

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



La visione IETF

- **L'obiettivo e' la connettivita'**
 - Il trasporto e' tramite protocollo IP
 - l'intelligenza e' ai bordi della rete (nei terminali) e non nascosta nella rete.
- Scalabilita` e Sicurezza sono requisiti primari
 - anche se la sicurezza
- Protocolli piccoli e mono-funzionali
 - Evitano duplicazione di funzioni
 - Modularita`



SIP: caratteristiche generali

- Protocollo **client - server**
- Utilizzato per "invitare" gli utenti a sessioni multimediali
- Utilizza diverse funzionalità di HTTP
- Indipendente dal trasporto
- Scalabile, Modulare, Semplice
- Impiega altri protocolli multimediali
 - RTP/RTCP (trasporto voce)
 - SDP: Session Description Protocol
 - SAP: Session Announcement Protocol
 - RTSP: Real Time Streaming Protocol



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: Elementi dell'architettura

- Client (o end system)
 - Invia le richieste SIP
 - Usualmente contiene un SIP User Agent Server
- User Agent Server
 - Soddisfa le richieste di chiamata entranti
- Redirect Server
 - Redirige una chiamata su un altro server
- Proxy Server
 - Invia la richiesta ad un altro server



SIP: Indirizzi e Metodi

- Gli indirizzi sono URI (Universal Resource Identifier):
 - sip:jdrosen@bell-labs.com:5067
 - sip:ann:passwd@lucent.com
- 6 metodi:
 - INVITE: Inizia o invita ad una conferenza
 - BYE: Termina la partecipazione ad una conferenza
 - CANCEL: Termina una ricerca
 - OPTIONS :Interroga un client sulle sue "capabilities"
 - ACK: Accetta la chiamata (invito)
 - REGISTER: Informa un SIP server sulla posizione di un utente



SIP: Sintassi dei messaggi

- La sintassi è ripresa da 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
```



Session Description Protocol

- Sintassi testuale per descrivere sessioni multimediali unicast e multicast
- Caratteristiche base
 - Descrive i flussi Audio/Video che formano la sessione ed i relativi parametri
 - Contiene gli indirizzi di destinazione dei diversi stream
 - "Governa" i tempi di inizio e fine di ogni sessione
 - Molto semplice



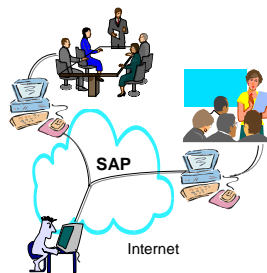
SDP: an example

```
v=0 Protocol version
o= ghittino 28908044538 289080890 IN IP4 93.175.132.118
  Creator and session identifier <address type>
  <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
a=rtpmap:98 L16/11025/2 Media name and transport address
  Media attribute line
```



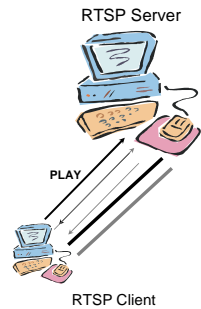
Session Announcement Protocol

- Annuncia sessioni multimediali via multicast
- Descrive sessioni (normalmente RTP) mediante SDP

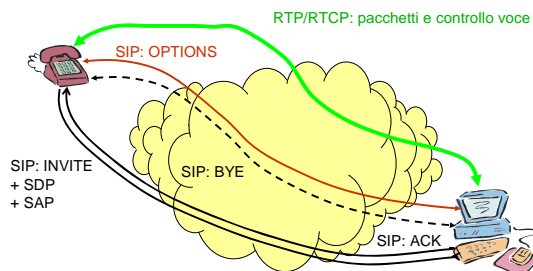


Real Time Streaming Protocol

- Controllo di un "media server" per servizi "on-demand"
- Controlli tipo VCR:
 - Play, Pause, Fast-forward, Rewind, Record, ...
- Un server RTSP puo` essere interrogato da un client mediante SIP (invito)
- La sessione e` descritta mediante SDP



SIP: Esempio di una chiamata vocale



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)

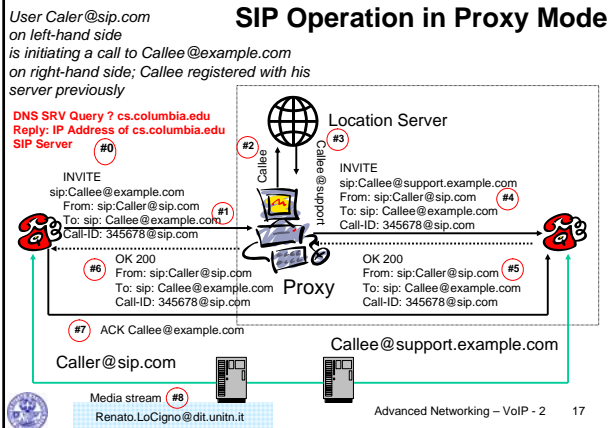


Proxy Server Functionality

- 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



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/ instant messaging
NOTIFY/ (draft-rosenberg-imp-*)
MESSAGE



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**
 - **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
- **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



SIP Message Structure

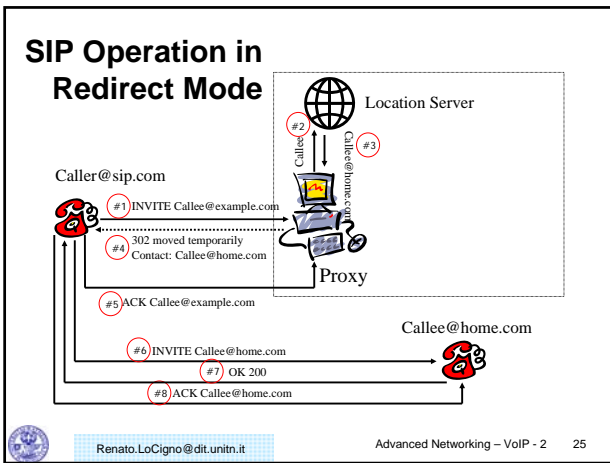
Request Method	Response Status
INVITE sip:1UserB@there.com SIP/2.0 Via: SIP/2.0/UDP here.com:5060 From: BigGuy <sip:UserA@here.com> To: LittleGuy <sip:UserB@here.com> Call-ID: 12345600@here.com CSeq: 1 INVITE Subject: Happy Christmas Contact: BigGuy <sip:UserA@here.com> Content-Type: application/sdp Content-Length: 147	SIP/2.0 200 OK Via: SIP/2.0/UDP here.com:5060 From: BigGuy <sip:UserA@here.com> To: LittleGuy <sip:UserB@here.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
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 v=0 o=UserB 2890844527 2890844527 IN IP4 here.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@example.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:, ...)

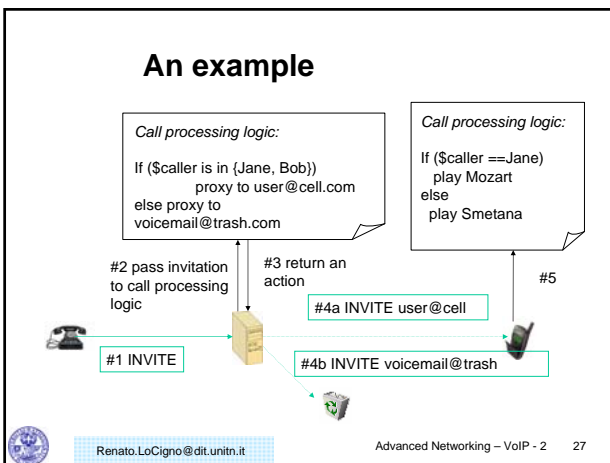
SIP Server -- Proxy versus Redirection

- 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



• Programming SIP - SIP Service Examples

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Programming SIP

- Users and third parties may program
- SIP follows HTTP programming model
- Need APIs that provide access to SIP specific constructs
 - Signaling protocol independent APIs won't work!
 - Parlay
 - TAPI/JTAPI
- Mechanisms suggested in IETF:
 - CGI
 - Call Processing Language (CPL)
 - Servlets
- If programming logic resides at external server?
 - Use SIP to get signaling to it
 - No need for "Remote Call Processing API"



SIP-CGI

- Follows Web-CGI
 - SIP-CGI supports proxying and processes responses as well.
- Language-independent (Perl, C, ...)
- Communicates through input/output and environment variables.
- CGI programs unlimited in their power
 - Drawback: Buggy scripts may affect server easily.
- Token is passed between SIP server and CGI to keep state across requests and related responses



Call Processing Language (CPL)

- Special-purpose call processing language
- May be used by both SIP and H.323 servers
- Target scenario: users determine call processing logic executed at a server
- Limited languages scope makes sure server's security will not get compromised
- Portability allows users to move CPL scripts across servers
- Manually written / generated using GUI tools / supplied by 3rd parties / ...



CPL Example

```
<incoming>
<address-switch field="origin" subfield="host">
  <address subdomain-of="example.com">
    <location url="sip:jones@example.com">
      <proxy timeout="10">
        <busy> <sub ref="voicemail" /> </busy>
        <noanswer> <sub ref="voicemail" /> </noanswer>
        <failure> <sub ref="voicemail" /> </failure>
      </proxy>
    </location>
  </address>
  <otherwise>
    <sub ref="voicemail" />
  </otherwise>
</address-switch>
</incoming>
```

- Based on XML
- Actions include: redirection, proxy, rejection
- Dynamically uploaded by Third-parties



SIP-Servlets

- Compromise between security and power:
 - a powerful generic language
 - security provided by Java "sand-box".
- SIP Servlet Benefits
 - SIP servlets process a particular SIP request or response
 - Supports: Sending responses, Proxying requests, Initiating requests
 - SIP servlets can create and access session data, call data, transaction data
 - Exposes key SIP headers, but handles most of the SIP details for you
- Inherits all scale and fault tolerance benefits of HTTP Servlets



Call Processing Tradeoffs

- Generality versus security
 - multipurpose programming languages (MPL) provide a huge service space
 - but also a huge vulnerability space
- Performance versus portability
 - portable languages need to be interpreted -> higher processing delay
 - portability needed if services deployed at multiple servers or end-devices



Generality versus Security

Generality

- CGI Highest
- Servlets Medium
- CPL Lowest

Security by

language	RT code	admin. verification	policy
CGI	x	x	✓
Servlets	x	✓	✓
CPL	✓	✓	✓



Performance and Portability

Performance

- CGIDOL*
- Servlets Medium
- CPL Lowest

Portability

- CGIDOL*
- Servlets Good
- CPL Good

- *DOL = depends on the language



• Mobility

- User mobility
- Service mobility
- 3GPP

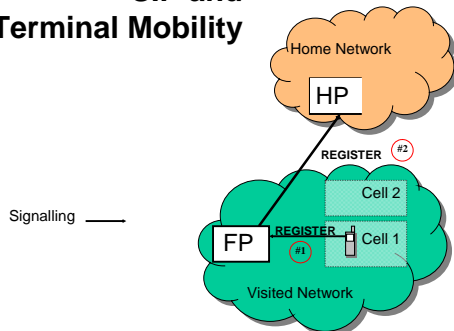


SIP and Terminal Mobility

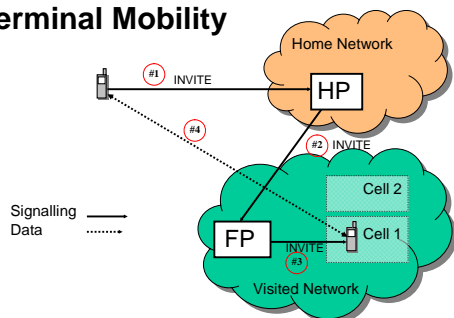
- Terminal can move between subnetworks
- Issues to consider:
 - Handoff performance
 - Redirection authentication
- Mobile hosts (MH) inform their home proxy about their new locations using REGISTER
- Mid-call mobility (Session mobility) is dealt with using reINVITE



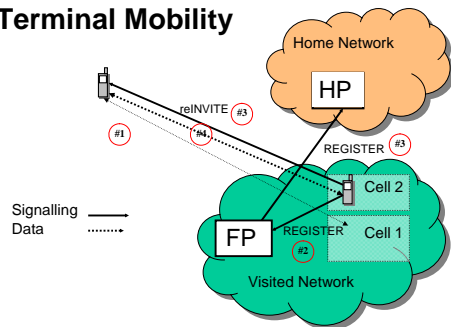
SIP and Terminal Mobility



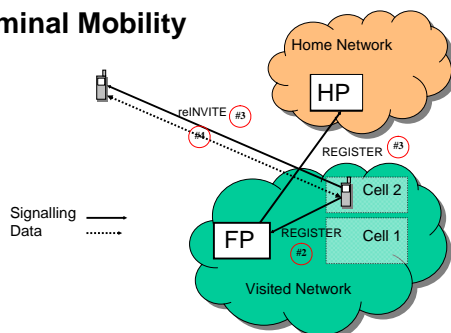
SIP and Terminal Mobility



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)



SIP & 3GPP

