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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 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 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

- 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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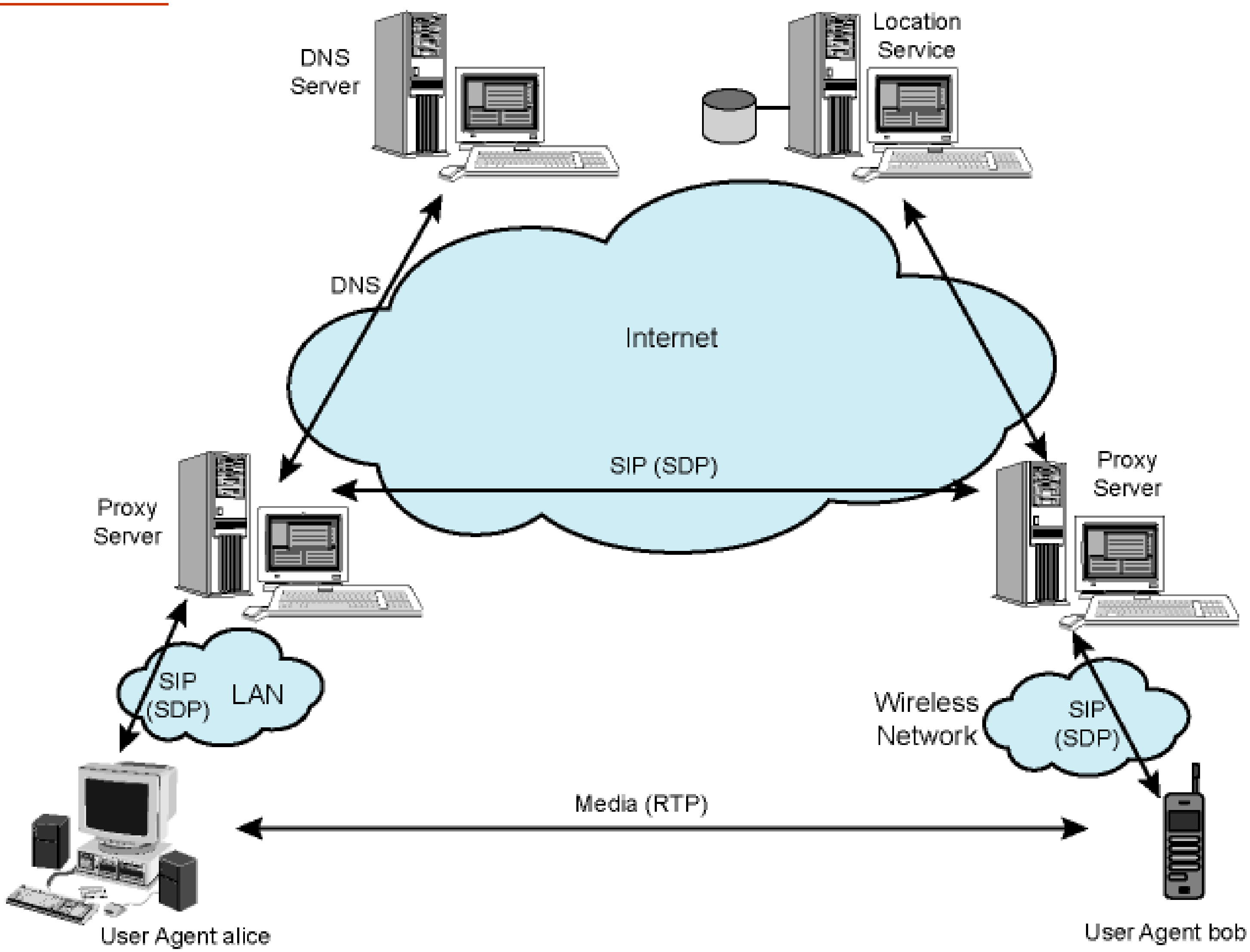

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SIP Messages - Requests

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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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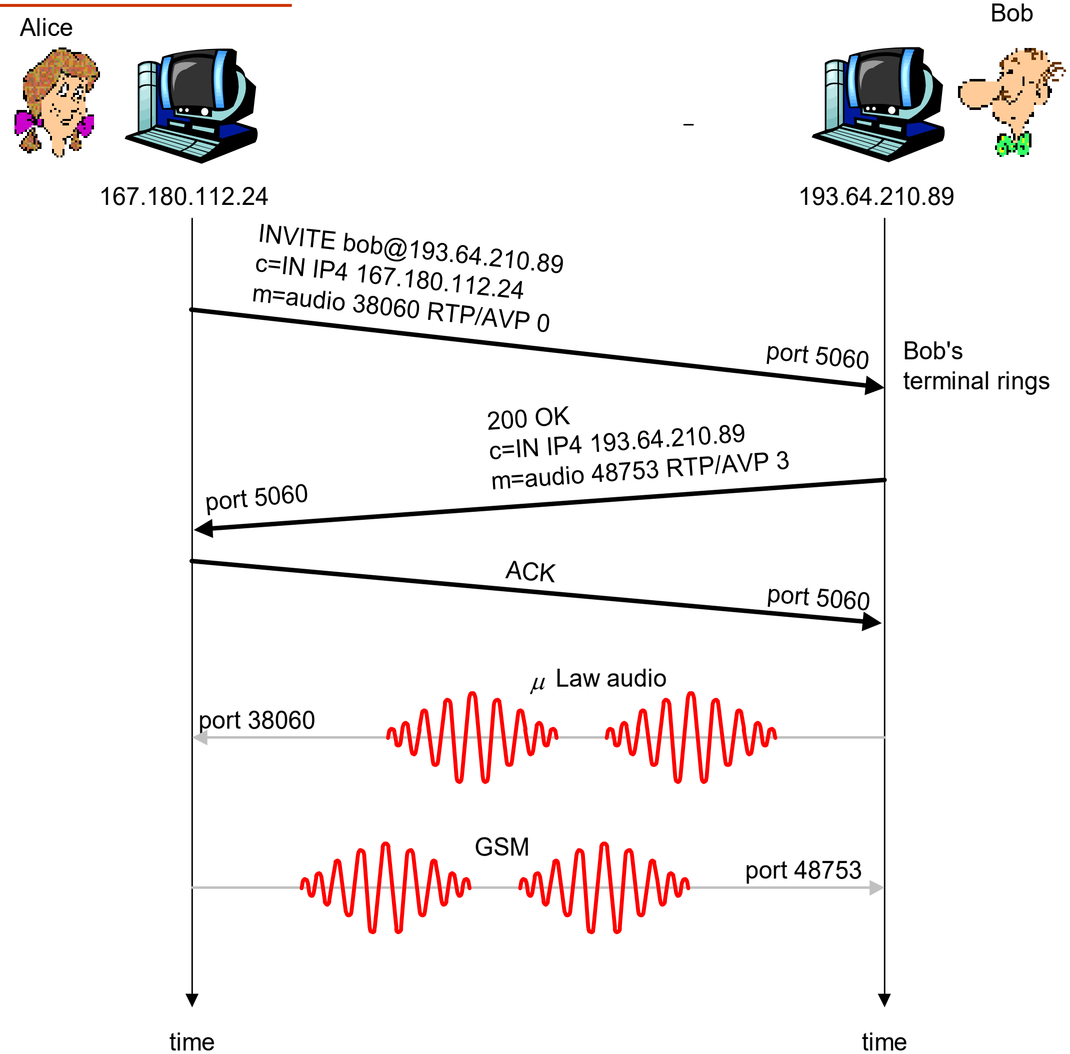

- Alice's SIP INVITE message indicates her port number, IP address, encoding she prefers to receive (PCM ulaw)
- Bob's 200 OK message indicates his port number, IP address, preferred encoding (GSM)
- SIP messages can be sent over TCP, UDP or other transport protocol; here sent over RTP/UDP.
- default 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

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- 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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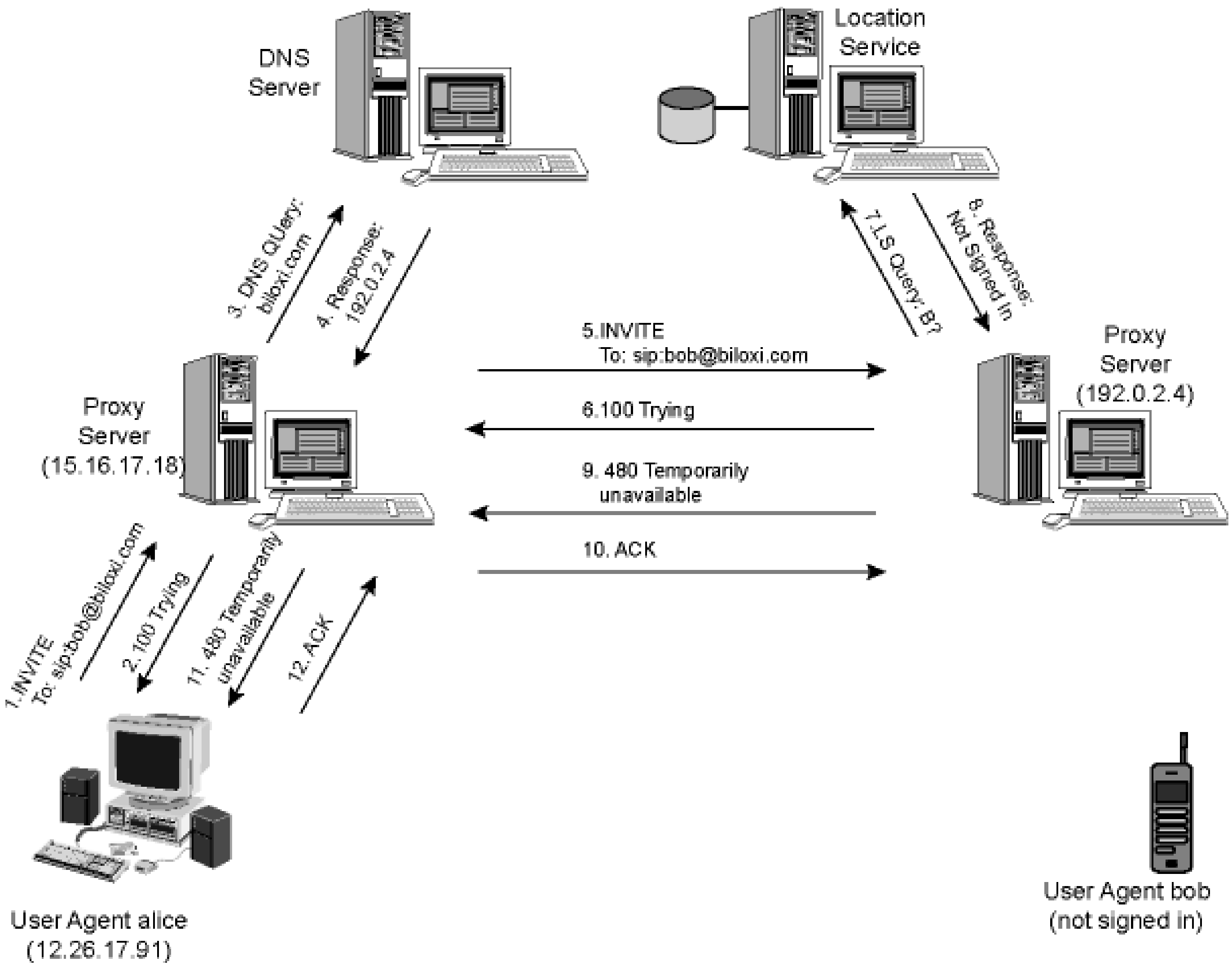

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SIP registrar

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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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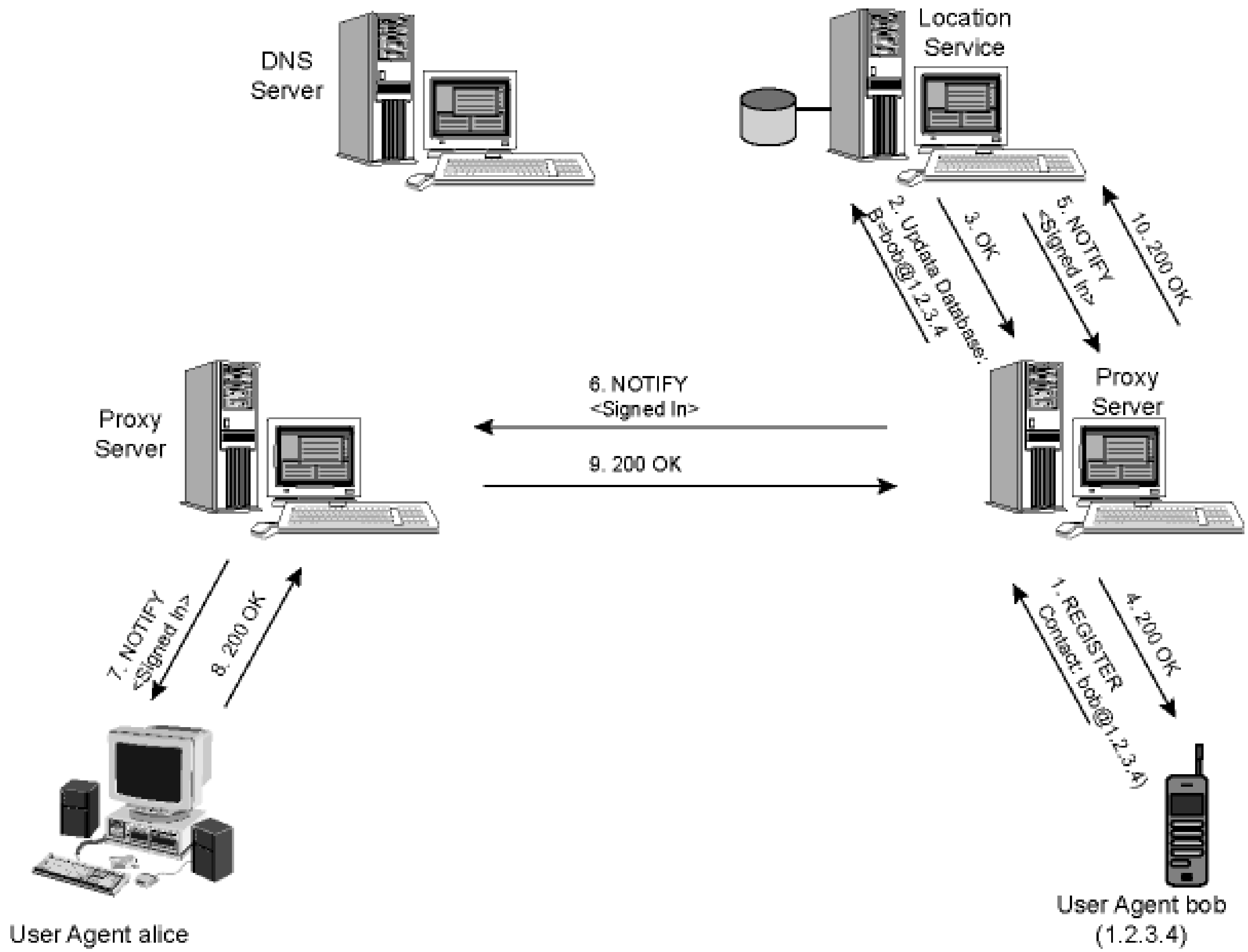

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SIP – Successful Call Setup

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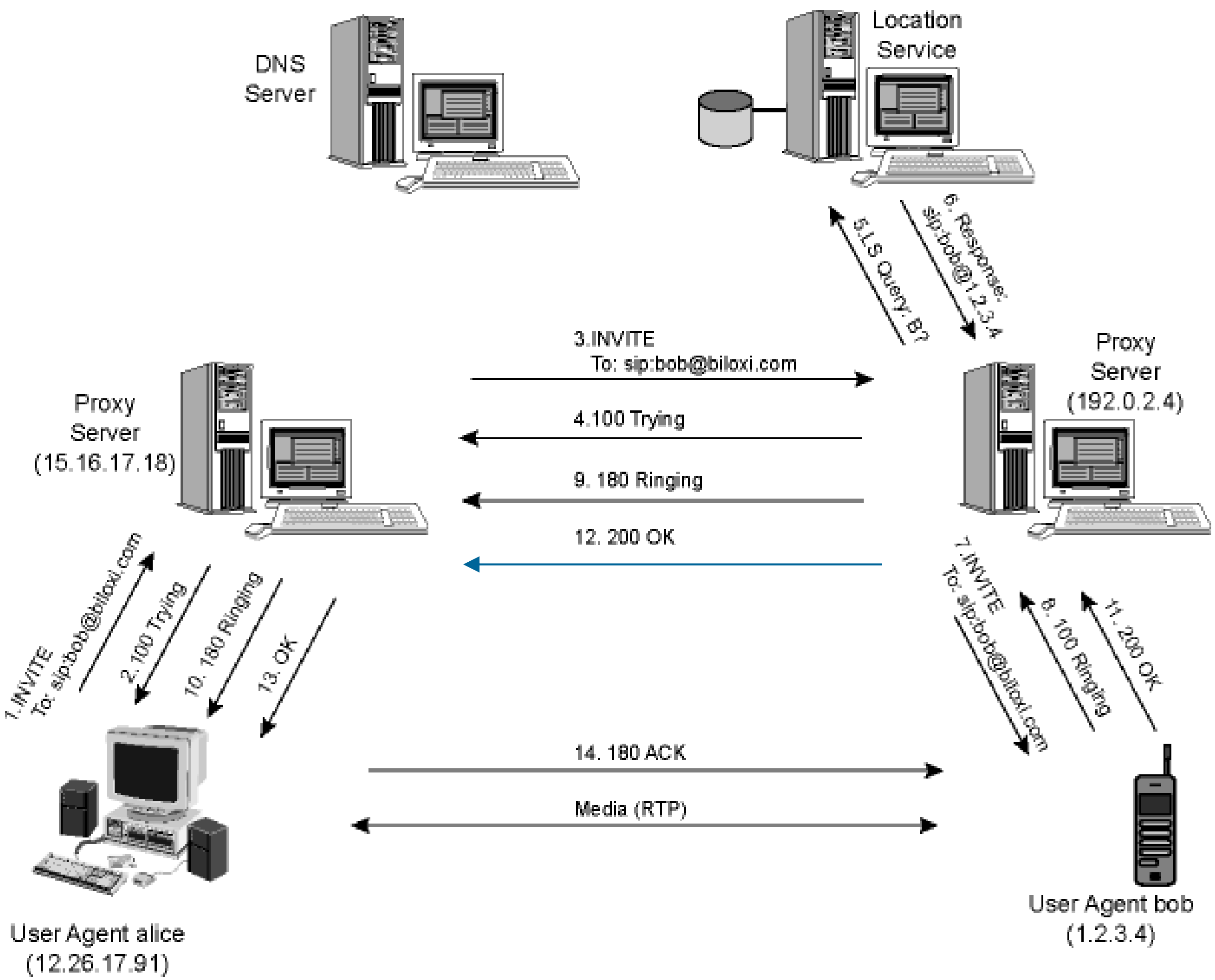

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SIP - Comparison with H.323

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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