Session Initiation Protocol (SIP)

Chapter 5
Introduction

- A powerful alternative to H.323
- More flexible, simpler
- Easier to implement
  - Advanced features
- Better suited to the support of intelligent user devices
- A part of IETF multimedia data and control architecture
- SDP, RTSP (Real-Time Streaming Protocol), SAP (Session Announcement Protocol)
The Popularity of SIP

- Originally Developed in the MMUSIC (Multiparty Multimedia Session Control)
  - A separate SIP working group
  - RFC 2543
  - Many developers
  - The latest version: RFC 3261

- SIP + MGCP/MEGACO
  - The VoIP signaling in the future

- “bake-off”
  - Various vendors come together and test their products against each other
    - to ensure that they have implemented the specification correctly
    - to ensure compatibility with other implementations
**SIP Architecture**

- A signaling protocol
  - The setup, modification, and tear-down of multimedia sessions
- SIP + SDP
  - Describe the session characteristics
- Separate signaling and media streams
SIP Network Entities [1/4]

- **Clients**
  - User agent clients
  - Application programs sending SIP requests

- **Servers**
  - Responds to clients’ requests

- **Clients and servers may be in the same platform**

- **Proxy**
  - Acts as both clients and servers
SIP Network Entities [2/4]

- Four types of servers
  - Proxy servers
    - Handle requests or forward requests to other servers
    - Can be used for call forwarding, time-of-day routing, or follow-me services

```
Caller@work.com
1. Request Collins@work.com
Proxy Server
2. Response
3. Request Collins@home.net
4. Response
Collins@home.net
```
SIP Network Entities [3/4]

- Redirect servers
  - Map the destination address to zero or more new addresses
SIP Network Entities [4/4]

- A user agent server
  - Accepts SIP requests and contacts the user
  - The user responds → an SIP response
  - A SIP device
  - E.g., a SIP-enabled telephone

- A registrar
  - Accepts SIP REGISTER requests
    - Indicating that the user is at a particular address
    - Personal mobility
  - Typically combined with a proxy or redirect server
SIP Call Establishment

- It is simple, which contains a number of interim responses.
SIP Advantages

- Attempt to keep the signaling as simple as possible
- Offer a great deal of flexibility
  - Does not care what type of media is to be exchanged during a session or the type of transport to be used for the media
- Various pieces of information can be included within the messages
  - Including non-standard information
  - Enable the users to make intelligent decisions
    - The control of the intelligent features is placed in the hands of the customer, not the network operator.
  - E.g., SUBJECT header
Call Completion to Busy Subscriber Service

1. INVITE
2. Busy (Try at 4pm)
3. ACK
4. INVITE
5. Ringing
6. OK
7. ACK
8. Conversation
9. BYE
10. OK
Overview of SIP Messaging Syntax

- **Text-based**
  - Similar to HTTP
  - Disadvantage – more bandwidth consumption

- **SIP messages**
  - message = start-line
    *message-header CRLF
    [message-body]
  - start-line = request-line | status-line

- Request-line specifies the type of request
- The response line indicates the success or failure of a given request.
Message headers
- Additional information of the request or response
- E.g.,
  - The originator and recipient
  - Retry-after header
  - Subject header

Message body
- Describe the type of session
- The most common structure for the message body is SDP (Session Description Protocol).
- Could include an ISDN User Part message
- Examined only at the two ends
SIP Requests [1/2]

- **Method SP Request-URI SP SIP-version CRLF**
- **Request-URI**
  - The address of the destination
- **Methods**
  - INVITE, ACK, OPTIONS, BYE, CANCLE, REGISTER
  - INVITE
    - Initiate a session
    - Information of the calling and called parties
    - The type of media
    - ~IAM (initial address message) of ISUP
    - ACK only when receiving the final response
SIP Requests [2/2]

- **BYE**
  - Terminate a session
  - Can be issued by either the calling or called party

- **Options**
  - Query a server as to its capabilities
    - A particular type of media

- **CANCEL**
  - Terminate a pending request
  - E.g., an INVITE did not receive a final response

- **REGISTER**
  - Log in and register the address with a SIP server
  - “all SIP servers” – multicast address (224.0.1.1750)
  - Can register with multiple servers
  - Can have several registrations with one server
"One number" service

User at Address 2

User at Address 1

Registrar / Proxy

Caller

a
b
REGISTER (address 1)
OK
c
REGISTER (address 2)
OK
d
e
OK
f
INVITE
Trying
gh
INVITE
g
OK
h
INVITE
i
OK
j
CANCEL
jk
OK (for CANCEL)
jl
OK (for INVITE)
km
l
n
Conversation

net Telephony  16
SIP INFO Method

- Specified in RFC 2976
  - For transferring information during an ongoing session
- The transfer of DTMF digits
- The transfer of account balance information
  - Pre-paid service
- The transfer of mid-call signaling information
SIP Responses

- SIP Version SP Status Code SP Reason-Phrase CRLF
- Reason-Phrase
  - A textual description of the outcome
  - Could be presented to the user
- status code
  - A three-digit number
  - 1XX Informational
  - 2XX Success (only code 200 is defined)
  - 3XX Redirection
  - 4XX Request Failure
  - 5XX Server Failure
  - 6XX Global Failure
  - All responses, except for 1XX, are considered final
    - Should be ACKed
SIP Addressing

- SIP URLs (Uniform Resource Locators)
  - user@host
  - sip:collins@home.net
  - sip:3344556789@telco.net
Message Headers

- Provide further information about the message
- E.g.,
  - To:header in an INVITE
    - The called party
  - From:header
    - The calling party
- Four main categories
  - General, Request, Response, and Entity headers
General Headers

- Used in both requests and responses
- Basic information
  - E.g., To:, From:, Call-ID: (uniquely identifies a specific invitation to a session), ...
- Contact:
  - Provides a URL for use in future communication regarding a particular session
  - Examples 1: In a SIP INVITE, the Contact header might be different from the From header.
    - An third-party administrator initiates a multiparty session.
  - Example 2: Used in response, it is useful for directing further requests directly to the called user.
  - Example 3: It is used to indicate a more appropriate address if an INVITE issued to a given URI failed to reach the user.
Request Headers
- Apply only to SIP requests
- Additional information about the request or the client
- E.g.,
  - Subject:
  - Priority:, urgency of the request (emergency, urgent, normal, or non-urgent)

Response Headers
- Further information about the response that cannot be included in the status line
- E.g.,
  - Unsupported
  - Retry-After
Entity Headers

- Indicate the type and format of information included in the message body
- Content-Length: the length of the message body
- Content-Type: the media type of the message body
  - E.g., application/sdp
- Content-Encoding: for message compression
- Content Disposition: how a message part should be interpreted
  - session, alert …
Examples of SIP Message Sequences

- **Registration**
  - Via:
  - From: and To:
  - Call-ID:
    - host-specific
  - Contact: (for future SIP message transmission)
    - *
  - Content-Length:
    - Zero, no msg body
  - CSeq:
    - A response to any request must use the same value of CSeq as used in the request.
  - Expires:
    - TTL
    - 0, unreg
Invitation

- A two-party call
  - Subject: optional
  - Content-Type: application/sdp
- A dialog ID
  - To identify a peer-to-peer relationship between two user agents
  - Tag in From
  - Tag in To
  - Call-ID
Termination of a Call

- **Cseq:**
  - Has changed

```
Daniel<sip:Collins@work.com>

a
BYE sip:manager@work.com SIP/2.0
Via: SIP/2.0/UDP station1.work.com;branch=z9hG4bK123
Max-Forwards: 70
From: Daniel<sip:Collins@work.com>; tag=44551
To: Boss<sip:Manager@station2.work.com>; tag=11222
Call-ID: 123456@station1.work.com
CSeq: 2 BYE
Content-Length: 0

b
SIP/2.0 200 OK
Via: SIP/2.0/UDP station1.work.com;branch=z9hG4bK123
From: Daniel<sip:Collins@work.com>; tag=44551
To: Boss<sip:Manager@station2.work.com>; tag=11222
Call-ID: 123456@station1.work.com
CSeq: 2 BYE
Content-Length: 0
```
Redirect Servers

- An alternative address
  - 302, Moved temporarily
- Another INVITE
  - Same Call-ID
  - CSeq ++
Proxy Servers

- Sits between a user-agent client and the far-end user-agent server
- Numerous proxies can reside in a chain between the caller and callee.
- The last proxy may change the Request-URI.
- **Via:**
  - The path taken by a request
  - Loop detected, 482 (status code)
  - For a response
    - The 1st Via: header
    - Checked
    - Removed
  - Branch: used to distinguish between multiple responses to the same request
    - Forking Proxy: Issue a single request to multiple destinations
Proxy state

- Can be either stateless or stateful
- Record-Route:
  - The messages and responses may not pass through the same proxy
    - Use Contact:
  - A Proxy might require that it remains in the signaling path
    - In particular, for a stateful proxy
  - Insert its address into the Record-Route: header
  - The response includes the Record-Route: header
  - The information contained in the Record-Route: header is used in the subsequent requests related to the same call.
  - The Route: header = the Record-Route: header in reverse order
Forking Proxy

- “fork” requests
- A user is registered at several locations
  - ;branch=xxx
- In order to handle such forking, a proxy must be stateful.