



SMU School of Engineering

SMU Course #: EETS 8393

NTU Course #: TC 725-N

**Internet Communications
Using SIP**

Session #4 -- Dr. Baker

August 30, 2002

“Live” Class Call-In Line: 214-768-3593

Engineering and Beyond

SIP Tutorial

Introduction to SIP

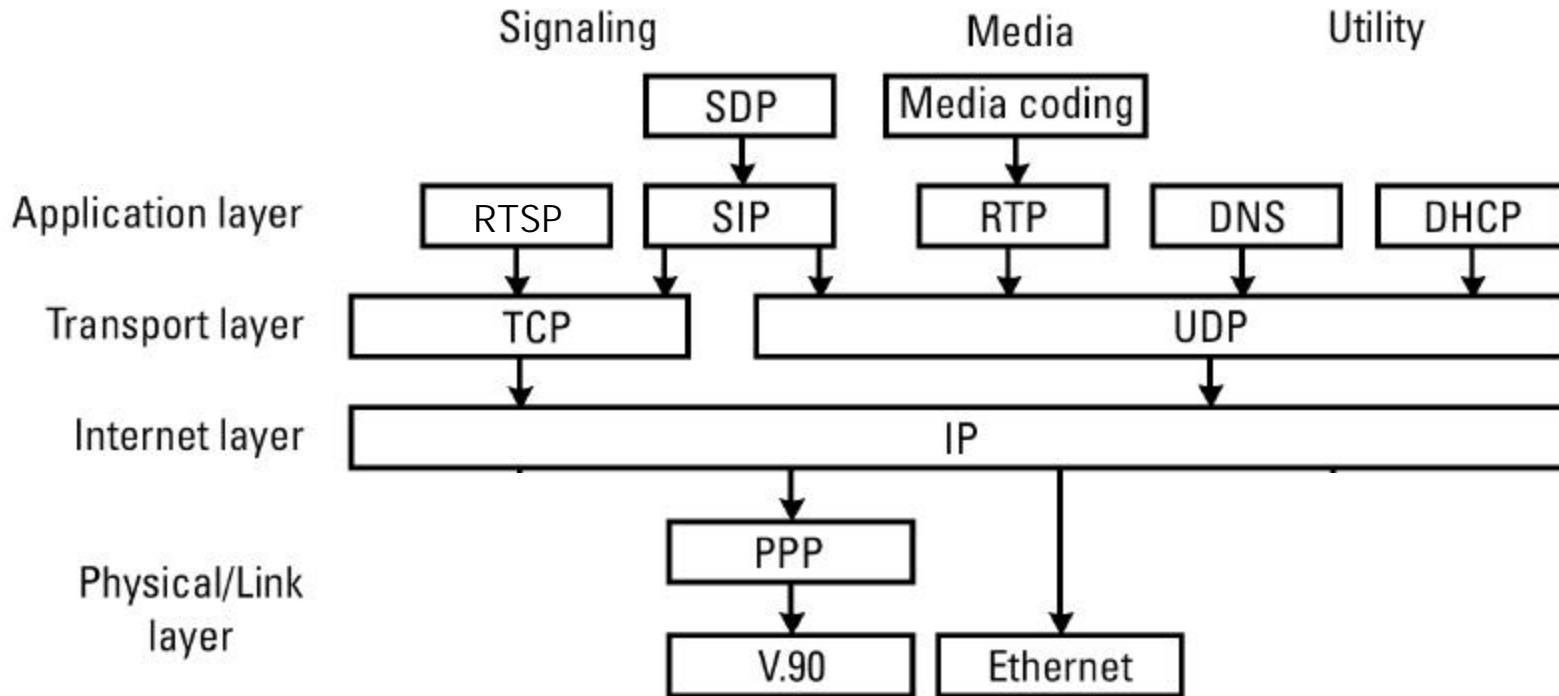
Henry Sinnreich

Alan Johnston

SIP Overview

- Application Layer Signaling Protocol
- Used to establish, modify, and terminate multimedia sessions
- Part of Internet Multimedia Architecture
- Can use UDP, TCP, TLS, SCTP, etc.
- Based on HTTP (Web)
 - Similar text-based structure
 - Uses URIs (Uniform Resource Indicators)
- Applications include (but not limited to):
 - Voice, video, gaming, instant messaging, presence, call control, etc.

Internet Multimedia Protocols



SIP Requests and Responses

SIP Request types are called "methods"

Methods in base spec:

INVITE

ACK

OPTIONS

CANCEL

BYE

REGISTER

SIP Responses use a numerical code:

Classes:

1xx

Informational

2xx

Final

3xx

Redirection

4xx

Client Error

5xx

Server Error

6xx

Global Failure

Example: 404 Not Found

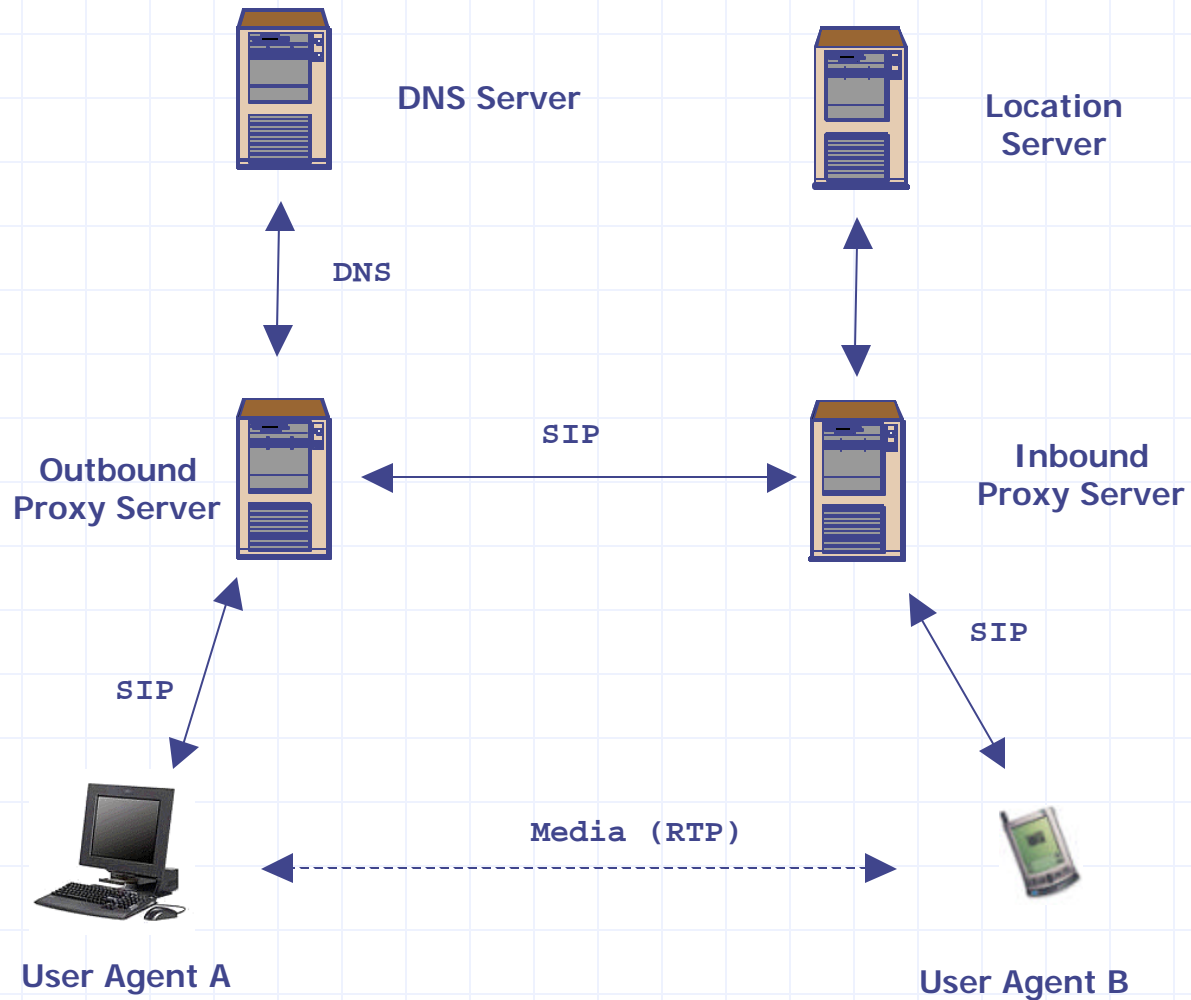
SIP Uniform Resource Indicators (URIs)

- Same form as email addresses: user@domain
- Two URI schemes:
 - sip:henry@siptest.wcom.com is a SIP URI
 - Most common form introduced in RFC 2543
 - sips:henry@siptest.wcom.com is a Secure SIP URI
 - New scheme introduced in RFC 3261
 - Requires TLS over TCP as transport for security
- Two types of SIP URIs:
 - Address of Record (AOR) (identifies a user)
 - sip:henry@wcom.com (Need DNS SRV records to locate SIP Servers for wcom.com domain)
 - Fully Qualified Domain Name (FQDN) (identifies a device)
 - sip:henry@127.24.45.4 or sip:henry@cube43.lab.wcom.com

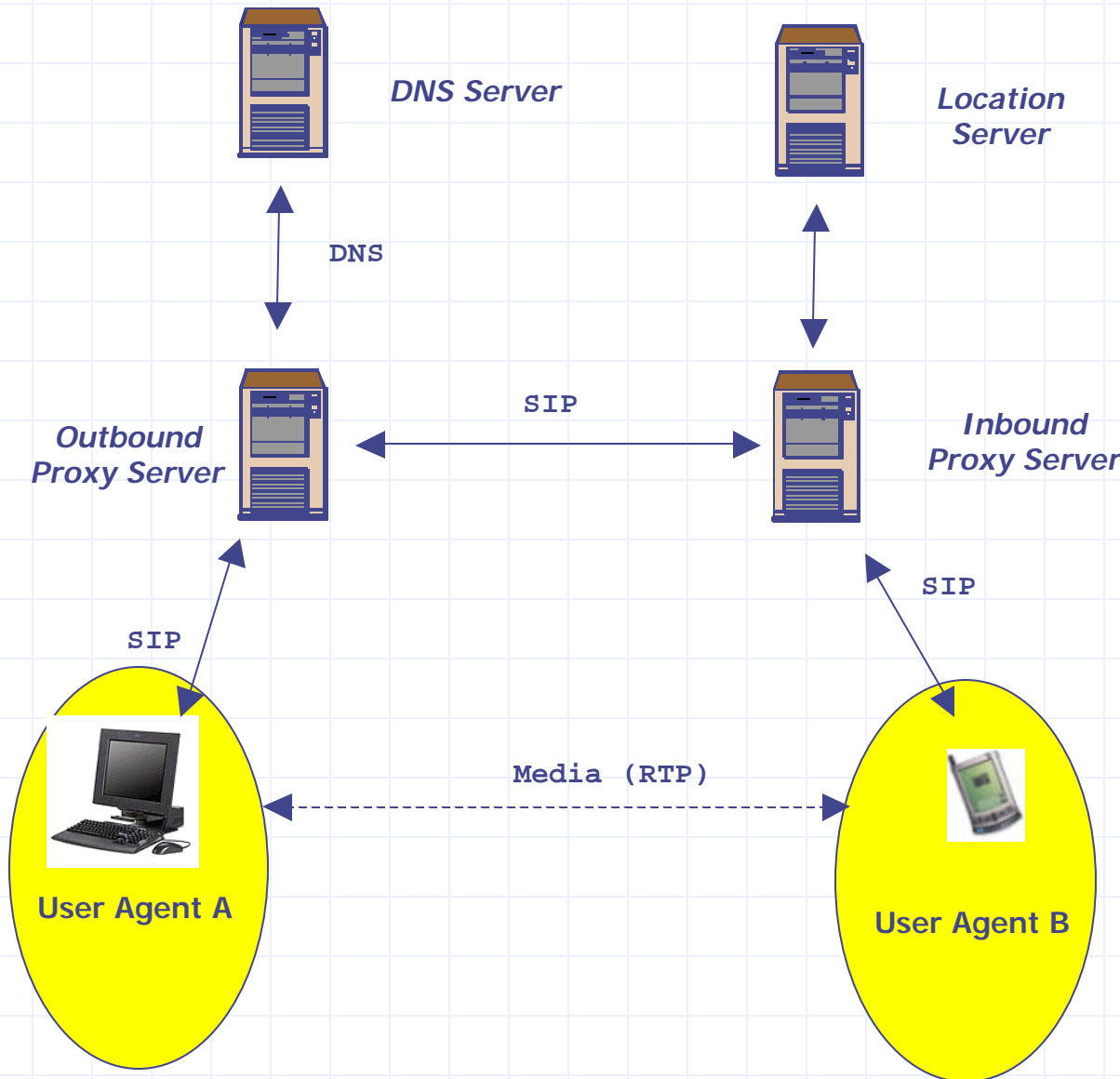
Related Protocols

- ◆ SDP – Session Description Protocol
 - Used to describe media session.
 - Carried as a message body in SIP messages.
 - Is a text-based protocol
 - Uses RTP/AVP Profiles for common media types
 - Defined by RFC 2327
- ◆ RTP – Real-time Transport Protocol
 - Used to transport media packets over IP
 - RTP adds a bit-oriented header containing:
 - name of media source
 - timestamp
 - codec type
 - sequence number
 - Defined by H. Schulzrinne et al, RFC 1889.
 - Profiles defined by RFC 1890.

SIP "Trapezoid"



SIP "Trapezoid"



SIP User Agents

Capable of sending and receiving SIP requests.

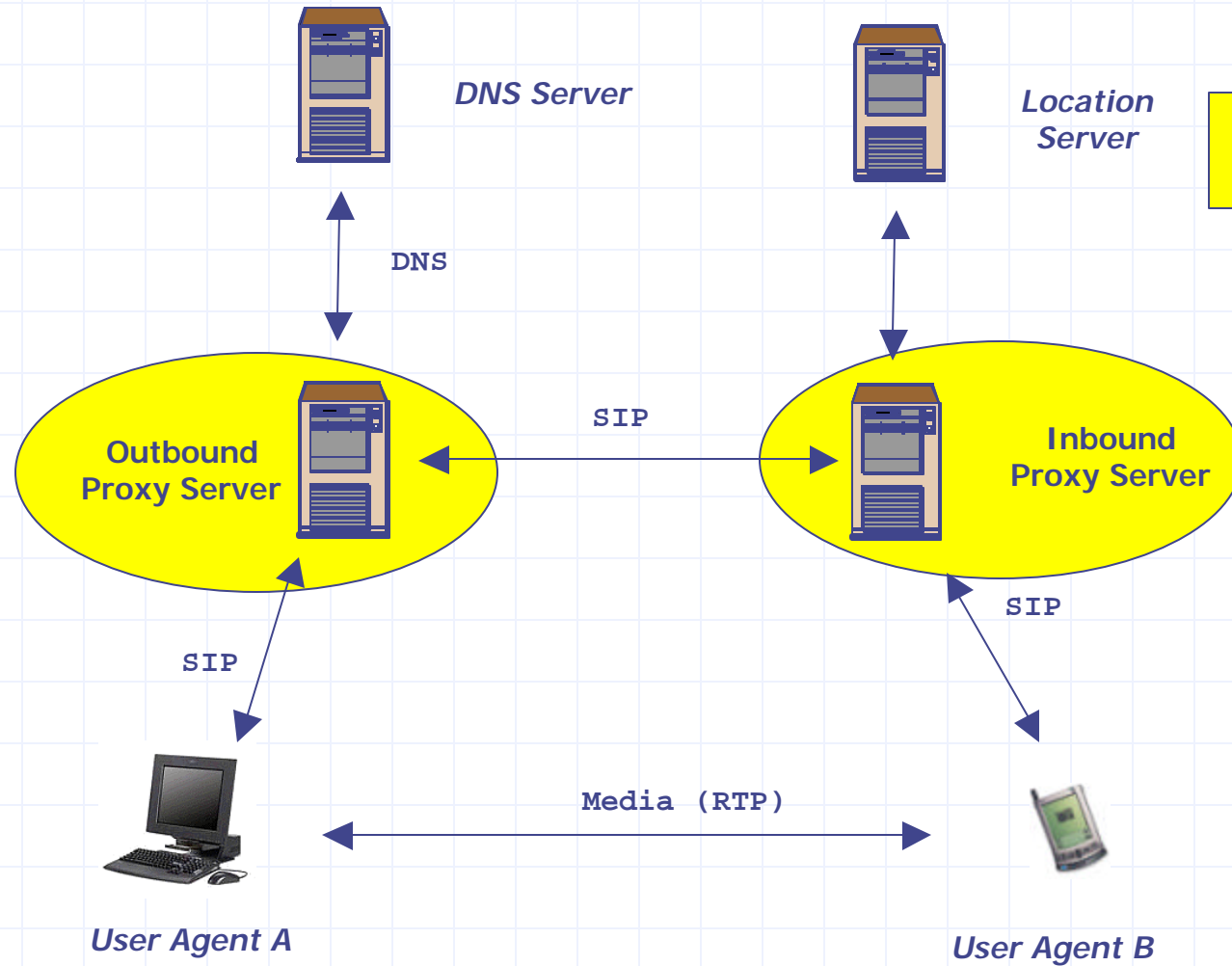
End Devices:

- SIP phone
- PC/laptop with SIP Client
- PDA
- mobile phone

PSTN Gateways

are a type of User Agent

SIP "Trapezoid"



SIP Proxy Servers

Forward or "proxy" requests on behalf of User Agents

Route requests

Consult databases:

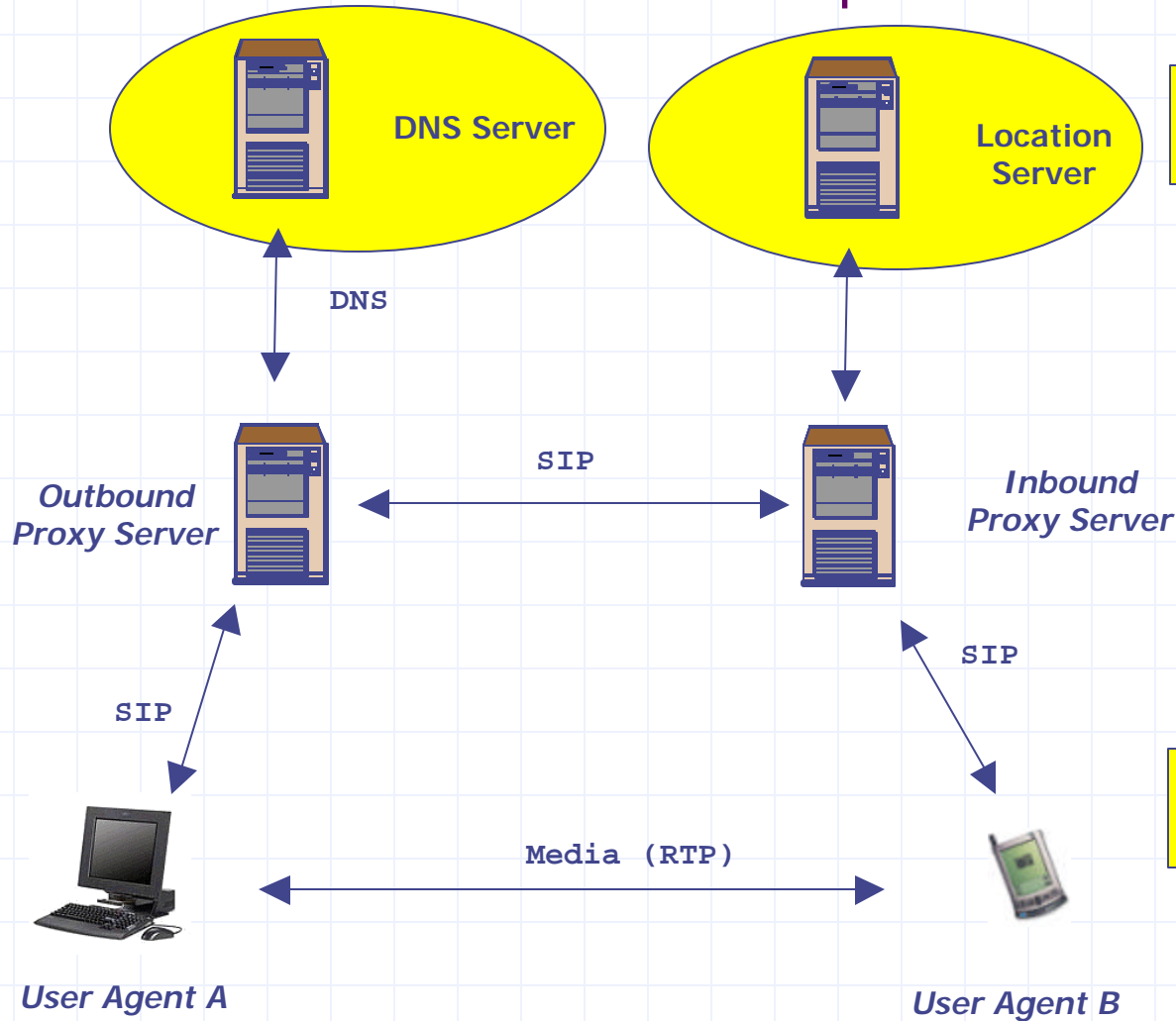
- DNS
- Location Server

Types:

- Stateless
- Transaction Stateful
- Call Stateful

Can be any number

SIP "Trapezoid"



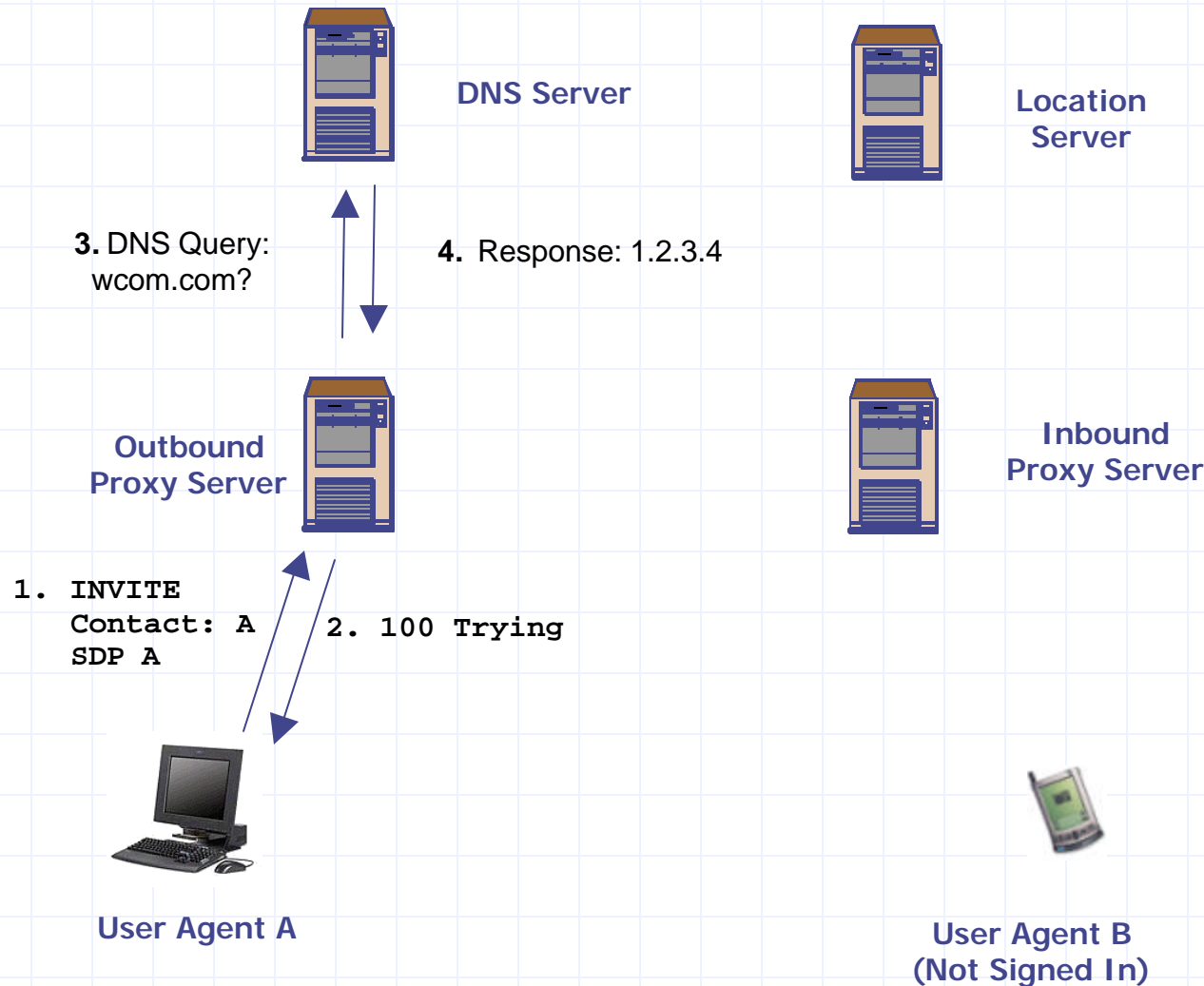
Location Server

Database of locations of SIP User Agents
Queried by Proxies in routing
Updated by User Agents by Registration

DNS Server

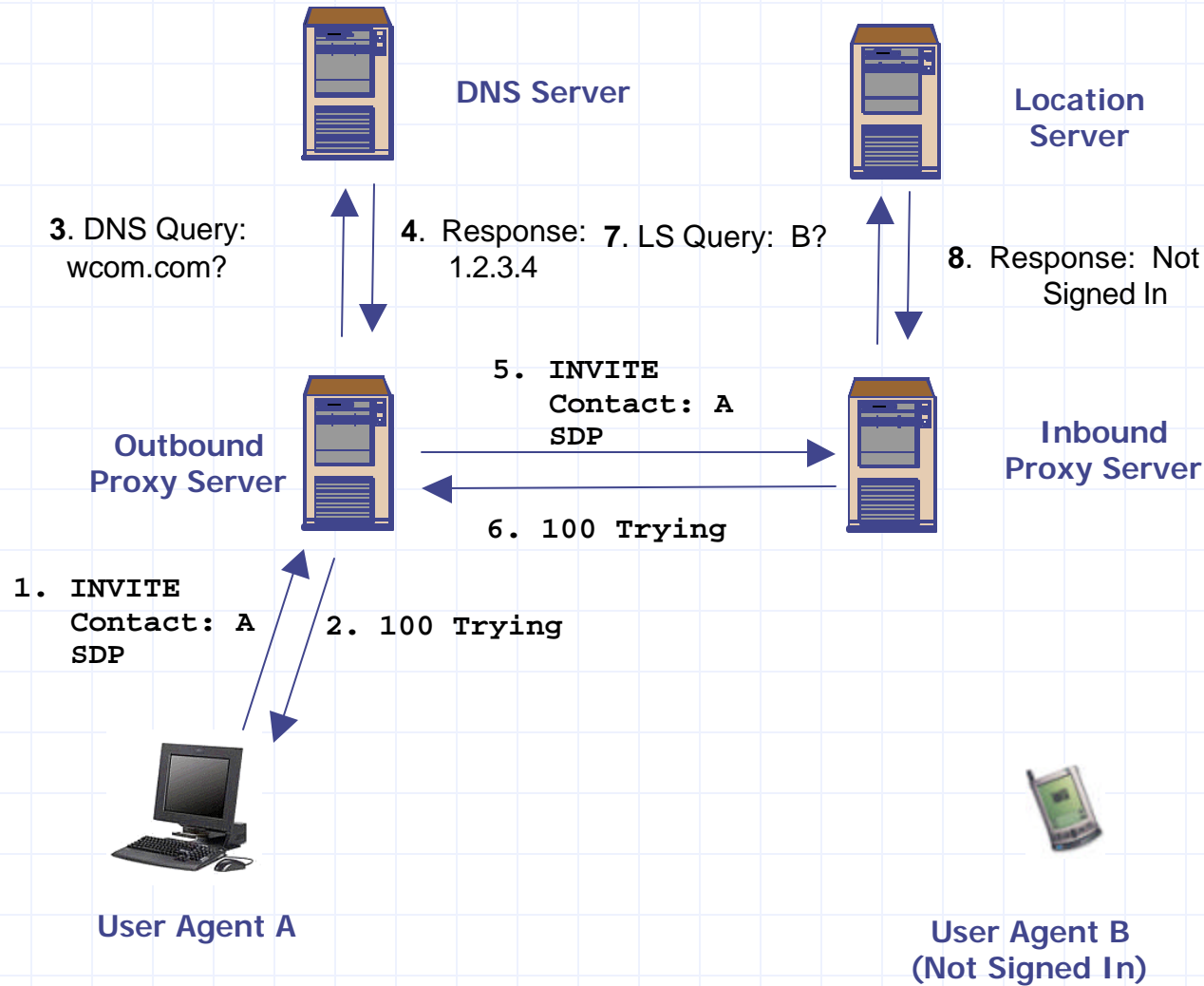
SRV (Service) Records used to locate Inbound Proxy Servers

SIP Call Setup Attempt Scenario



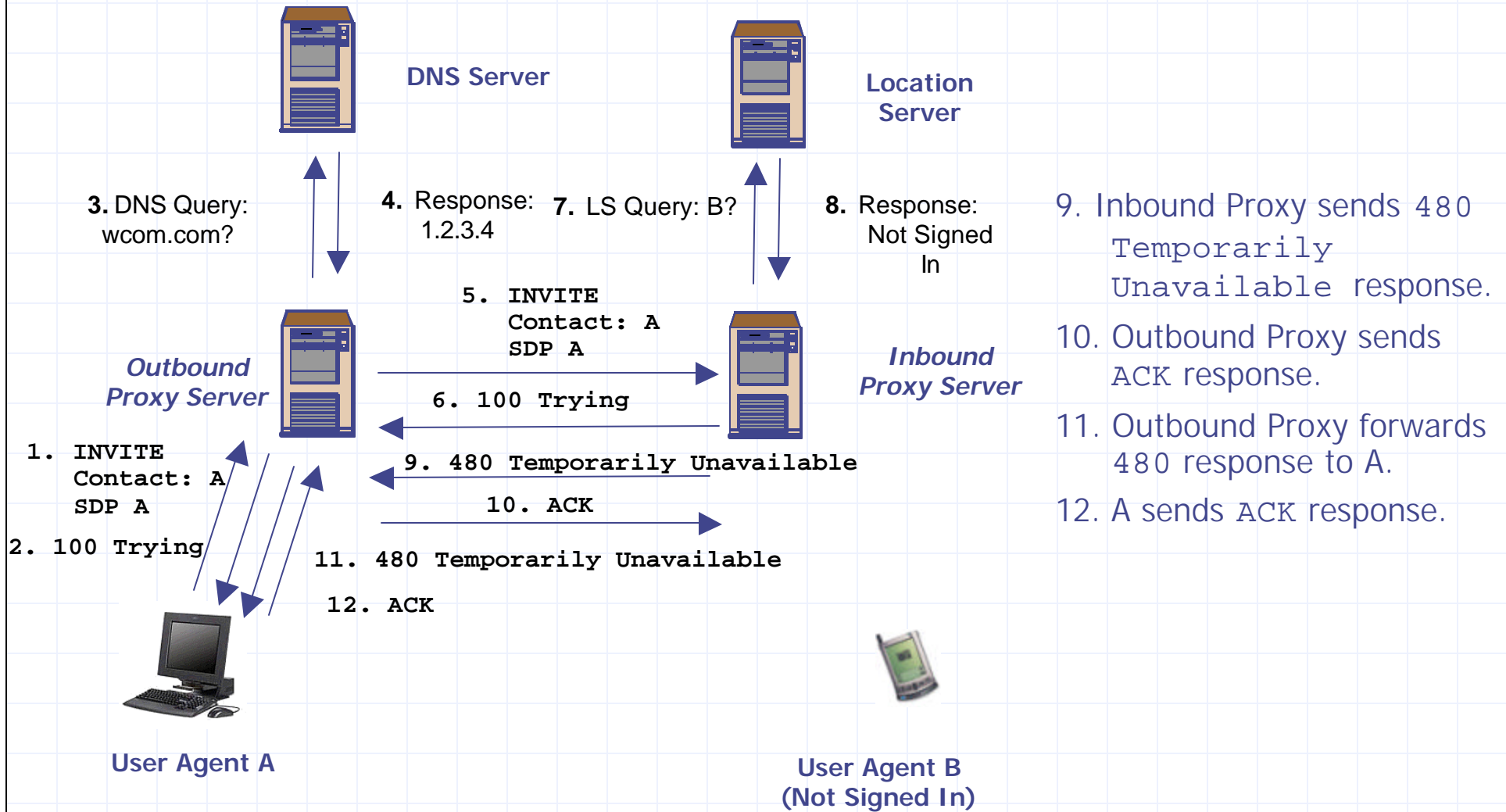
1. A "dials" SIP AOR URI sip:B@wcom.com. User Agent A sends INVITE to outbound Proxy Server.
2. Outbound Proxy sends 100 Trying response.
3. Outbound Proxy does DNS query to find proxy server for wcom.com domain
4. DNS responds with IP address of wcom.com Proxy Server

SIP Call Setup Attempt Scenario

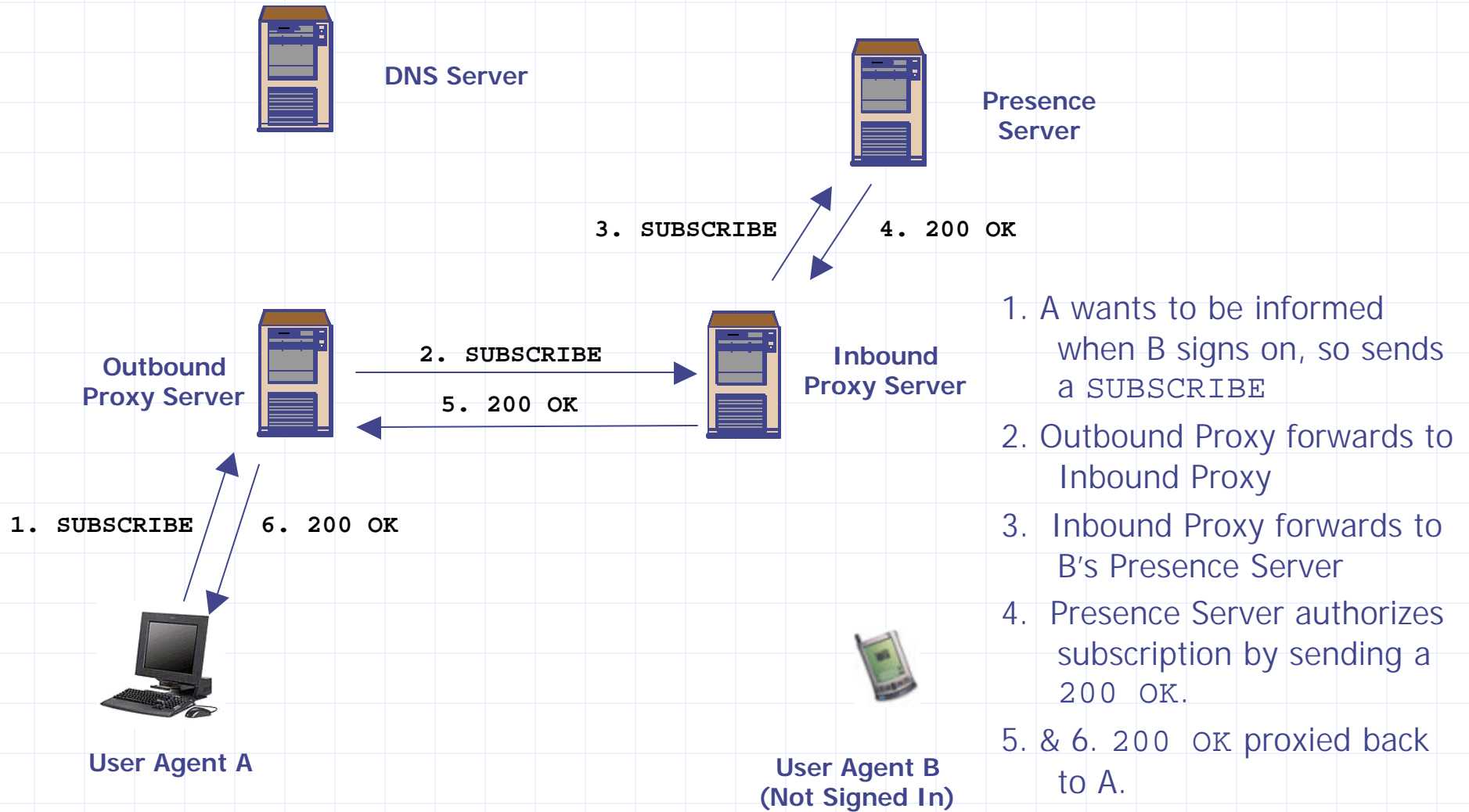


5. Outbound Proxy sends INVITE to Inbound Proxy Server.
6. Inbound Proxy sends 100 Trying response.
7. Inbound Proxy consults Location Server.
8. Location Server responds with "Not Signed In."

SIP Call Setup Attempt Scenario

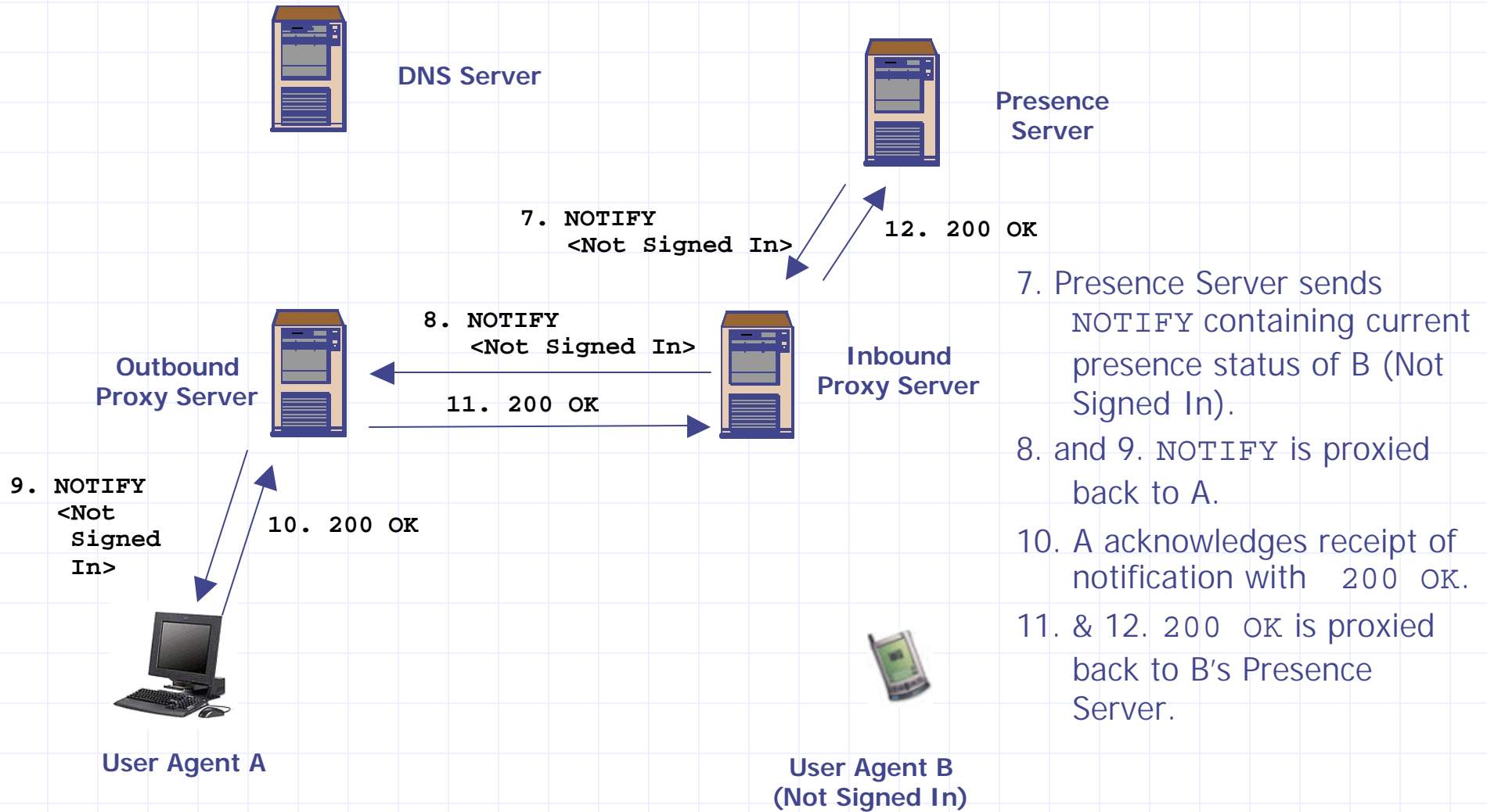


SIP Presence Example



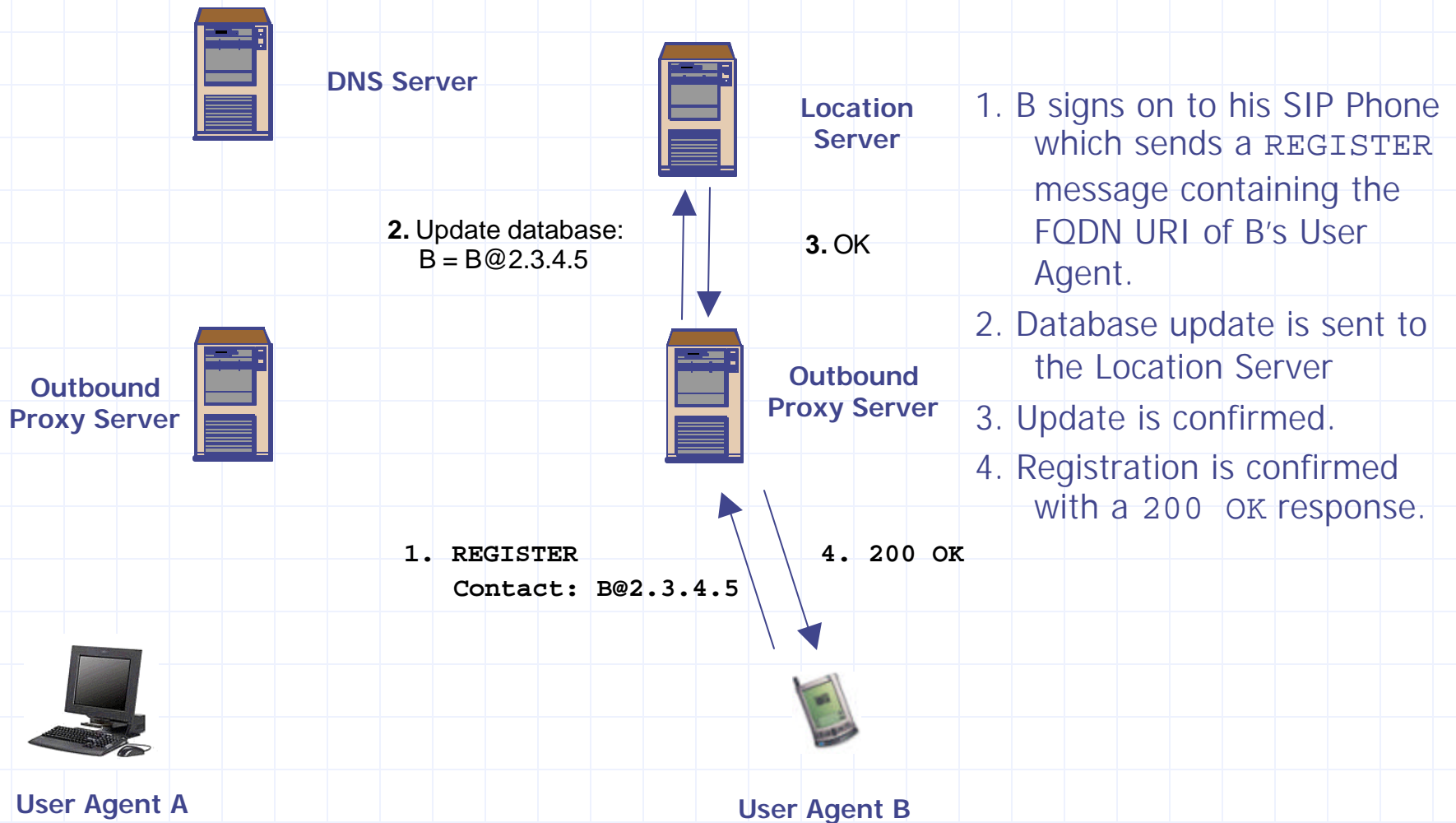
1. A wants to be informed when B signs on, so sends a SUBSCRIBE
2. Outbound Proxy forwards to Inbound Proxy
3. Inbound Proxy forwards to B's Presence Server
4. Presence Server authorizes subscription by sending a 200 OK.
5. & 6. 200 OK proxied back to A.

SIP Presence Example

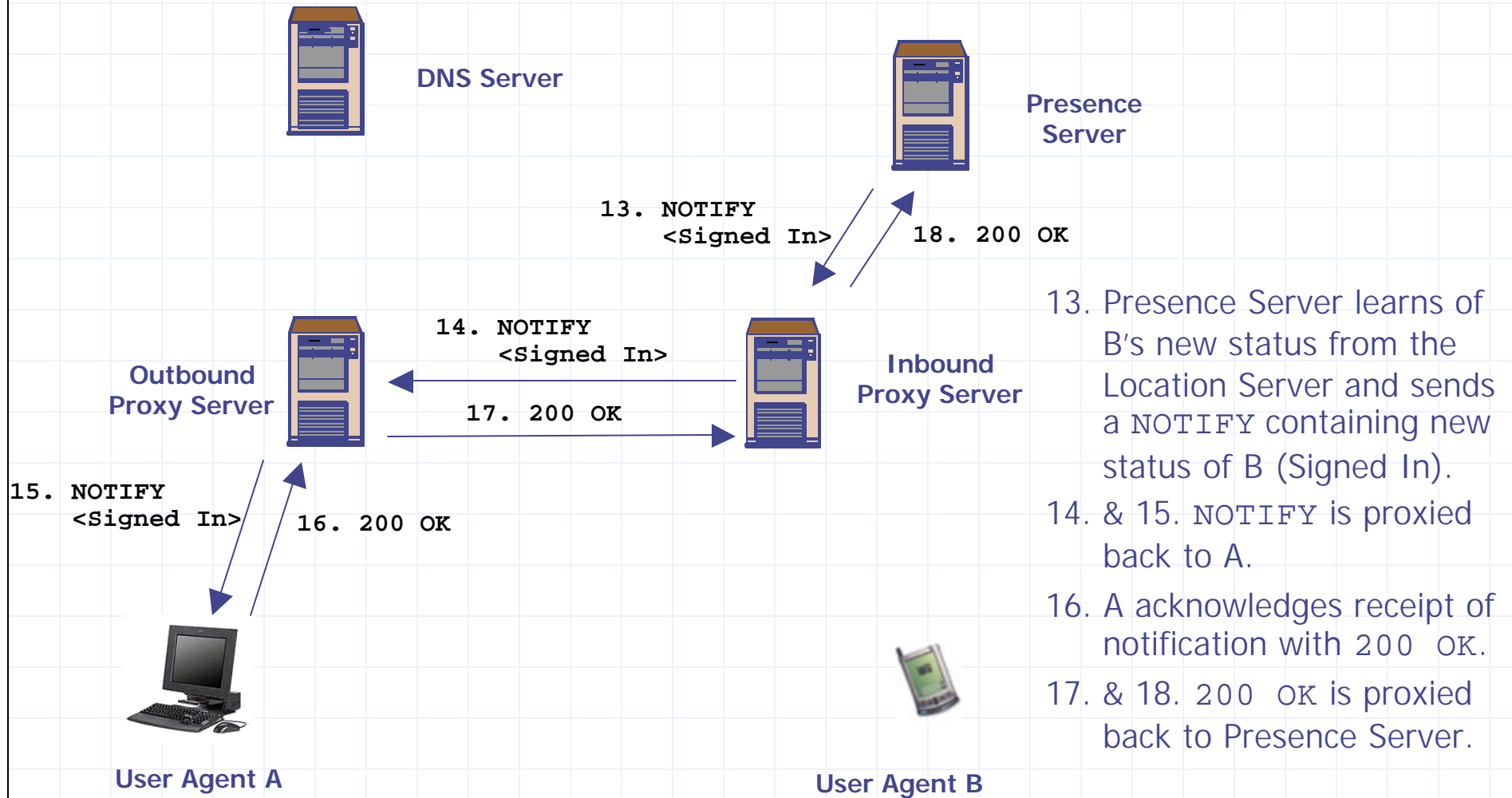


- 7. Presence Server sends NOTIFY containing current presence status of B (Not Signed In).
- 8. and 9. NOTIFY is proxied back to A.
- 10. A acknowledges receipt of notification with 200 OK.
- 11. & 12. 200 OK is proxied back to B's Presence Server.

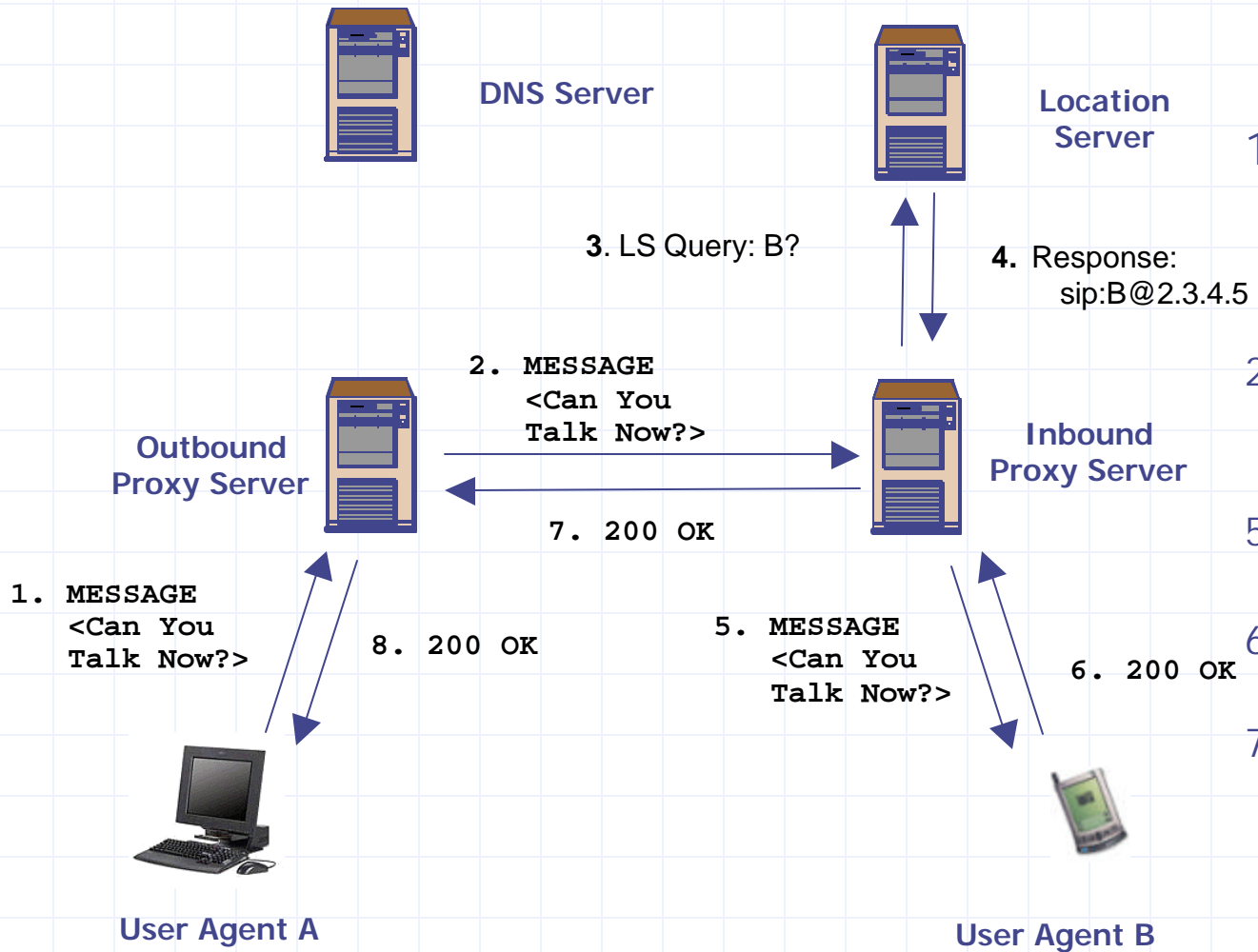
SIP Registration Example



SIP Presence Example

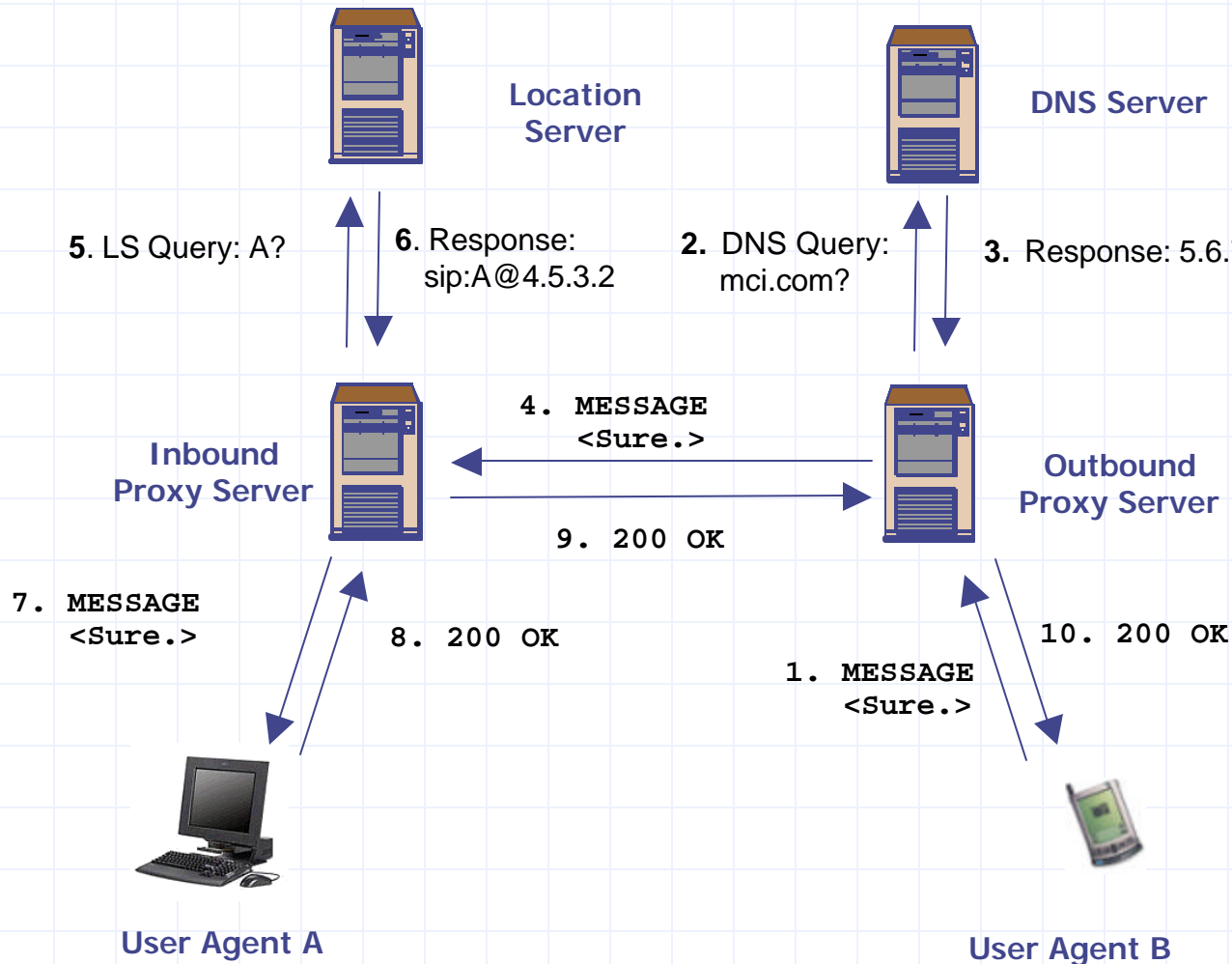


SIP Instant Message Scenario



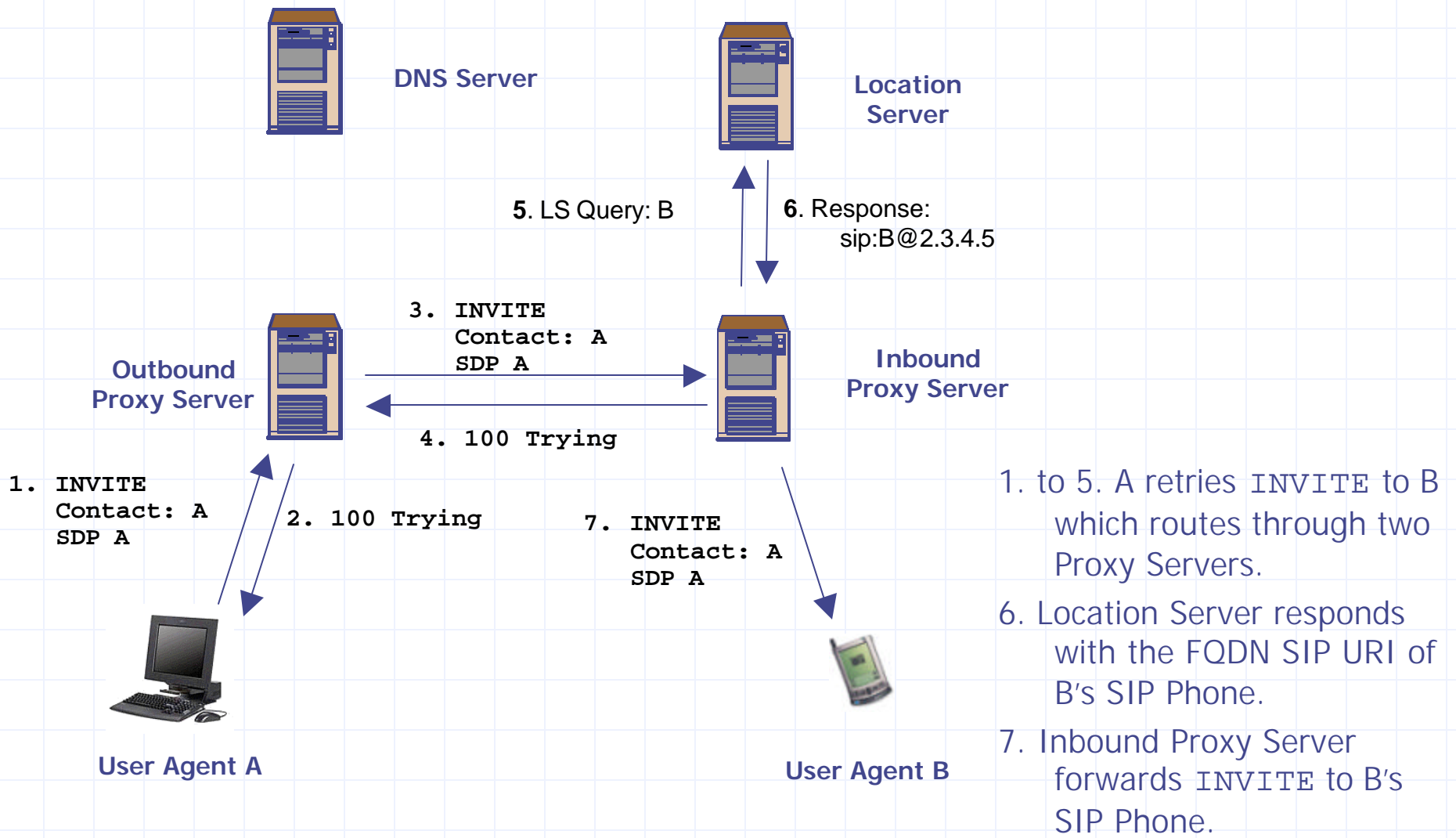
1. A sends an Instant Message to B saying "Can you talk now?" in a MESSAGE request.
- 2., 3. & 4. MESSAGE request is proxied, Location Server queried.
5. Inbound Proxy forwards MESSAGE to B.
6. User Agent B responds with 200 OK.
7. & 8. 200 OK is proxied back to A.

SIP Instant Message Scenario



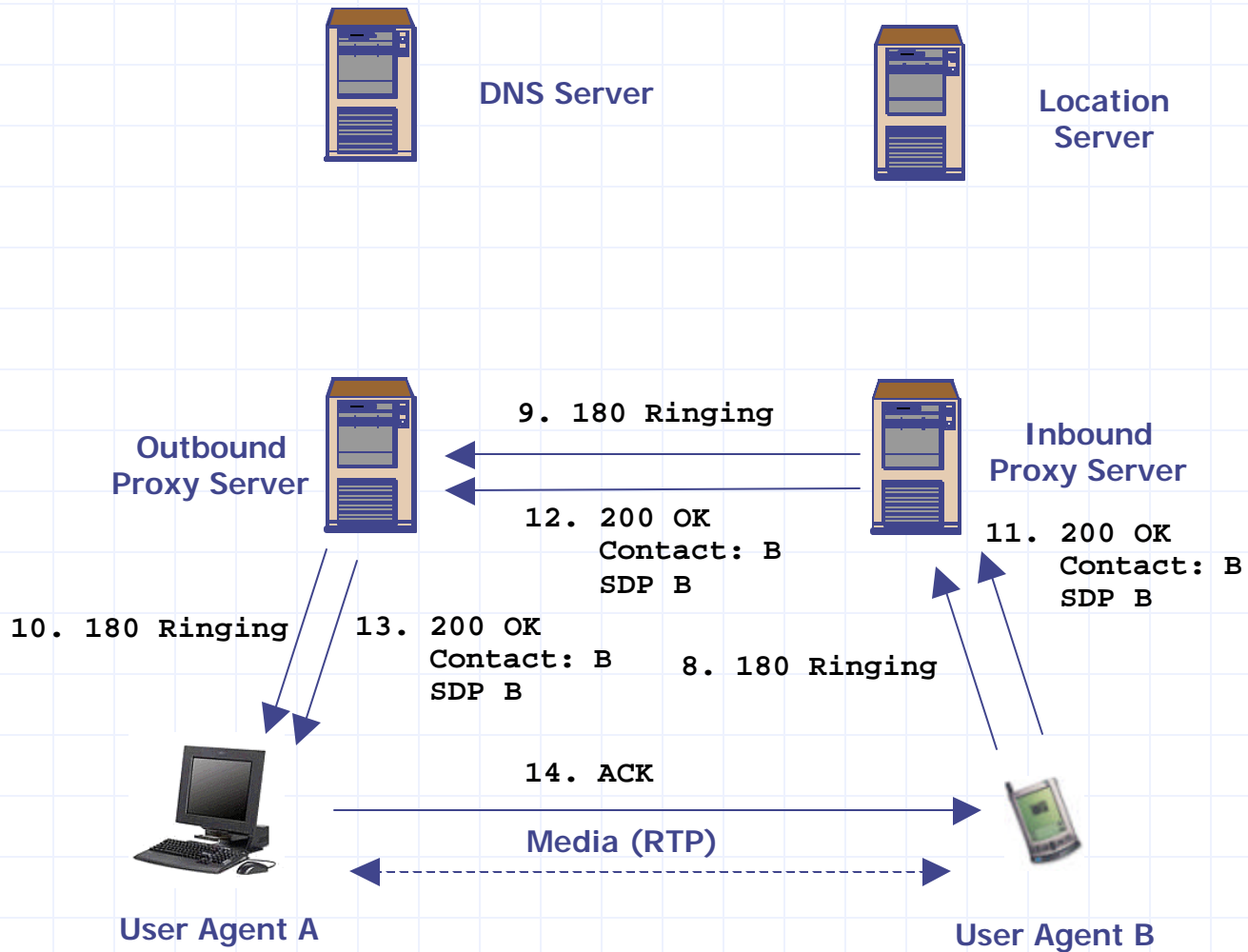
1. B sends an Instant Message to A saying "Sure." in a MESSAGE sent to A's AOR URI.
2. & 3. DNS Server is queried.
4. Outbound Proxy forwards MESSAGE to Inbound Server.
5. & 6. Location Server is queried.
7. Inbound Proxy forwards to A.
8. User Agent A responds with 200 OK.
9. & 10. 200 OK is proxied back to B.

SIP Call Setup Attempt Scenario



- 1. to 5. A retries INVITE to B which routes through two Proxy Servers.
- 6. Location Server responds with the FQDN SIP URI of B's SIP Phone.
- 7. Inbound Proxy Server forwards INVITE to B's SIP Phone.

SIP Call Setup Scenario

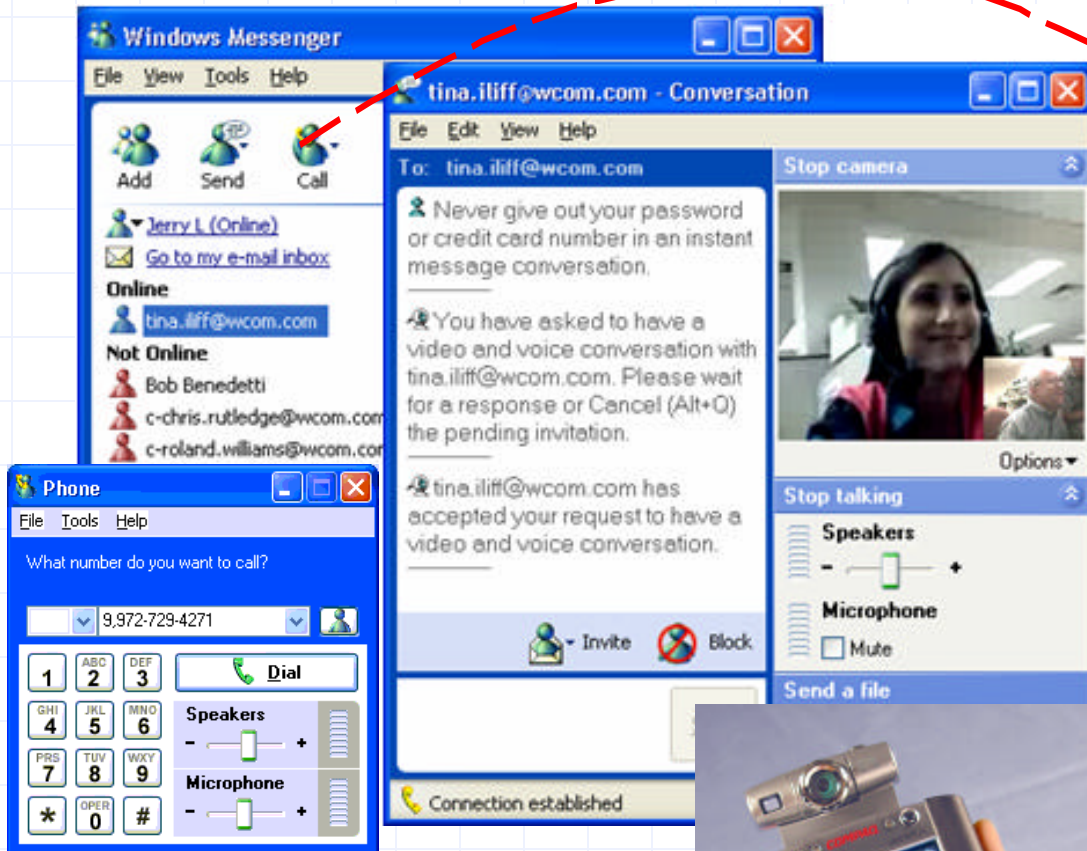


- 8. User Agent B alerts B and sends 180 Ringing response.
- 9. & 10. 180 Ringing is proxied back to A.
- 11. B accepts call and User Agent B sends 200 OK response.
- 12. & 13. 200 OK is proxied back to A.
- 14. ACK is sent by A to confirm setup call bypassing proxies.

SIP IP Communication Devices

Microsoft Messenger based on SIP

Commworks Network SIP Gateway



1. POTS Connectivity

- PC to phone

2. IP Communications

- Presence
- Text messaging (IM)
- Voice, telephony
- Video

3. Data Collaboration

- Whiteboard
- Application sharing
- FTP



CISCO

8/30/2002



Pingtel



Microsoft Portrait



Snom AG



Mitel

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SIP Based Service Example

WorldCom US - US Products - The WorldCom Connection - Technical Diagram - Microsoft Internet Explorer

Address: http://www.worldcom.com/us/products/connection/diagram/

US Home | About WorldCom | US Products | US Programs | Tools | Resources | News | Contact Us

THE WORLDCOM CONNECTION

A New World in Communications Management

The WorldCom ConnectionSM simplifies and streamlines network communication for easy mobility, efficient communication, and a single source for all services. The difference is visible in these network architecture diagrams.

Before The WorldCom Connection, a company might operate three or more different networks for data and IP, long distance calling, and local voice services. This model is inefficient and expensive, with a closed PBX architecture and extra charges for any moves, additions, or changes to the existing structure. Check it out:

Old World

- 3 networks
- Duration/geography-based pricing
- Expensive moves, adds, and changes
- Standalone application
- Closed PBX architecture

Enter WorldCom Connection, and the world of communications management gets a whole lot simpler for IT managers, offering potential cost savings on the bottom line. Since WorldCom is a premier network provider, we provide our services in the network - instead of at the customer's premises.

- First, by providing direct access to the PSTN, service is expanded beyond the LAN to the WAN.
- Second, centralized intelligence is offered. Customers are able to access reporting via a Web browser to conveniently control and manage their network.
- Third, WorldCom inures the costs of buying major equipment, again limiting the customer's risk and capital investment.

New World

- ONE source for all services
- Network access to PSTN - expands service to the WAN
- Centralized, network-based intelligence
- WorldCom provides the technology
- Easy mobility

Resources for SIP

- ◆ Henning Schulzrinne's SIP Page
<http://www.cs.columbia.edu/sip>
- ◆ IP Telephony Web Page
<http://iptel.org>
- ◆ SIP Forum
<http://www.sipforum.vom>
- ◆ SIP Center
<http://www.sipcenter.com>
- ◆ SIP Products at Pulver.com
<http://www.pulver.com/sip/products.html>
- ◆ Books
 - "SIP: Understanding the Session Initiation Protocol," Alan B. Johnston, Artech House, 2001.
 - "Internet Communications Using SIP" by Henry Sinnreich and Alan B. Johnston, Wiley, 2001.
 - "SIP Demystified" by Gonzalo Camarillo, McGraw-Hill, 2001.

Attachment:

SIP Message Details

SIP Message Details

```
INVITE sip:wh@200.201.202.203 SIP/2.0
Via: SIP/2.0/UDP proxy.munich.de:5060;branch=82.1
Via: SIP/2.0/UDP 100.101.102.103:5060
Max-Forwards: 70
To: Heisenberg <sip:w.heisenberg@munich.de>
From: E. Schroedinger <sip:schroed5244@aol.com>
Call-ID: 105637921@100.101.102.103
CSeq: 1 INVITE
Contact: sip:schroed5244@100.101.102.103
Content-Type: application/sdp
Content-Length: 159
```

- ◆ Start Line is the first line of a SIP message which contains:
 - the method or Request type: **INVITE**
 - the Request-URI which indicates who the request is for
`sip:wh@200.201.202.203`
 - the SIP version number **SIP/2.0**

SIP Message Details

```
INVITE sip:w.h@200.201.202.203 SIP/2.0
Via: SIP/2.0/UDP proxy.munich.de:5060;branch=82.1
Via: SIP/2.0/UDP 100.101.102.103:5060
Max-Forwards: 70
To: Heisenberg <sip:w.heisenberg@munich.de>
From: E. Schroedinger <sip:schroed5244@aol.com>
Call-ID: 105637921@100.101.102.103
CSeq: 1 INVITE
Contact: sip:schroed5244@100.101.102.103
Content-Type: application/sdp
Content-Length: 159
```

- ◆ Via headers show the path the request has taken in the SIP network
 - The bottom via header is inserted by the User Agent which initiated the request
 - The top via headers are inserted by proxies in the path
- ◆ The via headers are used to route responses back the same way

SIP Message Details

```
INVITE sip:w.h@200.201.202.203 SIP/2.0
Via: SIP/2.0/UDP proxy.munich.de:5060;branch=82.1
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```

- ◆ Max-Forwards is a count decremented by each proxy that forwards the request.
- ◆ When count goes to zero, request is discarded and 483 Too Many Hops response is sent.
- ◆ Used for stateless loop detection.

SIP Message Details

```
INVITE sip:w.h@200.201.202.203 SIP/2.0
Via: SIP/2.0/UDP proxy.munich.de:5060;branch=82.1
Via: SIP/2.0/UDP 100.101.102.103:5060
Max-Forwards: 70
To: Heisenberg <sip:w.heisenberg@munich.de>
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```

- ❖ Dialog (formerly call leg) information is in headers:
 - To tag, From tag, and Call-ID
- ❖ All requests and responses in this call will use this same Dialog information.
- ❖ Call-ID is unique identifier usually composed of
 - pseudo-random string "@" hostname or IP Address

SIP Message Details

```
INVITE sip:w.h@200.201.202.203 SIP/2.0
Via: SIP/2.0/UDP proxy.munich.de:5060;branch=82.1
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Content-Type: application/sdp
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```

- ◆ CSeq Command Sequence Number
 - Initialized at start of call (1 in this example)
 - Incremented for each subsequent request
 - Used to distinguish a retransmission from a new request
- ◆ Also contains the request type (method)

SIP Message Details

```
INVITE sip:w.h@200.201.202.203 SIP/2.0
Via: SIP/2.0/UDP proxy.munich.de:5060;branch=82.1
Via: SIP/2.0/UDP 100.101.102.103:5060
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Contact: sip:schroed5244@100.101.102.103
Content-Type: application/sdp
Content-Length: 159
```

- ◆ Contact header contains a SIP URL for direct communication between User Agents
 - If Proxies do not Record-Route, they can be bypassed
- ◆ Contact header is also present in 200 OK response

SIP Message Details

```
INVITE sip:w.h@200.201.202.203 SIP/2.0
Via: SIP/2.0/UDP proxy.munich.de:5060;branch=82.1
Via: SIP/2.0/UDP 100.101.102.103:5060
Max-Forwards: 70
To: Heisenberg <sip:w.heisenberg@munich.de>
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CSeq: 1 INVITE
Contact: sip:schroed5244@100.101.102.103
Content-Type: application/sdp
Content-Length: 159
```

- ◆ Content-Type indicates the type of message body attachment (others could be text/plain, application/cpl+xml, etc.)
- ◆ Content-Length indicates the octet (byte) count of the message body.

SDP Message Details

```
v=0
o=Tesla 289084526 28904526 IN IP4 lab.high-voltage.org
s=-
c=IN IP4 100.101.102.103
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

- Version number (*ignored by SIP*)
- Origin (*only version used by SIP*)
- Subject (*ignored by SIP*)
- Connection Data (IP Address)
- Time (*ignored by SIP*)
- Media (type, port, RTP/AVP Profile)
- Attribute (profile, codec, sampling rate)



SMU School of Engineering

End of Session #4
EETS 8393 TC 725-N
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