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Escola de Engenharia
Departamento de Informática

Multimedia Networking

Signaling Protocols

SIP – Session Initiation Protocol

Main Sources:

1. *Computer Networking: A Top Down Approach*, 7th Ed.
Jim Kurose, Keith Ross, Pearson, 2017.
2. *Computer Networks w/ Internet Protocols & Tech.*,
William Stallings, 2004
3. *RFC 3261, RFC 5411*

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SIP – Motivation



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Need to:

- ❑ create and manage sessions within applications
- ❑ handle heterogeneous users' behavior and practices (mobility, different devices and IDs/names)

SIP long-term vision:

- ❑ all telephone calls, video conference calls take place over the Internet
- ❑ people are identified by names or e-mail addresses, rather than by phone numbers
- ❑ you can reach callee, no matter where callee roams, no matter what IP device callee is currently using.

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SIP – overview



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Session Initiation Protocol

“An **application-layer control** (signaling) **protocol** for creating, modifying, and terminating **sessions** with one or more participants.”

“These sessions include Internet telephone calls, multimedia distribution, and multimedia conferences”, i.e., involving exchange of data among an association of participants.

[RFC3261,2002]

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SIP – overview



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"SIP works with existing real-time multimedia protocols by enabling Internet endpoints (called user agents) to **discover** one another and to **agree** on a characterization of a session they would like to share"

"For locating prospective session participants, and for other functions, **SIP enables the creation of an infrastructure of network hosts** (called proxy servers) to which user agents can send registrations, invitations to sessions, and other requests."

[RFC3261,2002]

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SIP - overview



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- **SIP Functionality**

- **User location**: find current user location; users can access application features from remote locations
- **User availability**: determine the willingness of called party to communicate
- **User capabilities**: determine media and parameters to be used
- **Session setup**: establish session parameters for point-to-point and multiparty calls (e.g., ringing)
- **Session management**: transfer and termination of a session, modifying session parameters, and invoking services

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- ❑ Setting up a call, SIP provides mechanisms...
 - for caller to let callee know someone wants to establish a call
 - so caller and callee can agree on media type, encoding
 - to end a call
- ❑ determine current IP address of callee:
 - maps mnemonic identifier to current IP address
- ❑ call management:
 - add new media streams during call
 - change encoding during call
 - invite others
 - transfer, hold calls

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SIP - overview



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- SIP Uniform Resource Identifier (URI) is similar to an email address
 - sip|sips:user:password@host:port;uri-parameters?headers
 - sip: bob@domain.com
- Based on HTTP-like request/response transaction model
- Uses most HTTP header fields, encoding rules, and status codes
 - readable format for displaying information
- Uses concept of recursive and iterative DNS searches
- Incorporates Session Description Protocol (SDP)
 - defines session contents using types similar to MIME (mail extensions)

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SIP - overview



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- SIP provides service primitives, **not** services!
- SIP is rather an **independent component** that can be used with other IETF protocols (RTP, RTSP, SDP, etc.) to build a complete multimedia architecture.
 - can run over TCP, UDP, DCCP, SCTP, RTP/RTCP, etc.
- SIP comprises four types of logical entities: **user agent, redirect server, proxy server, registrar**

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SIP - Components and Protocols



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- **Client**
 - sends requests and receives responses
 - user agent clients and proxies are clients
- **Server**
 - receives requests and sends back responses
 - proxies, user agent servers, redirect servers, and registrars
- **User Agent**
 - in every SIP end-station/terminal
 - User agent client (UAC): Issues requests
 - User agent server (UAS): Receives requests and responds
- **Redirect Server**
 - redirecting the client to contact an alternate set of URIs
 - like iterative searches in DNS

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SIP - Components and Protocols



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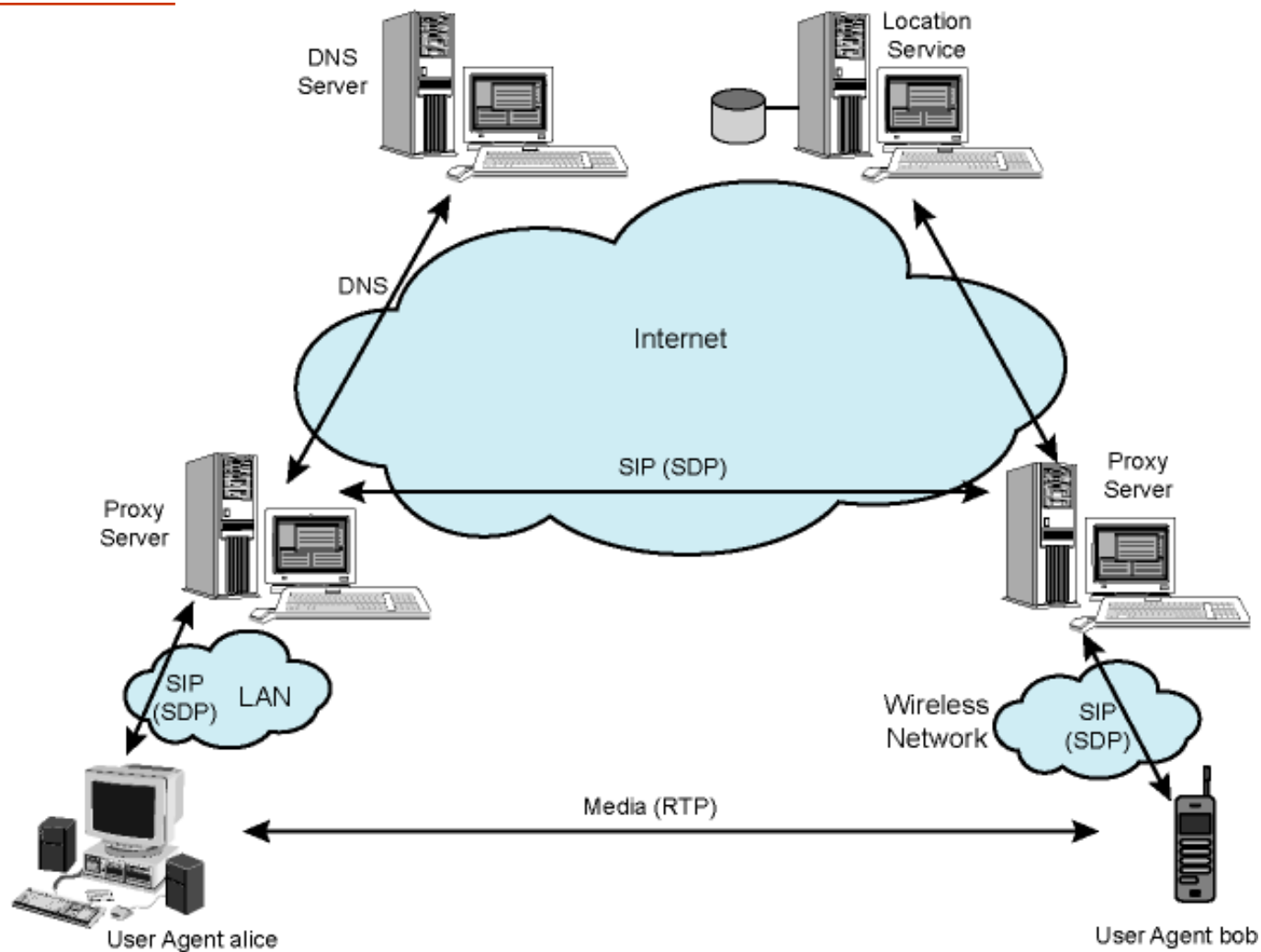
- **Proxy Server**
 - server and client
 - makes requests for other clients
 - routing and enforcing policy on calls
 - like recursive searches in DNS
- **Registrar**
 - server that accepts REGISTER requests
 - places information it receives in requests in the location service for the domain
 - SIP address, associated IP address of device, status...
- **Location Service**
 - used by redirect or proxy servers to obtain information about a callee's possible location(s)
 - maintains a database of SIP-address/IP-address mappings

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SIP - Components and Protocols



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SIP Messages - Requests



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Examples of Methods

- **REGISTER**: notifies SIP network of IP address(es) and URIs for which it would like to receive calls
- **INVITE**: establishes session between user agents
- **ACK**: confirms reliable message exchanges
- **CANCEL**: terminates pending request, but does not undo completed call
- **BYE**: terminates session between users in conference
- **OPTIONS**: solicits information about callee capabilities
- ...

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SIP Messages – Response



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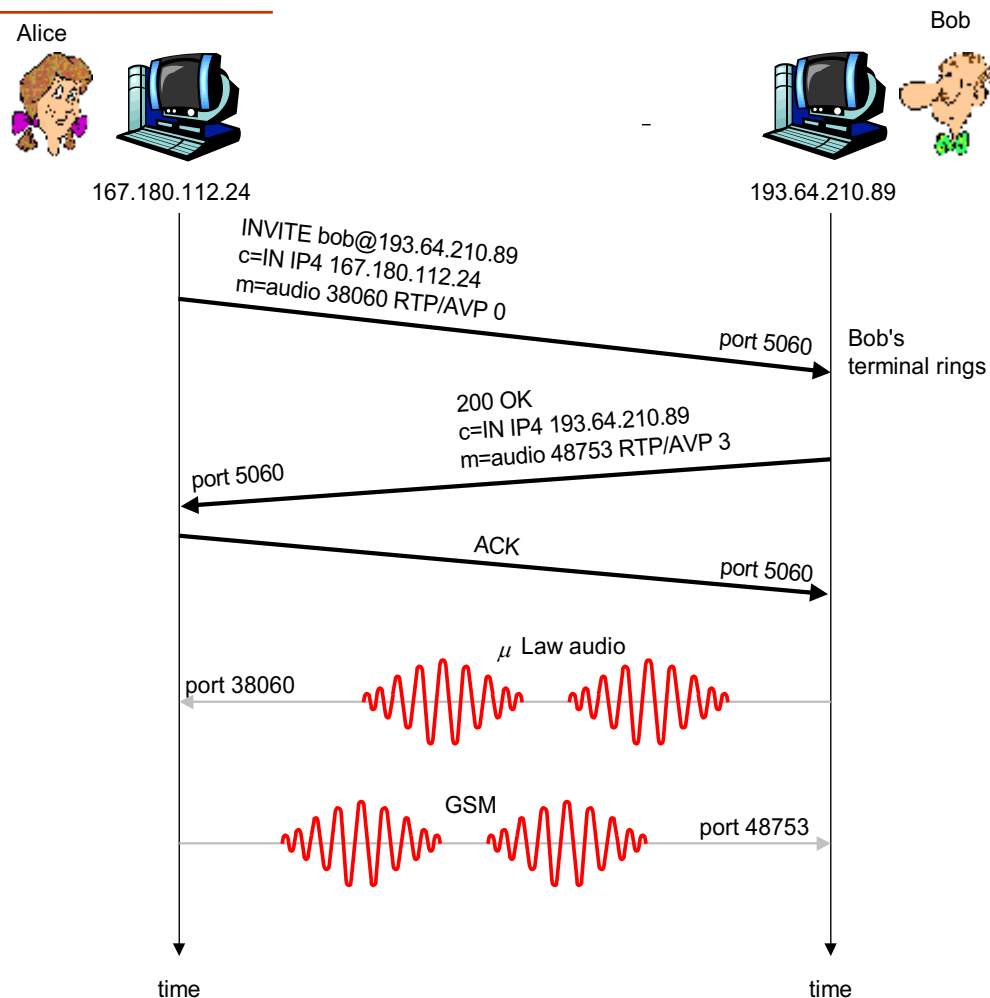
- **Provisional** (1xx): Request received and being processed
- **Success** (2xx): Action successfully received, understood, and accepted
- **Redirection** (3xx): Further action needed
- **Client Error** (4xx): Request contains bad syntax or cannot be fulfilled at this server
- **Server Error** (5xx): Server failed to fulfill apparently valid request
- **Global Failure** (6xx): Request cannot be fulfilled at any server

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SIP – setting up a call to a known IP



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□ Alice's SIP INVITE message indicates her port number, IP address, encoding she prefers to receive (PCM ulaw)

□ Bob's 200 OK message indicates his port number, IP address, preferred encoding (GSM)

□ SIP messages can be sent over TCP, UDP or other transport protocol; here sent over RTP/UDP.

□ default SIP port number is 5060.

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SIP – setting up a call



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❑ codec negotiation:

- suppose Bob doesn't have PCM ulaw encoder.
- Bob will instead reply with 606 Not Acceptable Reply, listing his encoders Alice can then send new INVITE message, advertising different encoder

❑ rejecting a call

- Bob can reject with replies "busy," "gone," "payment required," "forbidden"

❑ media can be sent over RTP or some other protocol

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SIP - request message example



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```
INVITE sip:bob@domain.com SIP/2.0
Via: SIP/2.0/UDP 167.180.112.24
From: sip:alice@hereway.com
To: sip:bob@domain.com
Call-ID: a2e3a@pigeon.hereway.com
Content-Type: application/sdp
Content-Length: 885
```

```
c=IN IP4 167.180.112.24
m=audio 38060 RTP/AVP 0
```

Notes:

- HTTP message syntax
- sdp = session description protocol
- Call-ID is unique for every call.

□ Here we don't know Bob's IP address. Intermediate SIP servers needed.

□ Alice sends, receives SIP messages using SIP default port 5060

□ Alice specifies in Via: header that SIP client sends, receives SIP messages over UDP



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SIP – name translation and user location

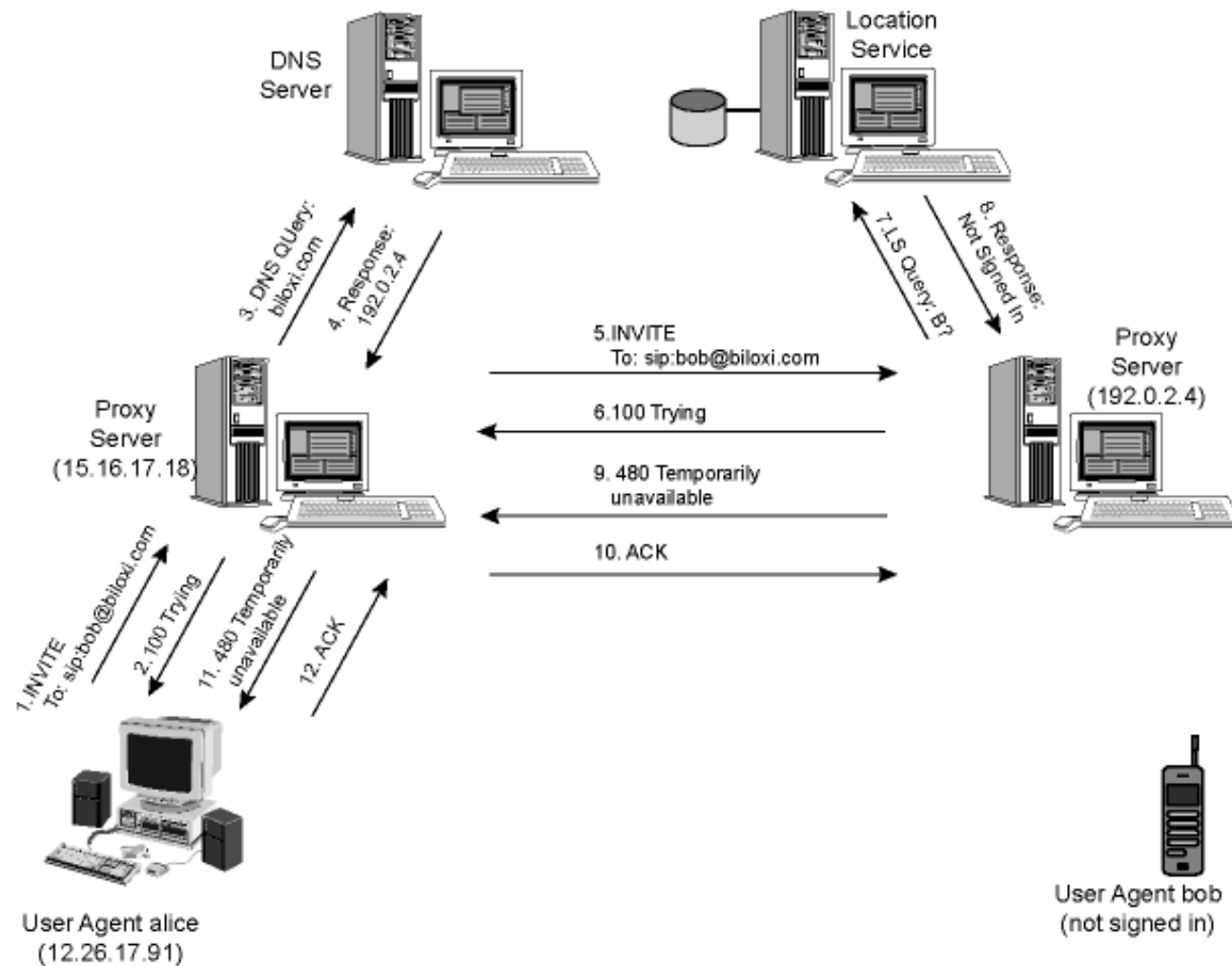
- ❑ caller wants to call callee, but only has callee's name or e-mail address.
 - ❑ need to get IP address of callee's current host:
 - user moves around
 - DHCP protocol
 - user has different IP devices (PC, mobile, tablet, car device)
 - ❑ result can be based on:
 - time of day (work, home)
 - caller (don't want boss to call you at home)
 - status of callee (calls sent to voicemail when callee is already talking to someone)
- Service provided by SIP servers:
- ❑ SIP registrar server
 - ❑ SIP proxy server

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SIP – Call Setup Attempt Scenario



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SIP registrar



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- when Bob starts SIP client, client sends SIP REGISTER message to Bob's registrar server
(similar function needed and used by Instant Messaging)

Register Message:

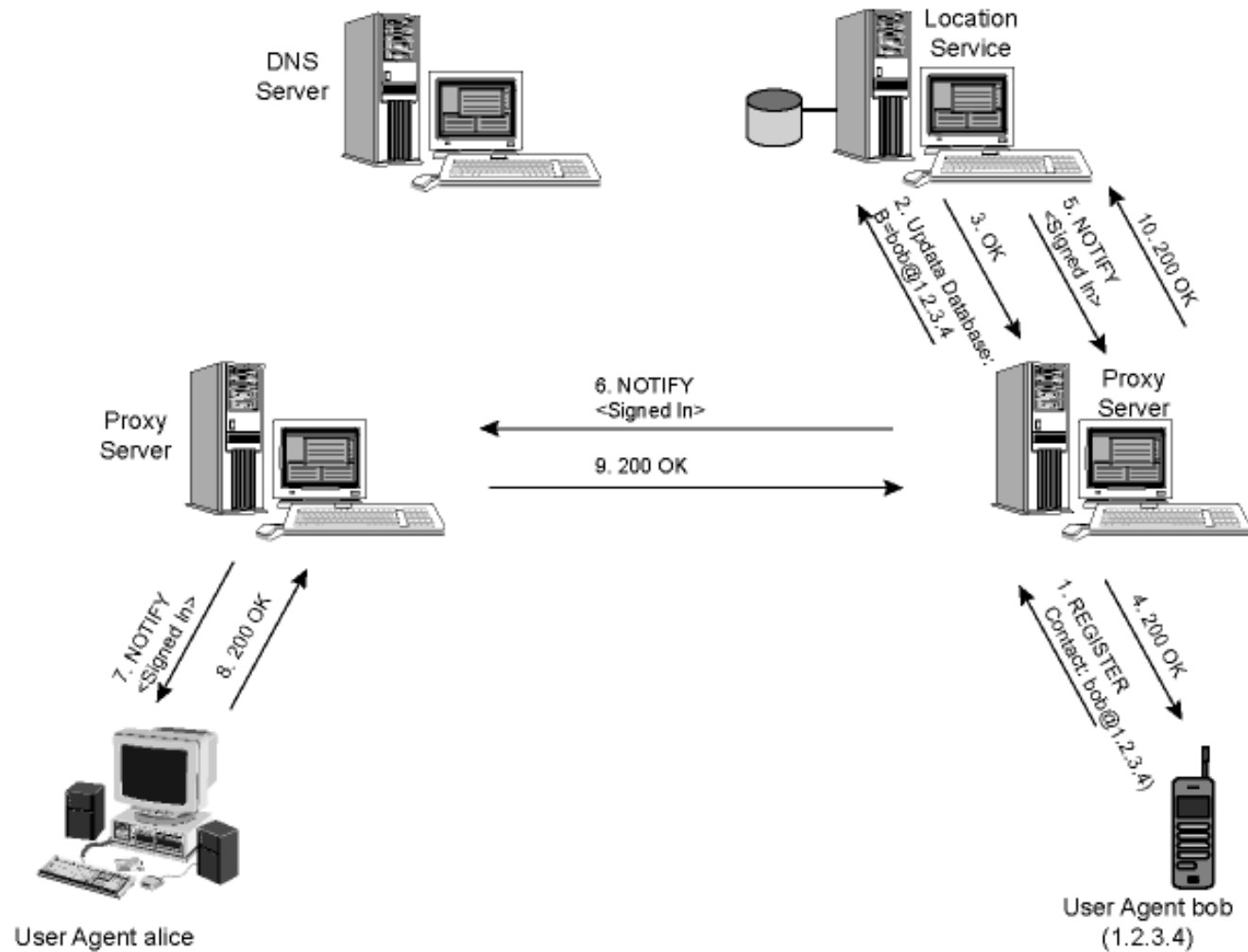
```
REGISTER sip:domain.com SIP/2.0  
Via: SIP/2.0/UDP 193.64.210.89  
From: sip:bob@domain.com  
To: sip:bob@domain.com  
Expires: 3600
```

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SIP – Notification Example



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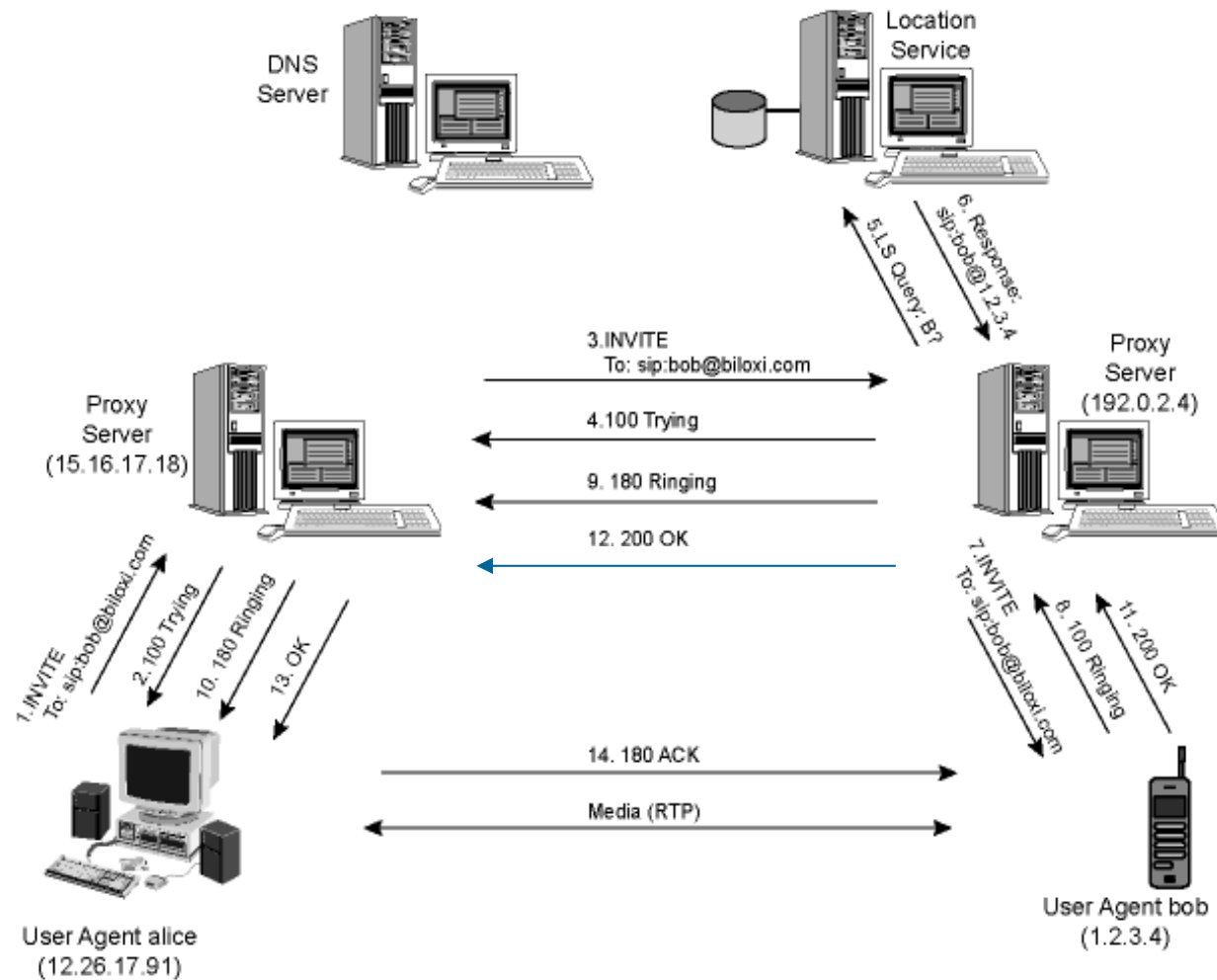


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SIP – Successful Call Setup



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SIP - Comparison with H.323



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- ❑ H.323 is another signaling protocol for real-time, interactive services
- ❑ H.323 is a complete, vertically integrated suite of protocols for multimedia conferencing: signaling, registration, admission control, transport, codecs
- ❑ SIP is a single component. Works with RTP, but does not mandate it. Can be combined with other protocols and services
- ❑ H.323 comes from the ITU (telephony).
- ❑ SIP comes from IETF: Borrows much of its concepts from HTTP
 - SIP has Web flavor, whereas H.323 has telephony flavor.
- ❑ SIP is globally simpler than H.323.
- ❑ It has been widely adopted (also within 3GPP, 3GPP2, 4G)

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SIP references



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- ❑ RFC 3261 – SIP standard
- ❑ RFC 5411 (2009) – A Hitchhiker's Guide to SIP
- ❑ lots of material, available implementations, etc, etc...
 - ❑ <http://www.cs.columbia.edu/sip/>

also...

- ❑ IETF Applications and Real-Time Area
 - ❑ <https://datatracker.ietf.org/wg/#art>