Mobility and Session Management: The Session Initiation Protocol

dit UPM

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SIP – Session Initiation Protocol

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Introduction

- Signalling protocol for initiating, modifying, and terminating sessions with one or more participants
  - Including, but not limited to multimedia sessions
- IETF (Internet Engineering Task Force) standard
  - RFC 2543 (March 1999), obsolete
  - RFC 3261 (June 2002)
  - Drafts for extensions and new features
- Characteristics
  - Client-server protocol
  - Text-based messages (~ http, smtp)
  - Simplicity, extensibility
- Increasing availability of products and applications
  - Adopted for UMTS - IMS (IP Multimedia Subsystem)

SIP – Session Initiation Protocol

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SIP Functions

- Establishment and termination of sessions
- Negociation of session capabilities (codecs, UDP ports, ...)
  - SDP (Session Description Protocol)
- Location of users (mobility)
- User availability
- Supplementary services

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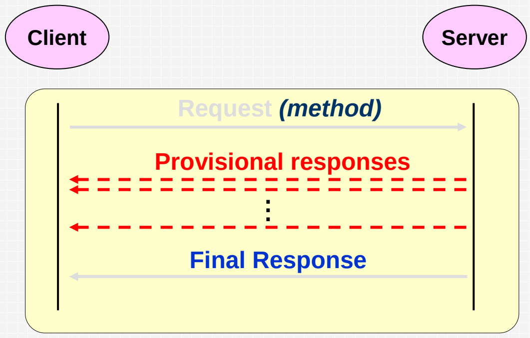
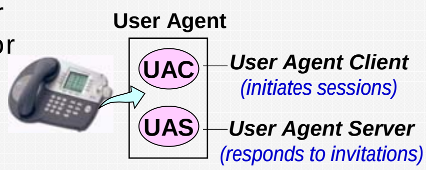
Operational Model

- Peer-to-peer communications based on client/server model
- Based on transactions
  - Transaction = request + [provisional answer(s)] + final answer

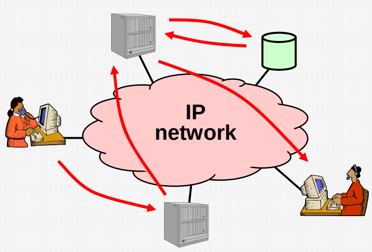


SIP Entities

- **User agents**
  - can act as client and server
  - can communicate directly or via intermediary network servers
  - e.g. PC client, SIP phone, VoIP gateway...



- **Network Servers**
  - Intermediary elements for assisting in session establishment
    - user location, mobility, and session routing
  - Different types of servers according to their funcionality



SIP Messages – Methods (requests)

Basic methods, defined in RFC 3261

INVITE	requests session establishment
ACK	signals end of establishment
BYE	terminates a session
CANCEL	cancels pending request
REGISTER	registers current user location
OPTIONS	asks for supported capabilities

Other methods, defined in extensions

INFO	transports mid-call signaling information
REFER	provides contact info (e.g. for call transfer)
PRACK	acknowledges receipt of provisional response
SUBSCRIBE	requests notification of an event
NOTIFY	indicates the ocurrence of an event
MESSAGE	carries information as MIME body

Note: Excepting ACK, all methods require a final response

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SIP Messages - Responses

- Organized in numerical ranges (as in other application protocols: ftp, smtp, http, ...)

	Code	Category	Examples
Provisional responses (do not end transaction)	1xx	Informational	*trying, ringing, forwarded, queued,...*
Final responses (end transaction)	2xx	Success	*ok*
	3xx	Redirection	*moved permanently, moved temporarily,...*
	4xx	Client error	*bad request, unauthorized, timeout, busy here, user not found,...*
	5xx	Server error	*not implemented, version not supported,...*
	6xx	Global error	*busy everywhere, user does not exist anywhere, session not acceptable,...*

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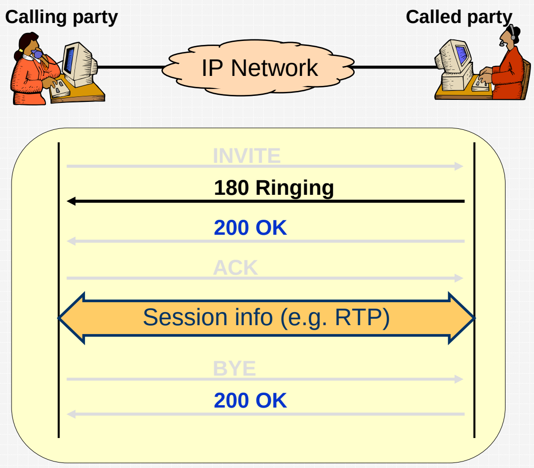
Example of SIP session



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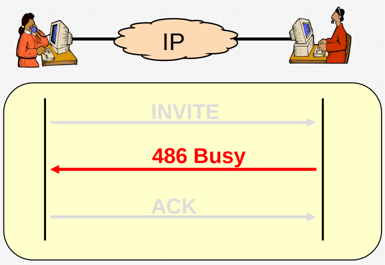
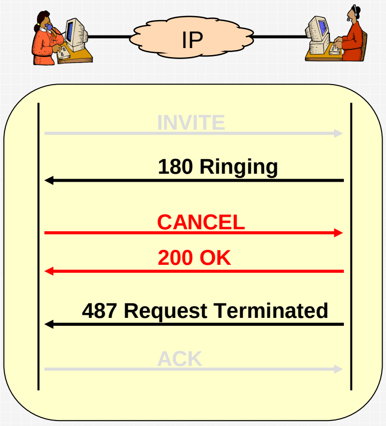
Other examples





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Transport of messages

- SIP can be used with different transport protocols
  - UDP, TCP, TLS, SCTP, ...
- UDP is usually preferred
  - It does not require openning a TCP connection
    - faster session establishment
  - SIP has its own reliability mechanisms
    - timers, retransmissions, ...
- Default port: 5060

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Format of messages

SIP URI (Uniform Resource Identifier)
several formats

Initial line	`INVITE sip:blas@company.com SIP/2.0`

SIP parameters	`Via: SIP/2.0/UDP 138.4.3.130:3456`
	`CallID: a2e3a@138.4.3.130`
	`From: sip:epi@dit.upm.es`
	`To: sip:blas@company.com`
	`Cseq: 1 INVITE`
	`Content-type: application/SDP`
	`Content-Length: 98`

Blank line	

Message Body
(SDP message for multimedia sessions)
	`v=0`
	`c=IN IP4 138.4.3.130`
	`m=audio 49170 RTP/AVP 0`
	`a=rtpmap:0 PCMU/8000`

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SIP addressing

- SIP URIs (Uniform Resource Identifiers)
  - Format similar to e-mail addresses
- Examples:
  - sip:blas@company.com
  - sip:blas@lab.company.com
  - sip:blas@proxy.company.com:6878
  - sip:blas@138.4.3.130
  - sip:+34-91-555-1234@optel.com; user=phone

Session Description Protocol (SDP)

- Sintaxis for describing characteristics of multimedia sessions (RFC 2327, April 1998)
- SIP uses a reduced subset of SDP sintaxis
- Example:

```
v=0
c=IN IP4 138.4.3.130
m=audio 4006 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

SDP version 0

Internet connection, IPv4, address 138.4.3.130

Media information: audio, port 4006, RTP, perfil 0

Attributes of perfil 0: codec PCM, mu law, sampling at 8KHz

SIP Servers

- Intermediary network elements for assisting in session establishment
  - Key for allowing mobility and location of SIP users
  - Process SIP requests for a given Internet domain
    - Registered in Domain Name System (DNS)

Server	Request	Function
REGISTRAR	REGISTER	Stores user´s location information on a data base (location server)
REDIRECT SERVER	INVITE	Responds to invitation requests with SIP addresses stored in location server
PROXY SERVER	INVITE	Forwards invitation requests to SIP addresses stored in location server

- Other types of servers: application servers, B2BUA (Back to back user agents)

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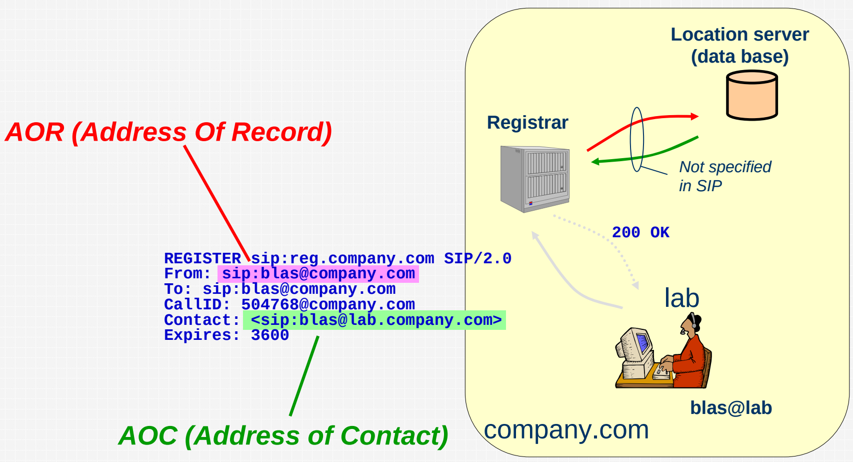
Registrar



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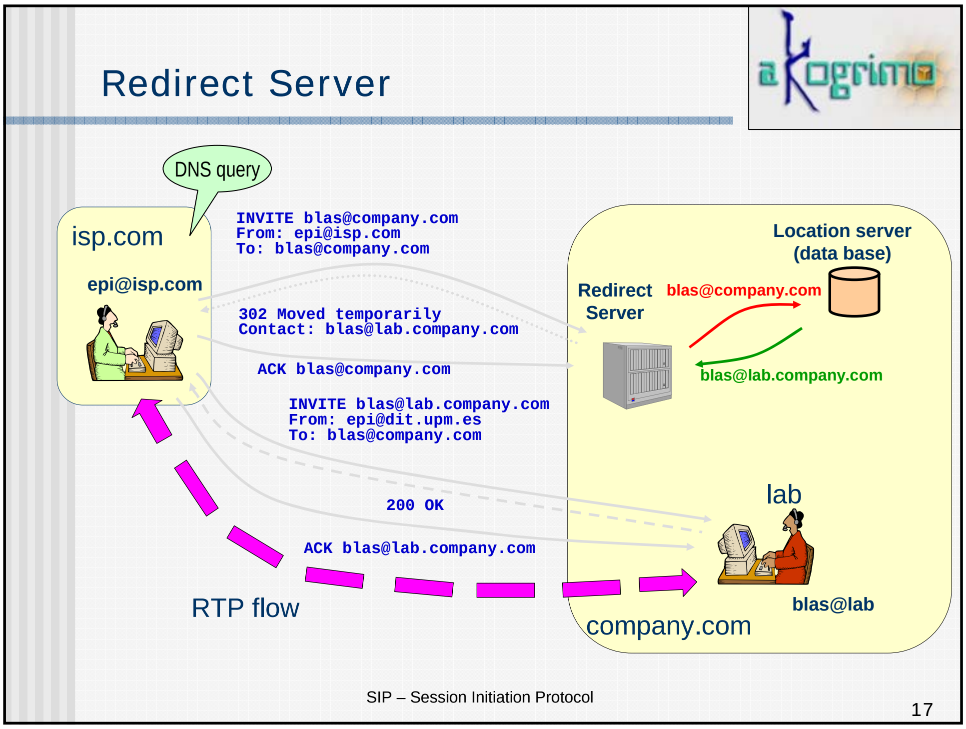
Redirect Server
DNS query
isp.com
epi@isp.com
INVITE blas@company.com
From: epi@isp.com
To: blas@company.com
302 Moved temporarily
Contact: blas@lab.company.com
ACK blas@company.com
INVITE blas@lab.company.com
From: epi@dit.upm.es
To: blas@company.com
200 OK
ACK blas@lab.company.com
Location server
(data base)
Redirect
Server
blas@company.com
blas@lab.company.com
lab
blas@lab
company.com
RTP flow
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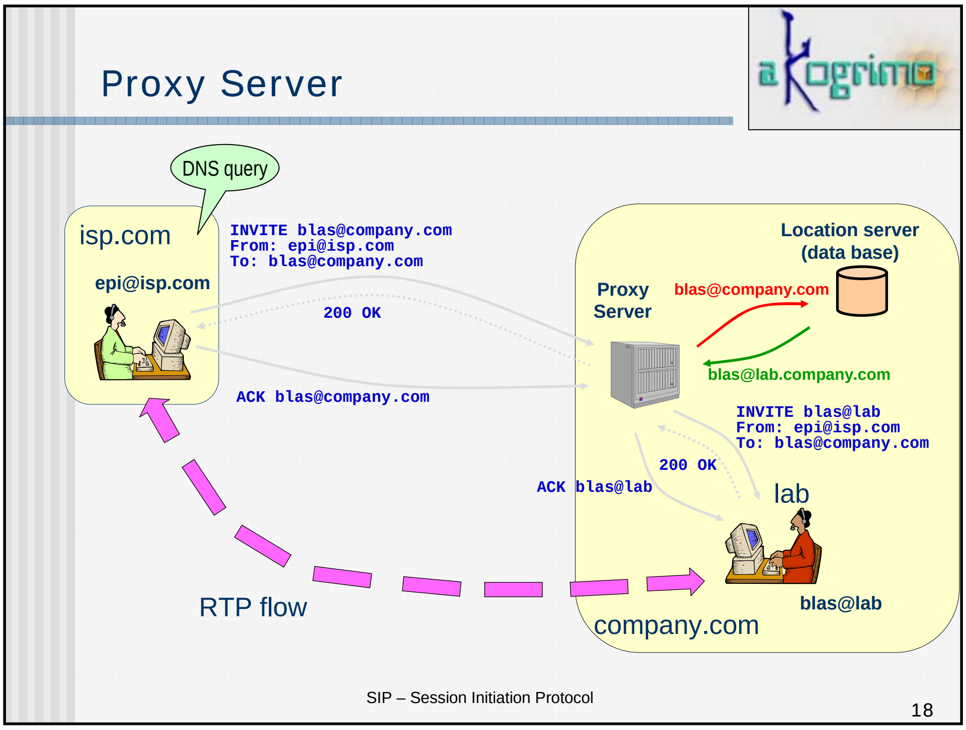
Proxy Server
DNS query
isp.com
epi@isp.com
INVITE blas@company.com
From: epi@isp.com
To: blas@company.com
200 OK
ACK blas@company.com
Location server
(data base)
Proxy
Server
blas@company.com
blas@lab.company.com
INVITE blas@lab
From: epi@isp.com
To: blas@company.com
200 OK
ACK blas@lab
lab
blas@lab
company.com
RTP flow
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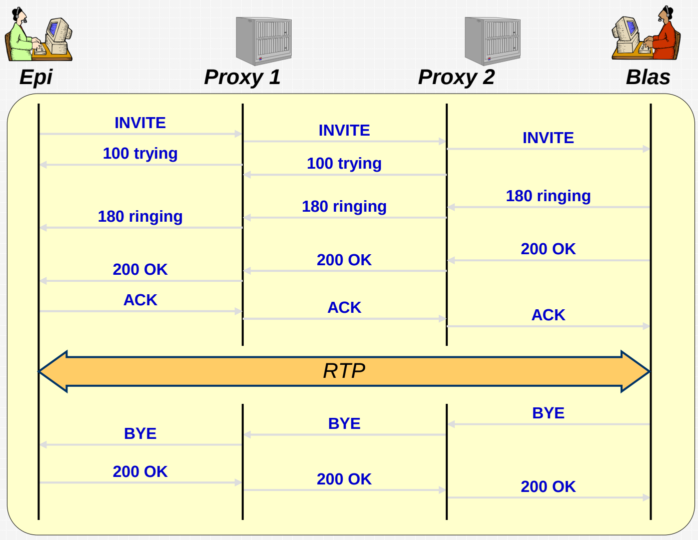
Session through two proxies



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Extensions to SIP

- Supplementary services:
  - Call transfer, caller ID and privacy, 3rd party call control, caller preferences, ...
- Non multimedia applications:
  - Presence indication, instant messaging, event notification, ...
- Protocol improvements
  - Reliability for provisional reponses, privacy and security, compression, ...
- Others
  - Interworking with other signalling protocols: ISUP, Q.931 (ISDN), ...
  - SNMP management (MIB)
  - Firewall and NAT traversing issues
  - ...

Large collection of RFCs and Internet drafts

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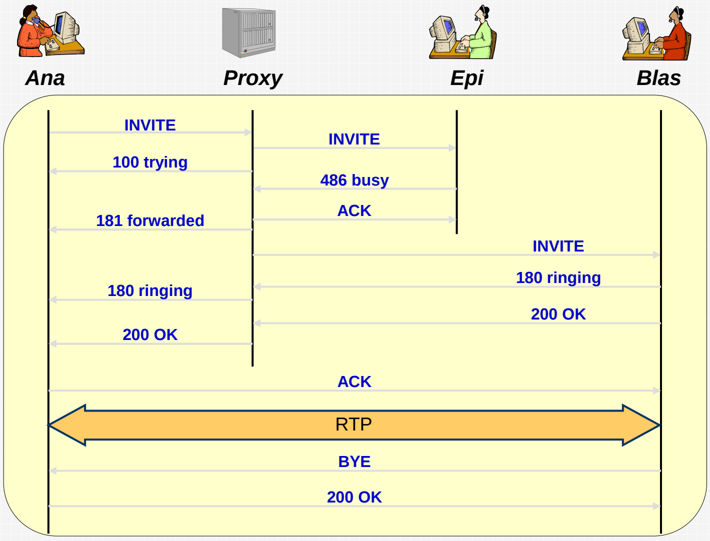
Call forwarding



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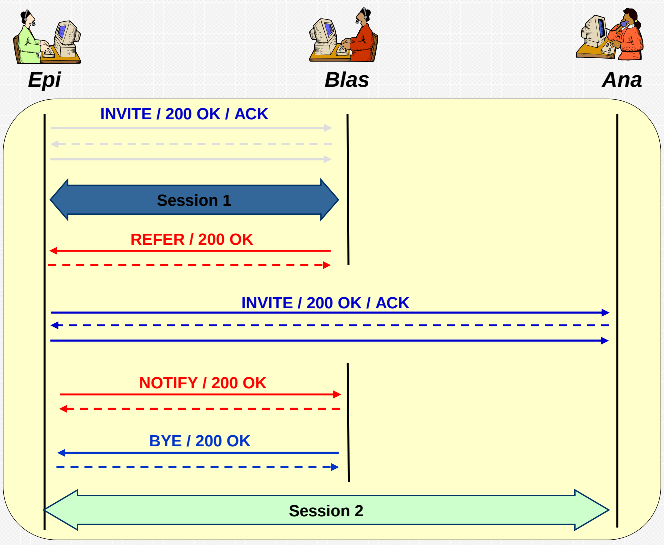
Call transfer



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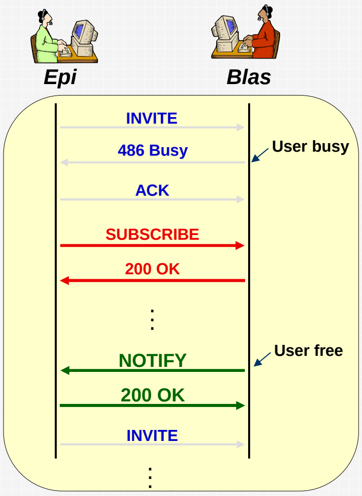
Event notification

- Asynchronous notification of events
  - Supplementary services
    - Call completion on busy subscriber
    - Message waiting indication
  - Others:
    - presence notification
    - stock exchange alarms
    - ...
- New SIP methods:
  - SUBSCRIBE
  - NOTIFY



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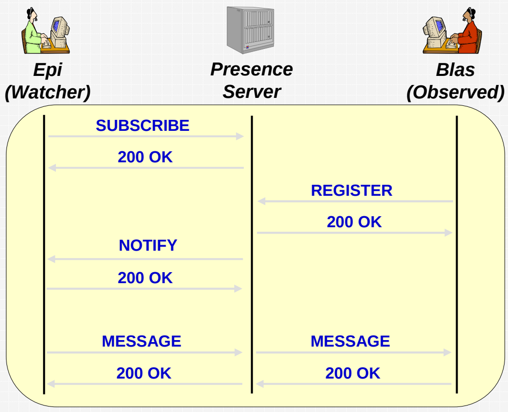
Instant Messaging and Presence

- IETF SIMPLE Working Group (SIP for Instant Messaging and Presence Leveraging Extensions)



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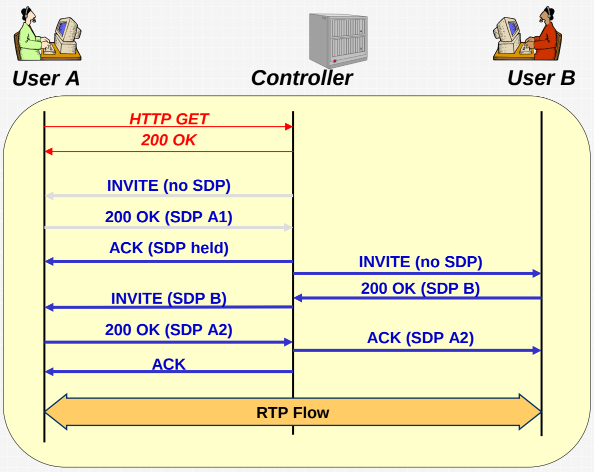
3rd Party Call Control (3pcc)

- Example: “click to dial”



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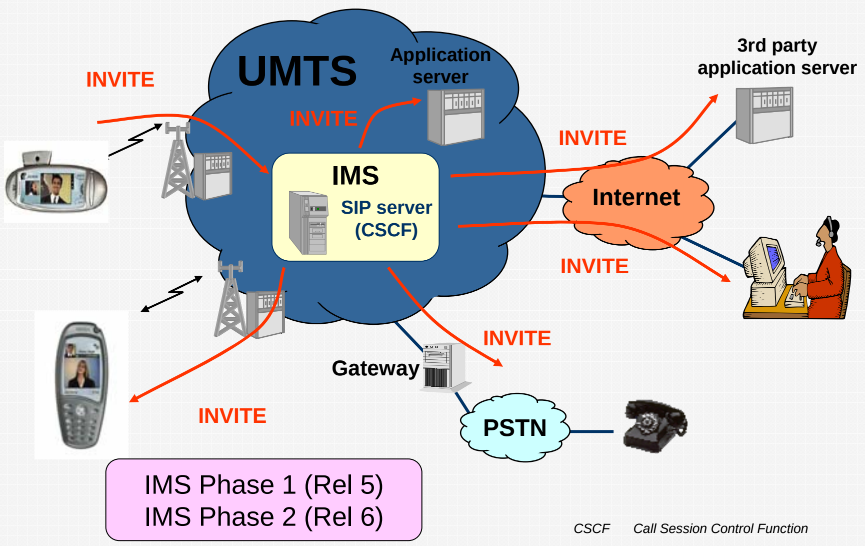
SIP in UMTS
The IP Multimedia Subsysem (IMS)



CSCF Call Session Control Function

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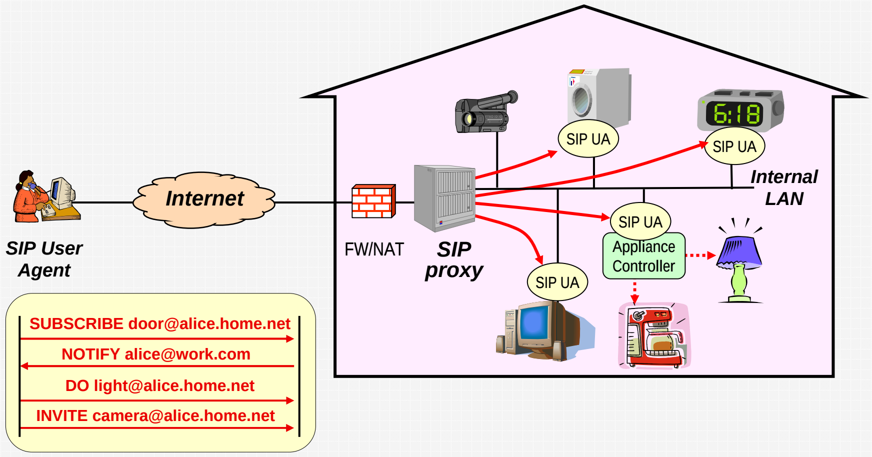
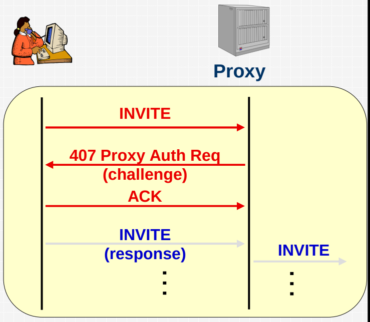
SIP for appliances



Framework Draft for Networked Appliances Using the Session Initiation Protocol
(draft-moyer-sip-appliances-framework-00.txt, Jul 2000)

Security in SIP

- Security mechanisms:
  - Authentication: challenge/response based on shared secret
  - Encryption using private/public keys
    - S/MIME, IPsec, TLS

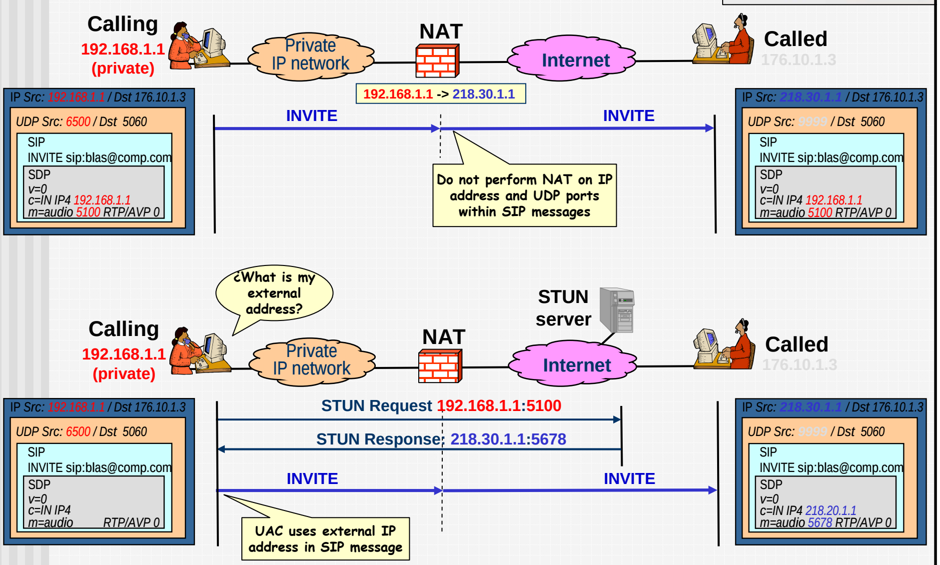


- Other issues
  - Security of user data (e.g. ciphering of voice packets)
  - Blocking of packets by firewalls

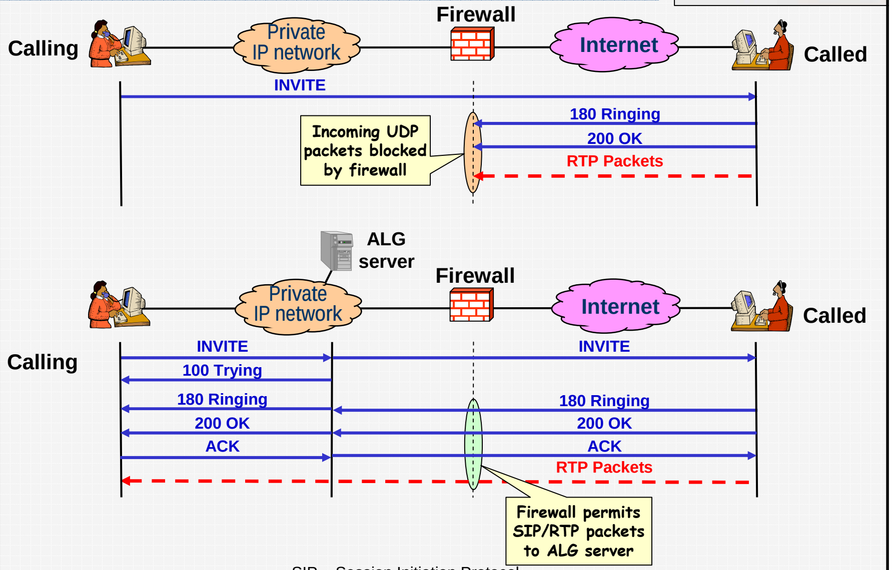
Firewall and NAT traversal problems

- Firewalls block incoming UDP packets
  - TCP can solve the problem for SIP signalling
  - But RTP packets are usually transported over UDP
- NAT (Network Address Translators) servers do not map IP address or TCP/UDP ports in application layer protocols
- Problem common to many P2P applications
- Several solutions in discussion
  - ALG (Application Level Gateway)
  - STUN (Simple Traversal of UDP Through NATs)
    - IETF MIDCOM Working Group
  - Universal Plug and Pray (UPnP)
  - ...

Simple Traversal of UDP Through NATs (STUN)



Application Layer Gateway (ALG)



Products and applications

- SIP phones
- PC Clients
- SIP Servers
- VoIP Gateways
- Software (libraries, development environments, ...)
- Test equipment
- ...

- Voice
- Video
- Net games
- Indication of presence
- Instant messaging
- ...

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- H. Sinnreich, A. Johnston, "Internet Communications Using SIP", John Wiley & Sons, 2001
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- *J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, E. Schooler, "SIP: Session Initiation Protocol", RFC 3261, June 2002*

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References on WWW

- IETF:
  - http://www.ietf.org/html.charters/sip-charter.html
  - http://www.ietf.org/html.charters/simple-charter.html
  - http://www.ietf.org/html.charters/sipping-charter.html
  - http://www.ietf.org/html.charters/iptel-charter.html
- Others:
  - http://www.sipforum.org/
  - http://www.cs.columbia.edu/sip/
  - http://www.sipcenter.com/
  - http://www.pulver.com/

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Glossary

3PCC	Third-party Call Control
AoC	Address of Contact
AoR	Address of Record
CGI	Common Gateway Protocol
CPL	Common Progrramming Language
DNS	Domain Name System
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
IP	Internet Protocol
ISUP	ISDN User Part
ITU	International Telecommunication Union
MIB	Management Information Base
NAT	Network Address Translation
P2P	Peer to peer
RFC	Request For Comments
RGW	Residential Gateway
RTP	Real Time Protocol
RTCP	Real Time Control Protocol
SCTP	Stream Control Transmission Protocol
SDP	Session Description Protocol
SNMP	Simple Network Management Protocol
SIP	Session Initiation Protocol
TCP	Transmission Control Protocol
TLS	Transport Layer Security
UAC	User Agent Client
UAS	User Agent Server
UDP	User Datagram Protocol
UMTS	Universal Mobile Telecommunications System
URL	Uniform Resource Locator
URI	Uniform Resource Idenifier
VoIP	Voice over IP
XML	Extensible Markup Language