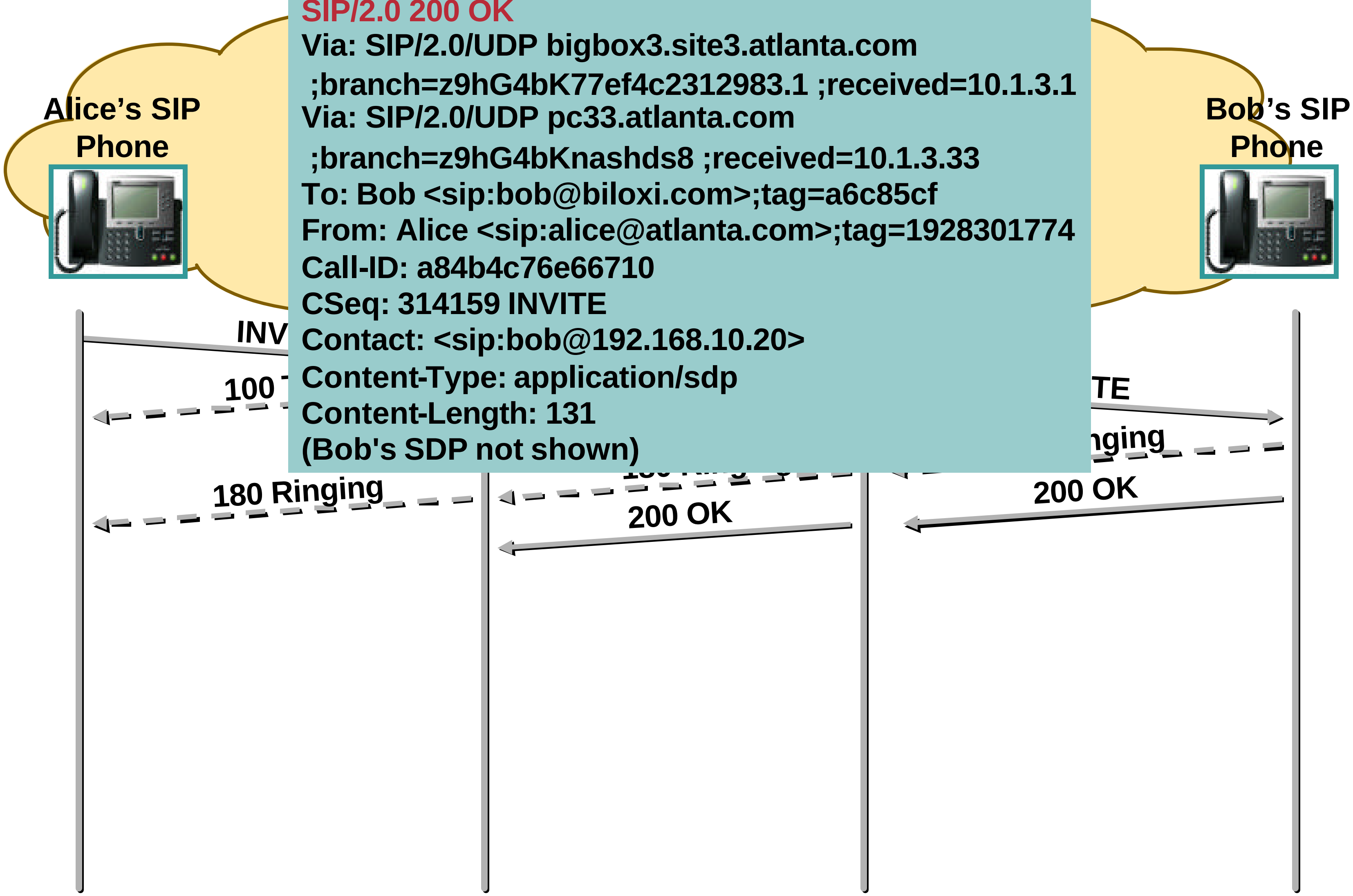
SIP/2.0 200 OK
Via: SIP/2.0/UDP server10.biloxi.com
;branch=z9hG4bK4b43c2ff8.1 ;received=192.168.10.1
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
;branch=z9hG4bK77ef4c2312983.1 ;received=10.1.3.1
Via: SIP/2.0/UDP pc33.atlanta.com
;branch=z9hG4bKnashds8 ;received=10.1.3.33
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.168.10.20>
Content-Type: application/sdp
Content-Length: 131

(Bob's SDP not shown)

SIP Call Set-up [F10]

Cisco.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
;branch=z9hG4bK77ef4c2312983.1 ;received=10.1.3.1
Via: SIP/2.0/UDP pc33.atlanta.com
;branch=z9hG4bKnashds8 ;received=10.1.3.33
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.168.10.20>
Content-Type: application/sdp
Content-Length: 131
(Bob's SDP not shown)



[F10] 200 OK biloxi.com Proxy -> atlanta.com Proxy

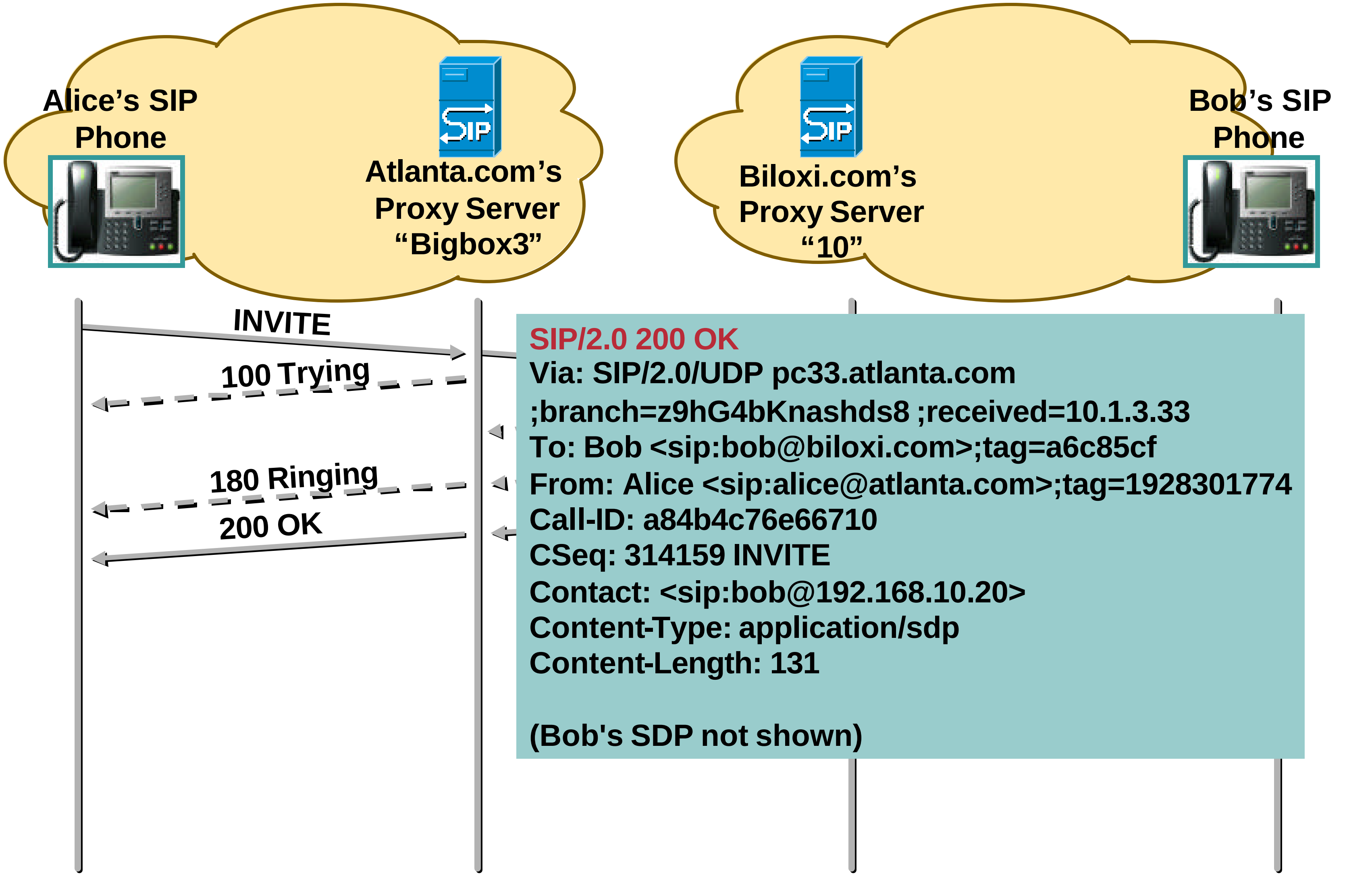
Cisco.com

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
 ;branch=z9hG4bK77ef4c2312983.1 ;received=10.1.3.1
Via: SIP/2.0/UDP pc33.atlanta.com
 ;branch=z9hG4bKnashds8 ;received=10.1.3.33
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.168.10.20>
Content-Type: application/sdp
Content-Length: 131

(Bob's SDP not shown)
```

SIP Call Set-up [F11]

Cisco.com




```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pc33.atlanta.com
;branch=z9hG4bKnashds8 ;received=10.1.3.33
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.168.10.20>
Content-Type: application/sdp
Content-Length: 131

(Bob's SDP not shown)
```

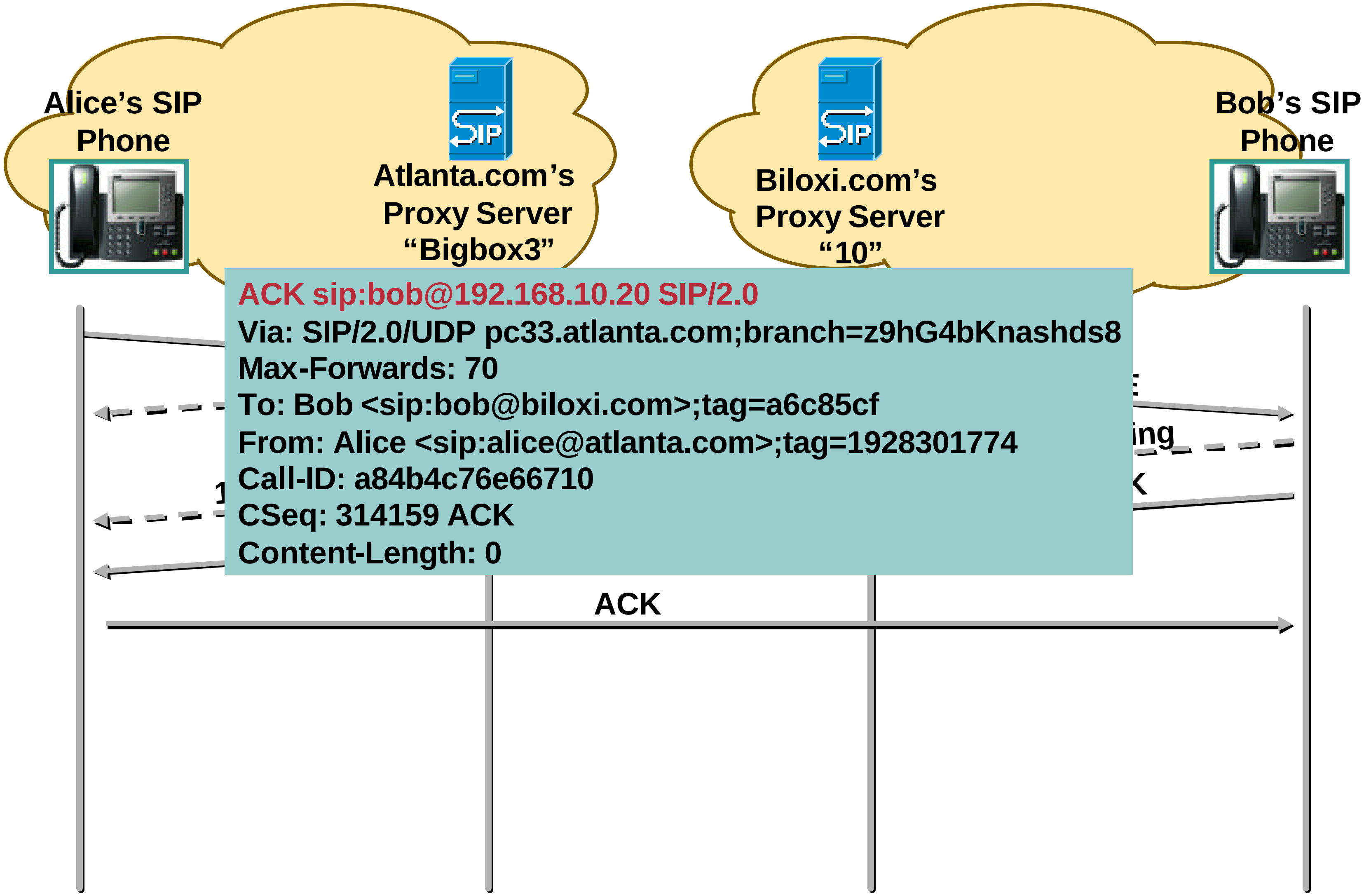
[F11] 200 OK atlanta.com Proxy -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP pc33.atlanta.com
;branch=z9hG4bKnashds8 ;received=10.1.3.33
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.168.10.20>
Content-Type: application/sdp
Content-Length: 131

(Bob's SDP not shown)

SIP Call Set-up [F12]

Cisco.com



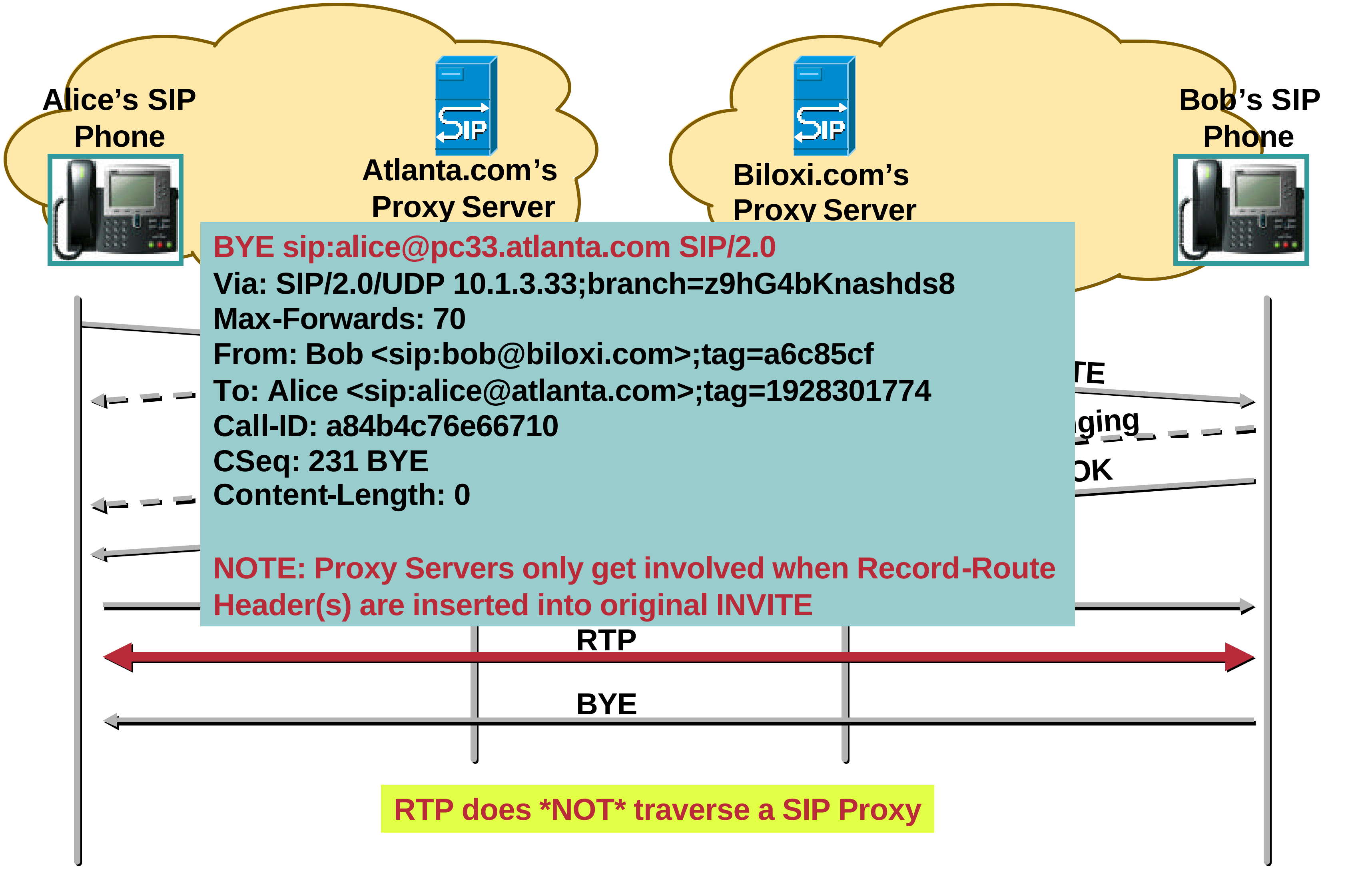
ACK sip:bob@192.168.10.20 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

[F12] ACK Alice -> Bob

ACK sip:bob@192.168.10.20 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

SIP Call Set-up [F13]

Cisco.com



BYE sip:alice@pc33.atlanta.com SIP/2.0
Via: SIP/2.0/UDP 10.1.3.33;branch=z9hG4bKnashds8
Max-Forwards: 70
From: Bob <sip:bob@biloxi.com>;tag=a6c85cf
To: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 231 BYE
Content-Length: 0

NOTE: Proxy Servers only get involved when Record-Route Header(s) are inserted into original INVITE

RTP does *NOT* traverse a SIP Proxy

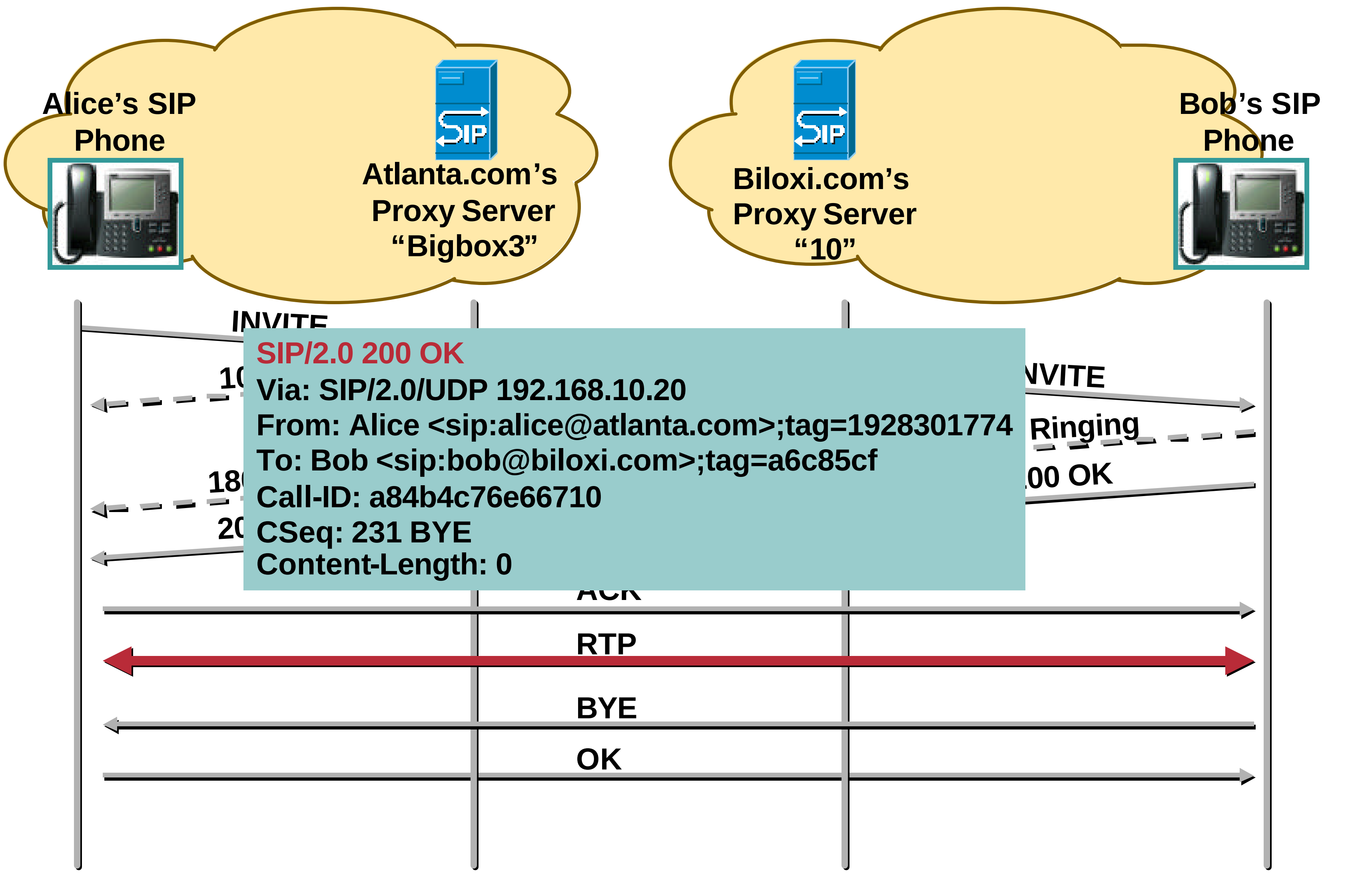
[F13] BYE Bob -> Alice

BYE sip:alice@pc33.atlanta.com SIP/2.0
Via: SIP/2.0/UDP 10.1.3.33;branch=z9hG4bKnashds8
Max-Forwards: 70
From: Bob <sip:bob@biloxi.com>;tag=a6c85cf
To: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 231 BYE
Content-Length: 0

Proxy Servers only get involved when Record Route Header(s) are inserted into original INVITE

SIP Call Set-up [F14]

Cisco.com



[F14] 200 OK Alice -> Bob

SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.10.20
From: Alice <sip:alice@atlanta.com>;tag=1928301774
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
Call-ID: a84b4c76e66710
CSeq: 231 BYE
Content-Length: 0

SIP Call Set-up [F1–F14]

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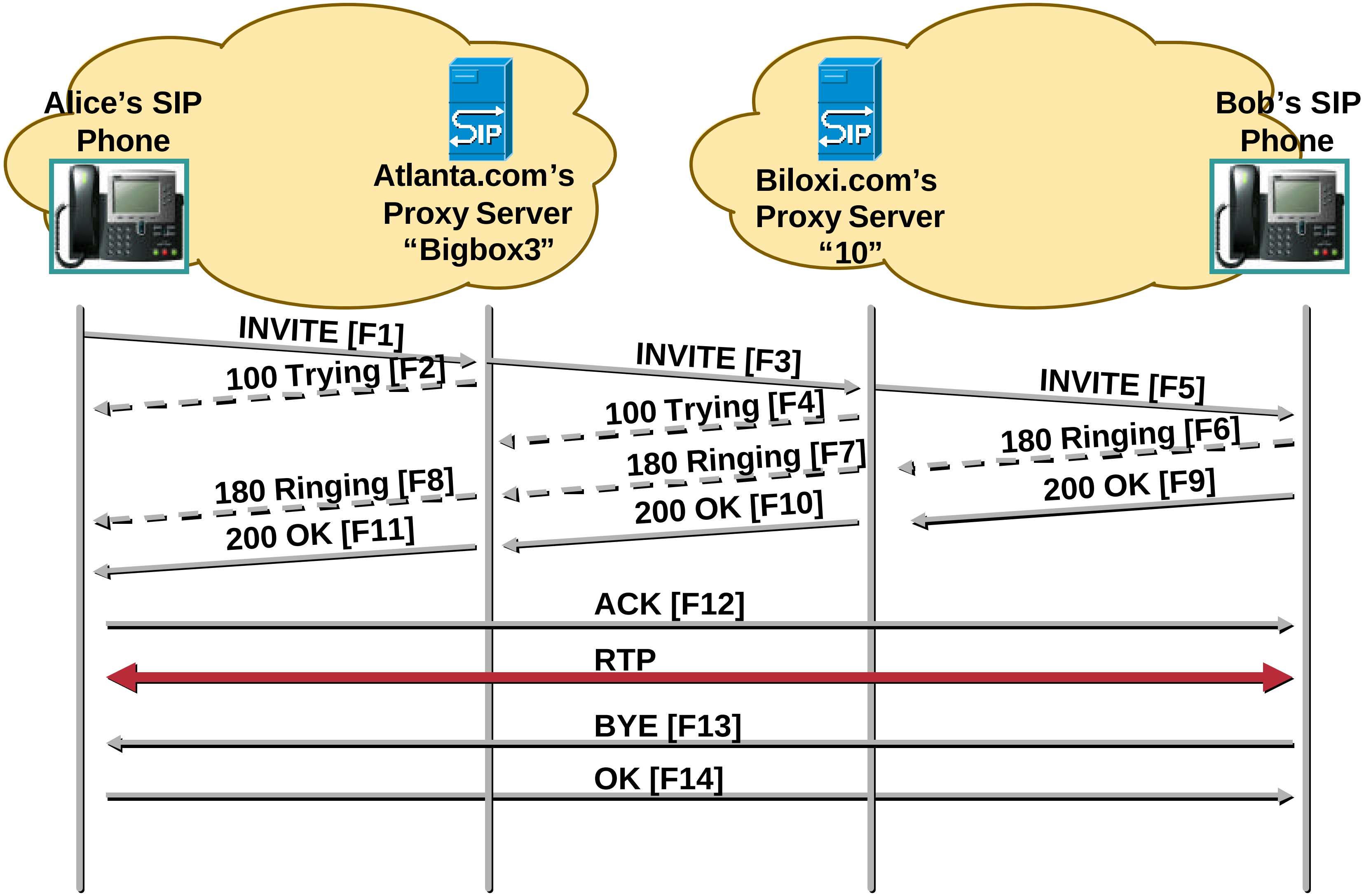

SIP Error Response Examples

- What if Bob is not able to answer Alice?
- What if Alice includes an unacceptable codec?
- Call Forward Busy No Answer Function to 3rd party
- Call Forward Busy No Answer Function to Voice Mail
- What if the SIP Gateway has no more ports available?

RFC 3261

SIP Example: Called Party Busy

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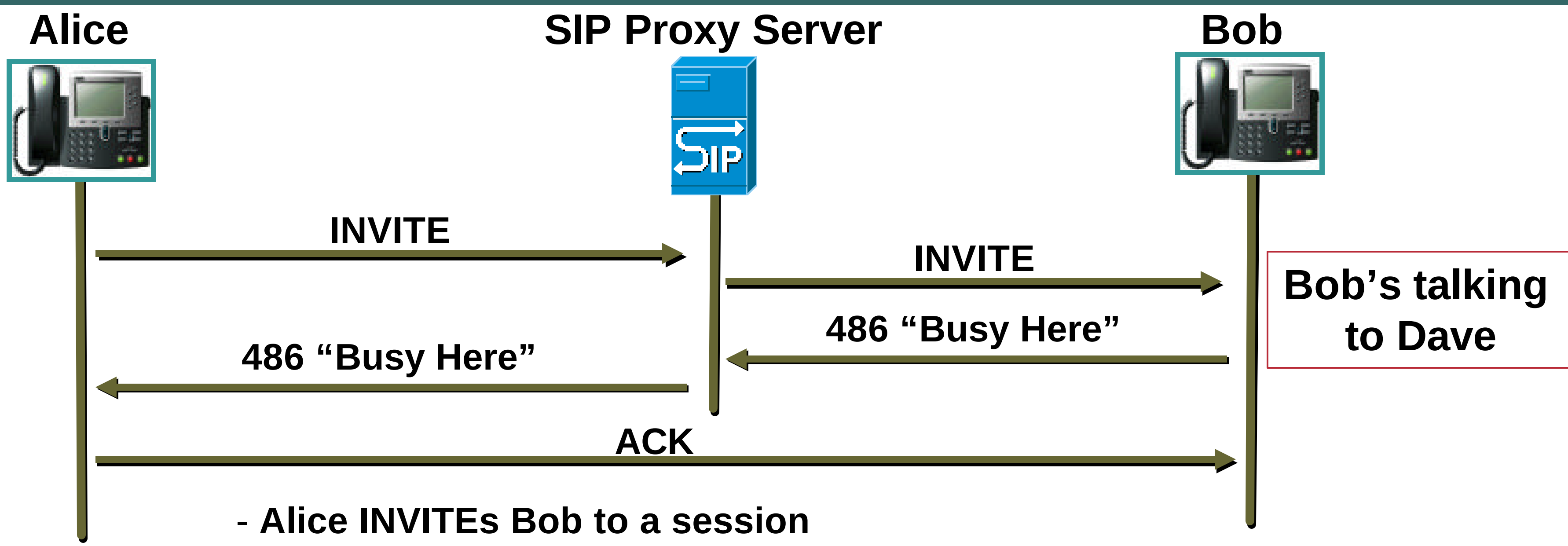
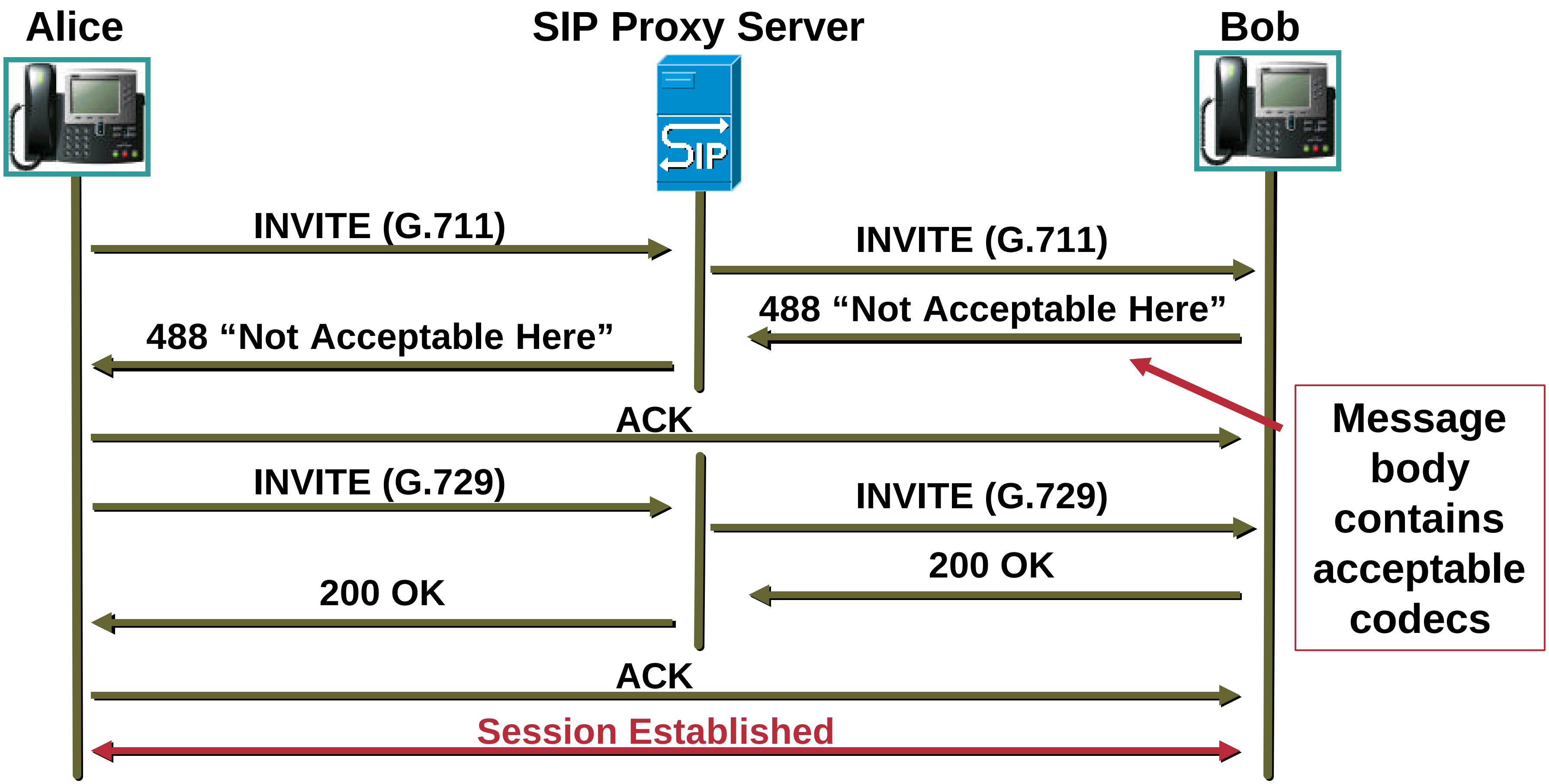


- Alice INVITEs Bob to a session
- Bob’s on the phone with Dave
- Bob’s UA responds with a “Busy Here” indication

General processing rule of thumb:

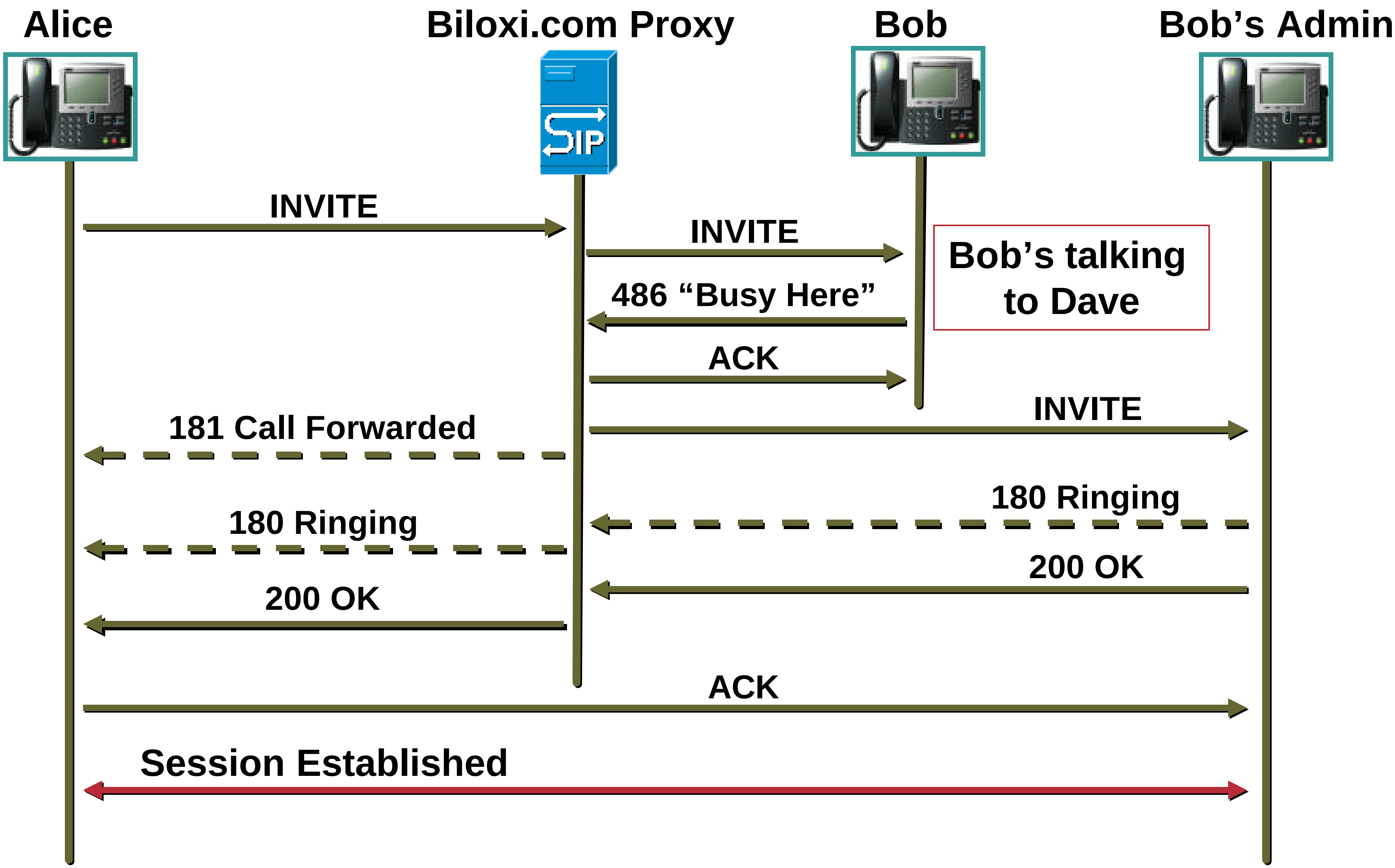
- respond with a 486 (Busy Here) if the UAS cannot or is currently not willing to take the call
- respond with a 600 (Busy Everywhere) if the UAS knows no other endsystem will answer (unlikely)
- respond with a 488 (Not Acceptable Here) if the UAS is rejecting the Offer (should state why)

SIP Example: Unsupported Codec

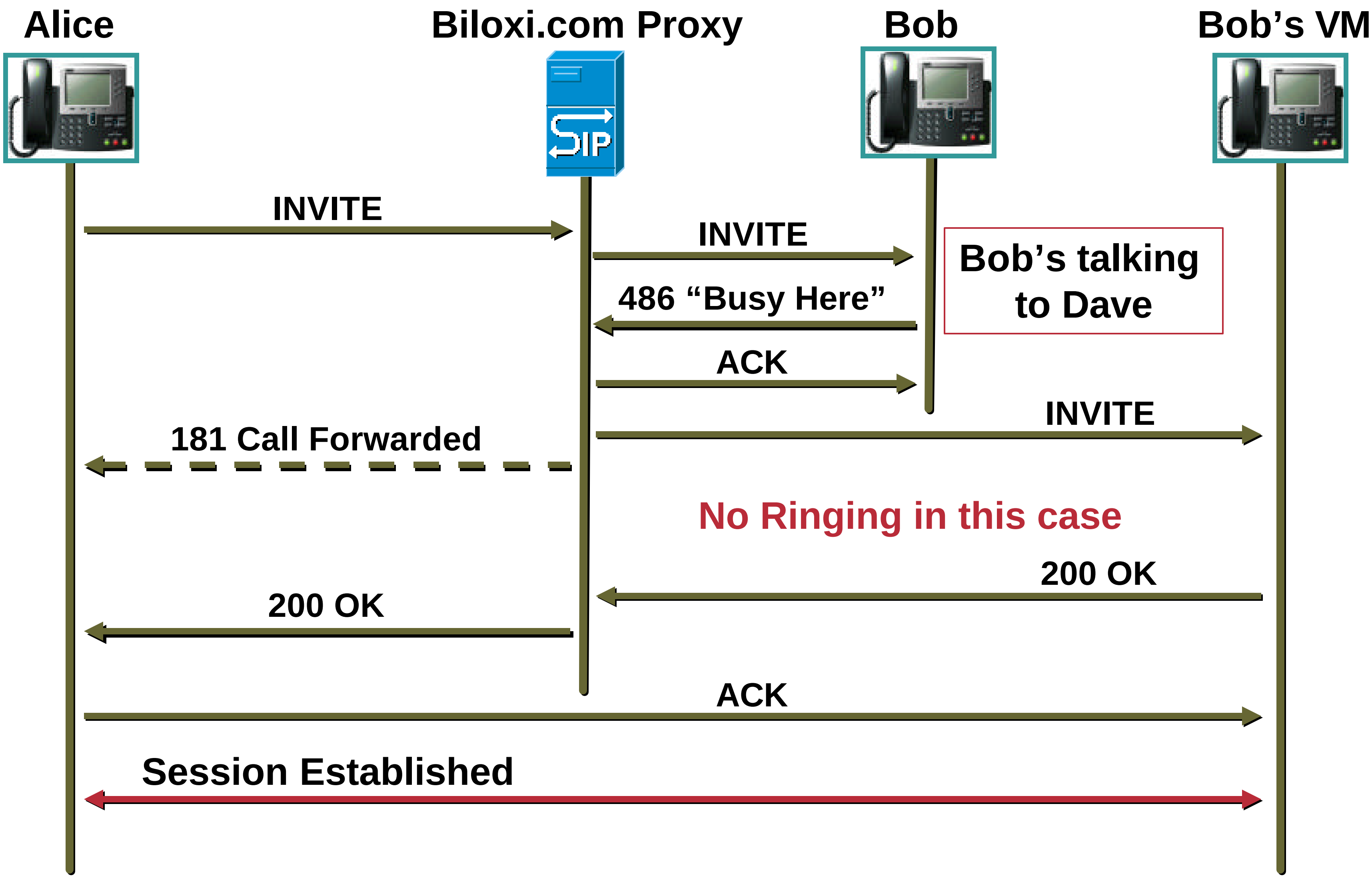


- Alice INVITEs Bob to a session offering G.711 only
- Bob's UA doesn't understand G.711, but can do G.729
- Bob's 488 Error message indicates which other codecs it supports in SDP
- Alice UA INVITEs Bob again offering G.729, which Bob's UA accepts

SIP Example: Call Forward Busy No Answer



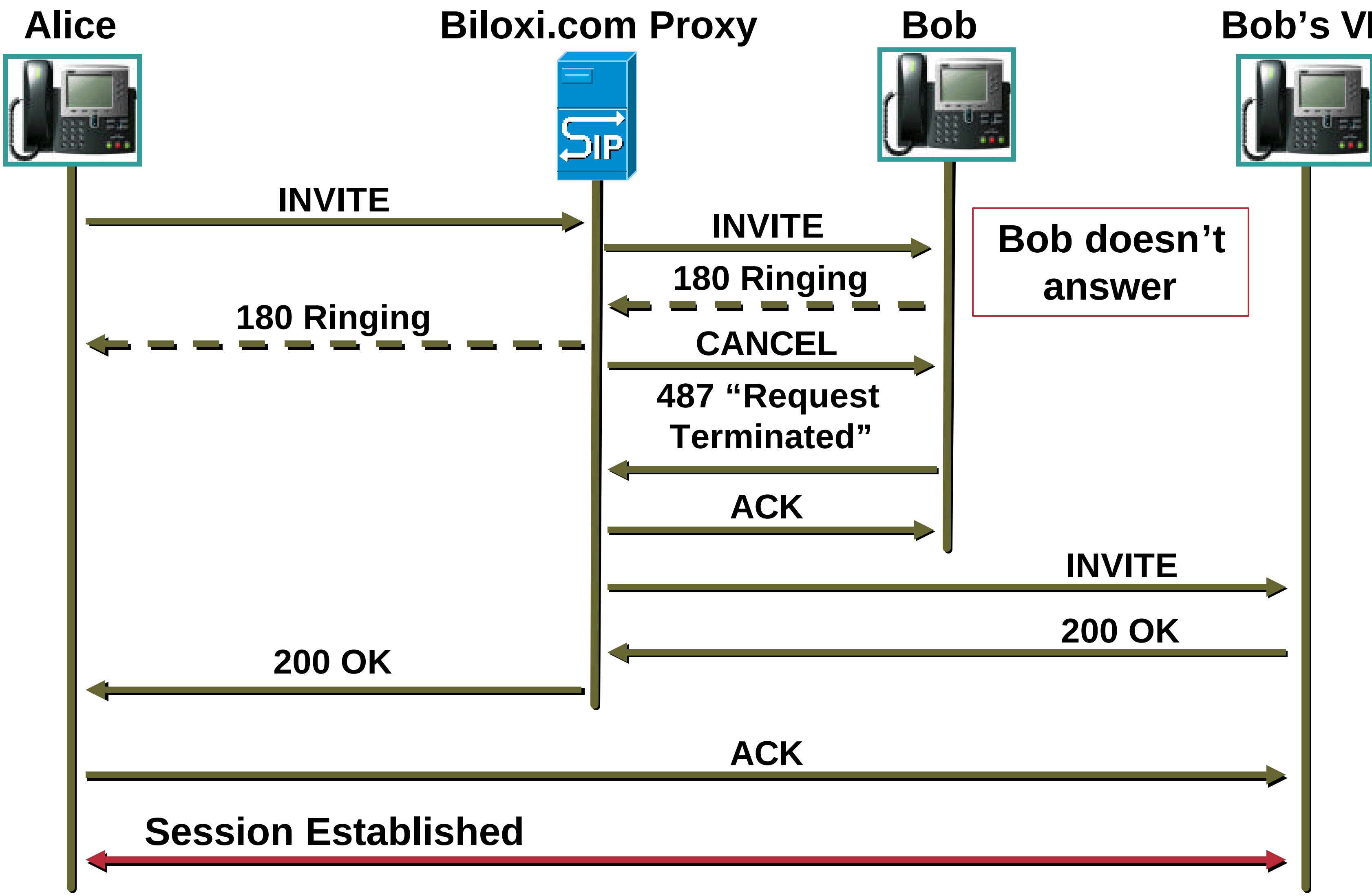
SIP Example: Call Forward Busy No Answer



RFC 3261

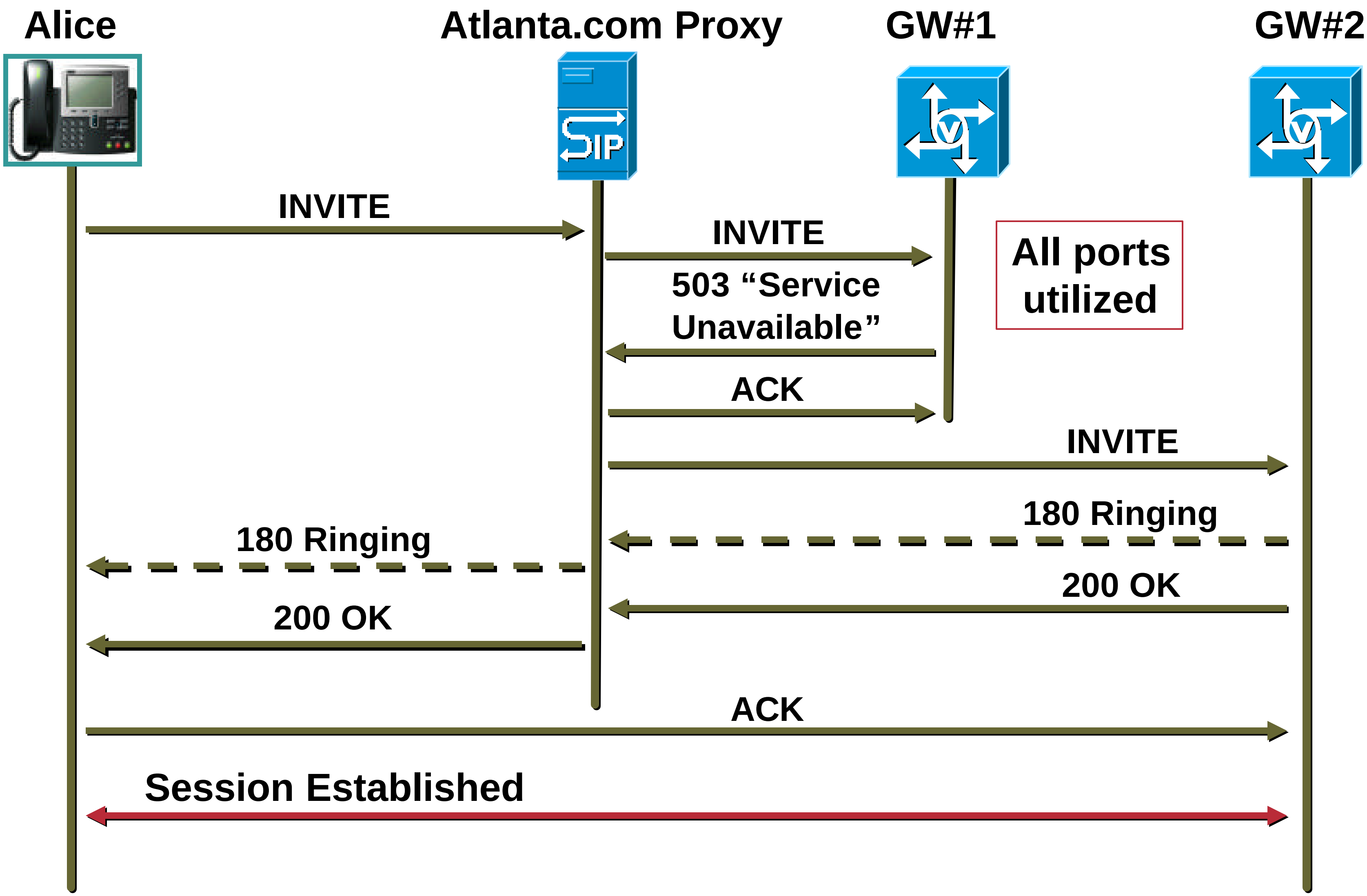
SIP Example: Call Forward Busy No Answer

Cisco.com



RFC 3261

SIP Example: Gateway Congestion



Session Behavior

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- **Once a session is established, media is transmitted between the two (or more) parties involved in the original session**
- **If a change in the characteristics is desired by either Bob or Alice, a new INVITE is sent to the other referencing the existing dialog making sure both know what this new request is about***

  A "200 OK" is the response to accept the change, always with an ACK returned

  A "406 Not Acceptable" is the possible return for not accepting the proposed changes, also with the ACK returned

- **This does not fail the existing call, this merely doesn't allow the proposed change to occur**

*** Proxy(ies) Only Involved if They Inserted the Record-Route Header into the Original Session Initiation**

What's the difference between this...

INVITE **sip:bob@biloxi.com SIP/2.0**
Via: **SIP/2.0/TCP pc33.atlanta.com;branch=z9hG4bK776asdhds**
Max-Forwards: **70**
To: **Bob <sip:bob@biloxi.com>**
From: **Alice <sip:alice@atlanta.com>;tag=1928301774**
Call-ID: **a84b4c76e66710@pc33.atlanta.com**
CSeq: **314159 INVITE**
Contact: **<sip:alice@pc33.atlanta.com>**
Content-Type: **application/sdp**
Content-Length: **142**

What’s the difference between this...and this?

MESSAGE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/TCP pc33.atlanta.com;branch=z9hG4bK776sgdkse
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301775
Call-ID: a84b4c76e66711@pc33.atlanta.com
CSeq: 10197 MESSAGE
Content-Type: text/plain
Content-Length: 37

Isn’t this a great presentation, Bob?

The Extensions of SIP

- **SIP functionality is spread over several Standards Track RFCs:**

  - Sending additional information about existing session (RFC 2976)
  - Provisional Acknowledgements (RFC 3262)
  - Discovering Remote SIP Server locations (RFC 3263)
  - Subscriptions to remote devices looking for events to occur (RFC 3265)
  - Notifications when events occur at remote devices (RFC 3265)
  - Updating an existing session set-up (RFC 3311)
  - Preconditions for QoS (RFC 3312)
  - Privacy Mechanisms (RFC 3323)
  - Asserted Identity (RFC 3325)
  - Sending messaging between users or to an automaton (RFC 3428)
  - Referring one user agent to communicate with another (RFC 3515)
  - Service Route Discovery during Registrations (RFC 3608)
  - Message Waiting Indicator (RFC 3842)
  - Publishing information about a user agent to a service (RFC 3903)

The Coming Extensions of SIP

Cisco.com

- **SIP functionality is spread over several Standards Track Internet Drafts:**
  - Location Conveyance
  - Emergency Session Identification and Routing (e911/112)
  - Message Prioritization and Session Preemption
  - Embedded Authorization for Requested Behavior
  - Preconditions for Security
  - History Information of a Message
  - Session Timers
  - Session Policy
  - Session Independent Policy
  - Conferencing
  - Transcoding

SIP References

Cisco.com

- http://www.cisco.com—Search for SIP, Cisco proxy server
- http://www.cs.columbia.edu/~hgs/sip/—SIP homepage
- http://www.ietf.org/html.charters/sip-charter.html/—IETF SIP WG
- http://www.ietf.org/html.charters/sipping-charter.html/—IETF SIPPING WG
- http://search.ietf.org/rfc.html—IETF RFC search page
- http://search.ietf.org/search/brokers/internet-drafts/query.html—Internet draft search page
- http://www.softarmor.com/sipwg/—SIP WG supplemental site
- http://www.softarmor.com/sipping/—SIPPING WG supplemental site
- http://www.sipcenter.com/—The SIP center

SIP Signaling Architecture

- Elements of SIP
- SIP Signaling
- SIP in a Network
- <u>SIP Summary</u>

James Polk 20050503

In Conclusion…

VVT-2001

In Conclusion…

- SIP is a session set-up protocol
- SIP is Layers 3 & 4 independent
- SIP utilizes many existing IETF protocols
- Used for Voice, Video and Instant Messaging
- Enables many ways to accomplish services based on the requirements of the situation
- The integration of call control and web services creates applications that reduce cost and increase productivity

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James M. Polk

Complete Your Session Evaluation Form

Thank you for attending this session.

Complete your session evaluation, por favor.

Give to the room attendants as you leave the room.

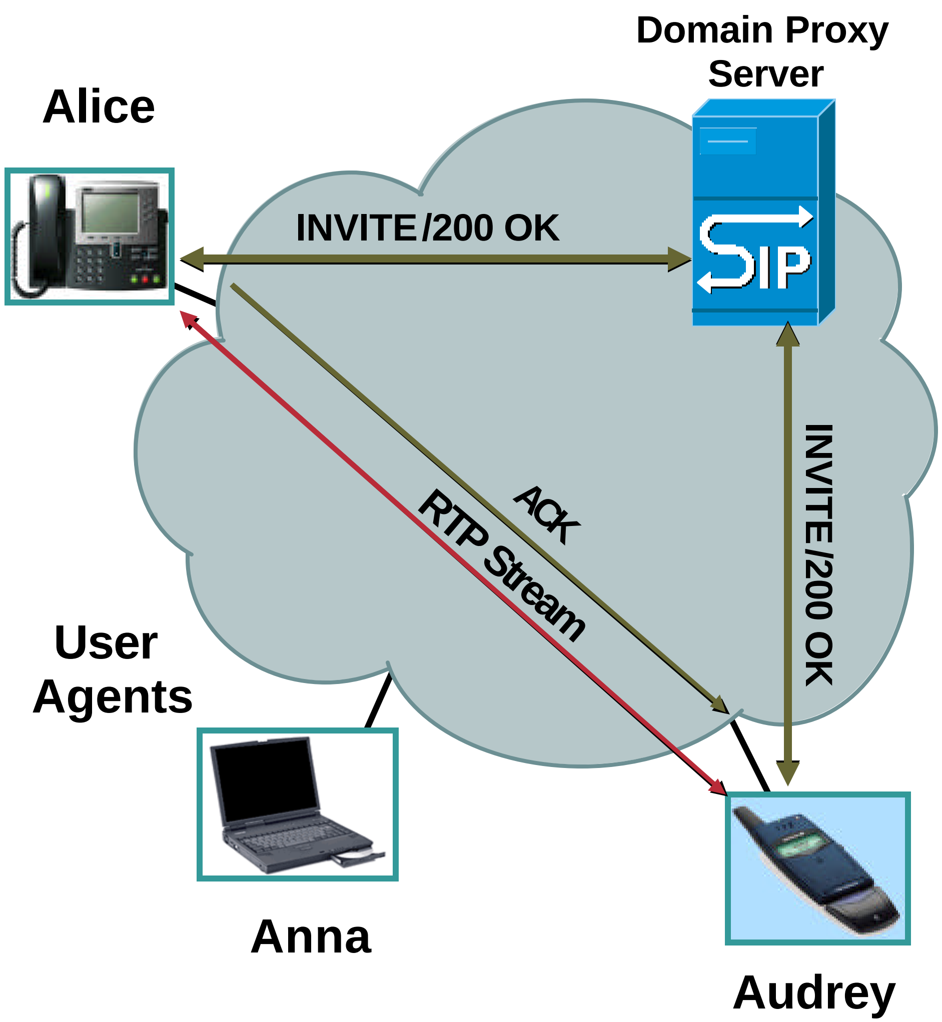
Muchas Gracias

James Polk 20050503

Interworking—Putting It All Together

VVT-2001—Module 2

Peer-to-Peer to/from Client/Server Interworking

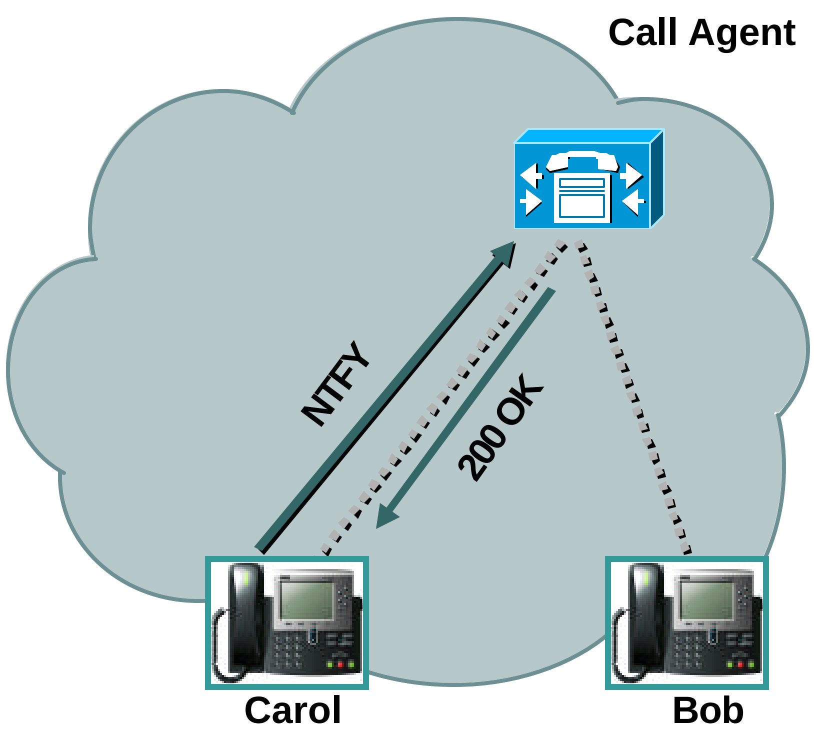


SIP Only Domain

Alice Calls Audrey

Peer-to-Peer to/from Client/Server Interworking

MGCP Only Domain



Carol Calls Bob

Peer-to-Peer to/from Client/Server Interworking

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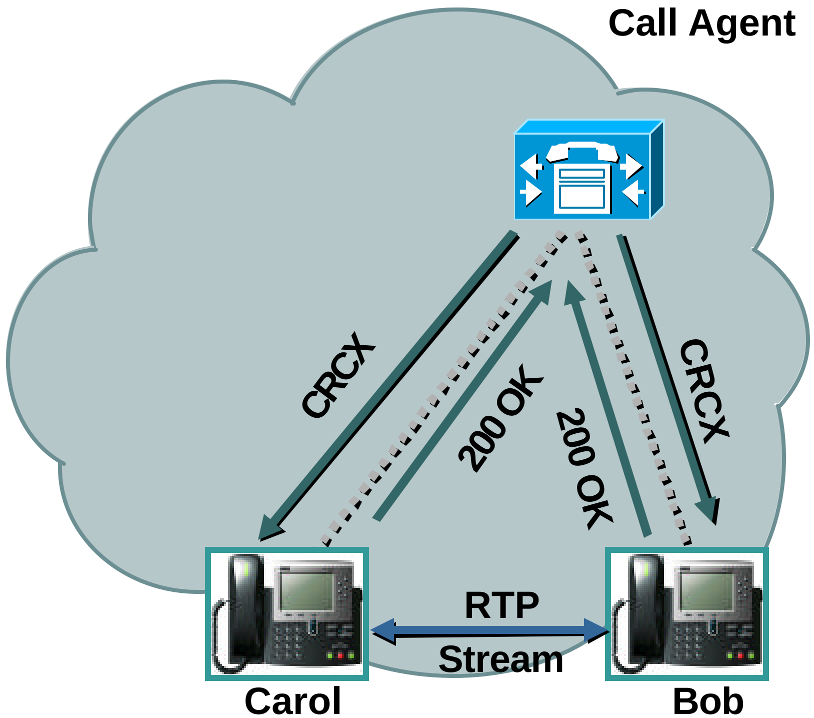
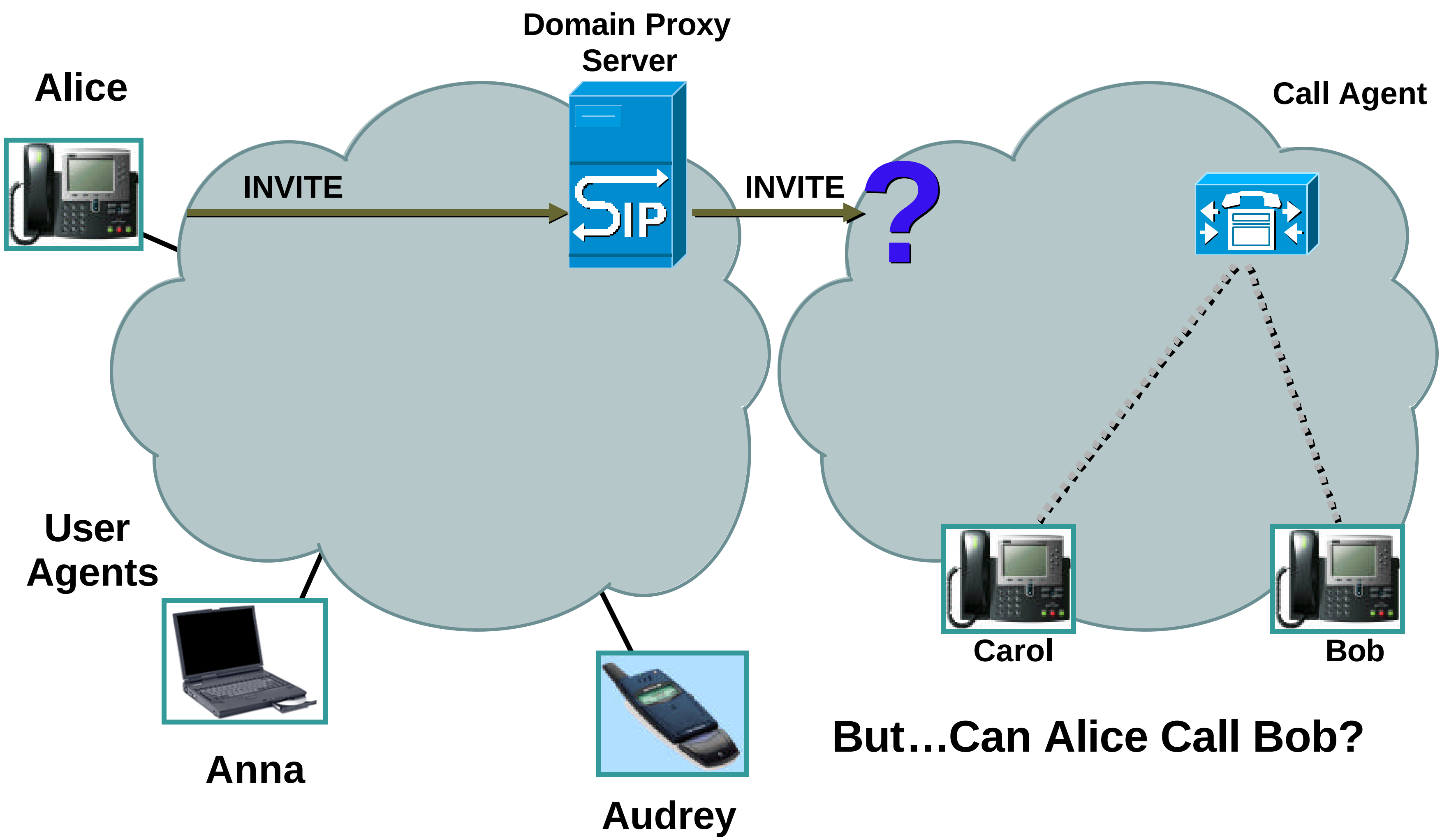
MGCP Only Domain



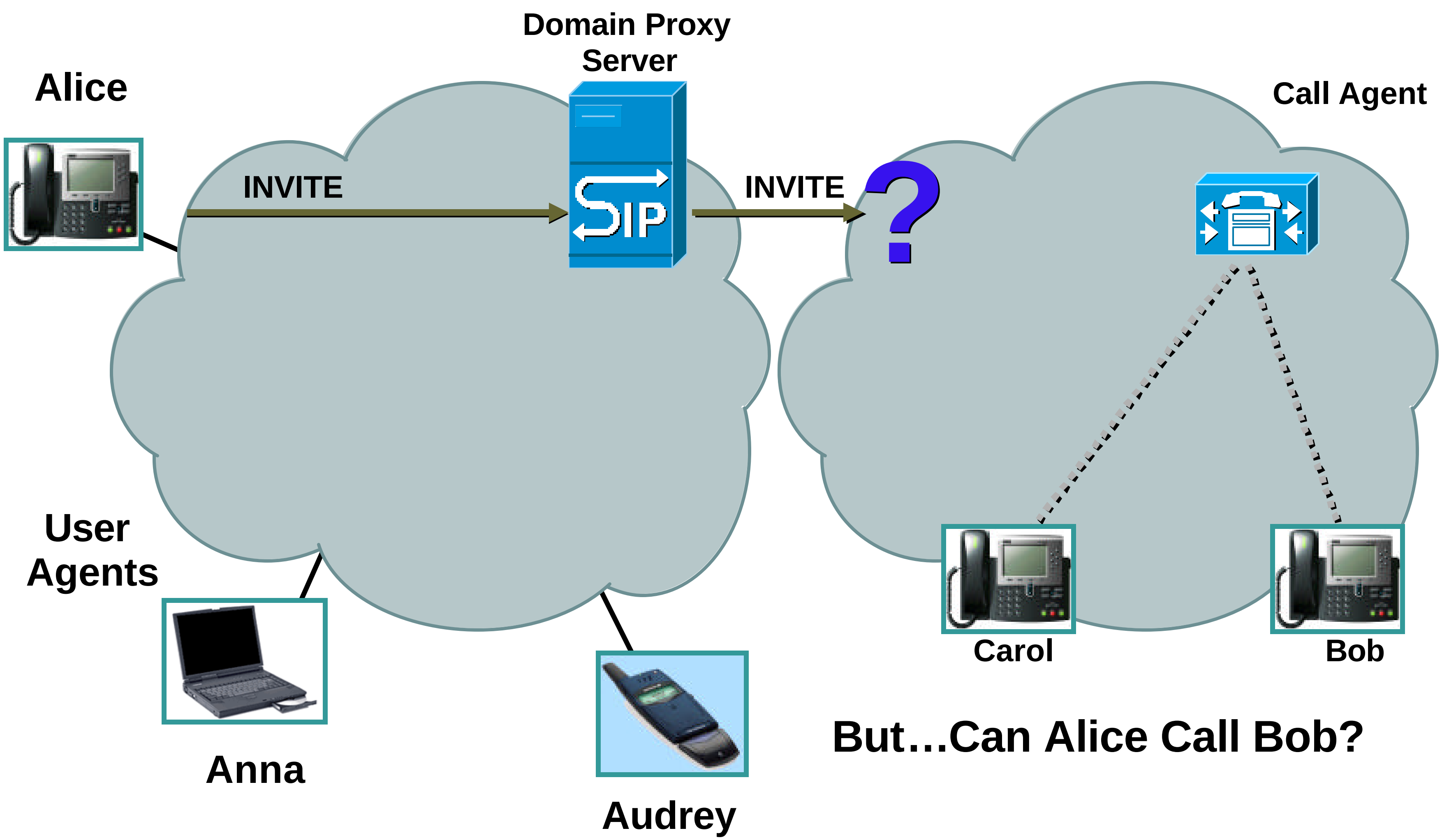
Carol Calls Bob

Peer-to-Peer to/from Client/Server Interworking



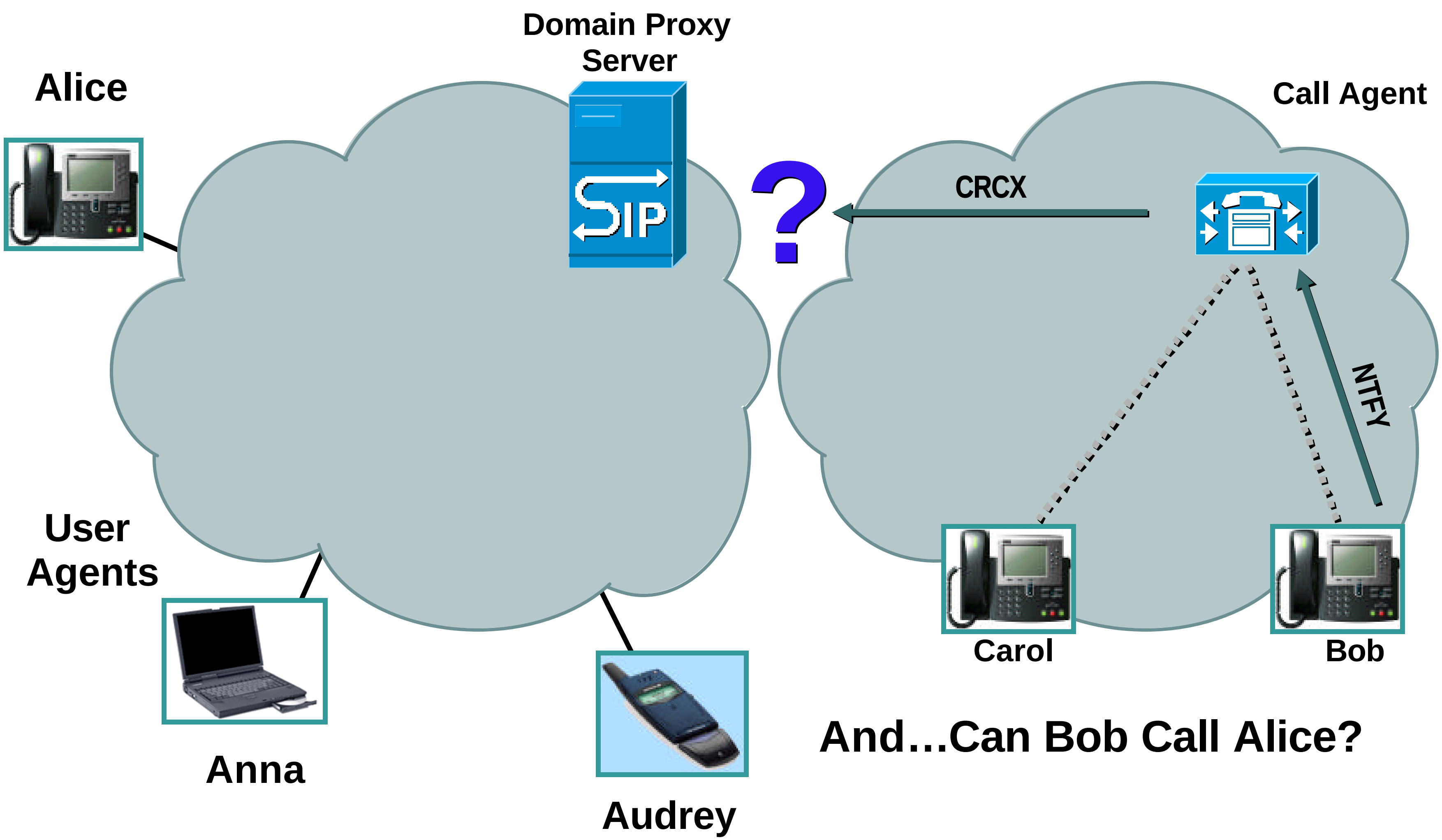
But…Can Alice Call Bob?

Peer-to-Peer to/from Client/Server Interworking



But…Can Alice Call Bob?

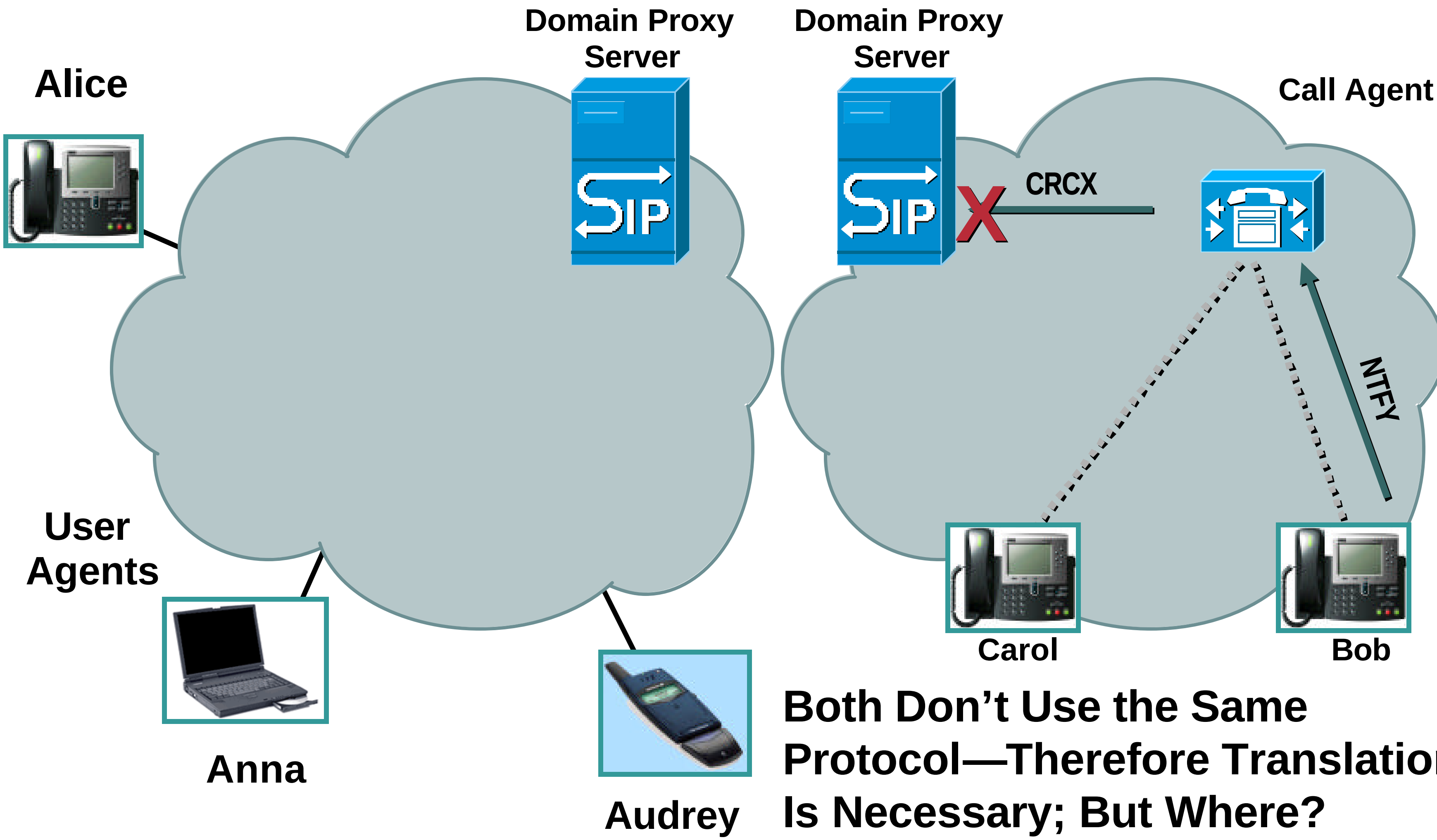
Peer-to-Peer to/from Client/Server Interworking



And…Can Bob Call Alice?

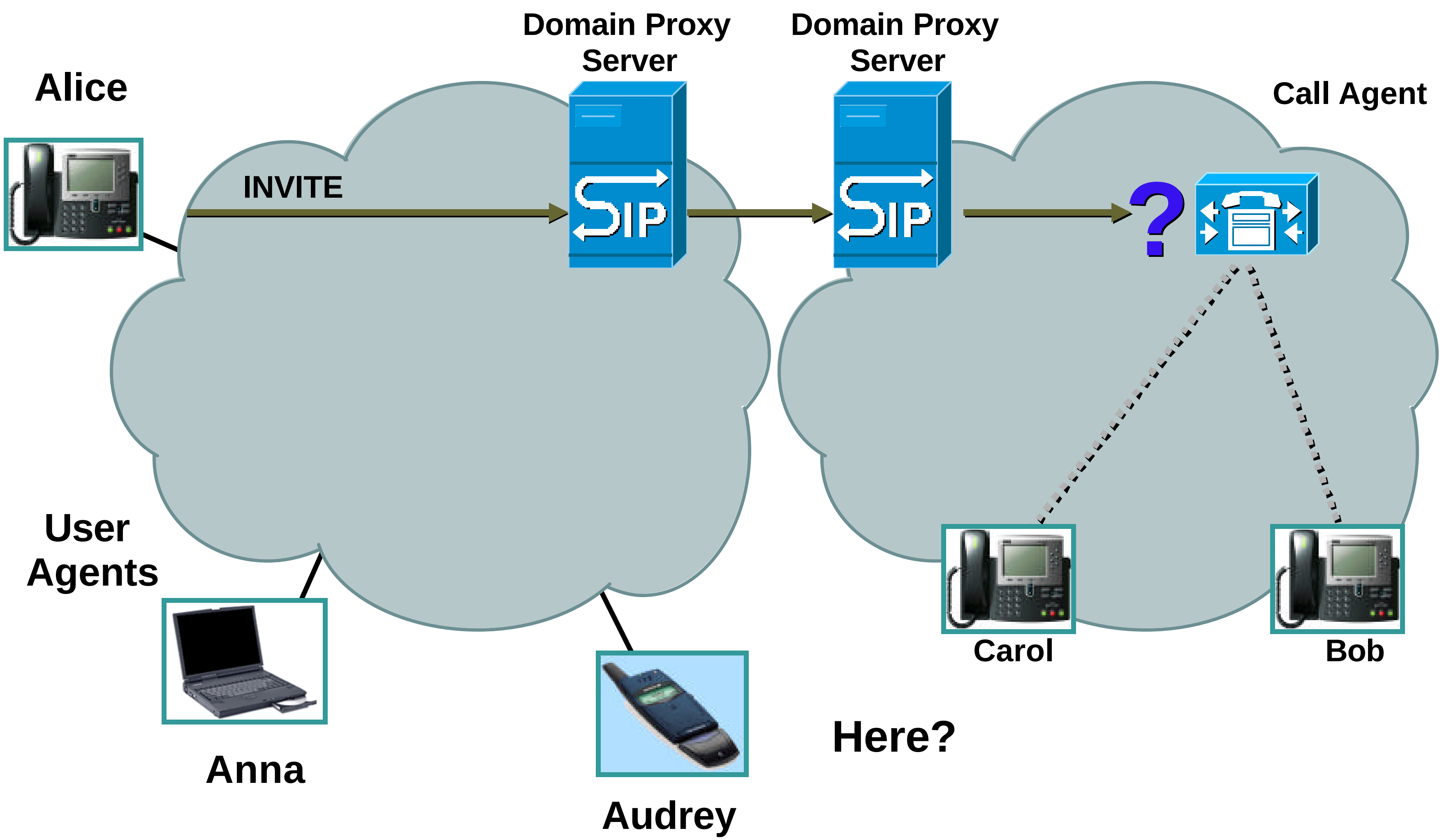
Peer-to-Peer to/from Client/Server Interworking

Cisco.com

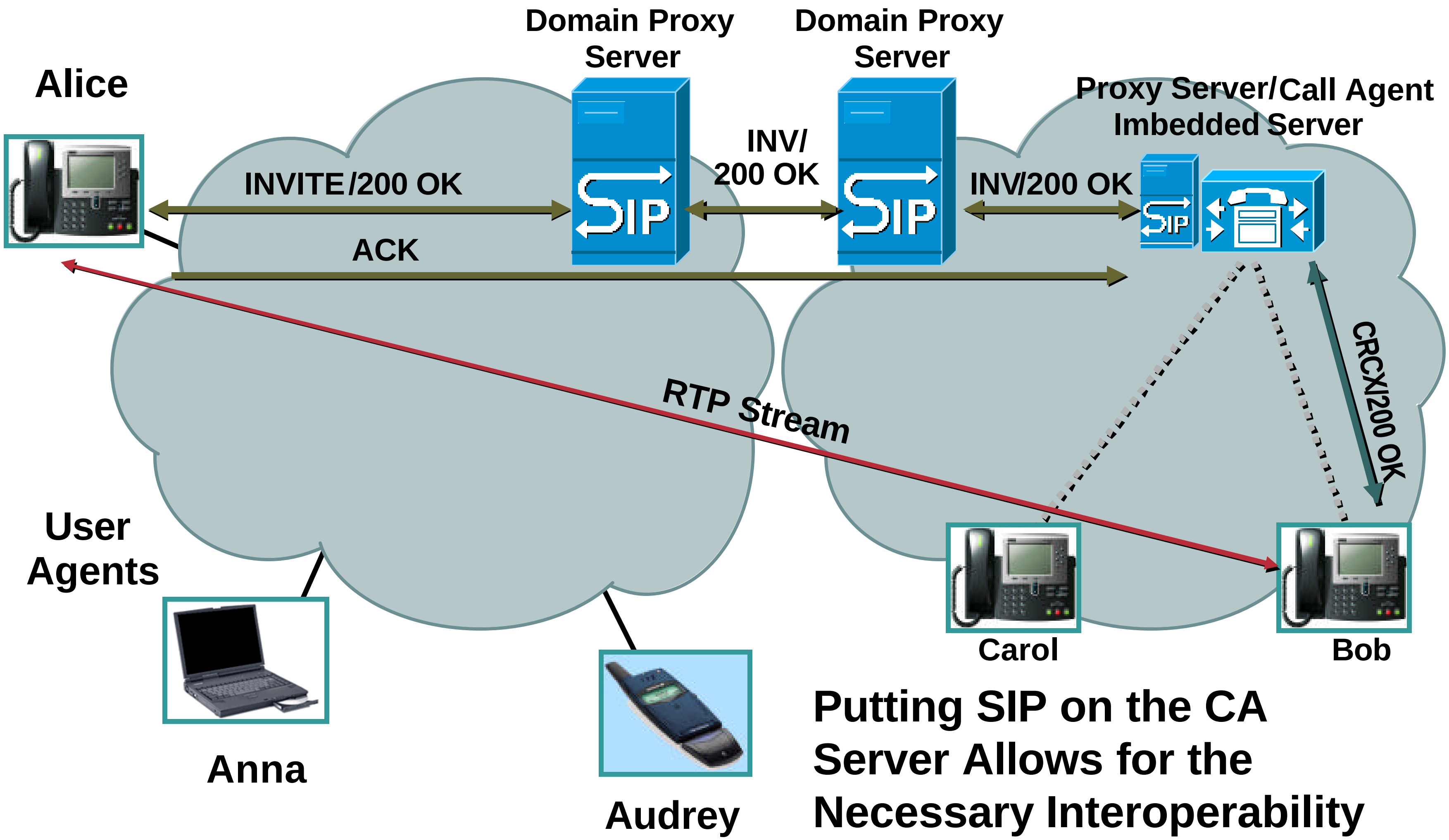


Both Don't Use the Same Protocol—Therefore Translation Is Necessary; But Where?

Peer-to-Peer to/from Client/Server Interworking



Peer-to-Peer to/from Client/Server Interworking



Putting SIP on the CA Server Allows for the Necessary Interoperability

SIP to H.323 Interworking Architectures

Below are some common architectures involving an IWF:

- **Basic Configuration:**

  SIP UA -- IWF -- H.323 Endpoint

- **Calls using H.323 GK:**

  SIP UA -- IWF -- H.323 GK -- H.323 Endpoint

- **Calls using SIP proxies:**

  SIP UA -- SIP proxies -- -- IWF H.323 Endpoint

- **Calls using both SIP proxy and H.323 GK:**

  SIP UA -- SIP proxies -- IWF -- H.323 GK -- H.323 Endpoint

- **SIP trunking between H.323 networks:**

  H.323 Endpoint -- IWF -- SIP network -- IWF -- H.323 Endpoint

- **H.323 trunking between SIP networks:**

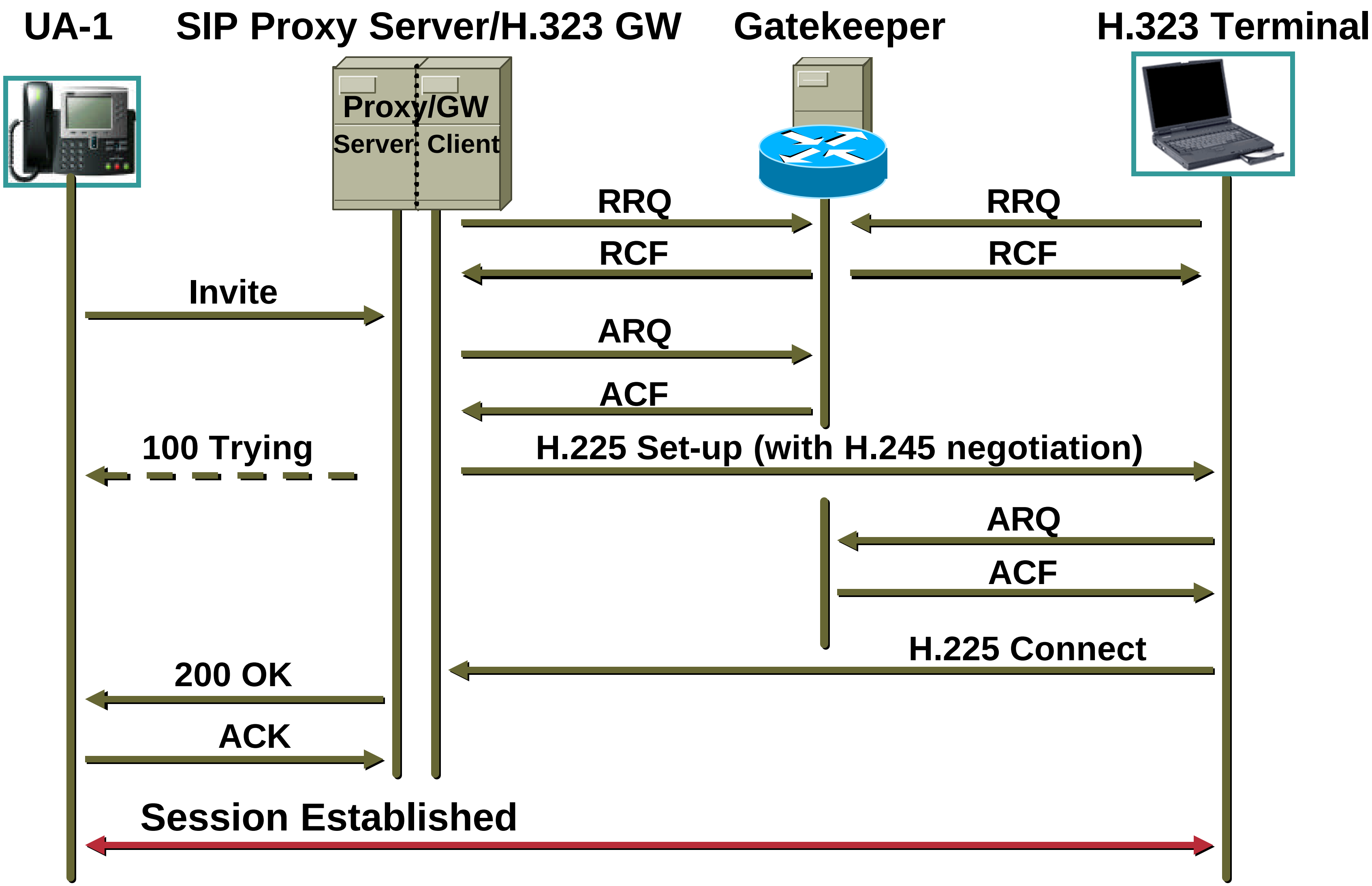
  SIP UA -- IWF -- H.323 network -- IWF -- SIP UA

SIP to H.323 Interworking Scenarios

Some possible call scenarios for the previous configurations are:

- Simple call from H.323 terminal to SIP UA;
- Call from H.323 terminal to SIP UA using H.245 tunneling;
- Call from H.323 terminal to SIP UA using early H.245;
- Call from H.323 terminal to SIP terminal using H.323 Fastconnect procedure;
- Call from H.323 terminal to SIP terminal using overlapped sending
- Call from H.323 terminal to SIP terminal using pre-granted ARQ (for configurations having H.323 GK);
- Simple call from SIP UA to H.323 terminal;
- Call from SIP UA to H.323 terminal using H.245 tunneling.
- Call from SIP UA to H.323 terminal using early H.245;
- Call from SIP UA to H.323 terminal using H.323 fast connect procedure;
- Call from SIP UA to H.323 terminal using overlapped sending;
- Call from SIP UA to H.323 terminal using pre-granted ARQ (for configuration having H.323 GK);
- Call from SIP UA to SIP UA using H.323 trunking between two IWFs;
- Call from a H.323 terminal to another H.323 terminal using SIP trunking between two IWFs.

SIP to H.323v2 Interworking Example



Complete Your Online Session Evaluation!

Por favor, complete el formulario de evaluación.

Muchas gracias.

Session ID: VVT-2001

Intermediate SIP for Voice, Video and Instant Messaging

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