Advanced Networking

Voice over IP
&
Other Multimedia Protocols

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SIP: Session Initiation Protocol

- Defined by IETF
- RFC 2543 (first release march 1999)
  - many other RFCs … see IETF site and later on
- Multiparty MUltimedia Session Control (Mmusic) WG
- Born from Mbone experience and as a more "Internet" alternative to H.323

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IETF service vision

- First objective is connectivity
  - Transport through IP
  - Iintelligence is in hosts and not in network nodes (routers) which only switch and forward datagrams
- Scalability and Security are primary concerns … although scalability is addressed, while security …
- SIP is an umbrella protocol suite using other light mono-function protocols
  - Avoid function duplication
  - Modular development

SIP: geenral characteristics

- Client – server protocol
- The usage is: "invite" users in participation to multimedia sessions
- Uses several http-derived functionalities
- Independent from the transport layer
- Should be Scalabile, Modular and Simple
- Defining a suite is based on the use of other protocols
  - RTP/RTCP (voice/video conversational transport)
  - SDP: Session Description Protocol
  - SAP: Session Announcement Protocol
  - RTSP: Real Time Streaming Protocol (VoD like)

SIP: A General Purpose Session Control Protocol?

- SIP is not limited to IP telephony
  - SDP quite flexible
  - arbitrary payloads allowed
- Other applications relying on notion of session:
  - distributed virtual reality systems
  - network games
  - video conferencing
- Applications may leverage SIP infrastructure (Call Processing, User Location, Auth., etc.)
  - Instant Messaging and Presence
  - SIP for Appliances !?!?!?

SIP: it's not…

- A transport Protocol
- A QoS Reservation Protocol
- A gateway Control Protocol
- It does NOT dictate ...
  - product features and services (color of your phone and distinctive ringing melodies, number of simultaneous calls your phone can handle, don't disturb feature, ...)
  - network configuration

SIP: Architectural Elements

- Client (o end system)
  - Send SIP requests
  - Normally embedded into a SIP User Agent Server
- User Agent Server (UAS)
  - Answers incoming queries and calls
- Redirect Server
  - Redirect calls to another server
- Proxy Server
  - Send Requests to another server, including UASs

SIP: Addresses and Methods

- Addresses are URI (Universal Resource Identifier):
  - sip:jdrosen@bell-labs.com:5067
  - sip:ann:passwd@lucent.com
- 6 methods (or primitives):
  - INVITE: Starts or invite to a converence
  - BYE: Closes a participation
  - CANCEL: Terminates a search (unsuccesful)
    OPTIONS: Query a client on his “capabilities”
  - ACK: Accept a call (IVITE)
  - REGISTER: Registers a client onto a server, normally a proxy, include location information

SIP: Message syntax

- Derived from HTTP:

INVITE gerla@cs.ucla.edu SIP/2.0
From: locigno@dit.unitn.it (Renato Lo Cigno)
Subject: Next visit to L.A.
To: gerla@cs.ucla.edu (Mario Gerla)
Call-ID: 1999284605.56.86@
Content-type: application/sdp
CSeq: 4711
Content-Length: 187

- Make use of the Session Description Protocol (SDP)

Session Description Protocol

- Textual syntax for multimedia sessions (unicast and multicast)
- Basic characteristics
  - Describes Audio/Video flows that from the session and the related parameters
  - Includes addresses (internal ports) for the termination of different streams
  - "Commands" initial and termination times

SDP: an example

v=0 **Protocol version**

Creator and session identifier **<address type>**

o= locigno 28908044538 289080890 IN IP4 93.175.132.118

<username> <session id> <version> <network type> <address>

s=SIP Tutorial **Session name**

e=ghittino@csp.it **Email address**

c=IN IP4 126.16.69.4 **Connection information**

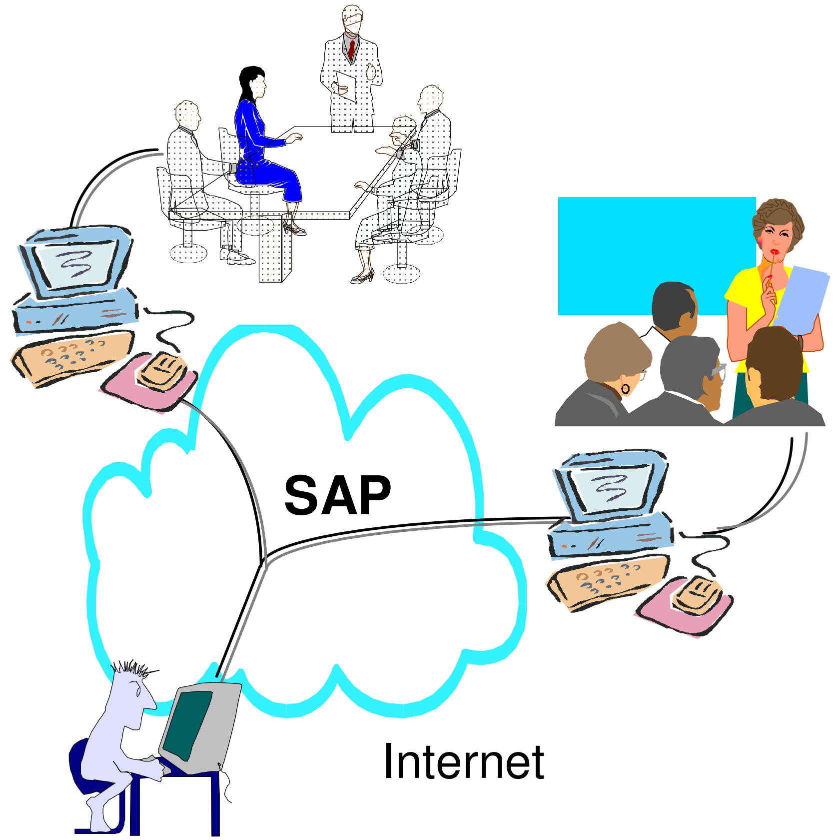
t=28908044900 28908045000 **Time the session is active (start – stop)**

m=audio 49170 RTP/AVP 0 98 **Media name and transport address**

a=rtpmap:98 L16/11025/2 **Media attribute line**

Session Announcement Protocol

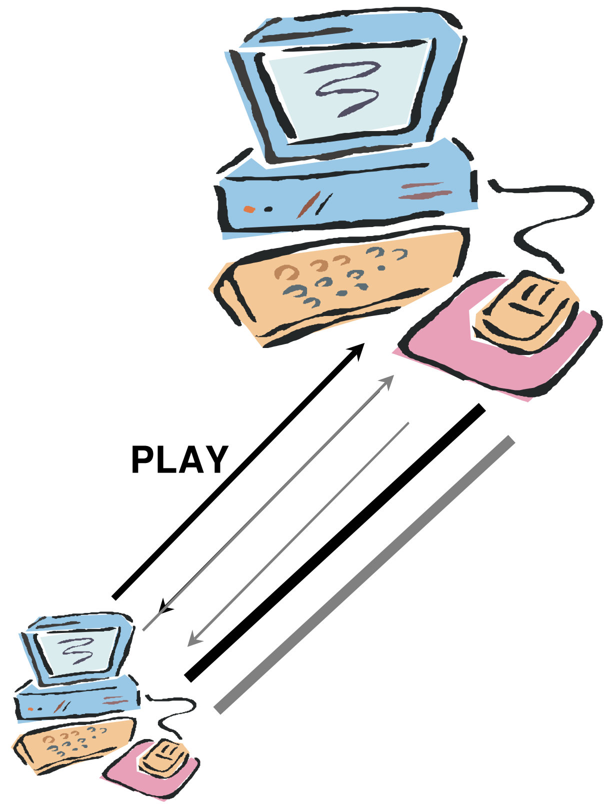
- Announces multimedia sessions via multicast
- Uses SDP for the description and RTP for media distribution
- Can be used for VoIP conferencing, but it is rarely done



Real Time Streaming Protocol

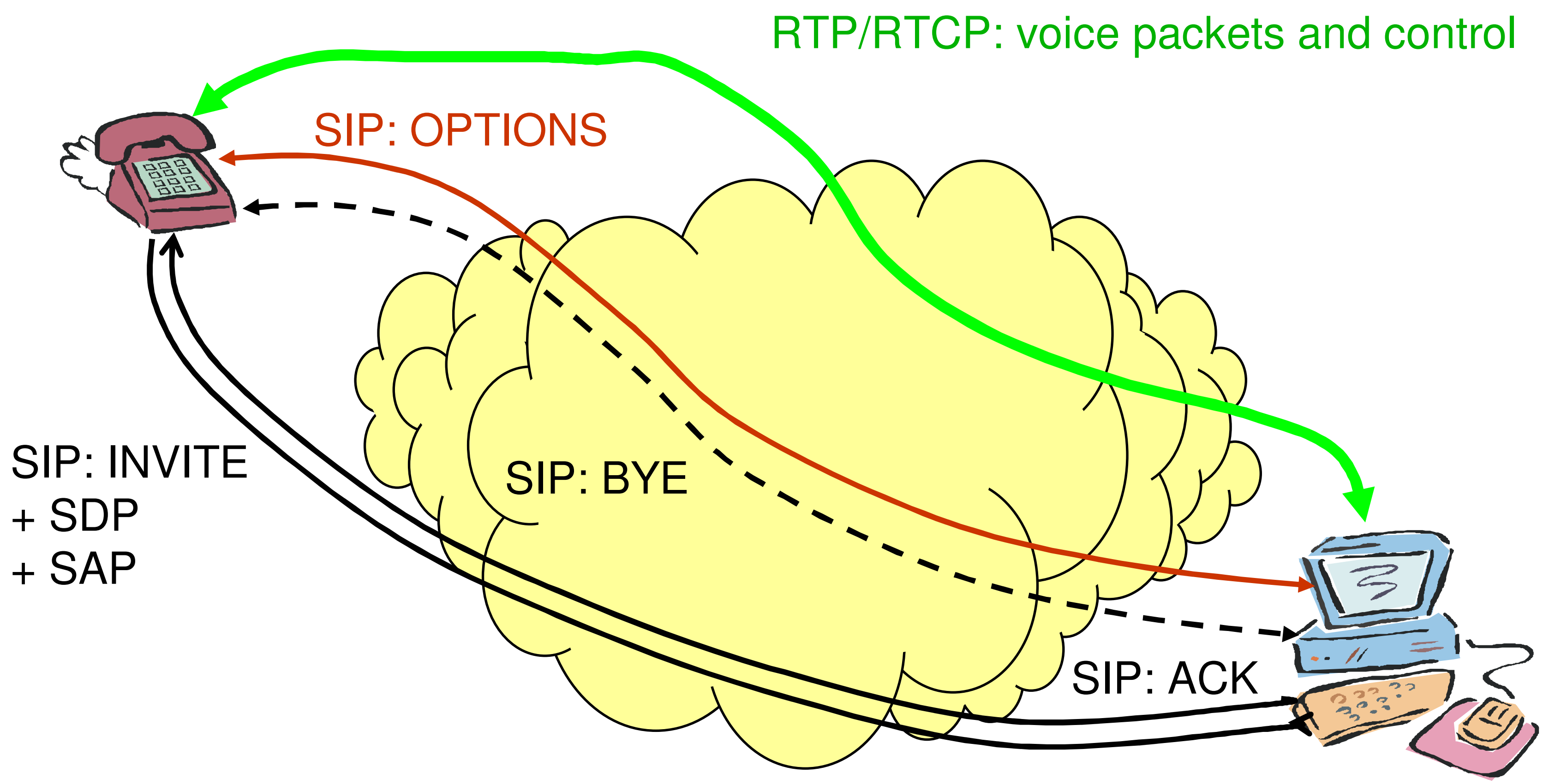
- "Media server" control for "on-demand" services
- VCR-like controls:
  - Play, Pause, Fast-forward, Rewind, Record, ...
- An RTSP server can be queried by a client using standard SIP invitation
- The session is again described through SDP

RTSP Server



RTSP Client

SIP: Voice Call example



SIP Servers and Clients

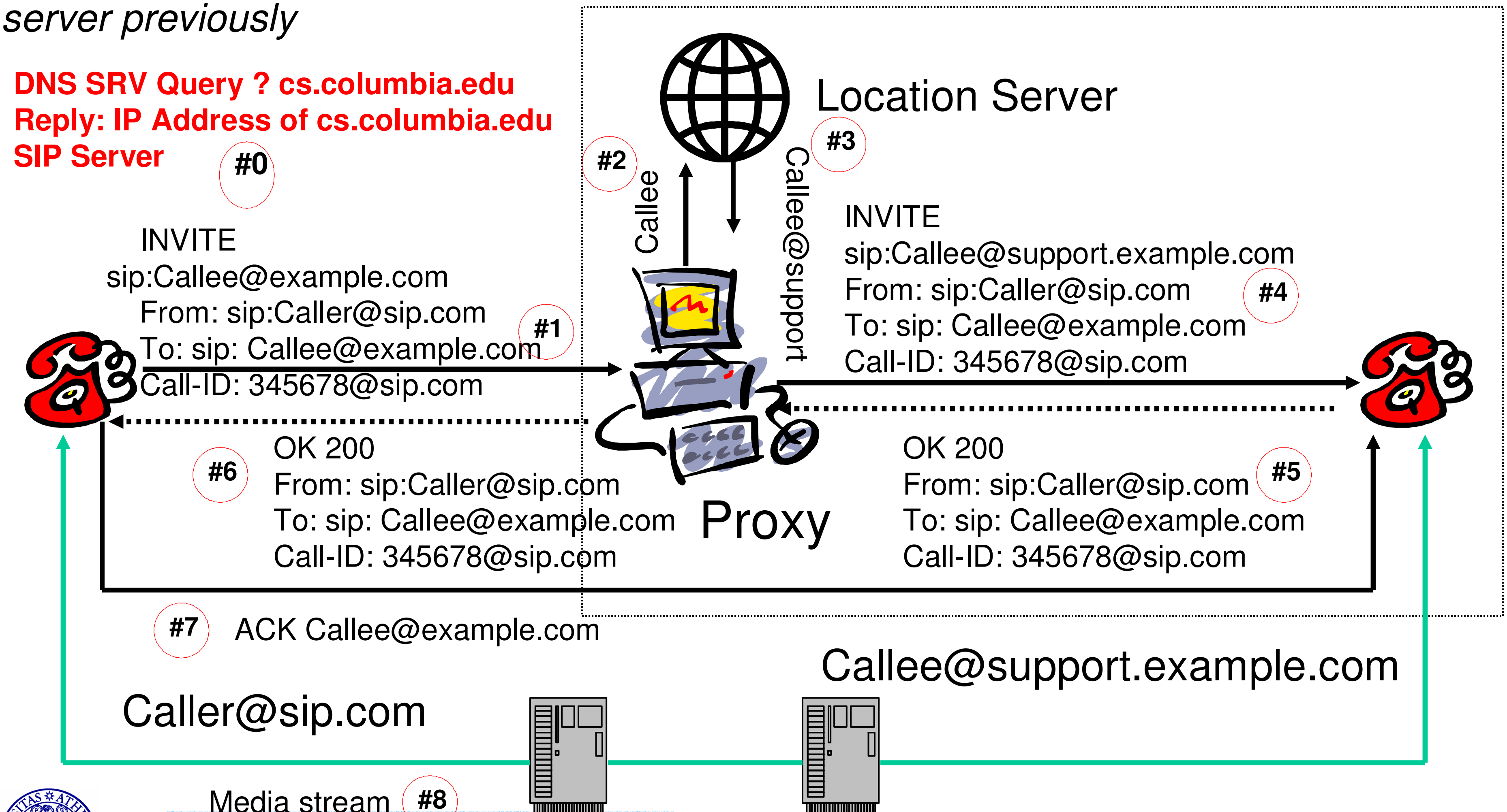
- User Agent (user application)
  - UA Client (originates calls)
  - UA Server (listens for incoming calls)
  - both SW and HW available
- SIP Proxy Server
  - relays call signaling, i.e. acts as both client and server
- SIP Redirect Server
  - redirects callers to other servers
- SIP Registrar
  - accept registration requests from users
  - maintains user's whereabouts at a Location Server (like GSM HLR)

Proxy Server Functionality

- Serve as rendezvous point at which callees are globally reachable
- Perform routing function, i.e., determine to which UA/proxy/redirect an incoming call should be relayed
- **Allow the routing function to be programmable**
- **Forking: Several destinations may be tried for a request sequentially or in parallel**
- May serve as AAA trigger points

SIP Operation in Proxy Mode

User Caler@sip.com
on left-hand side
is initiating a call to Callee@example.com
on right-hand side; Callee registered with his
server previously



SIP RFC2543 Methods

- INVITE: initiates sessions
  - Request URI indicated destination; may be changed on the path
  - session description included in message body
  - re-INVITEs used to change session state
- ACK: confirms session establishment
  - can only be used with INVITE
- BYE: terminates sessions
- CANCEL: cancels a pending INVITE
  - if a CANCEL follows a RE-INVITE the session is not torn down!
- OPTIONS: capability inquiry
  - replied as INVITE
  - may include Allow, Accept, Accept-Encoding, Accept-Language, Supported,...
- REGISTER:

SIP REGISTER Method

- REGISTER binds a permanent address to current location
- similar to registering with HLRs in GSM
- REGISTERs may be multicast
- may convey user data (e.g., CPL scripts)
- default registration timeout: 3600 s
- may be also used to cancel or query existing registrations

SIP Extension Methods

- INFO mid-call signaling (RFC 2976)
- COMET precondition met (draft-ietf-sip-manyfolks-resource)
- PRACK provisional reliable responses acknowledgement (draft-ietf-sip-100rel)
- SUBSCRIBE/ NOTIFY/ MESSAGE instant messaging (draft-rosenberg-impp-*)

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SIP Response Codes

- **Borrowed from HTTP: xyz explanatory text**
- **Receivers need to understand x**
- **x80 and higher codes avoid conflicts with future HTTP response codes**

- **1yz** Informational
  - 100 Trying
  - 180 Ringing (processed locally)
  - 181 Call is Being Forwarded
- **2yz** Success
  - 200 ok
- **3yz** Redirection
  - 300 Multiple Choices
  - 301 Moved Permanently
  - 302 Moved Temporarily
- **4yz** Client error
  - 400 Bad Request
  - 401 Unauthorized
  - 404 Not Found
  - 405 Method not Allowed
  - 407 Proxy Authentication Required
  - 415 Unsupported Media Type
  - 482 Loop Detected
  - 486 Busy Here
- **5yz** Server failure
  - 500 Server Internal Error
- **6yz** Global Failure
  - 600 Busy Everywhere

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SIP Message Structure

Request Method

INVITE sip:UserB@there.com SIP/2.0

Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Subject: Happy Christmas
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: 147

Message Header Fields

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

Payload

Response Status

SIP/2.0 200 OK

Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=65a35
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Subject: Happy Christmas
Contact: LittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: 134

v=0
o=UserB 2890844527 2890844527 IN IP4 there.com
s=Session SDP
c=IN IP4 110.111.112.113
t=0 0
m=audio 3456 RTP/AVP 0

“receive RTP G.711-encoded audio at 100.101.102.103:49172”

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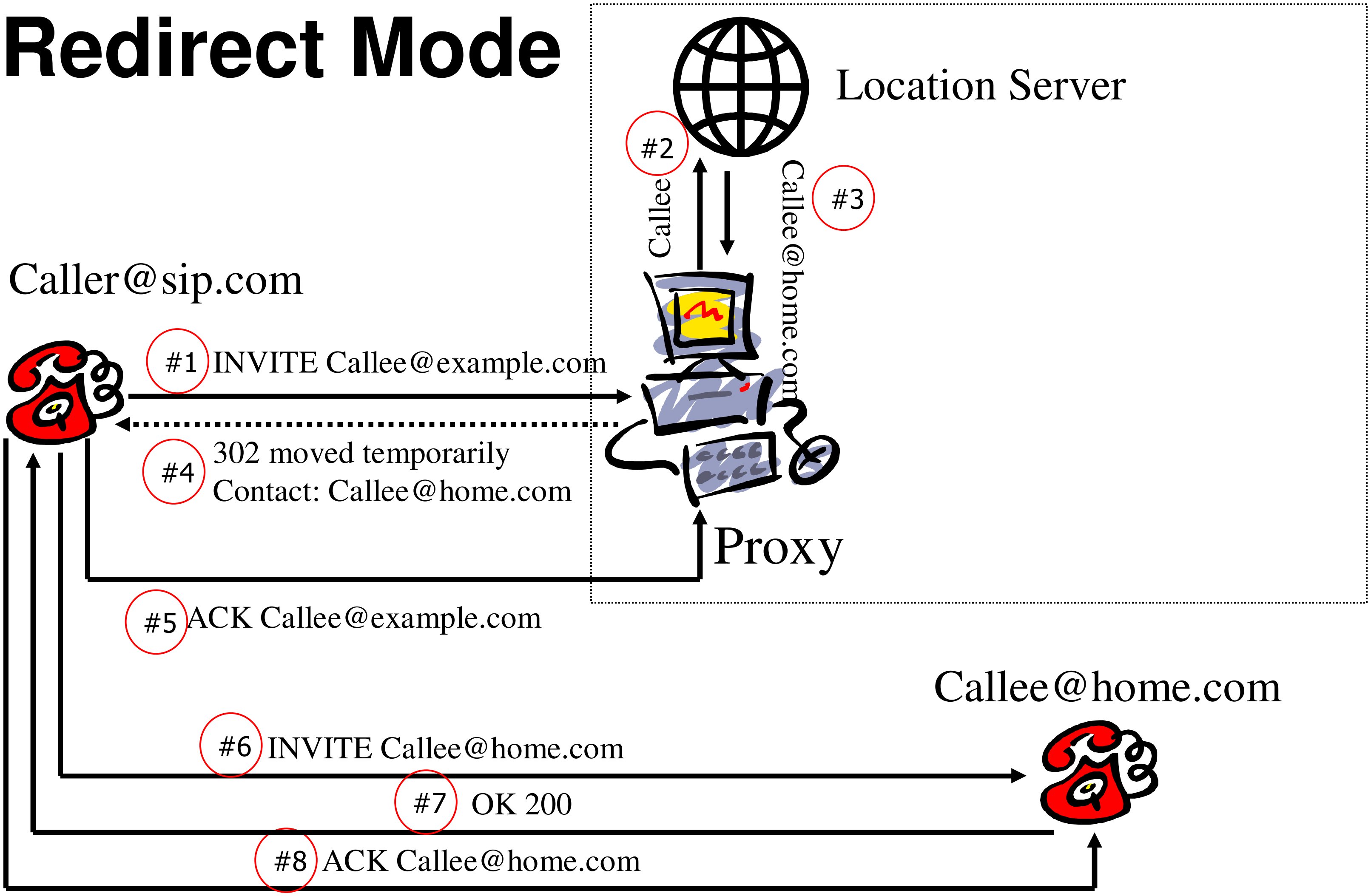
SIP Addresses

- URLs used to identify a call party a human being or an automated service
- examples:
  - sip:voicemail@examples.com?subject=callme
  - sip:sales@bigcom.com; geo.position:=48.54_-123.84_120
- must include host, may include user name, port number, parameters (e.g., transport), etc.
- may be embedded in Webpages, email signatures, printed on your business card, etc.
- address space unlimited
- non-SIP URLs can be used as well (mailto:, http:, ...)

SIP Server -- Proxy versus Redirection

- A SIP server may either **proxy** or **redirect** a request
  - statically configured
  - dynamically determined (CPL).
- Redirection
  - a user moves or changes her provider (PSTN: "The number you have dialed is not available.")
  - caller does not need to try the original server next time. Stateless.
- Proxy useful if
  - forking, AAA, firewall control needed
  - proxying grants more control to the server

SIP Operation in Redirect Mode



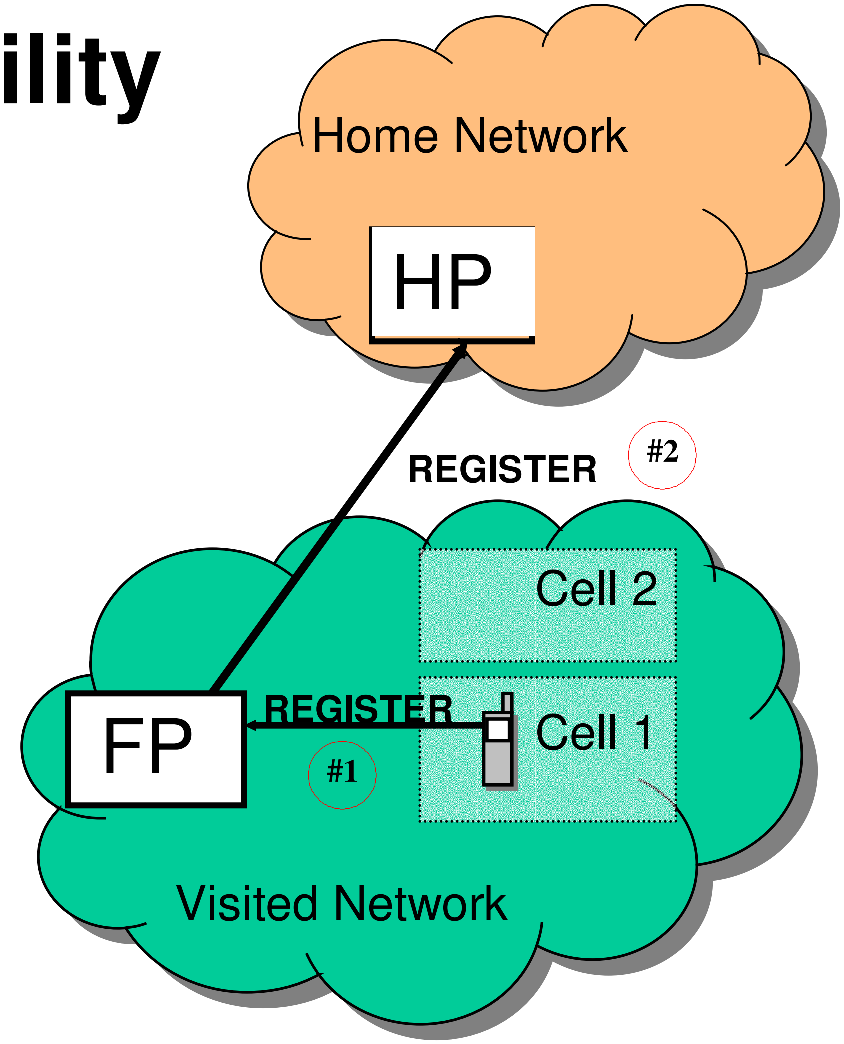
- Mobility
  - User mobility
  - Service mobility
  - 3GPP

SIP and Terminal Mobility

- Terminal can move between subnetworks
- Issues to consider:
  - Handoff performance
  - Redirection authentication
- Mobile hosts (MH) inform their home proxy about their new locations using REGISTER
- Mid-call mobility (Session mobility) is dealt with using reINVITE

SIP and Terminal Mobility

Signalling



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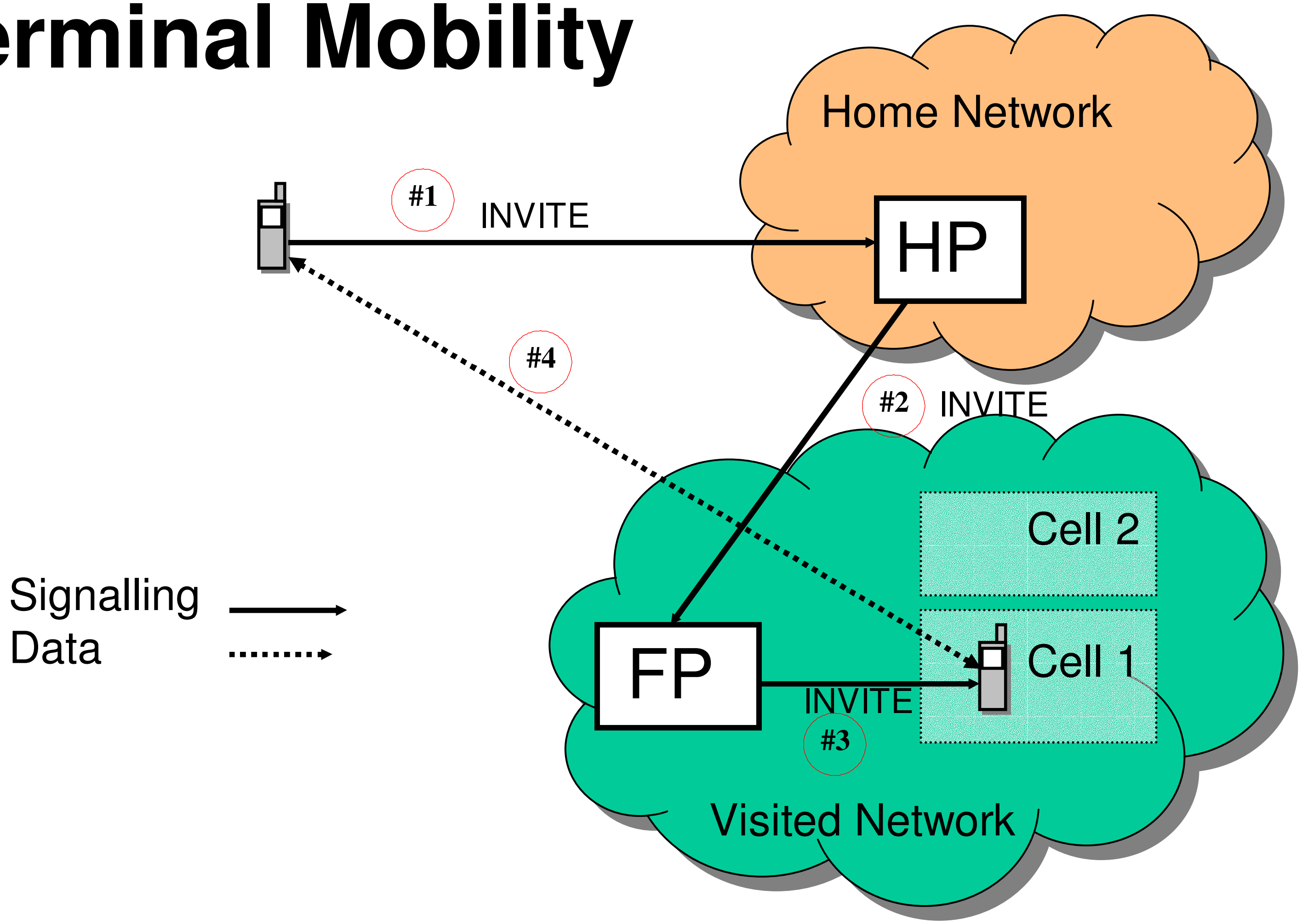
SIP and Terminal Mobility



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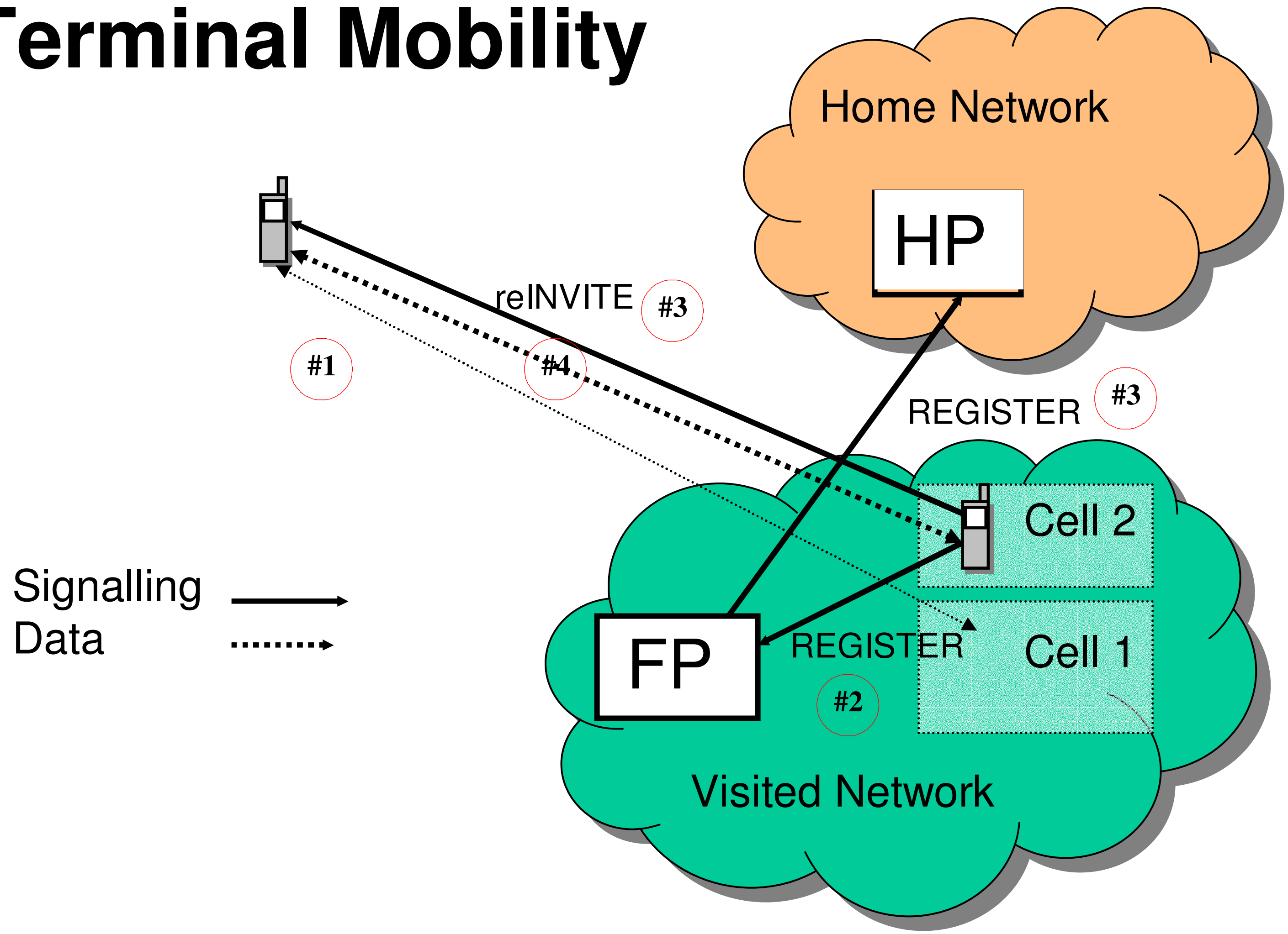
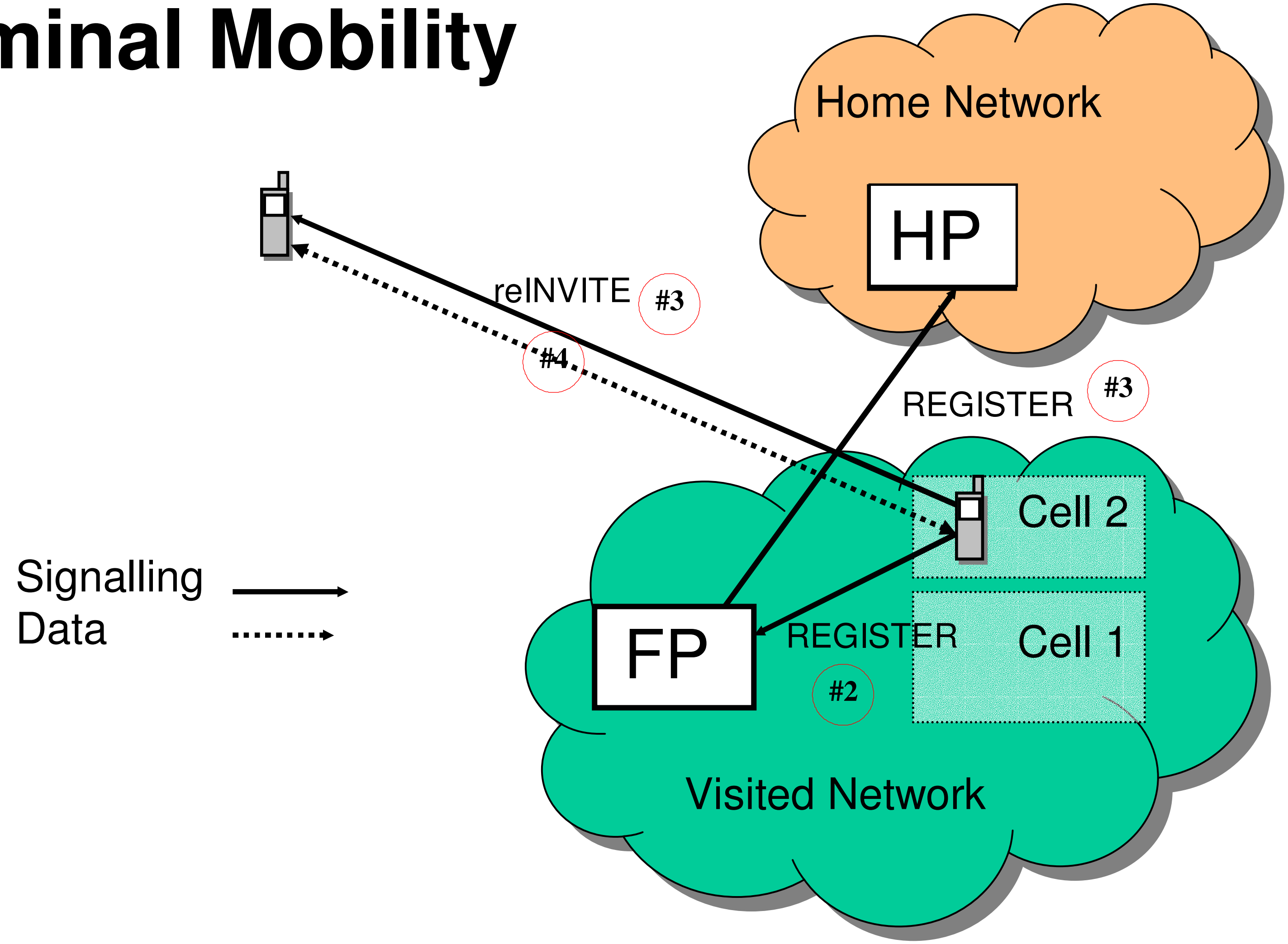
SIP and Terminal Mobility



SIP and Terminal Mobility



SIP and Personal Mobility

- Person uses different devices
- REGISTER binds a person to a device
- Proxy and redirect translate address to location and device
- Issues to consider:
  - Authentication
  - Binding different addresses to single person: LDAP ...

SIP and Service Mobility

- Use same services from different locations and devices
  - speed dial, address book, media preferences, call handling
- Services located at home server
  - RECORD-ROUTE home proxy to force calls to be processed by home servers
  - Services located at end systems
  - retrieve with REGISTER
- Issues to consider
  - services need to be device independent
    - standardised service description (CPL)
  - User recognition and authentication

SIP & 3GPP

- Può rappresentare una delle “alleanze” più promettenti per l’integrazione delle TLC (soprattutto fisso/mobile)
- Punta all’integrazione della mobilità con i servizi streaming in IP
- Introduce espressamente in 3G SIP ed un nodo di controllo della multimedialità su IP: CSCF (Call/Session Control Function)