Module Name
Author(s): Mario Apuy, Luis Ramirez

Objectives

- Understand the Session Initiation Protocol and it's components
- Understand the signaling exchange for call establishment

Module Overview

This module will help understand the basics of the Session Initiation Protocol (SIP) and the requirements to establish a voice, video or any multimedia session using SIP as the signaling protocol.

Material

High level Overview

SIP Basics

The Session Initiation Protocol (SIP) is an application layer control (signaling) protocol used for creating, modifying and terminating multimedia sessions between one or more participants.

SIP Sessions Include:

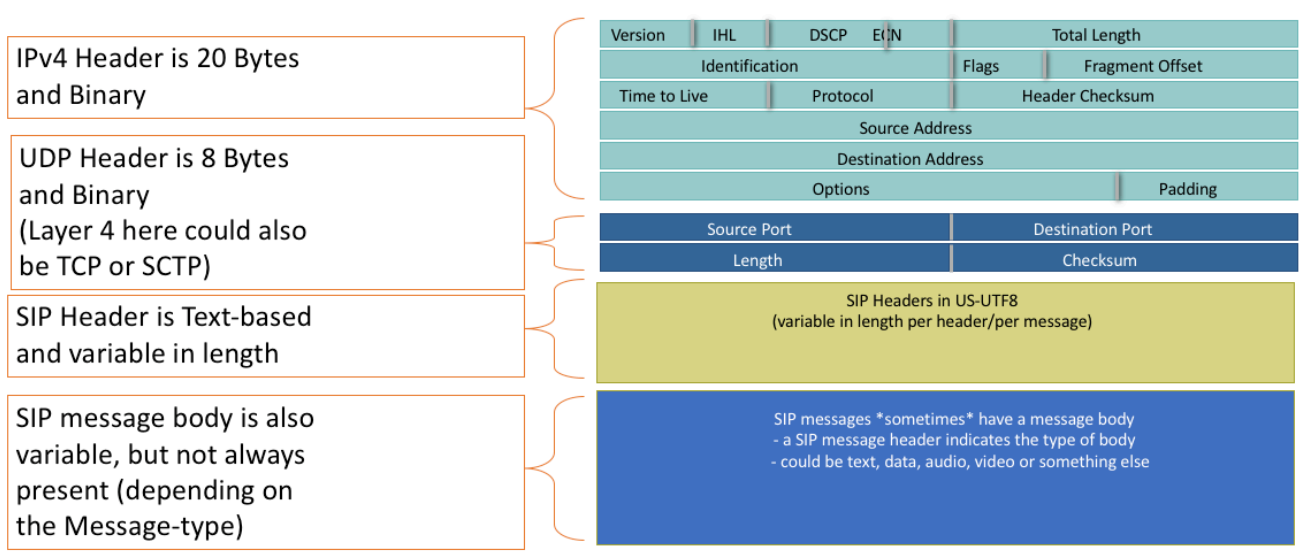
- Internet multimedia conferences
- Internet telephone calls
- Internet video sessions
- Multimedia distribution
- Subscription and Notification of Events
- Publications of State

Communication between the members of a SIP session can happen in a unicast format as well as multicast, a mesh of unicast relations or a combination of all.

On both IPv4 and IPv6 SIP allows for transport to be done over UDP, TCP or SCTP, it also allows for secure connections using TLS over TCP.

Protocol Specifics

Example of a SIP Packet with UDP Format

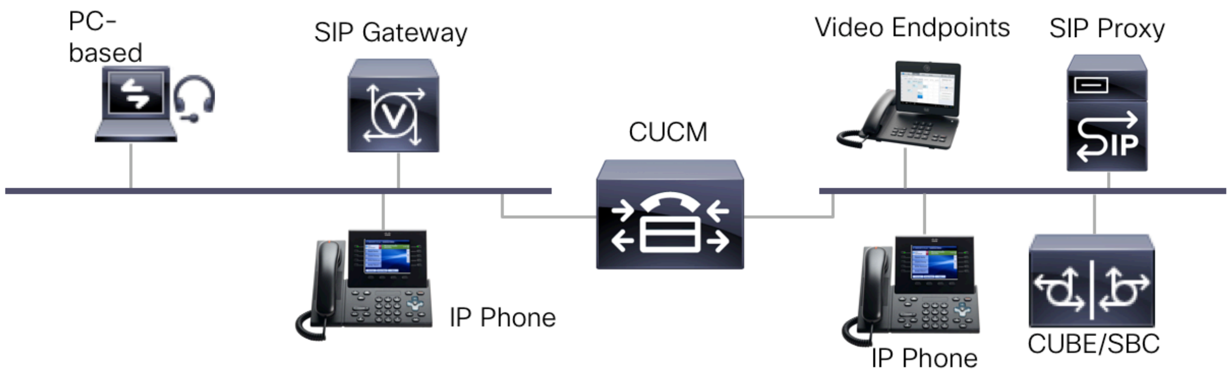


SIP Components

- User Agents (UAs)
- Gateways
- Registrar Servers
- Proxy Servers
- Redirect Servers

User Agents

- User Agent Clients (UAC) send requests to User Agent Servers (UAS)
- User Agent Servers send responses to the requests
- Most SIP devices are both a UAC and a UAS (they both initiate and accept requests)
- Unified CM and CUBE are both Back-to-Back User Agents (B2BUA) (as opposed to Proxies)



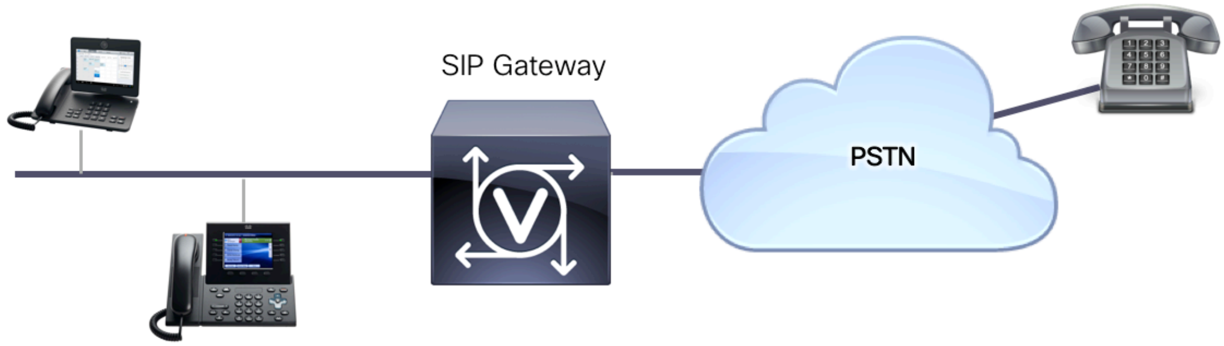
User Agents use a Client-Server model:

- User Agent Client (UAC)—Initiates sessions
- User Agent Server (UAS)—Responds to session requests
- User Agent = UAC + UAS
- All sorts of devices can be a User Agent

SIP Gateways

Translation between SIP protocol format to and from non-SIP protocol format

- SIP to ISDN or Analog
- SIP to H.323



Registrar Server

A registrar is a SIP endpoint that provides location services. It accepts incoming Register request recording the address and other parameters corresponding to that user agent.

SIP Proxy

A proxy is a SIP element that routes requests between an UAC and UAS. Proxies can work on a Stateful or Stateless entities where one retains information about requests that have been processed and the other does not.

Redirect Server

A redirect server provides routing information for requests without having to be involved in further messaging for the session that is been established, thus reducing load on SIP proxies by removing the routing decision.

SIP Addressing

- Fully-Qualified Domain Names
    - sip:jdoe.cisco.com
- SMTP-style Domain Names [RFC 2368]
    - sip:jdoe@cisco.com
- E.164 style addresses [RFC 2806]
    - sip:14085551234@gateway.com; user=phone
      user=phone means this is a gateway

- Mixed addresses
    - sip:14085551234@10.1.1.1; user=phone
      sip:jdoe@10.1.1.1

- Secure SIP Messaging (indicates TLS is used) [RFC 4346]
    - sips:jdoe@cisco.com
      called a “SIPS-URI” or just “SIPS”

SIP Headers

Request-URI header

The Request URI includes the destination of the call, it also identifies the UAS that is in charge of processing the request. The initial request URI value should match the value of the To field.

Via header

The Via header field indicates the transport used for the transaction and identifies the location where the response is to be sent.

Max Forwards

The Max-Forwards header field serves to limit the number of hops a request can transit on the way to its destination. The value set is decremented by one on each hop where the request is handled.

To

The To header field identifies the original recipient of the request designated by the user identified in the From field. The original recipient may or may not be the UAS processing the request, due to call forwarding or other proxy operations.

From

The From header indicates the originator of the request, possibly the user’s address.

Call ID

The Call-ID header is a unique identified for all messages in a dialog, it must be the same for all requests and responses sent by either UA.

Cseq

The Cseq header is used to identify and order transactions, it is composed by a number and the method which should match that of the request, all responses for that request should match.

Contact

The Contact header provides a SIP URI that can be used to contact the UAC that initiated the Invite request for any subsequent requests.

Session Description Protocol (SDP)

Session Description Protocol offers a set of parameters to describe a multimedia session, it's functionality is similar to H.245 and it was developed by the IETF MMUSIC Work Group.

When initiating multimedia sessions, there is a requirement to convey media details, transport addresses, and other session description metadata to the participants.
SDP provides a standard representation for such information.

- SDP includes:
    - The type of media (video, audio, etc)
    - The transport protocol (RTP/UDP/IP, H.320, etc)
    - The format of the media (H.261 video, MPEG video, etc)
- SDP is purely a format for session description
- SDP is NOT a transport protocol

SDP relays on the Offer/Answer described on RFC 3264 for media negotiation.

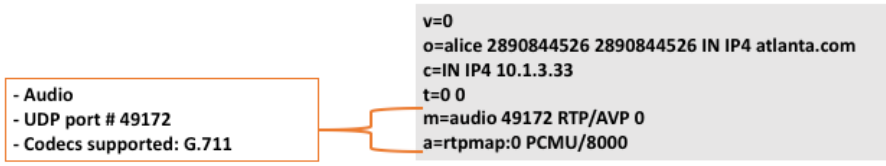
SDP Lines

* "Lines" below are in order

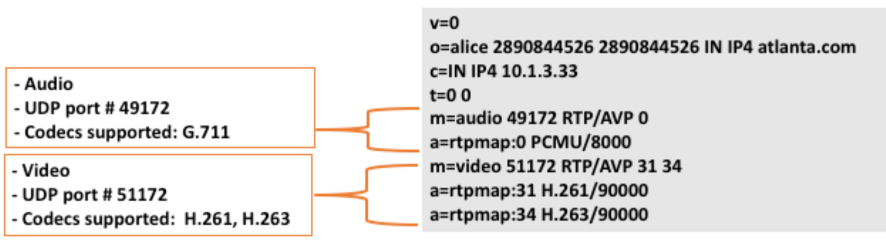
v = protocol version
o = owner/creator and session identifier
s = session name
c = connection information –not required if included in all media
k = encryption keys
t = time the session is active
m = media descriptions and transport address
a = (zero or more) media attributes lines

The SDP information on a SIP Message Body for a Multimedia session

- An SDP for a voice only session



- An SDP for a voice and video session



Media Negotiation

SIP uses the Session Description Protocol (SDP RFC 4566/3266/2327) to relay media information, it then leverages the offer/answer model described in RFC 3264 to negotiate media using the SDP information.

Offer/Answer Model

- One endpoint sends an offer SDP containing all the capabilities the endpoint wishes to negotiate.
- SDP contains m lines for each media stream being negotiated (i.e. audio, video, content channel, etc.9)
- Receiving endpoint sends an answer SDP that contains the same or a subset of capabilities received in the offer.
- Per RFC 3264, “For each "m=" line in the offer, there MUST be a corresponding "m=“ line in the answer. The answer MUST contain exactly the same number of "m=" lines as the offer.”

Early Offer and Delayed Offer

- Initiator of the call can send SDP offer in the INVITE – this is called an Early Offer (EO)
- Receiving endpoint can send the SDP offer in a response if the INVITE did not contain an offer – this is called a Delayed Offer (DO)
- For Early Offer, the answer is sent in a response (usually 200 OK).

- For Delayed Offer, the answer is typically sent in the ACK.

Early Media

- Delayed Offer calls do not set up media until the 200 OK (call is answered)
- If media is required prior to the call being connected, SIP has provisions for Early Media
- With Early Media on a Delayed Offer call, the offer comes from the terminating side in a provisional response (e.g. 183 Session Progress)
- Originating side sends SDP Answer in a PRACK message (defined in RFC 3262)

SIP Methods, Requests and Responses

Methods (Requests)

- INVITE
- ACK
- BYE
- CANCEL
- OPTIONS
- REGISTER

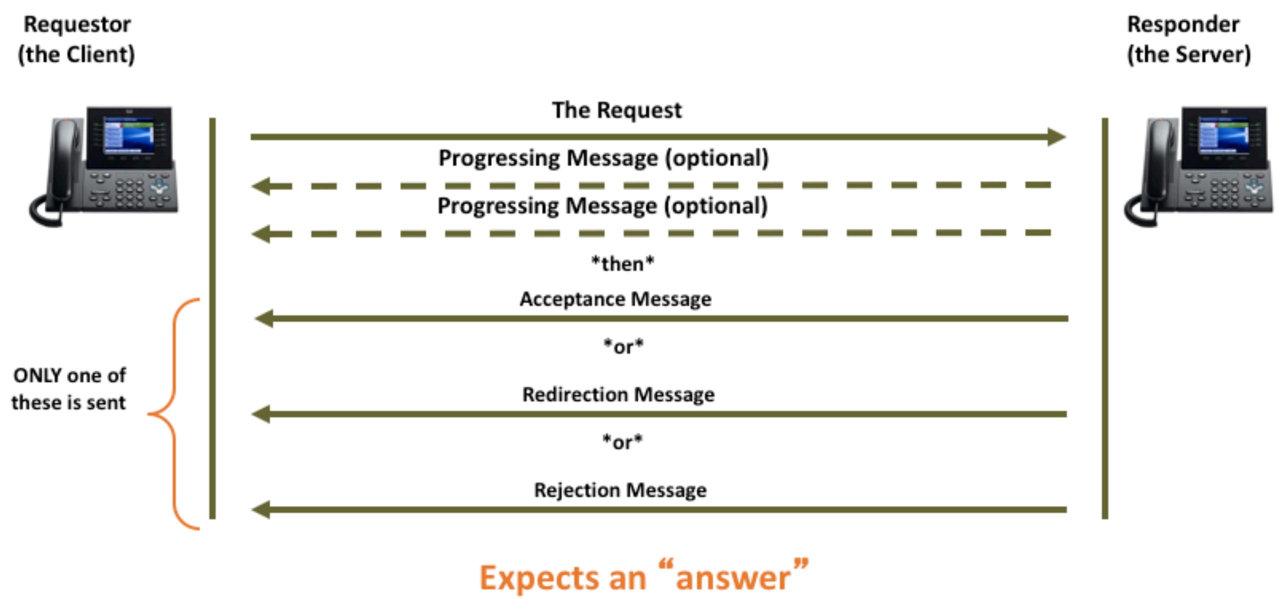
Additional Methods

- INFO (RFC 2976)—to send more information within an established dialog
- PRACK (RFC 3262)—for the UAC to provisionally acknowledge a response (part of very long call set-ups)
- SUBSCRIBE (RFC 3265)—to tell a remote node to look for a certain event
- NOTIFY (RFC 3265)—to respond when that certain event occurs
- UPDATE (RFC 3311)—to update parameters of a session set-up
- MESSAGE (RFC 3428)—SIP instant messaging
- REFER (RFC 3515)—to “refer” one UA to communicate with another UA
- PUBLISH (RFC 3903)—to push UA state information to a compositor/presence server

SIP Responses

Response Code	Description	Example
1xx	Informational – Request Received and Continuing to Process Request	100 Trying 180 Ringing 183 Session Progress
2xx	Success – Action was successfully received, understood, and accepted	200 OK 202 Acceptable
3xx	Redirection – Another SIP Element needs to be contacted in order to complete the request	300 Multiple Choices 301 Moved Permanently 302 Moved Temporarily
4xx	Client Error – Request contains bad syntax or cannot be fulfilled at this server	401 Unauthorized 404 Not Found 406 Not Acceptable 486 Busy Here 488 Not Acceptable Here
5xx	Server Error – Server failed to fulfill an apparently valid request	503 Service Unavailable
6xx	Global Failure – Request is invalid at any server	600 Busy Everywhere 603 Decline

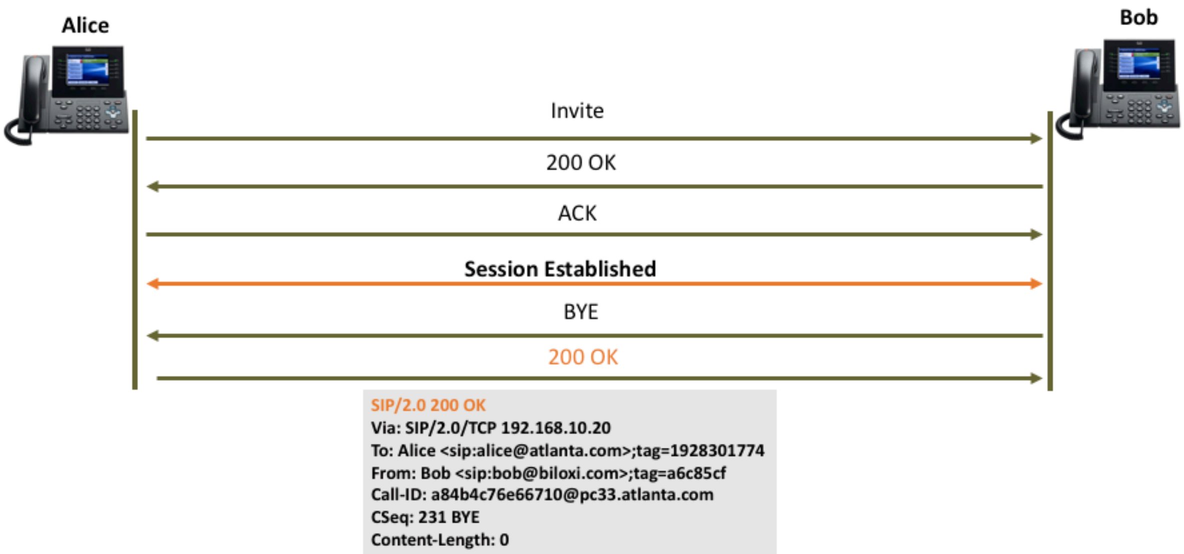
SIP is a request/response protocol



SIP Methods in detail

Invite, ACK and BYE

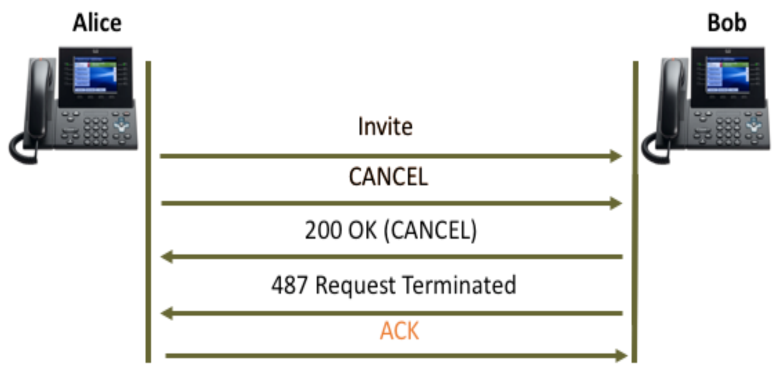
- INVITE—A user or service is being invited to participate in a multimedia session
- ACK—Confirms that a client has received a final response to an INVITE request
- BYE—Terminates an existing session; can be sent by any user agent in a dialog



Cancel

Cancels pending requests; does not terminate sessions that have already been accepted

- SHOULD only be for INVITE Requests
- CANCEL Requests cannot be challenged by Servers
- If a Request exceeds the time in the Expires header, a CANCEL Request should be sent immediately
- All Requests require a Final Response, here this is a 200 OK to the CANCEL
- 487 “Request Terminated” is the proper response to the INVITE as a result of the CANCEL
- An ACK is sent even though the INVITE did not establish a dialog
- If Alice receives Bob’s 200 OK [Final response] to the INVITE prior to receiving the 200 OK to the CANCEL, she will send an ACK immediately, then send a BYE immediately to terminate the session. The users will not be aware of this.




```
INVITE sip:bob@192.168.10.20 SIP/2.0
Via: SIP/2.0/TCP
10.1.3.33;branch=z9hG4bK776asdhds
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:alice@atlanta.com>
Content-Type: application/sdp
Content-Length: 142
```

```
CANCEL sip:bob@192.168.10.20 SIP/2.0
Via: SIP/2.0/TCP
10.1.3.33;branch=z9hG4bK776asdhds
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 10197 CANCEL
Contact: <sip:alice@atlanta.com>
Reason: SIP ;cause=486 ;text="Busy Here"
Content-Length: 0
```

Reason Header will give the reason for the Cancel

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.1.3.33
From: Alice <sip:alice@atlanta.com>;tag=1928301774
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 10197 CANCEL
Content-Length: 0
```

200 OK to the Cancel acknowledges the session will not be established

```
SIP/2.0 487 Request Terminated
Via: SIP/2.0/TCP 10.1.3.33
From: Alice <sip:alice@atlanta.com>;tag=1928301774
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Content-Length: 0
```

487 Request Terminated
is the proper Response to the CANCEL Request

```
ACK sip:bob@192.168.10.20 SIP/2.0
Via: SIP/2.0/TCP
10.1.3.33;branch=z9hG4bK776asdhds
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 ACK
Content-Length: 0
```

ACK always follows a 4xx Response

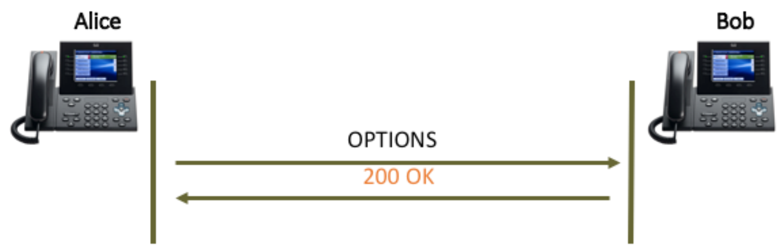
Options

Allows a UA to query another UA or a proxy server as to its capabilities

- Allows a UAC to discover the supported methods, content types, extensions, codecs, etc. without "ringing" the other party

- All UAs MUST support the OPTIONS method

- A 200 OK provides all:
    - Contacts known to that UAS
    - Methods supported by that UAS
    - Language supported by that UAS
    - Message Body type accepted
    - assumed to be “application/sdp” if no Accept header
    - SDP port MUST be set to zero (0) to ensure this isn’t considered an Offer by any SIP element
- A 486 “Busy Here” is returned if the UA is not ready to accept a new Request




```
OPTIONS sip:bob@192.168.10.20 SIP/2.0
Via: SIP/2.0/TCP pc33.atlanta.com;branch=z9hG4bK77i832k9
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e6Kr456@pc33.atlanta.com
CSeq: 22756 OPTIONS
Contact: <sip:alice@pc33.atlanta.com>
Allow: INVITE, ACK, OPTIONS, BYE, CANCEL, REFER, SUBSCRIBE,
NOTIFY, MESSAGE, UPDATE
Accept: application/sdp, application/pidf-xml
Content-Length: 0
```

Allows a UAC to discover the supported:

- methods,
- content types,
- extensions,
- codecs,
- etc.

Without "ringing" the other party

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP sip:alice@atlanta.com;branch=z9hG4bK77i832k9
To: Bob <sip:bob@biloxi.com>; tag=a6c85e3
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e6Kr456@pc33.atlanta.com
CSeq: 22756 OPTIONS
Contact: <sip:bob@biloxi.com>
Contact: <sip:bob_home@biloxi.com>
Allow: INVITE, ACK, OPTIONS, BYE, CANCEL, REFER,NOTIFY, MESSAGE
Accept: application/sdp, text/plain, image/jpeg
Accept-language: en, fr
Content-Type: application/sdp
Content-Length: 274
(Bob’s SDP indicating codecs supported)
```

200 OK provides all:

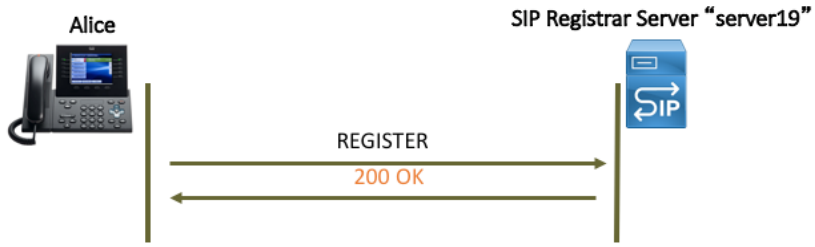
- Contacts known
- Methods supported
- Language supported
- Message Body type

A 486 “Busy Here” is returned if the UA is not ready to accept a new Request

Register

Registers the user agent with the registrar server of a domain

- Binds a SIP URI (called an Address of Record (AOR)) and a contact name in a location service.
- Enables UAs to receive SIP messages.
- Registrations represent a dynamic piece of state maintained in a network.
- Because devices can be “always on”, a domain can request that a SIP device re-authenticate to the domain.
- The 200 (OK) response from the registrar contains a list of Contact fields enumerating all current bindings.
- UAs can use three ways to determine the address to which to send registrations: by configuration, using the address-of-record and multicast.




```
REGISTER sip:server19.atlanta.com SIP/2.0
Via: SIP/2.0/TCP
pc33.atlanta.com;branch=z9hG4bk2l55n1
To: Alice <sip:alice@atlanta.com>
From: Alice <sip:alice@atlanta.com>;tag=283074
Call-ID: a84b4g96te10@pc33.atlanta.com
CSeq: 31862 REGISTER
Contact: <sip:alice@10.1.3.33>
Expires: 21600
Content-Length: 0
```

Binds a SIP URI (called an Address of Record (AOR)) with a contact name in a location service

- Enables UAs to receive SIP messages
- Registrations represent a dynamic piece of state maintained in a network

UAs can use three ways to determine the address to send registrations:

- Configuration
- Address-of-Record
- Multicast [224.0.1.75]

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP pc33.atlanta.com;branch=z9hG4bk2l55n1;
received=10.1.3.33
To: Alice <sip:alice@atlanta.com>; tag=a6c85e3
From: Alice <sip:alice@atlanta.com>;tag=283074
Call-ID: a84b4g96te10@pc33.atlanta.com
CSeq: 31862 REGISTER
Contact: <sip:alice@pc33.atlanta.com>
Contact: <sip:alice@cm9013.atlanta.com>
Service-Route: <sip:bigbox3.atlanta.com;lr>
Expires: 3600
Contact-Length: 0
```

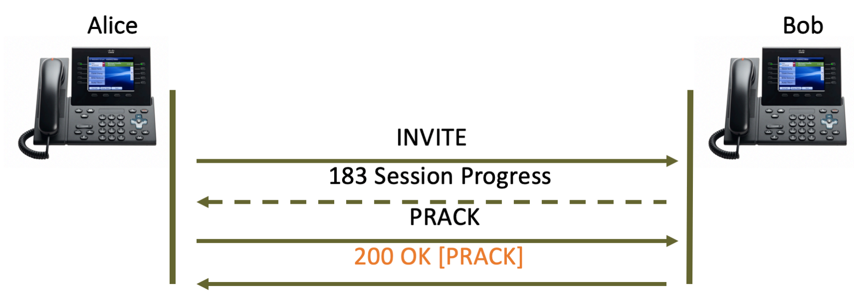
- The 200 (OK) response from the registrar contains a list of Contact fields enumerating all current bindings
- Expires Header informs UA how long Registration lasts before a refresh is required
- Because devices can be “always on”, a domain can request that a SIP device re-authenticate to the domain
- The return of the Service-Route header supplies this route for her UA

PRACK

Provisional Acknowledgment is a reliable provisional response.

- Its purpose is to acknowledge progress information on a requesting process (what's the status of an INVITE)
- Sent by UAC after receiving a non-100 Provisional response to an INVITE Request and before the Final Acknowledgement message, but only if asked to within the INVITE
- PRACKs require a 200 OK from the UAS/Server
- MUST be sent if an INVITE contained a Require Header with a 100rel option tag
- 100 Trying is not to be sent reliably

RFC 3662 describe in detail functionality of the PRACK.




```
INVITE sip:bob@192.168.10.20 SIP/2.0
Via: SIP/2.0/TCP pc33.atlanta.com;branch=z9hG4bK776asdhds
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Requires: 100rel
Content-Type: application/sdp
Content-Length: 142

(SDP omitted)
```

- The INVITE Includes a Require header stipulating the UAC wants a reliable provisional response

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/TCP pc33.atlanta.com ;branch=z9hG4bK776asdhds
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
RSeq: 813520
Contact: <sip:alice@pc33.atlanta.com>
Content-Type:  application/sdp
Content-Length: 235

(SDP omitted)
```

- Includes an RSeq header, which is the sequence number of the reliable message to be generated by the UAC
- This is necessary to synchronize both UAs with a multi-step session set-up similar to an ACK

```
PRACK sip:bob@192.168.10.20 SIP/2.0
Via: SIP/2.0/TCP pc33.atlanta.com;branch=z9hG4bK776asi98JK
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159
RAck: 813520 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 0
```

- UAC sends message
- RAck – Reliable ACK value matching RSeq

```
SIP/2.0 200 OK sip:bob@192.168.10.20
Via: SIP/2.0/TCP
pc33.atlanta.com;branch=z9hG4bK776asi98JK;received=10.1.3.33
To: Bob <sip:bob@biloxi.com>; tag=a6c85e3
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 PRACK
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 0
```

- 200 OK to the PRACK is sent by the UAS
- Message flows never end with a PRACK or its 200 OK

Subscribe and Notify

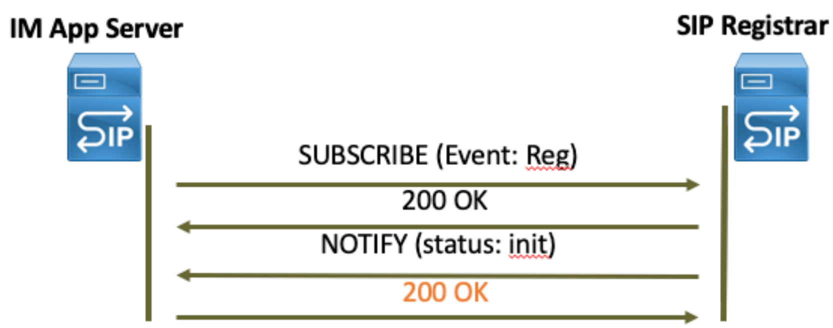
SUBSCRIBE

1. Method used to request current state and state updates from a remote node
2. Used to request asynchronous notification of an event or set of events at a later time
    - SUBSCRIBE requests SHOULD contain an "Expires" header which indicates the duration of the subscription
    - Subscriptions need to be refreshed periodically
    - 200-class responses to SUBSCRIBE requests also MUST contain an "Expires" header
    - Subscribers MUST include exactly one "Event" header in SUBSCRIBE requests, indicating to which event or class of events they are subscribing
    - SUBSCRIBE is a dialog-creating method
    - 200-class responses indicate that the subscription has been accepted, and that a NOTIFY will be sent immediately
    - If a proxy wishes to see all of the SUBSCRIBE and NOTIFY requests for a given dialog, it MUST record-route the initial SUBSCRIBE and any dialog-establishing NOTIFY requests
    - SUBSCRIBE Events are IANA registered

NOTIFY

1. Sent to inform subscribers of changes in state to which the subscriber has a subscription
2. Used to notify a SIP node that an event which has been requested by an earlier SUBSCRIBE method has occurred
   - Sending a NOTIFY message to an unsuspecting node is invalid behavior, it MUST receive a 481 “Subscription does not exist" response
   - NOTIFY "Event" headers will contain a single event package name for which a notification is being generated, it MUST match the "Event" header in the corresponding SUBSCRIBE message
   - Subscribers and Notifiers may choose to use S/MIME for their message bodies; therefore, Proxies might not be able to assess any information that is not explicitly required to be proxy-readable by SIP

RFCs 3265 and 3680 describe in detail the functionality of Subscribe and Notify



SUBSCRIBE sip:server19@atlanta.com SIP/2.0
Via: SIP/2.0/TCP app_IM.atlanta.com;branch=z9hG4bKnashds7
From: sip:app_IM.atlanta.com ;tag=123aa9
To: sip:server19@atlanta.com
Call-ID: 9987@app_IM.atlanta.com
CSeq: 9887 SUBSCRIBE
Contact: sip:app_IM.atlanta.com
Event: reg
Max-Forwards: 70
Expires: 21600
Accept: application/reginfo+xml

Used to request asynchronous notification of an event or set of events at a later time

- method used to request current state and state updates from a remote node
- Expires header SHOULD be present in Request
- Requests MUST have exactly one Event Header value

```
NOTIFY sip:app_IM.atlanta.com SIP/2.0
Via: SIP/2.0/TCP server1.atlanta.com;branch=z9hG4bKnasaii
From: sip:server19@atlanta.com ;tag=xyzygg
To: sip:app_IM.atlanta.com ;tag=123aa9
Max-Forwards: 70
Call-ID: 9987@app_IM.atlanta.com
CSeq: 1288 NOTIFY
Contact: sip:server19.atlanta.com
Event: reg
Subscription-State: active
Content-Type: application/reginfo+xml
Content-Length: 223

<?xml version="1.0"?>
  <reginfo xmlns=
          "urn:ietf:params:xml:ns:reginfo"
           version="0" state="full">
   <registration aor="sip:alice@atlanta.com"
           id="a7" state="init" />
  </reginfo>
```

Used to notify a SIP node that an event which has been requested by an earlier SUBSCRIBE method has occurred

- NOTIFY is sent to inform subscribers of changes in state to which the subscriber has a subscription
- Event Header MUST match

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP server19.atlanta.com  ;branch=z9hG4bKnasaii
;received=10.1.3.1
From: sip:app_IM.atlanta.com ;tag=123aa9
To: sip:server19@atlanta.com ;tag=xyzygg
Call-ID: 9987@app_IM.atlanta.com
CSeq: 1288 NOTIFY
Contact: sip:server1.atlanta.com
Content-Length: 0
```

- sending a NOTIFY message to an unsuspecting node is invalid behavior, MUST receive a 481 "Subscription does not exist" response

Info

Used for the carrying of session related control information that is generated during a session.

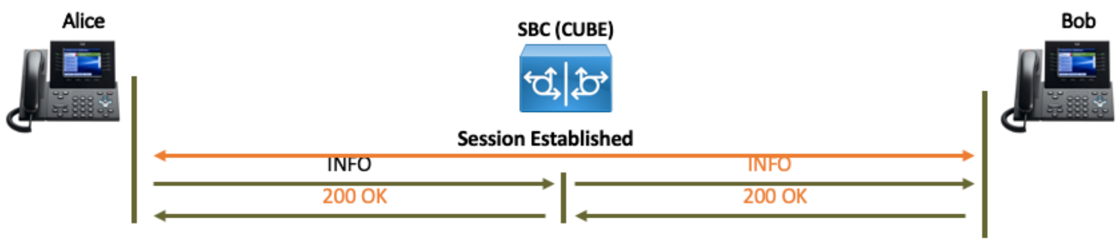
- The purpose of the INFO message is to carry application level information along the SIP signaling path
- The INFO method is not used to change the state of SIP calls, or the parameters of the sessions SIP initiates
- It is necessary that the mid-session signaling information traverse the post session setup SIP signaling path. This is the path taken by SIP re-INVITEs, BYEs and other SIP requests that are tied to an individual session. This allows SIP proxy servers to receive, and potentially act on, the mid-session signaling information
- Example Uses:
    - Carrying mid-call PSTN signaling messages between PSTN gateways
    - Carrying DTMF digits generated during a SIP session
    - Carrying wireless signal strength information in support of wireless mobility applications

- Carrying account balance information
- Carrying images or other non-streaming information between the participants of a session

This information will generally be carried in message bodies, although it can be carried in headers in the INFO message.

Endpoints need to have agreed upon a Content-Type for INFO exchange, as the RFC 2976 does not include any text to account for this.

If a CANCEL is sent to an INFO message, a 487 "Request Terminated" is the appropriate Response.



Update

- Allows a client to update parameters of a session but has no impact on the state of a dialog
    - such as the set of media streams and their codecs
    - can be sent before the initial INVITE has been completed
    - MAY be sent by either caller or callee
- Session Initiation occurs just as in 3261 (INVITE), a UAC compliant to RFC 3311 SHOULD also include an Allow header field in the INVITE request, listing the method UPDATE
- Although UPDATE can be used on confirmed (existing) dialogs, it is RECOMMENDED that a re-INVITE be used instead

```
INVITE sip:bob@biloxi.com/TCP SIP/2.0
Via: SIP/2.0/TCP pc33.atlanta.com;branch=z9hG4bK776asdhds
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928
Call-ID: a84b4c76e66710@pc33.atlanta.com
Allow: UPDATE
CSeq: 22756 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Requires: 100rel
Content-Type: application/sdp
Content-Length: 142

(SDP omitted)
(but calls for the codec G.711)
```

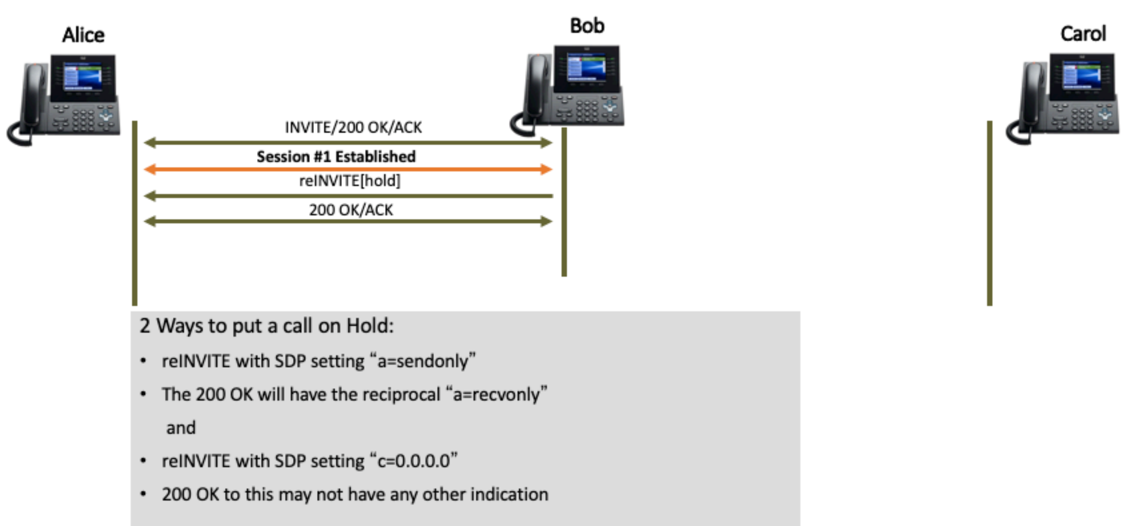
```
SIP/2.0 180 Ringing
Via: SIP/2.0/TCP pc33.atlanta.com
 ;branch=z9hG4bK776asdhds
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928
Call-ID: a84b4c76e66710@pc33.atlanta.com
Allow: UPDATE
CSeq: 22756 INVITE
RSeq: 813520
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

(SDP Omitted)
(suitable response with the codec G.711)
```

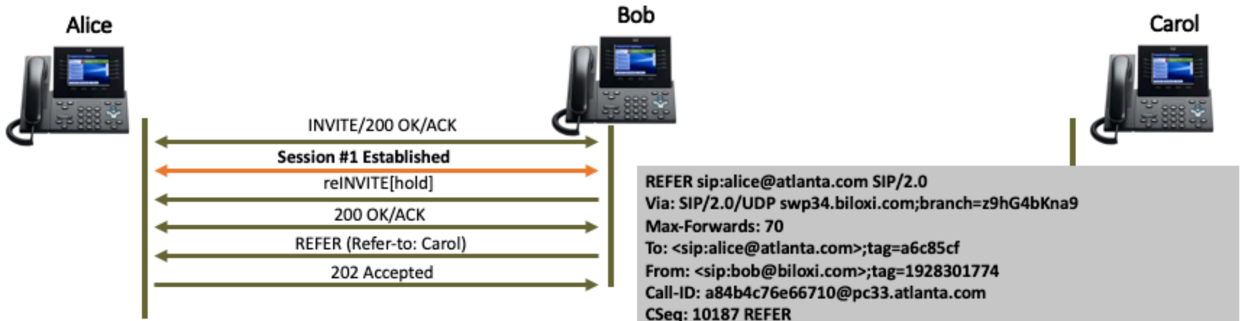
- UAC should include Allow Header indicating support for UPDATE
- UAS should include Allow Header indicating support for UPDATE

Refer

SIP Request directing one UA to INVITE another UA to a new session

- Call Transfer is an example
- REFER implicitly establishes a subscription with another UA
- The "Refer-to" SIP Header is mandatory in the REFER Request to let the referred party know where to go
- The user of a UA receiving a "well formed" REFER Request SHOULD request user input before proceeding with the request
- A 202 "Accept" is a good Response to a REFER

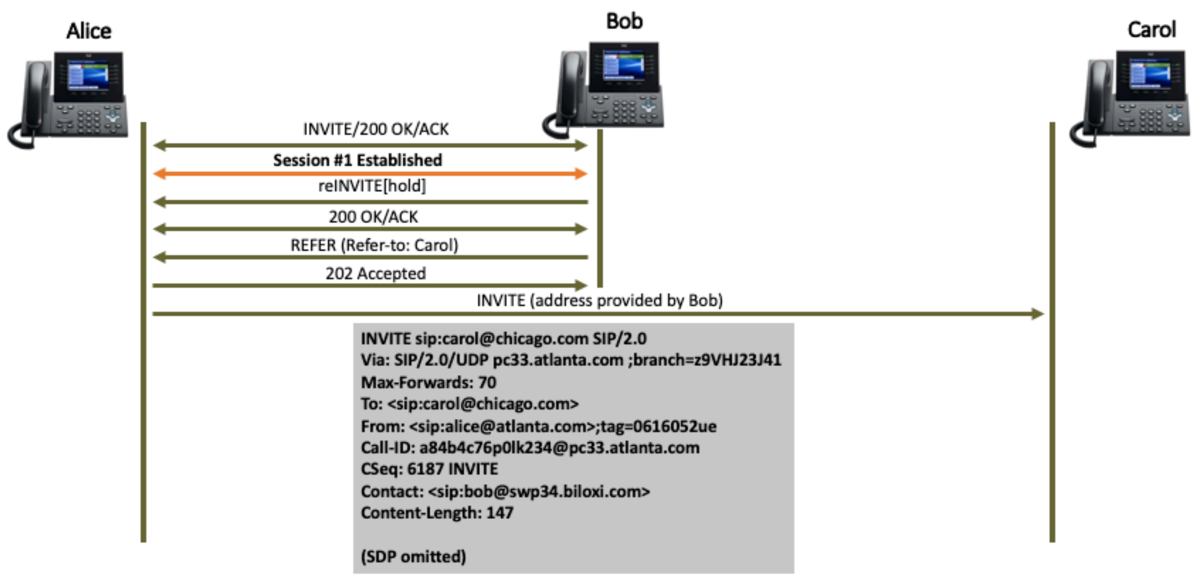


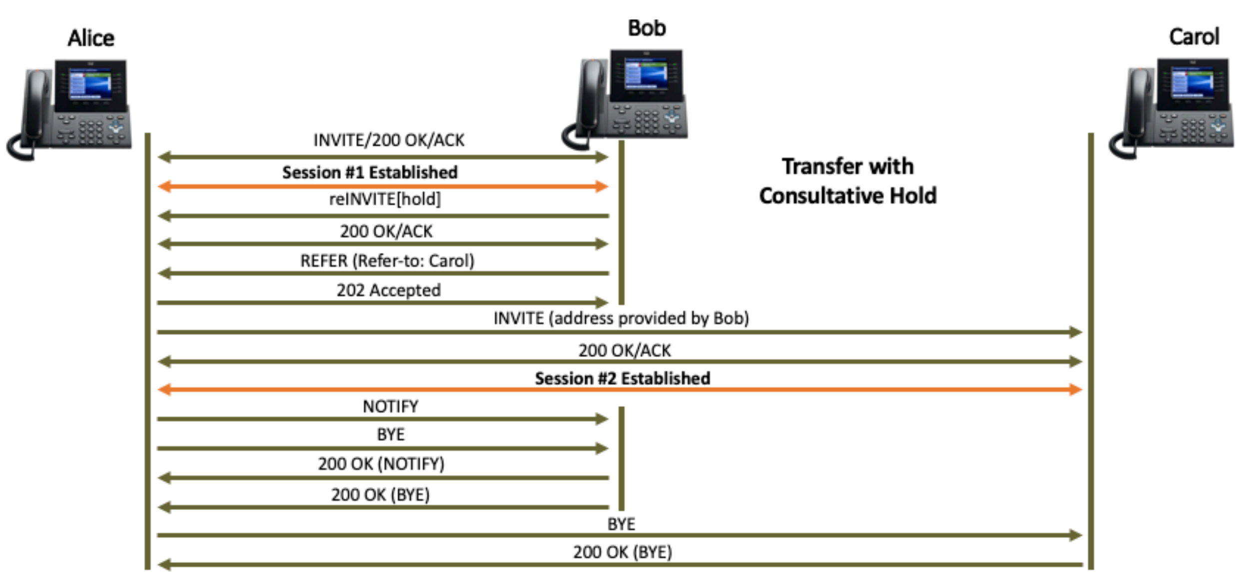


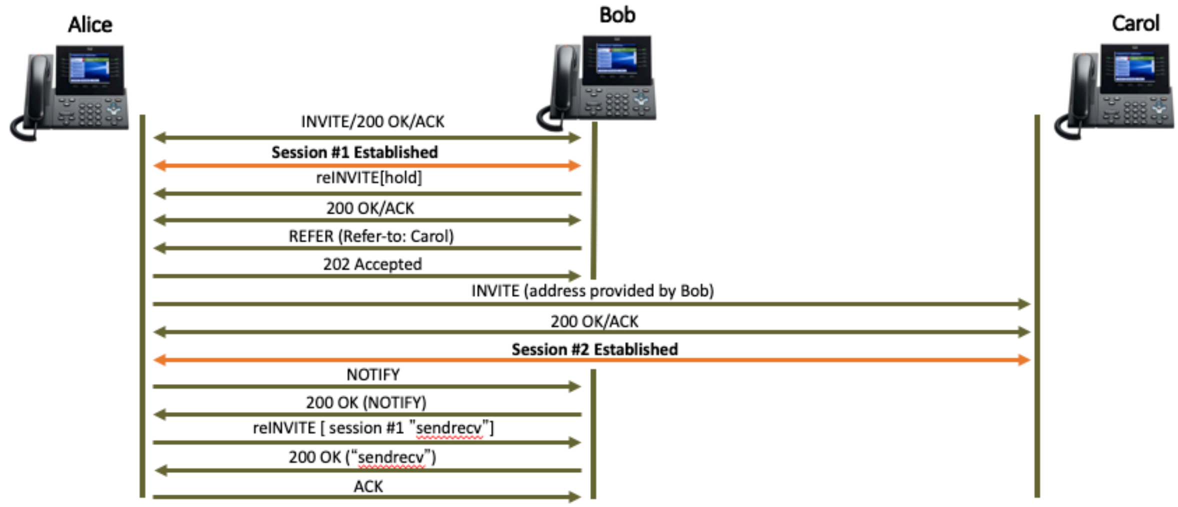
Refer Request from one UA to INVITE another to a session

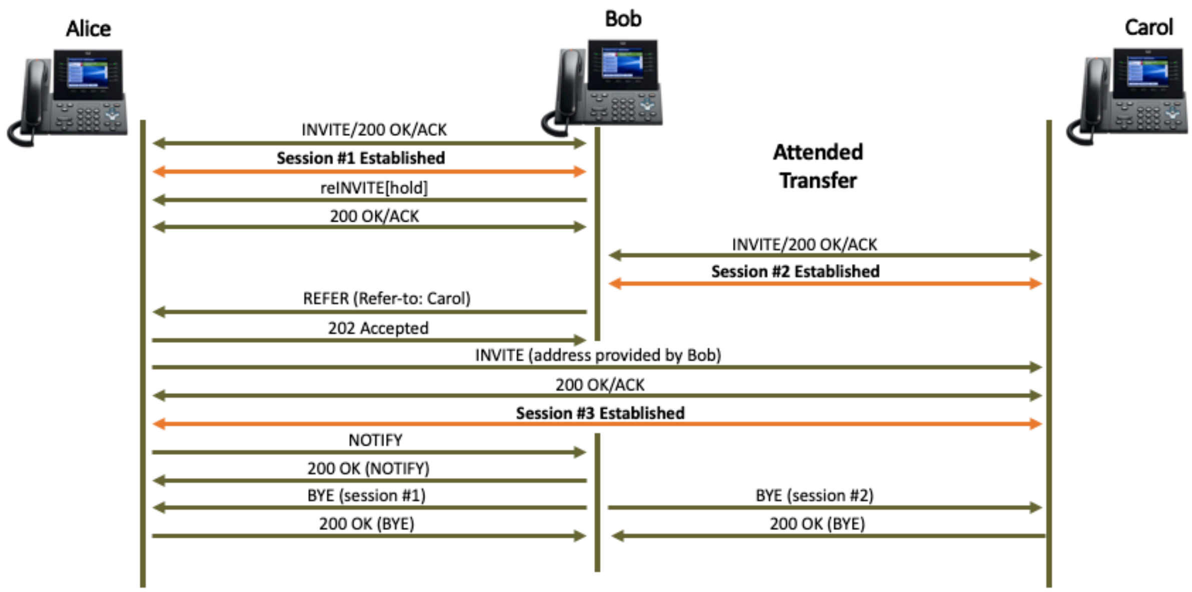
- REFER implicitly establishes a subscription with another UA
  - NOTIFY is required when completed
- The "Refer-to" SIP Header is mandatory in the REFER Request

```
REFER sip:alice@atlanta.com SIP/2.0
Via: SIP/2.0/UDP swp34.biloxi.com;branch=z9hG4bKna9
Max-Forwards: 70
To: <sip:alice@atlanta.com>;tag=a6c85cf
From: <sip:bob@biloxi.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 10187 REFER
Allow: INVITE, ACK, CANCEL, OPTIONS,
 BYE, REFER, NOTIFY, UPDATE
Supported: replaces
Refer-To: <sip:carol@chicago.com>
Contact: <sip:bob@swp34.biloxi.com>
Content-Length: 0
```









DTMF Relay

Three different methods for carrying DTMF signals over a SIP network.

- RFC2833
- SIP Notify
- SIP KPML

RFC2833

- Digits are passed in the RTP stream with a unique payload type
- Capability is negotiated in SDP like any other codec

Offer

m=audio 30414 RTP/AVP 0 8 116 18 100 101
c=IN IP4 172.18.106.231
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:100 X-NSE/800
a=fmtp:100 192-194
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16

Answer

m=audio 17236 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

SIP Notify

- Passes DTMF information in a SIP NOTIFY message telephone-event Event Negotiated in Call-Info header

Offer

INVITE sip:+19195553333@172.18.106.231:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.106.59:5060;branch=z9hG4bK9843c455840434
From: "Paul Giralt" <sip:9195551234@172.18.106.59>;tag=14902469~0d0d25d7-4931-4a07-83c6
To: <sip:+19195553333@172.18.106.231>
Date: Mon, 13 May 2013 14:48:00 GMT
Call-ID: 1a189580-1901fd20-962c99-3b6a12ac@172.18.106.59
... snip ...
Call-Info: <sip:172.18.106.59:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=DESKTOP
... snip ...
Max-Forwards: 69
Content-Length: 0

Answer

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.106.59:5060;branch=z9hG4bK9843c455840434
From: "Paul Giralt" <sip:9195551234@172.18.106.59>;tag=14902469~0d0d25d7-4931-4a07-83c6
To: <sip:+19195553333@172.18.106.231>;tag=4363A830-17FC
Call-ID: 1a189580-1901fd20-962c99-3b6a12ac@172.18.106.59
... snip ...
Allow-Events: telephone-event
Call-Info: <sip:172.18.106.231:5060>;method="NOTIFY;Event=telephone-event;Duration=500”
... snip ...
Content-Length: 601

Notify

NOTIFY sip:172.18.106.231:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.106.59:5060;branch=z9hG4bK98443140152a0a
From: "Paul Giralt" <sip:9195551234@172.18.106.59>;tag=14902469~0d0d25d7-4931-4a07-83c6
To: <sip:+19195553333@172.18.106.231>;tag=4363A830-17FC
Call-ID: 1a189580-1901fd20-962c99-3b6a12ac@172.18.106.59
CSeq: 104 NOTIFY
Max-Forwards: 70
Date: Mon, 13 May 2013 14:48:11 GMT
User-Agent: Cisco-CUCM10.0
Event: telephone-event
Subscription-State: active
Contact: <sip:172.18.106.59:5060>
P-Asserted-Identity: "Paul Giralt" <sip:9195551234@172.18.106.59>
Content-Type: audio/telephone-event
Content-Length: 4

SIP KPML

- Passes DTMF information in a SIP NOTIFY message kpml Event
- Capability advertised in Allow-Events – uses SUBSCRIBE message to subscribe

Offer

INVITE sip:+19195554444@172.18.106.231:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.106.59:5060;branch=z9hG4bK986efd6c4e51e4
From: "Paul Giralt" <sip:9195551234@172.18.106.59>;tag=14918970~0d0d25d7-4931-4a07-83c6
To: <sip:+19195554444@172.18.106.231>
Date: Mon, 13 May 2013 15:05:24 GMT
Call-ID: 885e5780-19110134-96567f-3b6a12ac@172.18.106.59
User-Agent: Cisco-CUCM10.0
... snip ...
Allow-Events: presence, kpml
... snip ...
Session-Expires: 18000
Max-Forwards: 69
Content-Length: 0

Answer

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.106.59:5060;branch=z9hG4bK986efd6c4e51e4
From: "Paul Giralt" <sip:9195551234@172.18.106.59>;tag=14918970~0d0d25d7-4931-4a07-83c6
To: <sip:+19195554444@172.18.106.231>;tag=437394E8-2E1
Date: Mon, 13 May 2013 15:05:26 GMT
Call-ID: 885e5780-19110134-96567f-3b6a12ac@172.18.106.59
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO
Allow-Events: kpml, telephone-event
Remote-Party-ID: <sip:9196247285@172.18.106.231>;party=called;screen=no;privacy=off
Contact: <sip:+19196247285@172.18.106.231:5060>
Supported: replaces
Server: Cisco-SIPGateway/IOS-15.2.4.M3
Require: timer

Session-Expires: 18000;refresher=uac
Content-Type: multipart/mixed;boundary=uniqueBoundary
Mime-Version: 1.0
Content-Length: 600

Subscribe to KPML

SUBSCRIBE sip:9195554444@172.18.106.59:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.106.231:5060;branch=z9hG4bKBAE27139E
From: <sip:+19195551234@172.18.106.231>;tag=437394E8-2E1
To: "Paul Giralt" <sip:9195554444@172.18.106.59>;tag=14918970~0d0d25d7-4931-4a07-83c6
Call-ID: 885e5780-19110134-96567f-3b6a12ac@172.18.106.59
CSeq: 101 SUBSCRIBE
Max-Forwards: 70
User-Agent: Cisco-SIPGateway/IOS-15.2.4.M3
Event: kpml
Expires: 7200
Contact: <sip:172.18.106.231:5060>
Content-Type: application/kpml-request+xml
Content-Length: 327

<?xml version="1.0" encoding="UTF-8"?><kpml-request xmlns="urn:ietf:params:xml:ns:kpml-request" xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-request kpml-request.xsd" version="1.0"><pattern persist="persist"><regex tag="dtmf">[x*#ABCD]</regex></pattern></kpml-request>

Digit been sent

NOTIFY sip:172.18.106.231:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.106.59:5060;branch=z9hG4bK986f73662cca3b
From: "Paul Giralt" <sip:9195554444@172.18.106.59>;tag=14918970~0d0d25d7-4931-4a07-83c6
To: <sip:+19195551234@172.18.106.231>;tag=437394E8-2E1
Call-ID: 885e5780-19110134-96567f-3b6a12ac@172.18.106.59
CSeq: 104 NOTIFY
Max-Forwards: 70
User-Agent: Cisco-CUCM10.0
Event: kpml
Subscription-State: active;expires=7197
Contact: <sip:9195554444@172.18.106.59:5060>
Content-Type: application/kpml-response+xml
Content-Length: 336

<?xml version="1.0" encoding="UTF-8" ?>
<kpml-response xmlns="urn:ietf:params:xml:ns:kpml-response" xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-response kpml-response.xsd" code="200" digits="1" forced_flush="false" suppressed="false" tag="dtmf" text="Success" version="1.0"/>

Additional References

- SIP RFC
  https://www.ietf.org/rfc/rfc3261.txt
- Session Description Protocol
  https://www.ietf.org/rfc/rfc4566.txt
  https://www.ietf.org/rfc/rfc3264.txt
- RFC 2833
  https://tools.ietf.org/html/rfc2833
- Cisco SIP configuration guide
  https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/sip/configuration/15-mt/sip-config-15-mt-book.html