Session Initiation Protocol (SIP)
SIP Extensions and Enhancements

- RFC 2543, March 1999
  - RFC 3261, June 2002
  - SIP has attracted enormous interest.
  - Traditional telecommunications companies, cable TV providers and ISP

- A large number of extensions to SIP have been proposed.
  - SIP will be enhanced considerably before it becomes an Internet standard.
183 Session Progress

- It has been included within the revised SIP spec.
  - To open one-way audio path from called end to calling end
    - Enable in-band call progress information to be transmitted
      - Tones or announcements
  - Interworking with SS7 network
    - ACM (Address Complete Message)
    - For SIP-PSTN-SIP connections
The Supported Header

- The Base RFC 2543
  - The Require: Header
    - In request (client -> server)
      - A client indicates that a server must support certain extension.
  - The Unsupported Header
    - In response (server -> client)
      - 420 (bad extension)
    - A cumbersome way of determining what extensions a server does or does not support

- The Supported: Header (RFC 3261)
  - May be included in OPTIONS request
    - Associated with the Supported: header is 421 (extension required) response.
  - Can also be included in responses
SIP INFO Method

- Be specified in RFC 2976
- For transferring information during an ongoing session
  - DTMF digits, account-balance information, mid-call signaling information (from PSTN)
  - Application-layer information could be transferred in the middle of a call.
- A powerful, flexible tool to support new services
Several SIP-based applications have been devised based on the concept of a user being informed of some event.

- E.g., Instant messaging
- RFC 3265 has addressed the issue of event notification.
  - SUBSCRIBE and NOTIFY
  - The Event header
SIP for Instant Messaging

- The IETF working group – SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE)

- A new SIP method – MESSAGE
  - This request carries the actual message in a message body.
  - A MESSAGE request does not establish a SIP dialog.
MESSAGE sip:Collins@work.com SIP/2.0
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK7890
Max-Forwards: 70
From: Boss<sip:Manager@home.net>
To: Daniel<sip:Collins@work.com>
Call-ID: 123456@pc1.home.net
CSeq: 1 MESSAGE
Content-Type: text/plain
Content-Length: 19
Content-Disposition: render

Hello. How are you?

MESSAGE sip:Collins@work.com SIP/2.0
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bKxyz1
Via: SIP/2.0/UDP server.work.com; branch=z9hG4bK7890
Max-Forwards: 69
From: Boss<sip:Manager@home.net>
To: Daniel<sip:Collins@work.com>
Call-ID: 123456@pc1.home.net
CSeq: 1 MESSAGE
Content-Type: text/plain
Content-Length: 19
Content-Disposition: render

Hello. How are you?

SIP/2.0 200 OK
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK7890
From: Boss<sip:Manager@home.net>
To: Daniel<sip:Collins@work.com>
Call-ID: 123456@pc1.home.net
CSeq: 1 MESSAGE
Content-Length: 0

SIP/2.0 200 OK
Via: SIP/2.0/UDP server.work.com;
From: Boss<sip:Manager@home.net>
To: Daniel<sip:Collins@work.com>
Call-ID: 123456@pc1.home.net
CSeq: 1 MESSAGE
Content-Length: 0
MESSAGE sip:Manager@home.net SIP/2.0
Via: SIP/2.0/UDP server.work.com; branch=z9hG4bKabcd
Via: SIP/2.0/UDP station1.work.com; branch=z9hG4bK123
Max-Forwards: 69
From: Daniel<sip:Collins@work.com>
To: Boss<sip:Manager@home.net>
Call-ID: 456789@station1.work.com
CSeq: 1101 MESSAGE
Content-Type: text/plain
Content-Length: 22
Content-Disposition: render
I'm fine. How are you?

SIP/2.0 200 OK
Via: SIP/2.0/UDP server.work.com; branch=z9hG4bKabcd
Via: SIP/2.0/UDP station1.work.com; branch=z9hG4bK123
From: Daniel<sip:Collins@work.com>
To: Boss<sip:Manager@home.net>
Call-ID: 456789@station1.work.com
CSeq: 1101 MESSAGE
Content-Length: 0
SIP REFER Method

- To enable the sender of the request to instruct the receiver to contact a third party
  - With the contact details for the third party included within the REFER request
  - For Call Transfer applications
- The Refer-to: and Refer-by: Headers
- The dialog between Mary and Joe remains established.
  - Joe could return to the dialog after consultation with Susan.
REFER sip:Joe@station2.work.com SIP/2.0
Via: SIP/2.0/UDP station1.work.com; branch=z9hG4bK789
Max-Forwards: 70
From: Mary<sip:Mary@work.com>; tag=123456
To: Joe<sip:Joe@work.com>; tag=67890
Contact: Mary<Mary@station1.work.com>
Refer-To: Susan<sip:Susan@station3.work.com>
Call-ID: 123456@station1.work.com
CSeq: 123 REFER
Content-Length: 0

INVITE sip:Susan@station3.work.com SIP/2.0
Via: SIP/2.0/UDP station2.work.com; branch=z9hG4bKxyz1
Max-Forwards: 70
From: Joe<sip:Joe@work.com>; tag=abcxyz
To: Susan<sip:Susan@station3.work.com>
Contact: Joe<Joe@station2.work.com>
Call-ID: 67890@station2.work.com
CSeq: 567 INVITE
Content-Type: application/sdp
Content-Length: xx
Content-Disposition: session
{message body}
sip:Mary@station1.work.com  sip:Joe@station2.work.com  sip:Susan@station3.work.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP station2.work.com; branch=z9hG4bk123
From: Joe@sip:Joe@work.com; tag=abcxyz
To: Susan@sip:Susan@station3.work.com; tag=123xyz
Call-ID: 67890@station2.work.com
CSeq: 567 INVITE
Content-Type: application/sdp
Content-Length: xx
Content-Disposition: session
{message body}

ACK sip:Susan@station3.work.com SIP/2.0
Via: SIP/2.0/UDP station2.work.com; branch=z9hG4bk123
Max-Forwards: 70
From: Joe@sip:Joe@work.com; tag=abcxyz
To: Susan@sip:Susan@station3.work.com; tag=123xyz
Call-ID: 67890@station2.work.com
CSeq: 567 ACK
Content-Length: 0

NOTIFY sip:Mary@station1.work.com SIP/2.0
Via: SIP/2.0/UDP station2.work.com; branch=z9hG4bK123
Max-Forwards: 70
To: Joe@sip:Joe@work.com; tag=67890
From: Mary@sip:Mary@work.com; tag=123456
Contact: Joe@sip:Joe@station2.work.com
Call-ID: 123456@station1.work.com
CSeq: 124 NOTIFY
Content-Type: message/sipfrag;version=2.0
Content-Length: 15

SIP/2.0 200 OK
Via: SIP/2.0/UDP station2.work.com; branch=z9hG4bk123
To: Joe@sip:Joe@work.com; tag=67890
From: Mary@sip:Mary@work.com; tag=123456
Call-ID: 123456@station1.work.com
CSeq: 124 NOTIFY
Content-Length: 0

IP Telephony 12
Reliability of Provisional Responses [1/2]

- **Provisional Responses**
  - 100 (trying), 180 (ringing), 183 (session in progress)
  - Are not answered with an ACK

- If the messages is sent over UDP
  - Unreliable

- Lost provisional response may cause problems when interoperating with other network
  - 180, 183 → Q.931 alerting or ISUP ACM
  - To drive a state machine
  - E.g., a call to an unassigned number
    - ACM to create a one-way path to relay an announcement such as “The number you have called has been changed”
    - If the provisional response is lost, the called might left in the dark and not understand why the call did not connect.
Reliability of Provisional Responses [2/2]

- RFC 3262
  - Reliability of Provisional Responses in SIP
- Supported: 100rel
- RSeq Header
  - Response Seq
  - +1, when retrans
- RAck Header
  - Response ACK
  - In PRACK
  - RSeq+CSeq
- PRACK
  - Prov. Resp. ACK
- Should not
  - Apply to 100
- Default timer value = 0.5 s
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ClientA.network.com; branch=z9hG4bK7890123
Require: 100rel
RSeq: 667891
From: sip:ClientA@network.com; tag=lmnop123
To: sip:ServerB@network.com; tag=xyz123
Call-ID: 123456@ClientA.network.com
CSseq: 1 INVITE

PRACK sip:ServerB@network.com SIP/2.0
Via: SIP/2.0/UDP ClientA.network.com; branch=z9hG4bK7890123
R Ack: 667891 1 INVITE
From: sip:ClientA@network.com; tag=lmnop123
To: sip:ServerB@network.com; tag=xyz123
Call-ID: 123456@ClientA.network.com
CS seq: 2 PRACK

SIP/2.0 200 OK
Via: SIP/2.0/UDP ClientA.network.com; branch=z9hG4bK7890123
From: sip:ClientA@network.com; tag=lmnop123
To: sip:ServerB@network.com; tag=xyz123
Call-ID: 123456@ClientA.network.com
CSeq: 2 PRACK
The SIP UPDATE Method

- To enable the modification of session information before a final response to an INVITE is received
  - The dialog is in the early state (An INVITE that receives a 183 response that includes a message body)
    - The message body might establish a media stream from callee to caller for sending a ring tone or music while the called party is alerted.
  - The UPDATE method can be used to change the codec
- Another important usage is when reserving network resources as part of a SIP session establishment.
Integration of SIP Signaling and Resource Management [1/2]

- Ensuring that sufficient resources are available to handle a media stream is very important.
  - To provide a high-quality service for a carrier-grade network
- The signaling might take a different path from the media.
  - The successful transfer of signaling messages does not imply to a successful transfer of media.
- Assume resource-reservation mechanisms are available (Chapter 8)
  - On a per-session basis
    - End-to-end network resources are reserved as part of session establishment.
  - On an aggregate basis
    - A certain amount of network resources are reserved in advance for a certain type of usage.
    - Policing functions at the edge of the network
Integration of SIP Signaling and Resource Management [2/2]

- Reserving network resources in advance of altering the called user
- A new draft – “Integration of Resource Management and SIP”
  - By using the provisional responses and UPDATE method
  - By involving extensions to SDP
Example of e2e Resource Reservation [1/2]

- **SDP for initial INVITE**
  
  ```markdown
  v=0
  o=userA 45678 001 IN IP4 stationA.network.com
  s=
  c=IN IP4 stationA.nework.com
  t=0 0
  m=audio 4444 RTP/AVP 0
  a=curr: qos e2e none
  a=des: qos mandatory e2e sendrecv
  ```

- **SDP for 183 response**
  
  ```markdown
  v=0
  o=userB 12345 001 IN IP4 stationB.network.com
  s=
  c=IN IP4 stationB.nework.com
  t=0 0
  m=audio 6666 RTP/AVP 0
  a=curr: qos e2e none
  a=des: qos mandatory e2e sendrecv
  a=conf: qos e2e recv
  ```
Example of e2e Resource Reservation [2/2]

- **SDP for UPDATE**
  
  ```
  v=0
  o=userA 45678 001 IN IP4 stationA.network.com
  s=
  c=IN IP4 stationA.nework.com
  t=0 0
  m=audio 4444 RTP/AVP 0
  a=curr: qos e2e send
  a=des: qos mandatory e2e sendrecv
  ```

- **SDP for 200 response**
  
  ```
  v=0
  o=userB 12345 001 IN IP4 stationB.network.com
  s=
  c=IN IP4 stationB.nework.com
  t=0 0
  m=audio 6666 RTP/AVP 0
  a=curr: qos e2e sendrecv
  a=des: qos mandatory e2e sendrecv
  ```
Example of Aggregate-based Reservation

- Each participant deals with network access permission at its own end.
- **Mandatory** means that the session can not continue unless the required resources are definitely available.
- **None** is the initial situation and indicates that no effort to reserve resources has yet taken place.
- **Response 580** (precondition failure)
Usage of SIP for Features/Services [1/2]

- Call-transfer application (with REFER method)
- Personal Mobility through the use of registration
- One number service through forking proxy
- Call-completion services by using Retry-After: header
- To carry MIME content as well as an SDP description
  - To include a piece of text, an HTML document, an image and so on
- SIP address is a URL
  - Click-to-call applications
- The existing supplementary services in traditional telephony
  - Call waiting, call forwarding, multi-party calling, call screening
Usage of SIP for Features/Services [2/2]

- Proxy invokes various types of advanced feature logic.
  - Policy server (call-routing, QoS)
  - Authentication server
  - Use the services of an IN SCP over INAP
- The network might use the Parley Open Service Access (OSA) approach, utilizing application programming interfaces (APIs) between the nodes.
Call Forwarding

- On busy
- 486, busy here
- With the same To, User 3 can recognize that this call is a forwarded call, originally sent to User 2.
- Contact: header in 200 response
- Call-forwarding-on-no-answer
  - Timeout
  - CANCEL method
Consultation Hold

- A SIP UPDATE
- User A asks User B a question, and User B needs to check with User C for the correct answer.
- If User C needs to talk to User A directly, User B could use the REFER method to transfer the call to User C.
PSTN Interworking

- PSTN Interworking
  - A SIP URL to a telephone number
  - A network gateway

- Seamless interworking between two different protocols is not quite easy.
  - One-to-one mapping between these protocols

- PSTN – SIP – PSTN
  - MIME media types
  - For ISUP
  - SIP for Telephony (SIP-T)

- The whole issue of interworking with SS7 is fundamental to the success of VoIP in the real world.
Interworking with H.323

- SIP-H.323 interworking gateway
INVITE
To: Terminal@H323.net

c=IN IP4 123.45.6.7
m=audio 8000 RTP/AVP 0

Setup
faststart [logical chan info = G711 Tx,
G711 RX 123.45.6.7:8000]

Alerting

180 (Ringing)

200 (OK)
To: Terminal@H323.net

c=IN IP4 123.67.8.9
m=audio 2000 RTP/AVP 0

Connect
faststart [logical chan info = G711 Tx,
G711 RX 123.67.8.9:2000]

Two-way voice
INVITE
To: Terminal@H323.net

Setup
faststart [logical chan info = G711 TX, G711 RX 123.45.6.7:8000]

Connect (no faststart)

Terminal Capability Set
G711 TX, G711 RX

Terminal Capability Set Ack

Terminal Capability Set
G711 TX, G711 RX

Terminal Capability Set Ack

Open Logical Channel

Open Logical Channel Ack
RX=123.45.6.7:8000

200 (OK)
To: Terminal@H323.net

c=IN IP4 123.45.6.7
m=audio 8000 RTP/AVP 0

c=IN IP4 123.45.6.7
m=audio 2000 RTP/AVP 0

ACK

Open Logical Channel

Open Logical Channel Ack
RX=123.45.6.7:8000

H.225.0
H.245
SIP
H.323 Terminal

Setup
faststart [logical chan info = G711 TX,
G711 RX 123.67.8.9:2000]

Gateway

INVITE
To: User@SIP.net

c=IN IP4 123.67.8.9
m=audio 2000 RTP/AVP 0

180 (Ringing)

200 (OK)
To: User@SIP.net

c=IN IP4 123.45.6.7
m=audio 8000 RTP/AVP 0

ACK

SIP Client

Alerting

Connect
faststart [logical chan info - G711 TX,
G711 RX 123.45.6.7:800]

Two-way voice

Two-way voice
Summary

- The future for signaling in VoIP networks
  - Simple, yet flexible
  - Easier to implement
  - Fit well with the media gateway control protocols
    - Coexisting with PSTN

- SIP is the protocol of choice for the evolution of third-generation wireless networks.
  - SIP-based mobile devices will become available.
  - SIP-based network elements will be introduced within mobile networks.