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

---

**a)**
```
REGISTER sip:registrar.work.com SIP/2.0
Via: SIP/2.0/UDP station1.work.com
Max-Forwards: 70
From: sip:Collins@work.com
To: sip:Collins@work.com
Call-ID: 123456@station1.work.com
CSeq: 1 REGISTER
Contact: sip:Collins@station1.work.com
Expires: 7200
Content-Length: 0
```

---

**b)**
```
SIP/2.0 200 OK
Via: SIP/2.0/UDP station1.work.com
From: sip:Collins@work.com
To: sip:Collins@work.com
Call-ID: 123456@station1.work.com
CSeq: 1 REGISTER
Contact: sip:Collins@station1.work.com
Expires: 3600
Content-Length: 0
```
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 first 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.
a) **INVITE** 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>; tag=ab12
   Contact: Boss<sip:manager@pc1.home.net>
   To: Daniel<sip:Collins@work.com>
   Call-ID: 123456@pc1.home.net
   CSeq: 1 INVITE

b) **INVITE** sip:Collins@station1.work.com SIP/2.0
   Via: SIP/2.0/UDP server.work.com;branch=z9hG4bKxyz1
   Via: SIP/2.0/UDP pc1.home.net;branch=z9hG4bK7890
   Max-Forwards: 69
   Record-coute: <sip:server.work.com>
   From: Boss<sip:Manager@home.net>; tag=ab12
   Contact: Boss<sip:manager@pc1.work.com>
   To: Daniel<sip:Collins@work.com>
   Call-ID: 123456@pc1.home.net
   CSeq: 1 INVITE

c) **SIP/2.0 100 Trying**
   Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK7890
   From: Boss<sip:Manager@home.net>; tag=ab12
   To: Daniel<sip:Collins@work.com>
   Call-ID: 123456@pc1.home.net
   CSeq: 1 INVITE

d) **SIP/2.0 200 OK**
   Via: SIP/2.0/UDP server.work.com;branch=z9hG4bKxyz1
   Via: SIP/2.0/UDP pc1.home.net;branch=z9hG4bK7890
   Record-coute: <sip:server.work.com>
   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

e) **SIP/2.0 200 OK**
   ...
SIP/2.0 200 OK
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK7890
Record-route: <sip:server.work.com>
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

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>
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.
a. INVITE sip:Collins@work.com SIP/2.0
   Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK789
   Max-Forwards: 70
   From: Boss<sip:Manager@home.net>; tag=ab12
   Contact: Boss<sip:manager@pc1.home.net>
   To: Daniel@sip:Collins@work.com>
   Call-ID: 123456@pc1.home.net
   CSeq: 1 INVITE

b. SIP/2.0 100 Trying
   Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK789
   From: Boss<sip:Manager@home.net>; tag=ab12
   To: Daniel<sip:Collins@work.com>
   Call-ID: 123456@pc1.home.net
   CSeq: 1 INVITE

b. INVITE sip:Collins@pc1.work.com SIP/2.0
   Via: SIP/2.0/UDP server.work.com; branch=z9hG4bK123
   Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK789
   Max-Forwards: 69
   Record-coute; <sip:server.work.com;lr>
   From: Boss<sip:Manager@home.net>; tag=ab12
   Contact: Boss<sip:manager@pc1.work.com>
   To: Daniel<sip:Collins@work.com>
   Call-ID: 123456@pc1.home.net
   CSeq: 1 INVITE

d. INVITE sip:Collins@pc2.work.com SIP/2.0
   Via: SIP/2.0/UDP server.work.com; branch=z9hG4bK456
   Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK789
   Max-Forwards: 69
   Record-coute; <sip:server.work.com;lr>
   From: Boss<sip:Manager@home.net>; tag=ab12
   Contact: Boss<sip:manager@pc1.work.com>
   To: Daniel<sip:Collins@work.com>
   Call-ID: 123456@pc1.home.net
   CSeq: 1 INVITE
f
SIP/2.0 200 OK
Via: SIP/2.0/UDP pc1.home.net;branch=z9hG4bK456
Record-route: <sip:server.work.com;l>
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@pc2.work.com

g
CANCEL sip:Collins@pc1.work.com SIP/2.0
Via: SIP/2.0/UDP server.work.com;branch=z9hG4bK456
Max-Forwards: 69
Record-route: <sip:server.work.com;l>
From: Boss<sip:Manager@home.net>; tag=ab12
Contact: Boss<sip:manager@pc1.work.com>
To: Daniel<sip:Collins@work.com>
Call-ID: 123456@pc1.home.net
CSeq: 1 CANCEL
The Session Description Protocol

- The Most Common Message Body
  - Session information describing the media to be exchanged between the parties
  - SDP, RFC 2327 (initial publication)
    - A number of modifications to the protocol have been suggested.

- SIP uses SDP in an answer/offer mode.
  - An agreement between the two parties as to the types of media they are willing to share
  - RFC 3264 (An Offer/Answer Model with SDP)
    - To describe how SDP and SIP should be used together
The Structure of SDP

- SDP simply provides a format for describing session information to potential session participants.
- Text-based Protocol
- The Structure of SDP
  - Session Level Info
    - Name of the session
    - Originator of the session
    - Time that the session is to be active
  - Media Level Info
    - Media type
    - Port number
    - Transport protocol
    - Media format
SDP Syntax

- A number of lines of text
- In each line
  - field=value
  - field is exactly one character (case-significant)
- Session-level fields
- Media-level fields
  - Begin with media description field (m=)
Mandatory Fields

- v=(protocol version)
- o=(session origin or creator)
- s=(session name), a text string
  - For multicast conference
- t=(time of the session), the start time and stop time
  - For pre-arranged multicast conference
- m=(media)
  - Media type
  - The transport port
  - The transport protocol
  - The media format (typically an RTP payload format)
Optional Fields [1/3]

- Some optional fields can be applied at both session and media levels.
  - The value applied at the media level overrides that at the session level

- i=(session information)
  - A text description
  - At both session and media levels
  - It would be somewhat superfluous since SIP already supports the Subject header.

- u=(URI of description)
  - Where further session information can be obtained
  - Only at session level
Optional Fields [2/3]

- e=(e-mail address)
  - Who is responsible for the session
  - Only at the session level
- p=(phone number)
  - Only at the session level
- c=(connection information)
  - Network type, address type and connection address
  - At session or media level
- b=(bandwidth information)
  - In kilobits per second
  - At session or media level
Optional Fields [3/3]

- **r=(repeat times)**
  - For regularly scheduled session a session is to be repeated
  - How often and how many times

- **z=(timezone adjustments)**
  - For regularly scheduled session
  - Standard time and daylight savings time

- **k=(encryption key)**
  - An encryption key or a mechanism to obtain it for the purposes of encrypting and decrypting the media
  - At session or media level

- **a=(attributes)**
  - Describe additional attributes
Ordering of Fields

- **Session Level**
  - Protocol version (v)
  - Origin (o)
  - Session name (s)
  - Session information (i)
  - URI (u)
  - E-mail address (e)
  - Phone number (p)
  - Connection info (c)
  - Bandwidth info (b)
  - Time description (t)
  - Repeat info (r)
  - Time zone adjustments (z)
  - Encryption key (k)
  - Attributes (a)

- **Media level**
  - Media description (m)
  - Media info (i)
  - Connection info (c)
    - Optional if specified at the session level
  - Bandwidth info (b)
  - Encryption key (k)
  - Attributes (a)
Subfields [1/3]

- **Field** = <value of subfield1> <value of subfield2> <value of subfield3>

- **Origin**
  - **Username**, the originator’s login id or “-”
  - **Session ID**
    - A unique ID
    - Make use of NTP timestamp
  - **Version**, a version number for this particular session
  - **Network type**
    - A text string
    - IN refers to Internet
  - **Address type**
    - IP4, IP6
  - **Address**, a fully-qualified domain name or the IP address
Subfields [2/3]

- **Connection Data**
  - The network and address at which media data will be received
  - Network type
  - Address type
  - Connection address

- **Media Information**
  - Media type
    - Audio, video, data, or control
  - Port
  - Format
    - List the various types of media format that can be supported
    - According to the RTP audio/video profile
  - m= audio 45678 RTP/AVP 15 3 0
    - G.728, GSM, G.711
Attributes

- To enable additional information to be included
- Property attribute
  - a=sendonly
  - a=recvonly
- Value attribute
  - a=orient: landscape used in a shared whiteboard session
- Rtpmap attribute
  - The use of dynamic payload type
  - a=rtpmap: <payload type> <encoding name>/ <clock rate> [/<encoding parameters>].
  - m=video 54678 RTP/AVP 98
  - a=rtpmap 98 L16/16000/2
    - 16-bit linear encoded stereo (2 channels) audio sampled at 16kHz
Usage of SDP with SIP

- SIP and SDP make a wonderful partnership for the transmission of session information.
- SIP provides the messaging mechanism for the establishment of multimedia sessions.
- SDP provides a structured language for describing the sessions.
  - The entity headers identifies the message body.
SIP Inclusion in SIP Messages

- Fig 5-15
  - G.728 is selected

- INVITE with multiple media streams
  - Unsupported should also be returned with a port number of zero

- An alternative
  - INVITE
    - m=audio 4444 RTP/AVP 2 4 15
    - a=rtpmap 2 G726-32/8000
    - a=rtpmap 4 G723/8000
    - a=rtpmap 15 G728/8000
  - 200 OK
    - m=audio 6666 RTP/AVP 15
    - a=rtpmap 15 G728/8000
INVITE sip:Manager@station2.work.com SIP/2.0
From: Daniel<sip:Collins@station1.work.com>; tag = abcd1234
To: Boss<sip:Manager@station2.work.com>
CSeq: 1 INVITE
Content-Length: 213
Content-Type: application/sdp
Content-Disposition: session

v=0
o=collins 123456 001 IN IP4 station1.work.com
s=
c=INET IP4 station1.work.com
t=0 0
m=audio 4444 RTP/AVP 2
a=rtpmap 2 G726-32/8000
m=audio 4666 RTP/AVP 4
a=rtpmap 4 G723/8000
m=audio 4888 RTP/AVP 15
a=rtpmap 15 G728/8000

SIP/2.0 200 OK
...
Daniel<sip:Collins@station1.work.com> > Boss<sip:Manager@station2.work.com>

SIP/2.0 200 OK
From: Daniel<sip:Collins@station1.work.com>; tag = abcd1234
To: Boss<sip:Manager@station2.work.com>; tag = xyz789
CSeq: 1 INVITE
Content-Length: 163
Content-Type: application/sdp
Content-Disposition: session

v=0
o=collins 45678 001 IN IP4 station2.work.com
s=
c=IN IP4 station2.work.com
t=0 0
m=audio 0 RTP/AVP 2
m=audio 0 RTP/AVP 4
m=audio 6666 RTP/AVP 15
a=rtpmap 15 G728/8000

ACK sip:manager@station2.work.com SIP/2.0
From: Daniel<sip:Collins@station1.work.com>; tag = abcd1234
To: Boss<sip:Manager@station2.work.com>; tag = xyz789
CSeq: 1 ACK
Content-Length: 0

Conversation
SIP and SDP Offer/Answer Model

- Re-INVITE is issued when the server replies with more than one codec.
  - With the same dialog identifier (To and From headers, including tag values), Call-ID and Request-URI
  - The session version is increased by 1 in o= line of message body.

- A mismatch
  - 488 or 606
  - Not Acceptable
  - A Warning header with warning code 304 (media type not available) or 305 (incompatible media type)
  - Then the caller issues a new INVITE request.
INVITE sip:manager@station2.work.com SIP/2.0
CSeq: 1 INVITE
Content-Length: 183
Content-Type: application/sdp
Content-Disposition: session
v=0
o=collins 123456 001 IN IP4 station1.work.com
s=
c=IN IP4 station1.work.com
t=0 0
m=audio 4444 RTP/AVP 2 4 15
a=rtpmap 2 G726-32/8000
a=rtpmap 4 G723/8000
a=rtpmap 15 G728/8000
a=inactive

SIP/2.0 200 OK
CSeq: 1 INVITE
Content-Length: 157
Content-Type: application/sdp
Content-Disposition: session
v=0
o=collins 45678 001 IN IP4 station2.work.com
s=
c=IN IP4 station2.work.com
t=0 0
m=audio 6666 RTP/AVP 4 15
a=rtpmap 4 G723/8000
a=rtpmap 15 G728/8000
a=inactive
Daniel<sip:Collins@station1.work.com>SENDING INVITE INVITE sip:manager@station2.work.com SIP/2.0
CSeq: 2 INVITE
Content-Length: 126
Content-Type: application/sdp
Content-Disposition: session
v=0
o=collins 123456 002 IN IP4 station1.work.com
s=
c=IN IP4 station1.work.com
t=0 0
m=audio 4444 RTP/AVP 15
a=rtpmap 15 G728/8000

ACK sip:manager@station2.work.com SIP/2.0
From: Daniel<sip:Collins@station1.work.com>; tag = abcd1234
To: Boss<sip:Manager@station2.work.com>; tag = xyz789
CSeq: 1 ACK
Content-Length: 0
OPTIONS Method

- Determine the capabilities of a potential called party
- Accept Header
  - Indicate the type of information that the sender hopes to receive
- Allow Header
  - Indicate the SIP methods that Boss can handle
- Supported Header
  - Indicate the SIP extensions that can be supported
Daniel <sip:Collins@station1.work.com>

Boss <sip:Manager@station2.work.com>

OPTIONS sip:manager@station2.work.com SIP/2.0
Via: SIP/2.0/UDP Station1.work.com; branch=z9hG4bK7890123
From: Daniel <sip:Collins@work.com>; tag=lmnop123
To: Boss <sip:Manager@station2.work.com>
Call-ID: 123456@station1.work.com
Contact: Daniel <sip:Collins@station1.work.com>
CSeq: 1 OPTIONS
Accept: application/sdp
Content-Length: 0

SIP/2.0 200 OK
Via: SIP/2.0/UDP Station1.work.com; branch=z9hG4bK7890123
From: Daniel <sip:Collins@work.com>; tag=lmnop123
To: Boss <sip:Manager@station2.work.com>; tag=xyz5678
Call-ID: 123456@station1.work.com
CSeq: 1 OPTIONS
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE
Supported: newfield
Content-Length: 146
Content-Type: application/sdp

v=0
o=manager 45678 001 IN IP4 station2.work.com
s=
c=IN IP4 station2.work.com
t=0 0
m=audio 0 RTP/AVP 4 15
a=rtpmap 4 G723/8000
a=rtpmap 15 G728/8000
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
SIP Event Notification

- 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=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?

SIP/2.0 200 OK
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK7890
From: Daniel <sip:Collins@work.com>
To: Boss <sip:Manager@home.net>
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; branch=z9hG4bKxyz1
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-Type: text/plain
Content-Length: 19
Content-Disposition: render
Hello. How are you?
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: 70
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

SIP/2.0 202 Accepted
Via: SIP/2.0/UDP station1.work.com; branch=z9hG4bK789
From: Mary<sip:Mary@work.com>; tag=123456
To: Joe<sip:Joe@work.com>; tag=67890
Contact: Joe<Joe@station2.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/2.0 200 OK
Via: SIP/2.0/UDP station2.work.com; branch=z9hG4bKxyz1
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=z9hG4bKxyz1
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
From: Joe<sip:Joe@work.com>
To: Mary<sip:Mary@work.com>
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
From: Joe<sip:Joe@work.com>
To: Mary<sip:Mary@work.com>
Call-ID: 123456@station1.work.com
CSeq: 124 NOTIFY
Content-Length: 0
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 retxm
- RAck Header
  - Response ACK
  - In PRACK
  - RSeq+CSeq
- PRACK
  - Prov. Resp. ACK
- Should not
  - Apply to 100
- Default timer value = 0.5 s

INVITE sip:ServerB@network.com SIP/2.0
Via: SIP/2.0/UDP ClientA.network.com; branch=z9hG4bK7890123
Supported: 100rel
Require: 100rel
From: sip:ClientA@network.com; tag=lmnop123
To: sip:ServerB@network.com
Call-ID: 123456@ClientA.network.com
CSeq: 1 INVITE

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ClientA.network.com; branch=z9hG4bK7890123
Require: 100rel
RSeq: 567890
From: sip:ClientA@network.com; tag=lmnop123
To: sip:ServerB@network.com; tag = xyz123
Call-ID: 123456@ClientA.network.com
CSeq: 1 INVITE

Response Lost

Response Retransmit

...
ClientA@network.com

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ClientA.network.com; branch=z9hG4bK7890123
Require: 100rel
RSeq: 567891
From: sip:ClientA@network.com; tag=lmnop123
To: sip:ServerB@network.com; tag=xyz123
Call-ID: 123456@ClientA.network.com
CSeq: 1 INVITE

PRACK sip:ServerB@network.com SIP/2.0
Via: SIP/2.0/UDP ClientA.network.com; branch=z9hG4bK7890123
RAck: 567891 1 INVITE
From: sip:ClientA@network.com; tag=lmnop123
To: sip:ServerB@network.com; tag=xyz123
Call-ID: 123456@ClientA.network.com
CSeq: 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

```
INVITE
Session Description (with pre-condition attributes)

PRACK

UPDATE
Session Description (with updated pre-condition attributes)

PRACK (for 180 response)

ACK
```
Example of e2e Resource Reservation [1/2]

- **SDP for initial INVITE**
  
  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**
  
  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.network.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.network.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)
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 need 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
SIP Client

INVITE
To: Terminal@H323.net

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

Gateway

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

Alerting

H.323 Terminal

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

180 (Ringing)

200 (OK)
To: Terminal@H323.net

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

ACK

Two-way voice

Two-way voice
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

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=123.67.8.9:2000

200 (OK)
To: Terminal@H323.net

c=IN IP4 123.67.8.9
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

c

d
Alerting

e

f
Connect
faststart [logical chan info = G711 TX,
G711 RX 123.45.6.7:800

g

h
Two-way voice

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

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.