universidade de aveiro

Session Initiation Protocol (SIP)

universidade de aveiro

SIP

- SIP = base protocol to establish sessions in the internet (peer-to-peer), low complexity and generic
  - Developed by IETF mmusic group, since 1995
  - Peer-to-peer signalling protocol (RFC 2543 in 1999, RFC 3261)
- Transports session description information of initiator (caller) to destination (callee)
  - Client-server approach (origination/answer)
  - Independent of protocol (UDP, TCP, AAL5, ...)
    - Supports multicast
    - But generally works through UDP…
    - Security at the transport and network layer provided with TLS (requires TCP) or IPSec
- Supports change of parameters in the middle of the session
- Signaling messages not frequent
  - Always with acknowledges
- Objective:
  - Allow maximum re-utilization of existent protocols
  - Use HTTP-alike coding (text-based)
  - Reuse already existent addresses (URLs, DNS, proxies…)
  - Be an alternative to H.323
  - Supporting new services
  - Being scalable, extensible

universidade de aveiro

SIP allows...

- Create, modify and terminate multimedia sessions with two or more participants
  - VoIP, distribution of multimedia data and multimedia conference
- Provides functionalities that can be used to implement the following services
  - Users location
  - Users availability
  - Determination of users capabilities
  - Negotiation (and re-negotiation) of the parameters of users participating in a session
  - Negotiation of session characteristics
    - Session Description Protocol
  - Users mobility
  - Security mechanisms
    - Prevention of denial of service attacks
    - Users authentication
    - Message integrity and confidentiality
- It does not distribute multimedia data
  - Part of IETF architecture of conference control (+SAP, + RTSP, + SDP, ...)
- It is not able to control media gateways

130

universidade de aveiro

SIP functionalities

- SIP supports five communication aspects:
  - User location - (given an e-mail type address) determination of the end system to be used for communication
    - Distributed directory lookups
  - User capabilities - determination of the media and media parameters to be used
  - User availability - determination of the willingness of the called party to engage in communications
  - Call (session) establishment - "ringing", establishment of session parameters at both called and calling party;
    - Including multi-party, using an MCU, or a fully-meshed strategy
  - Call (session) control - including transfer and termination of sessions, modifying session parameters, and invoking services
    - (Re)-negotiation of call parameters
    - Forwarding: manual and automatic
    - Personal mobility: different terminals with the same identifier
    - Call center: reach the first (load distribution) or all (conference)
    - Initiates, modifies and terminates sessions (conferences)
      - Including between gateways to the PSTN
- SIP has been heavily explored in current network concepts (IMS)

131

SIP clients and servers

User Agent Server, UAS
User Agent Client, UAC

UserAgent

Servers

Gateways
Registrar
Redirect
Proxy

- UAC: user-agent client (application that starts the call)
- UAS: user-agent server that accepts, redirects or rejects calls
- redirect server: redirect requests
- proxy server: server + client; controls the call, gets the address of the proxy callee, can also redirect
- registrar: registers the location of the user
- user agent = UAC + UAS
  - Usually combine a registrar + (proxy or redirect server)

SIP Clients and Servers

- SIP Clients
  - Phones (software based or hardware).
  - Gateways
  - User Agents
  - A User Agent acts as a
    - Client when it initiates a request (UAC),
    - Server when it responds to a request (UAS).
- SIP Servers
  - Proxy server
    - Receives SIP requests from a client and forwards them on the client's behalf.
    - Receives SIP messages and forward them to the next SIP server in the network.
    - Provides functions such as authentication, authorization, network access control, routing, reliable request retransmission, and security.
  - Redirect server
    - Provides the client with information about the next hop or hops that a message should take and then the client contacts the next-hop server or UAS directly.
  - Registrar server
    - Processes requests from UACs for registration of their current location.
    - Registrar servers are often co-located with a redirect or proxy server.

universidade de aveiro

Proxy servers

- Intermediate entities that behave as servers and clients
  - Make requests in name of other clients
- Get location of other endpoints
- Route SIP messages
- Optional
  - Authentication and accounting

134

universidade de aveiro

Registration and redirect servers

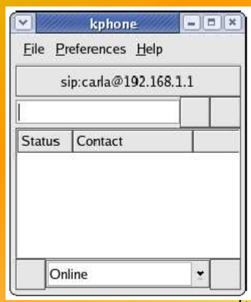
- Registration
  - Entities where users register their UAs
  - Allow the mapping between users addresses and their UAs addresses
    - Database of location service
    - Request to database by proxy and redirect servers
      - Routing and redirect of messages
- Redirect
  - Returns alternative locations of UAs and servers
    - Receives requests
    - Requests the location service
    - Returns a list of alternative addresses to where the request should be redirected

135

universidade
de aveiro

User Agent

- Endpoint of sessions
  - Initiates and terminates sessions
  - User Agent Server (UAS) and User Agent Client (UAC)
- UAC
  - Creates the requests (e.g. to initiate a session)
- UAS
  - Generates answers to requests (e.g. to answer a session request)
- Hardware or software equipment that implements UA functions



136

universidade
de aveiro

SIP addresses

- URI (*Uniform Resource Identifier*)
  - Translated, by proxy server, to the UA address used by the user
  - A same user can have and use different UAs
  - sip:user@host:port;uri-parameters?headers
    - *uri-parameters* are parameters that affect the request for the resource identified by SIP URI
    - *headers* are fields to be included in the request
- sip:275313364@telecom.pt;user=phone
  - Identifies a user or a resource through the phone number 275313364 in the *telecom.pt* domain
  - To enforce that it is a phone number, the parameter *user* with the value *phone* is used

137

universidade de aveiro

SIP Messages

- SIP used for Peer-to-Peer Communication though it uses a Client-Server model.
- SIP is a text-based protocol and uses the UTF-8 charset.
- A SIP message is either a **request** from a client to a server, or a **response** from a server to a client.
  - A request message consists of a Request-Line, one or more header fields, an empty line indicating the end of the header fields, and an optional message-body;
  - A response message consists of a Status-Line, one or more header fields, an empty line indicating the end of the header fields, and an optional message-body.
  - All lines (including empty ones) must be terminated by a carriage-return line-feed sequence (CRLF).

universidade de aveiro

SIP Methods and Purposes

- INVITE
  - Requests the establishment of a session.
- ACK
  - Completes a three way session handshake (INVITE request, responses, ACK).
- OPTIONS
  - Requests the capabilities of another User Agent
  - Response lists supported methods, extensions, codecs, etc.
- BYE
  - Terminates an established session
    - User Agents stop sending media packets.
- CANCEL
  - Terminates a pending session (INVITE sent but no final response yet received).
- REGISTER
  - Allows a User Agent to upload current location.

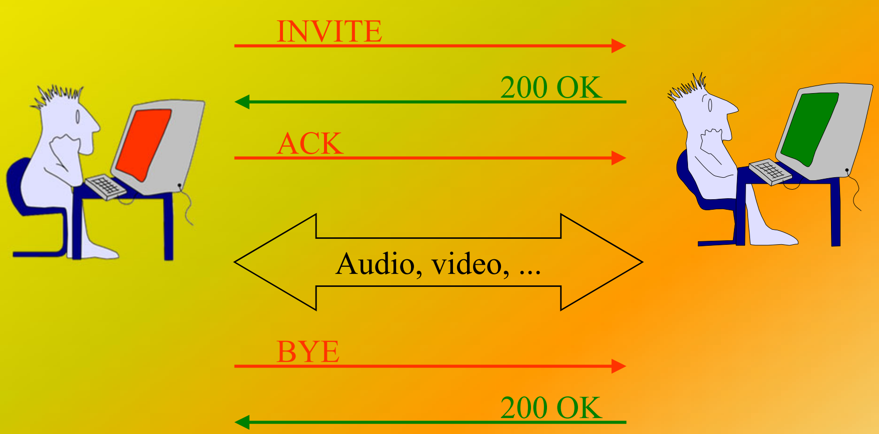
SIP Responses Codes and Purposes

- The first digit of the Status-Code defines the class of response.
  - 1xx: Provisional - request received, continuing to process the request;
  - 2xx: Success - the action was successfully received, understood, and accepted;
  - 3xx: Redirection - further action needs to be taken in order to complete the request;
  - 4xx: Client Error - the request contains bad syntax or cannot be fulfilled at this server;
  - 5xx: Server Error - the server failed to fulfill an apparently valid request;
  - 6xx: Global Failure - the request cannot be fulfilled at any server.

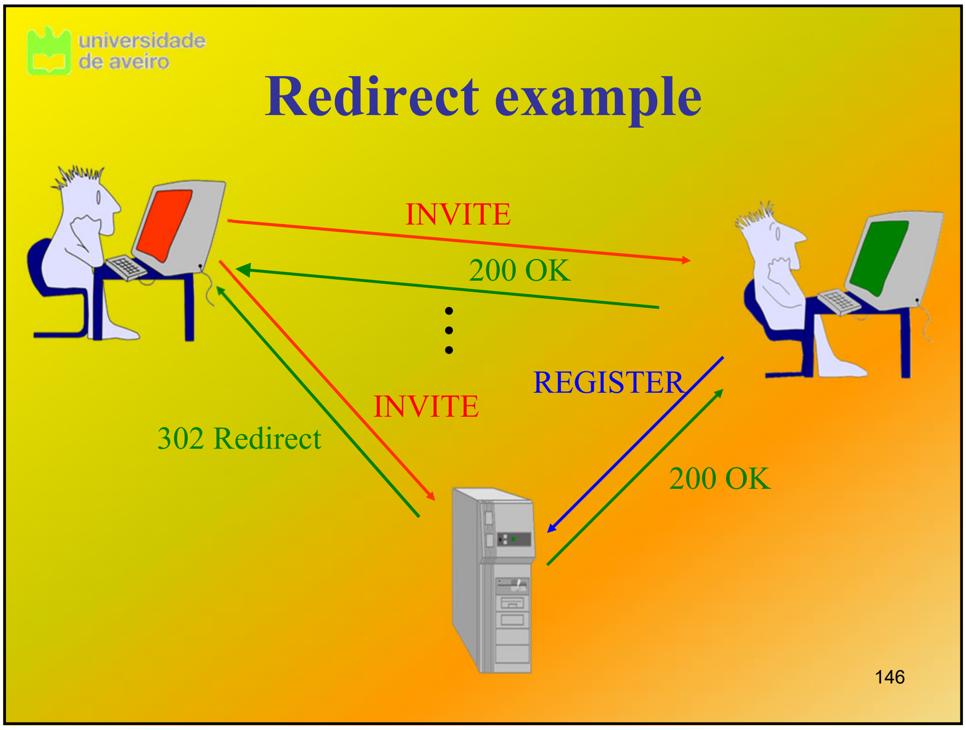
- Common Response codes:
  - 100 Trying
    - The request has been received and that some unspecified action is being taken.
  - 180 Ringing
    - Trying to alert the user.
  - 200 OK
  - 301 Moved Permanently and 302 Moved Temporarily
    - User can no longer be found at the address in the Request-URI.
  - 400 Bad Request
    - Request could not be understood.
  - 401 Unauthorized
    - Request requires user authentication.
  - 403 Forbidden
    - Server understood the request, but is refusing to fulfill it.
  - 404 Not Found
    - Server has definitive information that the user does not exist.

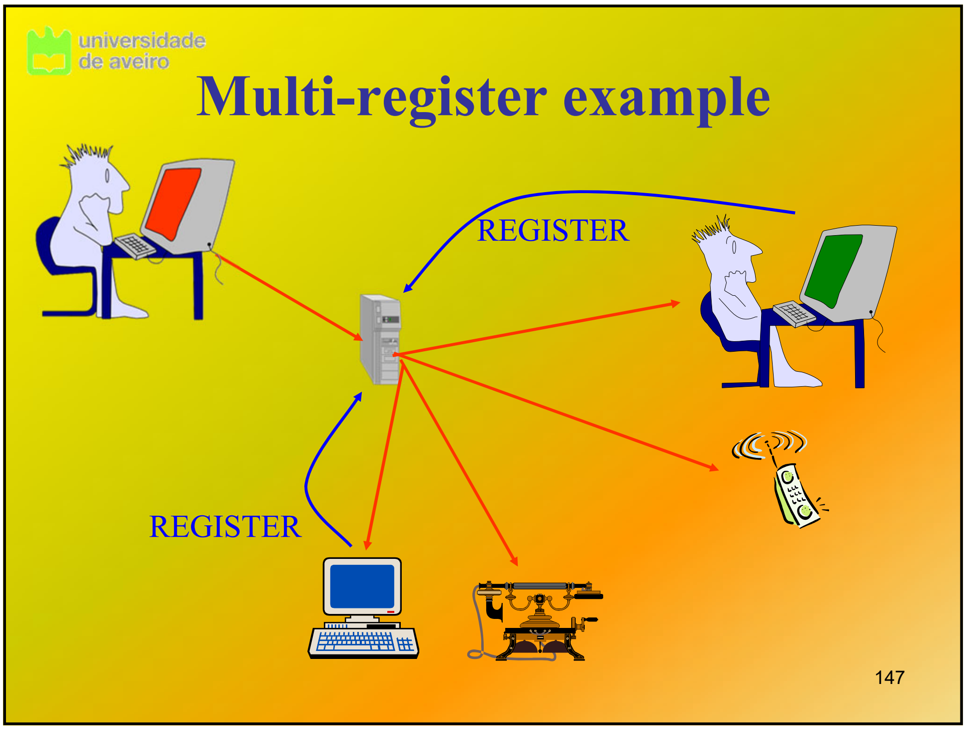
140

Signalling – basic example



145

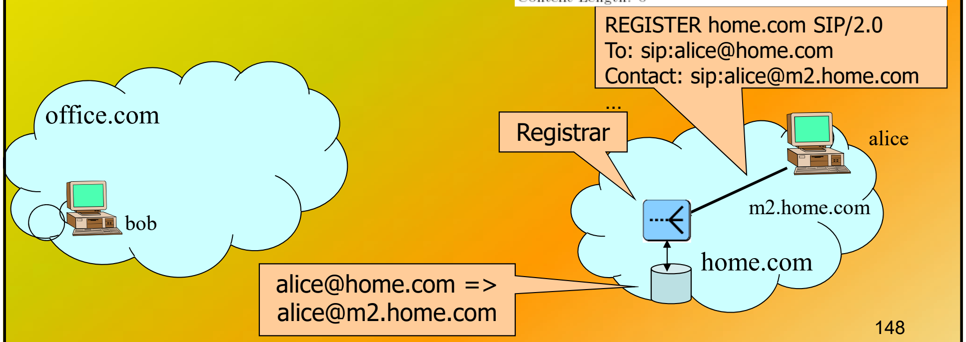

universidade
de aveiro
Redirect example
INVITE
200 OK
REGISTER
INVITE
302 Redirect
200 OK
146


universidade
de aveiro
Multi-register example
REGISTER
REGISTER
147

Setup of a basic call - registration

- Identifier of email type: <sip:alice@home.com>
- Alice phone registers in home.com

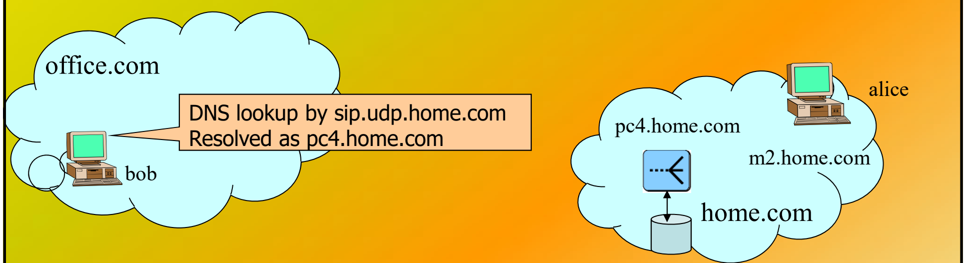
```
REGISTER sip:A.com SIP/2.0
Via: SIP/2.0/UDP pc.A.com:5060;branch=z9hG4bKnashds7
To: Carla <sip:carla@A.com>
From: Carla <sip:carla@A.com>;tag=456248
Call-ID: 843817637684230@998sdasdh09
CSeq: 1826 REGISTER
Contact: <sip:carla@pc.A.com>
Expires: 900
Content-Length: 0
```



148

Setup of a basic call - DNS lookup

- Identifier of email type: <sip:alice@home.com>
- Alice phone registers in home.com
- Bob calls alice@home.com; phone makes DNS lookup

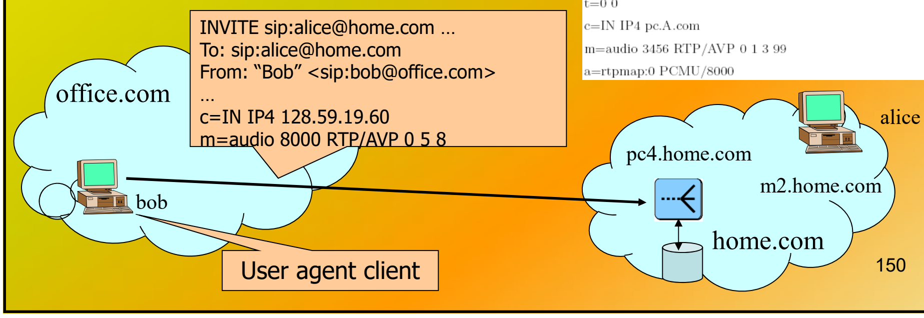


149

Setup of a basic call - INVITE

- Identifier of email type: <sip:alice@home.com
- Alice phone registers in home.com
- Bob calls alice@home.com; phone makes DNS lookup
- **Phone makes an INVITE; it acts as a UAC**

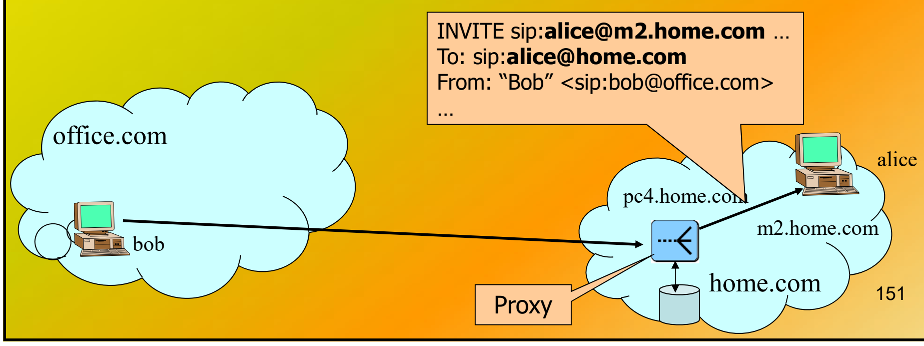
```
INVITE sip:romeu@B.com SIP/2.0
Via: SIP/2.0/UDP pc.A.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Romeu <sip:romeu@B.com>
From: Carla <sip:carla@A.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Content-Type: application/sdp
Content-Length: 142
==========corpo da mensagem==============
v=0
o=carla 53655765 2353687637 IN IP4 pc.A.com
s=Session SDP
t=0 0
c=IN IP4 pc.A.com
m=audio 3456 RTP/AVP 0 1 3 99
a=rtpmap:0 PCMU/8000
```



150

Setup of a basic call - INVITE

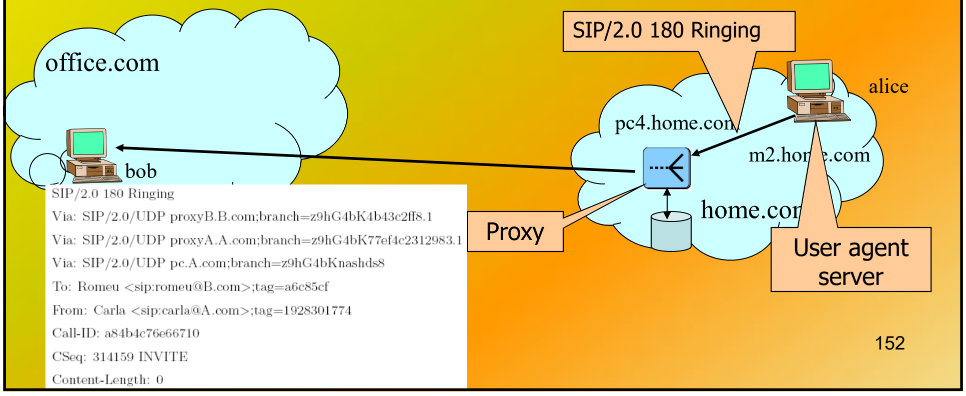
- Identifier of email type: <sip:alice@home.com>
- Alice phone registers in home.com
- Bob calls alice@home.com; phone makes DNS lookup
- Phone makes an INVITE; it acts as a UAC
- **Server is able to make the proxy of the call to the current location**



151

Setup of a basic call - Ringing

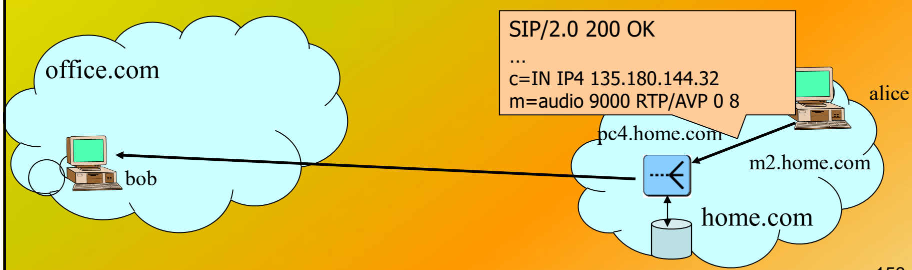
- Identifier of email type: <sip:alice@home.com>
- Alice phone registers in home.com
- Bob calls alice@home.com; phone makes DNS lookup
- Phone makes an INVITE; it acts as a UAC
- Server is able to make the proxy of the call to the current location
- **Alice phone rings; it acts as a UAS**



152

Setup of a basic call - 200 OK and ACK

- Phone makes an INVITE; it acts as a UAC
- Server is able to make the proxy of the call to the current location
- Alice phone rings; it acts as a UAS
- **When Alice hangs up the phone, the call is accepted, a 200 OK message is sent to Bob with the agreed configuration parameters, Bob phone sends a ACK to complete the setup.**



153

universidade de aveiro

Setup of a basic call - 200 OK and ACK

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP proxyB.B.com;branch=z9hG4bK4b43c2ff8.1
Via: SIP/2.0/UDP proxyA.A.com;branch=z9hG4bK77ef4c2312983.1
Via: SIP/2.0/UDP pc.A.com;branch=z9hG4bKnashds8
To: Romeu <sip:romeu@B.com>;tag=a6c85cf
From: Carla <sip:carla@A.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:romeu@pc.B.com>
Content-Type: application/sdp
Content-Length: 141
==========corpo da mensagem===============
v=0
o=carla 53655765 2353687637 IN IP4 pc.B.com
s=Session SDP
t=0 0
c=IN IP4 pc.B.com
m=audio 3456 RTP/AVP 0 1 3
a=rtpmap:0 PCMU/8000
```

```
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc.A.com;branch=z9hG4bKnashds9
Max-Forwards: 70
To: Romeu <sip:romeu@B.com>;tag=a6c85cf
From: Carla <sip:carla@A.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0
```

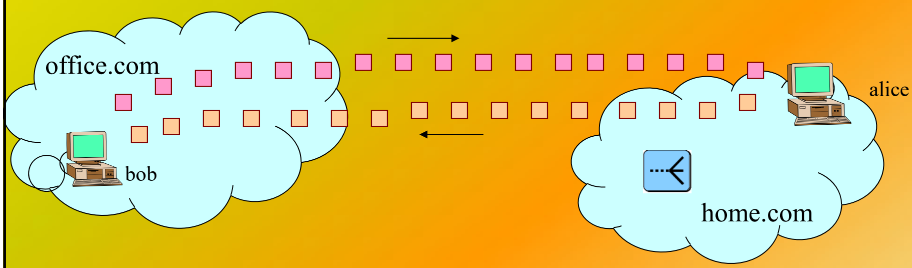
- ACK can be sent directly between UAs, since both addresses are now known

154

universidade de aveiro

Setup of a basic call - data

- Audio packets are transferred through RTP

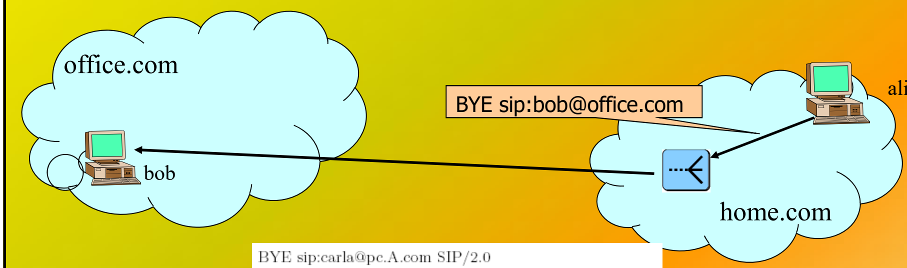


155

universidade de aveiro

Setup of a basic call - BYE

- **When one of the endpoints hangs off, a BYE is sent**

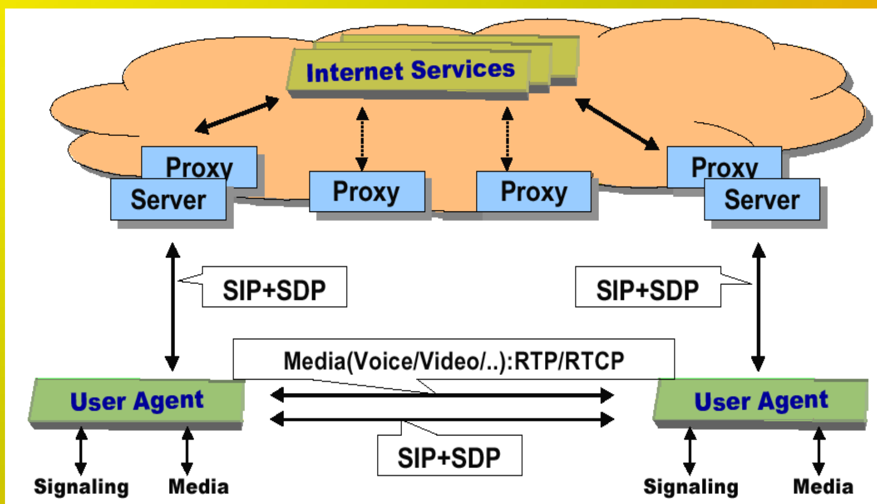



```
BYE sip:carla@pc.A.com SIP/2.0
Via: SIP/2.0/UDP pc.A.com;branch=z9hG4bKnashds10
Max-Forwards: 70
From: Carla <sip:carla@A.com>;tag=a6c85cf
To: Romeu <sip:romeu@B.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 231 BYE
Content-Length: 0
```

156

universidade de aveiro

End-to-End SIP model



157

SIP messages

Method	Propósito
REGISTER	Registar um UA no serviço de localização.
INVITE	Estabelecer ou alterar os parâmetros de uma sessão.
ACK	Confirmar a recepção da resposta a um pedido de sessão.
CANCEL	Terminar um pedido de sessão pendente.
BYE	Terminar uma sessão.
OPTIONS	Interrogar uma entidade acerca das suas capacidades.

Some fields

- *To* – address of the destination entity
- *From* – address of the entity that sends the message
- *Call-ID* - identifies, together with the parameters *tag* of fields *To* and *From*, each session SIP and all registration requests of a UA
- *Via* – contains information about a path followed by the request from its origin, that should be used to route the answer
- *Proxy-Authenticate* – contains a challenge sent by a proxy server to be used in the authentication
- *Proxy-Authorization* – contains the answer to the challenge sent by a proxy server
- *Route* – used to indicate the route of a request through a set of proxy servers

158

SIP answers

- SIP answers are of HTTP type

SIP-Version SP Status-Code SP Reason-Phrase CRLF (SP=Space, CRLF=Carriage Return and Line Feed).

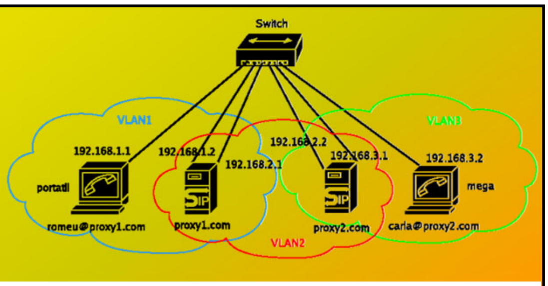
Example: SIP/2.0 404 Not Found

- First digit indicates the type of answer
  - 1xy – Informational
    Request received, being processed
  - 2xy – Success
    Request received and accepted
  - 3xy – Redirection
    More actions to be executed to conclude the request
  - 4xy – Client error
    Request contains a syntactic error that cannot be processed by this server
  - 5xy – Server error
    Server cannot serve an apparently valid request
  - 6xy – global failure
    Invalid request in any server

159

universidade de aveiro

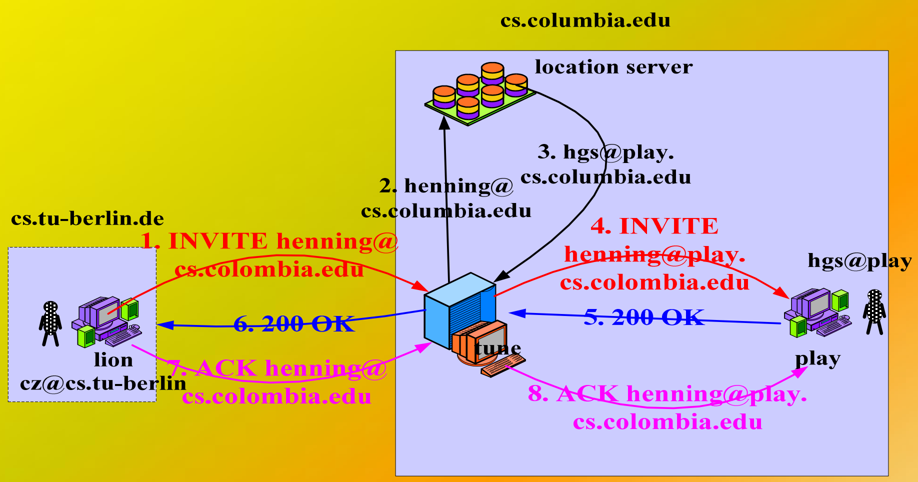
SIP Session

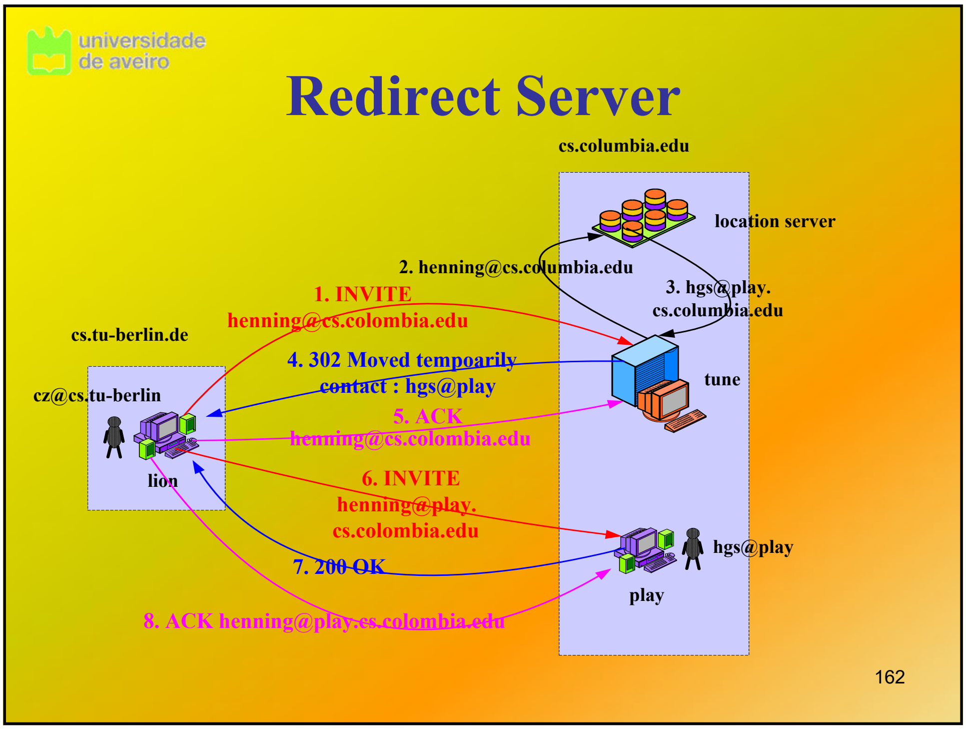


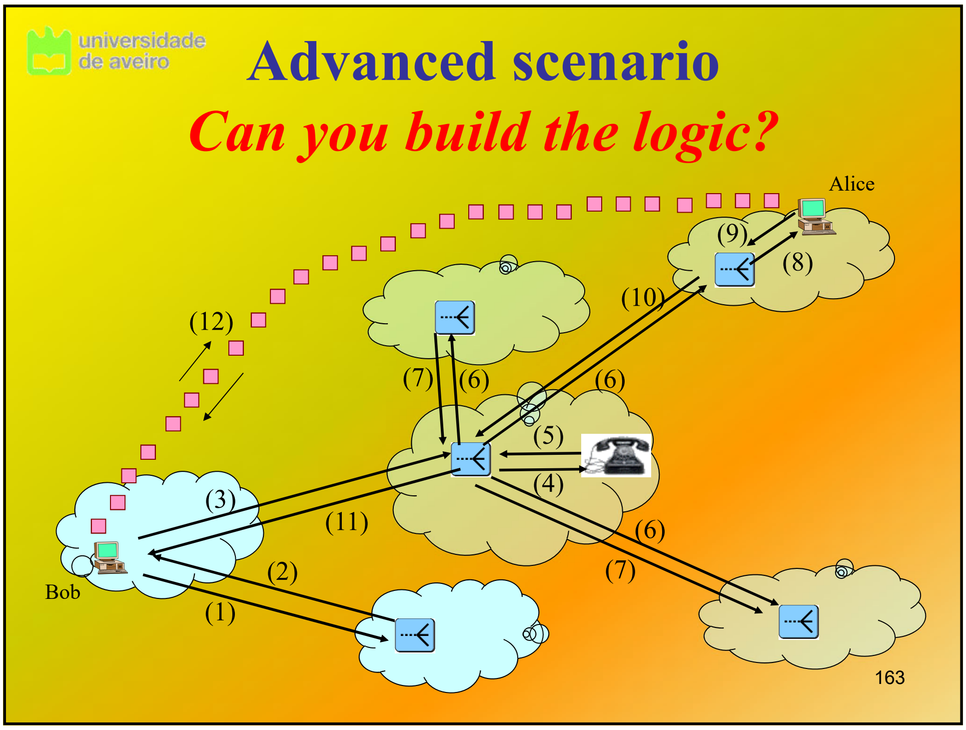
No.	Time	Source	Destination	Protocol	Info
1	0.000000	192.168.1.1	192.168.1.2	SIP	Request: REGISTER sip:proxy1.com
2	0.000530	192.168.1.2	192.168.1.1	SIP	Status: 401 Unauthorized (0 bindings)
3	5.121578	192.168.1.1	192.168.1.2	SIP	Request: REGISTER sip:proxy1.com
4	5.122829	192.168.1.2	192.168.1.1	SIP	Status: 200 OK (1 bindings)
5	21.65474	192.168.1.1	192.168.1.2	SIP/SDP	Request: INVITE sip:carla@proxy2.com, with session description
6	21.65531	192.168.1.2	192.168.1.1	SIP	Status: 100 trying -- your call is important to us
7	21.66426	192.168.1.2	192.168.1.1	SIP	Status: 180 Ringing
8	25.37596	192.168.1.2	192.168.1.1	SIP/SDP	Status: 200 OK, with session description
9	25.38055	192.168.1.1	192.168.1.2	SIP	Request: ACK sip:carla@192.168.3.2;transport=udp
10	25.50695	192.168.1.1	192.168.3.2	RTP	Payload type=ITU-T G.711 PCMU, SSRC=167772160, Seq=0, Time=20387
11	25.54920	192.168.1.1	192.168.3.2	RTP	Payload type=ITU-T G.711 PCMU, SSRC=167772160, Seq=1, Time=20387
12	25.54937	192.168.1.1	192.168.3.2	RTP	Payload type=ITU-T G.711 PCMU, SSRC=167772160, Seq=2, Time=20387
13	25.73593	192.168.3.2	192.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=167772160, Seq=0, Time=8174C
14	25.73595	192.168.3.2	192.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=167772160, Seq=1, Time=8174C
15	25.73601	192.168.3.2	192.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=167772160, Seq=2, Time=8174C
16	33.04344	192.168.1.2	192.168.1.1	SIP	Request: BYE sip:romeu@192.168.1.1;transport=udp
17	33.04890	192.168.1.1	192.168.1.2	SIP	Status: 200 OK

universidade de aveiro

Call Proxy scenario




universidade de aveiro
Redirect Server
cs.columbia.edu
location server
2. henning@cs.columbia.edu
3. hgs@play.
cs.columbia.edu
1. INVITE
henning@cs.colombia.edu
cs.tu-berlin.de
4. 302 Moved tempoarily
contact : hgs@play
tune
cz@cs.tu-berlin
5. ACK
henning@cs.colombia.edu
lion
6. INVITE
henning@play.
cs.colombia.edu
hgs@play
7. 200 OK
play
8. ACK henning@play.cs.colombia.edu
162


universidade de aveiro
Advanced scenario
Can you build the logic?
Alice
(9)
(8)
(10)
(12)
(7)
(6)
(6)
(5)
(4)
(3)
(11)
(6)
(7)
(2)
(1)
Bob
163

universidade de aveiro

Authentication and ciphering

- A proxy server or UA can oblige the initiator of the communication to provide authentication
  - Message #407 (proxy), or #401 (UA)
- It does not have recommended authorization systems
- RFC 3261 uses a mechanism of "digest", that provides authentication and reply protection; it does not offer integrity and confidentiality mechanisms
- SIP supports end-to-end or hop-to-hop ciphering
- Commercial systems can support
  - Kerberos
  - Certificates
- MANY extensions defined in IETF Drafts

164

universidade de aveiro

SIP potencialities

- Other defined facilities (generally in extensions)
  - *Presence and instant messaging* (methods of general notifications – IETF: SIMPLE WG), caller preference, callee capabilities, ...
    - Allow the unification of servers and common databases!
  - Integration of web, email, fax/video... In an unified way
  - Uses RTSP, similarly to HTTP (request-response)
- Programming of services
  - SIP-CGI, CPL, SIP-servlet
- SIP conference

165

universidade
de aveiro

SIP Extensions

- SIP Specific Event Notification (RFC 3265)
  - SUBSCRIBE and NOTIFY messages.
  - Extensible framework by which SIP nodes can request notification from remote nodes indicating that certain events have occurred.
  - E.g. request notifications for voicemail messages waiting.
- SIP INFO Method (RFC 6086)
  - INFO message.
  - Allow for the carrying of session related control information that is generated during a session.
  - E.g DTMF tones emulation.
- SIP Extension for Event State Publication (RFC 3903)
  - PUBLISH message.
  - Allows to publish event state used within the SIP Events framework.
  - E.g. User/terminal status change (Away, Busy, etc...)

universidade
de aveiro

SIP Presence and Instant Messaging

- SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE)
  - Provides for presence and buddy lists,
  - Instant Messaging in the enterprise,
  - Telephony enabled user lists.
- Presence
  - SIP-Specific Event Notification (RFC 6665).
    - SUBSCRIBE and NOTIFY methods.
  - Session Initiation Protocol (SIP) Extension for Event State Publication (RFC 3903)
    - PUBLISH mechanism.
- Instant Messaging
  - Page Mode
    - Doesn't require a session. Uses MESSAGE method (RFC 3428).
  - Session Mode
    - Message Session Relay Protocol (RFC 4975, RFC 4976).
    - Text-based protocol for exchanging content between users
    - Requires the establishment of an MSRP session.
      - Set-up using MSRP URI, within SIP and SDP signaling.

167

SIP for Presence

- The SUBSCRIBE method is used to request current state and state updates/notifications from a remote node for a specific event.
  - Must contain an "Event" header field with information to identify the resource for which event notification is desired.
    - e.g., Voicemail (`Event: message-summary`).
  - Should contain an "Expires" header field indicating the duration of the subscription.
    - Unsubscribing is handled as refreshing a subscription, with the "Expires" header field set to "0".
  - May contain an "Accept" header field indicating the body formats allowed in notifications.
- The NOTIFY requests are sent to inform subscribers of changes in state (events) to which the subscriber has a subscription.
  - Does not terminate its corresponding subscription.
- 200 OK responses are used to acknowledge SUBSCRIBE and NOTIFY requests.
- The PUBLISH method is used to create, modify, and remove an event state.
  - e.g., Presence (away, busy, available, etc...) - `Event: presence`

168

Sample SUBSCRIBE and NOTIFY

```
Session Initiation Protocol (SUBSCRIBE)
 Request-Line: SUBSCRIBE sip:PintoDaCosta@192.168.56.102 SIP/2.0
 Message Header
  CSeq: 2 SUBSCRIBE
  Via: SIP/2.0/UDP 10.0.2.15:5060;branch=z9hG4bK5f5c8d9e-af10-1910-9cfa-0800270fe441;rport
  User-Agent: Ekiga/4.0.2
  Authorization: Digest username="PintoDaCosta", realm="asterisk", nonce="48f80483", uri="s
  From: <sip:PintoDaCosta@192.168.56.102>;tag=0b5c8d9e-af10-1910-9cf9-0800270fe441
  Call-ID: 0b5c8d9e-af10-1910-9cf8-0800270fe441@Win81
  To: <sip:PintoDaCosta@192.168.56.102>
  Accept: application/simple-message-summary
  Contact: <sip:PintoDaCosta@192.168.56.1:56079>
  Allow: INVITE,ACK,OPTIONS,BYE,CANCEL,SUBSCRIBE,NOTIFY,REFER,MESSAGE,INFO,PING,PRACK
  Expires: 3600
  Event: message-summary
  Content-Length: 0
  Max-Forwards: 70
```

```
Session Initiation Protocol (NOTIFY)
 Request-Line: NOTIFY sip:PintoDaCosta@192.168.56.1:56079 SIP/2.0
 Message Header
  Via: SIP/2.0/UDP 192.168.56.102:5060;branch=z9hG4bK6c68d274;rport
  Max-Forwards: 70
  Route: <sip:PintoDaCosta@192.168.56.1:56079>
  From: "asterisk" <sip:asterisk@192.168.56.102>;tag=as0f02fcd3
  To: <sip:PintoDaCosta@192.168.56.1:56079>;tag=0b5c8d9e-af10-1910-9cf9-0800270fe441
  Contact: <sip:asterisk@192.168.56.102:5060>
  Call-ID: 0b5c8d9e-af10-1910-9cf8-0800270fe441@Win81
  CSeq: 102 NOTIFY
  User-Agent: Asterisk PBX 1.8.10.1~dfsg-1ubuntu1
  Event: message-summary
  Content-Type: application/simple-message-summary
  Subscription-State: active
  Content-Length: 95
 Message Body
  Messages-Waiting: yes\r\n
  Message-Account: sip:asterisk@192.168.56.102\r\n
  Voice-Message: 1/0 (0/0)\r\n
```

169

universidade de aveiro

Sample PUBLISH

```
Session Initiation Protocol (PUBLISH)
 Request-Line: PUBLISH sip:Vieira@192.168.56.102 SIP/2.0
 Message Header
  CSeq: 35 PUBLISH
  Via: SIP/2.0/UDP 193.136.93.144:5060;branch=z9hG4bKf09839d5-1f61-e511-9914-7824afcb1a1a;rport
  User-Agent: Ekiga/4.0.1
  From: <sip:Vieira@192.168.56.102>
  Call-ID: 9ef4fa6e-1f61-e511-9914-7824afcb1a1a@SalAsus
  To: <sip:Vieira@192.168.56.102>
  Expires: 300
  Event: presence
  Content-Length: 551
  Content-Type: application/pidf+xml
  Max-Forwards: 70
 Message Body
  eXtensible Markup Language
   <?xml
     version="1.0"
     encoding="UTF-8"
     ?>
   <presence
     xmlns="urn:ietf:params:xml:ns:pidf"
     xmlns:dm="urn:ietf:params:xml:ns:pidf:data-model"
     xmlns:rpid="urn:ietf:params:xml:ns:pidf:rpid"
     entity="pres:Vieira@192.168.56.102">
    <tuple
      id="TCA427E12">
     <status>
     <contact
     <timestamp>
      </tuple>
    <dm:person
      id="p8">
     <rpid:activities>
       <rpid:busy/>
       </rpid:activities>
      </dm:person>
     </presence>
```

- Content-Type header defines content format.
  - e.g., XML.
- Message Body contains presence description.

170

---

universidade de aveiro

SIP for Instant Message (IM)

- The MESSAGE method (an extension to SIP) allows the transfer of Instant Messages (IM).
- MESSAGE requests carry the content in the form of MIME body parts.
  - Content-Type header defines content format.
- MESSAGE requests do not themselves initiate a SIP dialog.
  - May be sent in the context of a dialog initiated by some other SIP request.

```
Session Initiation Protocol (MESSAGE)
 Request-Line: MESSAGE sip:2001@192.168.56.102 SIP/2.0
 Message Header
  CSeq: 29 MESSAGE
  Via: SIP/2.0/UDP 192.168.56.1:5060;branch=z9hG4bK6abbfdfc-2361-e511-8e33-7824afcb1a1a;rport
  User-Agent: Ekiga/4.0.1
  From: <sip:Vieira@192.168.56.102>
  Call-ID: d0affdfc-2361-e511-8e33-7824afcb1a1a@SalAsus
  To: <sip:2001@192.168.56.102>
  Expires: 5000
  Content-Length: 5
  Content-Type: text/plain;charset=UTF-8
  Max-Forwards: 70
 Message Body
  Line-based text data: text/plain
   teste
```

171

Session Description Protocol (SDP)

- Protocol used to describe multimedia sessions announcements, requests to join or other ways of starting a multimedia session
  - When initiating multimedia teleconferences, VoIP calls, streaming video, or other sessions, is required to transmit to participants media details, transport addresses, and other session description metadata.
- A multimedia session is a set of streams that is active for a period of time
- Not “exactly a protocol”, but describes data used in other protocols
  - SDP is purely a format for session description.
  - SDP (RFC 2327, RFC 4566) provides a standard representation for such information, irrespective of how that information is transported.
  - SDP is intended to be general purpose so that it can be used in a wide range of network environments and applications: SIP, RTSP, H.332, PINT.
  - SIP carries (encapsulates) SDP messages.

172

SDP Session Description

- An SDP session description is entirely textual.
- Consists of a number of lines of text of the form *<type>*=*<value>*
  - *<type> is* one case-significant character.
  - *<value>* is structured text whose format depends on *<type>*.
- Consists of a session-level section followed by zero or more media-level sections.
  - The session-level part starts with a "v=" line and continues to the first media-level section.
  - Each media-level section starts with an "m=" line.

- Types

Session description
- v= (protocol version)
- o= (originator and session identifier)
- s= (session name)
- i=* (session information)
- u=* (URI of description)
- e=* (email address)
- p=* (phone number)
- c=* (connection information -- not required if included in all media)
- b=* (zero or more bandwidth information lines)
- One or more time descriptions ("t=" and "r=" lines; see below)
- z=* (time zone adjustments)
- k=* (encryption key)
- a=* (zero or more session attribute lines)
- Zero or more media descriptions

Time description
- t= (time the session is active)
- r=* (zero or more repeat times)

Media description, if present
- m= (media name and transport address)
- i=* (media title)
- c=* (connection information -- optional if included at session level)
- b=* (zero or more bandwidth information lines)
- k=* (encryption key)
- a=* (zero or more media attribute lines)

173

universidade de aveiro

SDP: Session Description Protocol

- E.g:
  - v=0
  - o=g.bell 877283459 877283519 IN IP4 132.151.1.19
  - s=Come here, Watson!
  - u=http://www.ietf.org
  - e=g.bell@bell-telephone.com
  - c=IN IP4 132.151.1.19
  - b=CT:64
  - t=3086272736 0
  - k=clear:manhole cover
  - m=audio 3456 RTP/AVP 96 - media
  - a=rtpmap:96 VDVI/8000/1 - attributes
  - m=video 3458 RTP/AVP 31
  - m=application 32416 udp wb

174

universidade de aveiro

Interconnecting (two) Large Networks

How to interconnect PSTN and ISPs

179

Learning outcomes

- Understand the scope of VoIP models
- Describe RTP operation
- Understand the SIP and H.323 protocols
- Describe architectures for interconnecting POTS and the Internet.

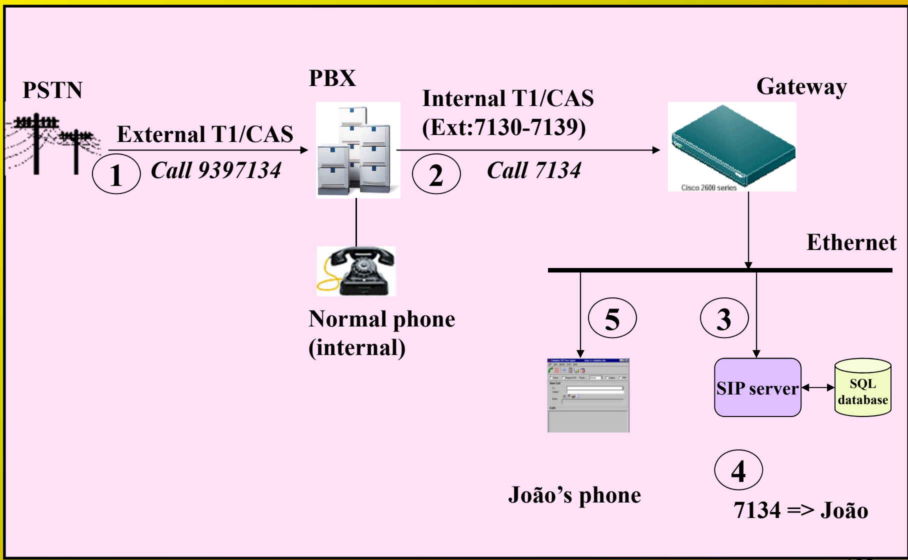
180

PSTN interoperation

- Translate audio (PCMU/PCMA)
- Translate signalling (PRI/T1,ISUP)
  - Different signalling
  - Advanced capabilities in SIP are not mapped in PSTN
- Translate identifiers (phone numbers)
- Determine transition points

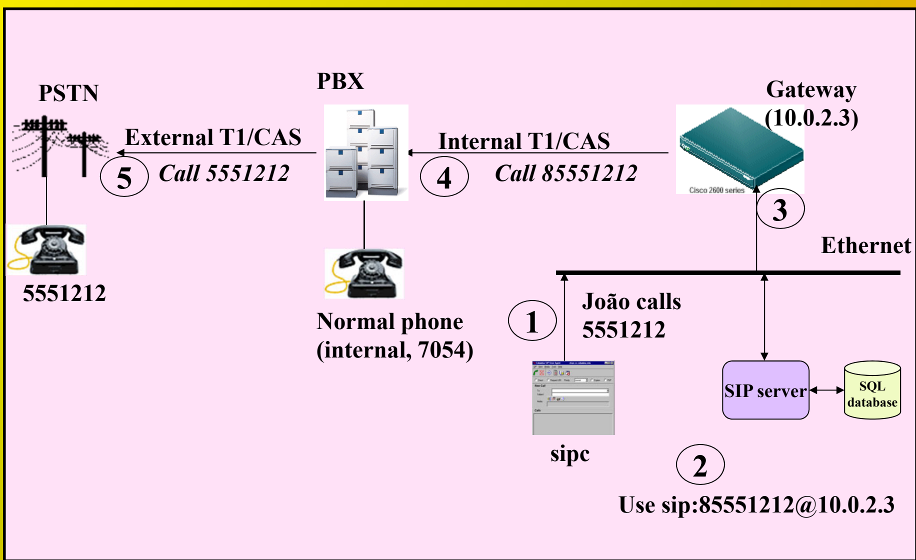
181

PSTN calls to IP



182

IP calls to PSTN



183

universidade de aveiro

ISPs and PSTN

- Having VoIP (specially voice) sessions connecting to old-style phone networks implies:
  1. Interconnecting voice signalling
  2. Interconnecting data (voice)
     - Typically this is set by routing tables in both sides
  3. Linking both inteconnection actions
  4. Selecting where to do each one of these

184

universidade de aveiro

What about REAL interoperation?

- Signaling boxes between the data and circuit systems must be interconnected
  - Multiple interconnection points may exist
- Systems must select best interconnection points
  - This implies best routing solution
    - And this is mixed routing – both in data and circuit systems
  - Interoperation points may be different for the data and control planes
- Different types of boxes may exist (interoperation of data/control/both)

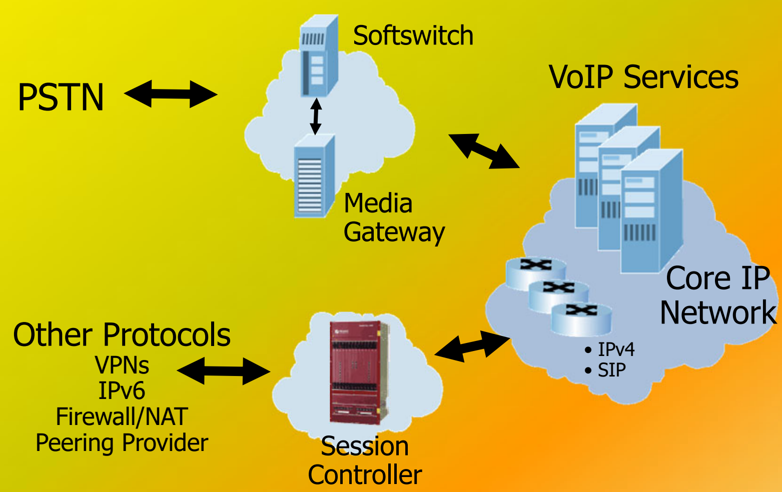
185

VoIP and PSTN Interoperability in Large Scalable Scenarios

- Requires an application programming interface and a corresponding protocol for controlling VoIP Gateways from external call control elements.
- Signaling must be inter-operable between PSTN and VoIP.
- Protocols:
  - Media Gateway Controller Protocol (MGCP) - RFC 2705
  - MGCP evolution/successor → H.248/Megaco (RFC 3015) → H.248.1/Gateway Control Protocol (RFC 3525)
    - These are control plane signaling only.
  - SIGTRAN (Signaling Transport) is the standard telephony protocol used to transport Signaling System 7 (SS7) signals over the Internet.
    - Stream Control Transmission Protocol (SCTP) – RFC 3286
      - Is an IP transport designed for transporting signaling information over an IP network.
      - Reliable transport protocol with support for framing of individual message boundaries.

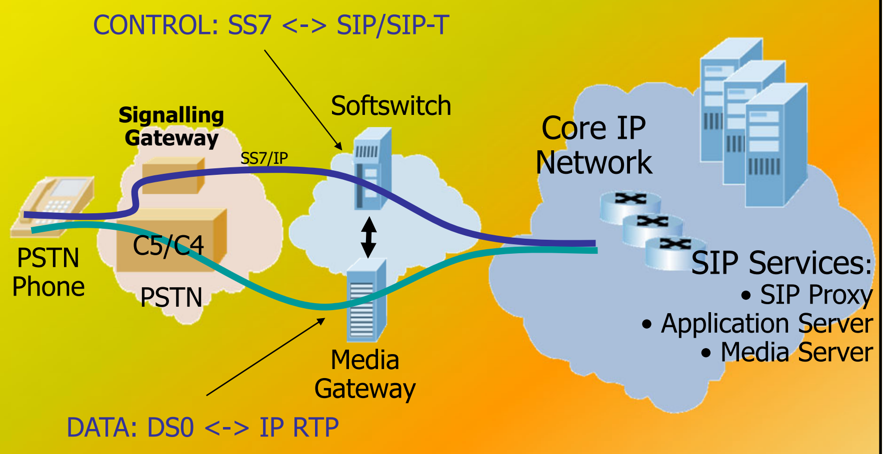
186

Network interoperation



Technologically addressed by media gateways and softswitches (PSTN ⇔ IP) and session controllers (IP ⇔ IP), and associated VoIP data plane protocols

PSTN/IP Interworking



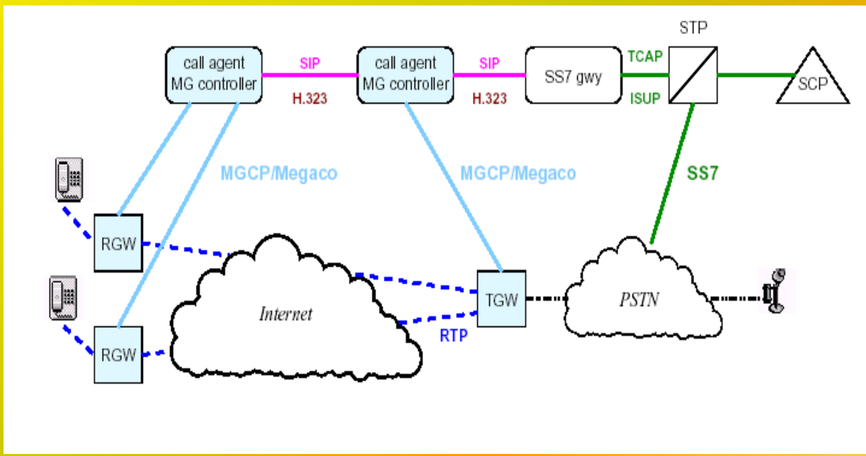
Softswitches are associated with Media Gateways (often colocated in the same box), to bridge both signalling and data.

MGCP e Megaco

- Media Gateway Controller Protocol (RFC 2705)
- Controls phone Gateways resorting to *external* control elements, the media gateway controllers (MGC) a.k.a. call agents
  - **Gateways: Eg: RGW (residential gateway): physical interconnection between VoIP networks and phone interfaces at homes**
  - **The call control “intelligence” is outside the gateways, and is controlled by external elements**
  - **master-slave philosophy**
- Objective: scalable gateway infrastructure between PSTN and IP networks
- MGCP Successor: H.248/Megaco
- These are control plane signaling ONLY.

MGCP Architecture

Objective: VoIP large scalable implementations

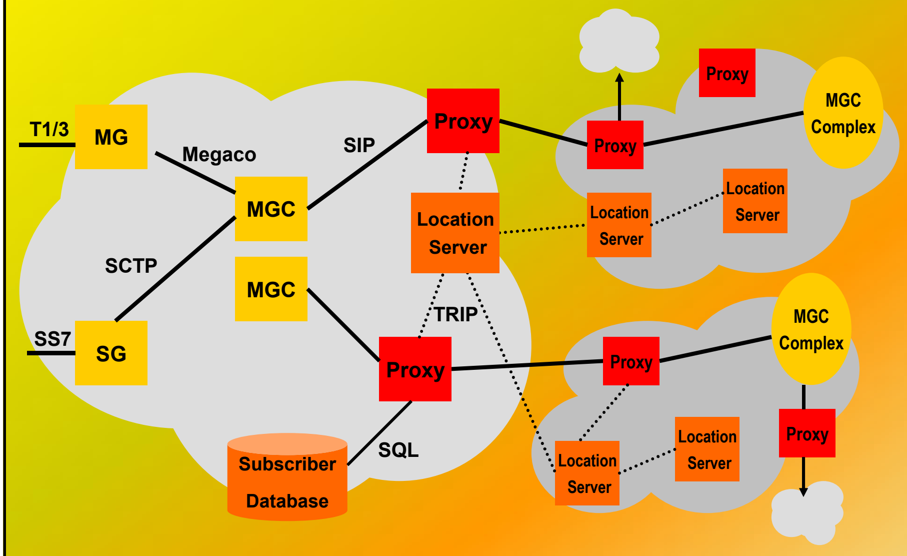


RGW: Residential Gateway
TGW: Trunk Gateway

MGCP/H.248 Elements

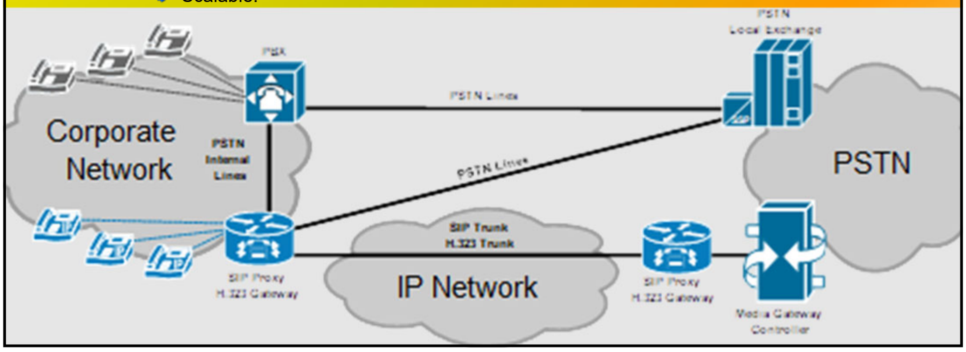
- Media Gateway Controller (MGC)
  - Controls the parts of the call state that pertain to connection control for media channels in a MG.
- Media Gateway (MG: RGW/TGW)
  - Converts media provided in one type of network to the format required in another type of network.
  - MG could terminate bearer channels from a switched circuit network (e.g., DS0s) and media streams from a packet network (e.g., RTP streams in an IP network).
- Signaling Gateway (SG)
  - Responsible for transferring signaling messages (e.g., SS7 messages) to different protocols and transports.
    - Signaling Transport (SIGTRAN)
    - e.g., SS7 to SIGTRAN (SCTP/IP).

Inter-MGC Architecture

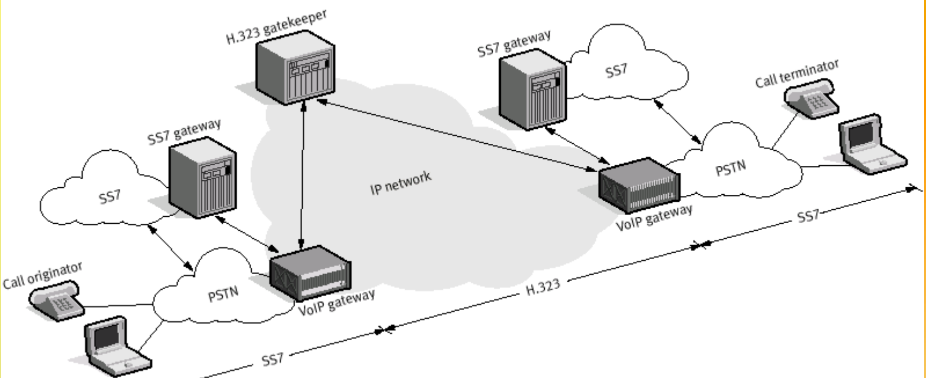


VoIP and PSTN Connectivity

- SIP proxy or H.323 gateway.
  - With PSTN interface (to ISP or local PBX).
    - Requires multiple PSTN Lines.
    - Not scalable.
  - With SIP or H.323 trunk to remote SIP proxy or H.323 Gateway.
    - Remote proxy/gateway interfaces with PSTN network.
    - Remote proxy/gateway owned by PSTN ISP or by a third-party entity.
    - Usually TCP/IP transport with a TLS security layer.
    - Scalable!



VoIP and SS7



SS7-Based VoIP Network