Session Initiation Protocol (SIP)
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 (June 2002)

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

- “bake-offs” or SIP Interoperability Tests
  - The development of SIP and its implementation by system developers has involved a number of events.
  - 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 to potential session participants
- Separate signaling and media streams
  - Signaling may pass via one or more proxy or redirect servers
  - Media stream takes a more direct path.
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
    - Act in a similar way to a proxy server used for web access
    - Handle requests or forward requests to other servers after some translation
    - Can be used for call forwarding, time-of-day routing, or follow-me services

```plaintext
1. Request
   Collins@work.com
2. Request
   Collins@home.net
3. Response
4. Response
```

Caller@work.com

SIP Proxy

Collins@home.net
SIP Network Entities [3/4]

- Redirect servers
  - Accept SIP requests
  - Map the destination address to zero or more new addresses
  - Return the new address(es) to the originator of the request

1. Request
   Collins@work.com

2. Moved temporarily
   Contact: Collins@home.net

3. ACK

4. Request
   Collins@home.net

5. Response

Redirect Server
SIP Network Entities [4/4]

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

- A registrar (location server)
  - 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

- A SIP call establishment is simple.
- A number of interim responses may be made to the INVITE prior to the called party accepting the call.

```
+-------------------+          +-------------------+          
|                   |          |                   |          
| INVITE            |          | Ringing           |          
|                   |          |                   |          
| a                 |          | OK                |          
|                   |          |                   |          
| b                 |          | ACK               |          
|                   |          |                   |          
| c                 |          | Conversation      |          
|                   |          |                   |          
| d                 |          |                   |          
|                   |          |                   |          
| e                 |          |                   |          
|                   |          |                   |          
| f                 |          | BYE               |          
|                   |          |                   |          
| g                 |          | OK                |          
|                   |          |                   |          
```
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
    - Text-based encoding
  - 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

INVITE

Busy (Try at 4pm)

ACK

INVITE

Ringing

OK

ACK

Conversation

BYE

OK
Overview of SIP Messaging Syntax

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

- **SIP messages**
  - \texttt{message} = \texttt{start-line}
    
    *message-header CRLF
    
    [message-body]
  - \texttt{start-line} = \texttt{request-line} | \texttt{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 SIP 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
    - To support a particular type of media

- **CANCEL**
  - Terminate a pending request
  - Pending Request: an INVITE did not receive a final response

- **REGISTER**
  - Log in and register the address with a SIP server
  - “all SIP servers” – multicast address (224.0.1.175)
  - Can register with multiple servers
  - Can have several registrations with one server
“One Number” Service
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
- Addition 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, render …
Examples of SIP Message Sequences

- **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>  Boss<sip:Manager@station2.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 [1/2]

- 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 most common scenario will have at least two proxies: one at the caller and one at the callee end.
  - It is likely that only the last proxy in the chain changes the Request-URI.
  - The other proxies in the chain would simply use the domain part of the received Request-URI as input to a location function (e.g., DNS) to determine the next hop.
Proxy Servers [2/2]

- **Via:**
  - The path taken by a request
  - Loop detected, 482 (status code)
  - For a response
    - The 1st Via: header is checked and removed.
    - The second Via: header is checked.
      - If it exists, perform forwarding.
      - If not, the response is destined to the proxy itself.
    - The response finds its way back to the originator of the request.
  - **Branch:** used to distinguish between multiple responses to the same request
    - Forking Proxy: Issue a single request to multiple destinations
Proxy State [1/2]

- Can be either stateless or stateful
- If stateless, the proxy takes an incoming request, performs whatever translation and forwards the corresponding outgoing request and forgets anything.
  - Retransmission takes the same path (no change on retransmission).
- If stateful, the proxy remembers incoming requests and corresponding outgoing request.
  - The proxy is able to act more intelligently on subsequent requests and responses related to the same session.
Proxy State [2/2]

- **Record-Route: and Route: Headers**
  - The subsequent requests may not pass through the same path as the initial request/response.
    - E.g., use Contact:
  - A Proxy might require that it remains in the signaling path for all subsequent requests to provide some advanced service.
    - 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 is used to record the path that the request is enforced to pass.
  - lr (loose routing) vs. sr (strict routing)
e  
SIP/2.0 200 OK
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK7890
Record-route: <sip:server.work.com;l>; tag=ab12
From: Boss <sip:Manager@home.net>; tag=ab12
To: Daniel <sip:Collins@work.com>; tag=xyz45
Call-ID: 123456@pc1.home.net
CSeq: 1 INVITE
Contact: sip:Collins@station1.work.com

f  
ACK sip:Collins@station1.work.com SIP/2.0
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK7891
Max-Forwards: 70
Route: <sip:server.work.com;l>; tag=ab12
From: Boss <sip:Manager@home.net>; tag=ab12
To: Daniel <sip:Collins@work.com>; tag=xyz45
Call-ID: 123456@pc1.home.net
CSeq: 1 ACK

ACK sip:Collins@station1.work.com SIP/2.0
Via: SIP/2.0/UDP server.work.com; branch=z9hG4bKxyz2
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK7891
Max-Forwards: 69
From: Boss <sip:Manager@home.net>; tag=ab12
To: Daniel <sip:Collins@work.com>; tag=xyz45
Call-ID: 123456@pc1.home.net
CSeq: 1 ACK
Forking Proxy

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