SIP for Voice, Video and Instant Messaging

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Agenda

- SIP History & Basics
- Elements of SIP
- SIP Message details
- SIP Signaling
- Interworking with MGCP/H.323

VoIP Signaling Overview

- Peer-to-Peer Signaling Protocols
  - H.323 Signaling Architecture
  - SIP Signaling Architecture
- Client/Server Signaling Protocols
  - MGCP Architecture
  - MEGACO/H.248 Architecture
  - Skinny (SCCP)
- Audio (Bearer) Path
  - Always Real Time Protocol (RTP)

VoIP Signaling Overview

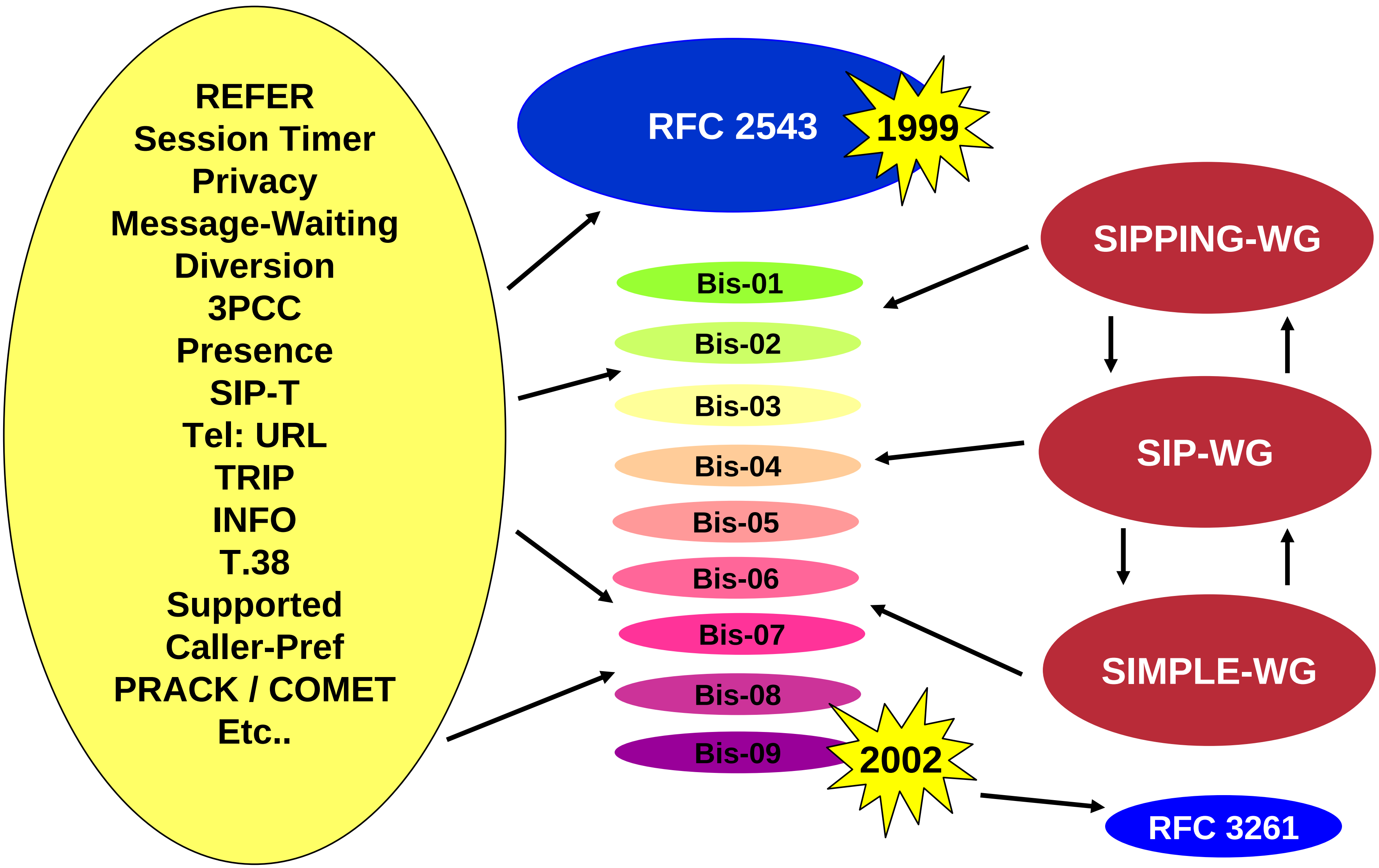
- **H.323 uses TCP or UDP**
  - **First to Market!**
  - **Complexity of Protocols: H.225, H.245**
  - **Intelligent endpoints**
  - **Gatekeeper routed call models**
- **MGCP uses UDP**
  - **Centralized Call Control**
  - **Decoupling of Call & Media Control**
  - **Event Triggered**

Session Initiation Protocol

- The **S**ession **I**nitiation **P**rotocol (SIP) is an application layer control (signaling) protocol for creating, modifying and terminating multimedia sessions with one or more participants
- SIP defined by IETF working group as RFC 2543, in March 1999, work continues within SIP WG

  Additional “feature” drafts have been written to address features & issues which concern SS7/ISUP handling, QoS, Alerting, DHCP, Firewalls & NAT, … etc.

SIP @ the IETF – A lot going on



SIP – What Is it?

SIP entities can communicate via:

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

in IPv4 and/or IPv6 environments using:

- – UDP
- – TCP
- – SCTP or
- – TLS over TCP

on a per SIP hop basis.

SIP – What Is it? (Cont.)

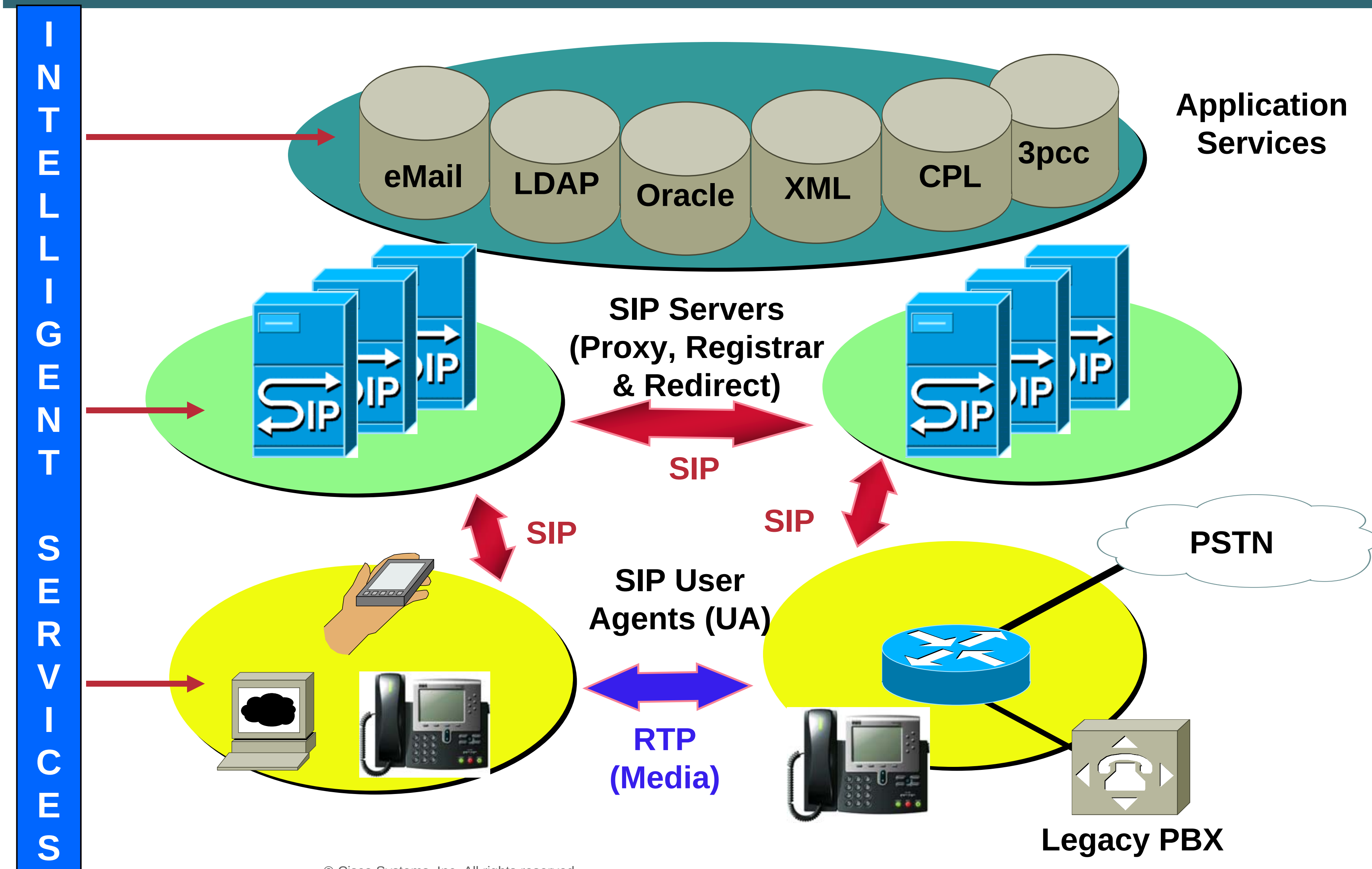
SIP is a simple extensible protocol

- **Methods** - Define the transaction
- **Headers** - Describe the transaction
- **Message Body** - SDP, Text, XML, jpeg, etc...

SIP Basic - Architecture

- SIP utilizes an Internet architecture, similar to the World Wide Web - intelligent clients utilizing services within the network
  - Uses URL style addresses and syntax
  - MIME definition for multimedia (SDP)
- Applications and SIP Services can be distributed throughout the network, with intelligence in the clients or in applications our the network
  - Additional services like Presence, Mobility, Find-Me/Follow-Me, etc.. are added into the network via basic SIP functionality or Application servers

SIP Basic - Architecture



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

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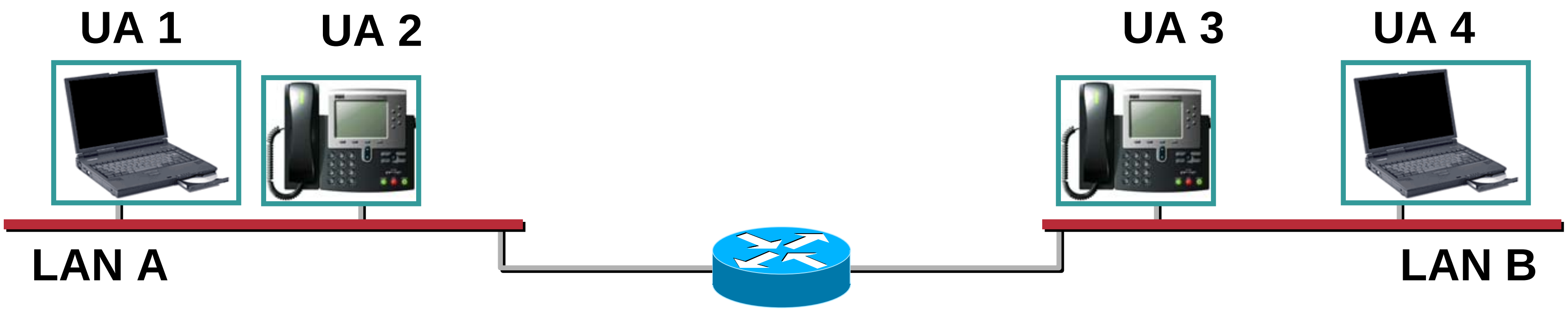
5 Components of SIP

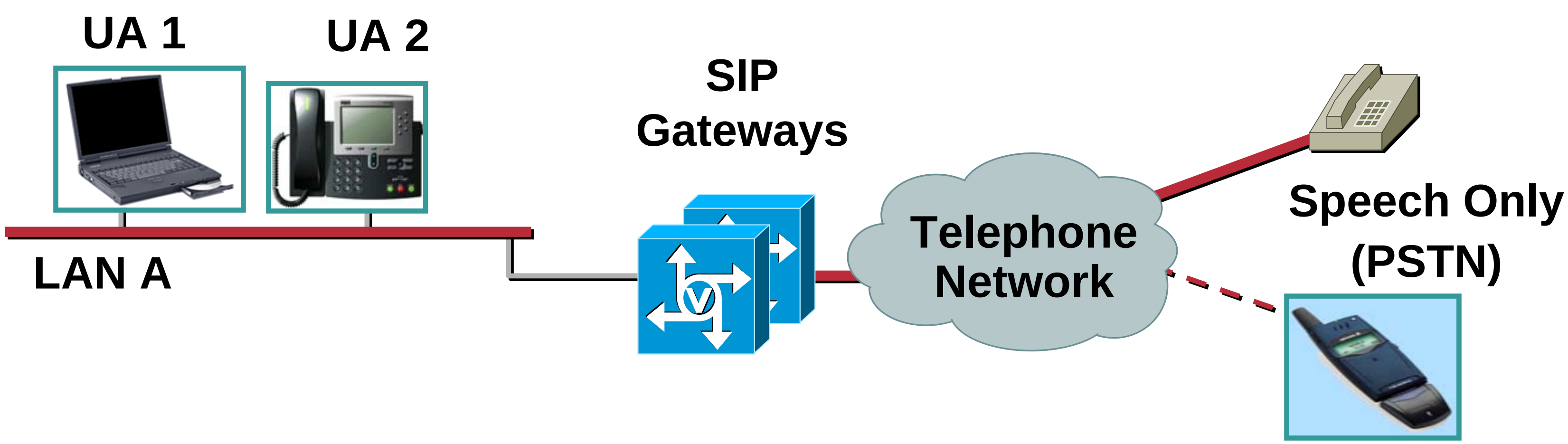
- **User Agent [Mandatory]**
- **SIP Gateway [*optional*]**
- **Registrar Server [*optional*]**
- **Proxy Server [*optional*]**
- **Redirect Server [*optional*]**

SIP User Agents



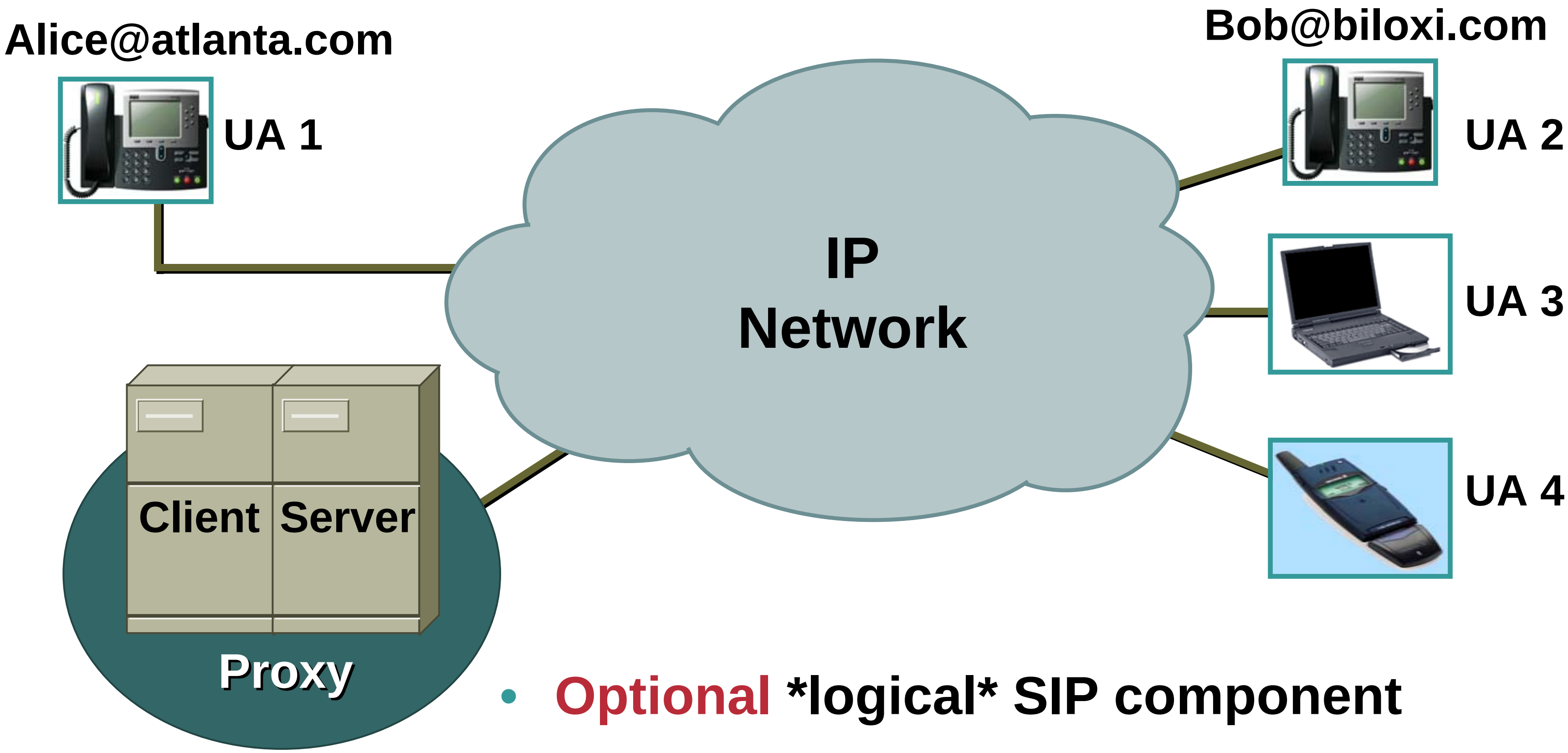
- **Client—Server model**
  - **User Agent Client (UAC)—Initiates sessions**
  - **User Agent Server (UAS)—Responds to session requests**
- **User Agent = UAC + UAS**

SIP Gateways



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

SIP Proxy Server



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

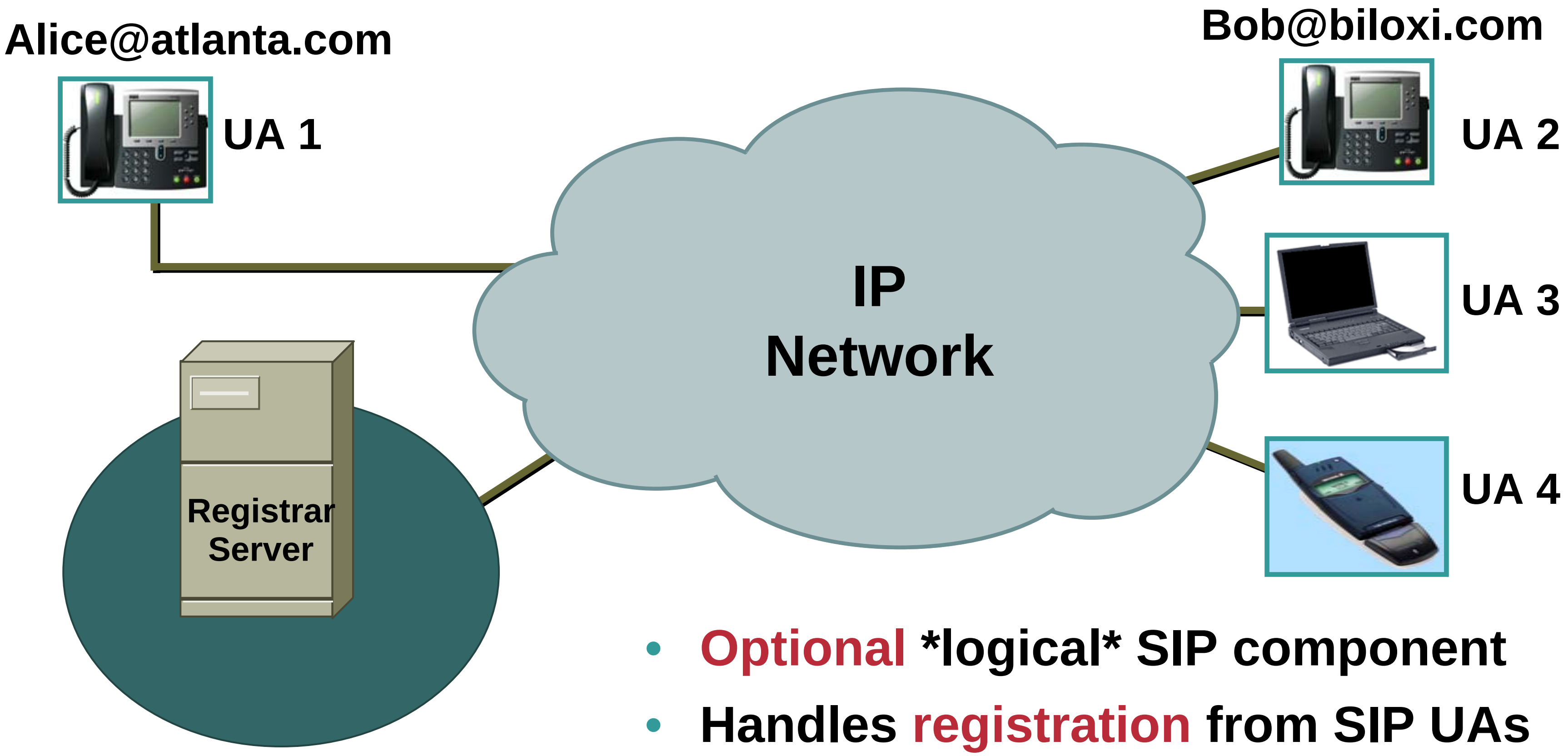
SIP Proxy Server

- **Handles the routing of SIP messages, but does not initiate any new SIP messages**
- **When a SIP Proxy is present, signaling can be routed in a similar manner to H.323 Gatekeeper Routed Signaling (GKRCS), where all signaling flows through the Proxy**

  SIP proxies can insert a *Record Route:* header into an INVITE message; when a record route header is inserted, all signaling messages flow through the proxy; this is useful for billing, or feature support

- **SIP Proxy can use any database, Registrar Server or DNS SRV query to determine the location of the next-hop of the message**

SIP Registrar Server

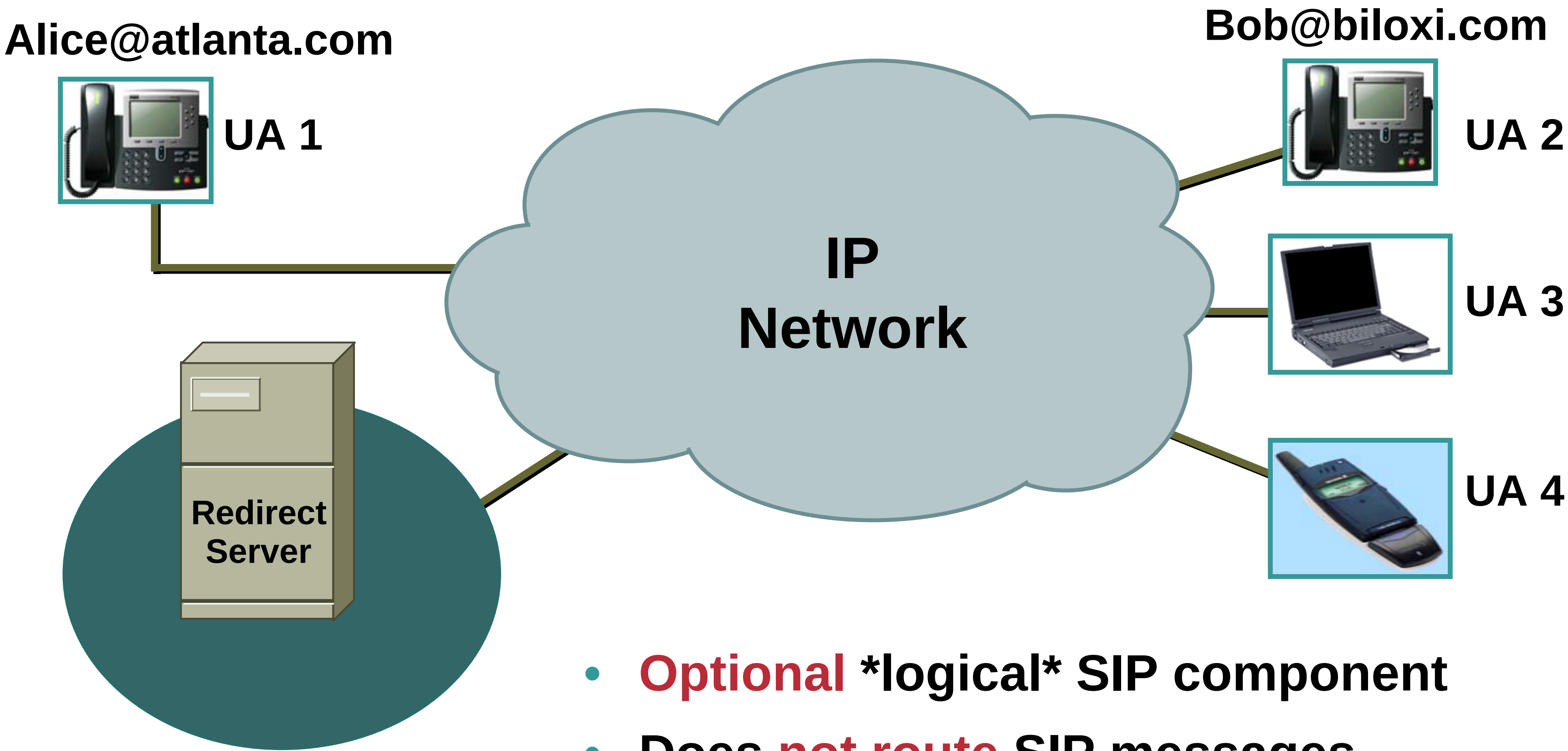


- Optional *logical* SIP component
- Handles registration from SIP UAs
- Also a mechanism for “Presence”
- Does not route SIP messages

SIP Registrar Server

- **Can be on the separate box, or a process that resides on a SIP Proxy or Redirect Server**
- **Registrar server is the device that handles SIP REGISTER messages from non-Gateway SIP User Agents**
  - **Registrar server stores the (*Contact:*) header from a User Agent REGISTER messages for location services**
  - **Once a SIP User Agent is registered within a domain, the domain Proxy Server is able to route session requests to that user (agent) properly**
- **Details from REGISTER messages are used by the Translation & Routing functionality of Proxy & Redirect servers.**

SIP Redirect Server

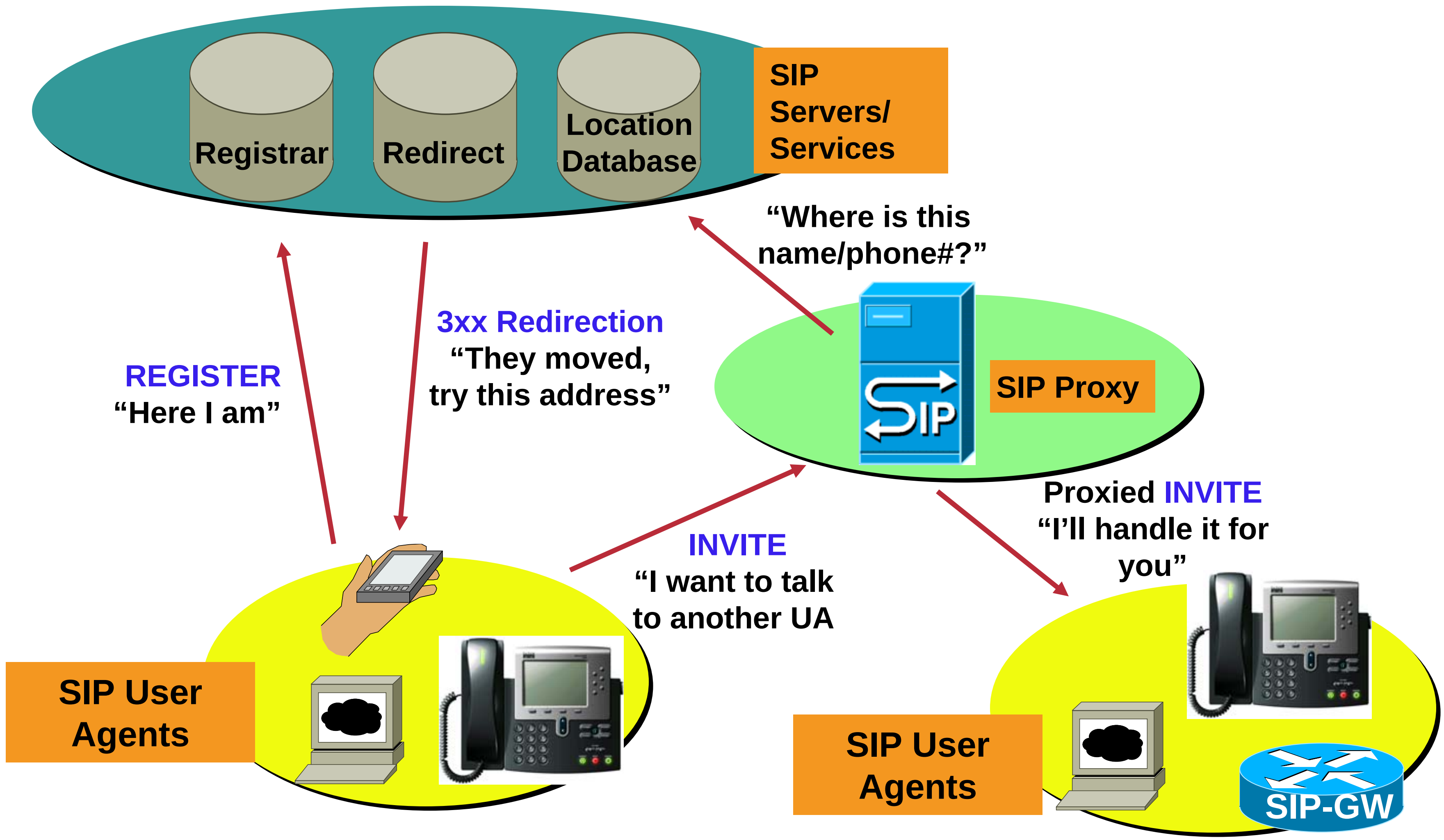


- Optional *logical* SIP component
- Does not route SIP messages
- Returns a *redirect* to UAC for directed routing to the given "new" destination

SIP Servers—Redirect Server

- **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 components in “action”



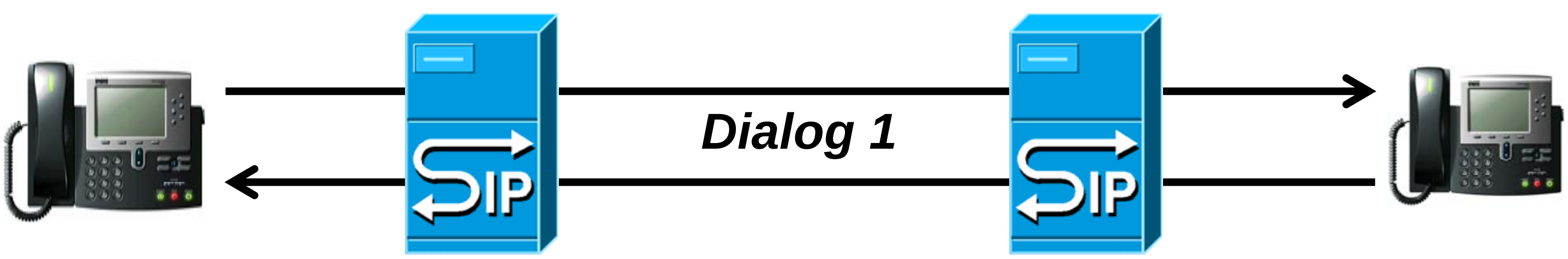
Back-to-Back User Agent

- **SIP Requests can be managed by intermediate components such as proxy servers**
- **Proxy servers have limited ability to modify SIP messages**

  Must obey strict rules regarding the modification of SIP headers

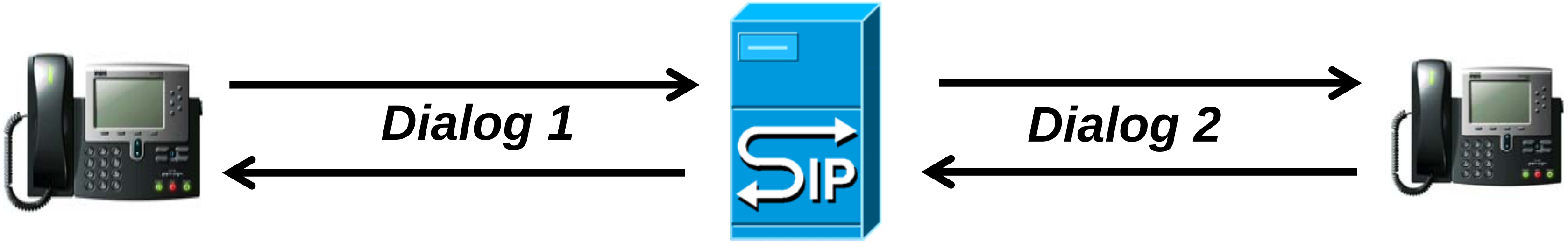
  Can’t touch SIP bodies, where the session’s media is defined

- **The dialog remains end-to-end**



Back-to-Back User Agent

- A commonly-adopted model, called a back-to-back user agent (B2BUA), combines a UAC and a UAS so that a request received by the UAS is reissued by the co-resident UAC
- The B2BUA generates a completely independent outgoing dialog, which affords it the ability to synthesize SIP headers and bodies of its choosing
- B2BUAs are inherently more stateful than proxy servers or redirect servers, and can more easily inter-work SIP with other protocols



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Example IPv4 SIP Packet Format with UDP

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 INVITE

SIP Header

INVITE sip:23198@172.17.207.91:5060 SIP/2.0
Via: SIP/2.0/UDP 10.80.17.134:5060
Via: SIP/2.0/UDP 172.18.192.232:5060;branch=1FV1xhfvxGJOK9rWcKdAKOA
To: <sip:23198@172.18.192.232>;tag=abc
From: <sip:15691@10.80.17.134>;tag=a73kszlfl
Call-ID: c2943000-50405d-6af10a-382e3031@10.80.17.134
CSeq: 100 INVITE
Contact: sip:15691@10.80.17.134:5060
Expires: 180
Content-Type: application/sdp
Content-Length: 219
User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled
Accept: application/sdp
Record-Route: <sip:23198@172.18.192.232:5060;maddr=172.18.192.232>

body

v=0
o=CiscoSystemsSIP-IPPhone-UserAgent 17045 11864 IN IP4 10.80.17.134
s=SIP Call
c=IN IP4 10.80.17.134
t=0 0
m=audio 29118 RTP/AVP 0 101
a=rtpmap:0 pcmu/8000
a=rtpmap:101 telephone-event/8000

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

- Mixed addresses

  sip:14085551234@10.1.1.1; user=phone

  sip:jdoe@10.1.1.1

- Secure SIP Messaging (indicates TLS is used) [RFC 4346]

  sips:jdoe@cisco.com

    called a "SIPS-URI" or just "SIPS"

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

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

RFC 2327, 3264

Session Description Protocol (SDP)

- 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

RFC 2327, 3264

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 49172 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
```

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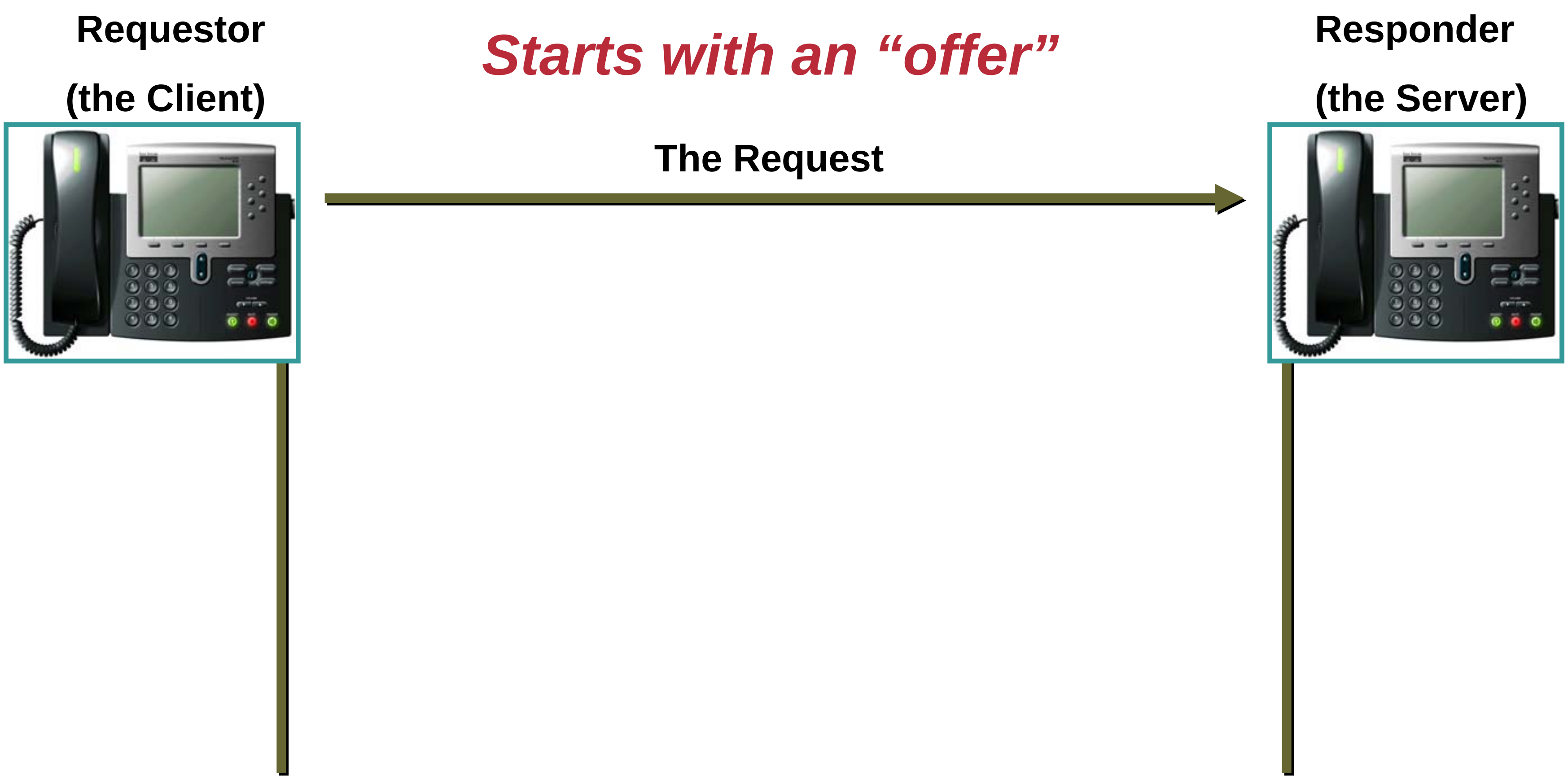
SIP Is a Request/Response Protocol

Starts with an "offer"



SIP Methods (Which Are Requests) from RFC 3261

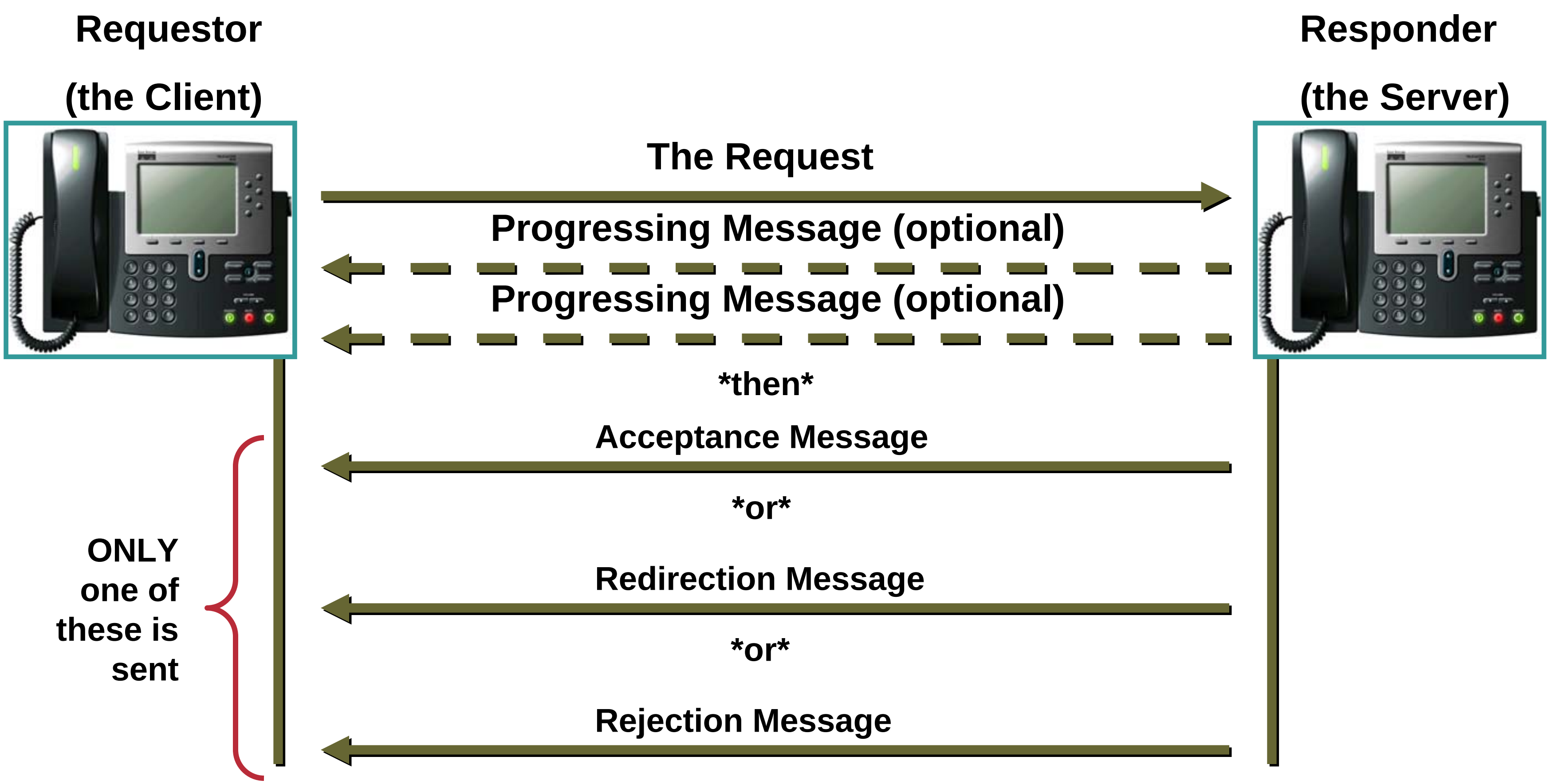
- **INVITE—A user or service is being invited to participate in a multimedia session**
- **ACK—Confirms that a client has received a final response to an INVITE request**
- **BYE—Terminates an existing session; can be sent by any user agent (in a multiparty session)**
- **CANCEL—Cancels pending requests; 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)

RFC 3264

SIP is a Request/Response Protocol



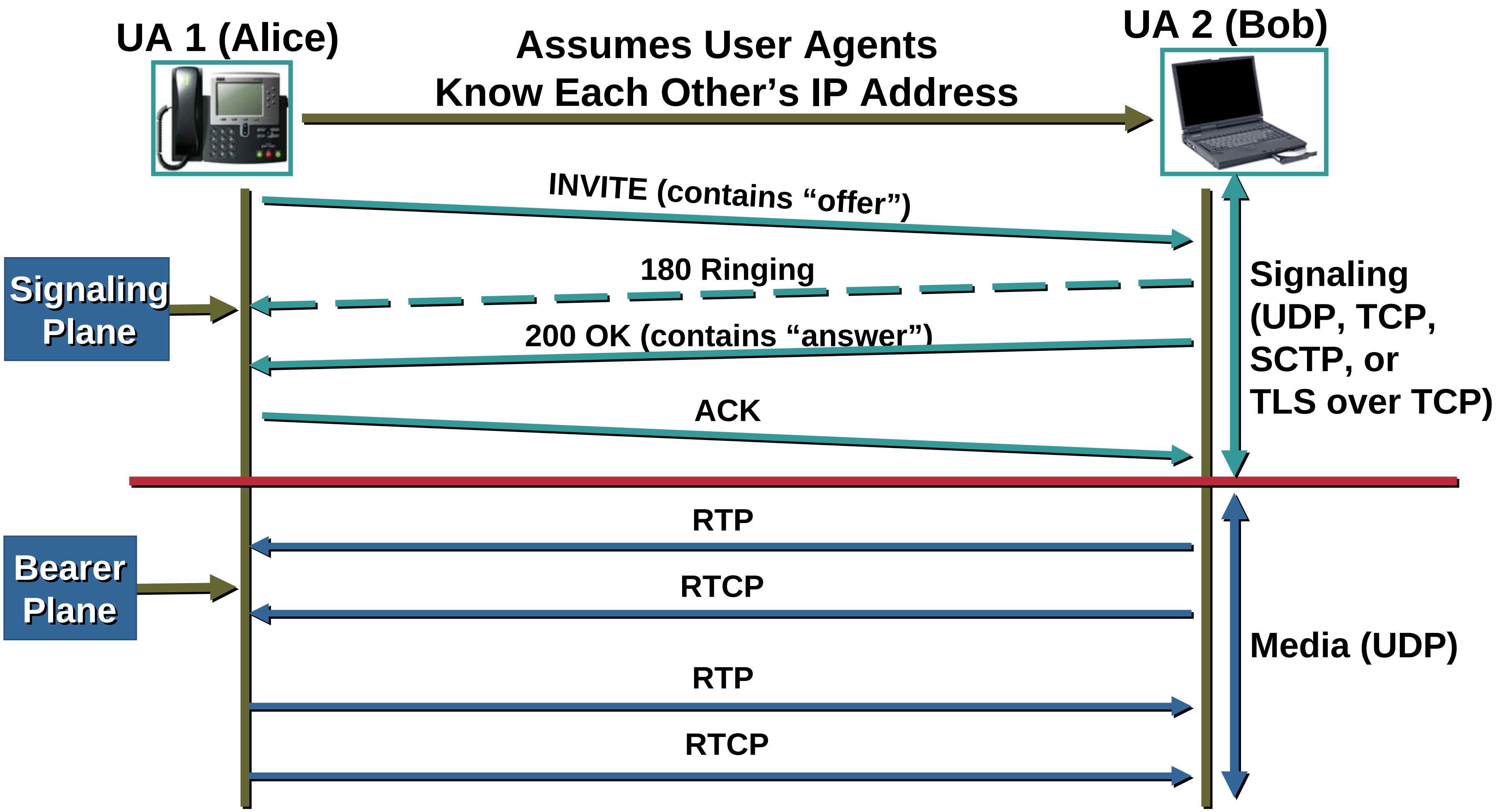
Expects an "answer"

RFC 3261

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

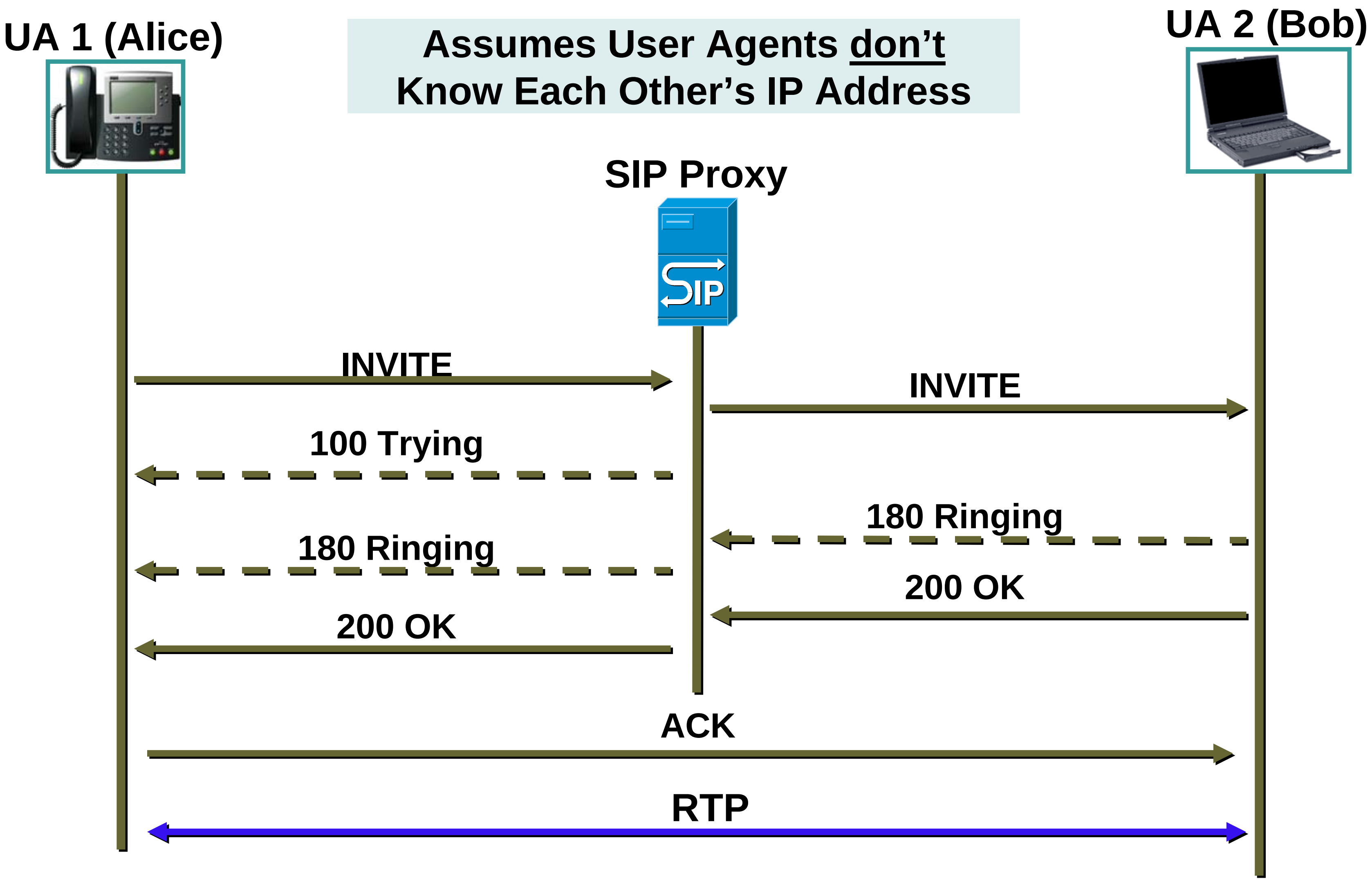
SIP Endpoint-to-Endpoint Signaling Without a Server



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

SIP Call Flow with Proxy Server

Assumes User Agents don't Know Each Other's IP Address



Stateless, Stateful and "Really" Stateful Proxy

Transaction Stateless

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

Transaction Stateful

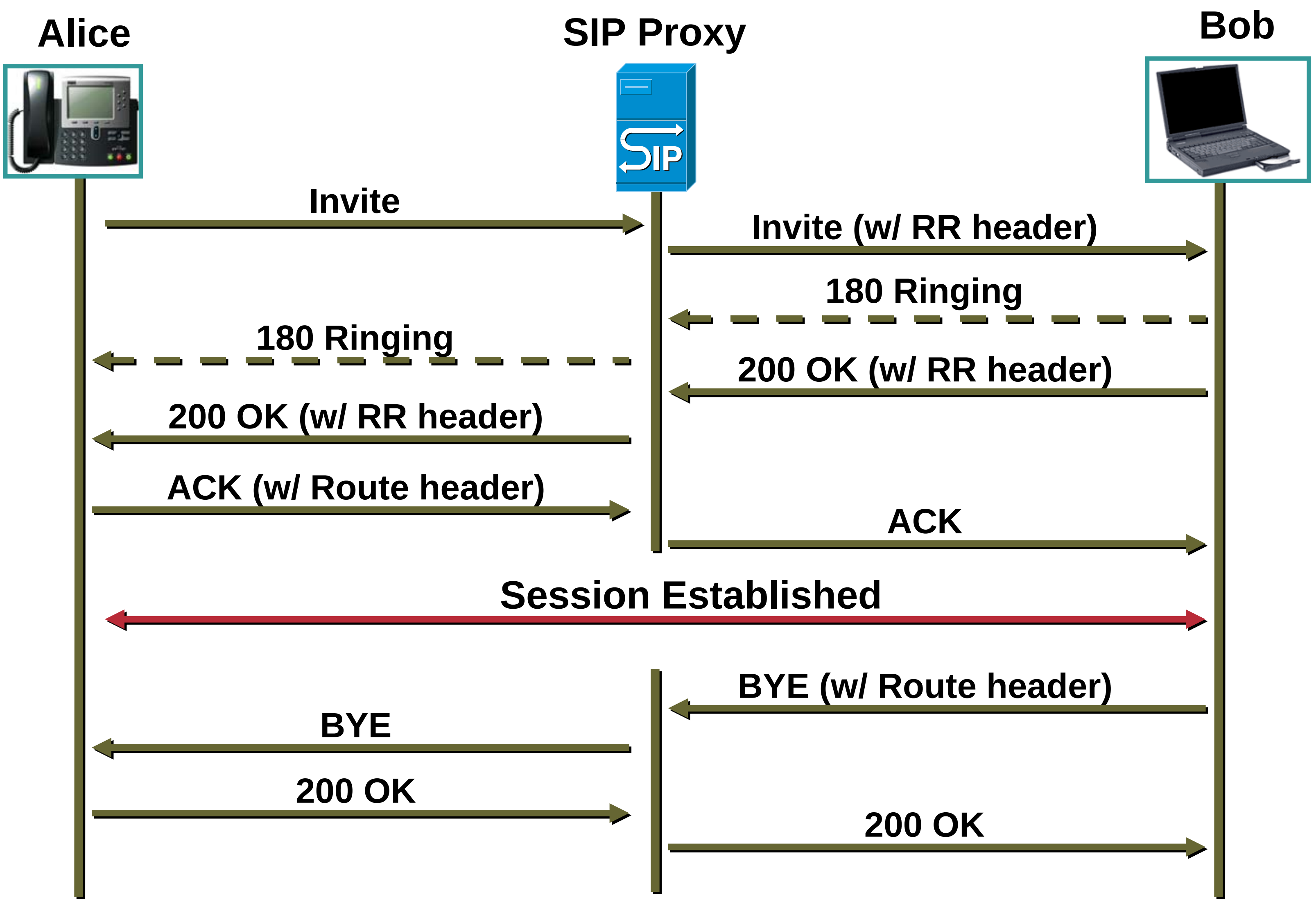
A Proxy Server that receives a SIP Request retains state of that transaction until that Server receives a Final Response (meaning a 2XX, 3XX, 4XX, 5XX or 6XX Response). Transaction Stateful has no knowledge of a session Update Request (UPDATE), a Transfer Request (REFER) or of a Termination Request (BYE)

Dialog Stateful

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

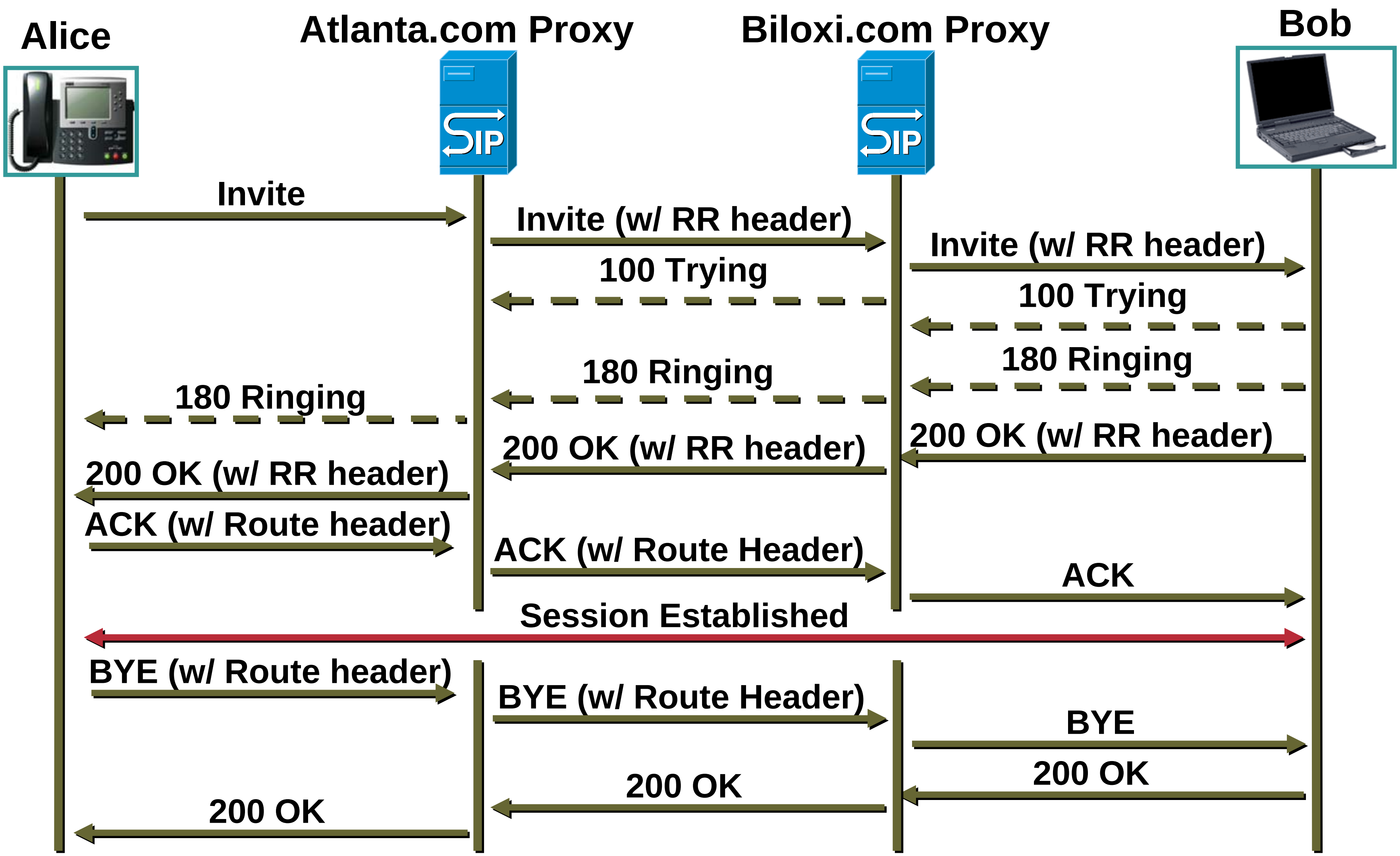
Dialog Stateful SIP Message Flow

RFC 3261

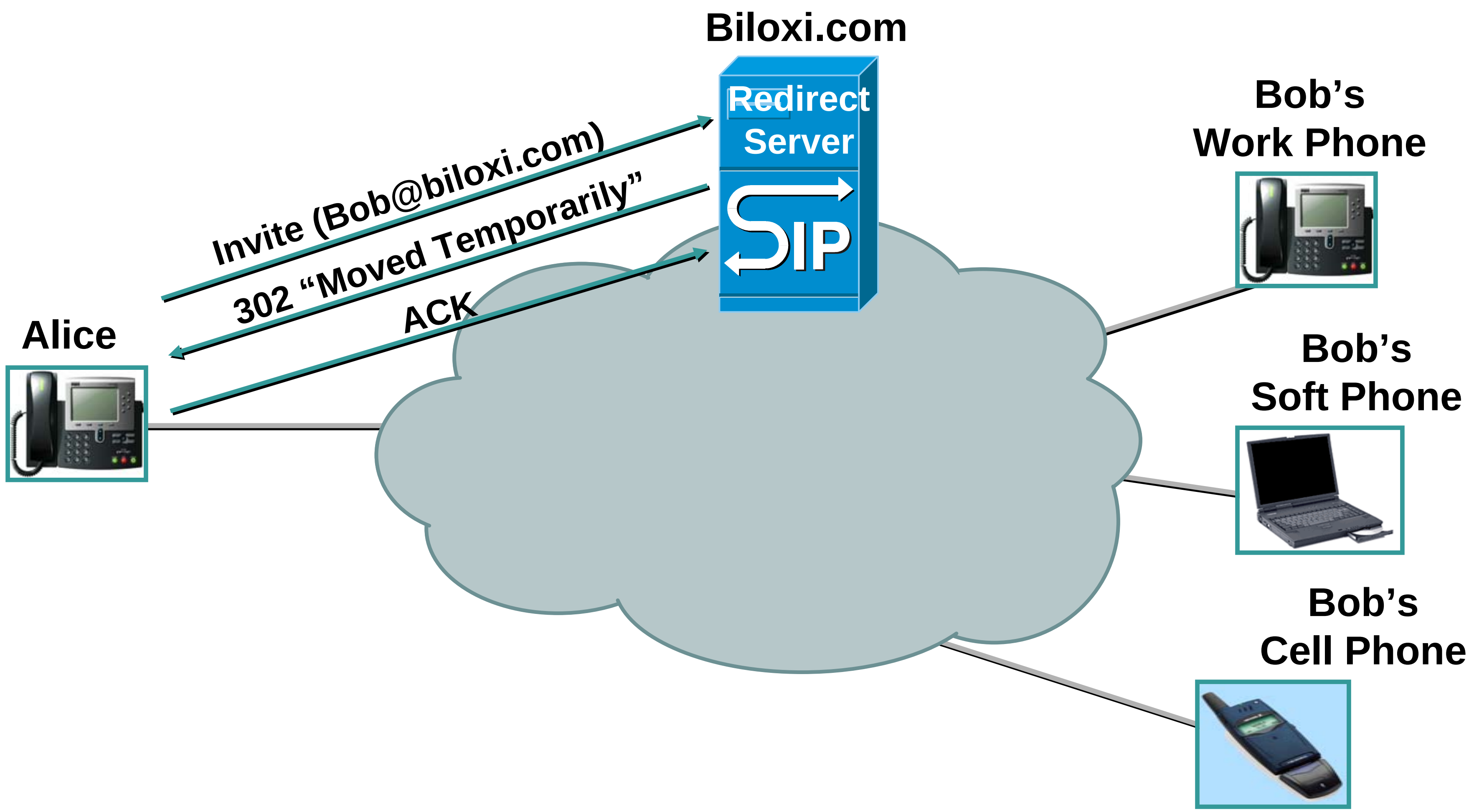


RFC 3261

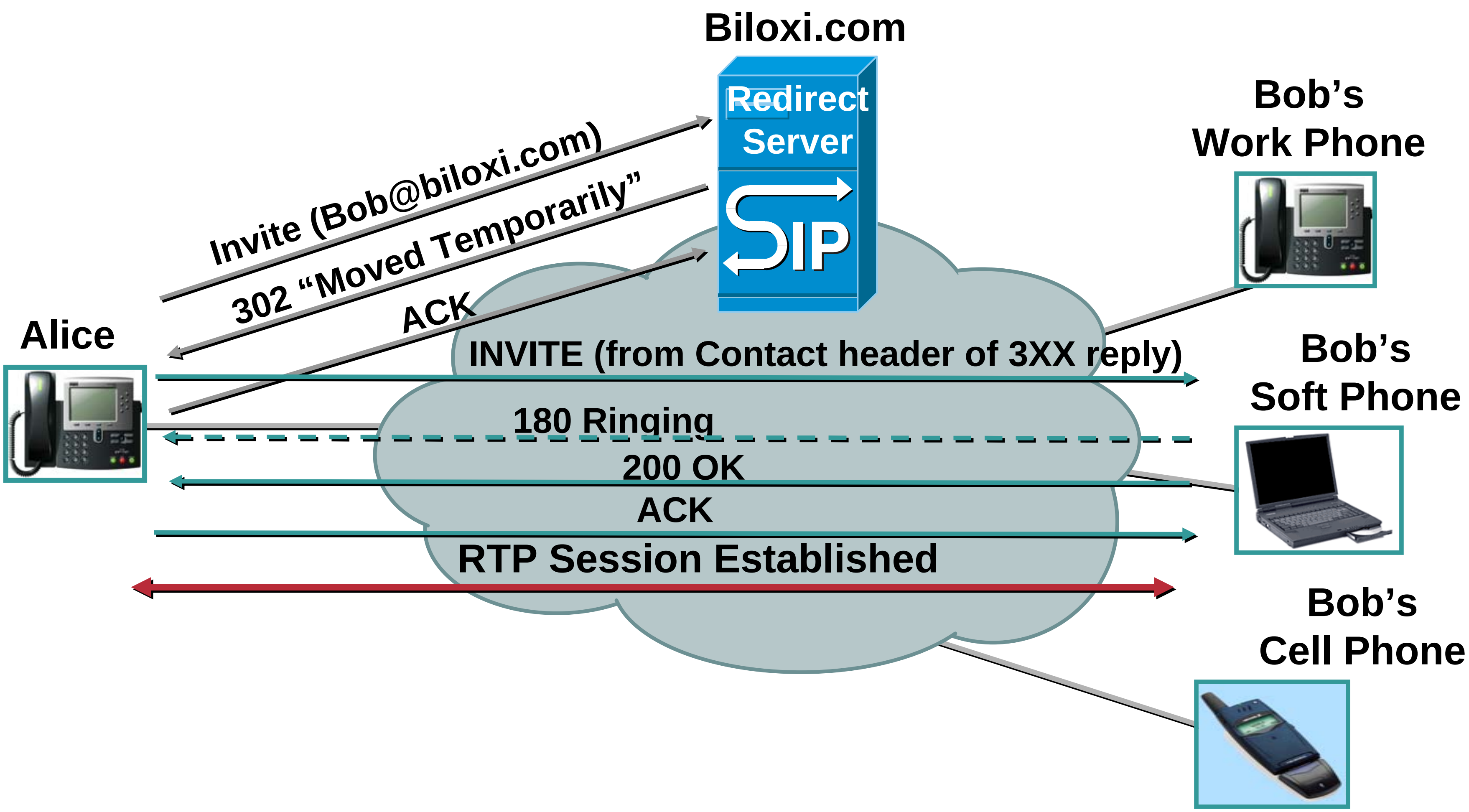
SIP Call Flow w/ 2 Proxies and Record Route



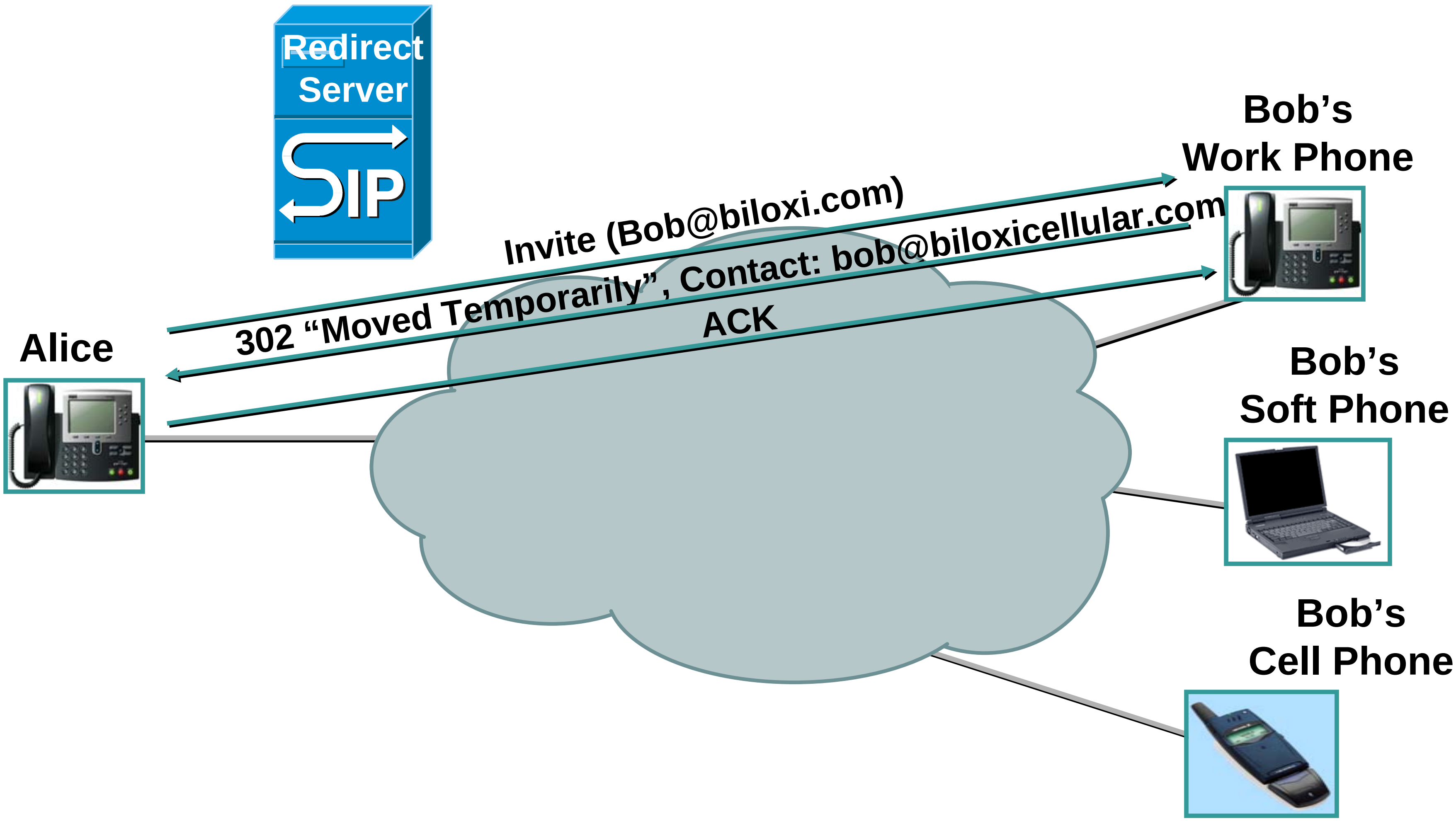
SIP Redirect Server



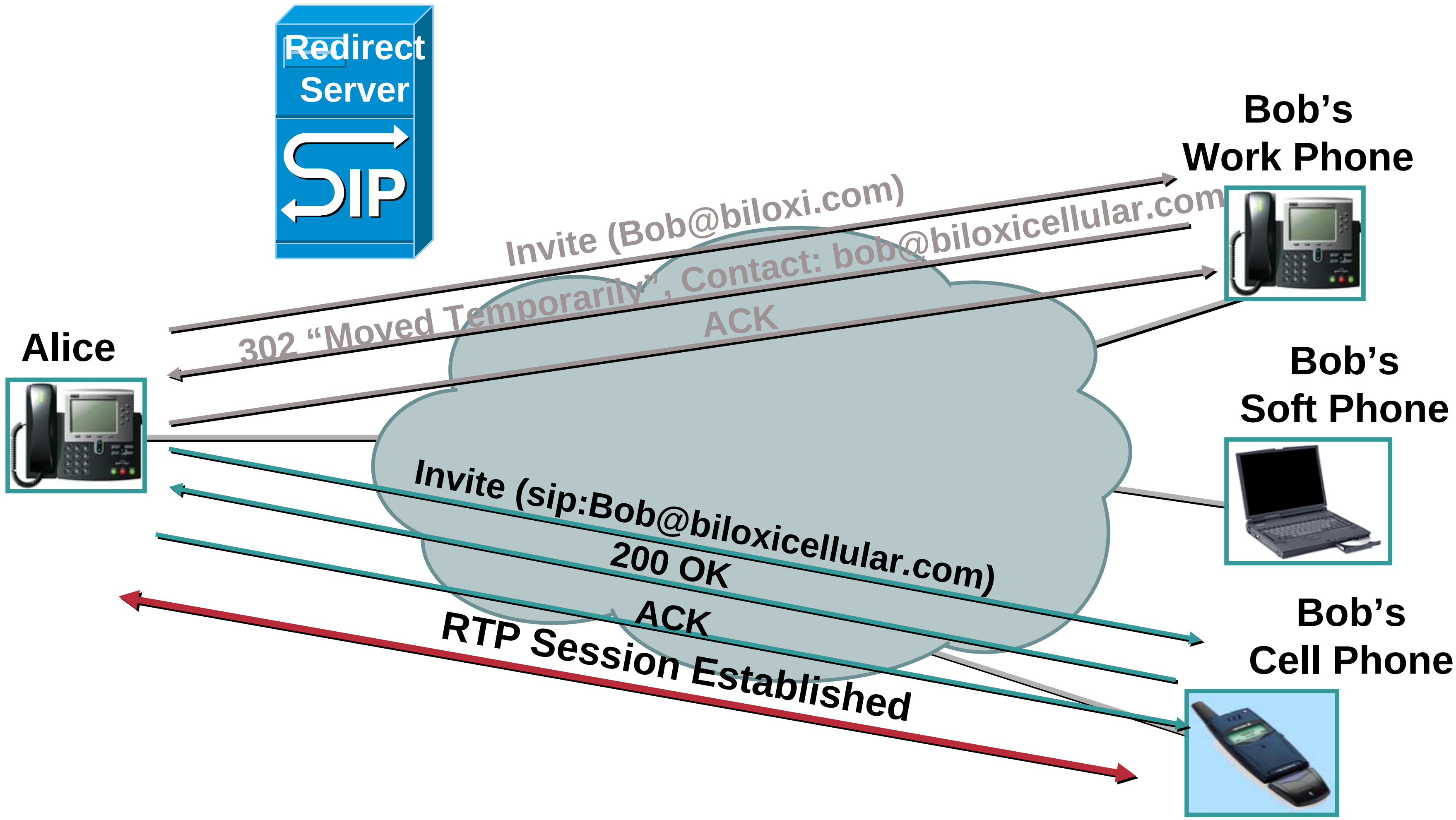
SIP Redirect Server



SIP Redirecting at the User Agent Server



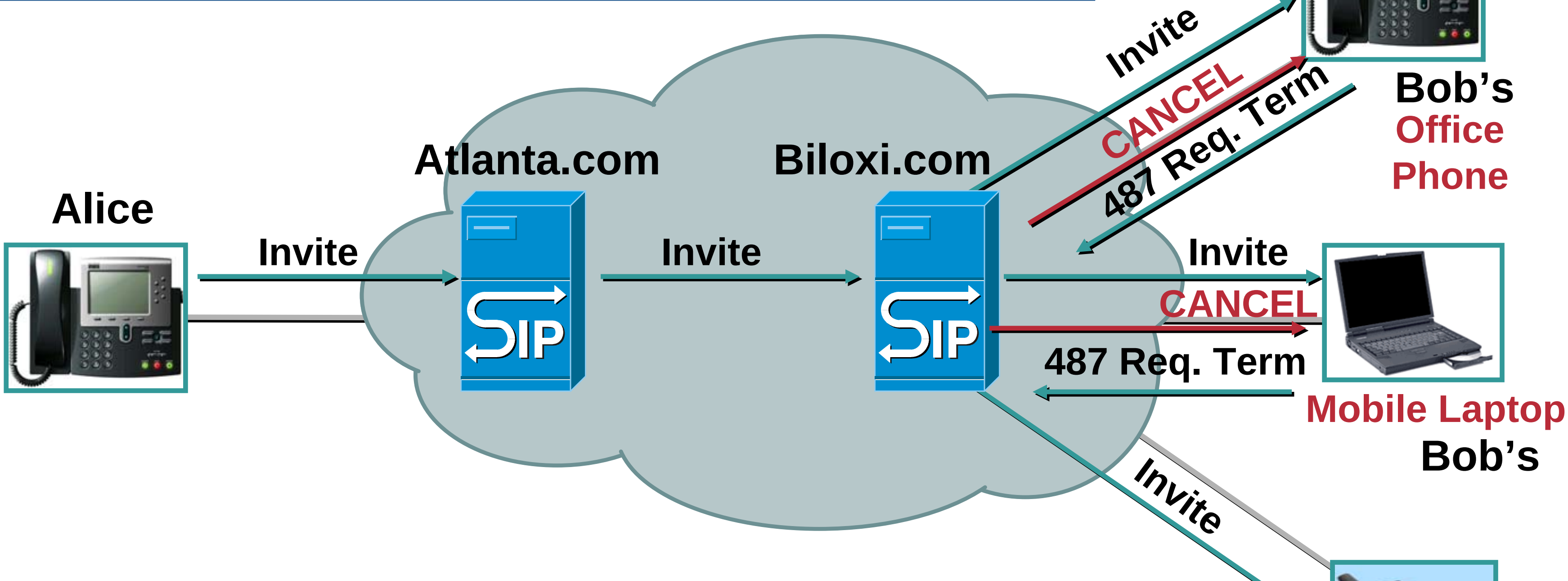
SIP Redirecting at the User Agent Server

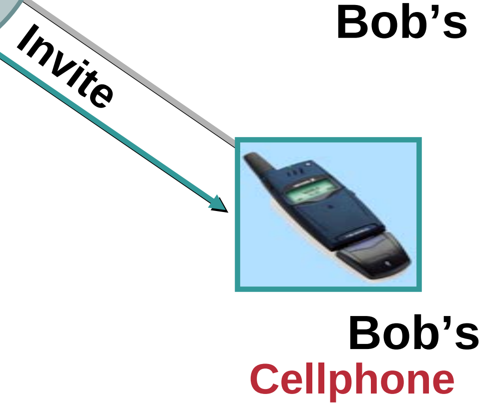


Call Forking (Sequentially)

RFC 3261

When Alice calls Bob, does she care which device Bob answers with?
In SIP, she's calling "Bob". Forking allows her to find "Bob".

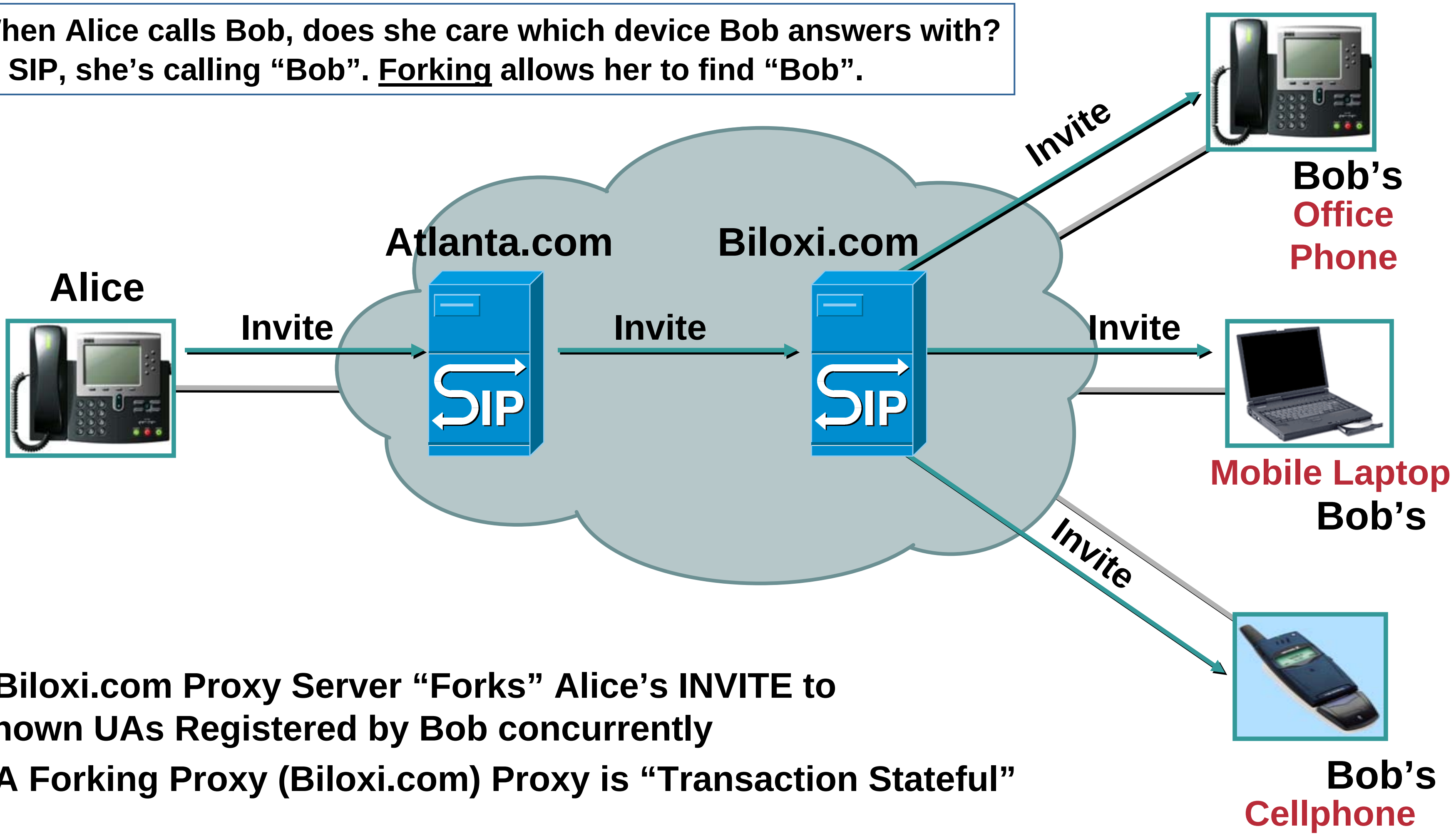




- Biloxi.com Proxy Server "Forks" Alice's INVITE to known UAs Registered by Bob (perhaps after 3 rings each) in a sequential pattern
- A Forking Proxy (Biloxi.com) Proxy is "Transaction Stateful"
- "branch" values are different per forked INVITE

Call Forking (Concurrently)

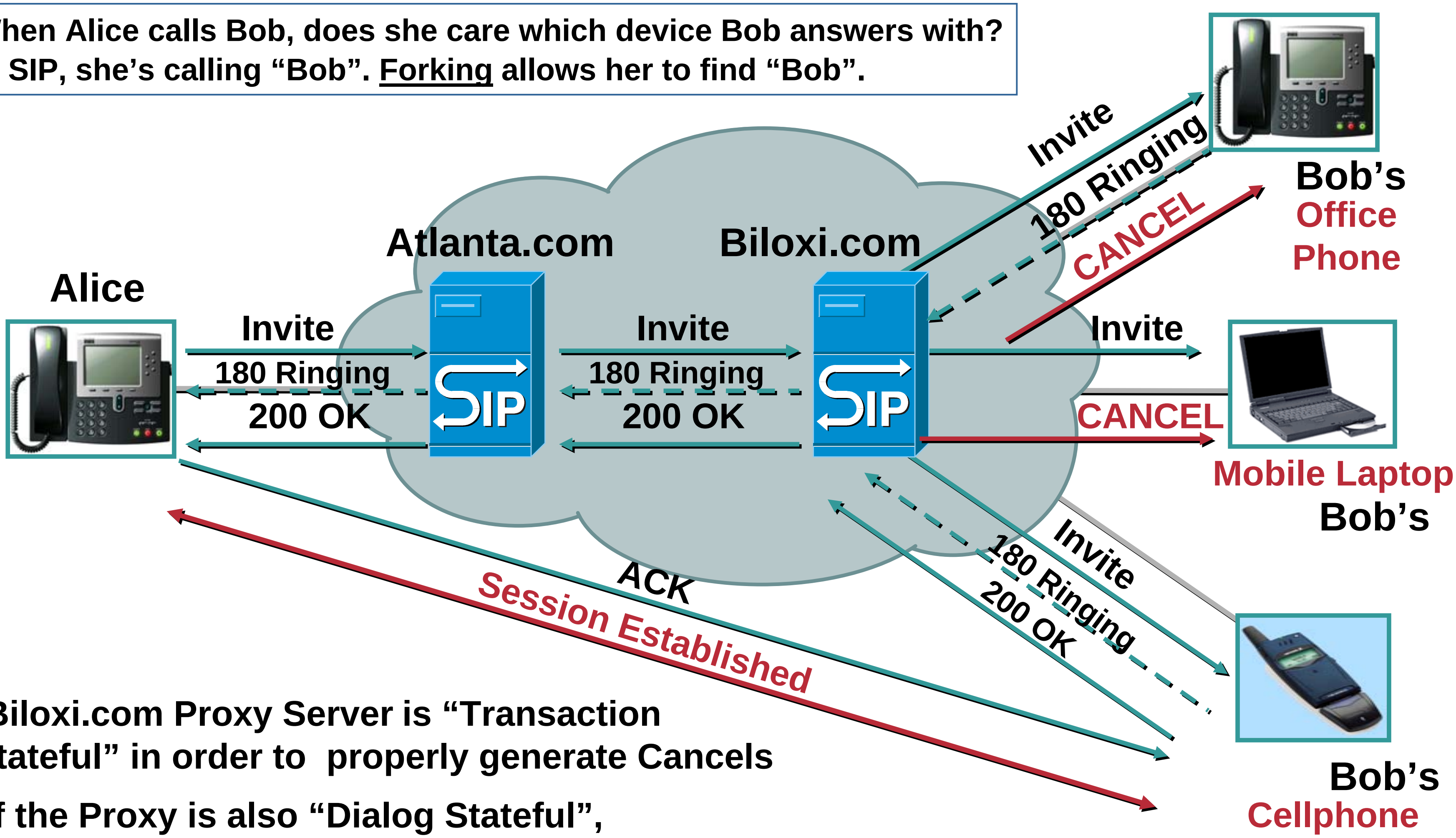
When Alice calls Bob, does she care which device Bob answers with?
In SIP, she's calling "Bob". Forking allows her to find "Bob".



- 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

When Alice calls Bob, does she care which device Bob answers with?
In SIP, she's calling "Bob". <u>Forking</u> allows her to find "Bob".

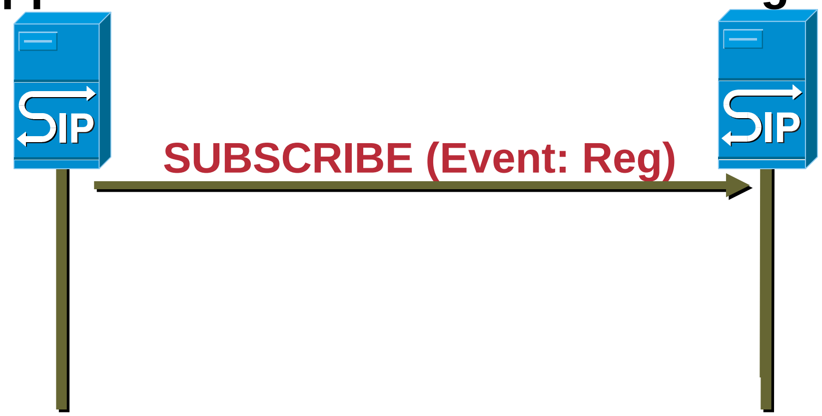


-Biloxi.com Proxy Server is "Transaction Stateful" in order to properly generate Cancels

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

RFC 3265&3680

SIP Methods: SUBSCRIBE & NOTIFY

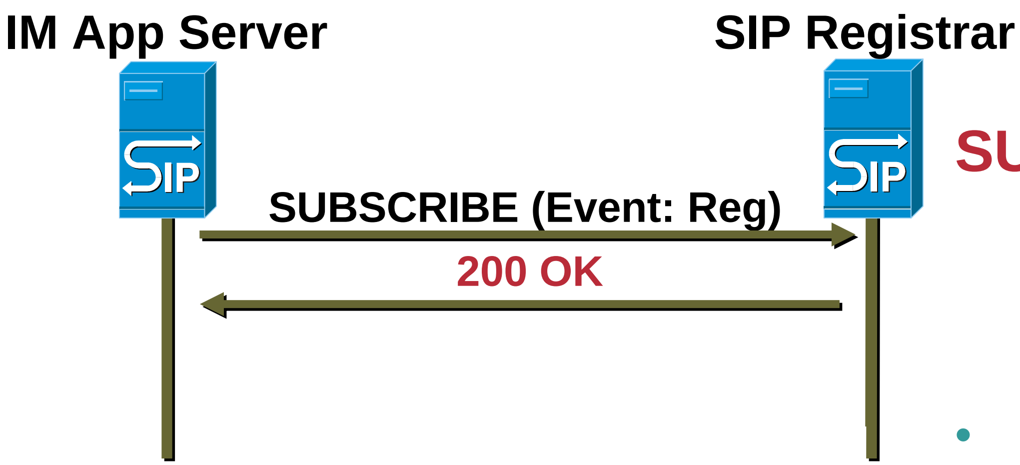


SUBSCRIBE sip:alice@atlanta.com SIP/2.0
Via: SIP/2.0/TCP app_IM.atlanta.com
;branch=z9hG4bKnashds7
From: sip:app_IM.atlanta.com ;tag=123aa9
To: sip:alice@atlanta.com
Call-ID: 9987@app_IM.atlanta.com
CSeq: 9887 SUBSCRIBE
Contact: sip:app_IM.atlanta.com
Event: reg
Max-Forwards: 70
Expires: 21600
Accept: application/reginfo+xml

SUBSCRIBE - used to request asynchronous notification of an event or set of events at a later time

- method used to request current state and state updates from a remote node
- Expires header SHOULD be present in Request
- Requests MUST have exactly one Event Header value

SIP Methods: SUBSCRIBE & NOTIFY

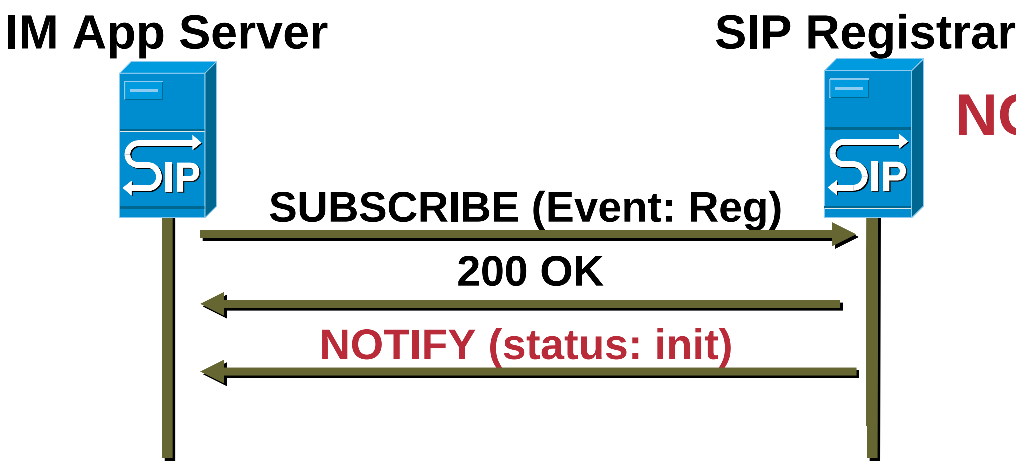


SIP/2.0 200 OK
Via: SIP/2.0/TCP app_IM.atlanta.com
 ;branch=z9hG4bKnashds7 ;received=10.1.3.2
From: sip:app_IM.atlanta.com ;tag=123aa9
To: sip:alice@atlanta.com ;tag=xyzygg
Call-ID: 9987@app_IM.atlanta.com
CSeq: 9987 SUBSCRIBE
Contact: sip:server19.atlanta.com
Expires: 3600

SUBSCRIBE - used to request asynchronous notification of an event or set of events at a later time

- Expires header MUST be present in Response
- 200-class responses indicate that the subscription has been accepted, and that a NOTIFY will be sent immediately

SIP Methods: SUBSCRIBE & NOTIFY



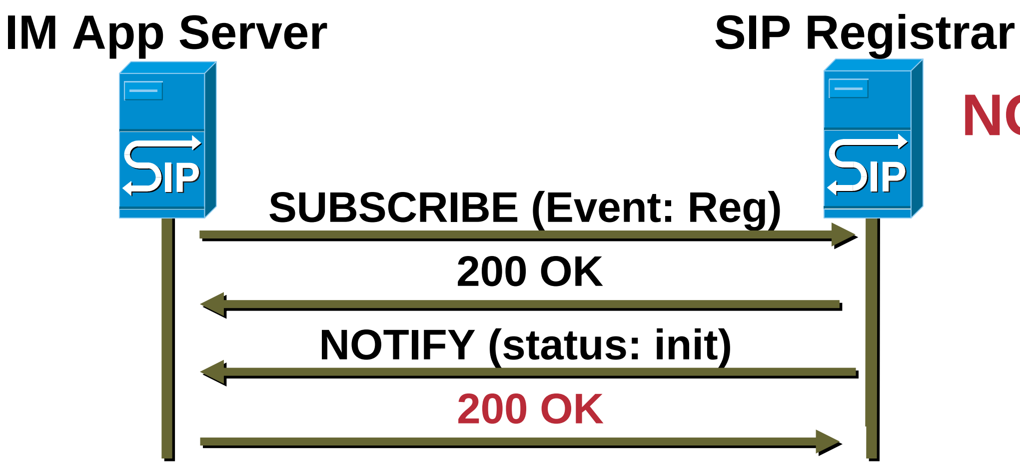

```
NOTIFY sip:app_IM.atlanta.com SIP/2.0
Via: SIP/2.0/TCP server1.atlanta.com
 ;branch=z9hG4bKnasaii
From: sip:alice@atlanta.com ;tag=xyzygg
To: sip:app_IM.atlanta.com ;tag=123aa9
Call-ID: 9987@app_IM.atlanta.com
CSeq: 1288 NOTIFY
Contact: sip:server19.atlanta.com
Event: reg
Max-Forwards: 70
Content-Type: application/reginfo+xml
Content-Length: 223
```

NOTIFY - used to notify a SIP node that an event which has been requested by an earlier SUBSCRIBE method has occurred

- NOTIFY is sent to inform subscribers of changes in state to which the subscriber has a subscription
- Event Header MUST match

```
<?xml version="1.0"?>
  <reginfo xmlns=
            "urn:ietf:params:xml:ns:reginfo"
             version="0" state="full">
    <registration aor="sip:alice@atlanta.com"
             id="a7" state="init" />
  </reginfo>
```

SIP Methods: SUBSCRIBE & NOTIFY



SIP/2.0 200 OK
Via: SIP/2.0/TCP server19.atlanta.com
;branch=z9hG4bKnasaii ;received=10.1.3.1
From: sip:app_IM.atlanta.com ;tag=123aa9
To: sip:alice@atlanta.com ;tag=xyzygg
Call-ID: 9987@app_IM.atlanta.com
CSeq: 1288 NOTIFY
Contact: sip:server1.atlanta.com
Content-Length: 0

NOTIFY - used to notify a SIP node that an event which has been requested by an earlier SUBSCRIBE method has occurred

- **sending a NOTIFY message to an unsuspecting node is invalid behavior, MUST receive a 481 "Subscription does not exist" response**

SIP Methods: MESSAGE

RFC 3428

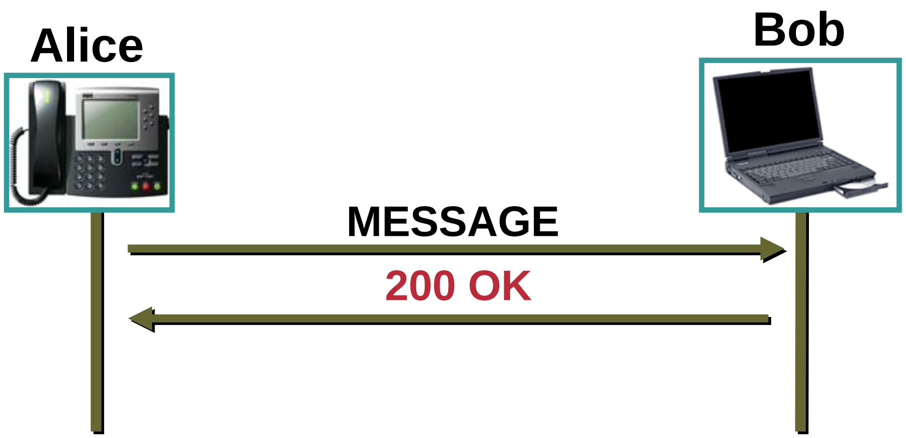

```
MESSAGE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/TCP pc33.atlanta.com
 ;branch=z9hG4bK776asegma
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 22756 MESSAGE
Content-Type: text/plain
Content-Length: 37

Isn’t this a great presentation ?
```

MESSAGE - the transfer of messages between users in near real-time

- Content (the payload) in MIME body parts
- MESSAGE does not initiate dialogs
- There is no explicit association between messages
- The body size MUST NOT exceed 1300 bytes

SIP Methods: MESSAGE



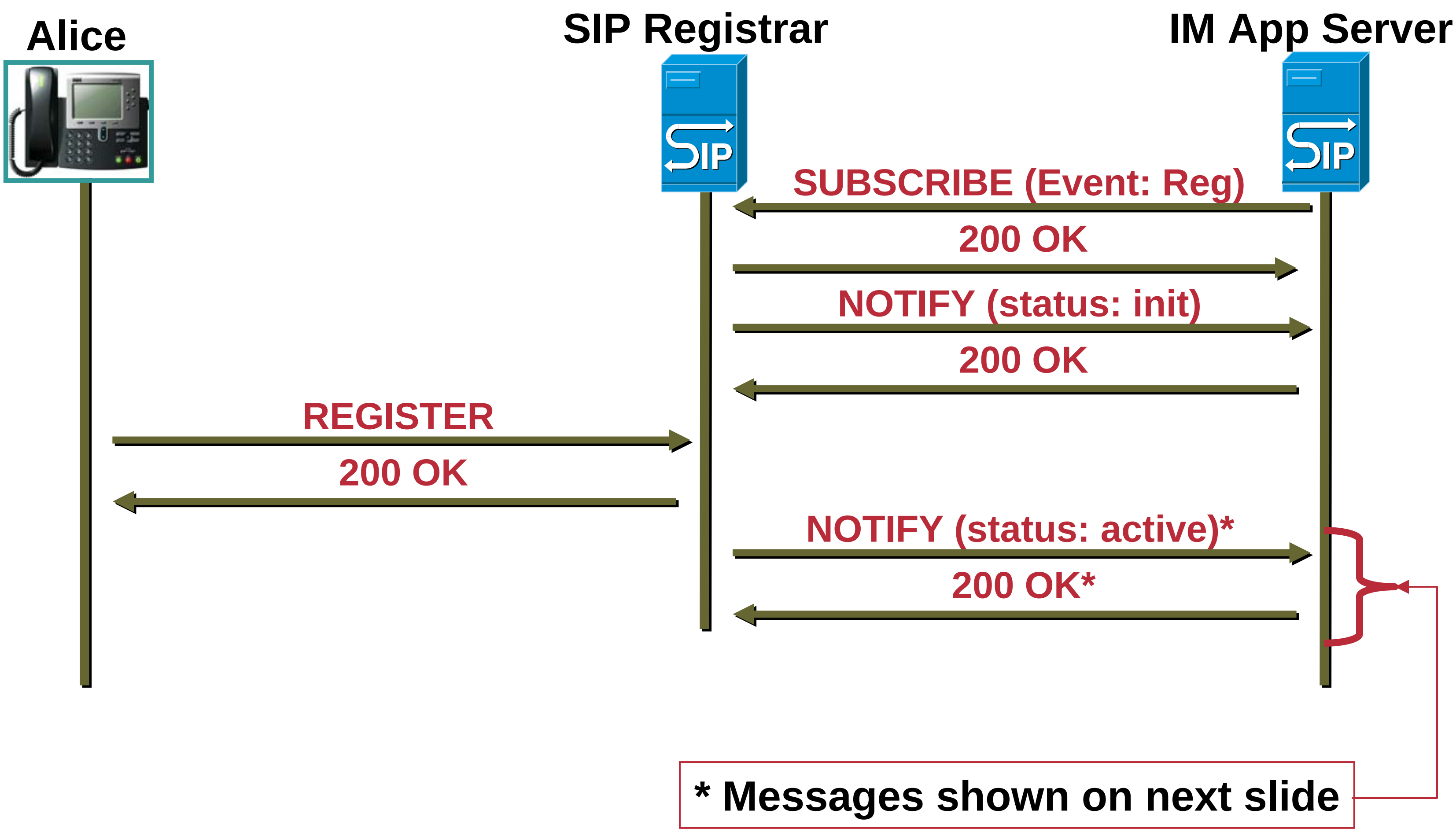
SIP/2.0 200 OK
Via: SIP/2.0/TCP pc33.atlanta.com
;branch=z9hG4bKnashds7 ;received=10.1.3.33
To: sip: sip:bob@biloxi.com>;tag=1928301774
From: alice@atlanta.com
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 22756 MESSAGE
Content-Length: 0

MESSAGE - the transfer of messages between users in near real-time

- **200 OK response does not necessarily mean the user has read the message**
- **A 4xx or 5xx response indicates that the message was not delivered successfully**
- **A 6xx response means it was delivered successfully, but refused**

RFC 3680

SIP Methods: Logging onto Service



SIP Methods: Logging onto Service (Cont.)

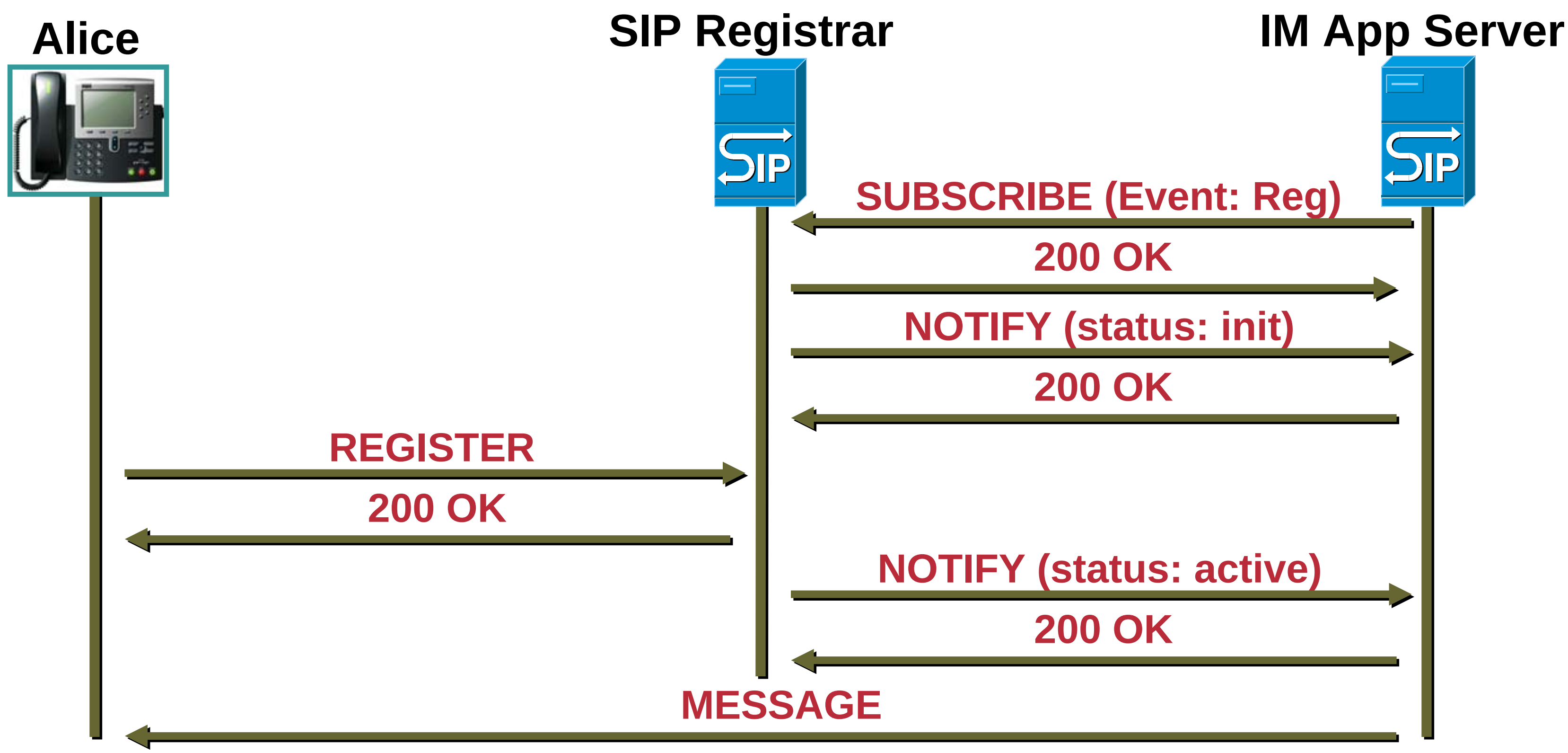
```
NOTIFY sip:app_IM.example.com SIP/2.0
Via: SIP/2.0/TCP server19.example.com
  ;branch=z9hG4bKnasaij
From: sip:alice@atlanta.com ;tag=xyzygg
To: sip:app_IM.atlanta.com ;tag=123aa9
Call-ID: 9987@app_IM.atlanta.com
CSeq: 1289 NOTIFY
Contact: sip:server19.atlanta.com
Event: reg
Max-Forwards: 70
Content-Type: application/reginfo+xml
Content-Length: ...
```

```
<?xml version="1.0"?>
 <reginfo xmlns="urn:ietf:params:xml:ns:reginfo"
         version="1" state="partial">
   <registration aor="sip:alice@atlanta.com"
         id="a7" state="active">
     <contact id="76" state="active" event="registered"
         duration-registered="0">
       <uri>sip:alice@pc33.atlanta.com</uri>
     </contact>
   </registration>
 </reginfo>
```

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP server19.atlanta.com
 ;branch=z9hG4bKnasaii ;received=10.1.3.1
From: sip:app_IM.atlanta.com ;tag=123aa9
To: sip:alice@atlanta.com ;tag=xyzygg
Call-ID: 9987@app_IM.atlanta.com
CSeq: 1289 NOTIFY
Contact: sip:server1.atlanta.com
Content-Length: 0
```

SIP Methods: Logging onto Service

RFC 3680

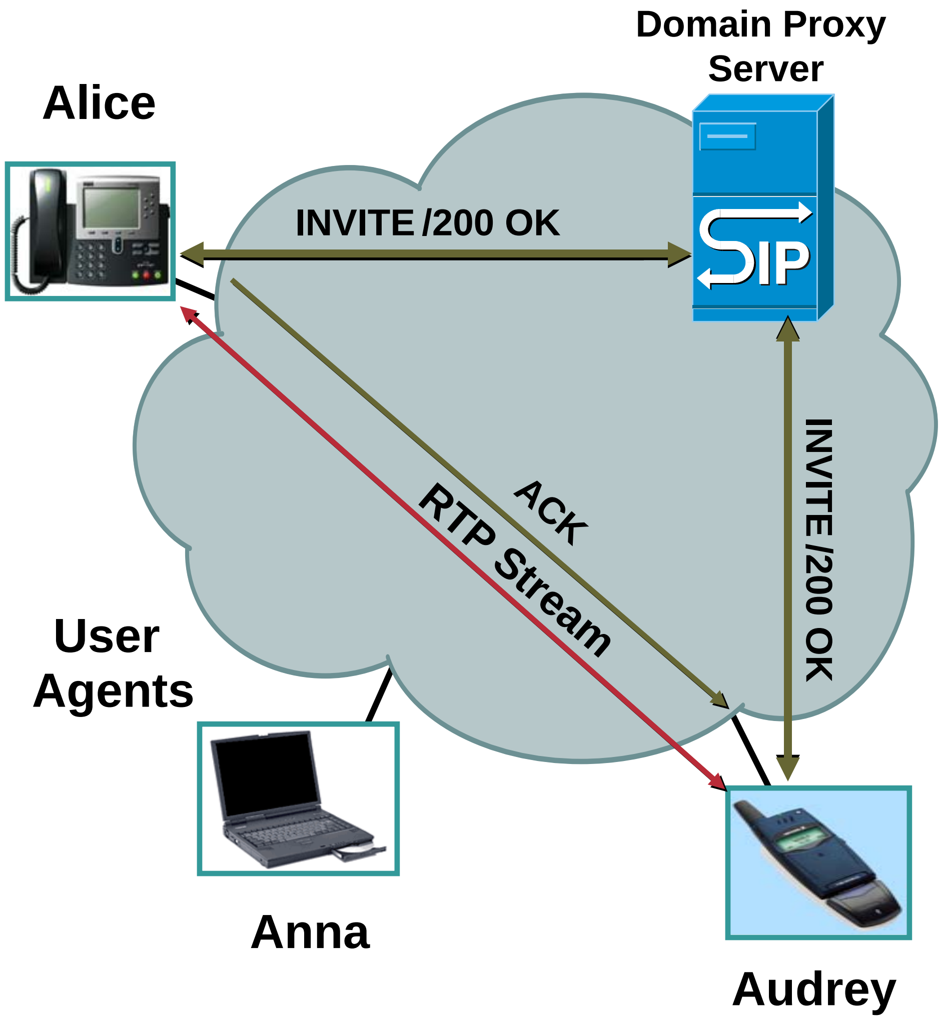


Welcome to the Atlanta Instant Messaging service!

Agenda

- SIP History & Basics
- Elements of SIP
- SIP Message details
- SIP Signaling
- Interworking with MGCP/H.323

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

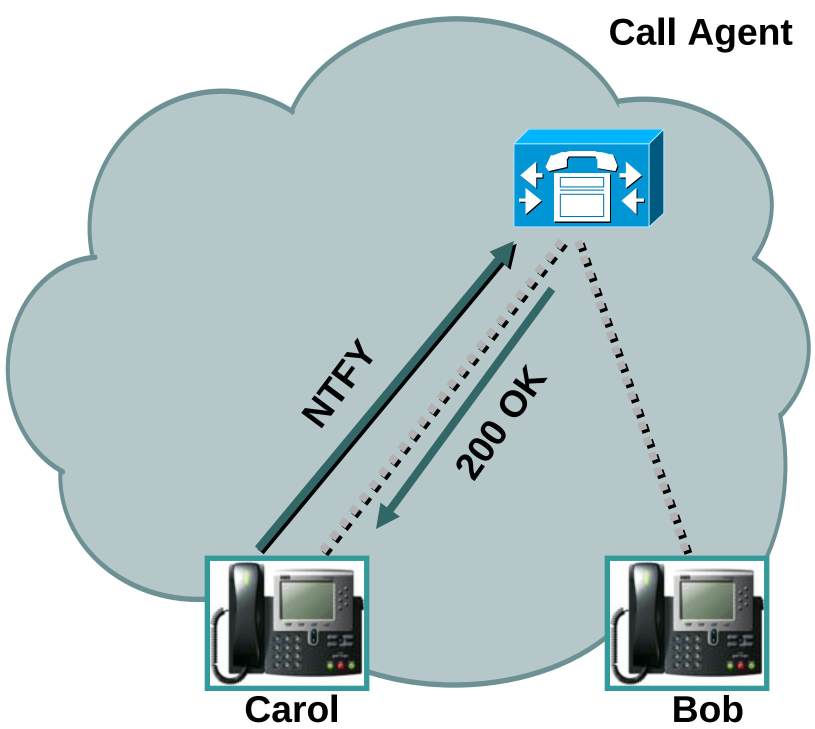


SIP Only Domain

Alice Calls Audrey

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

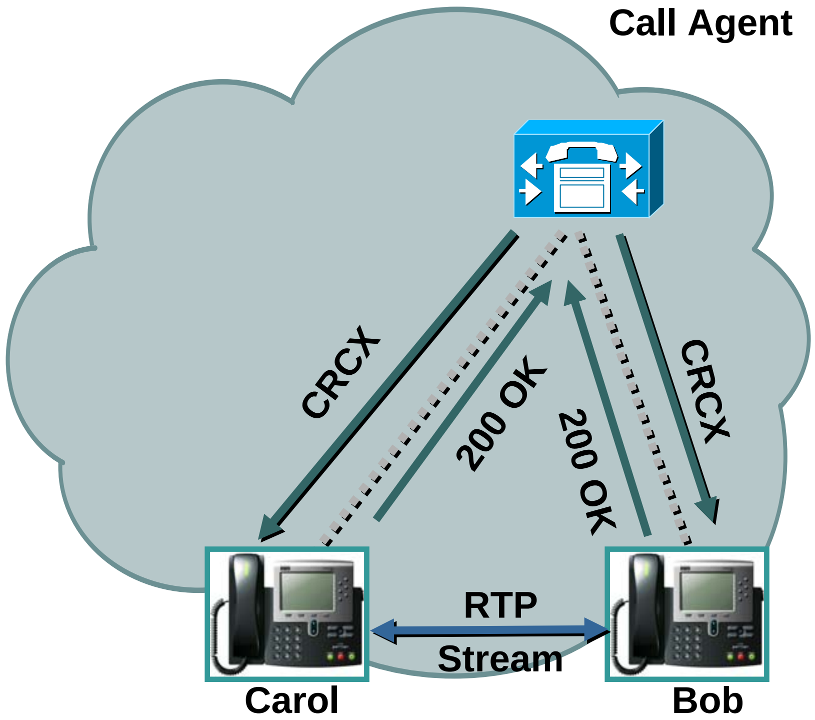
MGCP Only Domain



Carol Calls Bob

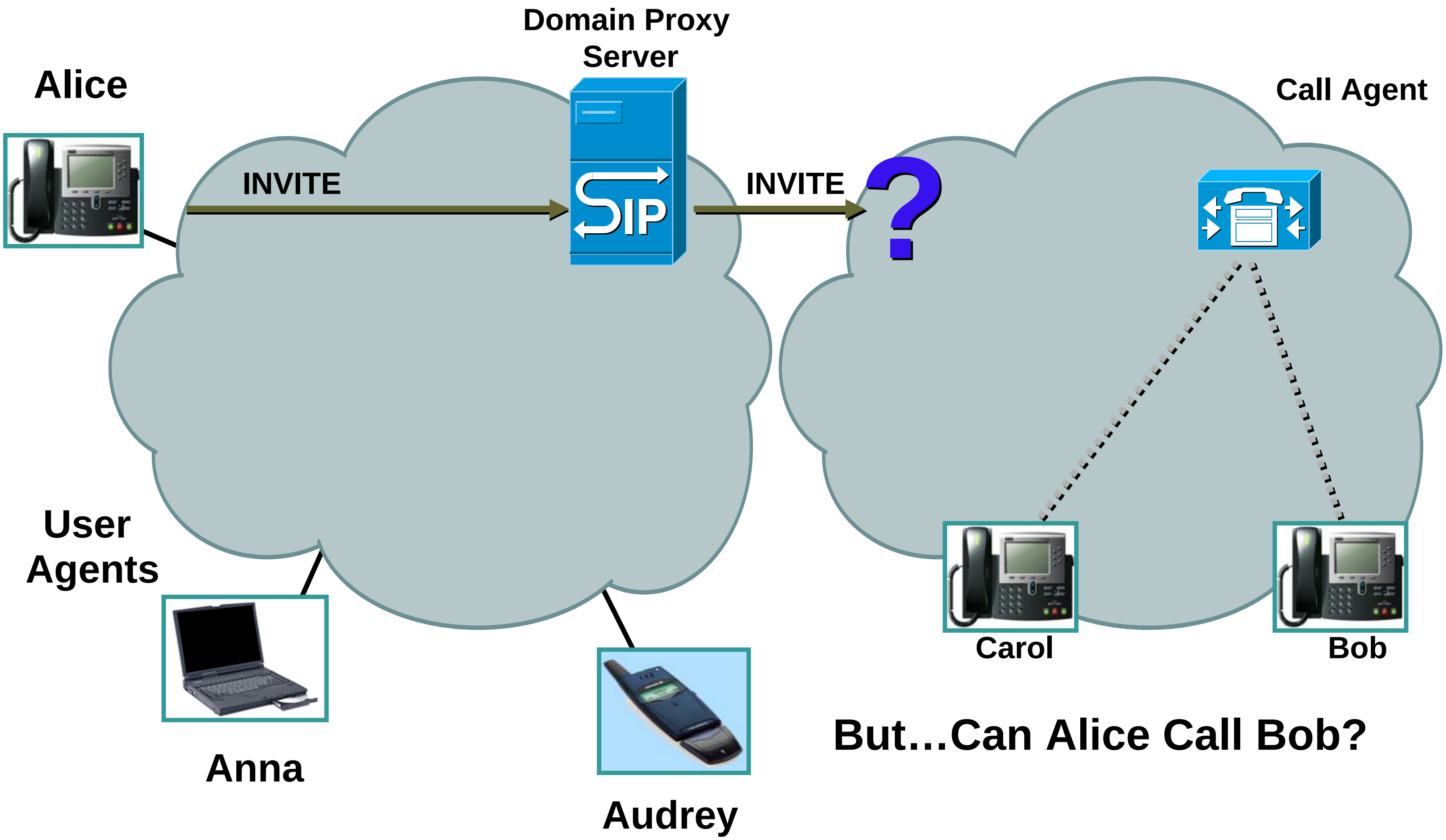
Peer-to-Peer to/from Client/Server Interworking

MGCP Only Domain



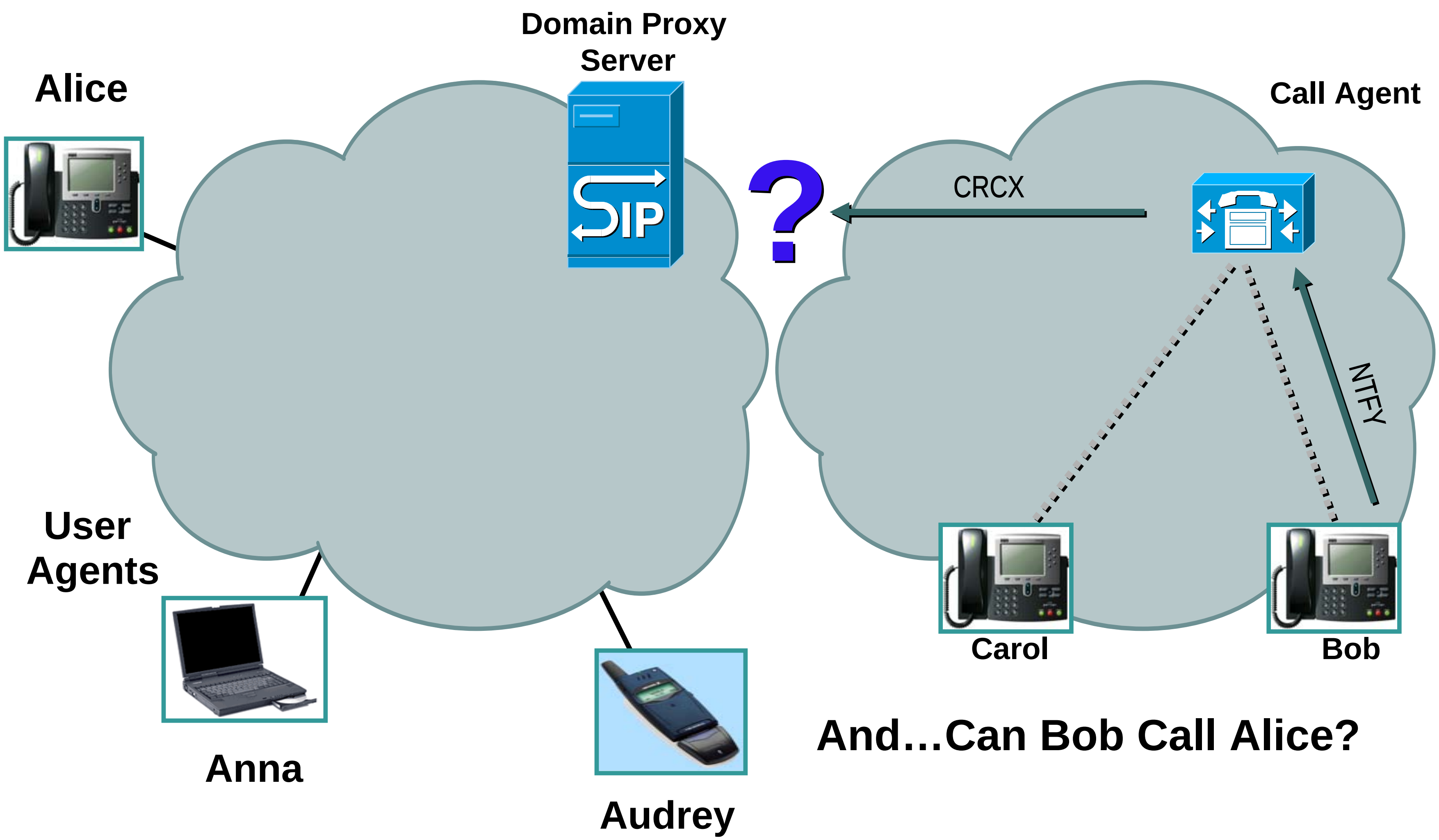
Carol Calls Bob

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



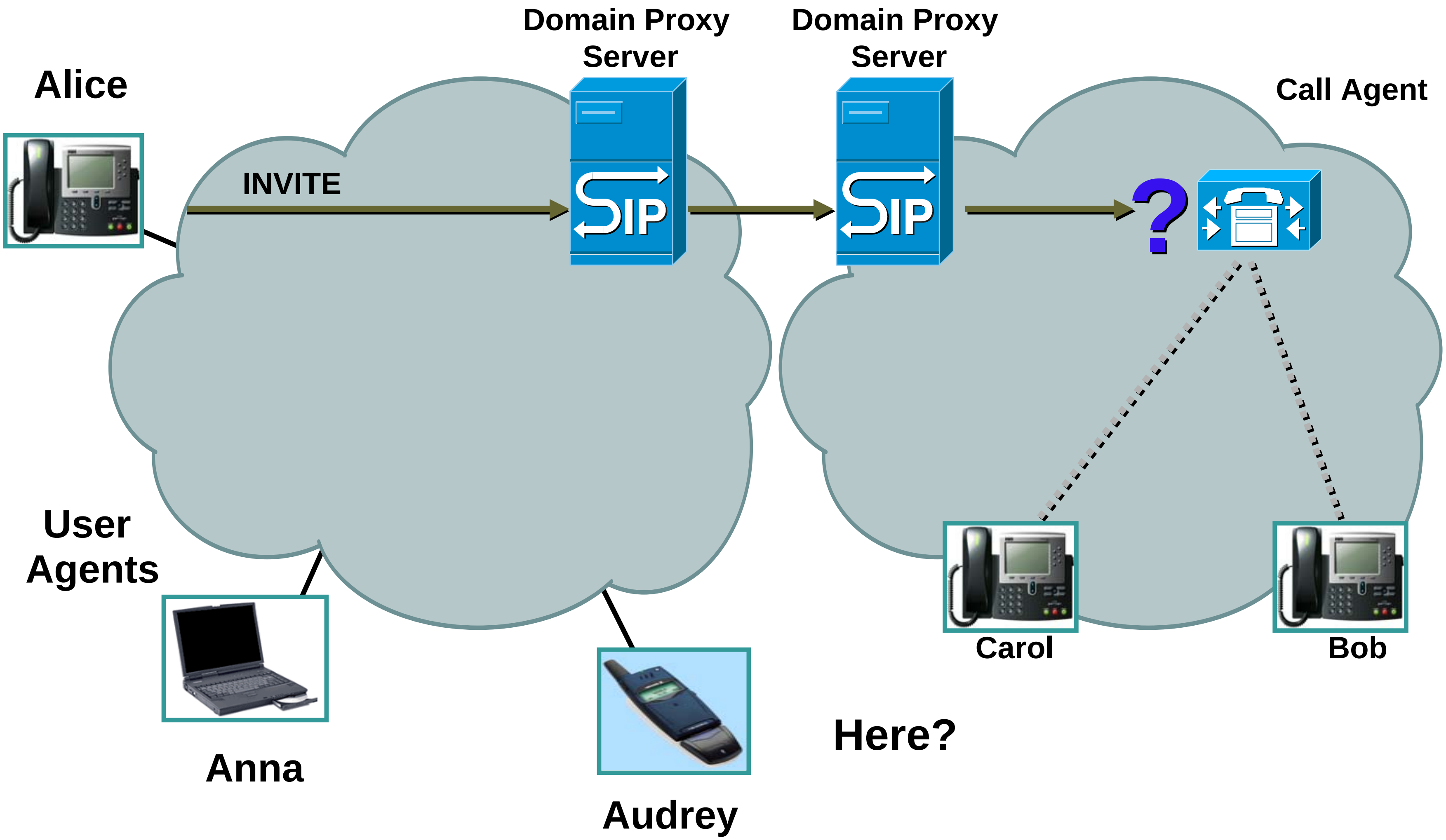
But…Can Alice Call Bob?

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

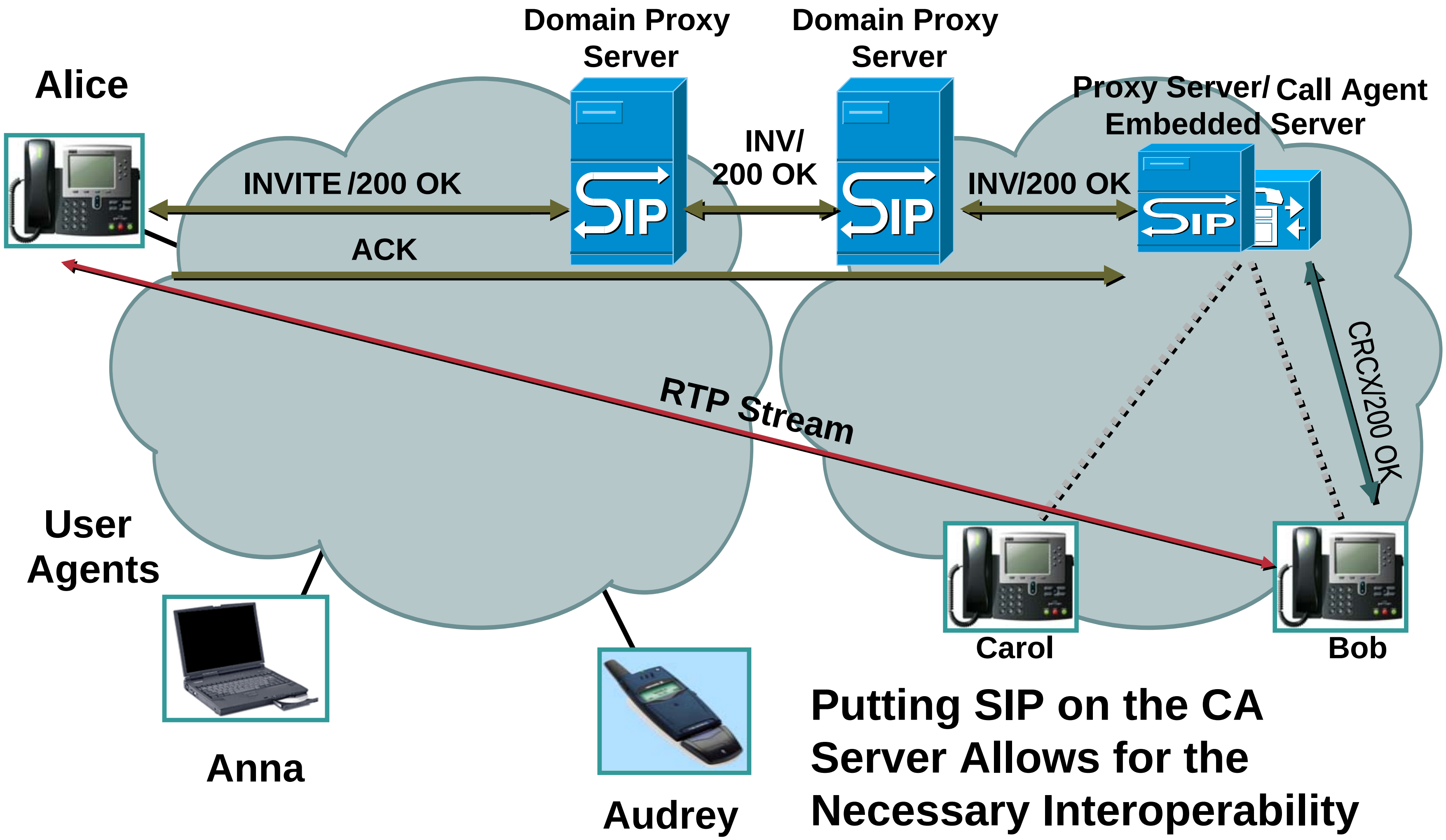


And…Can Bob Call Alice?

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

In Conclusion…

In Conclusion…

- SIP is a session set-up protocol, providing primitives to invoke services
- SIP is Layer 2, 3 & 4 independent
- SIP utilizes many existing IETF protocols and syntax
- Signals Instant Messaging, and transports another protocol (SDP) to establish Voice & Video
- Servers can have varying degrees of “awareness”
  - Amount of control based on configuration of the server
- The integration of call control and web services creates applications that reduce cost and increase productivity

SIP References

- http://www.cisco.com—Search for SIP, Cisco proxy server and CallManager 5.0
- 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.sipcenter.com/—The SIP center
- http://www.ietf.org/html.charters/ecrit-charter.html/—IETF ECRIT WG for emergency calling