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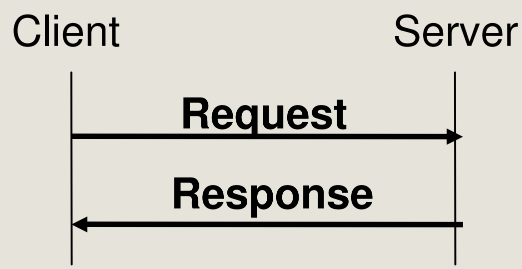
Session Initiation Protocol (SIP)

- SIP overview
- Logical entities in SIP
- SIP addresses
- Example of SIP messages
- SIP routing and Domain Name System (DNS)
- SIP registrations
- Routing: SIP server in proxy mode
- Routing: SIP server in redirect mode
- Summary of SIP methods
- Application areas of SIP
- Modularity of SIP
- The Session Description Protocol
- References, books, and further reading

Note well:
all of the following slides have been provided by Duncan Mills and Miguel.A.Garcia as part of a training course within the framework of the Vodafone/Ericsson IMS Trial Project

SIP general overview

- SIP is an end-to-end, client-server, extensible, text based protocol.
- The design base was HTTP and SMTP
- SIP was originally used to establish, modify and terminate multimedia sessions in the internet.
- SIP has evolved to be able to set-up a broad range of sessions:
  - Multimedia (e.g., voice, video, etc)
  - Gaming
  - Presence and Instant Messaging
- SIP messages are either requests or responses.
- SIP messages carry zero or more “bodies”.
- SDP is the common body for session initiation.
- SIP runs on any transport protocol (UDP, TCP, TLS, SCTP)
  - The spec mandates UDP and TCP. Other transport protocols are optional

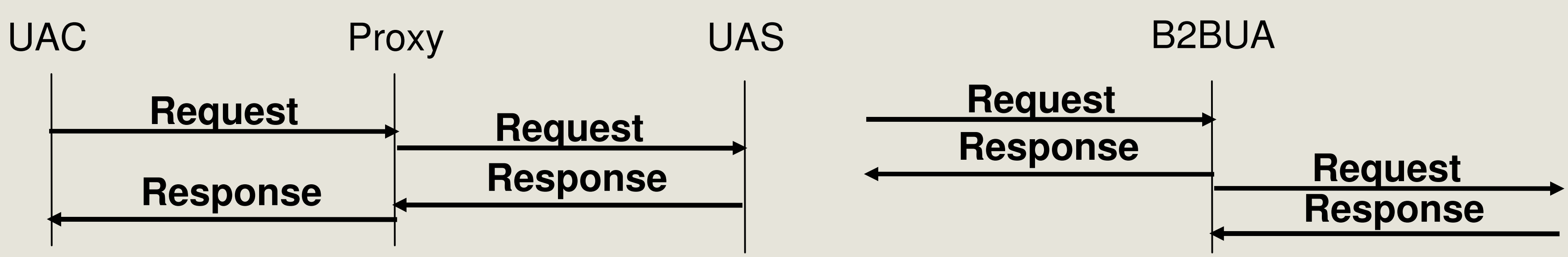


Overview of SIP functionality

- SIP provides the following functionality:
  - User location
  - User availability
  - User capabilities
  - Session set-up
  - Session management
- SIP does not provide services
  - But it enables the system to provide services
  - It has been demonstrated that it is easy to provide services with SIP

SIP logical entities

- User Agent (UA): An endpoint
  - User Agent Client (UAC): sends requests, receives responses
  - User Agent Server (UAS): receives requests, sends responses
- Proxy server: A network host that proxies requests and responses, i.e., acts as a UAC and as a UAS.
- Redirect server: a UAS that redirects requests to other servers.
- Back-to-back User Agent: a UAS linked to a UAC
  - Acts as a UAS and as a UAC linked by some application logic
- Registrar: A special UAS that accepts only registrations

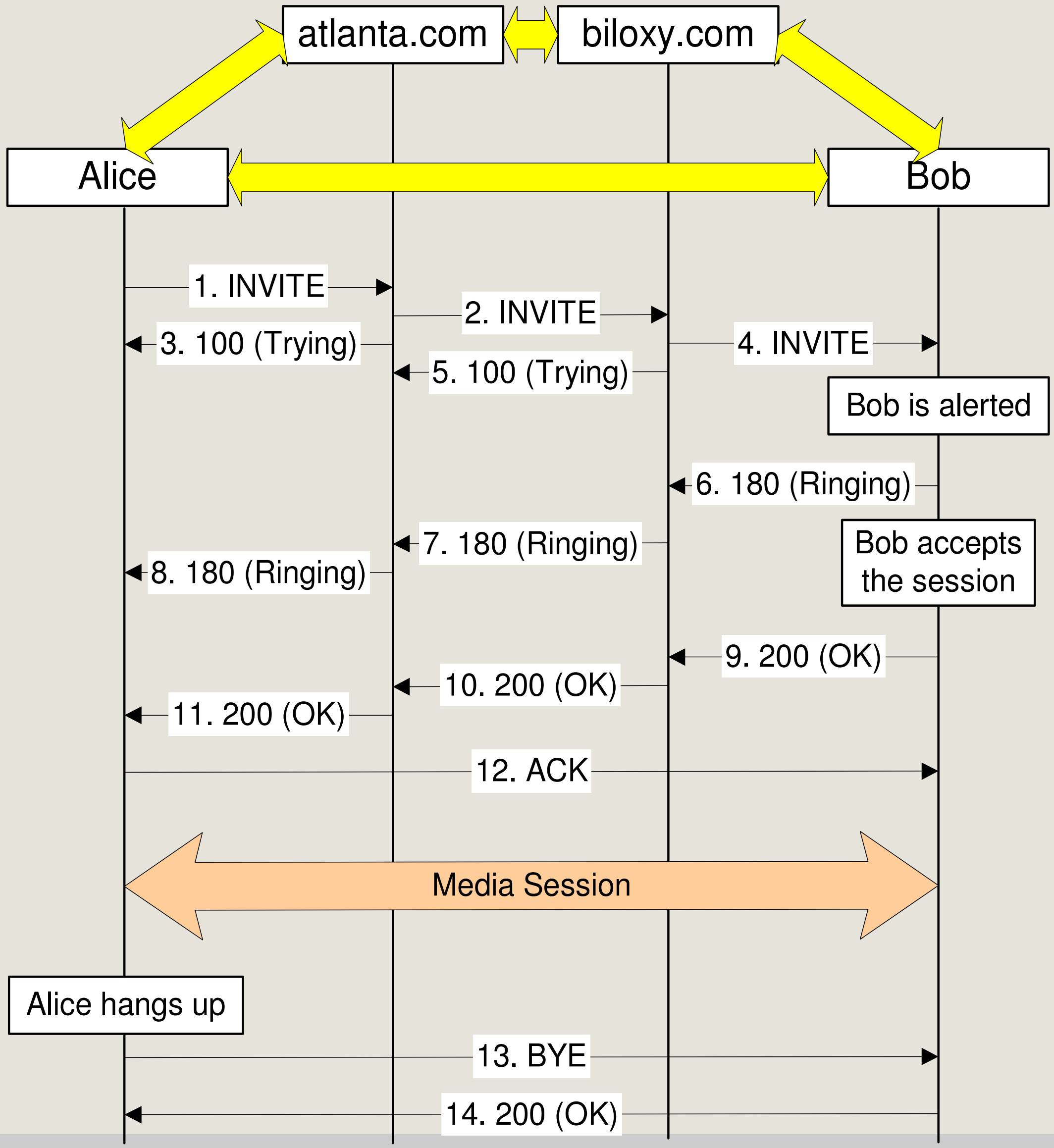


Stateless and stateful SIP proxies

There are several types of SIP proxies, depending on the state they keep:

- **Stateless proxy:** a proxy that does not keep any state when forwarding requests and responses.
- **Transaction stateful proxy, or stateful proxy:** a proxy that stores state during the duration of the transaction.
- **Call stateful proxy:** a proxy that stores all the state pertaining to a session (e.g., from INVITE to BYE). A call stateful proxy is always a transaction stateful proxy, but not the other way round.

The SIP trapezoid



SIP addresses

- SIP uses Uniform Resource Identifiers (URIs). At least, SIP URIs and SIPS URIs are supported, although others (such as TEL URL) are commonly supported.
  - sip:ciccio.pernacchio@pippozzo.com
  - sips:simon.romano@unina.it
  - tel:+358-9-299-3553
  - sip:proxy.atlanta.com:5060
  - sip:another-proxy.biloxi.com;transport=UDP
- SIP and SIPS URIs must include a host name, and may include username, may include port numbers, may include parameters
- Address space is unlimited
- Non SIP/TEL URIs are also valid under certain circumstances: HTTP, IM, PRES, MAILTO…

An example of a SIP request

Request Line

```
INVITE sip:miguel.a.garcia@ericsson.com SIP/2.0
```

Header

```
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd];branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.visited1.net;lr>, <sip:scscf1.home1.net;lr>
From: <sip:user1_public1@home1.net>;tag=171828
To: <sip:miguel.a.garcia@ericsson.com>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Contact: <sip:[5555::aaa:bbb:ccc:ddd]>
Content-Type: application/sdp
Content-Length: 248
```

Body

```
v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=audio 3458 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
```

Method

Version

Request-URI

Header Field

Header Field Name

Header Field Value

An example of a SIP response

Status Line

```
SIP/2.0 200 OK
```

SIP Version

Status code

Reason phrase

Header

```
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd];branch=z9hG4bKnashds7
Record-Route: <sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net;lr>
From: <sip:user1_public1@home1.net>;tag=171828
To: <sip:miguel.a.garcia@ericsson.com>;tag=314159
Call-ID: cb03a0s09a2sdfglkj490333
CSeq: 127 INVITE
Contact: <sip:[5555::eee:fff:aaa:bbb]>
Content-Type: application/sdp
Content-Length: 220
```

Body

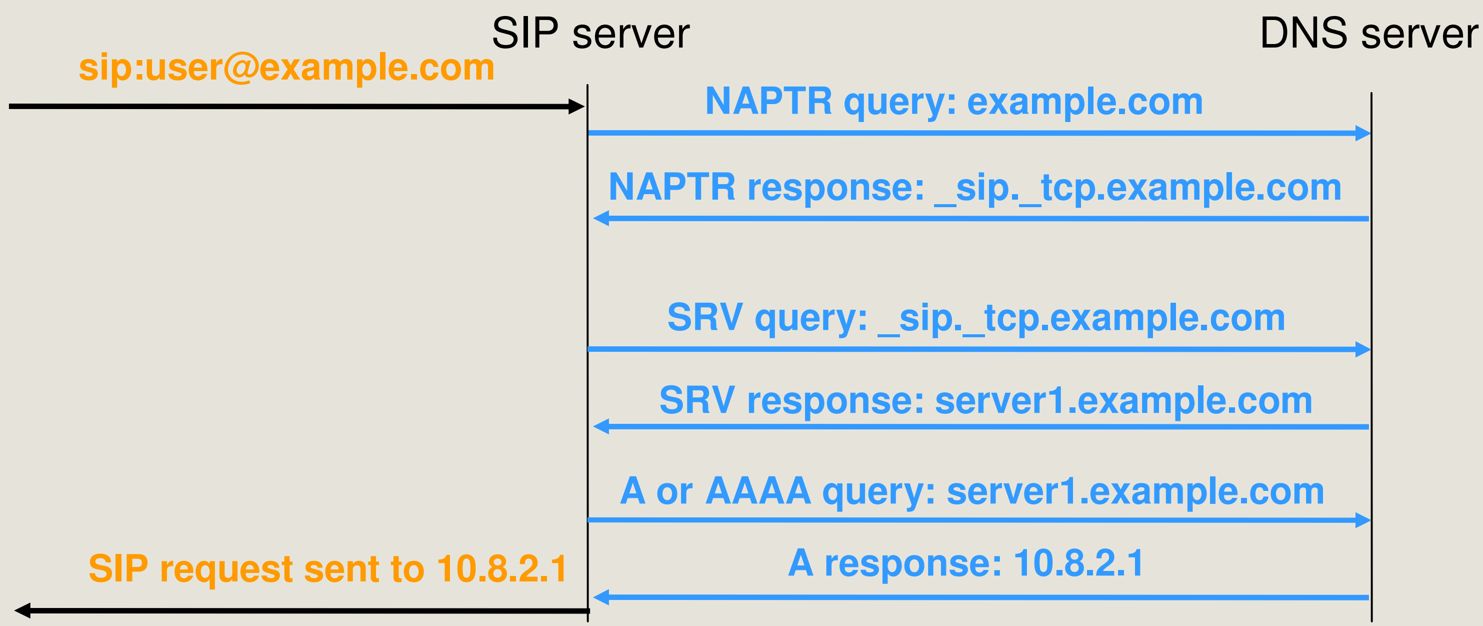
```
v=
o=- 2987933615 2987933615 IN IP6 5555::eee:fff:aaa:bbb
s=-
c=IN IP6 5555::eee:fff:aaa:bbb
t=907165275 0
m=audio 3458 RTP/AVP 97 0
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
```

Status codes in SIP

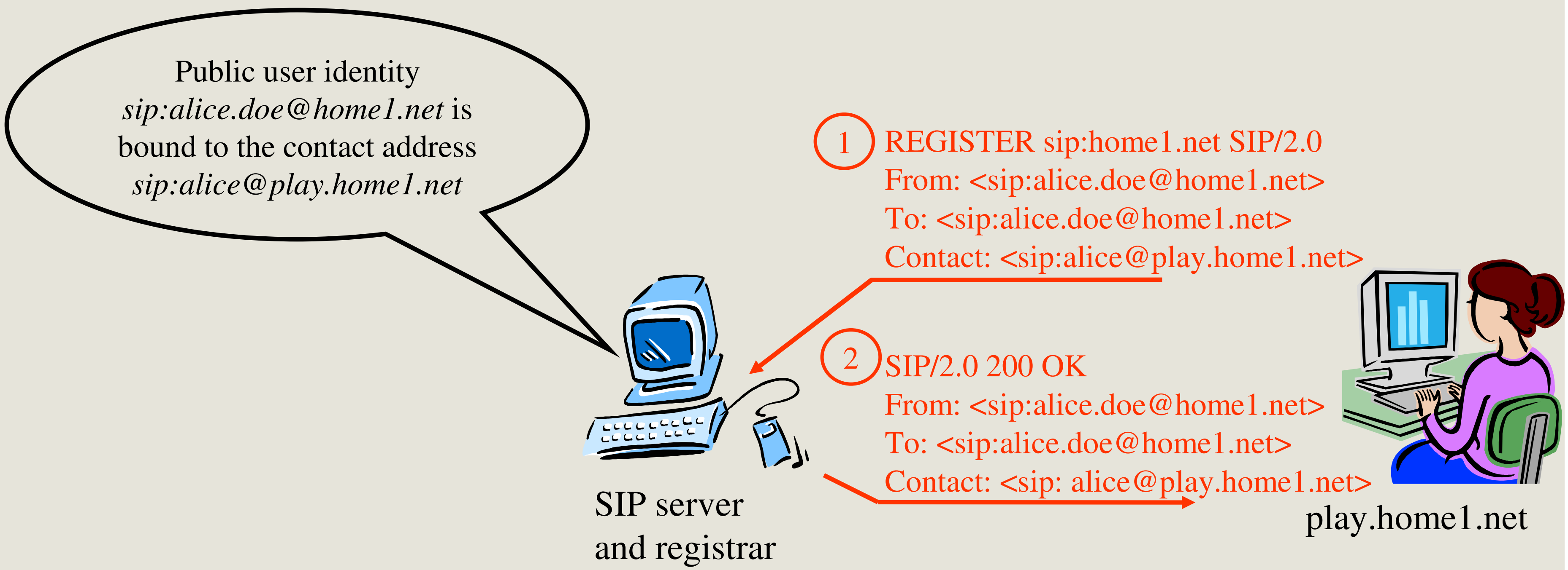
- **1xx - Provisional responses**
- **2xx - Success**
- **3xx - Redirection**
- **4xx - Client Error**
- **5xx - Server Error**
- **6xx - Global Failures**

SIP routing and Domain Name System

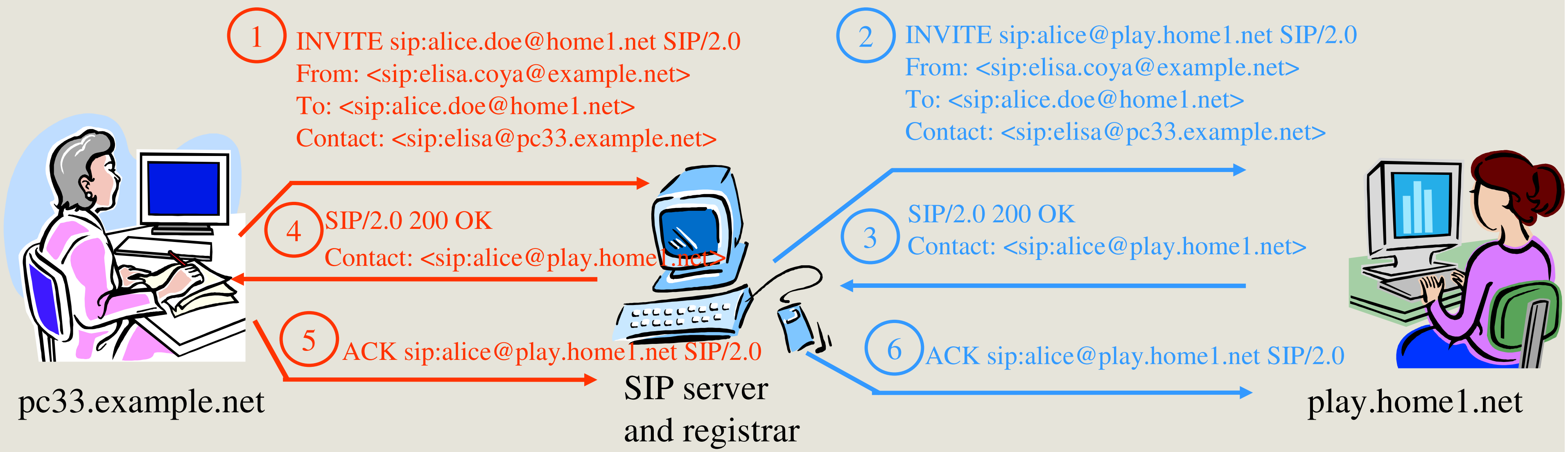
- SIP clients use DNS to route requests and find the next hop to route the request
  – By looking into a NAPTR (Naming Authority Pointer) record in DNS
  – By looking into a SRV (Services) record in DNS
  – By looking into A (IPv4) or AAAA (IPv6) records in DNS



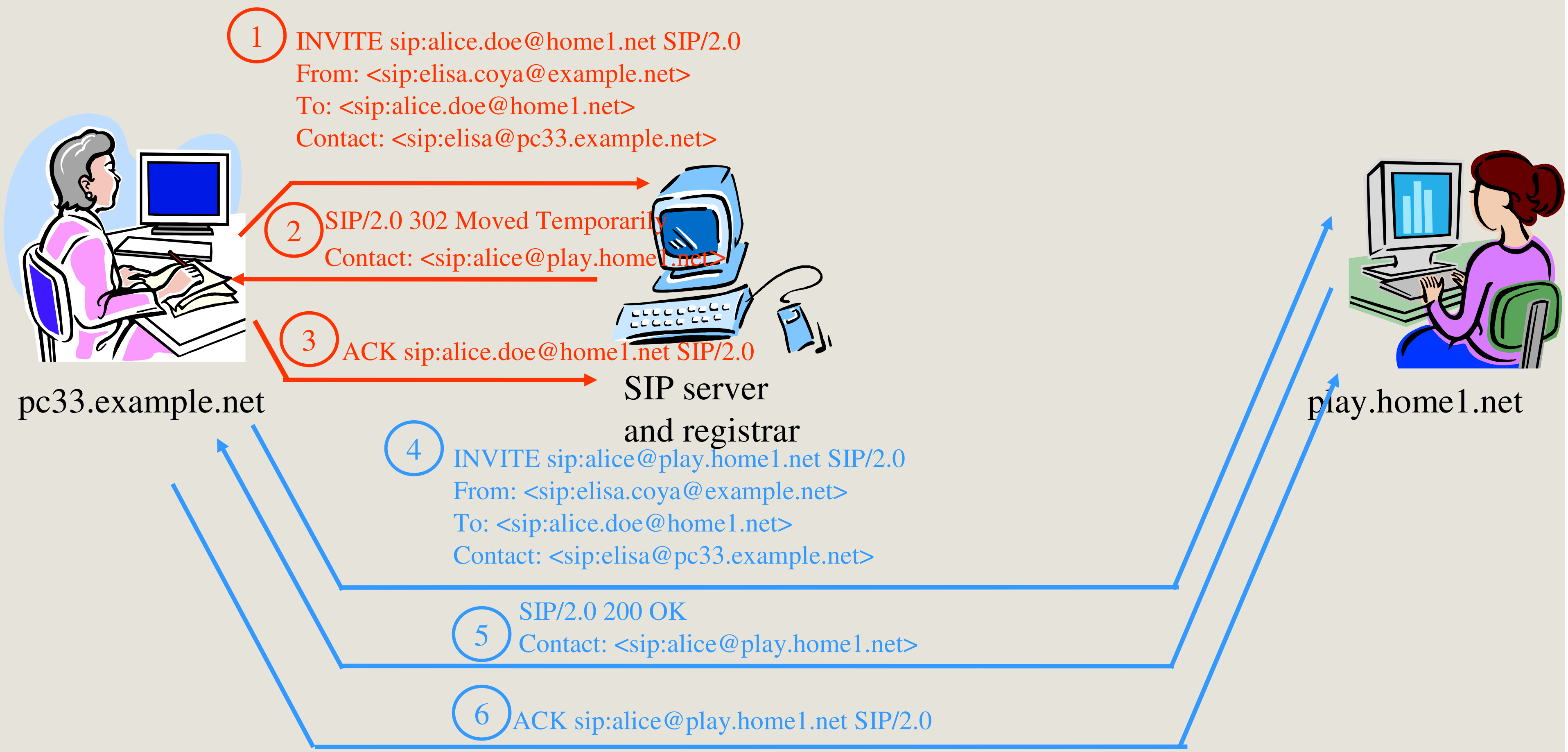
SIP registration



Routing: SIP server in proxy mode



Routing: SIP server in redirect mode



Summary of SIP methods

- INVITE: create a session
- BYE: terminates a session
- ACK: acknowledges a final response for an INVITE request
- CANCEL: cancels an INVITE request
- REGISTER: binds a public SIP URI to a Contact address
- OPTIONS: queries a server for capabilities
- SUBSCRIBE: installs a subscription for a resource
- NOTIFY: informs about changes in the state of the resource
- MESSAGE: delivers an Instant Message
- REFER: used for call transfer, call diversion, etc.
- PRACK: acknowledges a provisional response for an INVITE request
- UPDATE: changes the media description (e.g. SDP) in an existing session
- INFO: used to transport mid-session information
- PUBLISH: publication of presence information

Application areas of SIP

- SIP was originally developed to establish multimedia sessions on the internet (audio, video)
  - Mainly multicast sessions
  - Also unicast sessions
- But has evolved to support other aspects of communications:
  - Voice over IP calls between SIP terminals
  - Voice over IP calls between Gateways (SIP-T)
  - Gaming sessions
  - Instant messaging
  - Presence
  - Multimedia conferences
  - Machine-to-machine communication (e.g., vending machine notifications)

Modularity of SIP

- SIP is a modular extensible protocol. Depending on the desired application, some functions may or may not be present in an implementation.
  - Because of extensions
  - Or because of options in the core protocol.
  - Unlike ISUP, there are not different SIP flavours
  - But there are different functionalities, security mechanism, methods, headers, options, transport protocols, etc., that may or may not be implemented.
  - Generally, SIP contains mechanisms to discover what is supported by a proxy or remote end.
    - Require, Supported, Proxy-Require, Allow headers
    - Contact header in registration

Who is using SIP

- Third Generation Partnership Project (3GPP) uses SIP in the IP Multimedia Subsystem (IMS)
- Third Generation Partnership Project 2 (3GPP2) uses SIP in the IP Multimedia Subsystem (IMS)
- Microsoft uses SIP in Real-Time Communication in Windows XP
  - Note that earlier versions of MS Messenger had a non-standard pre-implementation of SIP for Presence and Instant Messaging
- PacketCable in Distributed Call Signalling
- The International Softswitch Consortium
- ETSI TIPHON program
- Wireless Village (now part of the Open Mobile Alliance) is looking at SIP for Instant Messaging and Presence
- America On Line (AOL) will support SIP for Instant Messaging and Presence
- The COMICS group uses SIP inside Meetecho (www.meetecho.com)!

Session Description Protocol (SDP)

- SDP is a session description protocol for multimedia sessions
- SDP is used to describe the set of media streams, codecs, and other media related parameters supported by either party.
- All SIP implementations MUST support SDP, although they can support other bodies
- Used by other protocols than SIP: RTSP, SAP, etc.
- SDP was initially developed to support multicast sessions in the Internet. Gradually tailored for SIP purposes.

SDP example

Session level	Version --->	v=0
	Origin --->	o=mgarcia 2987933615 2987933615 IN IP4 10.0.0.8
	Session Name -->	s=training course
	URI -->	u=http://standards.ericsson.net/sip
	E-mail address -->	e=miguel.a.garcia@ericsson.com
	Phone number -->	p=+358-9-299-3553
	Connection Data -->	c=IN IP4 10.0.0.8
	Times -->	t=907165275 0
Media level	Media -->	m=audio 3458 RTP/AVP 97 0
	Attributes -->	a=rtpmap:97 AMR
	Attributes -->	a=fmtp:97 mode-set=0,2,5,7; maxframes=2
	Media -->	m=video 3459 RTP/AVP 98
	Attributes -->	a=rtpmap:98 H263
	Media -->	m=application 32416 udp wb

IP address where media is received

Codecs

Type of media streams

Port number where media is received

References: SIP documents

- RFC 3261: Session Initiation Protocol
- RFC 3262: Reliable provisional responses in SIP
- RFC 3263: Locating SIP servers
- RFC 3265: SIP-specific event notification
- RFC 2617: HTTP Authentication: Basic and Digest Access Authentication
- RFC 2976: The SIP INFO method
- RFC 3050: Common Gateway Interface for SIP
- RFC 3310: HTTP Digest Authentication Using Authentication and Key Agreement (AKA)
- RFC 3311: The SIP UPDATE method
- RFC 3312: Integration of Resource Management and SIP
- RFC 3313: Media Authorization in SIP

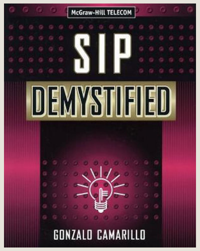
References: SIP documents (cont.)

- RFC 3323: SIP Privacy
- RFC 3325: The SIP P-Asserted-Identity header
- RFC 3326: The SIP Reason header
- RFC 3327: The SIP Path header
- RFC 3329: Security Agreement in SIP
- RFC 3428: The SIP MESSAGE method
- RFC 3455: P-Header Extensions to the SIP for 3GPP and SIP
- RFC 3485: The SIP and SDP Static Dictionary for Signaling Compression (SigComp)
- RFC 3486: Compressing SIP
- RFC 3515: The SIP REFER method

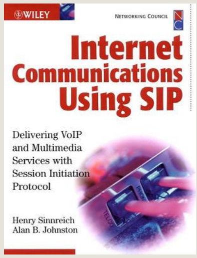
References: SDP documents

- RFC 2327: Session Description Protocol
- RFC 3108: ATM extensions for SDP
- RFC 3264: An offer/answer model with SDP
- RFC 3266: Support for IPv6 in SDP
- RFC 3388: Grouping of media lines in SDP

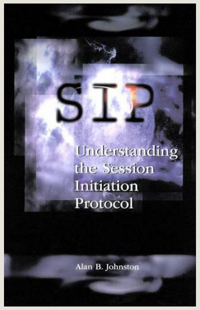
Books



- G. Camarillo (Ericsson employee): "SIP Demystified". McGraw-Hill Professional, 2002.



- H. Synnreich, A. B. Johnston: "Internet communications using SIP". John Wiley and Sons, 2001.



- A. B. Johnston: "SIP: Understanding the Session Initiation Protocol", Artech House Telecommunications Library, 2001

Asterisk

- http://www.asterisk.org
- È un PBX (*Private Branch eXchange*) open-source ibrido VoIP e TDM
- Può svolgere le funzioni di:
  – SIP Proxy Server
  – Gateway SIP – H.323
  – Gateway tra rete VoIP e PSTN/ISDN (mediante l'hardware opportuno...)
- Permette di realizzare a basso costo un centralino VoIP per la propria LAN casalinga/aziendale

Asterisk - Installazione

- È un applicativo sviluppato per ambienti Linux
- Il miglior modo per ottenere Asterisk è prelevarne una copia recente dal server CVS, nel seguente modo:

```
# cd /usr/src
# export CVSROOT=:pserver:anoncvs@cvs.digium.com:/usr/cvsroot
# cvs login          (la password è anoncvs)
# cvs checkout libpri asterisk
```

e di installarla con:

```
# cd ../libpri
# make clean ; make install
# cd ../asterisk
# make clean ; make install ; make samples
```

NB: con l'ultimo comando (`make samples`) si generano i file di configurazione di base nella directory `/etc/asterisk`, che andremo a modificare per quanto ci serve.

Asterisk - Il Dialplan

- È responsabile del *routing* delle chiamate dalla sorgente alla destinazione
- È immagazzinato nel file `extensions.conf`
- È formato da un insieme di *estensioni*, che hanno una struttura del genere:

```
exten => <extension_number>,<priority>,<application>[(arguments)]
```

Esempio:

```
[prova]
exten => 100,1,Wait(3)↰
exten => 100,2,Answer
exten => 100,3,Playback(demo.wav)↰
exten => 100,4,Hangup
```

Asterisk - Supporto SIP

- Per poter usare Asterisk come Proxy Server, è necessario:
  - Creare degli account personalizzati per i Client, abilitandoli ai servizi, nel file di configurazione `sip.conf`
  - Creare le opportune *estensioni* nel Dialplan
  - Configurare opportunamente gli User Agent

Asterisk - sip.conf

```
[general]
port=5060
bindaddr=0.0.0.0
context=voip
disallow=all
allow=ulaw

[mysjphone]
type=friend
host=dynamic
dtmfmode=rfc2833
context=voip
mailbox=1234

[mykphone]
type=friend
host=dynamic
dtmfmode=inband

[linuxsjphone]
type=friend
host=dynamic
dtmfmode=rfc2833
context=voip
```

Asterisk - extensions.conf

```
[voip]
exten => 1,1,Dial(SIP/mysjphone,30)
exten => 1,2,VoiceMail,u1234
exten => 1,102,VoiceMail,b1234
exten => 2,1,Dial(SIP/mykphone)
exten => 3,1,Dial(SIP/linuxsjphone,30)

exten => 12,1,Dial(SIP/mysjphone&SIP/mykphone)
exten => 13,1,Dial(SIP/mysjphone&SIP/linuxsjphone)
exten => 123,1,Dial(SIP/mysjphone&SIP/mykphone&SIP/linuxsjphone)

;Voice Mail
exten => 1001,1,Ringing
exten => 1001,2,Wait(2)
exten => 1001,3,VoicemailMain,s1234

;Nomi simbolici
exten => mysjphone,1,goto(1,1)
exten => mykphone,1,goto(2,1)
exten => linuxsjphone,1,goto(3,1)
exten => myvoicemail,1,goto(1001,1)
```

Asterisk – Servizi supplementari

- Voicemail
- Redirezione delle chiamate
- Trasferimento di chiamata
- Call parking
- Call queuing
- Call recording
- Filtri delle chiamate
- …

Link utili

- Sito ufficiale *Asterisk*: http://www.asterisk.org
- Sito ufficiale Digium: http://digium.com
- Sito ufficiale GnuGk: http://www.gnugk.org
- Sito ufficiale OpenH323 Project: www.openh323.org
- Sito ufficiale SJ Labs: http://sjlabs.com
- Tutorial "*Getting started with Asterisk*": http://www.automated.it/guidetoasterisk.htm
- Sito ufficiale The VOIP Wiki - *a reference guide to all things VOIP* – http://voip-info.org/
- *The Hitchhiker's Guide to Asterisk*: http://www.asteriskdocs.org/modules/tinycontent/content/docbook/current/docs-html/book1.html