

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



IETF service vision

- **First objective is connectivity**
 - Transport through IP
 - Intelligence 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: general 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 Scalable, Modular and Simple
- Defining a suite is based on the use of other protocols
 - SDP: Session Description Protocol
 - SAP: Session Announcement Protocol
 - RTP/RTCP (voice/video conversational transport)
 - 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 (unsuccessful)
 - 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**

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)**

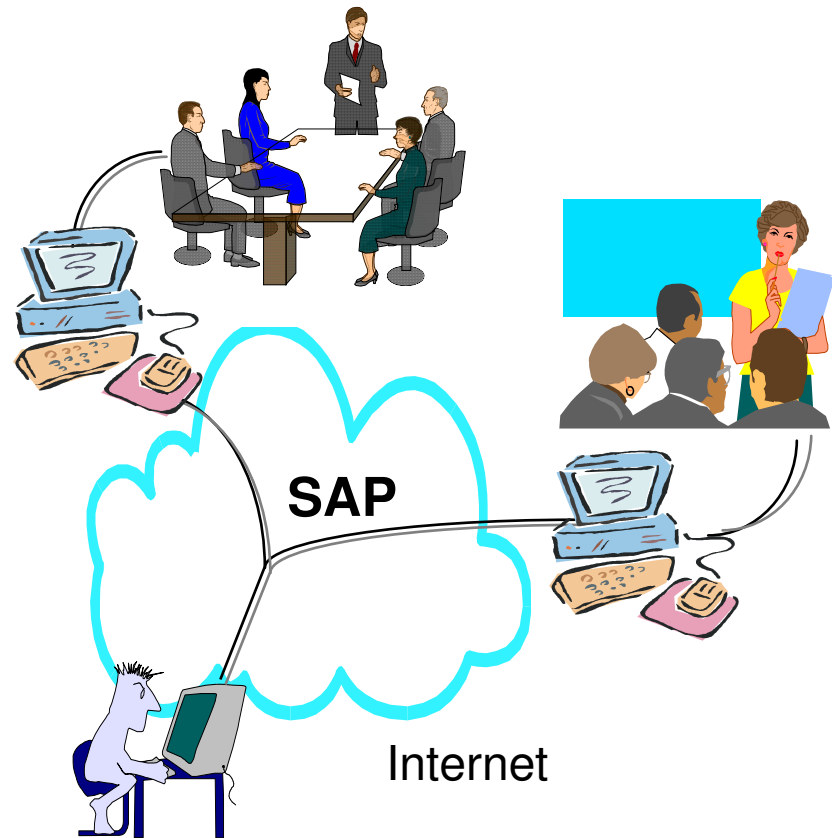
m=audio 49170 RTP/AVP 0 98 **Media name and transport address**

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



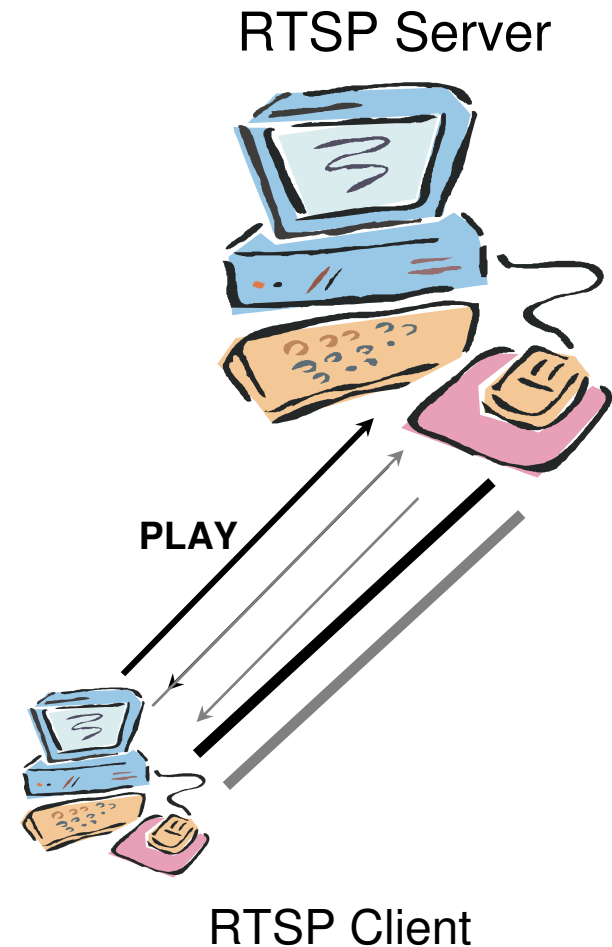
Session Announcement Protocol

- 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

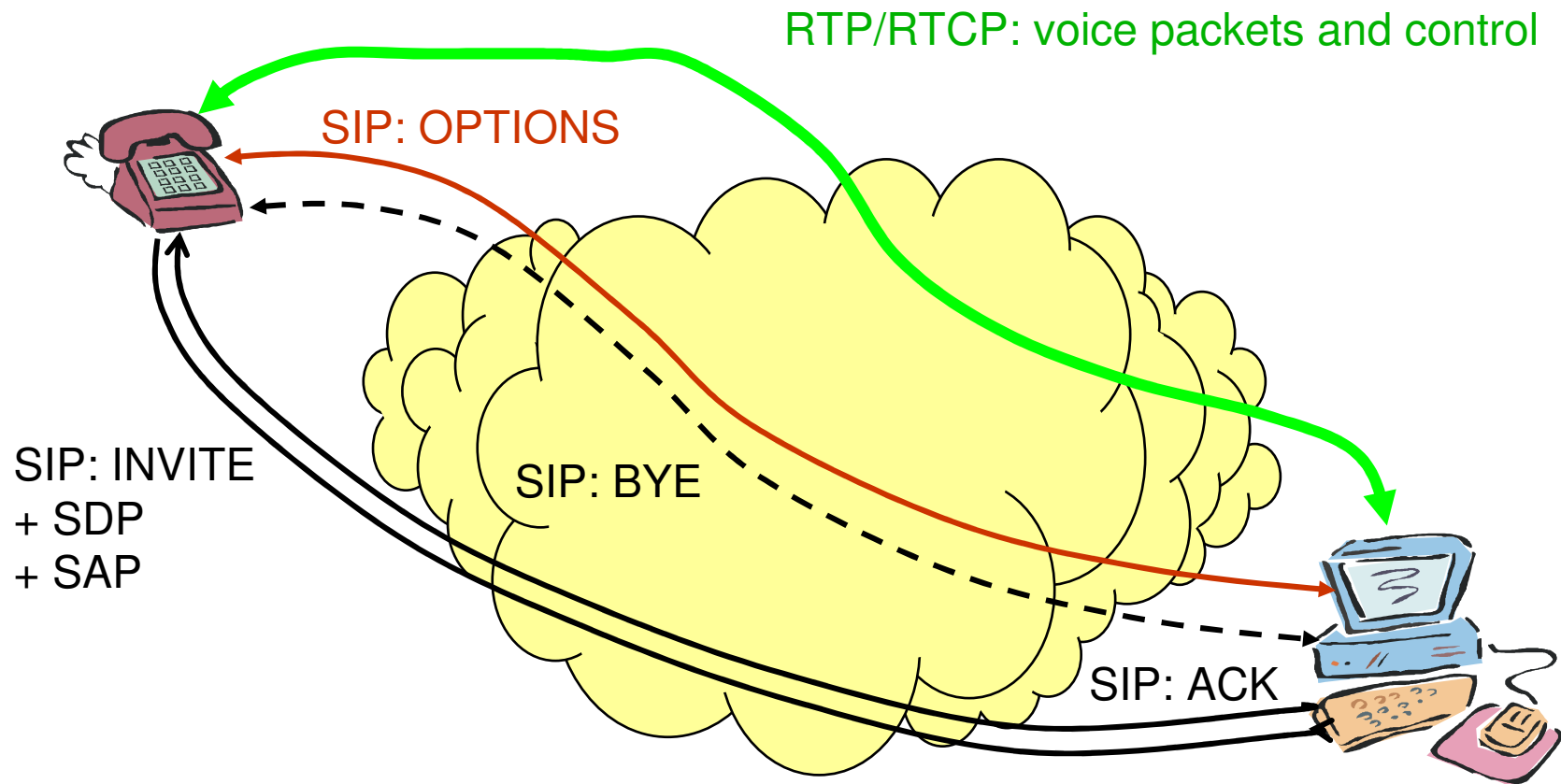


Real Time Streaming Protocol

- "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



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

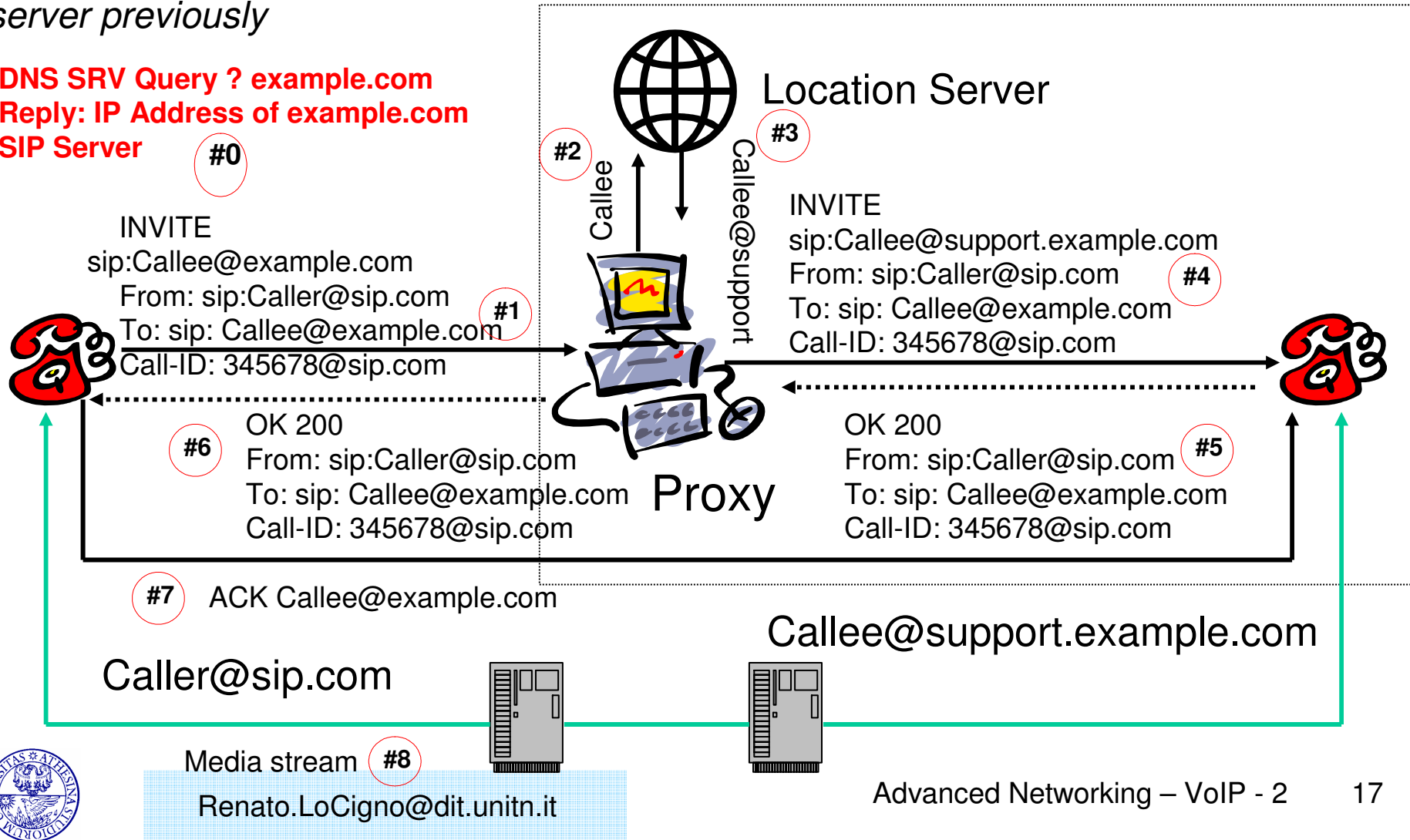
- Serve as rendezvous point at which callees are globally reachable
- Perform routing function, i.e., determine to which UA/proxy/redirect an incoming call should be relayed
- **Allow the routing function to be programmable**
- **Forking: Several destinations may be tried for a request sequentially or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

DNS SRV Query ? example.com
Reply: IP Address of example.com
SIP Server



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 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



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**

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=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

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”



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

