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
  - RTP/RTCP (voice/video conversational transport)
  - SDP: Session Description Protocol
  - SAP: Session Announcement Protocol
  - RTSP: Real Time Streaming Protocol (VoD like)

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**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?

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**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 AgentServer
- **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 conference
  - BYE: Closes a participation
  - CANCEL: Terminates a search (unsuccessful)
  - OPTIONS: Query a client on his “capabilities”
  - ACK: Accept a call (INVITE)
  - 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
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 Caller@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
- **BYE**: terminates sessions
- **CANCEL**: cancels a pending INVITE
  - if a CANCEL follows a re-INVITE the session is 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-impp-*)
  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
- 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**  
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

**Response Status**  
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

**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**

![Diagram of SIP Operation in Redirect Mode]

1. **Caller@sip.com** sends an INVITE to **Callee@example.com**.
2. **Proxy** receives the INVITE and redirects it to **Callee@home.com**.
3. **Proxy** sends an ACK to **Callee@example.com**.
4. **Callee@home.com** sends a 302 Moved Temporarily response with the updated contact Information.
5. **Proxy** sends an INVITE to **Callee@home.com**.
6. **Callee@home.com** sends an OK 200 response.
7. **Proxy** sends an ACK to **Callee@home.com**.

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**Mobility**

- **User mobility**
- **Service mobility**
- **3GPP**

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**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.

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SIP and Terminal 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 Personal 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)