IN THE UNITED STATES DISTRICT COURT
FOR THE EASTERN DISTRICT OF TEXAS

| | |
|---|---|
| METASWITCH NETWORKS LTD,<br>Plaintiff,<br>v.<br>GENBAND US LLC,<br>Defendant,<br>GENBAND US LLC,<br>Counterclaim Plaintiff,<br>v.<br>METASWITCH NETWORKS LTD AND METASWITCH NETWORKS CORP.,<br>Counterclaim Defendants. | Civil Action No.: 2:14-cv-744<br><br>**DECLARATION OF THE RFC PUBLISHER FOR THE INTERNET ENGINEERING TASK FORCE, AN ORGANIZED ACTIVITY OF THE INTERNET SOCIETY** |

I, Sandy Ginoza, based on my personal knowledge and information, hereby declare as follows:

1. I am an employee of Association Management Solutions, LLC (AMS), which acts under contract to the Internet Society (ISOC) as the operator of the RFC Production Center. The RFC Production Center is part of the "RFC Editor" function, which prepares documents for publication and places files in an online repository for the authoritative Request for Comments (RFC) series of documents (RFC Series), and perserves records relating to these documents. The RFC Series includes, among other things, the series of Internet standards developed by the Internet Engineering Task Force (IETF), an organized activity of

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2. Among my responsibilities as Director of the RFC Production Center, I act as the custodian of records relating to the RFC Series.

3. From June 1999 to 5 January 2010, I was an employee of the Information Sciences Institute at University of Southern California (ISI). I held various position titles with the RFC Editor project at ISI, ending with Senior Editor.

4. The RFC Editor function was conducted by ISI under contract to the United States government prior to 1998. In 1998, ISOC, in furtherance of its IETF activity, entered into the first in a series of contracts with ISI providing for ISI's performance of the RFC Editor function. Beginning in 2010, certain aspects of the RFC Editor function were assumed by the RFC Production Center operation of AMS under contract to ISOC (acting through its IETF function and, in particular, the IETF Administrative Oversight Committee). The business records of the RFC Editor function as it was conducted by ISI are currently housed on the computer systems of AMS, as contractor to ISOC.

5. I make this declaration based on my personal knowledge and information contained in the business records of the RFC Editor as they are currently housed at AMS, or confirmation with other responsible RFC Editor personnel with such knowledge.

6. Since approximately 1998, the RFC Editor's regular practice has been to publish RFCs and make them available to the public on its website at www.rfc-editor.org. The RFC Production Center currently makes available authoritative versions of all RFCs in the ordinary course of its regularly conducted activities on its website at www.rfc-editor.org.

7. Attachment A hereto lists five RFCs published by the RFC Editor, true and correct copies of which are attached as Exhibits 1 - 5.

8. I personally reviewed the documents attached as Exhibits 1 - 5.

9. I hereby certify, in accordance with the requirements of Federal Rule of Evidence 902, that the attached Exhibits 1 - 5 constitutes records of regularly conducted business activity which were (A) made at or near the time of the occurrence of the matters set forth by, or from information transmitted by, a person with knowledge of those matters; (B) kept in the course of the regularly conducted activity; and (C) made by the regularly conducted activity as a regular practice.

10. Based on a search of RFC Editor records, I have determined that the RFC Editor maintained copies of the documents attached as Exhibits 1 - 5 in the ordinary course of its regularly conducted activities.

11. Based on a search of RFC Editor records and the RFC Editor's course of conduct in publishing RFCs, I have also determined that the documents attached as Exhibits 1 – 5 were published on the RFC Editor website on or about the corresponding dates set forth in Attachment A. At such time, each such document was reasonably accessible to the public, and was disseminated or otherwise available to the extent that persons interested and ordinarily skilled in the subject matter or art exercising reasonable diligence could have located it.

Pursuant to Section 1746 of Title 28 of United States Code, I declare under penalty of perjury under the laws of the United States of America that the foregoing is true and correct and that the foregoing is based upon personal knowledge and information and is believed to be true.

Date: 27 OCTOBER 2015 By: ______________________

Sandy Ginoza

ATTACHMENT A

| Exhibit No. | RFC | Title | Publication Date |
|---|---|---|---|
| 1 | 2543 | SIP: Session Initiation Protocol | March 1999 |
| 2 | 3261 | SIP: Session Initiation Protocol | July 2002 |
| 3 | 5806 | Diversion Indication in SIP | March 2010 |
| 4 | 3891 | The Session Initiation Protocol (SIP) "Replaces" Header | September 2004 |
| 5 | 5359 | Session Initiation Protocol Service Examples | October 2008 |

Network Working Group
Request for Comments: 2543
Category: Standards Track

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March 1999

# SIP: Session Initiation Protocol

## Status of this Memo

This document specifies an Internet standards track protocol for the Internet community, and requests discussion and suggestions for improvements. Please refer to the current edition of the "Internet Official Protocol Standards" (STD 1) for the standardization state and status of this protocol. Distribution of this memo is unlimited.

## Copyright Notice



## IESG Note

The IESG intends to charter, in the near future, one or more working groups to produce standards for "name lookup", where such names would include electronic mail addresses and telephone numbers, and the result of such a lookup would be a list of attributes and characteristics of the user or terminal associated with the name. Groups which are in need of a "name lookup" protocol should follow the development of these new working groups rather than using SIP for this function. In addition it is anticipated that SIP will migrate towards using such protocols, and SIP implementors are advised to monitor these efforts.

## Abstract

The Session Initiation Protocol (SIP) is an application-layer control (signaling) protocol for creating, modifying and terminating sessions with one or more participants. These sessions include Internet multimedia conferences, Internet telephone calls and multimedia distribution. Members in a session can communicate via multicast or via a mesh of unicast relations, or a combination of these.

SIP invitations used to create sessions carry session descriptions which allow participants to agree on a set of compatible media types. SIP supports user mobility by proxying and redirecting requests to the user's current location. Users can register their current location. SIP is not tied to any particular conference control protocol. SIP is designed to be independent of the lower-layer transport protocol and can be extended with additional capabilities.

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# 1 Introduction

## 1.1 Overview of SIP Functionality

The Session Initiation Protocol (SIP) is an application-layer control protocol that can establish, modify and terminate multimedia sessions or calls. These multimedia sessions include multimedia conferences, distance learning, Internet telephony and similar applications. SIP can invite both persons and "robots", such as a media storage service. SIP can invite parties to both unicast and multicast sessions; the initiator does not necessarily have to be a member of the session to which it is inviting. Media and participants can be added to an existing session.

SIP can be used to initiate sessions as well as invite members to sessions that have been advertised and established by other means. Sessions can be advertised using multicast protocols such as SAP, electronic mail, news groups, web pages or directories (LDAP), among others.

SIP transparently supports name mapping and redirection services, allowing the implementation of ISDN and Intelligent Network telephony subscriber services. These facilities also enable personal mobility. In the parlance of telecommunications intelligent network services, this is defined as: "Personal mobility is the ability of end users to originate and receive calls and access subscribed telecommunication services on any terminal in any location, and the ability of the network to identify end users as they move. Personal mobility is based on the use of a unique personal identity (i.e., personal number)." [1]. Personal mobility complements terminal mobility, i.e., the ability to maintain communications when moving a single end system from one subnet to another.

SIP supports five facets of establishing and terminating multimedia communications:

User location: determination of the end system to be used for communication;

User capabilities: determination of the media and media parameters to be used;

User availability: determination of the willingness of the called party to engage in communications;

Call setup: "ringing", establishment of call parameters at both called and calling party;

Call handling: including transfer and termination of calls.

SIP can also initiate multi-party calls using a multipoint control unit (MCU) or fully-meshed interconnection instead of multicast. Internet telephony gateways that connect Public Switched Telephone Network (PSTN) parties can also use SIP to set up calls between them.

SIP is designed as part of the overall IETF multimedia data and control architecture currently incorporating protocols such as RSVP (RFC 2205 [2]) for reserving network resources, the real-time transport protocol (RTP) (RFC 1889 [3]) for transporting real-time data and providing QOS feedback, the real-time streaming protocol (RTSP) (RFC 2326 [4]) for controlling delivery of streaming media, the session announcement protocol (SAP) [5] for advertising multimedia sessions via multicast and the session description protocol (SDP) (RFC 2327 [6]) for describing multimedia sessions. However, the functionality and operation of SIP does not depend on any of these protocols.

SIP can also be used in conjunction with other call setup and signaling protocols. In that mode, an end system uses SIP exchanges to determine the appropriate end system address and protocol from a given address that is protocol-independent. For example, SIP could be used to determine that the party can be reached via H.323 [7], obtain the H.245 [8] gateway and user address and then use H.225.0 [9] to establish the call.

In another example, SIP might be used to determine that the callee is reachable via the PSTN and indicate the phone number to be called, possibly suggesting an Internet-to-PSTN gateway to be used.

SIP does not offer conference control services such as floor control or voting and does not prescribe how a conference is to be managed, but SIP can be used to introduce conference control protocols. SIP does not allocate multicast addresses.

SIP can invite users to sessions with and without resource reservation. SIP does not reserve resources, but can convey to the invited system the information necessary to do this.

## 1.2 Terminology

In this document, the key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" are to be interpreted as described in RFC 2119 [10] and indicate requirement levels for compliant SIP implementations.

## 1.3 Definitions

This specification uses a number of terms to refer to the roles played by participants in SIP communications. The definitions of client, server and proxy are similar to those used by the Hypertext Transport Protocol (HTTP) (RFC 2068 [11]). The terms and generic syntax of URI and URL are defined in RFC 2396 [12]. The following terms have special significance for SIP.

Call: A call consists of all participants in a conference invited by a common source. A SIP call is identified by a globally unique call-id (Section 6.12). Thus, if a user is, for example, invited to the same multicast session by several people, each of these invitations will be a unique call. A point-to-point Internet telephony conversation maps into a single SIP call. In a multiparty conference unit (MCU) based call-in conference, each participant uses a separate call to invite himself to the MCU.

Call leg: A call leg is identified by the combination of Call-ID, To and From.

Client: An application program that sends SIP requests. Clients may or may not interact directly with a human user. User agents and proxies contain clients (and servers).

Conference: A multimedia session (see below), identified by a common session description. A conference can have zero or more members and includes the cases of a multicast conference, a full-mesh conference and a two-party "telephone call", as well as combinations of these. Any number of calls can be used to create a conference.

Downstream: Requests sent in the direction from the caller to the callee (i.e., user agent client to user agent server).

Final response: A response that terminates a SIP transaction, as opposed to a provisional response that does not. All 2xx, 3xx, 4xx, 5xx and 6xx responses are final.

Initiator, calling party, caller: The party initiating a conference invitation. Note that the calling party does not have to be the same as the one creating the conference.

Invitation: A request sent to a user (or service) requesting participation in a session. A successful SIP invitation consists of two transactions: an INVITE request followed by an ACK request.

Invitee, invited user, called party, callee: The person or service that the calling party is trying to invite to a conference.

Isomorphic request or response: Two requests or responses are defined to be isomorphic for the purposes of this document if they have the same values for the Call-ID, To, From and CSeq header fields. In addition, isomorphic requests have to have the same Request-URI.

Location server: See location service.

Location service: A location service is used by a SIP redirect or proxy server to obtain information about a callee’s possible location(s). Location services are offered by location servers. Location servers MAY be co-located with a SIP server, but the manner in which a SIP server requests location services is beyond the scope of this document.

Parallel search: In a parallel search, a proxy issues several requests to possible user locations upon receiving an incoming request. Rather than issuing one request and then waiting for the final response before issuing the next request as in a sequential search , a parallel search issues requests without waiting for the result of previous requests.

Provisional response: A response used by the server to indicate progress, but that does not terminate a SIP transaction. 1xx responses are provisional, other responses are considered final.

Proxy, proxy server: An intermediary program that acts as both a server and a client for the purpose of making requests on behalf of other clients. Requests are serviced internally or by passing them on, possibly after translation, to other servers. A proxy interprets, and, if necessary, rewrites a request message before forwarding it.

Redirect server: A redirect server is a server that accepts a SIP request, maps the address into zero or more new addresses and returns these addresses to the client. Unlike a proxy server , it does not initiate its own SIP request. Unlike a user agent server , it does not accept calls.

Registrar: A registrar is a server that accepts REGISTER requests. A registrar is typically co-located with a proxy or redirect server and MAY offer location services.

Ringback: Ringback is the signaling tone produced by the calling client's application indicating that a called party is being alerted (ringing).

Server: A server is an application program that accepts requests in order to service requests and sends back responses to those requests. Servers are either proxy, redirect or user agent servers or registrars.

Session: From the SDP specification: "A multimedia session is a set of multimedia senders and receivers and the data streams flowing from senders to receivers. A multimedia conference is an example of a multimedia session." (RFC 2327 [6]) (A session as defined for SDP can comprise one or more RTP sessions.) As defined, a callee can be invited several times, by different calls, to the same session. If SDP is used, a session is defined by the concatenation of the user name , session id , network type , address type and address elements in the origin field.

(SIP) transaction: A SIP transaction occurs between a client and a server and comprises all messages from the first request sent from the client to the server up to a final (non-1xx) response sent from the server to the client. A transaction is identified by the CSeq sequence number (Section 6.17) within a single call leg. The ACK request has the same CSeq number as the corresponding INVITE request, but comprises a transaction of its own.

Upstream: Responses sent in the direction from the user agent server to the user agent client.

URL-encoded: A character string encoded according to RFC 1738, Section 2.2 [13].

User agent client (UAC), calling user agent: A user agent client is a client application that initiates the SIP request.

User agent server (UAS), called user agent: A user agent server is a server application that contacts the user when a SIP request is received and that returns a response on behalf of the user. The response accepts, rejects or redirects the request.

User agent (UA): An application which contains both a user agent client and user agent server.

An application program MAY be capable of acting both as a client and a server. For example, a typical multimedia conference control application would act as a user agent client to initiate calls or to invite others to conferences and as a user agent server to accept invitations. The properties of the different SIP server types are summarized in Table 1.

| property | redirect server | proxy server | user agent server | registrar |
|---|---|---|---|---|
| also acts as a SIP client | no | yes | no | no |
| returns 1xx status | yes | yes | yes | yes |
| returns 2xx status | no | yes | yes | yes |
| returns 3xx status | yes | yes | yes | yes |
| returns 4xx status | yes | yes | yes | yes |
| returns 5xx status | yes | yes | yes | yes |
| returns 6xx status | no | yes | yes | yes |
| inserts Via header | no | yes | no | no |
| accepts ACK | yes | yes | yes | no |

Table 1: Properties of the different SIP server types

## 1.4 Overview of SIP Operation

This section explains the basic protocol functionality and operation. Callers and callees are identified by SIP addresses, described in Section 1.4.1. When making a SIP call, a caller first locates the appropriate server (Section 1.4.2) and then sends a SIP request (Section 1.4.3). The most common SIP operation is the invitation (Section 1.4.4). Instead of directly reaching the intended callee, a SIP request may be redirected or may trigger a chain of new SIP requests by proxies (Section 1.4.5). Users can register their location(s) with SIP servers (Section 4.2.6).

### 1.4.1 SIP Addressing

The "objects" addressed by SIP are users at hosts, identified by a SIP URL. The SIP URL takes a form similar to a mailto or telnet URL, i.e., user@host. The user part is a user name or a telephone number. The host part is either a domain name or a numeric network address. See section 2 for a detailed discussion of SIP URL's.

A user's SIP address can be obtained out-of-band, can be learned via existing media agents, can be included in some mailers' message headers, or can be recorded during previous invitation interactions. In many cases, a user's SIP URL can be guessed from their email address.

A SIP URL address can designate an individual (possibly located at one of several end systems), the first available person from a group of individuals or a whole group. The form of the address, for example, sip:sales@example.com , is not sufficient, in general, to determine the intent of the caller.

If a user or service chooses to be reachable at an address that is guessable from the person's name and organizational affiliation, the traditional method of ensuring privacy by having an unlisted "phone" number is compromised. However, unlike traditional telephony, SIP offers authentication and access control mechanisms and can avail itself of lower-layer security mechanisms, so that client software can reject unauthorized or undesired call attempts.

### 1.4.2 Locating a SIP Server

When a client wishes to send a request, the client either sends it to a locally configured SIP proxy server (as in HTTP), independent of the Request-URI, or sends it to the IP address and port corresponding to the Request-URI.

For the latter case, the client must determine the protocol, port and IP address of a server to which to send the request. A client SHOULD follow the steps below to obtain this information, but MAY follow the alternative, optional procedure defined in Appendix D. At each step, unless stated otherwise, the client SHOULD try to contact a server at the port number listed in the Request-URI. If no port number is present in the Request-URI, the client uses port 5060. If the Request-URI specifies a protocol (TCP or UDP), the client contacts the server using that protocol. If no protocol is specified, the client tries UDP (if UDP is supported). If the attempt fails, or if the client doesn't support UDP but supports TCP, it then tries TCP.

A client SHOULD be able to interpret explicit network notifications (such as ICMP messages) which indicate that a server is not reachable, rather than relying solely on timeouts. (For socket-based programs: For TCP, connect() returns ECONNREFUSED if the client could not connect to a server at that address. For UDP, the socket needs to be bound to the destination address using connect() rather than sendto() or similar so that a second write() fails with ECONNREFUSED if there is no server listening) If the client finds the server is not reachable at a particular address, it SHOULD behave as if it had received a 400-class error response to that request.

The client tries to find one or more addresses for the SIP server by querying DNS. The procedure is as follows:

1. If the host portion of the Request-URI is an IP address, the client contacts the server at the given address. Otherwise, the client proceeds to the next step.

2. The client queries the DNS server for address records for the host portion of the Request-URI. If the DNS server returns no address records, the client stops, as it has been unable to locate a server. By address record, we mean A RR's, AAAA RR's, or other similar address records, chosen according to the client's network protocol capabilities.

There are no mandatory rules on how to select a host name for a SIP server. Users are encouraged to name their SIP servers using the sip.domainname (i.e., sip.example.com) convention, as specified in RFC 2219 [16]. Users may only know an email address instead of a full SIP URL for a callee, however. In that case, implementations may be able to increase the likelihood of reaching a SIP server for that domain by constructing a SIP URL from that email address by prefixing the host name with "sip.". In the future, this mechanism is likely to become unnecessary as better DNS techniques, such as the one in Appendix D, become widely available.

A client MAY cache a successful DNS query result. A successful query is one which contained records in the answer, and a server was contacted at one of the addresses from the answer. When the client wishes to send a request to the same host, it MUST start the search as if it had just received this answer from the name server. The client MUST follow the procedures in RFC1035 [15] regarding DNS cache invalidation when the DNS time-to-live expires.

### 1.4.3 SIP Transaction

Once the host part has been resolved to a SIP server, the client sends one or more SIP requests to that server and receives one or more responses from the server. A request (and its retransmissions) together with the responses triggered by that request make up a SIP transaction. All responses to a request contain the same values in the Call-ID, CSeq, To, and From fields (with the possible addition of a tag in the To field (section 6.37)). This allows responses to be matched with requests. The ACK request following an INVITE is not part of the transaction since it may traverse a different set of hosts.

If TCP is used, request and responses within a single SIP transaction are carried over the same TCP connection (see Section 10). Several SIP requests from the same client to the same server MAY use the same TCP connection or MAY use a new connection for each request.

If the client sent the request via unicast UDP, the response is sent to the address contained in the next Via header field (Section 6.40) of the response. If the request is sent via multicast UDP, the response is directed to the same multicast address and destination port. For UDP, reliability is achieved using retransmission (Section 10).

The SIP message format and operation is independent of the transport protocol.

### 1.4.4 SIP Invitation

A successful SIP invitation consists of two requests, INVITE followed by ACK. The INVITE (Section 4.2.1) request asks the callee to join a particular conference or establish a two-party conversation. After the callee has agreed to participate in the call, the caller confirms that it has received that response by sending an ACK (Section 4.2.2) request. If the caller no longer wants to participate in the call, it sends a BYE request instead of an ACK.

The INVITE request typically contains a session description, for example written in SDP (RFC 2327 [6]) format, that provides the called party with enough information to join the session. For multicast sessions, the session description enumerates the media types and formats that are allowed to be distributed to that session. For a unicast session, the session description enumerates the media types and formats that the caller is willing to use and where it wishes the media data to be sent. In either case, if the callee wishes to accept the call, it responds to the invitation by returning a similar description listing the media it wishes to use. For a multicast session, the callee SHOULD only return a session description if it is unable to receive the media indicated in the caller’s description or wants to receive data via unicast.

The protocol exchanges for the INVITE method are shown in Fig. 1 for a proxy server and in Fig. 2 for a redirect server. (Note that the messages shown in the figures have been abbreviated slightly.) In Fig. 1, the proxy server accepts the INVITE request (step 1), contacts the location service with all or parts of the address (step 2) and obtains a more precise location (step 3). The proxy server then issues a SIP INVITE request to the address(es) returned by the location service (step 4). The user agent server alerts the user (step 5) and returns a success indication to the proxy server (step

6). The proxy server then returns the success result to the original caller (step 7). The receipt of this message is confirmed by the caller using an ACK request, which is forwarded to the callee (steps 8 and 9). Note that an ACK can also be sent directly to the callee, bypassing the proxy. All requests and responses have the same Call-ID.

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                                         :     | hgs@lab                 :
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:                     : 7: 200 OK        :    (      )6: 200 OK (      ) :
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+.....................+                  +...............................+

  ====> SIP request
  ....> SIP response

   ^
   |    non-SIP protocols
   |
```

Figure 1: Example of SIP proxy server

The redirect server shown in Fig. 2 accepts the INVITE request (step 1), contacts the location service as before (steps 2 and 3) and, instead of contacting the newly found address itself, returns the address to the caller (step 4), which is then acknowledged via an ACK request (step 5). The caller issues a new request, with the same call-ID but a higher CSeq, to the address returned by the first server (step 6). In the example, the call succeeds (step 7). The caller and callee complete the handshake with an ACK (step 8).

The next section discusses what happens if the location service returns more than one possible alternative.

### 1.4.5 Locating a User

A callee may move between a number of different end systems over time. These locations can be dynamically registered with the SIP server (Sections 1.4.7, 4.2.6). A location server MAY also use one or more other protocols, such as finger (RFC 1288 [17]), rwhois (RFC 2167 [18]), LDAP (RFC 1777 [19]), multicast-based protocols [20] or operating-system dependent mechanisms to actively determine the end system where a user might be reachable. A location server MAY return several locations because the user is logged in at several hosts simultaneously or because the location server has (temporarily) inaccurate information. The SIP server combines the results to yield a list of a zero or more locations.

The action taken on receiving a list of locations varies with the type of SIP server. A SIP redirect server returns the list to the client as Contact headers (Section 6.13). A SIP proxy server can sequentially or in parallel try the addresses until the call is successful (2xx response) or the callee has declined the call (6xx response). With sequential attempts, a proxy server can implement an "anycast" service.

If a proxy server forwards a SIP request, it MUST add itself to the beginning of the list of forwarders noted in the Via (Section 6.40) headers. The Via trace ensures that replies can take the same path back, ensuring correct operation through compliant firewalls and avoiding request loops. On the response path, each host MUST remove its Via, so that routing internal information is hidden from the callee and outside networks. A proxy server MUST check that it does not generate a request to a host listed in the Via sent-by, via-received or via-maddr parameters (Section 6.40). (Note: If a host has several names or network addresses, this does not always work. Thus, each host also checks if it is part of the Via list.)

A SIP invitation may traverse more than one SIP proxy server. If one of these "forks" the request, i.e., issues more than one request in response to receiving the invitation request, it is possible that a client is reached, independently, by more than one copy of the invitation request. Each of these copies bears the same Call-ID. The user agent MUST return the same status response returned in the first response. Duplicate requests are not an error.

### 1.4.6 Changing an Existing Session

In some circumstances, it is desirable to change the parameters of an existing session. This is done by re-issuing the INVITE, using the same Call-ID, but a new or different body or header fields to convey the new information. This re INVITE MUST have a higher CSeq than any previous request from the client to the server.

For example, two parties may have been conversing and then want to add a third party, switching to multicast for efficiency. One of the participants invites the third party with the new multicast address and simultaneously sends an INVITE to the second party, with the new multicast session description, but with the old call identifier.

### 1.4.7 Registration Services

The REGISTER request allows a client to let a proxy or redirect server know at which address(es) it can be reached. A client MAY also use it to install call handling features at the server.

## 1.5 Protocol Properties

### 1.5.1 Minimal State

A single conference session or call involves one or more SIP request-response transactions. Proxy servers do not have to keep state for a particular call, however, they MAY maintain state for a single SIP transaction, as discussed in Section 12. For efficiency, a server MAY cache the results of location service requests.

### 1.5.2 Lower-Layer-Protocol Neutral

SIP makes minimal assumptions about the underlying transport and network-layer protocols. The lower-layer can provide either a packet or a byte stream service, with reliable or unreliable service.

In an Internet context, SIP is able to utilize both UDP and TCP as transport protocols, among others. UDP allows the application to more carefully control the timing of messages and their retransmission, to perform parallel searches without requiring TCP connection state for each outstanding request, and to use multicast. Routers can more readily snoop SIP UDP packets. TCP allows easier passage through existing firewalls.

```
                                         +....... cs.columbia.edu .......+
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                                         : (~~~~~~~~~~)                  :
                                         : ( location )                  :
                                         : ( service  )                  :
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                                         :    ^   |                      :
                                         :    | hgs@lab                  :
                                         :   2|  3|                      :
                                         :    |   |                      :
                                         : henning|                      :
+.. cs.tu-berlin.de ..+ 1: INVITE        :    |   |                      :
:                     :    henning@cs.col:    |   \/                     :
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        | . | 6: INVITE hgs@lab.cs.columbia.edu                 (~~~~~~) :
        | . ==================================================> (      ) :
        | ..................................................... (      ) :
        |     7: 200 OK                  :                      ( lab  ) :
        |                                :                      (      ) :
        |     8: ACK                     :                      (      ) :
        ======================================================> (~~~~~~) :
                                         +...............................+

  ====> SIP request
  ....> SIP response

    ^
    |   non-SIP protocols
    |
```

Figure 2: Example of SIP redirect server

When TCP is used, SIP can use one or more connections to attempt to contact a user or to modify parameters of an existing conference. Different SIP requests for the same SIP call MAY use different TCP connections or a single persistent connection, as appropriate.

For concreteness, this document will only refer to Internet protocols. However, SIP MAY also be used directly with protocols such as ATM AAL5, IPX, frame relay or X.25. The necessary naming conventions are beyond the scope of this document. User agents SHOULD implement both UDP and TCP transport. Proxy, registrar, and redirect servers MUST implement both UDP and TCP transport.

### 1.5.3 Text-Based

SIP is text-based, using ISO 10646 in UTF-8 encoding throughout. This allows easy implementation in languages such as Java, Tcl and Perl, allows easy debugging, and most importantly, makes SIP flexible and extensible. As SIP is used for initiating multimedia conferences rather than delivering media data, it is believed that the additional overhead of using a text-based protocol is not significant.

## 2 SIP Uniform Resource Locators

SIP URLs are used within SIP messages to indicate the originator (From), current destination (Request-URI) and final recipient (To) of a SIP request, and to specify redirection addresses (Contact). A SIP URL can also be embedded in web pages or other hyperlinks to indicate that a particular user or service can be called via SIP. When used as a hyperlink, the SIP URL indicates the use of the INVITE method.

The SIP URL scheme is defined to allow setting SIP request-header fields and the SIP message-body.

> This corresponds to the use of mailto: URLs. It makes it possible, for example, to specify the subject, urgency or media types of calls initiated through a web page or as part of an email message.

A SIP URL follows the guidelines of RFC 2396 [12] and has the syntax shown in Fig. 3. The syntax is described using Augmented Backus-Naur Form (See Section C). Note that reserved characters have to be escaped and that the "set of characters reserved within any given URI component is defined by that component. In general, a character is reserved if the semantics of the URI changes if the character is replaced with its escaped US-ASCII encoding" [12].

```
SIP-URL         = "sip:" [ userinfo "@" ] hostport
                  url-parameters [ headers ]
userinfo        = user [ ":" password ]
user            = *( unreserved | escaped
                | "&" | "=" | "+" | "$" | "," )
password        = *( unreserved | escaped
                | "&" | "=" | "+" | "$" | "," )
hostport        = host [ ":" port ]
host            = hostname | IPv4address
hostname        = *( domainlabel "." ) toplabel [ "." ]
domainlabel     = alphanum | alphanum *( alphanum | "-" ) alphanum
toplabel        = alpha | alpha *( alphanum | "-" ) alphanum
IPv4address     = 1*digit "." 1*digit "." 1*digit "." 1*digit
port            = *digit
url-parameters  = *( ";" url-parameter )
url-parameter   = transport-param | user-param | method-param
                | ttl-param | maddr-param | other-param
transport-param = "transport=" ( "udp" | "tcp" )
ttl-param       = "ttl=" ttl
ttl             = 1*3DIGIT       ; 0 to 255
maddr-param     = "maddr=" host
user-param      = "user=" ( "phone" | "ip" )
method-param    = "method=" Method
tag-param       = "tag=" UUID
UUID            = 1*( hex | "-" )
other-param     = ( token | ( token "=" ( token | quoted-string )))
headers         = "?" header *( "&" header )
header          = hname "=" hvalue
hname           = 1*uric
hvalue          = *uric
uric            = reserved | unreserved | escaped
reserved        = ";" | "/" | "?" | ":" | "@" | "&" | "=" | "+" |
                  "$" | ","
digits          = 1*DIGIT
```

Figure 3: SIP URL syntax

The URI character classes referenced above are described in Appendix C.

The components of the SIP URI have the following meanings.

```
telephone-subscriber  = global-phone-number | local-phone-number
   global-phone-number   = "+" 1*phonedigit [isdn-subaddress]
                             [post-dial]
   local-phone-number    = 1*(phonedigit | dtmf-digit |
                             pause-character) [isdn-subaddress]
                             [post-dial]
   isdn-subaddress       = ";isub=" 1*phonedigit
   post-dial             = ";postd=" 1*(phonedigit | dtmf-digit
                         |  pause-character)
   phonedigit            = DIGIT | visual-separator
   visual-separator      = "-" | "."
   pause-character       = one-second-pause | wait-for-dial-tone
   one-second-pause      = "p"
   wait-for-dial-tone    = "w"
   dtmf-digit            = "*" | "#" | "A" | "B" | "C" | "D"
```

Figure 4: SIP URL syntax; telephone subscriber

user: If the host is an Internet telephony gateway, the user field MAY also encode a telephone number using the notation of telephone-subscriber (Fig. 4). The telephone number is a special case of a user name and cannot be distinguished by a BNF. Thus, a URL parameter, user, is added to distinguish telephone numbers from user names. The phone identifier is to be used when connecting to a telephony gateway. Even without this parameter, recipients of SIP URLs MAY interpret the pre-@ part as a phone number if local restrictions on the name space for user name allow it.

password: The SIP scheme MAY use the format "user:password" in the userinfo field. The use of passwords in the userinfo is NOT RECOMMENDED, because the passing of authentication information in clear text (such as URIs) has proven to be a security risk in almost every case where it has been used.

host: The mailto: URL and RFC 822 email addresses require that numeric host addresses ("host numbers") are enclosed in square brackets (presumably, since host names might be numeric), while host numbers without brackets are used for all other URLs. The SIP URL requires the latter form, without brackets.

The issue of IPv6 literal addresses in URLs is being looked at elsewhere in the IETF. SIP implementers are advised to keep up to date on that activity.

port: The port number to send a request to. If not present, the procedures outlined in Section 1.4.2 are used to determine the port number to send a request to.

URL parameters: SIP URLs can define specific parameters of the request. URL parameters are added after the host component and are separated by semi-colons. The transport parameter determines the the transport mechanism (UDP or TCP). UDP is to be assumed when no explicit transport parameter is included. The maddr parameter provides the server address to be contacted for this user, overriding the address supplied in the host field. This address is typically a multicast address, but could also be the address of a backup server. The ttl parameter determines the time-to-live value of the UDP multicast packet and MUST only be used if maddr is a multicast address and the transport protocol is UDP. The user parameter was described above. For example, to specify to call j.doe@big.com using multicast to 239.255.255.1 with a ttl of 15, the following URL would be used:

```
sip:j.doe@big.com;maddr=239.255.255.1;ttl=15
```

The transport, maddr, and ttl parameters MUST NOT be used in the From and To header fields and the Request-URI; they are ignored if present.

Headers: Headers of the SIP request can be defined with the "?" mechanism within a SIP URL. The special hname "body" indicates that the associated hvalue is the message-body of the SIP INVITE request. Headers MUST NOT be used in the From and To header fields and the Request-URI; they are ignored if present. hname and hvalue are encodings of a SIP header name and value, respectively. All URL reserved characters in the header names and values MUST be escaped.

Method: The method of the SIP request can be specified with the method parameter. This parameter MUST NOT be used in the From and To header fields and the Request-URI; they are ignored if present.

Table 2 summarizes where the components of the SIP URL can be used and what default values they assume if not present.

Examples of SIP URLs are:

| | default | Req.-URI | To | From | Contact | external |
|---|---|---|---|---|---|---|
| user | -- | x | x | x | x | x |
| password | -- | x | x | | x | x |
| host | mandatory | x | x | x | x | x |
| port | 5060 | x | x | x | x | x |
| user-param | ip | x | x | x | x | x |
| method | INVITE | | | | x | x |
| maddr-param | -- | | | | x | x |
| ttl-param | 1 | | | | x | x |
| transp.-param | -- | | | | x | x |
| headers | -- | | | | x | x |

Table 2: Use and default values of URL components for SIP headers, Request-URI and references

```
sip:j.doe@big.com
sip:j.doe:secret@big.com;transport=tcp
sip:j.doe@big.com?subject=project
sip:+1-212-555-1212:1234@gateway.com;user=phone
sip:1212@gateway.com
sip:alice@10.1.2.3
sip:alice@example.com
sip:alice%40example.com@gateway.com
sip:alice@registrar.com;method=REGISTER
```

Within a SIP message, URLs are used to indicate the source and intended destination of a request, redirection addresses and the current destination of a request. Normally all these fields will contain SIP URLs.

SIP URLs are case-insensitive, so that for example the two URLs sip:j.doe@example.com and SIP:J.Doe@Example.com are equivalent. All URL parameters are included when comparing SIP URLs for equality.

SIP header fields MAY contain non-SIP URLs. As an example, if a call from a telephone is relayed to the Internet via SIP, the SIP From header field might contain a phone URL.

## 3 SIP Message Overview

SIP is a text-based protocol and uses the ISO 10646 character set in UTF-8 encoding (RFC 2279 [21]). Senders MUST terminate lines with a CRLF, but receivers MUST also interpret CR and LF by themselves as line terminators.

Except for the above difference in character sets, much of the message syntax is and header fields are identical to HTTP/1.1; rather than repeating the syntax and semantics here we use [HX.Y] to refer to Section X.Y of the current HTTP/1.1 specification (RFC 2068 [11]). In addition, we describe SIP in both prose and an augmented Backus-Naur form (ABNF). See section C for an overview of ABNF.

Note, however, that SIP is not an extension of HTTP.

Unlike HTTP, SIP MAY use UDP. When sent over TCP or UDP, multiple SIP transactions can be carried in a single TCP connection or UDP datagram. UDP datagrams, including all headers, SHOULD NOT be larger than the path maximum transmission unit (MTU) if the MTU is known, or 1500 bytes if the MTU is unknown.

> The 1500 bytes accommodates encapsulation within the "typical" ethernet MTU without IP fragmentation. Recent studies [22] indicate that an MTU of 1500 bytes is a reasonable assumption. The next lower common MTU values are 1006 bytes for SLIP and 296 for low-delay PPP (RFC 1191 [23]). Thus, another reasonable value would be a message size of 950 bytes, to accommodate packet headers within the SLIP MTU without fragmentation.

A SIP message is either a request from a client to a server, or a response from a server to a client.

```
SIP-message  =  Request | Response
```

Both Request (section 4) and Response (section 5) messages use the generic-message format of RFC 822 [24] for transferring entities (the body of the message). Both types of messages consist of a start-line, one or more header fields (also known as "headers"), an empty line (i.e., a line with nothing preceding the carriage-return line-feed (CRLF)) indicating the end of the header fields, and an optional message-body. To avoid confusion with similar-named headers in HTTP, we refer to the headers describing the message body as entity headers. These components are described in detail in the upcoming sections.

```
generic-message  =  start-line
                    *message-header
```

```
                          CRLF
                          [ message-body ]

        start-line       =  Request-Line |     ;Section 4.1
                            Status-Line        ;Section 5.1



        message-header  =  ( general-header
                           | request-header
                           | response-header
                           | entity-header )
```

In the interest of robustness, any leading empty line(s) MUST be ignored. In other words, if the Request or Response message begins with one or more CRLF, CR, or LFs, these characters MUST be ignored.

# 4 Request

The Request message format is shown below:

```
        Request  =  Request-Line       ;  Section 4.1
                    *( general-header
                    | request-header
                    | entity-header )
                    CRLF
                    [ message-body ]   ;  Section 8
```

## 4.1 Request-Line

The Request-Line begins with a method token, followed by the Request-URI and the protocol version, and ending with CRLF. The elements are separated by SP characters. No CR or LF are allowed except in the final CRLF sequence.

```
        Request-Line  =  Method SP Request-URI SP SIP-Version CRLF
```

```
general-header   =  Accept                 ; Section 6.7
                 |  Accept-Encoding        ; Section 6.8
                 |  Accept-Language        ; Section 6.9
                 |  Call-ID                ; Section 6.12
                 |  Contact                ; Section 6.13
                 |  CSeq                   ; Section 6.17
                 |  Date                   ; Section 6.18
                 |  Encryption             ; Section 6.19
                 |  Expires                ; Section 6.20
                 |  From                   ; Section 6.21
                 |  Record-Route           ; Section 6.29
                 |  Timestamp              ; Section 6.36
                 |  To                     ; Section 6.37
                 |  Via                    ; Section 6.40
entity-header    =  Content-Encoding       ; Section 6.14
                 |  Content-Length         ; Section 6.15
                 |  Content-Type           ; Section 6.16
request-header   =  Authorization          ; Section 6.11
                 |  Contact                ; Section 6.13
                 |  Hide                   ; Section 6.22
                 |  Max-Forwards           ; Section 6.23
                 |  Organization           ; Section 6.24
                 |  Priority               ; Section 6.25
                 |  Proxy-Authorization    ; Section 6.27
                 |  Proxy-Require          ; Section 6.28
                 |  Route                  ; Section 6.33
                 |  Require                ; Section 6.30
                 |  Response-Key           ; Section 6.31
                 |  Subject                ; Section 6.35
                 |  User-Agent             ; Section 6.39
response-header  =  Allow                  ; Section 6.10
                 |  Proxy-Authenticate     ; Section 6.26
                 |  Retry-After            ; Section 6.32
                 |  Server                 ; Section 6.34
                 |  Unsupported            ; Section 6.38
                 |  Warning                ; Section 6.41
                 |  WWW-Authenticate       ; Section 6.42
```

Table 3: SIP headers

## 4.2 Methods

The methods are defined below. Methods that are not supported by a proxy or redirect server are treated by that server as if they were an OPTIONS method and forwarded accordingly. Methods that are not supported by a user agent server or registrar cause a 501 (Not Implemented) response to be returned (Section 7). As in HTTP, the Method token is case-sensitive.

```
Method  =  "INVITE" | "ACK" | "OPTIONS" | "BYE"
           | "CANCEL" | "REGISTER"
```

### 4.2.1 INVITE

The INVITE method indicates that the user or service is being invited to participate in a session. The message body contains a description of the session to which the callee is being invited. For two-party calls, the caller indicates the type of media it is able to receive and possibly the media it is willing to send as well as their parameters such as network destination. A success response MUST indicate in its message body which media the callee wishes to receive and MAY indicate the media the callee is going to send.

> Not all session description formats have the ability to indicate sending media.

A server MAY automatically respond to an invitation for a conference the user is already participating in, identified either by the SIP Call-ID or a globally unique identifier within the session description, with a 200 (OK) response.

If a user agent receives an INVITE request for an existing call leg with a higher CSeq sequence number than any previous INVITE for the same Call-ID, it MUST check any version identifiers in the session description or, if there are no version identifiers, the content of the session description to see if it has changed. It MUST also inspect any other header fields for changes. If there is a change, the user agent MUST update any internal state or information generated as a result of that header. If the session description has changed, the user agent server MUST adjust the session parameters accordingly, possibly after asking the user for confirmation. (Versioning of the session description can be used to accommodate the capabilities of new arrivals to a conference, add or delete media or change from a unicast to a multicast conference.)

This method MUST be supported by SIP proxy, redirect and user agent servers as well as clients.

### 4.2.2 ACK

The ACK request confirms that the client has received a final response to an INVITE request. (ACK is used only with INVITE requests.) 2xx responses are acknowledged by client user agents, all other final responses by the first proxy or client user agent to receive the response. The Via is always initialized to the host that originates the ACK request, i.e., the client user agent after a 2xx response or the first proxy to receive a non-2xx final response. The ACK request is forwarded as the corresponding INVITE request, based on its Request-URI. See Section 10 for details.

The ACK request MAY contain a message body with the final session description to be used by the callee. If the ACK message body is empty, the callee uses the session description in the INVITE request.

A proxy server receiving an ACK request after having sent a 3xx, 4xx, 5xx, or 6xx response must make a determination about whether the ACK is for it, or for some user agent or proxy server further downstream. This determination is made by examining the tag in the To field. If the tag in the ACK To header field matches the tag in the To header field of the response, and the From, CSeq and Call-ID header fields in the response match those in the ACK, the ACK is meant for the proxy server. Otherwise, the ACK SHOULD be proxied downstream as any other request.

> It is possible for a user agent client or proxy server to receive multiple 3xx, 4xx, 5xx, and 6xx responses to a request along a single branch. This can happen under various error conditions, typically when a forking proxy transitions from stateful to stateless before receiving all responses. The various responses will all be identical, except for the tag in the To field, which is different for each one. It can therefore be used as a means to disambiguate them.

This method MUST be supported by SIP proxy, redirect and user agent servers as well as clients.

### 4.2.3 OPTIONS

The server is being queried as to its capabilities. A server that believes it can contact the user, such as a user agent where the user is logged in and has been recently active, MAY respond to this request with a capability set. A called user agent MAY return a status reflecting how it would have responded to an invitation, e.g., 600 (Busy). Such a server SHOULD return an Allow header field indicating the methods that it supports. Proxy and redirect servers simply forward the request without indicating their capabilities.

This method MUST be supported by SIP proxy, redirect and user agent servers, registrars and clients.

### 4.2.4 BYE

The user agent client uses BYE to indicate to the server that it wishes to release the call. A BYE request is forwarded like an INVITE request and MAY be issued by either caller or callee. A party to a call SHOULD issue a BYE request before releasing a call ("hanging up"). A party receiving a BYE request MUST cease transmitting media streams specifically directed at the party issuing the BYE request.

If the INVITE request contained a Contact header, the callee SHOULD send a BYE request to that address rather than the From address.

This method MUST be supported by proxy servers and SHOULD be supported by redirect and user agent SIP servers.

### 4.2.5 CANCEL

The CANCEL request cancels a pending request with the same Call-ID, To, From and CSeq (sequence number only) header field values, but does not affect a completed request. (A request is considered completed if the server has returned a final status response.)

A user agent client or proxy client MAY issue a CANCEL request at any time. A proxy, in particular, MAY choose to send a CANCEL to destinations that have not yet returned a final response after it has received a 2xx or 6xx response for one or more of the parallel-search requests. A proxy that receives a CANCEL request forwards the request to all destinations with pending requests.

The Call-ID, To, the numeric part of CSeq and From headers in the CANCEL request are identical to those in the original request. This allows a CANCEL request to be matched with the request it cancels. However, to allow the client to distinguish responses to the CANCEL from those to the original request, the CSeq Method component is set to CANCEL. The Via header field is initialized to the proxy issuing the CANCEL request. (Thus, responses to this CANCEL request only reach the issuing proxy.)

Once a user agent server has received a CANCEL, it MUST NOT issue a 2xx response for the cancelled original request.

A redirect or user agent server receiving a CANCEL request responds with a status of 200 (OK) if the transaction exists and a status of 481 (Transaction Does Not Exist) if not, but takes no further action. In particular, any existing call is unaffected.

> The BYE request cannot be used to cancel branches of a parallel search, since several branches may, through intermediate proxies, find the same user agent server and then terminate the call. To terminate a call instead of just pending searches, the UAC must use BYE instead of or in addition to CANCEL. While CANCEL can terminate any pending request other than ACK or CANCEL, it is typically useful only for INVITE. 200 responses to INVITE and 200 responses to CANCEL are distinguished by the method in the Cseq header field, so there is no ambiguity.

This method MUST be supported by proxy servers and SHOULD be supported by all other SIP server types.

#### 4.2.6 REGISTER

A client uses the REGISTER method to register the address listed in the To header field with a SIP server.

A user agent MAY register with a local server on startup by sending a REGISTER request to the well-known "all SIP servers" multicast address "sip.mcast.net" (224.0.1.75). This request SHOULD be scoped to ensure it is not forwarded beyond the boundaries of the administrative system. This MAY be done with either TTL or administrative scopes [25], depending on what is implemented in the network. SIP user agents MAY listen to that address and use it to become aware of the location of other local users [20]; however, they do not respond to the request. A user agent MAY also be configured with the address of a registrar server to which it sends a REGISTER request upon startup.

Requests are processed in the order received. Clients SHOULD avoid sending a new registration (as opposed to a retransmission) until they have received the response from the server for the previous one.

> Clients may register from different locations, by necessity using different Call-ID values. Thus, the CSeq value cannot be used to enforce ordering. Since registrations are additive, ordering is less of a problem than if each REGISTER request completely replaced all earlier ones.

The meaning of the REGISTER request-header fields is defined as follows. We define "address-of-record" as the SIP address that the registry knows the registrand, typically of the form "user@domain" rather than "user@host". In third-party registration, the entity issuing the request is different from the entity being registered.

To: The To header field contains the address-of-record whose registration is to be created or updated.

From: The From header field contains the address-of-record of the person responsible for the registration. For first-party registration, it is identical to the To header field value.

Request-URI: The Request-URI names the destination of the registration request, i.e., the domain of the registrar. The user name MUST be empty. Generally, the domains in the Request-URI and the To header field have the same value; however, it is possible to register as a "visitor", while maintaining one's name. For example, a traveler sip:alice@acme.com (To) might register under the Request-URI sip:atlanta.hiayh.org , with the former as the To header field and the latter as the Request-URI. The REGISTER request is no longer forwarded once it has reached the server whose authoritative domain is the one listed in the Request-URI.

Call-ID: All registrations from a client SHOULD use the same Call-ID header value, at least within the same reboot cycle.

Cseq: Registrations with the same Call-ID MUST have increasing CSeq header values. However, the server does not reject out-of-order requests.

Contact: The request MAY contain a Contact header field; future non-REGISTER requests for the URI given in the To header field SHOULD be directed to the address(es) given in the Contact header.

If the request does not contain a Contact header, the registration remains unchanged.

This is useful to obtain the current list of registrations in the response. Registrations using SIP URIs that differ in one or more of host, port, transport-param or maddr-param (see Figure 3) from an existing registration are added to the list of registrations. Other URI types are compared according to the standard URI equivalency rules for the URI schema. If the URIs are equivalent to that of an existing registration, the new registration replaces the old one if it has a higher q value or, for the same value of q, if the ttl value is higher. All current registrations MUST share the same action value. Registrations that have a different action than current registrations for the same user MUST be rejected with status of 409 (Conflict).

A proxy server ignores the q parameter when processing non-REGISTER requests, while a redirect server simply returns that parameter in its Contact response header field.

Having the proxy server interpret the q parameter is not sufficient to guide proxy behavior, as it is not clear, for example, how long it is supposed to wait between trying addresses.

If the registration is changed while a user agent or proxy server processes an invitation, the new information SHOULD be used.

This allows a service known as "directed pick-up". In the telephone network, directed pickup permits a user at a remote station who hears his own phone ringing to pick up at that station, dial an access code, and be connected to the calling user as if he had answered his own phone.

A server MAY choose any duration for the registration lifetime. Registrations not refreshed after this amount of time SHOULD be silently discarded. Responses to a registration SHOULD include an Expires header (Section 6.20) or expires Contact parameters (Section 6.13), indicating the time at which the server will drop the registration. If none is present, one hour is assumed. Clients MAY request a registration lifetime by indicating the time in an Expires header in the request. A server SHOULD NOT use a higher lifetime than the one requested, but MAY use a lower one. A single address (if host-independent) MAY be registered from several different clients.

A client cancels an existing registration by sending a REGISTER request with an expiration time (Expires) of zero seconds for a particular Contact or the wildcard Contact designated by a "*" for all registrations. Registrations are matched based on the user, host, port and maddr parameters.

The server SHOULD return the current list of registrations in the 200 response as Contact header fields.

It is particularly important that REGISTER requests are authenticated since they allow to redirect future requests (see Section 13.2).

> Beyond its use as a simple location service, this method is needed if there are several SIP servers on a single host. In that case, only one of the servers can use the default port number.

Support of this method is RECOMMENDED.

### 4.3 Request-URI

The Request-URI is a SIP URL as described in Section 2 or a general URI. It indicates the user or service to which this request is being addressed. Unlike the To field, the Request-URI MAY be re-written by proxies.

When used as a Request-URI, a SIP-URL MUST NOT contain the transport-param, maddr-param, ttl-param, or headers elements. A server that receives a SIP-URL with these elements removes them before further processing.

> Typically, the UAC sets the Request-URI and To to the same SIP URL, presumed to remain unchanged over long time periods. However, if the UAC has cached a more direct path to the callee, e.g., from the Contact header field of a response to a previous request, the To would still contain the long-term, "public" address, while the Request-URI would be set to the cached address.

Proxy and redirect servers MAY use the information in the Request-URI and request header fields to handle the request and possibly rewrite the Request-URI. For example, a request addressed to the generic address sip:sales@acme.com is proxied to the particular person, e.g., sip:bob@ny.acme.com , with the To field remaining as sip:sales@acme.com. At ny.acme.com , Bob then designates Alice as the temporary substitute.

The host part of the Request-URI typically agrees with one of the host names of the receiving server. If it does not, the server SHOULD proxy the request to the address indicated or return a 404 (Not Found) response if it is unwilling or unable to do so. For example, the Request-URI and server host name can disagree in the case of a firewall proxy that handles outgoing calls. This mode of operation is similar to that of HTTP proxies.

If a SIP server receives a request with a URI indicating a scheme other than SIP which that server does not understand, the server MUST return a 400 (Bad Request) response. It MUST do this even if the To header field contains a scheme it does understand. This is because proxies are responsible for processing the Request-URI; the To field is of end-to-end significance.

### 4.3.1 SIP Version

Both request and response messages include the version of SIP in use, and follow [H3.1] (with HTTP replaced by SIP, and HTTP/1.1 replaced by SIP/2.0) regarding version ordering, compliance requirements, and upgrading of version numbers. To be compliant with this specification, applications sending SIP messages MUST include a SIP-Version of "SIP/2.0".

## 4.4 Option Tags

Option tags are unique identifiers used to designate new options in SIP. These tags are used in Require (Section 6.30) and Unsupported (Section 6.38) fields.

Syntax:

```
option-tag  =  token
```

See Section C for a definition of token. The creator of a new SIP option MUST either prefix the option with their reverse domain name or register the new option with the Internet Assigned Numbers Authority (IANA). For example, "com.foo.mynewfeature" is an apt name for a feature whose inventor can be reached at "foo.com". Individual organizations are then responsible for ensuring that option names don’t collide. Options registered with IANA have the prefix "org.iana.sip.", options described in RFCs have the prefix "org.ietf.rfc.N", where N is the RFC number. Option tags are case-insensitive.

### 4.4.1 Registering New Option Tags with IANA

When registering a new SIP option, the following information MUST be provided:

- Name and description of option. The name MAY be of any length, but SHOULD be no more than twenty characters long. The name MUST consist of alphanum (See Figure 3) characters only;

- o Indication of who has change control over the option (for example, IETF, ISO, ITU-T, other international standardization bodies, a consortium or a particular company or group of companies);

- o A reference to a further description, if available, for example (in order of preference) an RFC, a published paper, a patent filing, a technical report, documented source code or a computer manual;

- o Contact information (postal and email address);

Registrations should be sent to iana@iana.org

This procedure has been borrowed from RTSP [4] and the RTP AVP [26].

# 5 Response

After receiving and interpreting a request message, the recipient responds with a SIP response message. The response message format is shown below:

```
Response  =  Status-Line        ;  Section 5.1
             *( general-header
             | response-header
             | entity-header )
             CRLF
             [ message-body ]   ;  Section 8
```

SIP's structure of responses is similar to [H6], but is defined explicitly here.

## 5.1 Status-Line

The first line of a Response message is the Status-Line, consisting of the protocol version (Section 4.3.1) followed by a numeric Status-Code and its associated textual phrase, with each element separated by SP characters. No CR or LF is allowed except in the final CRLF sequence.

```
Status-Line  =  SIP-version SP Status-Code SP Reason-Phrase CRLF
```

### 5.1.1 Status Codes and Reason Phrases

The Status-Code is a 3-digit integer result code that indicates the outcome of the attempt to understand and satisfy the request. The Reason-Phrase is intended to give a short textual description of the Status-Code. The Status-Code is intended for use by automata, whereas the Reason-Phrase is intended for the human user. The client is not required to examine or display the Reason-Phrase.

```
Status-Code     =  Informational                    ;Fig. 5
                |  Success                          ;Fig. 5
                |  Redirection                      ;Fig. 6
                |  Client-Error                     ;Fig. 7
                |  Server-Error                     ;Fig. 8
                |  Global-Failure                   ;Fig. 9
                |  extension-code
extension-code  =  3DIGIT
Reason-Phrase   =  *<TEXT-UTF8,  excluding CR, LF>
```

We provide an overview of the Status-Code below, and provide full definitions in Section 7. The first digit of the Status-Code defines the class of response. The last two digits do not have any categorization role. SIP/2.0 allows 6 values for the first digit:

1xx: Informational -- request received, continuing to process the request;

2xx: Success -- the action was successfully received, understood, and accepted;

3xx: Redirection -- further action needs to be taken in order to complete the request;

4xx: Client Error -- the request contains bad syntax or cannot be fulfilled at this server;

5xx: Server Error -- the server failed to fulfill an apparently valid request;

6xx: Global Failure -- the request cannot be fulfilled at any server.

Figures 5 through 9 present the individual values of the numeric response codes, and an example set of corresponding reason phrases for SIP/2.0. These reason phrases are only recommended; they may be replaced by local equivalents without affecting the protocol. Note that SIP adopts many HTTP/1.1 response codes. SIP/2.0 adds response codes in the range starting at x80 to avoid conflicts with newly defined HTTP response codes, and adds a new class, 6xx, of response codes.

SIP response codes are extensible. SIP applications are not required to understand the meaning of all registered response codes, though such understanding is obviously desirable. However, applications MUST understand the class of any response code, as indicated by the first digit, and treat any unrecognized response as being equivalent to the x00 response code of that class, with the exception that an unrecognized response MUST NOT be cached. For example, if a client receives an unrecognized response code of 431, it can safely assume that there was something wrong with its request and treat the response as if it had received a 400 (Bad Request) response code. In such cases, user agents SHOULD present to the user the message body returned with the response, since that message body is likely to include human-readable information which will explain the unusual status.

```
     Informational  =  "100"  ;  Trying
                   |   "180"  ;  Ringing
                   |   "181"  ;  Call Is Being Forwarded
                   |   "182"  ;  Queued
     Success        =  "200"  ;  OK
```

Figure 5: Informational and success status codes

```
     Redirection  =  "300"  ;  Multiple Choices
                 |   "301"  ;  Moved Permanently
                 |   "302"  ;  Moved Temporarily
                 |   "303"  ;  See Other
                 |   "305"  ;  Use Proxy
                 |   "380"  ;  Alternative Service
```

Figure 6: Redirection status codes

```
Client-Error  =  "400"  ;  Bad Request
              |  "401"  ;  Unauthorized
              |  "402"  ;  Payment Required
              |  "403"  ;  Forbidden
              |  "404"  ;  Not Found
              |  "405"  ;  Method Not Allowed
              |  "406"  ;  Not Acceptable
              |  "407"  ;  Proxy Authentication Required
              |  "408"  ;  Request Timeout
              |  "409"  ;  Conflict
              |  "410"  ;  Gone
              |  "411"  ;  Length Required
              |  "413"  ;  Request Entity Too Large
              |  "414"  ;  Request-URI Too Large
              |  "415"  ;  Unsupported Media Type
              |  "420"  ;  Bad Extension
              |  "480"  ;  Temporarily not available
              |  "481"  ;  Call Leg/Transaction Does Not Exist
              |  "482"  ;  Loop Detected
              |  "483"  ;  Too Many Hops
              |  "484"  ;  Address Incomplete
              |  "485"  ;  Ambiguous
              |  "486"  ;  Busy Here
```

Figure 7: Client error status codes

```
Server-Error  =  "500"  ;  Internal Server Error
              |  "501"  ;  Not Implemented
              |  "502"  ;  Bad Gateway
              |  "503"  ;  Service Unavailable
              |  "504"  ;  Gateway Time-out
              |  "505"  ;  SIP Version not supported
```

Figure 8: Server error status codes

# 6 Header Field Definitions

SIP header fields are similar to HTTP header fields in both syntax and semantics. In particular, SIP header fields follow the syntax for message-header as described in [H4.2]. The rules for extending header fields over multiple lines, and use of multiple message-header fields with the same field-name, described in [H4.2] also apply to SIP. The

```
      Global-Failure |  "600"  ;  Busy Everywhere
                     |  "603"  ;  Decline
                     |  "604"  ;  Does not exist anywhere
                     |  "606"  ;  Not Acceptable
```

Figure 9: Global failure status codes

rules in [H4.2] regarding ordering of header fields apply to SIP, with the exception of Via fields, see below, whose order matters. Additionally, header fields which are hop-by-hop MUST appear before any header fields which are end-to-end. Proxies SHOULD NOT reorder header fields. Proxies add Via header fields and MAY add other hop-by-hop header fields. They can modify certain header fields, such as Max-Forwards (Section 6.23) and "fix up" the Via header fields with "received" parameters as described in Section 6.40.1. Proxies MUST NOT alter any fields that are authenticated (see Section 13.2).

The header fields required, optional and not applicable for each method are listed in Table 4 and Table 5. The table uses "o" to indicate optional, "m" mandatory and "-" for not applicable. A "*" indicates that the header fields are needed only if message body is not empty. See sections 6.15, 6.16 and 8 for details.

The "where" column describes the request and response types with which the header field can be used. "R" refers to header fields that can be used in requests (that is, request and general header fields). "r" designates a response or general-header field as applicable to all responses, while a list of numeric values indicates the status codes with which the header field can be used. "g" and "e" designate general (Section 6.1) and entity header (Section 6.2) fields, respectively. If a header field is marked "c", it is copied from the request to the response.

The "enc." column describes whether this message header field MAY be encrypted end-to-end. A "n" designates fields that MUST NOT be encrypted, while "c" designates fields that SHOULD be encrypted if encryption is used.

The "e-e" column has a value of "e" for end-to-end and a value of "h" for hop-by-hop header fields.

| | where | enc. | e-e | ACK | BYE | CAN | INV | OPT | REG |
|---|---|---|---|---|---|---|---|---|---|
| Accept | R | | e | - | - | - | o | o | o |
| Accept | 415 | | e | - | - | - | o | o | o |
| Accept-Encoding | R | | e | - | - | - | o | o | o |
| Accept-Encoding | 415 | | e | - | - | - | o | o | o |
| Accept-Language | R | | e | - | o | o | o | o | o |
| Accept-Language | 415 | | e | - | o | o | o | o | o |
| Allow | 200 | | e | - | - | - | - | m | - |
| Allow | 405 | | e | o | o | o | o | o | o |
| Authorization | R | | e | o | o | o | o | o | o |
| Call-ID | gc | n | e | m | m | m | m | m | m |
| Contact | R | | e | o | - | - | o | o | o |
| Contact | 1xx | | e | - | - | - | o | o | - |
| Contact | 2xx | | e | - | - | - | o | o | o |
| Contact | 3xx | | e | - | o | - | o | o | o |
| Contact | 485 | | e | - | o | - | o | o | o |
| Content-Encoding | e | | e | o | - | - | o | o | o |
| Content-Length | e | | e | o | - | - | o | o | o |
| Content-Type | e | | e | * | - | - | * | * | * |
| CSeq | gc | n | e | m | m | m | m | m | m |
| Date | g | | e | o | o | o | o | o | o |
| Encryption | g | n | e | o | o | o | o | o | o |
| Expires | g | | e | - | - | - | o | - | o |
| From | gc | n | e | m | m | m | m | m | m |
| Hide | R | n | h | o | o | o | o | o | o |
| Max-Forwards | R | n | e | o | o | o | o | o | o |
| Organization | g | c | h | - | - | - | o | o | o |

Table 4: Summary of header fields, A--O

Other header fields can be added as required; a server MUST ignore header fields not defined in this specification that it does not understand. A proxy MUST NOT remove or modify header fields not defined in this specification that it does not understand. A compact form of these header fields is also defined in Section 9 for use over UDP when the request has to fit into a single packet and size is an issue.

Table 6 in Appendix A lists those header fields that different client and server types MUST be able to parse.

## 6.1 General Header Fields

General header fields apply to both request and response messages. The "general-header" field names can be extended reliably only in combination with a change in the protocol version. However, new or

| | where | enc. | e-e | ACK | BYE | CAN | INV | OPT | REG |
|---|---|---|---|---|---|---|---|---|---|
| Proxy-Authenticate | 407 | n | h | o | o | o | o | o | o |
| Proxy-Authorization | R | n | h | o | o | o | o | o | o |
| Proxy-Require | R | n | h | o | o | o | o | o | o |
| Priority | R | c | e | - | - | - | o | - | - |
| Require | R | | e | o | o | o | o | o | o |
| Retry-After | R | c | e | - | - | - | - | - | o |
| Retry-After | 404,480,486 | c | e | o | o | o | o | o | o |
| | 503 | c | e | o | o | o | o | o | o |
| | 600,603 | c | e | o | o | o | o | o | o |
| Response-Key | R | c | e | - | o | o | o | o | o |
| Record-Route | R | | h | o | o | o | o | o | o |
| Record-Route | 2xx | | h | o | o | o | o | o | o |
| Route | R | | h | o | o | o | o | o | o |
| Server | r | c | e | o | o | o | o | o | o |
| Subject | R | c | e | - | - | - | o | - | - |
| Timestamp | g | | e | o | o | o | o | o | o |
| To | gc(1) | n | e | m | m | m | m | m | m |
| Unsupported | 420 | | e | o | o | o | o | o | o |
| User-Agent | g | c | e | o | o | o | o | o | o |
| Via | gc(2) | n | e | m | m | m | m | m | m |
| Warning | r | | e | o | o | o | o | o | o |
| WWW-Authenticate | 401 | c | e | o | o | o | o | o | o |

Table 5: Summary of header fields, P--Z; (1): copied with possible addition of tag; (2): UAS removes first Via header field

experimental header fields MAY be given the semantics of general header fields if all parties in the communication recognize them to be "general-header" fields. Unrecognized header fields are treated as "entity-header" fields.

## 6.2 Entity Header Fields

The "entity-header" fields define meta-information about the message-body or, if no body is present, about the resource identified by the request. The term "entity header" is an HTTP 1.1 term where the response body can contain a transformed version of the message body. The original message body is referred to as the "entity". We retain the same terminology for header fields but usually refer to the "message body" rather then the entity as the two are the same in SIP.

## 6.3 Request Header Fields

The "request-header" fields allow the client to pass additional information about the request, and about the client itself, to the server. These fields act as request modifiers, with semantics equivalent to the parameters of a programming language method invocation.

The "request-header" field names can be extended reliably only in combination with a change in the protocol version. However, new or experimental header fields MAY be given the semantics of "request-header" fields if all parties in the communication recognize them to be request-header fields. Unrecognized header fields are treated as "entity-header" fields.

## 6.4 Response Header Fields

The "response-header" fields allow the server to pass additional information about the response which cannot be placed in the Status-Line. These header fields give information about the server and about further access to the resource identified by the Request-URI.

Response-header field names can be extended reliably only in combination with a change in the protocol version. However, new or experimental header fields MAY be given the semantics of "response-header" fields if all parties in the communication recognize them to be "response-header" fields. Unrecognized header fields are treated as "entity-header" fields.

## 6.5 End-to-end and Hop-by-hop Headers

End-to-end headers MUST be transmitted unmodified across all proxies, while hop-by-hop headers MAY be modified or added by proxies.

## 6.6 Header Field Format

Header fields ("general-header", "request-header", "response-header", and "entity-header") follow the same generic header format as that given in Section 3.1 of RFC 822 [24]. Each header field consists of a name followed by a colon (":") and the field value. Field names are case-insensitive. The field value MAY be preceded by any amount of leading white space (LWS), though a single space (SP) is preferred. Header fields can be extended over multiple lines by preceding each extra line with at least one SP or horizontal tab (HT). Applications MUST follow HTTP "common form" when generating these constructs, since there might exist some implementations that fail to accept anything beyond the common forms.

```
message-header  =  field-name ":" [ field-value ] CRLF
field-name      =  token
field-value     =  *( field-content | LWS )
field-content   =  < the OCTETs  making up the field-value
                    and consisting of either *TEXT-UTF8
                    or combinations of token,
                    separators, and quoted-string>
```

The relative order of header fields with different field names is not significant. Multiple header fields with the same field-name may be present in a message if and only if the entire field-value for that header field is defined as a comma-separated list (i.e., #(values)). It MUST be possible to combine the multiple header fields into one "field-name: field-value" pair, without changing the semantics of the message, by appending each subsequent field-value to the first, each separated by a comma. The order in which header fields with the same field-name are received is therefore significant to the interpretation of the combined field value, and thus a proxy MUST NOT change the order of these field values when a message is forwarded.

Field names are not case-sensitive, although their values may be.

## 6.7 Accept

The Accept header follows the syntax defined in [H14.1]. The semantics are also identical, with the exception that if no Accept header is present, the server SHOULD assume a default value of application/sdp.

This request-header field is used only with the INVITE, OPTIONS and REGISTER request methods to indicate what media types are acceptable in the response.

Example:

```
Accept: application/sdp;level=1, application/x-private, text/html
```

## 6.8 Accept-Encoding

The Accept-Encoding request-header field is similar to Accept, but restricts the content-codings [H3.4.1] that are acceptable in the response. See [H14.3]. The syntax of this header is defined in [H14.3]. The semantics in SIP are identical to those defined in [H14.3].

## 6.9 Accept-Language

The Accept-Language header follows the syntax defined in [H14.4]. The rules for ordering the languages based on the q parameter apply to SIP as well. When used in SIP, the Accept-Language request-header field can be used to allow the client to indicate to the server in which language it would prefer to receive reason phrases, session descriptions or status responses carried as message bodies. A proxy MAY use this field to help select the destination for the call, for example, a human operator conversant in a language spoken by the caller.

Example:

```
Accept-Language: da, en-gb;q=0.8, en;q=0.7
```

## 6.10 Allow

The Allow entity-header field lists the set of methods supported by the resource identified by the Request-URI. The purpose of this field is strictly to inform the recipient of valid methods associated with the resource. An Allow header field MUST be present in a 405 (Method Not Allowed) response and SHOULD be present in an OPTIONS response.

```
Allow  =  "Allow" ":" 1#Method
```

## 6.11 Authorization

A user agent that wishes to authenticate itself with a server -- usually, but not necessarily, after receiving a 401 response -- MAY do so by including an Authorization request-header field with the request. The Authorization field value consists of credentials containing the authentication information of the user agent for the realm of the resource being requested.

Section 13.2 overviews the use of the Authorization header, and section 15 describes the syntax and semantics when used with PGP based authentication.

## 6.12 Call-ID

The Call-ID general-header field uniquely identifies a particular invitation or all registrations of a particular client. Note that a single multimedia conference can give rise to several calls with different Call-IDs, e.g., if a user invites a single individual several times to the same (long-running) conference.

For an INVITE request, a callee user agent server SHOULD NOT alert the user if the user has responded previously to the Call-ID in the INVITE request. If the user is already a member of the conference and the conference parameters contained in the session description have not changed, a callee user agent server MAY silently accept the call, regardless of the Call-ID. An invitation for an existing Call-ID or session can change the parameters of the conference. A client application MAY decide to simply indicate to the user that the conference parameters have been changed and accept the invitation automatically or it MAY require user confirmation.

A user may be invited to the same conference or call using several different Call-IDs. If desired, the client MAY use identifiers within the session description to detect this duplication. For example, SDP contains a session id and version number in the origin (o) field.

The REGISTER and OPTIONS methods use the Call-ID value to unambiguously match requests and responses. All REGISTER requests issued by a single client SHOULD use the same Call-ID, at least within the same boot cycle.

> Since the Call-ID is generated by and for SIP, there is no reason to deal with the complexity of URL-encoding and case-ignoring string comparison.

```
Call-ID   =  ( "Call-ID" | "i" ) ":" local-id "@" host
local-id  =  1*uric
```

"host" SHOULD be either a fully qualified domain name or a globally routable IP address. If this is the case, the "local-id" SHOULD be an identifier consisting of URI characters that is unique within "host". Use of cryptographically random identifiers [27] is RECOMMENDED. If, however, host is not an FQDN or globally routable IP address (such as a net 10 address), the local-id MUST be globally unique, as opposed to unique within host. These rules guarantee overall global uniqueness of the Call-ID. The value for Call-ID MUST NOT be reused for a different call. Call-IDs are case-sensitive.

Using cryptographically random identifiers provides some protection against session hijacking. Call-ID, To and From are needed to identify a call leg. The distinction between call and call leg matters in calls with third-party control.

For systems which have tight bandwidth constraints, many of the mandatory SIP headers have a compact form, as discussed in Section 9. These are alternate names for the headers which occupy less space in the message. In the case of Call-ID, the compact form is i.

For example, both of the following are valid:

```
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@foo.bar.com
```

or

```
i:f81d4fae-7dec-11d0-a765-00a0c91e6bf6@foo.bar.com
```

## 6.13 Contact

The Contact general-header field can appear in INVITE, ACK, and REGISTER requests, and in 1xx, 2xx, 3xx, and 485 responses. In general, it provides a URL where the user can be reached for further communications.

INVITE and ACK requests: INVITE and ACK requests MAY contain Contact headers indicating from which location the request is originating.

This allows the callee to send future requests, such as BYE, directly to the caller instead of through a series of proxies. The Via header is not sufficient since the desired address may be that of a proxy.

INVITE 2xx responses: A user agent server sending a definitive, positive response (2xx) MAY insert a Contact response header field indicating the SIP address under which it is reachable most directly for future SIP requests, such as ACK, within the same Call-ID. The Contact header field contains the address of the server itself or that of a proxy, e.g., if the host is behind a firewall. The value of this Contact header is copied into the Request-URI of subsequent requests for this call if the response did not also contain a Record-Route header. If the response also contains a Record-Route header field, the address in the Contact header field is added as the last item in the Route header field. See Section 6.29 for details.

The Contact value SHOULD NOT be cached across calls, as it may not represent the most desirable location for a particular destination address.

INVITE 1xx responses: A UAS sending a provisional response (1xx) MAY insert a Contact response header. It has the same semantics in a 1xx response as a 2xx INVITE response. Note that CANCEL requests MUST NOT be sent to that address, but rather follow the same path as the original request.

REGISTER requests: REGISTER requests MAY contain a Contact header field indicating at which locations the user is reachable. The REGISTER request defines a wildcard Contact field, "*", which MUST only be used with Expires: 0 to remove all registrations for a particular user. An optional "expires" parameter indicates the desired expiration time of the registration. If a Contact entry does not have an "expires" parameter, the Expires header field is used as the default value. If neither of these mechanisms is used, SIP URIs are assumed to expire after one hour. Other URI schemes have no expiration times.

REGISTER 2xx responses: A REGISTER response MAY return all locations at which the user is currently reachable. An optional "expires" parameter indicates the expiration time of the registration. If a Contact entry does not have an "expires" parameter, the value of the Expires header field indicates the expiration time. If neither mechanism is used, the expiration time specified in the request, explicitly or by default, is used.

3xx and 485 responses: The Contact response-header field can be used with a 3xx or 485 (Ambiguous) response codes to indicate one or more alternate addresses to try. It can appear in responses to BYE, INVITE and OPTIONS methods. The Contact header field contains URIs giving the new locations or user names to try, or may simply specify additional transport parameters. A 300 (Multiple Choices), 301 (Moved Permanently), 302 (Moved Temporarily) or 485 (Ambiguous) response SHOULD contain a Contact field containing URIs of new addresses to be tried. A

301 or 302 response may also give the same location and username that was being tried but specify additional transport parameters such as a different server or multicast address to try or a change of SIP transport from UDP to TCP or vice versa. The client copies the "user", "password", "host", "port" and "user-param" elements of the Contact URI into the Request-URI of the redirected request and directs the request to the address specified by the "maddr" and "port" parameters, using the transport protocol given in the "transport" parameter. If "maddr" is a multicast address, the value of "ttl" is used as the time-to-live value.

Note that the Contact header field MAY also refer to a different entity than the one originally called. For example, a SIP call connected to GSTN gateway may need to deliver a special information announcement such as "The number you have dialed has been changed."

A Contact response header field can contain any suitable URI indicating where the called party can be reached, not limited to SIP URLs. For example, it could contain URL’s for phones, fax, or irc (if they were defined) or a mailto: (RFC 2368, [28]) URL.

The following parameters are defined. Additional parameters may be defined in other specifications.

q: The "qvalue" indicates the relative preference among the locations given. "qvalue" values are decimal numbers from 0 to 1, with higher values indicating higher preference.

action: The "action" parameter is used only when registering with the REGISTER request. It indicates whether the client wishes that the server proxy or redirect future requests intended for the client. If this parameter is not specified the action taken depends on server configuration. In its response, the registrar SHOULD indicate the mode used. This parameter is ignored for other requests.

expires: The "expires" parameter indicates how long the URI is valid. The parameter is either a number indicating seconds or a quoted string containing a SIP-date. If this parameter is not provided, the value of the Expires header field determines how long the URI is valid. Implementations MAY treat values larger than 2**32-1 (4294967295 seconds or 136 years) as equivalent to 2**32-1.

```
Contact = ( "Contact" | "m" ) ":"
          ("*" | (1# (( name-addr | addr-spec )
          [ *( ";" contact-params ) ] [ comment ] )))

name-addr      = [ display-name ] "<" addr-spec ">"
addr-spec      = SIP-URL | URI
display-name   = *token | quoted-string

contact-params = "q"       "=" qvalue
               | "action"  "=" "proxy" | "redirect"
               | "expires" "=" delta-seconds | <"> SIP-date <">
               | extension-attribute

extension-attribute = extension-name [ "=" extension-value ]
```

only allows one address, unquoted. Since URIs can contain commas and semicolons as reserved characters, they can be mistaken for header or parameter delimiters, respectively. The current syntax corresponds to that for the To and From header, which also allows the use of display names.

Example:

```
Contact: "Mr. Watson" <sip:watson@worcester.bell-telephone.com>
   ;q=0.7; expires=3600,
   "Mr. Watson" <mailto:watson@bell-telephone.com> ;q=0.1
```

## 6.14 Content-Encoding

```
Content-Encoding  =  ( "Content-Encoding" | "e" ) ":"
                     1#content-coding
```

The Content-Encoding entity-header field is used as a modifier to the "media-type". When present, its value indicates what additional content codings have been applied to the entity-body, and thus what decoding mechanisms MUST be applied in order to obtain the media-type referenced by the Content-Type header field. Content-Encoding is primarily used to allow a body to be compressed without losing the identity of its underlying media type.

If multiple encodings have been applied to an entity, the content codings MUST be listed in the order in which they were applied.

All content-coding values are case-insensitive. The Internet Assigned Numbers Authority (IANA) acts as a registry for content-coding value tokens. See [3.5] for a definition of the syntax for content-coding.

Clients MAY apply content encodings to the body in requests. If the server is not capable of decoding the body, or does not recognize any of the content-coding values, it MUST send a 415 "Unsupported Media Type" response, listing acceptable encodings in the Accept-Encoding header. A server MAY apply content encodings to the bodies in responses. The server MUST only use encodings listed in the Accept-Encoding header in the request.

## 6.15 Content-Length

The Content-Length entity-header field indicates the size of the message-body, in decimal number of octets, sent to the recipient.

```
Content-Length  =  ( "Content-Length" | "l" ) ":" 1*DIGIT
```

An example is

```
Content-Length: 3495
```

Applications SHOULD use this field to indicate the size of the message-body to be transferred, regardless of the media type of the entity. Any Content-Length greater than or equal to zero is a valid value. If no body is present in a message, then the Content-Length header field MUST be set to zero. If a server receives a UDP request without Content-Length, it MUST assume that the request encompasses the remainder of the packet. If a server receives a UDP request with a Content-Length, but the value is larger than the size of the body sent in the request, the client SHOULD generate a 400 class response. If there is additional data in the UDP packet after the last byte of the body has been read, the server MUST treat the remaining data as a separate message. This allows several messages to be placed in a single UDP packet.

If a response does not contain a Content-Length, the client assumes that it encompasses the remainder of the UDP packet or the data until the TCP connection is closed, as applicable. Section 8 describes how to determine the length of the message body.

## 6.16 Content-Type

The Content-Type entity-header field indicates the media type of the message-body sent to the recipient. The "media-type" element is defined in [H3.7].

```
Content-Type  =  ( "Content-Type" | "c" ) ":" media-type
```

Examples of this header field are

```
Content-Type: application/sdp
Content-Type: text/html; charset=ISO-8859-4
```

### 6.17 CSeq

Clients MUST add the CSeq (command sequence) general-header field to every request. A CSeq header field in a request contains the request method and a single decimal sequence number chosen by the requesting client, unique within a single value of Call-ID. The sequence number MUST be expressible as a 32-bit unsigned integer. The initial value of the sequence number is arbitrary, but MUST be less than 2**31. Consecutive requests that differ in request method, headers or body, but have the same Call-ID MUST contain strictly monotonically increasing and contiguous sequence numbers; sequence numbers do not wrap around. Retransmissions of the same request carry the same sequence number, but an INVITE with a different message body or different header fields (a "re-invitation") acquires a new, higher sequence number. A server MUST echo the CSeq value from the request in its response. If the Method value is missing in the received CSeq header field, the server fills it in appropriately.

The ACK and CANCEL requests MUST contain the same CSeq value as the INVITE request that it refers to, while a BYE request cancelling an invitation MUST have a higher sequence number. A BYE request with a CSeq that is not higher should cause a 400 response to be generated.

A user agent server MUST remember the highest sequence number for any INVITE request with the same Call-ID value. The server MUST respond to, and then discard, any INVITE request with a lower sequence number.

All requests spawned in a parallel search have the same CSeq value as the request triggering the parallel search.

```
CSeq  =  "CSeq" ":" 1*DIGIT Method
```

Strictly speaking, CSeq header fields are needed for any SIP request that can be cancelled by a BYE or CANCEL request or where a client can issue several requests for the same Call-ID in close succession. Without a sequence number, the response to an INVITE could be mistaken for the response to the cancellation (BYE or CANCEL). Also, if the network duplicates packets or if an ACK is delayed until the server has sent an additional response, the client could interpret an old response as the response to a re-invitation issued shortly thereafter. Using CSeq also makes it easy for the server to distinguish different versions of an invitation, without comparing the message body.

The Method value allows the client to distinguish the response to an INVITE request from that of a CANCEL response. CANCEL requests can be generated by proxies; if they were to increase the sequence number, it might conflict with a later request issued by the user agent for the same call.

With a length of 32 bits, a server could generate, within a single call, one request a second for about 136 years before needing to wrap around. The initial value of the sequence number is chosen so that subsequent requests within the same call will not wrap around. A non-zero initial value allows to use a time-based initial sequence number, if the client desires. A client could, for example, choose the 31 most significant bits of a 32-bit second clock as an initial sequence number.

Forked requests MUST have the same CSeq as there would be ambiguity otherwise between these forked requests and later BYE issued by the client user agent.

Example:

```
CSeq: 4711 INVITE
```

## 6.18 Date

Date is a general-header field. Its syntax is:

```
SIP-date  =  rfc1123-date
```

See [H14.19] for a definition of rfc1123-date. Note that unlike HTTP/1.1, SIP only supports the most recent RFC1123 [29] formatting for dates.

The Date header field reflects the time when the request or response is first sent. Thus, retransmissions have the same Date header field value as the original.

> The Date header field can be used by simple end systems without a battery-backed clock to acquire a notion of current time.

## 6.19 Encryption

The Encryption general-header field specifies that the content has been encrypted. Section 13 describes the overall SIP security architecture and algorithms. This header field is intended for end-to-end encryption of requests and responses. Requests are encrypted based on the public key belonging to the entity named in the To header field. Responses are encrypted based on the public key conveyed in the Response-Key header field. Note that the public keys themselves may not be used for the encryption. This depends on the particular algorithms used.

For any encrypted message, at least the message body and possibly other message header fields are encrypted. An application receiving a request or response containing an Encryption header field decrypts the body and then concatenates the plaintext to the request line and headers of the original message. Message headers in the decrypted part completely replace those with the same field name in the plaintext part. (Note: If only the body of the message is to be encrypted, the body has to be prefixed with CRLF to allow proper concatenation.) Note that the request method and Request-URI cannot be encrypted.

> Encryption only provides privacy; the recipient has no guarantee that the request or response came from the party listed in the From message header, only that the sender used the recipient’s public key. However, proxies will not be able to modify the request or response.

```
Encryption         =  "Encryption" ":" encryption-scheme 1*SP
                      #encryption-params
encryption-scheme  =  token
encryption-params  =  token "=" ( token | quoted-string )
```

The token indicates the form of encryption used; it is described in section 13.

The example in Figure 10 shows a message encrypted with ASCII-armored PGP that was generated by applying "pgp -ea" to the payload to be encrypted.

```
INVITE sip:watson@boston.bell-telephone.com SIP/2.0
Via: SIP/2.0/UDP 169.130.12.5
From: <sip:a.g.bell@bell-telephone.com>
To: T. A. Watson <sip:watson@bell-telephone.com>
Call-ID: 187602141351@worcester.bell-telephone.com
Content-Length: 885
Encryption: PGP version=2.6.2,encoding=ascii

hQEMAxkp5GPd+j5xAQf/ZDIfGD/PDOM1wayvwdQAKgGgjmZWe+MTy9NEX8O25Red
h0/pyrd/+DV5C2BYs7yzSOSXaj1C/tTK/4do6rtjhP8QA3vbDdVdaFciwEVAcuXs
ODxlNAVqyDi1RqFC28BJIvQ5KfEkPuACKTK7WlRSBc7vNPEA3nyqZGBTwhxRSbIR
RuFEsHSVojdCam4htcqxGnFwD9sksqs6LIyCFaiTAhWtwcCaN437G7mUYzy2KLcA
zPVGq1VQg83b99zPzIxRdlZ+K7+bAnu8Rtu+ohOCMLV3TPXbyp+err1YiThCZHIu
X9dOVj3CMjCP66RSHa/ea0wYTRRNYA/G+kdP8DSUcqYAAAE/hZPX6nFIqk7AVnf6
IpWHUPTelNUJpzUp5Ou+q/5P7ZAsn+cSAuF2YWtVjCf+SQmBR13p2EYYWHoxlA2/
GgKADYe4M3JSwOtqwU8zUJF3FIfk7vsxmSqtUQrRQaiIhqNyG7KxJt4YjWnEjF5E
WUIPhvyGFMJaeQXIyGRYZAYvKKklyAJcm29zLACxU5alX4M25lHQd9FR9Zmq6Jed
wbWvia6cAIfsvlZ9JGocmQYF7pcuz5pnczqP+/yvRqFJtDGD/v3s++G2R+ViVYJO
z/lxGUZaM4IWBCf+4DUjNanZM0oxAE28NjaIZ0rrldDQmO8V9FtPKdHxkqA5iJP+
6vGOFti1Ak4kmEz0vM/Nsv7kkubTFhRl05OiJIGr9S1UhenlZv9l6RuXsOY/EwH2
z8X9N4MhMyXEVuC9rt8/AUhmVQ==
=bOW+
```

Figure 10: PGP Encryption Example

Since proxies can base their forwarding decision on any combination of SIP header fields, there is no guarantee that an encrypted request "hiding" header fields will reach the same destination as an otherwise identical un-encrypted request.

## 6.20 Expires

The Expires entity-header field gives the date and time after which the message content expires.

This header field is currently defined only for the REGISTER and INVITE methods. For REGISTER, it is a request and response-header field. In a REGISTER request, the client indicates how long it wishes the registration to be valid. In the response, the server indicates the earliest expiration time of all registrations. The server MAY choose a shorter time interval than that requested by the client, but SHOULD NOT choose a longer one.

For INVITE requests, it is a request and response-header field. In a request, the caller can limit the validity of an invitation, for example, if a client wants to limit the time duration of a search or a conference invitation. A user interface MAY take this as a hint to leave the invitation window on the screen even if the user is not currently at the workstation. This also limits the duration of a search. If the request expires before the search completes, the proxy returns a 408 (Request Timeout) status. In a 302 (Moved Temporarily) response, a server can advise the client of the maximal duration of the redirection.

The value of this field can be either a SIP-date or an integer number of seconds (in decimal), measured from the receipt of the request. The latter approach is preferable for short durations, as it does not depend on clients and servers sharing a synchronized clock. Implementations MAY treat values larger than 2**32-1 (4294967295 or 136 years) as equivalent to 2**32-1.

```
Expires  =  "Expires" ":" ( SIP-date | delta-seconds )
```

Two examples of its use are

```
Expires: Thu, 01 Dec 1994 16:00:00 GMT
Expires: 5
```

## 6.21 From

Requests and responses MUST contain a From general-header field, indicating the initiator of the request. The From field MAY contain the "tag" parameter. The server copies the From header field from the request to the response. The optional "display-name" is meant to be rendered by a human-user interface. A system SHOULD use the display name "Anonymous" if the identity of the client is to remain hidden.

The SIP-URL MUST NOT contain the "transport-param", "maddr-param", "ttl-param", or "headers" elements. A server that receives a SIP-URL with these elements removes them before further processing.

Even if the "display-name" is empty, the "name-addr" form MUST be used if the "addr-spec" contains a comma, question mark, or semicolon.

```
From         =  ( "From" | "f" ) ":" ( name-addr | addr-spec )
                *( ";" addr-params )
addr-params  =  tag-param
tag-param    =  "tag=" UUID
UUID         =  1*( hex | "-" )
```

Examples:

```
From: "A. G. Bell" <sip:agb@bell-telephone.com>
From: sip:+12125551212@server.phone2net.com
From: Anonymous <sip:c8oqz84zk7z@privacy.org>
```

The "tag" MAY appear in the From field of a request. It MUST be present when it is possible that two instances of a user sharing a SIP address can make call invitations with the same Call-ID.

The "tag" value MUST be globally unique and cryptographically random with at least 32 bits of randomness. A single user maintains the same tag throughout the call identified by the Call-ID.

Call-ID, To and From are needed to identify a call leg. The distinction between call and call leg matters in calls with multiple responses to a forked request. The format is similar to the equivalent RFC 822 [24] header, but with a URI instead of just an email address.

## 6.22 Hide

A client uses the Hide request header field to indicate that it wants the path comprised of the Via header fields (Section 6.40) to be hidden from subsequent proxies and user agents. It can take two forms: Hide: route and Hide:  hop. Hide header fields are typically added by the client user agent, but MAY be added by any proxy along the path.

If a request contains the "Hide: route" header field, all following proxies SHOULD hide their previous hop. If a request contains the "Hide: hop" header field, only the next proxy SHOULD hide the previous hop and then remove the Hide option unless it also wants to remain anonymous.

A server hides the previous hop by encrypting the "host" and "port" parts of the top-most Via header field with an algorithm of its choice. Servers SHOULD add additional "salt" to the "host" and "port" information prior to encryption to prevent malicious downstream proxies from guessing earlier parts of the path based on seeing identical encrypted Via headers. Hidden Via fields are marked with the "hidden" Via option, as described in Section 6.40.

A server that is capable of hiding Via headers MUST attempt to decrypt all Via headers marked as "hidden" to perform loop detection. Servers that are not capable of hiding can ignore hidden Via fields in their loop detection algorithm.

> If hidden headers were not marked, a proxy would have to decrypt all headers to detect loops, just in case one was encrypted, as the Hide: Hop option may have been removed along the way.

A host MUST NOT add such a "Hide: hop" header field unless it can guarantee it will only send a request for this destination to the same next hop. The reason for this is that it is possible that the request will loop back through this same hop from a downstream proxy. The loop will be detected by the next hop if the choice of next hop is fixed, but could loop an arbitrary number of times otherwise.

A client requesting "Hide: route" can only rely on keeping the request path private if it sends the request to a trusted proxy. Hiding the route of a SIP request is of limited value if the request results in data packets being exchanged directly between the calling and called user agent.

The use of Hide header fields is discouraged unless path privacy is truly needed; Hide fields impose extra processing costs and restrictions for proxies and can cause requests to generate 482 (Loop Detected) responses that could otherwise be avoided.

The encryption of Via header fields is described in more detail in Section 13.

The Hide header field has the following syntax:

```
Hide  =  "Hide" ":" ( "route" | "hop" )
```

## 6.23 Max-Forwards

The Max-Forwards request-header field may be used with any SIP method to limit the number of proxies or gateways that can forward the request to the next downstream server. This can also be useful when the client is attempting to trace a request chain which appears to be failing or looping in mid-chain.

```
Max-Forwards  =  "Max-Forwards" ":" 1*DIGIT
```

The Max-Forwards value is a decimal integer indicating the remaining number of times this request message is allowed to be forwarded.

Each proxy or gateway recipient of a request containing a Max-Forwards header field MUST check and update its value prior to forwarding the request. If the received value is zero (0), the recipient MUST NOT forward the request. Instead, for the OPTIONS and REGISTER methods, it MUST respond as the final recipient. For all other methods, the server returns 483 (Too many hops).

If the received Max-Forwards value is greater than zero, then the forwarded message MUST contain an updated Max-Forwards field with a value decremented by one (1).

Example:

```
Max-Forwards: 6
```

## 6.24 Organization

The Organization general-header field conveys the name of the organization to which the entity issuing the request or response belongs. It MAY also be inserted by proxies at the boundary of an organization.

The field MAY be used by client software to filter calls.

```
Organization  =  "Organization" ":" *TEXT-UTF8
```

## 6.25 Priority

The Priority request-header field indicates the urgency of the request as perceived by the client.

```
Priority        =  "Priority" ":" priority-value
priority-value  =  "emergency" | "urgent" | "normal"
                |  "non-urgent"
```

It is RECOMMENDED that the value of "emergency" only be used when life, limb or property are in imminent danger.

Examples:

```
Subject: A tornado is heading our way!
Priority: emergency

Subject: Weekend plans
Priority: non-urgent
```

These are the values of RFC 2076 [30], with the addition of "emergency".

## 6.26 Proxy-Authenticate

The Proxy-Authenticate response-header field MUST be included as part of a 407 (Proxy Authentication Required) response. The field value consists of a challenge that indicates the authentication scheme and parameters applicable to the proxy for this Request-URI.

Unlike its usage within HTTP, the Proxy-Authenticate header MUST be passed upstream in the response to the UAC. In SIP, only UAC's can authenticate themselves to proxies.

The syntax for this header is defined in [H14.33]. See 14 for further details on its usage.

A client SHOULD cache the credentials used for a particular proxy server and realm for the next request to that server. Credentials are, in general, valid for a specific value of the Request-URI at a particular proxy server. If a client contacts a proxy server that has required authentication in the past, but the client does not have credentials for the particular Request-URI, it MAY attempt to use the most-recently used credential. The server responds with 401 (Unauthorized) if the client guessed wrong.

> This suggested caching behavior is motivated by proxies restricting phone calls to authenticated users. It seems likely that in most cases, all destinations require the same password. Note that end-to-end authentication is likely to be destination-specific.

## 6.27 Proxy-Authorization

The Proxy-Authorization request-header field allows the client to identify itself (or its user) to a proxy which requires authentication. The Proxy-Authorization field value consists of credentials containing the authentication information of the user agent for the proxy and/or realm of the resource being requested.

Unlike Authorization, the Proxy-Authorization header field applies only to the next outbound proxy that demanded authentication using the Proxy- Authenticate field. When multiple proxies are used in a chain, the Proxy-Authorization header field is consumed by the first outbound proxy that was expecting to receive credentials. A proxy MAY relay the credentials from the client request to the next proxy if that is the mechanism by which the proxies cooperatively authenticate a given request.

See [H14.34] for a definition of the syntax, and section 14 for a discussion of its usage.

## 6.28 Proxy-Require

The Proxy-Require header field is used to indicate proxy-sensitive features that MUST be supported by the proxy. Any Proxy-Require header field features that are not supported by the proxy MUST be negatively acknowledged by the proxy to the client if not supported. Proxy servers treat this field identically to the Require field.

See Section 6.30 for more details on the mechanics of this message and a usage example.

## 6.29 Record-Route

The Record-Route request and response header field is added to a request by any proxy that insists on being in the path of subsequent requests for the same call leg. It contains a globally reachable Request-URI that identifies the proxy server. Each proxy server adds its Request-URI to the beginning of the list.

The server copies the Record-Route header field unchanged into the response. (Record-Route is only relevant for 2xx responses.)

The calling user agent client copies the Record-Route header into a Route header field of subsequent requests within the same call leg, reversing the order of requests, so that the first entry is closest to the user agent client. If the response contained a Contact header field, the calling user agent adds its content as the last Route header. Unless this would cause a loop, any client MUST send any subsequent requests for this call leg to the first Request-URI in the Route request header field and remove that entry.

The calling user agent MUST NOT use the Record-Route header field in requests that contain Route header fields.

> Some proxies, such as those controlling firewalls or in an automatic call distribution (ACD) system, need to maintain call state and thus need to receive any BYE and ACK packets for the call.

The Record-Route header field has the following syntax:

```
Record-Route  =  "Record-Route" ":" 1# name-addr
```

Proxy servers SHOULD use the "maddr" URL parameter containing their address to ensure that subsequent requests are guaranteed to reach exactly the same server.

Example for a request that has traversed the hosts ieee.org and bell-telephone.com , in that order:

```
Record-Route: <sip:a.g.bell@bell-telephone.com>,
  <sip:a.bell@ieee.org>
```

## 6.30 Require

The Require request-header field is used by clients to tell user agent servers about options that the client expects the server to support in order to properly process the request. If a server does not understand the option, it MUST respond by returning status code 420 (Bad Extension) and list those options it does not understand in the Unsupported header.

```
Require  =  "Require" ":" 1#option-tag
```

Example:

```
C->S:   INVITE sip:watson@bell-telephone.com SIP/2.0
        Require: com.example.billing
        Payment: sheep_skins, conch_shells

S->C:   SIP/2.0 420 Bad Extension
        Unsupported: com.example.billing
```

> This is to make sure that the client-server interaction will proceed without delay when all options are understood by both sides, and only slow down if options are not understood (as in the example above). For a well-matched client-server pair, the interaction proceeds quickly, saving a round-trip often required by negotiation mechanisms. In addition, it also removes ambiguity when the client requires features that the server does not understand. Some features, such as call handling fields, are only of interest to end systems.

Proxy and redirect servers MUST ignore features that are not understood. If a particular extension requires that intermediate devices support it, the extension MUST be tagged in the Proxy-Require field as well (see Section 6.28).

## 6.31 Response-Key

The Response-Key request-header field can be used by a client to request the key that the called user agent SHOULD use to encrypt the response with. The syntax is:

```
Response-Key  =  "Response-Key" ":" key-scheme 1*SP #key-param
key-scheme    =  token
key-param     =  token "=" ( token | quoted-string )
```

The "key-scheme" gives the type of encryption to be used for the response. Section 13 describes security schemes.

If the client insists that the server return an encrypted response, it includes a

```
Require: org.ietf.sip.encrypt-response
```

header field in its request. If the server cannot encrypt for whatever reason, it MUST follow normal Require header field procedures and return a 420 (Bad Extension) response. If this Require header field is not present, a server SHOULD still encrypt if it can.

## 6.32 Retry-After

The Retry-After general-header field can be used with a 503 (Service Unavailable) response to indicate how long the service is expected to be unavailable to the requesting client and with a 404 (Not Found), 600 (Busy), or 603 (Decline) response to indicate when the called party anticipates being available again. The value of this field can be either an SIP-date or an integer number of seconds (in decimal) after the time of the response.

A REGISTER request MAY include this header field when deleting registrations with "Contact: * ;expires: 0". The Retry-After value then indicates when the user might again be reachable. The registrar MAY then include this information in responses to future calls.

An optional comment can be used to indicate additional information about the time of callback. An optional "duration" parameter indicates how long the called party will be reachable starting at the initial time of availability. If no duration parameter is given, the service is assumed to be available indefinitely.

```
Retry-After  =  "Retry-After" ":" ( SIP-date | delta-seconds )
                [ comment ] [ ";" "duration" "=" delta-seconds ]
```

Examples of its use are

```
Retry-After: Mon, 21 Jul 1997 18:48:34 GMT (I'm in a meeting)
```

```
Retry-After: Mon, 01 Jan 9999 00:00:00 GMT
  (Dear John: Don’t call me back, ever)
Retry-After: Fri, 26 Sep 1997 21:00:00 GMT;duration=3600
Retry-After: 120
```

In the third example, the callee is reachable for one hour starting at 21:00 GMT. In the last example, the delay is 2 minutes.

## 6.33 Route

The Route request-header field determines the route taken by a request. Each host removes the first entry and then proxies the request to the host listed in that entry, also using it as the Request-URI. The operation is further described in Section 6.29.

The Route header field has the following syntax:

```
Route  =  "Route" ":" 1# name-addr
```

## 6.34 Server

The Server response-header field contains information about the software used by the user agent server to handle the request. The syntax for this field is defined in [H14.39].

## 6.35 Subject

This is intended to provide a summary, or to indicate the nature, of the call, allowing call filtering without having to parse the session description. (Also, the session description does not have to use the same subject indication as the invitation.)

```
Subject  =  ( "Subject" | "s" ) ":" *TEXT-UTF8
```

Example:

```
Subject: Tune in - they are talking about your work!
```

## 6.36 Timestamp

The timestamp general-header field describes when the client sent the request to the server. The value of the timestamp is of significance only to the client and it MAY use any timescale. The server MUST echo the exact same value and MAY, if it has accurate information about this, add a floating point number indicating the number of seconds that have elapsed since it has received the request. The timestamp is used by the client to compute the round-trip time to the server so that it can adjust the timeout value for retransmissions.

```
Timestamp  =  "Timestamp" ":" *(DIGIT) [ "." *(DIGIT) ] [ delay ]
delay      =  *(DIGIT) [ "." *(DIGIT) ]
```

Note that there MUST NOT be any LWS between a DIGIT and the decimal point.

## 6.37 To

The To general-header field specifies recipient of the request, with the same SIP URL syntax as the From field.

```
To  =  ( "To" | "t" ) ":" ( name-addr | addr-spec )
       *( ";" addr-params )
```

Requests and responses MUST contain a To general-header field, indicating the desired recipient of the request. The optional "display-name" is meant to be rendered by a human-user interface. The UAS or redirect server copies the To header field into its response, and MUST add a "tag" parameter if the request contained more than one Via header field.

> If there was more than one Via header field, the request was handled by at least one proxy server. Since the receiver cannot know whether any of the proxy servers forked the request, it is safest to assume that they might have.

The SIP-URL MUST NOT contain the "transport-param", "maddr-param", "ttl-param", or "headers" elements. A server that receives a SIP-URL with these elements removes them before further processing.

The "tag" parameter serves as a general mechanism to distinguish multiple instances of a user identified by a single SIP URL. As proxies can fork requests, the same request can reach multiple instances of a user (mobile and home phones, for example). As each can respond, there needs to be a means to distinguish the responses from each at the caller. The situation also arises with multicast requests. The tag in the To header field serves to distinguish responses at the UAC. It MUST be placed in the To field of the response by each instance when there is a possibility that the request was forked at an intermediate proxy. The "tag" MUST be added by UAS, registrars and redirect servers, but MUST NOT be inserted into responses forwarded upstream by proxies. The "tag" is added for all definitive responses for all methods, and MAY be added for informational responses from a UAS or redirect server. All subsequent transactions between two entities MUST include the "tag" parameter, as described in Section 11.

See Section 6.21 for details of the "tag" parameter.

The "tag" parameter in To headers is ignored when matching responses to requests that did not contain a "tag" in their To header.

A SIP server returns a 400 (Bad Request) response if it receives a request with a To header field containing a URI with a scheme it does not recognize.

Even if the "display-name" is empty, the "name-addr" form MUST be used if the "addr-spec" contains a comma, question mark, or semicolon.

The following are examples of valid To headers:

```
To: The Operator <sip:operator@cs.columbia.edu>;tag=287447
To: sip:+12125551212@server.phone2net.com
```

> Call-ID, To and From are needed to identify a call leg. The distinction between call and call leg matters in calls with multiple responses from a forked request. The "tag" is added to the To header field in the response to allow forking of future requests for the same call by proxies, while addressing only one of the possibly several responding user agent servers. It also allows several instances of the callee to send requests that can be distinguished.

### 6.38 Unsupported

The Unsupported response-header field lists the features not supported by the server. See Section 6.30 for a usage example and motivation.

Syntax:

```
Unsupported  =  "Unsupported" ":" 1#option-tag
```

### 6.39 User-Agent

The User-Agent general-header field contains information about the client user agent originating the request. The syntax and semantics are defined in [H14.42].

### 6.40 Via

The Via field indicates the path taken by the request so far. This prevents request looping and ensures replies take the same path as the requests, which assists in firewall traversal and other unusual routing situations.

#### 6.40.1 Requests

The client originating the request MUST insert into the request a Via field containing its host name or network address and, if not the default port number, the port number at which it wishes to receive responses. (Note that this port number can differ from the UDP source port number of the request.) A fully-qualified domain name is RECOMMENDED. Each subsequent proxy server that sends the request onwards MUST add its own additional Via field before any existing Via fields. A proxy that receives a redirection (3xx) response and then searches recursively, MUST use the same Via headers as on the original proxied request.

A proxy SHOULD check the top-most Via header field to ensure that it contains the sender’s correct network address, as seen from that proxy. If the sender’s address is incorrect, the proxy MUST add an additional "received" attribute, as described 6.40.2.

> A host behind a network address translator (NAT) or firewall may not be able to insert a network address into the Via header that can be reached by the next hop beyond

the NAT. Use of the received attribute allows SIP requests to traverse NAT's which only modify the source IP address. NAT's which modify port numbers, called Network Address Port Translator's (NAPT) will not properly pass SIP when transported on UDP, in which case an application layer gateway is required. When run over TCP, SIP stands a better chance of traversing NAT's, since its behavior is similar to HTTP in this case (but of course on different ports).

A proxy sending a request to a multicast address MUST add the "maddr" parameter to its Via header field, and SHOULD add the "ttl" parameter. If a server receives a request which contained an "maddr" parameter in the topmost Via field, it SHOULD send the response to the multicast address listed in the "maddr" parameter.

If a proxy server receives a request which contains its own address in the Via header value, it MUST respond with a 482 (Loop Detected) status code.

A proxy server MUST NOT forward a request to a multicast group which already appears in any of the Via headers.

This prevents a malfunctioning proxy server from causing loops. Also, it cannot be guaranteed that a proxy server can always detect that the address returned by a location service refers to a host listed in the Via list, as a single host may have aliases or several network interfaces.

### 6.40.2 Receiver-tagged Via Header Fields

Normally, every host that sends or forwards a SIP message adds a Via field indicating the path traversed. However, it is possible that Network Address Translators (NATs) changes the source address and port of the request (e.g., from net-10 to a globally routable address), in which case the Via header field cannot be relied on to route replies. To prevent this, a proxy SHOULD check the top-most Via header field to ensure that it contains the sender's correct network address, as seen from that proxy. If the sender's address is incorrect, the proxy MUST add a "received" parameter to the Via header field inserted by the previous hop. Such a modified Via header field is known as a receiver-tagged Via header field. An example is:

```
Via: SIP/2.0/UDP erlang.bell-telephone.com:5060
Via: SIP/2.0/UDP 10.0.0.1:5060 ;received=199.172.136.3
```

In this example, the message originated from 10.0.0.1 and traversed a NAT with the external address border.ieee.org (199.172.136.3) to reach erlang.bell-telephone.com. The latter noticed the mismatch, and added a parameter to the previous hop's Via header field, containing the address that the packet actually came from. (Note that the NAT border.ieee.org is not a SIP server.)

### 6.40.3 Responses

Via header fields in responses are processed by a proxy or UAC according to the following rules:

1. The first Via header field should indicate the proxy or client processing this response. If it does not, discard the message. Otherwise, remove this Via field.

2. If there is no second Via header field, this response is destined for this client. Otherwise, the processing depends on whether the Via field contains a "maddr" parameter or is a receiver-tagged field:

   - If the second Via header field contains a "maddr" parameter, send the response to the multicast address listed there, using the port indicated in "sent-by", or port 5060 if none is present. The response SHOULD be sent using the TTL indicated in the "ttl" parameter, or with a TTL of 1 if that parameter is not present. For robustness, responses MUST be sent to the address indicated in the "maddr" parameter even if it is not a multicast address.

   - If the second Via header field does not contain a "maddr" parameter and is a receiver-tagged field (Section 6.40.2), send the message to the address in the "received" parameter, using the port indicated in the "sent-by" value, or using port 5060 if none is present.

   - If neither of the previous cases apply, send the message to the address indicated by the "sent-by" value in the second Via header field.

### 6.40.4 User Agent and Redirect Servers

A UAS or redirect server sends a response based on one of the following rules:

- o If the first Via header field in the request contains a "maddr" parameter, send the response to the multicast address listed there, using the port indicated in "sent-by", or port 5060 if none is present. The response SHOULD be sent using the TTL indicated in the "ttl" parameter, or with a TTL of 1 if that parameter is not present. For robustness, responses MUST be sent to the address indicated in the "maddr" parameter even if it is not a multicast address.

- If the address in the "sent-by" value of the first Via field differs from the source address of the packet, send the response to the actual packet source address, similar to the treatment for receiver-tagged Via header fields (Section 6.40.2).

- If neither of these conditions is true, send the response to the address contained in the "sent-by" value. If the request was sent using TCP, use the existing TCP connection if available.

### 6.40.5 Syntax

The format for a Via header field is shown in Fig. 11. The defaults for "protocol-name" and "transport" are "SIP" and "UDP", respectively. The "maddr" parameter, designating the multicast address, and the "ttl" parameter, designating the time-to-live (TTL) value, are included only if the request was sent via multicast. The "received" parameter is added only for receiver-added Via fields (Section 6.40.2). For reasons of privacy, a client or proxy may wish to hide its Via information by encrypting it (see Section 6.22). The "hidden" parameter is included if this header field was hidden by the upstream proxy (see 6.22). Note that privacy of the proxy relies on the cooperation of the next hop, as the next-hop proxy will, by necessity, know the IP address and port number of the source host.

The "branch" parameter is included by every forking proxy. The token MUST be unique for each distinct request generated when a proxy forks. CANCEL requests MUST have the same branch value as the corresponding forked request. When a response arrives at the proxy it can use the branch value to figure out which branch the response corresponds to. A proxy which generates a single request (non-forking) MAY also insert the "branch" parameter. The identifier has to be unique only within a set of isomorphic requests.

```
Via: SIP/2.0/UDP first.example.com:4000;ttl=16
  ;maddr=224.2.0.1 ;branch=a7c6a8dlze (Example)
Via: SIP/2.0/UDP adk8
```

```
Via              = ( "Via" | "v") ":" 1#( sent-protocol sent-by
                   *( ";" via-params ) [ comment ] )
via-params       = via-hidden | via-ttl | via-maddr
                 | via-received | via-branch
via-hidden       = "hidden"
via-ttl          = "ttl" "=" ttl
via-maddr        = "maddr" "=" maddr
via-received     = "received" "=" host
via-branch       = "branch" "=" token
sent-protocol    = protocol-name "/" protocol-version "/" transport
protocol-name    = "SIP" | token
protocol-version = token
transport        = "UDP" | "TCP" | token
sent-by          = ( host [ ":" port ] ) | ( concealed-host )
concealed-host   = token
ttl              = 1*3DIGIT     ; 0 to 255
```

Figure 11: Syntax of Via header field

## 6.41 Warning

The Warning response-header field is used to carry additional information about the status of a response. Warning headers are sent with responses and have the following format:

```
Warning        =  "Warning" ":" 1#warning-value
warning-value  =  warn-code SP warn-agent SP warn-text
warn-code      =  3DIGIT
warn-agent     =  ( host [ ":" port ] ) | pseudonym
                  ;  the name or pseudonym of the server adding
                  ;  the Warning header, for use in debugging
warn-text      =  quoted-string
```

A response MAY carry more than one Warning header.

The "warn-text" should be in a natural language that is most likely to be intelligible to the human user receiving the response.  This decision can be based on any available knowledge, such as the location of the cache or user, the Accept-Language field in a request, or the Content-Language field in a response. The default language is i-default [31].

Any server MAY add Warning headers to a response. Proxy servers MUST place additional Warning headers before any Authorization headers. Within that constraint, Warning headers MUST be added after any existing Warning headers not covered by a signature. A proxy server MUST NOT delete any Warning header field that it received with a response.

When multiple Warning headers are attached to a response, the user agent SHOULD display as many of them as possible, in the order that they appear in the response. If it is not possible to display all of the warnings, the user agent first displays warnings that appear early in the response.

The warn-code consists of three digits. A first digit of "3" indicates warnings specific to SIP.

This is a list of the currently-defined "warn-code"s, each with a recommended warn-text in English, and a description of its meaning. Note that these warnings describe failures induced by the session description.

Warnings 300 through 329 are reserved for indicating problems with keywords in the session description, 330 through 339 are warnings related to basic network services requested in the session description, 370 through 379 are warnings related to quantitative QoS parameters requested in the session description, and 390 through 399 are miscellaneous warnings that do not fall into one of the above categories.

300 Incompatible network protocol: One or more network protocols contained in the session description are not available.

301 Incompatible network address formats: One or more network address formats contained in the session description are not available.

302 Incompatible transport protocol: One or more transport protocols described in the session description are not available.

303 Incompatible bandwidth units: One or more bandwidth measurement units contained in the session description were not understood.

304 Media type not available: One or more media types contained in the session description are not available.

305 Incompatible media format: One or more media formats contained in the session description are not available.

306 Attribute not understood: One or more of the media attributes in the session description are not supported.

307 Session description parameter not understood: A parameter other than those listed above was not understood.

330 Multicast not available: The site where the user is located does not support multicast.

331 Unicast not available: The site where the user is located does not support unicast communication (usually due to the presence of a firewall).

370 Insufficient bandwidth: The bandwidth specified in the session description or defined by the media exceeds that known to be available.

399 Miscellaneous warning: The warning text can include arbitrary information to be presented to a human user, or logged. A system receiving this warning MUST NOT take any automated action.

1xx and 2xx have been taken by HTTP/1.1.

Additional "warn-code"s, as in the example below, can be defined through IANA.

Examples:

```
Warning: 307 isi.edu "Session parameter 'foo' not understood"
Warning: 301 isi.edu "Incompatible network address type 'E.164'"
```

## 6.42 WWW-Authenticate

The WWW-Authenticate response-header field MUST be included in 401 (Unauthorized) response messages. The field value consists of at least one challenge that indicates the authentication scheme(s) and parameters applicable to the Request-URI. See [H14.46] for a definition of the syntax, and section 14 for an overview of usage.

The content of the "realm" parameter SHOULD be displayed to the user. A user agent SHOULD cache the authorization credentials for a given value of the destination (To header) and "realm" and attempt to re-use these values on the next request for that destination.

In addition to the "basic" and "digest" authentication schemes defined in the specifications cited above, SIP defines a new scheme, PGP (RFC 2015, [32]), Section 15. Other schemes, such as S/MIME, are for further study.

# 7 Status Code Definitions

The response codes are consistent with, and extend, HTTP/1.1 response codes. Not all HTTP/1.1 response codes are appropriate, and only those that are appropriate are given here. Other HTTP/1.1 response codes SHOULD NOT be used. Response codes not defined by HTTP/1.1 have codes x80 upwards to avoid clashes with future HTTP response codes. Also, SIP defines a new class, 6xx. The default behavior for unknown response codes is given for each category of codes.

## 7.1 Informational 1xx

Informational responses indicate that the server or proxy contacted is performing some further action and does not yet have a definitive response. The client SHOULD wait for a further response from the server, and the server SHOULD send such a response without further prompting. A server SHOULD send a 1xx response if it expects to take more than 200 ms to obtain a final response. A server MAY issue zero or more 1xx responses, with no restriction on their ordering or uniqueness. Note that 1xx responses are not transmitted reliably, that is, they do not cause the client to send an ACK. Servers are free to retransmit informational responses and clients can inquire about the current state of call processing by re-sending the request.

### 7.1.1 100 Trying

Some unspecified action is being taken on behalf of this call (e.g., a database is being consulted), but the user has not yet been located.

### 7.1.2 180 Ringing

The called user agent has located a possible location where the user has registered recently and is trying to alert the user.

### 7.1.3 181 Call Is Being Forwarded

A proxy server MAY use this status code to indicate that the call is being forwarded to a different set of destinations.

#### 7.1.4 182 Queued

The called party is temporarily unavailable, but the callee has decided to queue the call rather than reject it. When the callee becomes available, it will return the appropriate final status response. The reason phrase MAY give further details about the status of the call, e.g., "5 calls queued; expected waiting time is 15 minutes". The server MAY issue several 182 responses to update the caller about the status of the queued call.

### 7.2 Successful 2xx

The request was successful and MUST terminate a search.

#### 7.2.1 200 OK

The request has succeeded. The information returned with the response depends on the method used in the request, for example:

BYE: The call has been terminated. The message body is empty.

CANCEL: The search has been cancelled. The message body is empty.

INVITE: The callee has agreed to participate; the message body indicates the callee’s capabilities.

OPTIONS: The callee has agreed to share its capabilities, included in the message body.

REGISTER: The registration has succeeded. The client treats the message body according to its Content-Type.

### 7.3 Redirection 3xx

3xx responses give information about the user’s new location, or about alternative services that might be able to satisfy the call. They SHOULD terminate an existing search, and MAY cause the initiator to begin a new search if appropriate.

Any redirection (3xx) response MUST NOT suggest any of the addresses in the Via (Section 6.40) path of the request in the Contact header field. (Addresses match if their host and port number match.)

To avoid forwarding loops, a user agent client or proxy MUST check whether the address returned by a redirect server equals an address tried earlier.

#### 7.3.1 300 Multiple Choices

The address in the request resolved to several choices, each with its own specific location, and the user (or user agent) can select a preferred communication end point and redirect its request to that location.

The response SHOULD include an entity containing a list of resource characteristics and location(s) from which the user or user agent can choose the one most appropriate, if allowed by the Accept request header. The entity format is specified by the media type given in the Content-Type header field. The choices SHOULD also be listed as Contact fields (Section 6.13). Unlike HTTP, the SIP response MAY contain several Contact fields or a list of addresses in a Contact field. User agents MAY use the Contact header field value for automatic redirection or MAY ask the user to confirm a choice. However, this specification does not define any standard for such automatic selection.

> This status response is appropriate if the callee can be reached at several different locations and the server cannot or prefers not to proxy the request.

#### 7.3.2 301 Moved Permanently

The user can no longer be found at the address in the Request-URI and the requesting client SHOULD retry at the new address given by the Contact header field (Section 6.13). The caller SHOULD update any local directories, address books and user location caches with this new value and redirect future requests to the address(es) listed.

#### 7.3.3 302 Moved Temporarily

The requesting client SHOULD retry the request at the new address(es) given by the Contact header field (Section 6.13). The duration of the redirection can be indicated through an Expires (Section 6.20) header. If there is no explicit expiration time, the address is only valid for this call and MUST NOT be cached for future calls.

#### 7.3.4 305 Use Proxy

The requested resource MUST be accessed through the proxy given by the Contact field. The Contact field gives the URI of the proxy. The recipient is expected to repeat this single request via the proxy. 305 responses MUST only be generated by user agent servers.

### 7.3.5 380 Alternative Service

The call was not successful, but alternative services are possible. The alternative services are described in the message body of the response. Formats for such bodies are not defined here, and may be the subject of future standardization.

## 7.4 Request Failure 4xx

4xx responses are definite failure responses from a particular server. The client SHOULD NOT retry the same request without modification (e.g., adding appropriate authorization). However, the same request to a different server might be successful.

### 7.4.1 400 Bad Request

The request could not be understood due to malformed syntax.

### 7.4.2 401 Unauthorized

The request requires user authentication.

### 7.4.3 402 Payment Required

Reserved for future use.

### 7.4.4 403 Forbidden

The server understood the request, but is refusing to fulfill it. Authorization will not help, and the request SHOULD NOT be repeated.

### 7.4.5 404 Not Found

The server has definitive information that the user does not exist at the domain specified in the Request-URI. This status is also returned if the domain in the Request-URI does not match any of the domains handled by the recipient of the request.

### 7.4.6 405 Method Not Allowed

The method specified in the Request-Line is not allowed for the address identified by the Request-URI. The response MUST include an Allow header field containing a list of valid methods for the indicated address.

### 7.4.7 406 Not Acceptable

The resource identified by the request is only capable of generating response entities which have content characteristics not acceptable according to the accept headers sent in the request.

### 7.4.8 407 Proxy Authentication Required

This code is similar to 401 (Unauthorized), but indicates that the client MUST first authenticate itself with the proxy. The proxy MUST return a Proxy-Authenticate header field (section 6.26) containing a challenge applicable to the proxy for the requested resource. The client MAY repeat the request with a suitable Proxy-Authorization header field (section 6.27). SIP access authentication is explained in section 13.2 and 14.

This status code is used for applications where access to the communication channel (e.g., a telephony gateway) rather than the callee requires authentication.

### 7.4.9 408 Request Timeout

The server could not produce a response, e.g., a user location, within the time indicated in the Expires request-header field. The client MAY repeat the request without modifications at any later time.

### 7.4.10 409 Conflict

The request could not be completed due to a conflict with the current state of the resource. This response is returned if the action parameter in a REGISTER request conflicts with existing registrations.

### 7.4.11 410 Gone

The requested resource is no longer available at the server and no forwarding address is known. This condition is expected to be considered permanent. If the server does not know, or has no facility to determine, whether or not the condition is permanent, the status code 404 (Not Found) SHOULD be used instead.

### 7.4.12 411 Length Required

The server refuses to accept the request without a defined Content-Length. The client MAY repeat the request if it adds a valid Content-Length header field containing the length of the message-body in the request message.

### 7.4.13 413 Request Entity Too Large

The server is refusing to process a request because the request entity is larger than the server is willing or able to process. The server MAY close the connection to prevent the client from continuing the request.

If the condition is temporary, the server SHOULD include a Retry-After header field to indicate that it is temporary and after what time the client MAY try again.

### 7.4.14 414 Request-URI Too Long

The server is refusing to service the request because the Request-URI is longer than the server is willing to interpret.

### 7.4.15 415 Unsupported Media Type

The server is refusing to service the request because the message body of the request is in a format not supported by the requested resource for the requested method. The server SHOULD return a list of acceptable formats using the Accept, Accept-Encoding and Accept-Language header fields.

### 7.4.16 420 Bad Extension

The server did not understand the protocol extension specified in a Require (Section 6.30) header field.

### 7.4.17 480 Temporarily Unavailable

The callee's end system was contacted successfully but the callee is currently unavailable (e.g., not logged in or logged in in such a manner as to preclude communication with the callee). The response MAY indicate a better time to call in the Retry-After header. The user could also be available elsewhere (unbeknownst to this host), thus, this response does not terminate any searches. The reason phrase SHOULD indicate a more precise cause as to why the callee is unavailable. This value SHOULD be setable by the user agent. Status 486 (Busy Here) MAY be used to more precisely indicate a particular reason for the call failure.

This status is also returned by a redirect server that recognizes the user identified by the Request-URI, but does not currently have a valid forwarding location for that user.

### 7.4.18 481 Call Leg/Transaction Does Not Exist

This status is returned under two conditions: The server received a BYE request that does not match any existing call leg or the server received a CANCEL request that does not match any existing transaction. (A server simply discards an ACK referring to an unknown transaction.)

### 7.4.19 482 Loop Detected

The server received a request with a Via (Section 6.40) path containing itself.

### 7.4.20 483 Too Many Hops

The server received a request that contains more Via entries (hops) (Section 6.40) than allowed by the Max-Forwards (Section 6.23) header field.

### 7.4.21 484 Address Incomplete

The server received a request with a To (Section 6.37) address or Request-URI that was incomplete. Additional information SHOULD be provided.

> This status code allows overlapped dialing. With overlapped dialing, the client does not know the length of the dialing string. It sends strings of increasing lengths, prompting the user for more input, until it no longer receives a 484 status response.

### 7.4.22 485 Ambiguous

The callee address provided in the request was ambiguous. The response MAY contain a listing of possible unambiguous addresses in Contact headers.

Revealing alternatives can infringe on privacy concerns of the user or the organization. It MUST be possible to configure a server to respond with status 404 (Not Found) or to suppress the listing of possible choices if the request address was ambiguous.

Example response to a request with the URL lee@example.com :

```
485 Ambiguous SIP/2.0
Contact: Carol Lee <sip:carol.lee@example.com>
```

Contact: Ping Lee <sip:p.lee@example.com>
Contact: Lee M. Foote <sip:lee.foote@example.com>

Some email and voice mail systems provide this functionality. A status code separate from 3xx is used since the semantics are different: for 300, it is assumed that the same person or service will be reached by the choices provided. While an automated choice or sequential search makes sense for a 3xx response, user intervention is required for a 485 response.

### 7.4.23 486 Busy Here

The callee's end system was contacted successfully but the callee is currently not willing or able to take additional calls. The response MAY indicate a better time to call in the Retry-After header. The user could also be available elsewhere, such as through a voice mail service, thus, this response does not terminate any searches. Status 600 (Busy Everywhere) SHOULD be used if the client knows that no other end system will be able to accept this call.

## 7.5 Server Failure 5xx

5xx responses are failure responses given when a server itself has erred. They are not definitive failures, and MUST NOT terminate a search if other possible locations remain untried.

### 7.5.1 500 Server Internal Error

The server encountered an unexpected condition that prevented it from fulfilling the request. The client MAY display the specific error condition, and MAY retry the request after several seconds.

### 7.5.2 501 Not Implemented

The server does not support the functionality required to fulfill the request. This is the appropriate response when the server does not recognize the request method and is not capable of supporting it for any user.

### 7.5.3 502 Bad Gateway

The server, while acting as a gateway or proxy, received an invalid response from the downstream server it accessed in attempting to fulfill the request.

### 7.5.4 503 Service Unavailable

The server is currently unable to handle the request due to a temporary overloading or maintenance of the server. The implication is that this is a temporary condition which will be alleviated after some delay. If known, the length of the delay MAY be indicated in a Retry-After header. If no Retry-After is given, the client MUST handle the response as it would for a 500 response.

Note: The existence of the 503 status code does not imply that a server has to use it when becoming overloaded. Some servers MAY wish to simply refuse the connection.

### 7.5.5 504 Gateway Time-out

The server, while acting as a gateway, did not receive a timely response from the server (e.g., a location server) it accessed in attempting to complete the request.

### 7.5.6 505 Version Not Supported

The server does not support, or refuses to support, the SIP protocol version that was used in the request message. The server is indicating that it is unable or unwilling to complete the request using the same major version as the client, other than with this error message. The response MAY contain an entity describing why that version is not supported and what other protocols are supported by that server. The format for such an entity is not defined here and may be the subject of future standardization.

## 7.6 Global Failures 6xx

6xx responses indicate that a server has definitive information about a particular user, not just the particular instance indicated in the Request-URI. All further searches for this user are doomed to failure and pending searches SHOULD be terminated.

### 7.6.1 600 Busy Everywhere

The callee's end system was contacted successfully but the callee is busy and does not wish to take the call at this time. The response MAY indicate a better time to call in the Retry-After header. If the callee does not wish to reveal the reason for declining the call, the callee uses status code 603 (Decline) instead. This status response is returned only if the client knows that no other end point (such as a voice mail system) will answer the request. Otherwise, 486 (Busy Here) should be returned.

### 7.6.2 603 Decline

The callee's machine was successfully contacted but the user explicitly does not wish to or cannot participate. The response MAY indicate a better time to call in the Retry-After header.

### 7.6.3 604 Does Not Exist Anywhere

The server has authoritative information that the user indicated in the To request field does not exist anywhere. Searching for the user elsewhere will not yield any results.

### 7.6.4 606 Not Acceptable

The user's agent was contacted successfully but some aspects of the session description such as the requested media, bandwidth, or addressing style were not acceptable.

A 606 (Not Acceptable) response means that the user wishes to communicate, but cannot adequately support the session described. The 606 (Not Acceptable) response MAY contain a list of reasons in a Warning header field describing why the session described cannot be supported. Reasons are listed in Section 6.41. It is hoped that negotiation will not frequently be needed, and when a new user is being invited to join an already existing conference, negotiation may not be possible. It is up to the invitation initiator to decide whether or not to act on a 606 (Not Acceptable) response.

# 8 SIP Message Body

## 8.1 Body Inclusion

Requests MAY contain message bodies unless otherwise noted. Within this specification, the BYE request MUST NOT contain a message body. For ACK, INVITE and OPTIONS, the message body is always a session description. The use of message bodies for REGISTER requests is for further study.

For response messages, the request method and the response status code determine the type and interpretation of any message body. All responses MAY include a body. Message bodies for 1xx responses contain advisory information about the progress of the request. 2xx responses to INVITE requests contain session descriptions. In 3xx responses, the message body MAY contain the description of alternative destinations or services, as described in Section 7.3. For responses with status 400 or greater, the message body MAY contain additional, human-readable information about the reasons for failure. It is RECOMMENDED that information in 1xx and 300 and greater responses be of type text/plain or text/html

## 8.2 Message Body Type

The Internet media type of the message body MUST be given by the Content-Type header field. If the body has undergone any encoding (such as compression) then this MUST be indicated by the Content-Encoding header field, otherwise Content-Encoding MUST be omitted. If applicable, the character set of the message body is indicated as part of the Content-Type header-field value.

## 8.3 Message Body Length

The body length in bytes SHOULD be given by the Content-Length header field. Section 6.15 describes the behavior in detail.

The "chunked" transfer encoding of HTTP/1.1 MUST NOT be used for SIP. (Note: The chunked encoding modifies the body of a message in order to transfer it as a series of chunks, each with its own size indicator.)

# 9 Compact Form

When SIP is carried over UDP with authentication and a complex session description, it may be possible that the size of a request or response is larger than the MTU. To address this problem, a more compact form of SIP is also defined by using abbreviations for the common header fields listed below:

| short field name | long field name | note |
|---|---|---|
| c | Content-Type | |
| e | Content-Encoding | |
| f | From | |
| i | Call-ID | |
| m | Contact | from "moved" |
| l | Content-Length | |
| s | Subject | |
| t | To | |
| v | Via | |

Thus, the message in section 16.2 could also be written:

```
INVITE sip:schooler@vlsi.caltech.edu SIP/2.0
v:SIP/2.0/UDP 131.215.131.131;maddr=239.128.16.254;ttl=16
v:SIP/2.0/UDP 128.16.64.19
f:sip:mjh@isi.edu
t:sip:schooler@cs.caltech.edu
i:62729-27@128.16.64.19
c:application/sdp
CSeq: 4711 INVITE
l:187

v=0
o=user1 53655765 2353687637 IN IP4 128.3.4.5
s=Mbone Audio
i=Discussion of Mbone Engineering Issues
e=mbone@somewhere.com
c=IN IP4 224.2.0.1/127
t=0 0
m=audio 3456 RTP/AVP 0
```

Clients MAY mix short field names and long field names within the same request. Servers MUST accept both short and long field names for requests. Proxies MAY change header fields between their long and short forms, but this MUST NOT be done to fields following an Authorization header.

# 10 Behavior of SIP Clients and Servers

## 10.1 General Remarks

SIP is defined so it can use either UDP (unicast or multicast) or TCP as a transport protocol; it provides its own reliability mechanism.

### 10.1.1 Requests

Servers discard isomorphic requests, but first retransmit the appropriate response. (SIP requests are said to be idempotent , i.e., receiving more than one copy of a request does not change the server state.)

After receiving a CANCEL request from an upstream client, a stateful proxy server MAY send a CANCEL on all branches where it has not yet received a final response.

When a user agent receives a request, it checks the Call-ID against those of in-progress calls. If the Call-ID was found, it compares the tag value of To with the user’s tag and rejects the request if the two do not match. If the From header, including any tag value, matches the value for an existing call leg, the server compares the CSeq header field value. If less than or equal to the current sequence number, the request is a retransmission. Otherwise, it is a new request. If the From header does not match an existing call leg, a new call leg is created.

If the Call-ID was not found, a new call leg is created, with entries for the To, From and Call-ID headers. In this case, the To header field should not have contained a tag. The server returns a response containing the same To value, but with a unique tag added. The tag MAY be omitted if the request contained only one Via header field.

### 10.1.2 Responses

A server MAY issue one or more provisional responses at any time before sending a final response. If a stateful proxy, user agent server, redirect server or registrar cannot respond to a request with a final response within 200 ms, it SHOULD issue a provisional (1xx) response as soon as possible. Stateless proxies MUST NOT issue provisional responses on their own.

Responses are mapped to requests by the matching To, From, Call-ID, CSeq headers and the branch parameter of the first Via header. Responses terminate request retransmissions even if they have Via headers that cause them to be delivered to an upstream client.

A stateful proxy may receive a response that it does not have state for, that is, where it has no a record of an associated request. If the Via header field indicates that the upstream server used TCP, the proxy actively opens a TCP connection to that address. Thus, proxies have to be prepared to receive responses on the incoming side of passive TCP connections, even though most responses will arrive on the incoming side of an active connection. (An active connection is a TCP connection initiated by the proxy, a passive connection is one accepted by the proxy, but initiated by another entity.)

100 responses SHOULD NOT be forwarded, other 1xx responses MAY be forwarded, possibly after the server eliminates responses with status codes that had already been sent earlier. 2xx responses are forwarded according to the Via header. Once a stateful proxy has received a 2xx response, it MUST NOT forward non-2xx final responses. Responses with status 300 and higher are retransmitted by each stateful proxy until the next upstream proxy sends an ACK (see below for timing details) or CANCEL.

A stateful proxy SHOULD maintain state for at least 32 seconds after the receipt of the first definitive non-200 response, in order to handle retransmissions of the response.

> The 32 second window is given by the maximum retransmission duration of 200-class responses using the default timers, in case the ACK is lost somewhere on the way to the called user agent or the next stateful proxy.

### 10.2 Source Addresses, Destination Addresses and Connections

#### 10.2.1 Unicast UDP

Responses are returned to the address listed in the Via header field (Section 6.40), not the source address of the request.

> Recall that responses are not generated by the next-hop stateless server, but generated by either a proxy server or the user agent server. Thus, the stateless proxy can only use the Via header field to forward the response.

#### 10.2.2 Multicast UDP

Requests MAY be multicast; multicast requests likely feature a host-independent Request-URI. This request SHOULD be scoped to ensure it is not forwarded beyond the boundaries of the administrative system. This MAY be done with either TTL or administrative scopes[25], depending on what is implemented in the network.

A client receiving a multicast query does not have to check whether the host part of the Request-URI matches its own host or domain name. If the request was received via multicast, the response is also returned via multicast. Responses to multicast requests are multicast with the same TTL as the request, where the TTL is derived from the ttl parameter in the Via header (Section 6.40).

To avoid response implosion, servers MUST NOT answer multicast requests with a status code other than 2xx or 6xx. The server delays its response by a random interval uniformly distributed between zero and one second. Servers MAY suppress responses if they hear a lower-numbered or 6xx response from another group member prior to sending. Servers do not respond to CANCEL requests received via multicast to avoid request implosion. A proxy or UAC SHOULD send a CANCEL on receiving the first 2xx or 6xx response to a multicast request.

Server response suppression is a MAY since it requires a server to violate some basic message processing rules. Lets say A sends a multicast request, and it is received by B,C, and D. B sends a 200 response. The topmost Via field in the response will contain the address of A. C will also receive this response, and could use it to suppress its own response. However, C would normally not examine this response, as the topmost Via is not its own. Normally, a response received with an incorrect topmost Via MUST be dropped, but not in this case. To distinguish this packet from a misrouted or multicast looped packet is fairly complex, and for this reason the procedure is a MAY. The CANCEL, instead, provides a simpler and more standard way to perform response suppression. It is for this reason that the use of CANCEL here is a SHOULD

## 10.3 TCP

A single TCP connection can serve one or more SIP transactions. A transaction contains zero or more provisional responses followed by one or more final responses. (Typically, transactions contain exactly one final response, but there are exceptional circumstances, where, for example, multiple 200 responses can be generated.)

The client SHOULD keep the connection open at least until the first final response arrives. If the client closes or resets the TCP connection prior to receiving the first final response, the server treats this action as equivalent to a CANCEL request.

This behavior makes it less likely that malfunctioning clients cause a proxy server to keep connection state indefinitely.

The server SHOULD NOT close the TCP connection until it has sent its final response, at which point it MAY close the TCP connection if it wishes to. However, normally it is the client’s responsibility to close the connection.

If the server leaves the connection open, and if the client so desires it MAY re-use the connection for further SIP requests or for requests from the same family of protocols (such as HTTP or stream control commands).

If a server needs to return a response to a client and no longer has a connection open to that client, it MAY open a connection to the address listed in the Via header. Thus, a proxy or user agent MUST be prepared to receive both requests and responses on a "passive" connection.

### 10.4 Reliability for BYE, CANCEL, OPTIONS, REGISTER Requests

#### 10.4.1 UDP

A SIP client using UDP SHOULD retransmit a BYE, CANCEL, OPTIONS, or REGISTER request with an exponential backoff, starting at a T1 second interval, doubling the interval for each packet, and capping off at a T2 second interval. This means that after the first packet is sent, the second is sent T1 seconds later, the next 2*T1 seconds after that, the next 4*T1 seconds after that, and so on, until the interval hits T2. Subsequent retransmissions are spaced by T2 seconds. If the client receives a provisional response, it continues to retransmit the request, but with an interval of T2 seconds. Retransmissions cease when the client has sent a total of eleven packets, or receives a definitive response. Default values for T1 and T2 are 500 ms and 4 s, respectively. Clients MAY use larger values, but SHOULD NOT use smaller ones. Servers retransmit the response upon receipt of a request retransmission. After the server sends a final response, it cannot be sure the client has received the response, and thus SHOULD cache the results for at least 10*T2 seconds to avoid having to, for example, contact the user or location server again upon receiving a request retransmission.

> Use of the exponential backoff is for congestion control purposes. However, the back-off must cap off, since request retransmissions are used to trigger response retransmissions at the server. Without a cap, the loss of a single response could significantly increase transaction latencies.

The value of the initial retransmission timer is smaller than that that for TCP since it is expected that network paths suitable for interactive communications have round-trip times smaller than 500 ms. For congestion control purposes, the retransmission count has to be bounded. Given that most transactions are expected to consist of one request and a few responses, round-trip time estimation is not likely to be very useful. If RTT estimation is desired to more quickly discover a missing final response, each request retransmission needs to be labeled with its own Timestamp (Section 6.36), returned in the response. The server caches the result until it can be sure that the client will not retransmit the same request again.

Each server in a proxy chain generates its own final response to a CANCEL request. The server responds immediately upon receipt of the CANCEL request rather than waiting until it has received final responses from the CANCEL requests it generates.

BYE and OPTIONS final responses are generated by redirect and user agent servers; REGISTER final responses are generated by registrars. Note that in contrast to the reliability mechanism described in Section 10.5, responses to these requests are not retransmitted periodically and not acknowledged via ACK.

### 10.4.2 TCP

Clients using TCP do not need to retransmit requests.

## 10.5 Reliability for INVITE Requests

Special considerations apply for the INVITE method.

1. After receiving an invitation, considerable time can elapse before the server can determine the outcome. For example, if the called party is "rung" or extensive searches are performed, delays between the request and a definitive response can reach several tens of seconds. If either caller or callee are automated servers not directly controlled by a human being, a call attempt could be unbounded in time.

2. If a telephony user interface is modeled or if we need to interface to the PSTN, the caller’s user interface will provide "ringback", a signal that the callee is being alerted. (The status response 180 (Ringing) MAY be used to initiate ringback.) Once the callee picks up, the caller needs to know so that it can enable the voice path and stop ringback. The callee’s response to the invitation could get lost. Unless the response is transmitted reliably, the caller will continue to hear ringback while the callee assumes that the call exists.

3. The client has to be able to terminate an on-going request, e.g., because it is no longer willing to wait for the connection or search to succeed. The server will have to wait several retransmission intervals to interpret the lack of request retransmissions as the end of a call. If the call succeeds shortly after the caller has given up, the callee will "pick up the phone" and not be "connected".

### 10.5.1 UDP

For UDP, A SIP client SHOULD retransmit a SIP INVITE request with an interval that starts at T1 seconds, and doubles after each packet transmission. The client ceases retransmissions if it receives a provisional or definitive response, or once it has sent a total of 7 request packets.

A server which transmits a provisional response should retransmit it upon reception of a duplicate request. A server which transmits a final response should retransmit it with an interval that starts at T1 seconds, and doubles for each subsequent packet. Response retransmissions cease when any one of the following occurs:

1. An ACK request for the same transaction is received;

2. a BYE request for the same call leg is received;

3. a CANCEL request for the same call leg is received and the final response status was equal or greater to 300;

4. the response has been transmitted 7 times.

Only the user agent client generates an ACK for 2xx final responses, If the response contained a Contact header field, the ACK MAY be sent to the address listed in that Contact header field. If the response did not contain a Contact header, the client uses the same To header field and Request-URI as for the INVITE request and sends the ACK to the same destination as the original INVITE request. ACKs for final responses other than 2xx are sent to the same server that the original request was sent to, using the same Request-URI as the original request. Note, however, that the To header field in the ACK is copied from the response being acknowledged, not the request, and thus MAY additionally contain the tag parameter. Also note than unlike 2xx final responses, a proxy generates an ACK for non-2xx final responses.

The ACK request MUST NOT be acknowledged to prevent a response-ACK feedback loop. Fig. 12 and 13 show the client and server state diagram for invitations.

> The mechanism in Sec. 10.4 would not work well for INVITE because of the long delays between INVITE and a final response. If the 200 response were to get lost, the callee would believe the call to exist, but the voice path would

```
              +===========+
              *           *
  ...........>*  Initial  *<;;;;;;;;;;
  : 7 INVITE  *           *          ;
  :   sent    +===========+          ;
  :                 |                ;
  :                 |    -           ;
  :                 |  INVITE        ;
  :                 |                ;
  :                 v                ;
  :           *************          ;
  : T1*2^n <--*           *          ;
  : INVITE -->*  Calling  *--------+ ;
  :           *           *        | ;
  :           *************        | ;
  :             :   |              | ;
  :.............:   | 1xx      xxx | ;
                    |  -       ACK | ;
                    |              | ;
                    v              | ;
              *************        | ;
              *           *        | ;
              *  Ringing  *<->1xx  | ;
              *           *        | ;
              *************        | ;
                    |              | ;
                    |<-------------+ ;
                    |                ;
                    v                ;
              *************          ;
      xxx  <--*           *          ;
      ACK  -->* Completed *          ;
              *           *          ;
              *************          ;
                    ; 32s (for proxy);
                    ;;;;;;;;;;;;;;;;;;

 event (xxx=status)
     message
```

Figure 12: State transition diagram of client for INVITE method

```
   7 pkts sent  +===============+
+-------------->*               *
|               *   Initial     *<...............
|;;;;;;;;;;;;;;>*               *               :
|;              +===============+               :
|; CANCEL               !                       :
|;  200                 !  INVITE               :
|;                      !   1xx                 :
|;                      !                       :
|;                      v                       :
|;              *****************          BYE  :
|;    INVITE -->*               *          200  :
|;      1xx  <--* Call proceed. *..............>:
|;              *               *               :
|;;;;;;;;;;;;;;;*****************               :
|;                    !   !                     :
|:                    !   !                     :
|;         failure    !   !  picks up           :
|;         >= 300     !   !    200              :
|;            +-------+   +-------+             :
|;            v                   v             :
|;       ***********         ***********        :
|;INVITE<*         *<T1*2^n->*         *>INVITE :
|;status>* failure *>status<-* success *<status :
|;       *         *         *         *        :
|;;;;;;;;***********         ***********        :
|             ! : |             |  !  :         :
|             ! : |             |  !  :         :
+-------------!-:-+-------------+  !  :         :
              ! :................!..:.........>:
              !                  !          BYE :
              +---------+--------+          200 :
  event                 ! ACK                   :
message sent            v                       :
                ****************                :
            V---*               *               :
           ACK  *   Confirmed   *               :
            |-->*               *               :
                ****************                .
                        :.......................>:
```

   Figure 13: State transition diagram of server for INVITE method

be dead since the caller does not know that the callee has picked up. Thus, the INVITE retransmission interval would have to be on the order of a second or two to limit the duration of this state confusion. Retransmitting the response with an exponential back-off helps ensure that the response is received, without placing an undue burden on the network.

### 10.5.2 TCP

A user agent using TCP MUST NOT retransmit requests, but uses the same algorithm as for UDP (Section 10.5.1) to retransmit responses until it receives an ACK.

It is necessary to retransmit 2xx responses as their reliability is assured end-to-end only. If the chain of proxies has a UDP link in the middle, it could lose the response, with no possibility of recovery. For simplicity, we also retransmit non-2xx responses, although that is not strictly necessary.

## 10.6 Reliability for ACK Requests

The ACK request does not generate responses. It is only generated when a response to an INVITE request arrives (see Section 10.5). This behavior is independent of the transport protocol. Note that the ACK request MAY take a different path than the original INVITE request, and MAY even cause a new TCP connection to be opened in order to send it.

## 10.7 ICMP Handling

Handling of ICMP messages in the case of UDP messages is straightforward. For requests, a host, network, port, or protocol unreachable error SHOULD be treated as if a 400-class response was received. For responses, these errors SHOULD cause the server to cease retransmitting the response.

Source quench ICMP messages SHOULD be ignored. TTL exceeded errors SHOULD be ignored. Parameter problem errors SHOULD be treated as if a 400-class response was received.

# 11 Behavior of SIP User Agents

This section describes the rules for user agent client and servers for generating and processing requests and responses.

### 11.1 Caller Issues Initial INVITE Request

When a user agent client desires to initiate a call, it formulates an INVITE request. The To field in the request contains the address of the callee. The Request-URI contains the same address. The From field contains the address of the caller. If the From address can appear in requests generated by other user agent clients for the same call, the caller MUST insert the tag parameter in the From field. A UAC MAY optionally add a Contact header containing an address where it would like to be contacted for transactions from the callee back to the caller.

### 11.2 Callee Issues Response

When the initial INVITE request is received at the callee, the callee can accept, redirect, or reject the call. In all of these cases, it formulates a response. The response MUST copy the To, From, Call-ID, CSeq and Via fields from the request. Additionally, the responding UAS MUST add the tag parameter to the To field in the response if the request contained more than one Via header field. Since a request from a UAC may fork and arrive at multiple hosts, the tag parameter serves to distinguish, at the UAC, multiple responses from different UAS’s. The UAS MAY add a Contact header field in the response. It contains an address where the callee would like to be contacted for subsequent transactions, including the ACK for the current INVITE. The UAS stores the values of the To and From field, including any tags. These become the local and remote addresses of the call leg, respectively.

### 11.3 Caller Receives Response to Initial Request

Multiple responses may arrive at the UAC for a single INVITE request, due to a forking proxy. Each response is distinguished by the "tag" parameter in the To header field, and each represents a distinct call leg. The caller MAY choose to acknowledge or terminate the call with each responding UAS. To acknowledge, it sends an ACK request, and to terminate it sends a BYE request. The To header field in the ACK or BYE MUST be the same as the To field in the 200 response, including any tag. The From header field MUST be the same as the From header field in the 200 (OK) response, including any tag. The Request-URI of the ACK or BYE request MAY be set to whatever address was found in the Contact header field in the 200 (OK) response, if present. Alternately, a UAC may copy the address from the To header field into the Request-URI. The UAC also notes the value of the To and From header fields in each response. For each call leg, the To header field becomes the remote address, and the From header field becomes the local address.

### 11.4 Caller or Callee Generate Subsequent Requests

Once the call has been established, either the caller or callee MAY generate INVITE or BYE requests to change or terminate the call. Regardless of whether the caller or callee is generating the new request, the header fields in the request are set as follows. For the desired call leg, the To header field is set to the remote address, and the From header field is set to the local address (both including any tags). The Contact header field MAY be different than the Contact header field sent in a previous response or request. The Request-URI MAY be set to the value of the Contact header field received in a previous request or response from the remote party, or to the value of the remote address.

### 11.5 Receiving Subsequent Requests

When a request is received subsequently, the following checks are made:

1. If the Call-ID is new, the request is for a new call, regardless of the values of the To and From header fields.

2. If the Call-ID exists, the request is for an existing call. If the To, From, Call-ID, and CSeq values exactly match (including tags) those of any requests received previously, the request is a retransmission.

3. If there was no match to the previous step, the To and From fields are compared against existing call leg local and remote addresses. If there is a match, and the CSeq in the request is higher than the last CSeq received on that leg, the request is a new transaction for an existing call leg.

## 12 Behavior of SIP Proxy and Redirect Servers

This section describes behavior of SIP redirect and proxy servers in detail. Proxy servers can "fork" connections, i.e., a single incoming request spawns several outgoing (client) requests.

### 12.1 Redirect Server

A redirect server does not issue any SIP requests of its own. After receiving a request other than CANCEL, the server gathers the list of alternative locations and returns a final response of class 3xx or it refuses the request. For well-formed CANCEL requests, it SHOULD return a 2xx response. This response ends the SIP transaction. The redirect server maintains transaction state for the whole SIP transaction. It is up to the client to detect forwarding loops between redirect servers.

## 12.2 User Agent Server

User agent servers behave similarly to redirect servers, except that they also accept requests and can return a response of class 2xx.

## 12.3 Proxy Server

This section outlines processing rules for proxy servers. A proxy server can either be stateful or stateless. When stateful, a proxy remembers the incoming request which generated outgoing requests, and the outgoing requests. A stateless proxy forgets all information once an outgoing request is generated. A forking proxy SHOULD be stateful. Proxies that accept TCP connections MUST be stateful.

> Otherwise, if the proxy were to lose a request, the TCP client would never retransmit it.

A stateful proxy SHOULD NOT become stateless until after it sends a definitive response upstream, and at least 32 seconds after it received a definitive response.

A stateful proxy acts as a virtual UAS/UAC. It implements the server state machine when receiving requests, and the client state machine for generating outgoing requests, with the exception of receiving a 2xx response to an INVITE. Instead of generating an ACK, the 2xx response is always forwarded upstream towards the caller. Furthermore, ACK's for 200 responses to INVITE's are always proxied downstream towards the UAS, as they would be for a stateless proxy.

A stateless proxy does not act as a virtual UAS/UAC (as this would require state). Rather, a stateless proxy forwards every request it receives downstream, and every response it receives upstream.

### 12.3.1 Proxying Requests

To prevent loops, a server MUST check if its own address is already contained in the Via header field of the incoming request.

The To, From, Call-ID, and Contact tags are copied exactly from the original request. The proxy SHOULD change the Request-URI to indicate the server where it intends to send the request.

A proxy server always inserts a Via header field containing its own address into those requests that are caused by an incoming request. Each proxy MUST insert a "branch" parameter (Section 6.40).

#### 12.3.2 Proxying Responses

A proxy only processes a response if the topmost Via field matches one of its addresses. A response with a non-matching top Via field MUST be dropped.

#### 12.3.3 Stateless Proxy: Proxying Responses

A stateless proxy removes its own Via field, and checks the address in the next Via field. In the case of UDP, the response is sent to the address listed in the "maddr" tag if present, otherwise to the "received" tag if present, and finally to the address in the "sent-by" field. A proxy MUST remain stateful when handling requests received via TCP.

A stateless proxy MUST NOT generate its own provisional responses.

#### 12.3.4 Stateful Proxy: Receiving Requests

When a stateful proxy receives a request, it checks the To, From (including tags), Call-ID and CSeq against existing request records. If the tuple exists, the request is a retransmission. The provisional or final response sent previously is retransmitted, as per the server state machine. If the tuple does not exist, the request corresponds to a new transaction, and the request should be proxied.

A stateful proxy server MAY generate its own provisional (1xx) responses.

#### 12.3.5 Stateful Proxy: Receiving ACKs

When an ACK request is received, it is either processed locally or proxied. To make this determination, the To, From, CSeq and Call-ID fields are compared against those in previous requests. If there is no match, the ACK request is proxied as if it were an INVITE request. If there is a match, and if the server had ever sent a 200 response upstream, the ACK is proxied. If the server had never sent any responses upstream, the ACK is also proxied. If the server had sent a 3xx, 4xx, 5xx or 6xx response, but no 2xx response, the ACK is processed locally if the tag in the To field of the ACK matches the tag sent by the proxy in the response.

### 12.3.6 Stateful Proxy: Receiving Responses

When a proxy server receives a response that has passed the Via checks, the proxy server checks the To (without the tag), From (including the tag), Call-ID and CSeq against values seen in previous requests. If there is no match, the response is forwarded upstream to the address listed in the Via field. If there is a match, the "branch" tag in the Via field is examined. If it matches a known branch identifier, the response is for the given branch, and processed by the virtual client for the given branch. Otherwise, the response is dropped.

A stateful proxy should obey the rules in Section 12.4 to determine if the response should be proxied upstream. If it is to be proxied, the same rules for stateless proxies above are followed, with the following addition for TCP. If a request was received via TCP (indicated by the protocol in the top Via header), the proxy checks to see if it has a connection currently open to that address. If so, the response is sent on that connection. Otherwise, a new TCP connection is opened to the address and port in the Via field, and the response is sent there. Note that this implies that a UAC or proxy MUST be prepared to receive responses on the incoming side of a TCP connection. Definitive non 200-class responses MUST be retransmitted by the proxy, even over a TCP connection.

### 12.3.7 Stateless, Non-Forking Proxy

Proxies in this category issue at most a single unicast request for each incoming SIP request, that is, they do not "fork" requests. However, servers MAY choose to always operate in a mode that allows issuing of several requests, as described in Section 12.4.

The server can forward the request and any responses. It does not have to maintain any state for the SIP transaction. Reliability is assured by the next redirect or stateful proxy server in the server chain.

A proxy server SHOULD cache the result of any address translations and the response to speed forwarding of retransmissions. After the cache entry has been expired, the server cannot tell whether an incoming request is actually a retransmission of an older request. The server will treat it as a new request and commence another search.

## 12.4 Forking Proxy

The server MUST respond to the request immediately with a 100 (Trying) response.

Successful responses to an INVITE request MAY contain a Contact header field so that the following ACK or BYE bypasses the proxy search mechanism. If the proxy requires future requests to be routed through it, it adds a Record-Route header to the request (Section 6.29).

The following C-code describes the behavior of a proxy server issuing several requests in response to an incoming INVITE request. The function request(r, a, b) sends a SIP request of type r to address a, with branch id b. await_response() waits until a response is received and returns the response. close(a) closes the TCP connection to client with address a. response(r) sends a response to the client. ismulticast() returns 1 if the location is a multicast address and zero otherwise. The variable timeleft indicates the amount of time left until the maximum response time has expired. The variable recurse indicates whether the server will recursively try addresses returned through a 3xx response. A server MAY decide to recursively try only certain addresses, e.g., those which are within the same domain as the proxy server. Thus, an initial multicast request can trigger additional unicast requests.

```
/* request type */
typedef enum {INVITE, ACK, BYE, OPTIONS, CANCEL, REGISTER} Method;

process_request(Method R, int N, address_t address[])
{
  struct {
    int branch;          /* branch id */
    int done;            /* has responded */
  } outgoing[];
  int done[];            /* address has responded */
  char *location[];      /* list of locations */
  int heard = 0;         /* number of sites heard from */
  int class;             /* class of status code */
  int timeleft = 120;    /* sample timeout value */
  int loc = 0;           /* number of locations */
  struct {               /* response */
    int status;          /* response: CANCEL=-1 */
    int locations;       /* number of redirect locations */
    char *location[];    /* redirect locations */
    address_t a;         /* address of respondent */
    int branch;          /* branch identifier */
  } r, best;             /* response, best response */
  int i;

  best.status = 1000;
  for (i = 0; i < N; i++) {
```

```
          request(R, address[i], i);
          outgoing[i].done = 0;
          outgoing[i].branch = i;
        }

        while (timeleft > 0 && heard < N) {
          r = await_response();
          class = r.status / 100;

          /* If final response, mark branch as done. */
          if (class >= 2) {
            heard++;
            for (i = 0; i < N; i++) {
              if (r.branch == outgoing[i].branch) {
                outgoing[i].done = 1;
                break;
              }
            }
          }
          /* CANCEL: respond, fork and wait for responses */
          else if (class < 0) {
            best.status = 200;
            response(best);
            for (i = 0; i < N; i++) {
              if (!outgoing[i].done)
                request(CANCEL, address[i], outgoing[i].branch);
            }
            best.status = -1;
          }

          /* Send an ACK */

          if (class != 2) {
            if (R == INVITE) request(ACK, r.a, r.branch);
          }


          if (class == 2) {
            if (r.status < best.status) best = r;
            break;
          }
          else if (class == 3) {
            /* A server MAY optionally recurse.  The server MUST check
             * whether it has tried this location before and whether
             * the location is part of the Via path of the incoming
             * request.  This check is omitted here for brevity.
             * Multicast locations MUST NOT be returned to the client if
             * the server is not recursing.
```

```
            */
           if (recurse) {
             multicast = 0;
             N += r.locations;
             for (i = 0; i < r.locations; i++) {
               request(R, r.location[i]);
             }
           } else if (!ismulticast(r.location)) {
             best = r;
           }
         }
         else if (class == 4) {
           if (best.status >= 400) best = r;
         }
         else if (class == 5) {
           if (best.status >= 500) best = r;
         }
         else if (class == 6) {
           best = r;
           break;
         }
       }

       /* We haven't heard anything useful from anybody. */
       if (best.status == 1000) {
         best.status = 404;
       }
       if (best.status/100 != 3) loc = 0;
       response(best);
     }
```

Responses are processed as follows. The process completes (and state can be freed) when all requests have been answered by final status responses (for unicast) or 60 seconds have elapsed (for multicast). A proxy MAY send a CANCEL to all branches and return a 408 (Timeout) to the client after 60 seconds or more.

1xx: The proxy MAY forward the response upstream towards the client.

2xx: The proxy MUST forward the response upstream towards the client, without sending an ACK downstream. After receiving a 2xx, the server MAY terminate all other pending requests by sending a CANCEL request and closing the TCP connection, if applicable. (Terminating pending requests is advisable as searches consume resources. Also, INVITE requests could "ring" on a number of workstations if the callee is currently logged in more than once.)

3xx: The proxy MUST send an ACK and MAY recurse on the listed Contact addresses. Otherwise, the lowest-numbered response is returned if there were no 2xx responses.

Location lists are not merged as that would prevent forwarding of authenticated responses. Also, responses can have message bodies, so that merging is not feasible.

4xx, 5xx: The proxy MUST send an ACK and remember the response if it has a lower status code than any previous 4xx and 5xx responses. On completion, the lowest-numbered response is returned if there were no 2xx or 3xx responses.

6xx: The proxy MUST forward the response to the client and send an ACK. Other pending requests MAY be terminated with CANCEL as described for 2xx responses.

A proxy server forwards any response for Call-IDs for which it does not have a pending transaction according to the response's Via header. User agent servers respond to BYE requests for unknown call legs with status code 481 (Transaction Does Not Exist); they drop ACK requests with unknown call legs silently.

Special considerations apply for choosing forwarding destinations for ACK and BYE requests. In most cases, these requests will bypass proxies and reach the desired party directly, keeping proxies from having to make forwarding decisions.

A proxy MAY maintain call state for a period of its choosing. If a proxy still has list of destinations that it forwarded the last INVITE to, it SHOULD direct ACK requests only to those downstream servers.

# 13 Security Considerations

## 13.1 Confidentiality and Privacy: Encryption

### 13.1.1 End-to-End Encryption

SIP requests and responses can contain sensitive information about the communication patterns and communication content of individuals. The SIP message body MAY also contain encryption keys for the session itself. SIP supports three complementary forms of encryption to protect privacy:

- End-to-end encryption of the SIP message body and certain sensitive header fields;

- o hop-by-hop encryption to prevent eavesdropping that tracks who is calling whom;

- o hop-by-hop encryption of Via fields to hide the route a request has taken.

Not all of the SIP request or response can be encrypted end-to-end because header fields such as To and Via need to be visible to proxies so that the SIP request can be routed correctly. Hop-by-hop encryption encrypts the entire SIP request or response on the wire so that packet sniffers or other eavesdroppers cannot see who is calling whom. Hop-by-hop encryption can also encrypt requests and responses that have been end-to-end encrypted. Note that proxies can still see who is calling whom, and this information is also deducible by performing a network traffic analysis, so this provides a very limited but still worthwhile degree of protection.

SIP Via fields are used to route a response back along the path taken by the request and to prevent infinite request loops. However, the information given by them can also provide useful information to an attacker. Section 6.22 describes how a sender can request that Via fields be encrypted by cooperating proxies without compromising the purpose of the Via field.

End-to-end encryption relies on keys shared by the two user agents involved in the request. Typically, the message is sent encrypted with the public key of the recipient, so that only that recipient can read the message. All implementations SHOULD support PGP-based encryption [33] and MAY implement other schemes.

A SIP request (or response) is end-to-end encrypted by splitting the message to be sent into a part to be encrypted and a short header that will remain in the clear. Some parts of the SIP message, namely the request line, the response line and certain header fields marked with "n" in the "enc." column in Table 4 and 5 need to be read and returned by proxies and thus MUST NOT be encrypted end-to-end. Possibly sensitive information that needs to be made available as plaintext include destination address (To) and the forwarding path (Via) of the call. The Authorization header field MUST remain in the clear if it contains a digital signature as the signature is generated after encryption, but MAY be encrypted if it contains "basic" or "digest" authentication. The From header field SHOULD normally remain in the clear, but MAY be encrypted if required, in which case some proxies MAY return a 401 (Unauthorized) status if they require a From field.

Other header fields MAY be encrypted or MAY travel in the clear as desired by the sender. The Subject, Allow and Content-Type header fields will typically be encrypted. The Accept, Accept-Language, Date, Expires, Priority, Require, Call-ID, Cseq, and Timestamp header fields will remain in the clear.

All fields that will remain in the clear MUST precede those that will be encrypted. The message is encrypted starting with the first character of the first header field that will be encrypted and continuing through to the end of the message body. If no header fields are to be encrypted, encrypting starts with the second CRLF pair after the last header field, as shown below. Carriage return and line feed characters have been made visible as "$", and the encrypted part of the message is outlined.

```
  INVITE sip:watson@boston.bell-telephone.com SIP/2.0$
  Via: SIP/2.0/UDP 169.130.12.5$
  To: T. A. Watson <sip:watson@bell-telephone.com>$
  From: A. Bell <sip:a.g.bell@bell-telephone.com>$
  Encryption: PGP version=5.0$
  Content-Length: 224$
  Call-ID: 187602141351@worcester.bell-telephone.com$
  CSeq: 488$
  $
*******************************************************
* Subject: Mr. Watson, come here.$                    *
* Content-Type: application/sdp$                      *
* $                                                   *
* v=0$                                                *
* o=bell 53655765 2353687637 IN IP4 128.3.4.5$        *
* c=IN IP4 135.180.144.94$                            *
* m=audio 3456 RTP/AVP 0 3 4 5$                       *
*******************************************************
```

An Encryption header field MUST be added to indicate the encryption mechanism used. A Content-Length field is added that indicates the length of the encrypted body. The encrypted body is preceded by a blank line as a normal SIP message body would be.

Upon receipt by the called user agent possessing the correct decryption key, the message body as indicated by the Content-Length field is decrypted, and the now-decrypted body is appended to the clear-text header fields. There is no need for an additional Content-Length header field within the encrypted body because the length of the actual message body is unambiguous after decryption.

Had no SIP header fields required encryption, the message would have been as below. Note that the encrypted body MUST then include a blank line (start with CRLF) to disambiguate between any possible SIP header fields that might have been present and the SIP message body.

```
  INVITE sip:watson@boston.bell-telephone.com SIP/2.0$
  Via: SIP/2.0/UDP 169.130.12.5$
  To: T. A. Watson <sip:watson@bell-telephone.com>$
  From: A. Bell <a.g.bell@bell-telephone.com>$
  Encryption: PGP version=5.0$
  Content-Type: application/sdp$
  Content-Length: 107$
  $
*************************************************
* $                                             *
* v=0$                                          *
* o=bell 53655765 2353687637 IN IP4 128.3.4.5$  *
* c=IN IP4 135.180.144.94$                      *
* m=audio 3456 RTP/AVP 0 3 4 5$                 *
*************************************************
```

### 13.1.2 Privacy of SIP Responses

SIP requests can be sent securely using end-to-end encryption and authentication to a called user agent that sends an insecure response. This is allowed by the SIP security model, but is not a good idea. However, unless the correct behavior is explicit, it would not always be possible for the called user agent to infer what a reasonable behavior was. Thus when end-to-end encryption is used by the request originator, the encryption key to be used for the response SHOULD be specified in the request. If this were not done, it might be possible for the called user agent to incorrectly infer an appropriate key to use in the response. Thus, to prevent key-guessing becoming an acceptable strategy, we specify that a called user agent receiving a request that does not specify a key to be used for the response SHOULD send that response unencrypted.

Any SIP header fields that were encrypted in a request SHOULD also be encrypted in an encrypted response. Contact response fields MAY be encrypted if the information they contain is sensitive, or MAY be left in the clear to permit proxies more scope for localized searches.

### 13.1.3 Encryption by Proxies

Normally, proxies are not allowed to alter end-to-end header fields and message bodies. Proxies MAY, however, encrypt an unsigned request or response with the key of the call recipient.

> Proxies need to encrypt a SIP request if the end system cannot perform encryption or to enforce organizational security policies.

### 13.1.4 Hop-by-Hop Encryption

SIP requests and responses MAY also be protected by security mechanisms at the transport or network layer. No particular mechanism is defined or recommended here. Two possibilities are IPSEC [34] or TLS [35]. The use of a particular mechanism will generally need to be specified out of band, through manual configuration, for example.

### 13.1.5 Via field encryption

When Via header fields are to be hidden, a proxy that receives a request containing an appropriate "Hide: hop" header field (as specified in section 6.22) SHOULD encrypt the header field. As only the proxy that encrypts the field will decrypt it, the algorithm chosen is entirely up to the proxy implementor. Two methods satisfy these requirements:

- o The server keeps a cache of Via header fields and the associated To header field, and replaces the Via header field with an index into the cache. On the reverse path, take the Via header field from the cache rather than the message.

  This is insufficient to prevent message looping, and so an additional ID MUST be added so that the proxy can detect loops. This SHOULD NOT normally be the address of the proxy as the goal is to hide the route, so instead a sufficiently large random number SHOULD be used by the proxy and maintained in the cache.

  It is possible for replies to get directed to the wrong originator if the cache entry gets reused, so great care needs to be taken to ensure this does not happen.

- o The server MAY use a secret key to encrypt the Via field, a timestamp and an appropriate checksum in any such message with the same secret key. The checksum is needed to detect whether successful decoding has occurred, and the timestamp is required to prevent possible replay attacks and to ensure that no two requests from the same previous hop have the same encrypted Via field. This is the preferred solution.

## 13.2 Message Integrity and Access Control: Authentication

Protective measures need to be taken to prevent an active attacker from modifying and replaying SIP requests and responses. The same cryptographic measures that are used to ensure the authenticity of the SIP message also serve to authenticate the originator of the message. However, the "basic" and "digest" authentication mechanism offer authentication only, without message integrity.

Transport-layer or network-layer authentication MAY be used for hop-by-hop authentication. SIP also extends the HTTP WWW-Authenticate (Section 6.42) and Authorization (Section 6.11) header field and their Proxy counterparts to include cryptographically strong signatures. SIP also supports the HTTP "basic" and "digest" schemes (see Section 14) and other HTTP authentication schemes to be defined that offer a rudimentary mechanism of ascertaining the identity of the caller.

> Since SIP requests are often sent to parties with which no prior communication relationship has existed, we do not specify authentication based on shared secrets.

SIP requests MAY be authenticated using the Authorization header field to include a digital signature of certain header fields, the request method and version number and the payload, none of which are modified between client and called user agent. The Authorization header field is used in requests to authenticate the request originator end-to-end to proxies and the called user agent, and in responses to authenticate the called user agent or proxies returning their own failure codes. If required, hop-by-hop authentication can be provided, for example, by the IPSEC Authentication Header.

SIP does not dictate which digital signature scheme is used for authentication, but does define how to provide authentication using PGP in Section 15. As indicated above, SIP implementations MAY also use "basic" and "digest" authentication and other authentication mechanisms defined for HTTP. Note that "basic" authentication has severe security limitations. The following does not apply to these schemes.

To cryptographically sign a SIP request, the order of the SIP header fields is important. When an Authorization header field is present, it indicates that all header fields following the Authorization header field have been included in the signature. Therefore, hop-by-hop header fields which MUST or SHOULD be modified by proxies MUST precede the Authorization header field as they will generally be modified or added-to by proxy servers. Hop-by-hop header fields which MAY be modified by a proxy MAY appear before or after the Authorization header. When they appear before, they MAY be modified by a proxy. When they appear after, they MUST NOT be modified by a proxy. To sign a request, a client constructs a message from the request method (in upper case) followed, without LWS, by the SIP version number, followed, again without LWS, by the request headers to be signed and the message body. The message thus constructed is then signed.

For example, if the SIP request is to be:

```
INVITE sip:watson@boston.bell-telephone.com SIP/2.0
Via: SIP/2.0/UDP 169.130.12.5
Authorization: PGP version=5.0, signature=...
From: A. Bell <sip:a.g.bell@bell-telephone.com>
To: T. A. Watson <sip:watson@bell-telephone.com>
Call-ID: 187602141351@worcester.bell-telephone.com
Subject: Mr. Watson, come here.
Content-Type: application/sdp
Content-Length: ...

v=0
o=bell 53655765 2353687637 IN IP4 128.3.4.5
c=IN IP4 135.180.144.94
m=audio 3456 RTP/AVP 0 3 4 5
```

Then the data block that is signed is:

```
INVITESIP/2.0From: A. Bell <sip:a.g.bell@bell-telephone.com>
To: T. A. Watson <sip:watson@bell-telephone.com>
Call-ID: 187602141351@worcester.bell-telephone.com
Subject: Mr. Watson, come here.
Content-Type: application/sdp
Content-Length: ...

v=0
o=bell 53655765 2353687637 IN IP4 128.3.4.5
c=IN IP4 135.180.144.94
m=audio 3456 RTP/AVP 0 3 4 5
```

Clients wishing to authenticate requests MUST construct the portion of the message below the Authorization header using a canonical form. This allows a proxy to parse the message, take it apart, and reconstruct it, without causing an authentication failure due to extra white space, for example. Canonical form consists of the following rules:

- o No short form header fields
- o Header field names are capitalized as shown in this document
- o No white space between the header name and the colon
- o A single space after the colon
- o Line termination with a CRLF
- o No line folding
- o No comma separated lists of header values; each must appear as a separate header
- o Only a single SP between tokens, between tokens and quoted strings, and between quoted strings; no SP after last token or quoted string
- o No LWS between tokens and separators, except as described above for after the colon in header fields

Note that if a message is encrypted and authenticated using a digital signature, when the message is generated encryption is performed before the digital signature is generated. On receipt, the digital signature is checked before decryption.

A client MAY require that a server sign its response by including a Require: org.ietf.sip.signed-response request header field. The client indicates the desired authentication method via the WWW-Authenticate header.

The correct behavior in handling unauthenticated responses to a request that requires authenticated responses is described in section 13.2.1.

### 13.2.1 Trusting responses

There is the possibility that an eavesdropper listens to requests and then injects unauthenticated responses that terminate, redirect or otherwise interfere with a call. (Even encrypted requests contain enough information to fake a response.)

Clients need to be particularly careful with 3xx redirection responses. Thus a client receiving, for example, a 301 (Moved Permanently) which was not authenticated when the public key of the called user agent is known to the client, and authentication was requested in the request SHOULD be treated as suspicious. The correct behavior in such a case would be for the called-user to form a dated response containing the Contact field to be used, to sign it, and give this signed stub response to the proxy that will provide the redirection. Thus the response can be authenticated correctly. A client SHOULD NOT automatically redirect such a request to the new location without alerting the user to the authentication failure before doing so.

Another problem might be responses such as 6xx failure responses which would simply terminate a search, or "4xx" and "5xx" response failures.

If TCP is being used, a proxy SHOULD treat 4xx and 5xx responses as valid, as they will not terminate a search. However, fake 6xx responses from a rogue proxy terminate a search incorrectly. 6xx responses SHOULD be authenticated if requested by the client, and failure to do so SHOULD cause such a client to ignore the 6xx response and continue a search.

With UDP, the same problem with 6xx responses exists, but also an active eavesdropper can generate 4xx and 5xx responses that might cause a proxy or client to believe a failure occurred when in fact it did not. Typically 4xx and 5xx responses will not be signed by the called user agent, and so there is no simple way to detect these rogue responses. This problem is best prevented by using hop-by-hop encryption of the SIP request, which removes any additional problems that UDP might have over TCP.

These attacks are prevented by having the client require response authentication and dropping unauthenticated responses. A server user agent that cannot perform response authentication responds using the normal Require response of 420 (Bad Extension).

### 13.3 Callee Privacy

User location and SIP-initiated calls can violate a callee's privacy. An implementation SHOULD be able to restrict, on a per-user basis, what kind of location and availability information is given out to certain classes of callers.

### 13.4 Known Security Problems

With either TCP or UDP, a denial of service attack exists by a rogue proxy sending 6xx responses. Although a client SHOULD choose to ignore such responses if it requested authentication, a proxy cannot do so. It is obliged to forward the 6xx response back to the client. The client can then ignore the response, but if it repeats the request it will probably reach the same rogue proxy again, and the process will repeat.

## 14 SIP Authentication using HTTP Basic and Digest Schemes

SIP implementations MAY use HTTP's basic and digest authentication mechanisms to provide a rudimentary form of security. This section overviews usage of these mechanisms in SIP. The basic operation is almost completely identical to that for HTTP [36]. This section outlines this operation, pointing to [36] for details, and noting the differences when used in SIP.

### 14.1 Framework

The framework for SIP authentication parallels that for HTTP [36]. In particular, the BNF for auth-scheme, auth-param, challenge, realm, realm-value, and credentials is identical. The 401 response is used by user agent servers in SIP to challenge the authorization of a user agent client. Additionally, registrars and redirect servers MAY make use of 401 responses for authorization, but proxies MUST NOT, and instead MAY use the 407 response. The requirements for inclusion of the Proxy-Authenticate, Proxy-Authorization, WWW-Authenticate, and Authorization in the various messages is identical to [36].

Since SIP does not have the concept of a canonical root URL, the notion of protections spaces are interpreted differently for SIP. The realm is a protection domain for all SIP URIs with the same value for the userinfo, host and port part of the SIP Request-URI. For example:

```
INVITE sip:alice.wonderland@example.com SIP/2.0
WWW-Authenticate:  Basic realm="business"
```

and

```
INVITE sip:aw@example.com SIP/2.0
WWW-Authenticate: Basic realm="business"
```

define different protection realms according to this rule.

When a UAC resubmits a request with its credentials after receiving a 401 or 407 response, it MUST increment the CSeq header field as it would normally do when sending an updated request.

## 14.2 Basic Authentication

The rules for basic authentication follow those defined in [36], but with the words "origin server" replaced with "user agent server, redirect server , or registrar".

Since SIP URIs are not hierarchical, the paragraph in [36] that states that "all paths at or deeper than the depth of the last symbolic element in the path field of the Request-URI also are within the protection space specified by the Basic realm value of the current challenge" does not apply for SIP. SIP clients MAY preemptively send the corresponding Authorization header with requests for SIP URIs within the same protection realm (as defined above) without receipt of another challenge from the server.

## 14.3 Digest Authentication

The rules for digest authentication follow those defined in [36], with "HTTP 1.1" replaced by "SIP/2.0" in addition to the following differences:

1. The URI included in the challenge has the following BNF:

   ```
   URI  =  SIP-URL
   ```

2. The BNF for digest-uri-value is:

   ```
   digest-uri-value  =  Request-URI ; a defined in Section
   4.3
   ```

3. The example procedure for choosing a nonce based on Etag does not work for SIP.

4. The Authentication-Info and Proxy-Authentication-Info fields are not used in SIP.

5. The text in [36] regarding cache operation does not apply to SIP.

6. [36] requires that a server check that the URI in the request line, and the URI included in the Authorization header, point to the same resource. In a SIP context, these two URI's may actually refer to different users, due to forwarding at some proxy. Therefore, in SIP, a server MAY check that the request-uri in the Authorization header corresponds to a user that the server is willing to accept forwarded or direct calls for.

## 14.4 Proxy-Authentication

The use of the Proxy-Authentication and Proxy-Authorization parallel that as described in [36], with one difference. Proxies MUST NOT add the Proxy-Authorization header. 407 responses MUST be forwarded upstream towards the client following the procedures for any other response. It is the client's responsibility to add the Proxy-Authorization header containing credentials for the proxy which has asked for authentication.

> If a proxy were to resubmit a request with a Proxy-Authorization header field, it would need to increment the CSeq in the new request. However, this would mean that the UAC which submitted the original request would discard a response from the UAS, as the CSeq value would be different.

See sections 6.26 and 6.27 for additional information on usage of these fields as they apply to SIP.

# 15 SIP Security Using PGP

## 15.1 PGP Authentication Scheme

The "pgp" authentication scheme is based on the model that the client authenticates itself with a request signed with the client's private key. The server can then ascertain the origin of the request if it has access to the public key, preferably signed by a trusted third party.

### 15.1.1 The WWW-Authenticate Response Header

```
WWW-Authenticate =  "WWW-Authenticate" ":" "pgp" pgp-challenge
pgp-challenge    =  * (";" pgp-params )
pgp-params       =  realm | pgp-version | pgp-algorithm | nonce
realm            =  "realm" "=" realm-value
realm-value      =  quoted-string
pgp-version      =  "version" "="
                     <"> digit *( "." digit ) *letter <">
pgp-algorithm    =  "algorithm" "=" ( "md5" | "sha1" | token )
nonce            =  "nonce" "=" nonce-value
nonce-value      =  quoted-string
```

The meanings of the values of the parameters used above are as follows:

realm: A string to be displayed to users so they know which identity to use. This string SHOULD contain at least the name of the host performing the authentication and MAY additionally indicate the collection of users who might have access. An example might be " Users with call-out privileges ".

pgp-algorithm: The value of this parameter indicates the PGP message integrity check (MIC) to be used to produce the signature. If this not present it is assumed to be "md5". The currently defined values are "md5" for the MD5 checksum, and "sha1" for the SHA.1 algorithm.

pgp-version: The version of PGP that the client MUST use. Common values are "2.6.2" and "5.0". The default is 5.0.

nonce: A server-specified data string which should be uniquely generated each time a 401 response is made. It is RECOMMENDED that this string be base64 or hexadecimal data. Specifically, since the string is passed in the header lines as a quoted string, the double-quote character is not allowed. The contents of the nonce are implementation dependent. The quality of the implementation depends on a good choice. Since the nonce is used only to prevent replay attacks and is signed, a time stamp in units convenient to the server is sufficient.

Replay attacks within the duration of the call setup are of limited interest, so that timestamps with a resolution of a few seconds are often should be sufficient. In that case, the server does not have to keep a record of the nonces.

Example:

```
WWW-Authenticate: pgp ;version="5.0"
  ;realm="Your Startrek identity, please" ;algorithm=md5
  ;nonce="913082051"
```

### 15.1.2 The Authorization Request Header

The client is expected to retry the request, passing an Authorization header line, which is defined as follows.

```
Authorization  =  "Authorization" ":" "pgp" *( ";" pgp-response )
pgp-response   =  realm | pgp-version | pgp-signature
                  | signed-by | nonce
pgp-signature  =  "signature" "=" quoted-string
signed-by      =  "signed-by" "=" <"> URI <">
```

The client MUST increment the CSeq header before resubmitting the request. The signature MUST correspond to the From header of the request unless the signed-by parameter is provided.

pgp-signature: The PGP ASCII-armored signature [33], as it appears between the "BEGIN PGP MESSAGE" and "END PGP MESSAGE" delimiters, without the version indication. The signature is included without any linebreaks.

The signature is computed across the nonce (if present), request method, request version and header fields following the Authorization header and the message body, in the same order as they appear in the message. The request method and version are prepended to the header fields without any white space. The signature is computed across the headers as sent, and the terminating CRLF. The CRLF following the Authorization header is NOT included in the signature.

A server MAY be configured not to generate nonces only if replay attacks are not a concern.

Not generating nonces avoids the additional set of request, 401 response and possibly ACK messages and reduces delay by one round-trip time.

Using the ASCII-armored version is about 25% less space-efficient than including the binary signature, but it is significantly easier for the receiver to piece together. Versions of the PGP program always include the full (compressed) signed text in their output unless ASCII-armored mode ( -sta ) is specified. Typical signatures are about 200 bytes long. -- The PGP signature mechanism allows the client to simply pass the request to an external PGP program. This relies on the requirement that proxy servers are not allowed to reorder or change header fields.

realm: The realm is copied from the corresponding WWW-Authenticate header field parameter.

signed-by: If and only if the request was not signed by the entity listed in the From header, the signed-by header indicates the name of the signing entity, expressed as a URI.

Receivers of signed SIP messages SHOULD discard any end-to-end header fields above the Authorization header, as they may have been maliciously added en route by a proxy.

Example:

```
Authorization: pgp version="5.0"
  ;realm="Your Startrek identity, please"
  ;nonce="913082051"
  ;signature="iQB1AwUBNNJiUaYBnHmiiQh1AQFYsgL/Wt3dk6TWK81/b0gcNDf
  VAUGU4rhEBW972IPxFSOZ94L1qhCLInTPaqhHFw1cb3lB01rA0RhpV4t5yCdUt
  SRYBSkOK29o5e1KlFeW23EzYPVUm2TlDAhbcjbMdfC+KLFX
  =aIrx"
```

## 15.2 PGP Encryption Scheme

The PGP encryption scheme uses the following syntax:

```
Encryption    =  "Encryption" ":" "pgp" pgp-eparams
pgp-eparams   =  1# ( pgp-version | pgp-encoding )
pgp-encoding  =  "encoding" "=" "ascii" | token
```

encoding: Describes the encoding or "armor" used by PGP. The value "ascii" refers to the standard PGP ASCII armor, without the lines containing "BEGIN PGP MESSAGE" and "END PGP MESSAGE" and without the version identifier. By default, the encrypted part is included as binary.

Example:

```
Encryption: pgp version="2.6.2", encoding="ascii"
```

## 15.3 Response-Key Header Field for PGP

```
     Response-Key  =  "Response-Key" ":" "pgp" pgp-eparams
     pgp-eparams   =  1# ( pgp-version | pgp-encoding | pgp-key)
     pgp-key       =  "key" "=" quoted-string
```

If ASCII encoding has been requested via the encoding parameter, the key parameter contains the user's public key as extracted from the pgp key ring with the "pgp -kxa user ".

Example:

```
Response-Key: pgp version="2.6.2", encoding="ascii",
  key="mQBtAzNWHNYAAAEDAL7QvAdK2utY05wuUG+ItYK5tCF8HNJM60sU4rLaV+eUnkMk
  mOmJWtc2wXcZx1XaXb2lkydTQOesrUR75IwNXBuZXPEIMThEa5WLsT7VLme7njnx
  sE86SgWmAZx5ookIdQAFEbQxSGVubmluZyBTY2h1bHpyaW5uZSA8c2NodWx6cmlu
  bmVAY3MuY29sdW1iaWEuZWR1Pg==
  =+y19"
```

# 16 Examples

In the following examples, we often omit the message body and the corresponding Content-Length and Content-Type headers for brevity.

## 16.1 Registration

A user at host saturn.bell-tel.com registers on start-up, via multicast, with the local SIP server named bell-tel.com. In the example, the user agent on saturn expects to receive SIP requests on UDP port 3890.

```
C->S: REGISTER sip:bell-tel.com SIP/2.0
      Via: SIP/2.0/UDP saturn.bell-tel.com
      From: sip:watson@bell-tel.com
      To: sip:watson@bell-tel.com
      Call-ID: 70710@saturn.bell-tel.com
      CSeq: 1 REGISTER
      Contact: <sip:watson@saturn.bell-tel.com:3890;transport=udp>
      Expires: 7200
```

The registration expires after two hours. Any future invitations for watson@bell-tel.com arriving at sip.bell-tel.com will now be redirected to watson@saturn.bell-tel.com, UDP port 3890.

If Watson wants to be reached elsewhere, say, an on-line service he uses while traveling, he updates his reservation after first cancelling any existing locations:

```
C->S: REGISTER sip:bell-tel.com SIP/2.0
      Via: SIP/2.0/UDP saturn.bell-tel.com
      From: sip:watson@bell-tel.com
      To: sip:watson@bell-tel.com
      Call-ID: 70710@saturn.bell-tel.com
      CSeq: 2 REGISTER
      Contact: *
      Expires: 0

C->S: REGISTER sip:bell-tel.com SIP/2.0
      Via: SIP/2.0/UDP saturn.bell-tel.com
      From: sip:watson@bell-tel.com
      To: sip:watson@bell-tel.com
      Call-ID: 70710@saturn.bell-tel.com
      CSeq: 3 REGISTER
      Contact: sip:tawatson@example.com
```

Now, the server will forward any request for Watson to the server at example.com, using the Request-URI tawatson@example.com. For the server at example.com to reach Watson, he will need to send a REGISTER there, or inform the server of his current location through some other means.

It is possible to use third-party registration. Here, the secretary jon.diligent registers his boss, T. Watson:

```
C->S: REGISTER sip:bell-tel.com SIP/2.0
      Via: SIP/2.0/UDP pluto.bell-tel.com
      From: sip:jon.diligent@bell-tel.com
      To: sip:watson@bell-tel.com
      Call-ID: 17320@pluto.bell-tel.com
      CSeq: 1 REGISTER
      Contact: sip:tawatson@example.com
```

The request could be sent to either the registrar at bell-tel.com or the server at example.com. In the latter case, the server at example.com would proxy the request to the address indicated in the Request-URI. Then, Max-Forwards header could be used to restrict the registration to that server.

### 16.2 Invitation to a Multicast Conference

The first example invites schooler@vlsi.cs.caltech.edu to a multicast session. All examples use the Session Description Protocol (SDP) (RFC 2327 [6]) as the session description format.

#### 16.2.1 Request

```
C->S: INVITE sip:schooler@cs.caltech.edu SIP/2.0
      Via: SIP/2.0/UDP csvax.cs.caltech.edu;branch=8348
        ;maddr=239.128.16.254;ttl=16
      Via: SIP/2.0/UDP north.east.isi.edu
      From: Mark Handley <sip:mjh@isi.edu>
      To: Eve Schooler <sip:schooler@caltech.edu>
      Call-ID: 2963313058@north.east.isi.edu
      CSeq: 1 INVITE
      Subject: SIP will be discussed, too
      Content-Type: application/sdp
      Content-Length: 187

      v=0
      o=user1 53655765 2353687637 IN IP4 128.3.4.5
      s=Mbone Audio
      i=Discussion of Mbone Engineering Issues
      e=mbone@somewhere.com
      c=IN IP4 224.2.0.1/127
      t=0 0
      m=audio 3456 RTP/AVP 0
```

The From request header above states that the request was initiated by mjh@isi.edu and addressed to schooler@caltech.edu (From header fields). The Via fields list the hosts along the path from invitation initiator (the last element of the list) towards the callee. In the example above, the message was last multicast to the administratively scoped group 239.128.16.254 with a ttl of 16 from the host csvax.cs.caltech.edu. The second Via header field indicates that it was originally sent from the host north.east.isi.edu. The Request-URI indicates that the request is currently being being addressed to schooler@cs.caltech.edu, the local address that csvax looked up for the callee.

In this case, the session description is using the Session Description Protocol (SDP), as stated in the Content-Type header.

The header is terminated by an empty line and is followed by a message body containing the session description.

### 16.2.2 Response

The called user agent, directly or indirectly through proxy servers, indicates that it is alerting ("ringing") the called party:

```
S->C: SIP/2.0 180 Ringing
      Via: SIP/2.0/UDP csvax.cs.caltech.edu;branch=8348
        ;maddr=239.128.16.254;ttl=16
      Via: SIP/2.0/UDP north.east.isi.edu
      From: Mark Handley <sip:mjh@isi.edu>
      To: Eve Schooler <sip:schooler@caltech.edu> ;tag=9883472
      Call-ID: 2963313058@north.east.isi.edu
      CSeq: 1 INVITE
```

A sample response to the invitation is given below. The first line of the response states the SIP version number, that it is a 200 (OK) response, which means the request was successful. The Via headers are taken from the request, and entries are removed hop by hop as the response retraces the path of the request. A new authentication field MAY be added by the invited user's agent if required. The Call-ID is taken directly from the original request, along with the remaining fields of the request message. The original sense of From field is preserved (i.e., it is the session initiator).

In addition, the Contact header gives details of the host where the user was located, or alternatively the relevant proxy contact point which should be reachable from the caller's host.

```
S->C: SIP/2.0 200 OK
      Via: SIP/2.0/UDP csvax.cs.caltech.edu;branch=8348
        ;maddr=239.128.16.254;ttl=16
      Via: SIP/2.0/UDP north.east.isi.edu
      From: Mark Handley <sip:mjh@isi.edu>
      To: Eve Schooler <sip:schooler@caltech.edu> ;tag=9883472
      Call-ID: 2963313058@north.east.isi.edu
      CSeq: 1 INVITE
      Contact: sip:es@jove.cs.caltech.edu
```

The caller confirms the invitation by sending an ACK request to the location named in the Contact header:

```
C->S: ACK sip:es@jove.cs.caltech.edu SIP/2.0
      Via: SIP/2.0/UDP north.east.isi.edu
      From: Mark Handley <sip:mjh@isi.edu>
      To: Eve Schooler <sip:schooler@caltech.edu> ;tag=9883472
      Call-ID: 2963313058@north.east.isi.edu
      CSeq: 1 ACK
```

## 16.3 Two-party Call

For two-party Internet phone calls, the response must contain a description of where to send the data. In the example below, Bell calls Watson. Bell indicates that he can receive RTP audio codings 0 (PCMU), 3 (GSM), 4 (G.723) and 5 (DVI4).

```
C->S: INVITE sip:watson@boston.bell-tel.com SIP/2.0
      Via: SIP/2.0/UDP kton.bell-tel.com
      From: A. Bell <sip:a.g.bell@bell-tel.com>
      To: T. Watson <sip:watson@bell-tel.com>
      Call-ID: 3298420296@kton.bell-tel.com
      CSeq: 1 INVITE
      Subject: Mr. Watson, come here.
      Content-Type: application/sdp
      Content-Length: ...

      v=0
      o=bell 53655765 2353687637 IN IP4 128.3.4.5
      s=Mr. Watson, come here.
      c=IN IP4 kton.bell-tel.com
      m=audio 3456 RTP/AVP 0 3 4 5
```

```
S->C: SIP/2.0 100 Trying
      Via: SIP/2.0/UDP kton.bell-tel.com
      From: A. Bell <sip:a.g.bell@bell-tel.com>
      To: T. Watson <sip:watson@bell-tel.com> ;tag=37462311
      Call-ID: 3298420296@kton.bell-tel.com
      CSeq: 1 INVITE
      Content-Length: 0

S->C: SIP/2.0 180 Ringing
      Via: SIP/2.0/UDP kton.bell-tel.com
      From: A. Bell <sip:a.g.bell@bell-tel.com>
      To: T. Watson <sip:watson@bell-tel.com> ;tag=37462311
      Call-ID: 3298420296@kton.bell-tel.com
      CSeq: 1 INVITE
      Content-Length: 0

S->C: SIP/2.0 182 Queued, 2 callers ahead
      Via: SIP/2.0/UDP kton.bell-tel.com
      From: A. Bell <sip:a.g.bell@bell-tel.com>
      To: T. Watson <sip:watson@bell-tel.com> ;tag=37462311
      Call-ID: 3298420296@kton.bell-tel.com
      CSeq: 1 INVITE
      Content-Length: 0

S->C: SIP/2.0 182 Queued, 1 caller ahead
      Via: SIP/2.0/UDP kton.bell-tel.com
      From: A. Bell <sip:a.g.bell@bell-tel.com>
      To: T. Watson <sip:watson@bell-tel.com> ;tag=37462311
      Call-ID: 3298420296@kton.bell-tel.com
      CSeq: 1 INVITE
      Content-Length: 0

S->C: SIP/2.0 200 OK
      Via: SIP/2.0/UDP kton.bell-tel.com
      From: A. Bell <sip:a.g.bell@bell-tel.com>
      To: <sip:watson@bell-tel.com> ;tag=37462311
      Call-ID: 3298420296@kton.bell-tel.com
      CSeq: 1 INVITE
      Contact: sip:watson@boston.bell-tel.com
      Content-Type: application/sdp
      Content-Length: ...

      v=0
      o=watson 4858949 4858949 IN IP4 192.1.2.3
      s=I’m on my way
      c=IN IP4 boston.bell-tel.com
      m=audio 5004 RTP/AVP 0 3
```

The example illustrates the use of informational status responses. Here, the reception of the call is confirmed immediately (100), then, possibly after some database mapping delay, the call rings (180) and is then queued, with periodic status updates.

Watson can only receive PCMU and GSM. Note that Watson's list of codecs may or may not be a subset of the one offered by Bell, as each party indicates the data types it is willing to receive. Watson will send audio data to port 3456 at c.bell-tel.com, Bell will send to port 5004 at boston.bell-tel.com.

By default, the media session is one RTP session. Watson will receive RTCP packets on port 5005, while Bell will receive them on port 3457.

Since the two sides have agreed on the set of media, Bell confirms the call without enclosing another session description:

```
C->S: ACK sip:watson@boston.bell-tel.com SIP/2.0
      Via: SIP/2.0/UDP kton.bell-tel.com
      From: A. Bell <sip:a.g.bell@bell-tel.com>
      To: T. Watson <sip:watson@bell-tel.com> ;tag=37462311
      Call-ID: 3298420296@kton.bell-tel.com
      CSeq: 1 ACK
```

## 16.4 Terminating a Call

To terminate a call, caller or callee can send a BYE request:

```
C->S: BYE sip:watson@boston.bell-tel.com SIP/2.0
      Via: SIP/2.0/UDP kton.bell-tel.com
      From: A. Bell <sip:a.g.bell@bell-tel.com>
      To: T. A. Watson <sip:watson@bell-tel.com> ;tag=37462311
      Call-ID: 3298420296@kton.bell-tel.com
      CSeq: 2 BYE
```

If the callee wants to abort the call, it simply reverses the To and From fields. Note that it is unlikely that a BYE from the callee will traverse the same proxies as the original INVITE.

## 16.5 Forking Proxy

In this example, Bell (a.g.bell@bell-tel.com) (C), currently seated at host c.bell-tel.com wants to call Watson (t.watson@ieee.org). At the time of the call, Watson is logged in at two workstations, t.watson@x.bell-tel.com (X) and watson@y.bell-tel.com (Y), and has registered with the IEEE proxy server (P) called sip.ieee.org. The IEEE server also has a registration for the home machine of Watson, at watson@h.bell-tel.com (H), as well as a permanent registration at watson@acm.org (A). For brevity, the examples omit the session description and Via header fields.

Bell's user agent sends the invitation to the SIP server for the ieee.org domain:

```
C->P: INVITE sip:t.watson@ieee.org SIP/2.0
      Via:     SIP/2.0/UDP c.bell-tel.com
      From:    A. Bell <sip:a.g.bell@bell-tel.com>
      To:      T. Watson <sip:t.watson@ieee.org>
      Call-ID: 31415@c.bell-tel.com
      CSeq:    1 INVITE
```

The SIP server at ieee.org tries the four addresses in parallel. It sends the following message to the home machine:

```
P->H: INVITE sip:watson@h.bell-tel.com SIP/2.0
      Via:     SIP/2.0/UDP sip.ieee.org ;branch=1
      Via:     SIP/2.0/UDP c.bell-tel.com
      From:    A. Bell <sip:a.g.bell@bell-tel.com>
      To:      T. Watson <sip:t.watson@ieee.org>
      Call-ID: 31415@c.bell-tel.com
      CSeq:    1 INVITE
```

This request immediately yields a 404 (Not Found) response, since Watson is not currently logged in at home:

```
H->P: SIP/2.0 404 Not Found
      Via:     SIP/2.0/UDP sip.ieee.org ;branch=1
      Via:     SIP/2.0/UDP c.bell-tel.com
      From:    A. Bell <sip:a.g.bell@bell-tel.com>
      To:      T. Watson <sip:t.watson@ieee.org>;tag=87454273
```

```
        Call-ID: 31415@c.bell-tel.com
        CSeq:    1 INVITE
```

The proxy ACKs the response so that host H can stop retransmitting it:

```
P->H: ACK sip:watson@h.bell-tel.com SIP/2.0
      Via:     SIP/2.0/UDP sip.ieee.org ;branch=1
      From:    A. Bell <sip:a.g.bell@bell-tel.com>
      To:      T. Watson <sip:t.watson@ieee.org>;tag=87454273
      Call-ID: 31415@c.bell-tel.com
      CSeq:    1 ACK
```

Also, P attempts to reach Watson through the ACM server:

```
P->A: INVITE sip:watson@acm.org SIP/2.0
      Via:     SIP/2.0/UDP sip.ieee.org ;branch=2
      Via:     SIP/2.0/UDP c.bell-tel.com
      From:    A. Bell <sip:a.g.bell@bell-tel.com>
      To:      T. Watson <sip:t.watson@ieee.org>
      Call-ID: 31415@c.bell-tel.com
      CSeq:    1 INVITE
```

In parallel, the next attempt proceeds, with an INVITE to X and Y:

```
P->X: INVITE sip:t.watson@x.bell-tel.com SIP/2.0
      Via:     SIP/2.0/UDP sip.ieee.org ;branch=3
      Via:     SIP/2.0/UDP c.bell-tel.com
      From:    A. Bell <sip:a.g.bell@bell-tel.com>
      To:      T. Watson <sip:t.watson@ieee.org>
      Call-ID: 31415@c.bell-tel.com
      CSeq:    1 INVITE

P->Y: INVITE sip:watson@y.bell-tel.com SIP/2.0
      Via:     SIP/2.0/UDP sip.ieee.org ;branch=4
      Via:     SIP/2.0/UDP c.bell-tel.com
      From:    A. Bell <sip:a.g.bell@bell-tel.com>
      To:      T. Watson <sip:t.watson@ieee.org>
      Call-ID: 31415@c.bell-tel.com
      CSeq:    1 INVITE
```

As it happens, both Watson at X and a colleague in the other lab at host Y hear the phones ringing and pick up. Both X and Y return 200s via the proxy to Bell.

```
X->P: SIP/2.0 200 OK
      Via:      SIP/2.0/UDP sip.ieee.org ;branch=3
      Via:      SIP/2.0/UDP c.bell-tel.com
      From:     A. Bell <sip:a.g.bell@bell-tel.com>
      To:       T. Watson <sip:t.watson@ieee.org> ;tag=192137601
      Call-ID:  31415@c.bell-tel.com
      CSeq:     1 INVITE
      Contact:  sip:t.watson@x.bell-tel.com

Y->P: SIP/2.0 200 OK
      Via:      SIP/2.0/UDP sip.ieee.org ;branch=4
      Via:      SIP/2.0/UDP c.bell-tel.com
      Contact:  sip:t.watson@y.bell-tel.com
      From:     A. Bell <sip:a.g.bell@bell-tel.com>
      To:       T. Watson <sip:t.watson@ieee.org> ;tag=35253448
      Call-ID:  31415@c.bell-tel.com
      CSeq:     1 INVITE
```

Both responses are forwarded to Bell, using the Via information. At this point, the ACM server is still searching its database. P can now cancel this attempt:

```
P->A: CANCEL sip:watson@acm.org SIP/2.0
      Via:     SIP/2.0/UDP sip.ieee.org ;branch=2
      From:    A. Bell <sip:a.g.bell@bell-tel.com>
      To:      T. Watson <sip:t.watson@ieee.org>
      Call-ID: 31415@c.bell-tel.com
      CSeq:    1 CANCEL
```

The ACM server gladly stops its neural-network database search and responds with a 200. The 200 will not travel any further, since P is the last Via stop.

```
A->P: SIP/2.0 200 OK
      Via:     SIP/2.0/UDP sip.ieee.org ;branch=2
      From:    A. Bell <sip:a.g.bell@bell-tel.com>
      To:      T. Watson <sip:t.watson@ieee.org>
```

```
        Call-ID: 31415@c.bell-tel.com
        CSeq:    1 CANCEL
```

Bell gets the two 200 responses from X and Y in short order. Bell's reaction now depends on his software. He can either send an ACK to both if human intelligence is needed to determine who he wants to talk to or he can automatically reject one of the two calls. Here, he acknowledges both, separately and directly to the final destination:

```
C->X: ACK sip:t.watson@x.bell-tel.com SIP/2.0
      Via:      SIP/2.0/UDP c.bell-tel.com
      From:     A. Bell <sip:a.g.bell@bell-tel.com>
      To:       T. Watson <sip:t.watson@ieee.org>;tag=192137601
      Call-ID:  31415@c.bell-tel.com
      CSeq:     1 ACK

C->Y: ACK sip:watson@y.bell-tel.com SIP/2.0
      Via:      SIP/2.0/UDP c.bell-tel.com
      From:     A. Bell <sip:a.g.bell@bell-tel.com>
      To:       T. Watson <sip:t.watson@ieee.org>;tag=35253448
      Call-ID:  31415@c.bell-tel.com
      CSeq:     1 ACK
```

After a brief discussion between Bell with X and Y, it becomes clear that Watson is at X. (Note that this is not a three-way call; only Bell can talk to X and Y, but X and Y cannot talk to each other.) Thus, Bell sends a BYE to Y, which is replied to:

```
C->Y: BYE sip:watson@y.bell-tel.com SIP/2.0
      Via:      SIP/2.0/UDP c.bell-tel.com
      From:     A. Bell <sip:a.g.bell@bell-tel.com>
      To:       T. Watson <sip:t.watson@ieee.org>;tag=35253448
      Call-ID:  31415@c.bell-tel.com
      CSeq:     2 BYE

Y->C: SIP/2.0 200 OK
      Via:      SIP/2.0/UDP c.bell-tel.com
      From:     A. Bell <sip:a.g.bell@bell-tel.com>
      To:       T. Watson <sip:t.watson@ieee.org>;tag=35253448
      Call-ID:  31415@c.bell-tel.com
      CSeq:     2 BYE
```

## 16.6 Redirects

Replies with status codes 301 (Moved Permanently) or 302 (Moved Temporarily) specify another location using the Contact field. Continuing our earlier example, the server P at ieee.org decides to redirect rather than proxy the request:

```
P->C: SIP/2.0 302 Moved temporarily
      Via:     SIP/2.0/UDP c.bell-tel.com
      From:    A. Bell <sip:a.g.bell@bell-tel.com>
      To:      T. Watson <sip:t.watson@ieee.org>;tag=72538263
      Call-ID: 31415@c.bell-tel.com
      CSeq:    1 INVITE
      Contact: sip:watson@h.bell-tel.com,
                sip:watson@acm.org, sip:t.watson@x.bell-tel.com,
                sip:watson@y.bell-tel.com
      CSeq: 1 INVITE
```

As another example, assume Alice (A) wants to delegate her calls to Bob (B) while she is on vacation until July 29th, 1998. Any calls meant for her will reach Bob with Alice’s To field, indicating to him what role he is to play. Charlie (C) calls Alice (A), whose server returns:

```
A->C: SIP/2.0 302 Moved temporarily
      From: Charlie <sip:charlie@caller.com>
      To: Alice <sip:alice@anywhere.com> ;tag=2332462
      Call-ID: 27182@caller.com
      Contact: sip:bob@anywhere.com
      Expires: Wed, 29 Jul 1998 9:00:00 GMT
      CSeq: 1 INVITE
```

Charlie then sends the following request to the SIP server of the anywhere.com domain. Note that the server at anywhere.com forwards the request to Bob based on the Request-URI.

```
C->B: INVITE sip:bob@anywhere.com SIP/2.0
      From: sip:charlie@caller.com
      To: sip:alice@anywhere.com
      Call-ID: 27182@caller.com
      CSeq: 2 INVITE
```

In the third redirection example, we assume that all outgoing requests are directed through a local firewall F at caller.com, with Charlie again inviting Alice:

```
C->F: INVITE sip:alice@anywhere.com SIP/2.0
      From: sip:charlie@caller.com
      To: Alice <sip:alice@anywhere.com>
      Call-ID: 27182@caller.com
      CSeq: 1 INVITE
```

The local firewall at caller.com happens to be overloaded and thus redirects the call from Charlie to a secondary server S:

```
F->C: SIP/2.0 302 Moved temporarily
      From: sip:charlie@caller.com
      To: Alice <sip:alice@anywhere.com>
      Call-ID: 27182@caller.com
      CSeq: 1 INVITE
      Contact: <sip:alice@anywhere.com:5080;maddr=spare.caller.com>
```

Based on this response, Charlie directs the same invitation to the secondary server spare.caller.com at port 5080, but maintains the same Request-URI as before:

```
C->S: INVITE sip:alice@anywhere.com SIP/2.0
      From: sip:charlie@caller.com
      To: Alice <sip:alice@anywhere.com>
      Call-ID: 27182@caller.com
      CSeq: 2 INVITE
```

## 16.7 Negotiation

An example of a 606 (Not Acceptable) response is:

```
S->C: SIP/2.0 606 Not Acceptable
      From: sip:mjh@isi.edu
      To: <sip:schooler@cs.caltech.edu> ;tag=7434264
      Call-ID: 14142@north.east.isi.edu
```

```
      CSeq: 1 INVITE
      Contact: sip:mjh@north.east.isi.edu
      Warning: 370 "Insufficient bandwidth (only have ISDN)",
        305 "Incompatible media format",
        330 "Multicast not available"
      Content-Type: application/sdp
      Content-Length: 50

      v=0
      s=Let’s talk
      b=CT:128
      c=IN IP4 north.east.isi.edu
      m=audio 3456 RTP/AVP 5 0 7
      m=video 2232 RTP/AVP 31
```

In this example, the original request specified a bandwidth that was higher than the access link could support, requested multicast, and requested a set of media encodings. The response states that only 128 kb/s is available and that (only) DVI, PCM or LPC audio could be supported in order of preference.

The response also states that multicast is not available. In such a case, it might be appropriate to set up a transcoding gateway and re-invite the user.

## 16.8 OPTIONS Request

A caller Alice can use an OPTIONS request to find out the capabilities of a potential callee Bob, without "ringing" the designated address. Bob returns a description indicating that he is capable of receiving audio encodings PCM Ulaw (payload type 0), 1016 (payload type 1), GSM (payload type 3), and SX7300/8000 (dynamic payload type 99), and video encodings H.261 (payload type 31) and H.263 (payload type 34).

```
C->S: OPTIONS sip:bob@example.com SIP/2.0
      From: Alice <sip:alice@anywhere.org>
      To: Bob <sip:bob@example.com>
      Call-ID: 6378@host.anywhere.org
      CSeq: 1 OPTIONS
      Accept: application/sdp

S->C: SIP/2.0 200 OK
      From: Alice <sip:alice@anywhere.org>
      To: Bob <sip:bob@example.com> ;tag=376364382
```

```
Call-ID: 6378@host.anywhere.org
Content-Length: 81
Content-Type: application/sdp

v=0
m=audio 0 RTP/AVP 0 1 3 99
m=video 0 RTP/AVP 31 34
a=rtpmap:99 SX7300/8000
```

# A Minimal Implementation

## A.1 Client

All clients MUST be able to generate the INVITE and ACK requests. Clients MUST generate and parse the Call-ID, Content-Length, Content-Type, CSeq, From and To headers. Clients MUST also parse the Require header. A minimal implementation MUST understand SDP (RFC 2327, [6]). It MUST be able to recognize the status code classes 1 through 6 and act accordingly.

The following capability sets build on top of the minimal implementation described in the previous paragraph. In general, each capability listed below builds on the ones above it:

Basic: A basic implementation adds support for the BYE method to allow the interruption of a pending call attempt. It includes a User-Agent header in its requests and indicates its preferred language in the Accept-Language header.

Redirection: To support call forwarding, a client needs to be able to understand the Contact header, but only the SIP-URL part, not the parameters.

Firewall-friendly: A firewall-friendly client understands the Route and Record-Route header fields and can be configured to use a local proxy for all outgoing requests.

Negotiation: A client MUST be able to request the OPTIONS method and understand the 380 (Alternative Service) status and the Contact parameters to participate in terminal and media negotiation. It SHOULD be able to parse the Warning response header to provide useful feedback to the caller.

Authentication: If a client wishes to invite callees that require caller authentication, it MUST be able to recognize the 401 (Unauthorized) status code, MUST be able to generate the Authorization request header and MUST understand the WWW-Authenticate response header.

If a client wishes to use proxies that require caller authentication, it MUST be able to recognize the 407 (Proxy Authentication Required) status code, MUST be able to generate the Proxy-Authorization request header and understand the Proxy-Authenticate response header.

## A.2 Server

A minimally compliant server implementation MUST understand the INVITE, ACK, OPTIONS and BYE requests. A proxy server MUST also understand CANCEL. It MUST parse and generate, as appropriate, the Call-ID, Content-Length, Content-Type, CSeq, Expires, From, Max-Forwards, Require, To and Via headers. It MUST echo the CSeq and Timestamp headers in the response. It SHOULD include the Server header in its responses.

## A.3 Header Processing

Table 6 lists the headers that different implementations support. UAC refers to a user-agent client (calling user agent), UAS to a user-agent server (called user-agent).

The fields in the table have the following meaning. Type is as in Table 4 and 5. "-" indicates the field is not meaningful to this system (although it might be generated by it). "m" indicates the field MUST be understood. "b" indicates the field SHOULD be understood by a Basic implementation. "r" indicates the field SHOULD be understood if the system claims to understand redirection. "a" indicates the field SHOULD be understood if the system claims to support authentication. "e" indicates the field SHOULD be understood if the system claims to support encryption. "o" indicates support of the field is purely optional. Headers whose support is optional for all implementations are not shown.

| | type | UAC | proxy | UAS | registrar |
|---|---|---|---|---|---|
| Accept | R | - | o | m | m |
| Accept-Encoding | R | - | - | m | m |
| Accept-Language | R | - | b | b | b |
| Allow | 405 | o | - | - | - |
| Authorization | R | a | o | a | a |
| Call-ID | g | m | m | m | m |
| Content-Encoding | g | m | - | m | m |
| Content-Length | g | m | m | m | m |
| Content-Type | g | m | - | m | m |
| CSeq | g | m | m | m | m |
| Encryption | g | e | - | e | e |
| Expires | g | - | o | o | m |
| From | g | m | o | m | m |
| Hide | R | - | m | - | - |
| Contact | R | - | - | - | m |
| Contact | r | r | r | - | - |
| Max-Forwards | R | - | b | - | - |
| Proxy-Authenticate | 407 | a | - | - | - |
| Proxy-Authorization | R | - | a | - | - |
| Proxy-Require | R | - | m | - | - |
| Require | R | m | - | m | m |
| Response-Key | R | - | - | e | e |
| Route | R | - | m | - | - |
| Timestamp | g | o | o | m | m |
| To | g | m | m | m | m |
| Unsupported | r | b | b | - | - |
| User-Agent | g | b | - | b | - |
| Via | g | m | m | m | m |
| WWW-Authenticate | 401 | a | - | - | - |

Table 6: Header Field Processing Requirements

# B Usage of the Session Description Protocol (SDP)

This section describes the use of the Session Description Protocol (SDP) (RFC 2327 [6]).

## B.1 Configuring Media Streams

The caller and callee align their media descriptions so that the nth media stream ("m=" line) in the caller's session description corresponds to the nth media stream in the callee's description.

All media descriptions SHOULD contain "a=rtpmap" mappings from RTP payload types to encodings.

> This allows easier migration away from static payload types.

If the callee wants to neither send nor receive a stream offered by the caller, the callee sets the port number of that stream to zero in its media description.

> There currently is no other way than port zero for the callee to refuse a bidirectional stream offered by the caller. Both caller and callee need to be aware what media tools are to be started.

For example, assume that the caller Alice has included the following description in her INVITE request. It includes an audio stream and two bidirectional video streams, using H.261 (payload type 31) and MPEG (payload type 32).

```
v=0
o=alice 2890844526 2890844526 IN IP4 host.anywhere.com
c=IN IP4 host.anywhere.com
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
m=video 51372 RTP/AVP 31
a=rtpmap:31 H261/90000
m=video 53000 RTP/AVP 32
a=rtpmap:32 MPV/90000
```

The callee, Bob, does not want to receive or send the first video stream, so it returns the media description below:

```
v=0
o=bob 2890844730 2890844730 IN IP4 host.example.com
c=IN IP4 host.example.com
m=audio 47920 RTP/AVP 0 1
a=rtpmap:0 PCMU/8000
a=rtpmap:1 1016/8000
m=video 0 RTP/AVP 31
m=video 53000 RTP/AVP 32
a=rtpmap:32 MPV/90000
```

## B.2 Setting SDP Values for Unicast

If a session description from a caller contains a media stream which is listed as send (receive) only, it means that the caller is only willing to send (receive) this stream, not receive (send). The same is true for the callee.

For receive-only and send-or-receive streams, the port number and address in the session description indicate where the media stream should be sent to by the recipient of the session description, either caller or callee. For send-only streams, the address and port number have no significance and SHOULD be set to zero.

The list of payload types for each media stream conveys two pieces of information, namely the set of codecs that the caller or callee is capable of sending or receiving, and the RTP payload type numbers used to identify those codecs. For receive-only or send-and-receive media streams, a caller SHOULD list all of the codecs it is capable of supporting in the session description in an INVITE or ACK. For send-only streams, the caller SHOULD indicate only those it wishes to send for this session. For receive-only streams, the payload type numbers indicate the value of the payload type field in RTP packets the caller is expecting to receive for that codec type. For send-only streams, the payload type numbers indicate the value of the payload type field in RTP packets the caller is planning to send for that codec type. For send-and-receive streams, the payload type numbers indicate the value of the payload type field the caller expects to both send and receive.

If a media stream is listed as receive-only by the caller, the callee lists, in the response, those codecs it intends to use from among the ones listed in the request. If a media stream is listed as send-only by the caller, the callee lists, in the response, those codecs it is willing to receive among the ones listed in the the request. If the media stream is listed as both send and receive, the callee lists those codecs it is capable of sending or receiving among the ones listed by the caller in the INVITE. The actual payload type numbers in the callee’s session description corresponding to a particular codec MUST be the same as the caller’s session description.

If caller and callee have no media formats in common for a particular stream, the callee MUST return a session description containing the particular "m=" line, but with the port number set to zero, and no payload types listed.

If there are no media formats in common for all streams, the callee SHOULD return a 400 response, with a 304 Warning header field.

## B.3 Multicast Operation

The interpretation of send-only and receive-only for multicast media sessions differs from that for unicast sessions. For multicast, send-only means that the recipient of the session description (caller or callee) SHOULD only send media streams to the address and port indicated. Receive-only means that the recipient of the session description SHOULD only receive media on the address and port indicated.

For multicast, receive and send multicast addresses are the same and all parties use the same port numbers to receive media data. If the session description provided by the caller is acceptable to the callee, the callee can choose not to include a session description or MAY echo the description in the response.

A callee MAY, in the response, return a session description with some of the payload types removed, or port numbers set to zero (but no other value). This indicates to the caller that the callee does not support the given stream or media types which were removed. A callee MUST NOT change whether a given stream is send-only, receive-only, or send-and-receive.

If a callee does not support multicast at all, it SHOULD return a 400 status response and include a 330 Warning.

## B.4 Delayed Media Streams

In some cases, a caller may not know the set of media formats which it can support at the time it would like to issue an invitation. This is the case when the caller is actually a gateway to another protocol which performs media format negotiation after call setup. When this occurs, a caller MAY issue an INVITE with a session description that contains no media lines. The callee SHOULD interpret this to mean that the caller wishes to participate in a multimedia session described by the session description, but that the media streams are not yet known. The callee SHOULD return a session description indicating the streams and media formats it is willing to support, however. The caller MAY update the session description either in the ACK request or in a re-INVITE at a later time, once the streams are known.

## B.5 Putting Media Streams on Hold

If a party in a call wants to put the other party "on hold", i.e., request that it temporarily stops sending one or more media streams, a party re-invites the other by sending an INVITE request with a modified session description. The session description is the same as in the original invitation (or response), but the "c" destination addresses for the media streams to be put on hold are set to zero (0.0.0.0).

## B.6 Subject and SDP "s=" Line

The SDP "s=" line and the SIP Subject header field have different meanings when inviting to a multicast session. The session description line describes the subject of the multicast session, while the SIP Subject header field describes the reason for the invitation. The example in Section 16.2 illustrates this point. For invitations to two-party sessions, the SDP "s=" line MAY be left empty.

## B.7 The SDP "o=" Line

The "o=" line is not strictly necessary for two-party sessions, but MUST be present to allow re-use of SDP-based tools.

## C Summary of Augmented BNF

All of the mechanisms specified in this document are described in both prose and an augmented Backus-Naur Form (BNF) similar to that used by RFC 822 [9]. Implementors will need to be familiar with the notation in order to understand this specification. The augmented BNF includes the following constructs:

```
name  =  definition
```

The name of a rule is simply the name itself (without any enclosing "<" and ">") and is separated from its definition by the equal "=" character. White space is only significant in that indentation of continuation lines is used to indicate a rule definition that spans more than one line. Certain basic rules are in uppercase, such as SP, LWS, HT, CRLF, DIGIT, ALPHA, etc. Angle brackets are used within definitions whenever their presence will facilitate discerning the use of rule names.

```
"literal"
```

Quotation marks surround literal text. Unless stated otherwise, the text is case-insensitive.

```
rule1 | rule2
```

Elements separated by a bar ("|") are alternatives, e.g., "yes | no" will accept yes or no.

```
(rule1 rule2)
```

Elements enclosed in parentheses are treated as a single element. Thus, "(elem (foo | bar) elem)" allows the token sequences "elem foo elem" and "elem bar elem".

*rule

The character "*" preceding an element indicates repetition. The full form is "<n>*<m>element" indicating at least <n> and at most <m> occurrences of element. Default values are 0 and infinity so that "*(element)" allows any number, including zero; "1*element" requires at least one; and "1*2element" allows one or two.

[rule]

Square brackets enclose optional elements; "[foo bar]" is equivalent to "*1(foo bar)".

N rule

Specific repetition: "<n>(element)" is equivalent to "<n>*<n>(element)"; that is, exactly <n> occurrences of (element). Thus 2DIGIT is a 2-digit number, and 3ALPHA is a string of three alphabetic characters.

#rule

A construct "#" is defined, similar to "*", for defining lists of elements. The full form is "<n>#<m> element" indicating at least <n> and at most <m> elements, each separated by one or more commas (",") and OPTIONAL linear white space (LWS). This makes the usual form of lists very easy; a rule such as

```
( *LWS element *( *LWS "," *LWS element ))
```

can be shown as 1# element. Wherever this construct is used, null elements are allowed, but do not contribute to the count of elements present. That is, "(element), , (element)" is permitted, but counts as only two elements. Therefore, where at least one element is required, at least one non-null element MUST be present. Default values are 0 and infinity so that "#element" allows any number, including zero; "1#element" requires at least one; and "1#2element" allows one or two.

; comment

A semi-colon, set off some distance to the right of rule text, starts a comment that continues to the end of line. This is a simple way of including useful notes in parallel with the specifications.

implied *LWS

The grammar described by this specification is word-based. Except where noted otherwise, linear white space (LWS) can be included between any two adjacent words (token or quoted-string), and between adjacent tokens and separators, without changing the interpretation of a field. At least one delimiter (LWS and/or separators) MUST exist between any two tokens (for the definition of "token" below), since they would otherwise be interpreted as a single token.

## C.1 Basic Rules

The following rules are used throughout this specification to describe basic parsing constructs. The US-ASCII coded character set is defined by ANSI X3.4-1986.

```
OCTET     =  <any 8-bit sequence of data>
CHAR      =  <any US-ASCII character (octets 0 - 127)>
upalpha   =  "A" | "B" | "C" | "D" | "E" | "F" | "G" | "H" | "I" |
             "J" | "K" | "L" | "M" | "N" | "O" | "P" | "Q" | "R" |
             "S" | "T" | "U" | "V" | "W" | "X" | "Y" | "Z"
lowalpha  =  "a" | "b" | "c" | "d" | "e" | "f" | "g" | "h" | "i" |
             "j" | "k" | "l" | "m" | "n" | "o" | "p" | "q" | "r" |
             "s" | "t" | "u" | "v" | "w" | "x" | "y" | "z"
alpha     =  lowalpha | upalpha
digit     =  "0" | "1" | "2" | "3" | "4" | "5" | "6" | "7" |
             "8" | "9"
alphanum  =  alpha | digit
CTL       =  <any US-ASCII control character
             (octets 0 -- 31) and DEL (127)>
CR        =  %d13 ; US-ASCII CR, carriage return character
LF        =  %d10 ; US-ASCII LF, line feed character
SP        =  %d32 ; US-ASCII SP, space character
HT        =  %d09 ; US-ASCII HT, horizontal tab character
CRLF      =  CR LF ; typically the end of a line
```

The following are defined in RFC 2396 [12] for the SIP URI:

```
unreserved  =  alphanum | mark
mark        =  "-" | "_" | "." | "!" | "~" | "*" | "'"
           |   "(" | ")"
escaped     =  "%" hex hex
```

SIP header field values can be folded onto multiple lines if the continuation line begins with a space or horizontal tab. All linear white space, including folding, has the same semantics as SP. A recipient MAY replace any linear white space with a single SP before interpreting the field value or forwarding the message downstream.

```
LWS  =  [CRLF] 1*( SP | HT ) ; linear whitespace
```

The TEXT-UTF8 rule is only used for descriptive field contents and values that are not intended to be interpreted by the message parser. Words of *TEXT-UTF8 contain characters from the UTF-8 character set (RFC 2279 [21]). In this regard, SIP differs from HTTP, which uses the ISO 8859-1 character set.

```
TEXT-UTF8  =  <any UTF-8 character encoding, except CTLs,
              but including LWS>
```

A CRLF is allowed in the definition of TEXT-UTF8 only as part of a header field continuation. It is expected that the folding LWS will be replaced with a single SP before interpretation of the TEXT-UTF8 value.

Hexadecimal numeric characters are used in several protocol elements.

```
hex  =  "A" | "B" | "C" | "D" | "E" | "F"
        | "a" | "b" | "c" | "d" | "e" | "f" | digit
```

Many SIP header field values consist of words separated by LWS or special characters. These special characters MUST be in a quoted string to be used within a parameter value.

```
token       =  1*< any CHAR  except CTL's  or separators>
separators  =  "(" | ")" | "<" | ">" | "@" |
               "," | ";" | ":" | "\" | <"> |
               "/" | "[" | "]" | "?" | "=" |
               "{" | "}" | SP | HT
```

Comments can be included in some SIP header fields by surrounding the comment text with parentheses. Comments are only allowed in fields containing "comment" as part of their field value definition. In all other fields, parentheses are considered part of the field value.

```
comment  =  "(" *(ctext | quoted-pair | comment) ")"
ctext    =  < any TEXT-UTF8  excluding "("  and ")">
```

A string of text is parsed as a single word if it is quoted using double-quote marks.

```
quoted-string  =  ( <"> *(qdtext | quoted-pair ) <"> )
qdtext         =  <any TEXT-UTF8 except <">>
```

The backslash character ("\") MAY be used as a single-character quoting mechanism only within quoted-string and comment constructs.

```
quoted-pair  =  " \ " CHAR
```

## D Using SRV DNS Records

The following procedure is experimental and relies on DNS SRV records (RFC 2052 [14]). The steps listed below are used in place of the two steps in section 1.4.2.

If a step elicits no addresses, the client continues to the next step. However if a step elicits one or more addresses, but no SIP server at any of those addresses responds, then the client concludes the server is down and doesn't continue on to the next step.

When SRV records are to be used, the protocol to use when querying for the SRV record is "sip". SRV records contain port numbers for servers, in addition to IP addresses; the client always uses this port number when contacting the SIP server. Otherwise, the port number in the SIP URI is used, if present. If there is no port number in the URI, the default port, 5060, is used.

1. If the host portion of the Request-URI is an IP address, the client contacts the server at the given address. If the host portion of the Request-URI is not an IP address, the client proceeds to the next step.

2. The Request-URI is examined. If it contains an explicit port number, the next two steps are skipped.

3. The Request-URI is examined. If it does not specify a protocol (TCP or UDP), the client queries the name server for SRV records for both UDP (if supported by the client) and TCP (if supported by the client) SIP servers. The format of these queries is defined in RFC 2052 [14]. The results of the query or queries are merged together and ordered based on priority. Then, the searching technique outlined in RFC 2052 [14] is used to select servers in order. If DNS doesn't return any records, the user goes to the last step. Otherwise, the user attempts to contact each server in the order listed. If no server is contacted, the user gives up.

4. If the Request-URI specifies a protocol (TCP or UDP) that is supported by the client, the client queries the name server for SRV records for SIP servers of that protocol type only. If the client does not support the protocol specified in the Request-URI, it gives up. The searching technique outlined in RFC 2052 [14] is used to select servers from the DNS response in order. If DNS doesn't

return any records, the user goes to the last step. Otherwise, the user attempts to contact each server in the order listed. If no server is contacted, the user gives up.

5. The client queries the name server for address records for the host portion of the Request-URI. If there were no address records, the client stops, as it has been unable to locate a server. By address record, we mean A RR's, AAAA RR's, or their most modern equivalent.

A client MAY cache a successful DNS query result. A successful query is one which contained records in the answer, and a server was contacted at one of the addresses from the answer. When the client wishes to send a request to the same host, it starts the search as if it had just received this answer from the name server. The server uses the procedures specified in RFC1035 [15] regarding cache invalidation when the time-to-live of the DNS result expires. If the client does not find a SIP server among the addresses listed in the cached answer, it starts the search at the beginning of the sequence described above.

For example, consider a client that wishes to send a SIP request. The Request-URI for the destination is sip:user@company.com. The client only supports UDP. It would follow these steps:

1. The host portion is not an IP address, so the client goes to step 2 above.

2. The client does a DNS query of QNAME="sip.udp.company.com", QCLASS=IN, QTYPE=SRV. Since it doesn't support TCP, it omits the TCP query. There were no addresses in the DNS response, so the client goes to the next step.

3. The client does a DNS query for A records for "company.com". An address is found, so that client attempts to contact a server at that address at port 5060.

## E IANA Considerations

Section 4.4 describes a name space and mechanism for registering SIP options.

Section 6.41 describes the name space for registering SIP warn-codes.

## F Acknowledgments

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# SIP: Session Initiation Protocol

## Status of this Memo

This document specifies an Internet standards track protocol for the Internet community, and requests discussion and suggestions for improvements. Please refer to the current edition of the "Internet Official Protocol Standards" (STD 1) for the standardization state and status of this protocol. Distribution of this memo is unlimited.

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## Abstract

This document describes Session Initiation Protocol (SIP), an application-layer control (signaling) protocol for creating, modifying, and terminating sessions with one or more participants. These sessions include Internet telephone calls, multimedia distribution, and multimedia conferences.

SIP invitations used to create sessions carry session descriptions that allow participants to agree on a set of compatible media types. SIP makes use of elements called proxy servers to help route requests to the user's current location, authenticate and authorize users for services, implement provider call-routing policies, and provide features to users. SIP also provides a registration function that allows users to upload their current locations for use by proxy servers. SIP runs on top of several different transport protocols.

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# 1 Introduction

There are many applications of the Internet that require the creation and management of a session, where a session is considered an exchange of data between an association of participants. The implementation of these applications is complicated by the practices of participants: users may move between endpoints, they may be addressable by multiple names, and they may communicate in several different media - sometimes simultaneously. Numerous protocols have been authored that carry various forms of real-time multimedia session data such as voice, video, or text messages. The Session Initiation Protocol (SIP) works in concert with these protocols by enabling Internet endpoints (called user agents) to discover one another and to agree on a characterization of a session they would like to share. For locating prospective session participants, and for other functions, SIP enables the creation of an infrastructure of network hosts (called proxy servers) to which user agents can send registrations, invitations to sessions, and other requests. SIP is an agile, general-purpose tool for creating, modifying, and terminating sessions that works independently of underlying transport protocols and without dependency on the type of session that is being established.

## 2 Overview of SIP Functionality

SIP is an application-layer control protocol that can establish, modify, and terminate multimedia sessions (conferences) such as Internet telephony calls. SIP can also invite participants to already existing sessions, such as multicast conferences. Media can be added to (and removed from) an existing session. SIP transparently supports name mapping and redirection services, which supports personal mobility [27] - users can maintain a single externally visible identifier regardless of their network location.

SIP supports five facets of establishing and terminating multimedia communications:

User location: determination of the end system to be used for communication;

User availability: determination of the willingness of the called party to engage in communications;

User capabilities: determination of the media and media parameters to be used;

Session setup: "ringing", establishment of session parameters at both called and calling party;

Session management: including transfer and termination of sessions, modifying session parameters, and invoking services.

SIP is not a vertically integrated communications system. SIP is rather a component that can be used with other IETF protocols to build a complete multimedia architecture. Typically, these architectures will include protocols such as the Real-time Transport Protocol (RTP) (RFC 1889 [28]) for transporting real-time data and providing QoS feedback, the Real-Time streaming protocol (RTSP) (RFC 2326 [29]) for controlling delivery of streaming media, the Media

Gateway Control Protocol (MEGACO) (RFC 3015 [30]) for controlling gateways to the Public Switched Telephone Network (PSTN), and the Session Description Protocol (SDP) (RFC 2327 [1]) for describing multimedia sessions. Therefore, SIP should be used in conjunction with other protocols in order to provide complete services to the users. However, the basic functionality and operation of SIP does not depend on any of these protocols.

SIP does not provide services. Rather, SIP provides primitives that can be used to implement different services. For example, SIP can locate a user and deliver an opaque object to his current location. If this primitive is used to deliver a session description written in SDP, for instance, the endpoints can agree on the parameters of a session. If the same primitive is used to deliver a photo of the caller as well as the session description, a "caller ID" service can be easily implemented. As this example shows, a single primitive is typically used to provide several different services.

SIP does not offer conference control services such as floor control or voting and does not prescribe how a conference is to be managed. SIP can be used to initiate a session that uses some other conference control protocol. Since SIP messages and the sessions they establish can pass through entirely different networks, SIP cannot, and does not, provide any kind of network resource reservation capabilities.

The nature of the services provided make security particularly important. To that end, SIP provides a suite of security services, which include denial-of-service prevention, authentication (both user to user and proxy to user), integrity protection, and encryption and privacy services.

SIP works with both IPv4 and IPv6.

# 3 Terminology

In this document, the key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" are to be interpreted as described in BCP 14, RFC 2119 [2] and indicate requirement levels for compliant SIP implementations.

# 4 Overview of Operation

This section introduces the basic operations of SIP using simple examples. This section is tutorial in nature and does not contain any normative statements.

The first example shows the basic functions of SIP: location of an end point, signal of a desire to communicate, negotiation of session parameters to establish the session, and teardown of the session once established.

Figure 1 shows a typical example of a SIP message exchange between two users, Alice and Bob. (Each message is labeled with the letter "F" and a number for reference by the text.) In this example, Alice uses a SIP application on her PC (referred to as a softphone) to call Bob on his SIP phone over the Internet. Also shown are two SIP proxy servers that act on behalf of Alice and Bob to facilitate the session establishment. This typical arrangement is often referred to as the "SIP trapezoid" as shown by the geometric shape of the dotted lines in Figure 1.

Alice "calls" Bob using his SIP identity, a type of Uniform Resource Identifier (URI) called a SIP URI. SIP URIs are defined in Section 19.1. It has a similar form to an email address, typically containing a username and a host name. In this case, it is sip:bob@biloxi.com, where biloxi.com is the domain of Bob's SIP service provider. Alice has a SIP URI of sip:alice@atlanta.com. Alice might have typed in Bob's URI or perhaps clicked on a hyperlink or an entry in an address book. SIP also provides a secure URI, called a SIPS URI. An example would be sips:bob@biloxi.com. A call made to a SIPS URI guarantees that secure, encrypted transport (namely TLS) is used to carry all SIP messages from the caller to the domain of the callee. From there, the request is sent securely to the callee, but with security mechanisms that depend on the policy of the domain of the callee.

SIP is based on an HTTP-like request/response transaction model. Each transaction consists of a request that invokes a particular method, or function, on the server and at least one response. In this example, the transaction begins with Alice's softphone sending an INVITE request addressed to Bob's SIP URI. INVITE is an example of a SIP method that specifies the action that the requestor (Alice) wants the server (Bob) to take. The INVITE request contains a number of header fields. Header fields are named attributes that provide additional information about a message. The ones present in an INVITE include a unique identifier for the call, the destination address, Alice's address, and information about the type of session that Alice wishes to establish with Bob. The INVITE (message F1 in Figure 1) might look like this:

```
                     atlanta.com  . . . biloxi.com
                 .      proxy              proxy     .
               .                                       .
       Alice’s  . . . . . . . . . . . . . . . . . . . .  Bob’s
      softphone                                        SIP Phone
         |                |                |                |
         |    INVITE F1   |                |                |
         |--------------->|    INVITE F2   |                |
         |  100 Trying F3 |--------------->|    INVITE F4   |
         |<---------------|  100 Trying F5 |--------------->|
         |                |<-------------- | 180 Ringing F6 |
         |                | 180 Ringing F7 |<---------------|
         | 180 Ringing F8 |<---------------|     200 OK F9  |
         |<---------------|    200 OK F10  |<---------------|
         |    200 OK F11  |<---------------|                |
         |<---------------|                |                |
         |                       ACK F12                    |
         |------------------------------------------------->|
         |                   Media Session                  |
         |<================================================>|
         |                       BYE F13                    |
         |<-------------------------------------------------|
         |                     200 OK F14                   |
         |------------------------------------------------->|
         |                                                  |
```

Figure 1: SIP session setup example with SIP trapezoid

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP 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>
Content-Type: application/sdp
Content-Length: 142

(Alice’s SDP not shown)
```

The first line of the text-encoded message contains the method name (INVITE). The lines that follow are a list of header fields. This example contains a minimum required set. The header fields are briefly described below:

Via contains the address (pc33.atlanta.com) at which Alice is expecting to receive responses to this request. It also contains a branch parameter that identifies this transaction.

To contains a display name (Bob) and a SIP or SIPS URI (sip:bob@biloxi.com) towards which the request was originally directed. Display names are described in RFC 2822 [3].

From also contains a display name (Alice) and a SIP or SIPS URI (sip:alice@atlanta.com) that indicate the originator of the request. This header field also has a tag parameter containing a random string (1928301774) that was added to the URI by the softphone. It is used for identification purposes.

Call-ID contains a globally unique identifier for this call, generated by the combination of a random string and the softphone's host name or IP address. The combination of the To tag, From tag, and Call-ID completely defines a peer-to-peer SIP relationship between Alice and Bob and is referred to as a dialog.

CSeq or Command Sequence contains an integer and a method name. The CSeq number is incremented for each new request within a dialog and is a traditional sequence number.

Contact contains a SIP or SIPS URI that represents a direct route to contact Alice, usually composed of a username at a fully qualified domain name (FQDN). While an FQDN is preferred, many end systems do not have registered domain names, so IP addresses are permitted. While the Via header field tells other elements where to send the response, the Contact header field tells other elements where to send future requests.

Max-Forwards serves to limit the number of hops a request can make on the way to its destination. It consists of an integer that is decremented by one at each hop.

Content-Type contains a description of the message body (not shown).

Content-Length contains an octet (byte) count of the message body.

The complete set of SIP header fields is defined in Section 20.

The details of the session, such as the type of media, codec, or sampling rate, are not described using SIP. Rather, the body of a SIP message contains a description of the session, encoded in some other protocol format. One such format is the Session Description Protocol (SDP) (RFC 2327 [1]). This SDP message (not shown in the example) is carried by the SIP message in a way that is analogous to a document attachment being carried by an email message, or a web page being carried in an HTTP message.

Since the softphone does not know the location of Bob or the SIP server in the biloxi.com domain, the softphone sends the INVITE to the SIP server that serves Alice's domain, atlanta.com. The address of the atlanta.com SIP server could have been configured in Alice's softphone, or it could have been discovered by DHCP, for example.

The atlanta.com SIP server is a type of SIP server known as a proxy server. A proxy server receives SIP requests and forwards them on behalf of the requestor. In this example, the proxy server receives the INVITE request and sends a 100 (Trying) response back to Alice's softphone. The 100 (Trying) response indicates that the INVITE has been received and that the proxy is working on her behalf to route the INVITE to the destination. Responses in SIP use a three-digit code followed by a descriptive phrase. This response contains the same To, From, Call-ID, CSeq and branch parameter in the Via as the INVITE, which allows Alice's softphone to correlate this response to the sent INVITE. The atlanta.com proxy server locates the proxy server at biloxi.com, possibly by performing a particular type of DNS (Domain Name Service) lookup to find the SIP server that serves the biloxi.com domain. This is described in [4]. As a result, it obtains the IP address of the biloxi.com proxy server and forwards, or proxies, the INVITE request there. Before forwarding the request, the atlanta.com proxy server adds an additional Via header field value that contains its own address (the INVITE already contains Alice's address in the first Via). The biloxi.com proxy server receives the INVITE and responds with a 100 (Trying) response back to the atlanta.com proxy server to indicate that it has received the INVITE and is processing the request. The proxy server consults a database, generically called a location service, that contains the current IP address of Bob. (We shall see in the next section how this database can be populated.) The biloxi.com proxy server adds another Via header field value with its own address to the INVITE and proxies it to Bob's SIP phone.

Bob's SIP phone receives the INVITE and alerts Bob to the incoming call from Alice so that Bob can decide whether to answer the call, that is, Bob's phone rings. Bob's SIP phone indicates this in a 180 (Ringing) response, which is routed back through the two proxies in the reverse direction. Each proxy uses the Via header field to determine where to send the response and removes its own address from the top. As a result, although DNS and location service lookups were required to route the initial INVITE, the 180 (Ringing) response can be returned to the caller without lookups or without state being maintained in the proxies. This also has the desirable property that each proxy that sees the INVITE will also see all responses to the INVITE.

When Alice's softphone receives the 180 (Ringing) response, it passes this information to Alice, perhaps using an audio ringback tone or by displaying a message on Alice's screen.

In this example, Bob decides to answer the call. When he picks up the handset, his SIP phone sends a 200 (OK) response to indicate that the call has been answered. The 200 (OK) contains a message body with the SDP media description of the type of session that Bob is willing to establish with Alice. As a result, there is a two-phase exchange of SDP messages: Alice sent one to Bob, and Bob sent one back to Alice. This two-phase exchange provides basic negotiation capabilities and is based on a simple offer/answer model of SDP exchange. If Bob did not wish to answer the call or was busy on another call, an error response would have been sent instead of the 200 (OK), which would have resulted in no media session being established. The complete list of SIP response codes is in Section 21. The 200 (OK) (message F9 in Figure 1) might look like this as Bob sends it out:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP server10.biloxi.com
   ;branch=z9hG4bKnashds8;received=192.0.2.3
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
   ;branch=z9hG4bK77ef4c2312983.1;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com
   ;branch=z9hG4bK776asdhds ;received=192.0.2.1
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131

(Bob's SDP not shown)
```

The first line of the response contains the response code (200) and the reason phrase (OK). The remaining lines contain header fields. The Via, To, From, Call-ID, and CSeq header fields are copied from the INVITE request. (There are three Via header field values - one added by Alice's SIP phone, one added by the atlanta.com proxy, and one added by the biloxi.com proxy.) Bob's SIP phone has added a tag parameter to the To header field. This tag will be incorporated by both endpoints into the dialog and will be included in all future requests and responses in this call. The Contact header field contains a URI at which Bob can be directly reached at his SIP phone. The Content-Type and Content-Length refer to the message body (not shown) that contains Bob's SDP media information.

In addition to DNS and location service lookups shown in this example, proxy servers can make flexible "routing decisions" to decide where to send a request. For example, if Bob's SIP phone returned a 486 (Busy Here) response, the biloxi.com proxy server could proxy the INVITE to Bob's voicemail server. A proxy server can also send an INVITE to a number of locations at the same time. This type of parallel search is known as forking.

In this case, the 200 (OK) is routed back through the two proxies and is received by Alice's softphone, which then stops the ringback tone and indicates that the call has been answered. Finally, Alice's softphone sends an acknowledgement message, ACK, to Bob's SIP phone to confirm the reception of the final response (200 (OK)). In this example, the ACK is sent directly from Alice's softphone to Bob's SIP phone, bypassing the two proxies. This occurs because the endpoints have learned each other's address from the Contact header fields through the INVITE/200 (OK) exchange, which was not known when the initial INVITE was sent. The lookups performed by the two proxies are no longer needed, so the proxies drop out of the call flow. This completes the INVITE/200/ACK three-way handshake used to establish SIP sessions. Full details on session setup are in Section 13.

Alice and Bob's media session has now begun, and they send media packets using the format to which they agreed in the exchange of SDP. In general, the end-to-end media packets take a different path from the SIP signaling messages.

During the session, either Alice or Bob may decide to change the characteristics of the media session. This is accomplished by sending a re-INVITE containing a new media description. This re-INVITE references the existing dialog so that the other party knows that it is to modify an existing session instead of establishing a new session. The other party sends a 200 (OK) to accept the change. The requestor responds to the 200 (OK) with an ACK. If the other party does not accept the change, he sends an error response such as 488 (Not Acceptable Here), which also receives an ACK. However, the failure of the re-INVITE does not cause the existing call to fail - the session continues using the previously negotiated characteristics. Full details on session modification are in Section 14.

At the end of the call, Bob disconnects (hangs up) first and generates a BYE message. This BYE is routed directly to Alice's softphone, again bypassing the proxies. Alice confirms receipt of the BYE with a 200 (OK) response, which terminates the session and the BYE transaction. No ACK is sent - an ACK is only sent in response to a response to an INVITE request. The reasons for this special handling for INVITE will be discussed later, but relate to the reliability mechanisms in SIP, the length of time it can take for a ringing phone to be answered, and forking. For this reason, request handling in SIP is often classified as either INVITE or non-INVITE, referring to all other methods besides INVITE. Full details on session termination are in Section 15.

Section 24.2 describes the messages shown in Figure 1 in full.

In some cases, it may be useful for proxies in the SIP signaling path to see all the messaging between the endpoints for the duration of the session. For example, if the biloxi.com proxy server wished to remain in the SIP messaging path beyond the initial INVITE, it would add to the INVITE a required routing header field known as Record-Route that contained a URI resolving to the hostname or IP address of the proxy. This information would be received by both Bob's SIP phone and (due to the Record-Route header field being passed back in the 200 (OK)) Alice's softphone and stored for the duration of the dialog. The biloxi.com proxy server would then receive and proxy the ACK, BYE, and 200 (OK) to the BYE. Each proxy can independently decide to receive subsequent messages, and those messages will pass through all proxies that elect to receive it. This capability is frequently used for proxies that are providing mid-call features.

Registration is another common operation in SIP. Registration is one way that the biloxi.com server can learn the current location of Bob. Upon initialization, and at periodic intervals, Bob's SIP phone sends REGISTER messages to a server in the biloxi.com domain known as a SIP registrar. The REGISTER messages associate Bob's SIP or SIPS URI (sip:bob@biloxi.com) with the machine into which he is currently logged (conveyed as a SIP or SIPS URI in the Contact header field). The registrar writes this association, also called a binding, to a database, called the location service, where it can be used by the proxy in the biloxi.com domain. Often, a registrar server for a domain is co-located with the proxy for that domain. It is an important concept that the distinction between types of SIP servers is logical, not physical.

Bob is not limited to registering from a single device. For example, both his SIP phone at home and the one in the office could send registrations. This information is stored together in the location service and allows a proxy to perform various types of searches to locate Bob. Similarly, more than one user can be registered on a single device at the same time.

The location service is just an abstract concept. It generally contains information that allows a proxy to input a URI and receive a set of zero or more URIs that tell the proxy where to send the request. Registrations are one way to create this information, but not the only way. Arbitrary mapping functions can be configured at the discretion of the administrator.

Finally, it is important to note that in SIP, registration is used for routing incoming SIP requests and has no role in authorizing outgoing requests. Authorization and authentication are handled in SIP either on a request-by-request basis with a challenge/response mechanism, or by using a lower layer scheme as discussed in Section 26.

The complete set of SIP message details for this registration example is in Section 24.1.

Additional operations in SIP, such as querying for the capabilities of a SIP server or client using OPTIONS, or canceling a pending request using CANCEL, will be introduced in later sections.

# 5 Structure of the Protocol

SIP is structured as a layered protocol, which means that its behavior is described in terms of a set of fairly independent processing stages with only a loose coupling between each stage. The protocol behavior is described as layers for the purpose of presentation, allowing the description of functions common across elements in a single section. It does not dictate an implementation in any way. When we say that an element "contains" a layer, we mean it is compliant to the set of rules defined by that layer.

Not every element specified by the protocol contains every layer. Furthermore, the elements specified by SIP are logical elements, not physical ones. A physical realization can choose to act as different logical elements, perhaps even on a transaction-by-transaction basis.

The lowest layer of SIP is its syntax and encoding. Its encoding is specified using an augmented Backus-Naur Form grammar (BNF). The complete BNF is specified in Section 25; an overview of a SIP message's structure can be found in Section 7.

The second layer is the transport layer. It defines how a client sends requests and receives responses and how a server receives requests and sends responses over the network. All SIP elements contain a transport layer. The transport layer is described in Section 18.

The third layer is the transaction layer. Transactions are a fundamental component of SIP. A transaction is a request sent by a client transaction (using the transport layer) to a server transaction, along with all responses to that request sent from the server transaction back to the client. The transaction layer handles application-layer retransmissions, matching of responses to requests, and application-layer timeouts. Any task that a user agent client (UAC) accomplishes takes place using a series of transactions. Discussion of transactions can be found in Section 17. User agents contain a transaction layer, as do stateful proxies. Stateless proxies do not contain a transaction layer. The transaction layer has a client component (referred to as a client transaction) and a server component (referred to as a server transaction), each of which are represented by a finite state machine that is constructed to process a particular request.

The layer above the transaction layer is called the transaction user (TU). Each of the SIP entities, except the stateless proxy, is a transaction user. When a TU wishes to send a request, it creates a client transaction instance and passes it the request along with the destination IP address, port, and transport to which to send the request. A TU that creates a client transaction can also cancel it. When a client cancels a transaction, it requests that the server stop further processing, revert to the state that existed before the transaction was initiated, and generate a specific error response to that transaction. This is done with a CANCEL request, which constitutes its own transaction, but references the transaction to be cancelled (Section 9).

The SIP elements, that is, user agent clients and servers, stateless and stateful proxies and registrars, contain a core that distinguishes them from each other. Cores, except for the stateless proxy, are transaction users. While the behavior of the UAC and UAS cores depends on the method, there are some common rules for all methods (Section 8). For a UAC, these rules govern the construction of a request; for a UAS, they govern the processing of a request and generating a response. Since registrations play an important role in SIP, a UAS that handles a REGISTER is given the special name registrar. Section 10 describes UAC and UAS core behavior for the REGISTER method. Section 11 describes UAC and UAS core behavior for the OPTIONS method, used for determining the capabilities of a UA.

Certain other requests are sent within a dialog. A dialog is a peer-to-peer SIP relationship between two user agents that persists for some time. The dialog facilitates sequencing of messages and proper routing of requests between the user agents. The INVITE method is the only way defined in this specification to establish a dialog. When a UAC sends a request that is within the context of a dialog, it follows the common UAC rules as discussed in Section 8 but also the rules for mid-dialog requests. Section 12 discusses dialogs and presents the procedures for their construction and maintenance, in addition to construction of requests within a dialog.

The most important method in SIP is the INVITE method, which is used to establish a session between participants. A session is a collection of participants, and streams of media between them, for the purposes of communication. Section 13 discusses how sessions are initiated, resulting in one or more SIP dialogs. Section 14 discusses how characteristics of that session are modified through the use of an INVITE request within a dialog. Finally, section 15 discusses how a session is terminated.

The procedures of Sections 8, 10, 11, 12, 13, 14, and 15 deal entirely with the UA core (Section 9 describes cancellation, which applies to both UA core and proxy core). Section 16 discusses the proxy element, which facilitates routing of messages between user agents.

# 6 Definitions

The following terms have special significance for SIP.

Address-of-Record: An address-of-record (AOR) is a SIP or SIPS URI that points to a domain with a location service that can map the URI to another URI where the user might be available. Typically, the location service is populated through registrations. An AOR is frequently thought of as the "public address" of the user.

Back-to-Back User Agent: A back-to-back user agent (B2BUA) is a logical entity that receives a request and processes it as a user agent server (UAS). In order to determine how the request should be answered, it acts as a user agent client (UAC) and generates requests. Unlike a proxy server, it maintains dialog state and must participate in all requests sent on the dialogs it has established. Since it is a concatenation of a UAC and UAS, no explicit definitions are needed for its behavior.

Call: A call is an informal term that refers to some communication between peers, generally set up for the purposes of a multimedia conversation.

Call Leg: Another name for a dialog [31]; no longer used in this specification.

Call Stateful: A proxy is call stateful if it retains state for a dialog from the initiating INVITE to the terminating BYE request. A call stateful proxy is always transaction stateful, but the converse is not necessarily true.

Client: A client is any network element that sends SIP requests and receives SIP responses. Clients may or may not interact directly with a human user. User agent clients and proxies are clients.

Conference: A multimedia session (see below) that contains multiple participants.

Core: Core designates the functions specific to a particular type of SIP entity, i.e., specific to either a stateful or stateless proxy, a user agent or registrar. All cores, except those for the stateless proxy, are transaction users.

Dialog: A dialog is a peer-to-peer SIP relationship between two UAs that persists for some time. A dialog is established by SIP messages, such as a 2xx response to an INVITE request. A dialog is identified by a call identifier, local tag, and a remote tag. A dialog was formerly known as a call leg in RFC 2543.

Downstream: A direction of message forwarding within a transaction that refers to the direction that requests flow from the user agent client to user agent server.

Final Response: A response that terminates a SIP transaction, as opposed to a provisional response that does not. All 2xx, 3xx, 4xx, 5xx and 6xx responses are final.

Header: A header is a component of a SIP message that conveys information about the message. It is structured as a sequence of header fields.

Header Field: A header field is a component of the SIP message header. A header field can appear as one or more header field rows. Header field rows consist of a header field name and zero or more header field values. Multiple header field values on a given header field row are separated by commas. Some header fields can only have a single header field value, and as a result, always appear as a single header field row.

Header Field Value: A header field value is a single value; a header field consists of zero or more header field values.

Home Domain: The domain providing service to a SIP user. Typically, this is the domain present in the URI in the address-of-record of a registration.

Informational Response: Same as a provisional response.

Initiator, Calling Party, Caller: The party initiating a session (and dialog) with an INVITE request. A caller retains this role from the time it sends the initial INVITE that established a dialog until the termination of that dialog.

Invitation: An INVITE request.

Invitee, Invited User, Called Party, Callee: The party that receives an INVITE request for the purpose of establishing a new session. A callee retains this role from the time it receives the INVITE until the termination of the dialog established by that INVITE.

Location Service: A location service is used by a SIP redirect or proxy server to obtain information about a callee’s possible location(s). It contains a list of bindings of address-of-record keys to zero or more contact addresses. The bindings can be created and removed in many ways; this specification defines a REGISTER method that updates the bindings.

Loop: A request that arrives at a proxy, is forwarded, and later arrives back at the same proxy. When it arrives the second time, its Request-URI is identical to the first time, and other header fields that affect proxy operation are unchanged, so that the proxy would make the same processing decision on the request it made the first time. Looped requests are errors, and the procedures for detecting them and handling them are described by the protocol.

Loose Routing: A proxy is said to be loose routing if it follows the procedures defined in this specification for processing of the Route header field. These procedures separate the destination of the request (present in the Request-URI) from the set of proxies that need to be visited along the way (present in the Route header field). A proxy compliant to these mechanisms is also known as a loose router.

Message: Data sent between SIP elements as part of the protocol. SIP messages are either requests or responses.

Method: The method is the primary function that a request is meant to invoke on a server. The method is carried in the request message itself. Example methods are INVITE and BYE.

Outbound Proxy: A proxy that receives requests from a client, even though it may not be the server resolved by the Request-URI. Typically, a UA is manually configured with an outbound proxy, or can learn about one through auto-configuration protocols.

Parallel Search: In a parallel search, a proxy issues several requests to possible user locations upon receiving an incoming request. Rather than issuing one request and then waiting for the final response before issuing the next request as in a sequential search, a parallel search issues requests without waiting for the result of previous requests.

Provisional Response: A response used by the server to indicate progress, but that does not terminate a SIP transaction. 1xx responses are provisional, other responses are considered final.

Proxy, Proxy Server: An intermediary entity that acts as both a server and a client for the purpose of making requests on behalf of other clients. A proxy server primarily plays the role of routing, which means its job is to ensure that a request is sent to another entity "closer" to the targeted user. Proxies are also useful for enforcing policy (for example, making sure a user is allowed to make a call). A proxy interprets, and, if necessary, rewrites specific parts of a request message before forwarding it.

Recursion: A client recurses on a 3xx response when it generates a new request to one or more of the URIs in the Contact header field in the response.

Redirect Server: A redirect server is a user agent server that generates 3xx responses to requests it receives, directing the client to contact an alternate set of URIs.

Registrar: A registrar is a server that accepts REGISTER requests and places the information it receives in those requests into the location service for the domain it handles.

Regular Transaction: A regular transaction is any transaction with a method other than INVITE, ACK, or CANCEL.

Request: A SIP message sent from a client to a server, for the purpose of invoking a particular operation.

Response: A SIP message sent from a server to a client, for indicating the status of a request sent from the client to the server.

Ringback: Ringback is the signaling tone produced by the calling party's application indicating that a called party is being alerted (ringing).

Route Set: A route set is a collection of ordered SIP or SIPS URI which represent a list of proxies that must be traversed when sending a particular request. A route set can be learned, through headers like Record-Route, or it can be configured.

Server: A server is a network element that receives requests in order to service them and sends back responses to those requests. Examples of servers are proxies, user agent servers, redirect servers, and registrars.

Sequential Search: In a sequential search, a proxy server attempts each contact address in sequence, proceeding to the next one only after the previous has generated a final response. A 2xx or 6xx class final response always terminates a sequential search.

Session: From the SDP specification: "A multimedia session is a set of multimedia senders and receivers and the data streams flowing from senders to receivers. A multimedia conference is an example of a multimedia session." (RFC 2327 [1]) (A session as defined for SDP can comprise one or more RTP sessions.) As defined, a callee can be invited several times, by different calls, to the same session. If SDP is used, a session is defined by the concatenation of the SDP user name, session id, network type, address type, and address elements in the origin field.

SIP Transaction: A SIP transaction occurs between a client and a server and comprises all messages from the first request sent from the client to the server up to a final (non-1xx) response sent from the server to the client. If the request is INVITE and the final response is a non-2xx, the transaction also includes an ACK to the response. The ACK for a 2xx response to an INVITE request is a separate transaction.

Spiral: A spiral is a SIP request that is routed to a proxy, forwarded onwards, and arrives once again at that proxy, but this time differs in a way that will result in a different processing decision than the original request. Typically, this means that the request's Request-URI differs from its previous arrival. A spiral is not an error condition, unlike a loop. A typical cause for this is call forwarding. A user calls joe@example.com. The example.com proxy forwards it to Joe's PC, which in turn, forwards it to bob@example.com. This request is proxied back to the example.com proxy. However, this is not a loop. Since the request is targeted at a different user, it is considered a spiral, and is a valid condition.

Stateful Proxy: A logical entity that maintains the client and server transaction state machines defined by this specification during the processing of a request, also known as a transaction stateful proxy. The behavior of a stateful proxy is further defined in Section 16. A (transaction) stateful proxy is not the same as a call stateful proxy.

Stateless Proxy: A logical entity that does not maintain the client or server transaction state machines defined in this specification when it processes requests. A stateless proxy forwards every request it receives downstream and every response it receives upstream.

Strict Routing: A proxy is said to be strict routing if it follows the Route processing rules of RFC 2543 and many prior work in progress versions of this RFC. That rule caused proxies to destroy the contents of the Request-URI when a Route header field was present. Strict routing behavior is not used in this specification, in favor of a loose routing behavior. Proxies that perform strict routing are also known as strict routers.

Target Refresh Request: A target refresh request sent within a dialog is defined as a request that can modify the remote target of the dialog.

Transaction User (TU): The layer of protocol processing that resides above the transaction layer. Transaction users include the UAC core, UAS core, and proxy core.

Upstream: A direction of message forwarding within a transaction that refers to the direction that responses flow from the user agent server back to the user agent client.

URL-encoded: A character string encoded according to RFC 2396, Section 2.4 [5].

User Agent Client (UAC): A user agent client is a logical entity that creates a new request, and then uses the client transaction state machinery to send it. The role of UAC lasts only for the duration of that transaction. In other words, if a piece of software initiates a request, it acts as a UAC for the duration of that transaction. If it receives a request later, it assumes the role of a user agent server for the processing of that transaction.

UAC Core: The set of processing functions required of a UAC that reside above the transaction and transport layers.

User Agent Server (UAS): A user agent server is a logical entity that generates a response to a SIP request. The response accepts, rejects, or redirects the request. This role lasts only for the duration of that transaction. In other words, if a piece of software responds to a request, it acts as a UAS for the duration of that transaction. If it generates a request later, it assumes the role of a user agent client for the processing of that transaction.

UAS Core: The set of processing functions required at a UAS that resides above the transaction and transport layers.

User Agent (UA): A logical entity that can act as both a user agent client and user agent server.

The role of UAC and UAS, as well as proxy and redirect servers, are defined on a transaction-by-transaction basis. For example, the user agent initiating a call acts as a UAC when sending the initial INVITE request and as a UAS when receiving a BYE request from the callee. Similarly, the same software can act as a proxy server for one request and as a redirect server for the next request.

Proxy, location, and registrar servers defined above are logical entities; implementations MAY combine them into a single application.

# 7 SIP Messages

SIP is a text-based protocol and uses the UTF-8 charset (RFC 2279 [7]).

A SIP message is either a request from a client to a server, or a response from a server to a client.

Both Request (section 7.1) and Response (section 7.2) messages use the basic format of RFC 2822 [3], even though the syntax differs in character set and syntax specifics. (SIP allows header fields that would not be valid RFC 2822 header fields, for example.) Both types of messages consist of a start-line, one or more header fields, an empty line indicating the end of the header fields, and an optional message-body.

```
generic-message  =  start-line
                    *message-header
                    CRLF
                    [ message-body ]
start-line       =  Request-Line / Status-Line
```

The start-line, each message-header line, and the empty line MUST be terminated by a carriage-return line-feed sequence (CRLF). Note that the empty line MUST be present even if the message-body is not.

Except for the above difference in character sets, much of SIP's message and header field syntax is identical to HTTP/1.1. Rather than repeating the syntax and semantics here, we use [HX.Y] to refer to Section X.Y of the current HTTP/1.1 specification (RFC 2616 [8]).

However, SIP is not an extension of HTTP.

## 7.1 Requests

SIP requests are distinguished by having a Request-Line for a start-line. A Request-Line contains a method name, a Request-URI, and the protocol version separated by a single space (SP) character.

The Request-Line ends with CRLF. No CR or LF are allowed except in the end-of-line CRLF sequence. No linear whitespace (LWS) is allowed in any of the elements.

```
Request-Line  =  Method SP Request-URI SP SIP-Version CRLF
```

Method: This specification defines six methods: REGISTER for registering contact information, INVITE, ACK, and CANCEL for setting up sessions, BYE for terminating sessions, and OPTIONS for querying servers about their capabilities. SIP extensions, documented in standards track RFCs, may define additional methods.

Request-URI: The Request-URI is a SIP or SIPS URI as described in Section 19.1 or a general URI (RFC 2396 [5]). It indicates the user or service to which this request is being addressed. The Request-URI MUST NOT contain unescaped spaces or control characters and MUST NOT be enclosed in "<>".

SIP elements MAY support Request-URIs with schemes other than "sip" and "sips", for example the "tel" URI scheme of RFC 2806 [9]. SIP elements MAY translate non-SIP URIs using any mechanism at their disposal, resulting in SIP URI, SIPS URI, or some other scheme.

SIP-Version: Both request and response messages include the version of SIP in use, and follow [H3.1] (with HTTP replaced by SIP, and HTTP/1.1 replaced by SIP/2.0) regarding version ordering, compliance requirements, and upgrading of version numbers. To be compliant with this specification, applications sending SIP messages MUST include a SIP-Version of "SIP/2.0". The SIP-Version string is case-insensitive, but implementations MUST send upper-case.

Unlike HTTP/1.1, SIP treats the version number as a literal string. In practice, this should make no difference.

## 7.2 Responses

SIP responses are distinguished from requests by having a Status-Line as their start-line. A Status-Line consists of the protocol version followed by a numeric Status-Code and its associated textual phrase, with each element separated by a single SP character.

No CR or LF is allowed except in the final CRLF sequence.

```
Status-Line  =  SIP-Version SP Status-Code SP Reason-Phrase CRLF
```

The Status-Code is a 3-digit integer result code that indicates the outcome of an attempt to understand and satisfy a request. The Reason-Phrase is intended to give a short textual description of the Status-Code. The Status-Code is intended for use by automata, whereas the Reason-Phrase is intended for the human user. A client is not required to examine or display the Reason-Phrase.

While this specification suggests specific wording for the reason phrase, implementations MAY choose other text, for example, in the language indicated in the Accept-Language header field of the request.

The first digit of the Status-Code defines the class of response. The last two digits do not have any categorization role. For this reason, any response with a status code between 100 and 199 is referred to as a "1xx response", any response with a status code between 200 and 299 as a "2xx response", and so on. SIP/2.0 allows six values for the first digit:

1xx: Provisional -- request received, continuing to process the request;

2xx: Success -- the action was successfully received, understood, and accepted;

3xx: Redirection -- further action needs to be taken in order to complete the request;

4xx: Client Error -- the request contains bad syntax or cannot be fulfilled at this server;

5xx: Server Error -- the server failed to fulfill an apparently valid request;

6xx: Global Failure -- the request cannot be fulfilled at any server.

Section 21 defines these classes and describes the individual codes.

## 7.3 Header Fields

SIP header fields are similar to HTTP header fields in both syntax and semantics. In particular, SIP header fields follow the [H4.2] definitions of syntax for the message-header and the rules for extending header fields over multiple lines. However, the latter is specified in HTTP with implicit whitespace and folding. This specification conforms to RFC 2234 [10] and uses only explicit whitespace and folding as an integral part of the grammar.

[H4.2] also specifies that multiple header fields of the same field name whose value is a comma-separated list can be combined into one header field. That applies to SIP as well, but the specific rule is different because of the different grammars. Specifically, any SIP header whose grammar is of the form

```
header  =  "header-name" HCOLON header-value *(COMMA header-value)
```

allows for combining header fields of the same name into a comma-separated list. The Contact header field allows a comma-separated list unless the header field value is "*".

### 7.3.1 Header Field Format

Header fields follow the same generic header format as that given in Section 2.2 of RFC 2822 [3]. Each header field consists of a field name followed by a colon (":") and the field value.

```
field-name: field-value
```

The formal grammar for a message-header specified in Section 25 allows for an arbitrary amount of whitespace on either side of the colon; however, implementations should avoid spaces between the field name and the colon and use a single space (SP) between the colon and the field-value.

```
Subject:            lunch
Subject      :      lunch
Subject            :lunch
Subject: lunch
```

Thus, the above are all valid and equivalent, but the last is the preferred form.

Header fields can be extended over multiple lines by preceding each extra line with at least one SP or horizontal tab (HT). The line break and the whitespace at the beginning of the next line are treated as a single SP character. Thus, the following are equivalent:

```
Subject: I know you’re there, pick up the phone and talk to me!
Subject: I know you’re there,
         pick up the phone
         and talk to me!
```

The relative order of header fields with different field names is not significant. However, it is RECOMMENDED that header fields which are needed for proxy processing (Via, Route, Record-Route, Proxy-Require, Max-Forwards, and Proxy-Authorization, for example) appear towards the top of the message to facilitate rapid parsing. The relative order of header field rows with the same field name is important. Multiple header field rows with the same field-name MAY be present in a message if and only if the entire field-value for that header field is defined as a comma-separated list (that is, if follows the grammar defined in Section 7.3). It MUST be possible to combine the multiple header field rows into one "field-name: field-value" pair, without changing the semantics of the message, by appending each subsequent field-value to the first, each separated by a comma. The exceptions to this rule are the WWW-Authenticate, Authorization, Proxy-Authenticate, and Proxy-Authorization header fields. Multiple header field rows with these names MAY be present in a message, but since their grammar does not follow the general form listed in Section 7.3, they MUST NOT be combined into a single header field row.

Implementations MUST be able to process multiple header field rows with the same name in any combination of the single-value-per-line or comma-separated value forms.

The following groups of header field rows are valid and equivalent:

```
Route: <sip:alice@atlanta.com>
Subject: Lunch
Route: <sip:bob@biloxi.com>
Route: <sip:carol@chicago.com>

Route: <sip:alice@atlanta.com>, <sip:bob@biloxi.com>
Route: <sip:carol@chicago.com>
Subject: Lunch

Subject: Lunch
Route: <sip:alice@atlanta.com>, <sip:bob@biloxi.com>,
       <sip:carol@chicago.com>
```

Each of the following blocks is valid but not equivalent to the others:

```
Route: <sip:alice@atlanta.com>
Route: <sip:bob@biloxi.com>
Route: <sip:carol@chicago.com>

Route: <sip:bob@biloxi.com>
Route: <sip:alice@atlanta.com>
Route: <sip:carol@chicago.com>

Route: <sip:alice@atlanta.com>,<sip:carol@chicago.com>,
       <sip:bob@biloxi.com>
```

The format of a header field-value is defined per header-name. It will always be either an opaque sequence of TEXT-UTF8 octets, or a combination of whitespace, tokens, separators, and quoted strings. Many existing header fields will adhere to the general form of a value followed by a semi-colon separated sequence of parameter-name, parameter-value pairs:

```
field-name: field-value *(;parameter-name=parameter-value)
```

Even though an arbitrary number of parameter pairs may be attached to a header field value, any given parameter-name MUST NOT appear more than once.

When comparing header fields, field names are always case-insensitive. Unless otherwise stated in the definition of a particular header field, field values, parameter names, and parameter values are case-insensitive. Tokens are always case-insensitive. Unless specified otherwise, values expressed as quoted strings are case-sensitive. For example,

```
Contact: <sip:alice@atlanta.com>;expires=3600
```

is equivalent to

```
CONTACT: <sip:alice@atlanta.com>;ExPiReS=3600
```

and

```
Content-Disposition: session;handling=optional
```

is equivalent to

```
content-disposition: Session;HANDLING=OPTIONAL
```

The following two header fields are not equivalent:

```
Warning: 370 devnull "Choose a bigger pipe"
Warning: 370 devnull "CHOOSE A BIGGER PIPE"
```

### 7.3.2 Header Field Classification

Some header fields only make sense in requests or responses. These are called request header fields and response header fields, respectively. If a header field appears in a message not matching its category (such as a request header field in a response), it MUST be ignored. Section 20 defines the classification of each header field.

### 7.3.3 Compact Form

SIP provides a mechanism to represent common header field names in an abbreviated form. This may be useful when messages would otherwise become too large to be carried on the transport available to it (exceeding the maximum transmission unit (MTU) when using UDP, for example). These compact forms are defined in Section 20. A compact form MAY be substituted for the longer form of a header field name at any time without changing the semantics of the message. A header field name MAY appear in both long and short forms within the same message. Implementations MUST accept both the long and short forms of each header name.

### 7.4 Bodies

Requests, including new requests defined in extensions to this specification, MAY contain message bodies unless otherwise noted. The interpretation of the body depends on the request method.

For response messages, the request method and the response status code determine the type and interpretation of any message body. All responses MAY include a body.

#### 7.4.1 Message Body Type

The Internet media type of the message body MUST be given by the Content-Type header field. If the body has undergone any encoding such as compression, then this MUST be indicated by the Content-Encoding header field; otherwise, Content-Encoding MUST be omitted. If applicable, the character set of the message body is indicated as part of the Content-Type header-field value.

The "multipart" MIME type defined in RFC 2046 [11] MAY be used within the body of the message. Implementations that send requests containing multipart message bodies MUST send a session description as a non-multipart message body if the remote implementation requests this through an Accept header field that does not contain multipart.

SIP messages MAY contain binary bodies or body parts. When no explicit charset parameter is provided by the sender, media subtypes of the "text" type are defined to have a default charset value of "UTF-8".

#### 7.4.2 Message Body Length

The body length in bytes is provided by the Content-Length header field. Section 20.14 describes the necessary contents of this header field in detail.

The "chunked" transfer encoding of HTTP/1.1 MUST NOT be used for SIP. (Note: The chunked encoding modifies the body of a message in order to transfer it as a series of chunks, each with its own size indicator.)

## 7.5 Framing SIP Messages

Unlike HTTP, SIP implementations can use UDP or other unreliable datagram protocols. Each such datagram carries one request or response. See Section 18 on constraints on usage of unreliable transports.

Implementations processing SIP messages over stream-oriented transports MUST ignore any CRLF appearing before the start-line [H4.1].

> The Content-Length header field value is used to locate the end of each SIP message in a stream. It will always be present when SIP messages are sent over stream-oriented transports.

# 8 General User Agent Behavior

A user agent represents an end system. It contains a user agent client (UAC), which generates requests, and a user agent server (UAS), which responds to them. A UAC is capable of generating a request based on some external stimulus (the user clicking a button, or a signal on a PSTN line) and processing a response. A UAS is capable of receiving a request and generating a response based on user input, external stimulus, the result of a program execution, or some other mechanism.

When a UAC sends a request, the request passes through some number of proxy servers, which forward the request towards the UAS. When the UAS generates a response, the response is forwarded towards the UAC.

UAC and UAS procedures depend strongly on two factors. First, based on whether the request or response is inside or outside of a dialog, and second, based on the method of a request. Dialogs are discussed thoroughly in Section 12; they represent a peer-to-peer relationship between user agents and are established by specific SIP methods, such as INVITE.

In this section, we discuss the method-independent rules for UAC and UAS behavior when processing requests that are outside of a dialog. This includes, of course, the requests which themselves establish a dialog.

Security procedures for requests and responses outside of a dialog are described in Section 26. Specifically, mechanisms exist for the UAS and UAC to mutually authenticate. A limited set of privacy features are also supported through encryption of bodies using S/MIME.

## 8.1 UAC Behavior

This section covers UAC behavior outside of a dialog.

### 8.1.1 Generating the Request

A valid SIP request formulated by a UAC MUST, at a minimum, contain the following header fields: To, From, CSeq, Call-ID, Max-Forwards, and Via; all of these header fields are mandatory in all SIP requests. These six header fields are the fundamental building blocks of a SIP message, as they jointly provide for most of the critical message routing services including the addressing of messages, the routing of responses, limiting message propagation, ordering of messages, and the unique identification of transactions. These header fields are in addition to the mandatory request line, which contains the method, Request-URI, and SIP version.

Examples of requests sent outside of a dialog include an INVITE to establish a session (Section 13) and an OPTIONS to query for capabilities (Section 11).

#### 8.1.1.1 Request-URI

The initial Request-URI of the message SHOULD be set to the value of the URI in the To field. One notable exception is the REGISTER method; behavior for setting the Request-URI of REGISTER is given in Section 10. It may also be undesirable for privacy reasons or convenience to set these fields to the same value (especially if the originating UA expects that the Request-URI will be changed during transit).

In some special circumstances, the presence of a pre-existing route set can affect the Request-URI of the message. A pre-existing route set is an ordered set of URIs that identify a chain of servers, to which a UAC will send outgoing requests that are outside of a dialog. Commonly, they are configured on the UA by a user or service provider manually, or through some other non-SIP mechanism. When a provider wishes to configure a UA with an outbound proxy, it is RECOMMENDED that this be done by providing it with a pre-existing route set with a single URI, that of the outbound proxy.

When a pre-existing route set is present, the procedures for populating the Request-URI and Route header field detailed in Section 12.2.1.1 MUST be followed (even though there is no dialog), using the desired Request-URI as the remote target URI.

#### 8.1.1.2 To

The To header field first and foremost specifies the desired "logical" recipient of the request, or the address-of-record of the user or resource that is the target of this request. This may or may not be the ultimate recipient of the request. The To header field MAY contain a SIP or SIPS URI, but it may also make use of other URI schemes (the tel URL (RFC 2806 [9]), for example) when appropriate. All SIP implementations MUST support the SIP URI scheme. Any implementation that supports TLS MUST support the SIPS URI scheme. The To header field allows for a display name.

A UAC may learn how to populate the To header field for a particular request in a number of ways. Usually the user will suggest the To header field through a human interface, perhaps inputting the URI manually or selecting it from some sort of address book. Frequently, the user will not enter a complete URI, but rather a string of digits or letters (for example, "bob"). It is at the discretion of the UA to choose how to interpret this input. Using the string to form the user part of a SIP URI implies that the UA wishes the name to be resolved in the domain to the right-hand side (RHS) of the at-sign in the SIP URI (for instance, sip:bob@example.com). Using the string to form the user part of a SIPS URI implies that the UA wishes to communicate securely, and that the name is to be resolved in the domain to the RHS of the at-sign. The RHS will frequently be the home domain of the requestor, which allows for the home domain to process the outgoing request. This is useful for features like "speed dial" that require interpretation of the user part in the home domain. The tel URL may be used when the UA does not wish to specify the domain that should interpret a telephone number that has been input by the user. Rather, each domain through which the request passes would be given that opportunity. As an example, a user in an airport might log in and send requests through an outbound proxy in the airport. If they enter "411" (this is the phone number for local directory assistance in the United States), that needs to be interpreted and processed by the outbound proxy in the airport, not the user's home domain. In this case, tel:411 would be the right choice.

A request outside of a dialog MUST NOT contain a To tag; the tag in the To field of a request identifies the peer of the dialog. Since no dialog is established, no tag is present.

For further information on the To header field, see Section 20.39. The following is an example of a valid To header field:

```
To: Carol <sip:carol@chicago.com>
```

#### 8.1.1.3 From

The From header field indicates the logical identity of the initiator of the request, possibly the user's address-of-record. Like the To header field, it contains a URI and optionally a display name. It is used by SIP elements to determine which processing rules to apply to a request (for example, automatic call rejection). As such, it is very important that the From URI not contain IP addresses or the FQDN of the host on which the UA is running, since these are not logical names.

The From header field allows for a display name. A UAC SHOULD use the display name "Anonymous", along with a syntactically correct, but otherwise meaningless URI (like sip:thisis@anonymous.invalid), if the identity of the client is to remain hidden.

Usually, the value that populates the From header field in requests generated by a particular UA is pre-provisioned by the user or by the administrators of the user's local domain. If a particular UA is used by multiple users, it might have switchable profiles that include a URI corresponding to the identity of the profiled user. Recipients of requests can authenticate the originator of a request in order to ascertain that they are who their From header field claims they are (see Section 22 for more on authentication).

The From field MUST contain a new "tag" parameter, chosen by the UAC. See Section 19.3 for details on choosing a tag.

For further information on the From header field, see Section 20.20. Examples:

```
From: "Bob" <sips:bob@biloxi.com> ;tag=a48s
From: sip:+12125551212@phone2net.com;tag=887s
From: Anonymous <sip:c8oqz84zk7z@privacy.org>;tag=hyh8
```

#### 8.1.1.4 Call-ID

The Call-ID header field acts as a unique identifier to group together a series of messages. It MUST be the same for all requests and responses sent by either UA in a dialog. It SHOULD be the same in each registration from a UA.

In a new request created by a UAC outside of any dialog, the Call-ID header field MUST be selected by the UAC as a globally unique identifier over space and time unless overridden by method-specific behavior. All SIP UAs must have a means to guarantee that the Call-ID header fields they produce will not be inadvertently generated by any other UA. Note that when requests are retried after certain failure responses that solicit an amendment to a request (for example, a challenge for authentication), these retried requests are not considered new requests, and therefore do not need new Call-ID header fields; see Section 8.1.3.5.

Use of cryptographically random identifiers (RFC 1750 [12]) in the generation of Call-IDs is RECOMMENDED. Implementations MAY use the form "localid@host". Call-IDs are case-sensitive and are simply compared byte-by-byte.

> Using cryptographically random identifiers provides some protection against session hijacking and reduces the likelihood of unintentional Call-ID collisions.

No provisioning or human interface is required for the selection of the Call-ID header field value for a request.

For further information on the Call-ID header field, see Section 20.8.

Example:

```
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@foo.bar.com
```

#### 8.1.1.5 CSeq

The CSeq header field serves as a way to identify and order transactions. It consists of a sequence number and a method. The method MUST match that of the request. For non-REGISTER requests outside of a dialog, the sequence number value is arbitrary. The sequence number value MUST be expressible as a 32-bit unsigned integer and MUST be less than 2**31. As long as it follows the above guidelines, a client may use any mechanism it would like to select CSeq header field values.

Section 12.2.1.1 discusses construction of the CSeq for requests within a dialog.

Example:

```
CSeq: 4711 INVITE
```

#### 8.1.1.6 Max-Forwards

The Max-Forwards header field serves to limit the number of hops a request can transit on the way to its destination. It consists of an integer that is decremented by one at each hop. If the Max-Forwards value reaches 0 before the request reaches its destination, it will be rejected with a 483(Too Many Hops) error response.

A UAC MUST insert a Max-Forwards header field into each request it originates with a value that SHOULD be 70. This number was chosen to be sufficiently large to guarantee that a request would not be dropped in any SIP network when there were no loops, but not so large as to consume proxy resources when a loop does occur. Lower values should be used with caution and only in networks where topologies are known by the UA.

#### 8.1.1.7 Via

The Via header field indicates the transport used for the transaction and identifies the location where the response is to be sent. A Via header field value is added only after the transport that will be used to reach the next hop has been selected (which may involve the usage of the procedures in [4]).

When the UAC creates a request, it MUST insert a Via into that request. The protocol name and protocol version in the header field MUST be SIP and 2.0, respectively. The Via header field value MUST contain a branch parameter. This parameter is used to identify the transaction created by that request. This parameter is used by both the client and the server.

The branch parameter value MUST be unique across space and time for all requests sent by the UA. The exceptions to this rule are CANCEL and ACK for non-2xx responses. As discussed below, a CANCEL request will have the same value of the branch parameter as the request it cancels. As discussed in Section 17.1.1.3, an ACK for a non-2xx response will also have the same branch ID as the INVITE whose response it acknowledges.

> The uniqueness property of the branch ID parameter, to facilitate its use as a transaction ID, was not part of RFC 2543.

The branch ID inserted by an element compliant with this specification MUST always begin with the characters "z9hG4bK". These 7 characters are used as a magic cookie (7 is deemed sufficient to ensure that an older RFC 2543 implementation would not pick such a value), so that servers receiving the request can determine that the branch ID was constructed in the fashion described by this specification (that is, globally unique). Beyond this requirement, the precise format of the branch token is implementation-defined.

The Via header maddr, ttl, and sent-by components will be set when the request is processed by the transport layer (Section 18).

Via processing for proxies is described in Section 16.6 Item 8 and Section 16.7 Item 3.

#### 8.1.1.8 Contact

The Contact header field provides a SIP or SIPS URI that can be used to contact that specific instance of the UA for subsequent requests. The Contact header field MUST be present and contain exactly one SIP or SIPS URI in any request that can result in the establishment of a dialog. For the methods defined in this specification, that includes only the INVITE request. For these requests, the scope of the Contact is global. That is, the Contact header field value contains the URI at which the UA would like to receive requests, and this URI MUST be valid even if used in subsequent requests outside of any dialogs.

If the Request-URI or top Route header field value contains a SIPS URI, the Contact header field MUST contain a SIPS URI as well.

For further information on the Contact header field, see Section 20.10.

#### 8.1.1.9 Supported and Require

If the UAC supports extensions to SIP that can be applied by the server to the response, the UAC SHOULD include a Supported header field in the request listing the option tags (Section 19.2) for those extensions.

The option tags listed MUST only refer to extensions defined in standards-track RFCs. This is to prevent servers from insisting that clients implement non-standard, vendor-defined features in order to receive service. Extensions defined by experimental and informational RFCs are explicitly excluded from usage with the Supported header field in a request, since they too are often used to document vendor-defined extensions.

If the UAC wishes to insist that a UAS understand an extension that the UAC will apply to the request in order to process the request, it MUST insert a Require header field into the request listing the option tag for that extension. If the UAC wishes to apply an extension to the request and insist that any proxies that are traversed understand that extension, it MUST insert a Proxy-Require header field into the request listing the option tag for that extension.

As with the Supported header field, the option tags in the Require and Proxy-Require header fields MUST only refer to extensions defined in standards-track RFCs.

#### 8.1.1.10 Additional Message Components

After a new request has been created, and the header fields described above have been properly constructed, any additional optional header fields are added, as are any header fields specific to the method.

SIP requests MAY contain a MIME-encoded message-body. Regardless of the type of body that a request contains, certain header fields must be formulated to characterize the contents of the body. For further information on these header fields, see Sections 20.11 through 20.15.

### 8.1.2 Sending the Request

The destination for the request is then computed. Unless there is local policy specifying otherwise, the destination MUST be determined by applying the DNS procedures described in [4] as follows. If the first element in the route set indicated a strict router (resulting in forming the request as described in Section 12.2.1.1), the procedures MUST be applied to the Request-URI of the request. Otherwise, the procedures are applied to the first Route header field value in the request (if one exists), or to the request's Request-URI if there is no Route header field present. These procedures yield an ordered set of address, port, and transports to attempt. Independent of which URI is used as input to the procedures of [4], if the Request-URI specifies a SIPS resource, the UAC MUST follow the procedures of [4] as if the input URI were a SIPS URI.

Local policy MAY specify an alternate set of destinations to attempt. If the Request-URI contains a SIPS URI, any alternate destinations MUST be contacted with TLS. Beyond that, there are no restrictions on the alternate destinations if the request contains no Route header field. This provides a simple alternative to a pre-existing route set as a way to specify an outbound proxy. However, that approach for configuring an outbound proxy is NOT RECOMMENDED; a pre-existing route set with a single URI SHOULD be used instead. If the request contains a Route header field, the request SHOULD be sent to the locations derived from its topmost value, but MAY be sent to any server that the UA is certain will honor the Route and Request-URI policies specified in this document (as opposed to those in RFC 2543). In particular, a UAC configured with an outbound proxy SHOULD attempt to send the request to the location indicated in the first Route header field value instead of adopting the policy of sending all messages to the outbound proxy.

> This ensures that outbound proxies that do not add Record-Route header field values will drop out of the path of subsequent requests. It allows endpoints that cannot resolve the first Route URI to delegate that task to an outbound proxy.

The UAC SHOULD follow the procedures defined in [4] for stateful elements, trying each address until a server is contacted. Each try constitutes a new transaction, and therefore each carries a different topmost Via header field value with a new branch parameter. Furthermore, the transport value in the Via header field is set to whatever transport was determined for the target server.

### 8.1.3 Processing Responses

Responses are first processed by the transport layer and then passed up to the transaction layer. The transaction layer performs its processing and then passes the response up to the TU. The majority of response processing in the TU is method specific. However, there are some general behaviors independent of the method.

#### 8.1.3.1 Transaction Layer Errors

In some cases, the response returned by the transaction layer will not be a SIP message, but rather a transaction layer error. When a timeout error is received from the transaction layer, it MUST be treated as if a 408 (Request Timeout) status code has been received. If a fatal transport error is reported by the transport layer (generally, due to fatal ICMP errors in UDP or connection failures in TCP), the condition MUST be treated as a 503 (Service Unavailable) status code.

#### 8.1.3.2 Unrecognized Responses

A UAC MUST treat any final response it does not recognize as being equivalent to the x00 response code of that class, and MUST be able to process the x00 response code for all classes. For example, if a UAC receives an unrecognized response code of 431, it can safely assume that there was something wrong with its request and treat the response as if it had received a 400 (Bad Request) response code. A UAC MUST treat any provisional response different than 100 that it does not recognize as 183 (Session Progress). A UAC MUST be able to process 100 and 183 responses.

#### 8.1.3.3 Vias

If more than one Via header field value is present in a response, the UAC SHOULD discard the message.

> The presence of additional Via header field values that precede the originator of the request suggests that the message was misrouted or possibly corrupted.

#### 8.1.3.4 Processing 3xx Responses

Upon receipt of a redirection response (for example, a 301 response status code), clients SHOULD use the URI(s) in the Contact header field to formulate one or more new requests based on the redirected request. This process is similar to that of a proxy recursing on a 3xx class response as detailed in Sections 16.5 and 16.6. A client starts with an initial target set containing exactly one URI, the Request-URI of the original request. If a client wishes to formulate new requests based on a 3xx class response to that request, it places the URIs to try into the target set. Subject to the restrictions in this specification, a client can choose which Contact URIs it places into the target set. As with proxy recursion, a client processing 3xx class responses MUST NOT add any given URI to the target set more than once. If the original request had a SIPS URI in the Request-URI, the client MAY choose to recurse to a non-SIPS URI, but SHOULD inform the user of the redirection to an insecure URI.

> Any new request may receive 3xx responses themselves containing the original URI as a contact. Two locations can be configured to redirect to each other. Placing any given URI in the target set only once prevents infinite redirection loops.

As the target set grows, the client MAY generate new requests to the URIs in any order. A common mechanism is to order the set by the "q" parameter value from the Contact header field value. Requests to the URIs MAY be generated serially or in parallel. One approach is to process groups of decreasing q-values serially and process the URIs in each q-value group in parallel. Another is to perform only serial processing in decreasing q-value order, arbitrarily choosing between contacts of equal q-value.

If contacting an address in the list results in a failure, as defined in the next paragraph, the element moves to the next address in the list, until the list is exhausted. If the list is exhausted, then the request has failed.

Failures SHOULD be detected through failure response codes (codes greater than 399); for network errors the client transaction will report any transport layer failures to the transaction user. Note that some response codes (detailed in 8.1.3.5) indicate that the request can be retried; requests that are reattempted should not be considered failures.

When a failure for a particular contact address is received, the client SHOULD try the next contact address. This will involve creating a new client transaction to deliver a new request.

In order to create a request based on a contact address in a 3xx response, a UAC MUST copy the entire URI from the target set into the Request-URI, except for the "method-param" and "header" URI parameters (see Section 19.1.1 for a definition of these parameters). It uses the "header" parameters to create header field values for the new request, overwriting header field values associated with the redirected request in accordance with the guidelines in Section 19.1.5.

Note that in some instances, header fields that have been communicated in the contact address may instead append to existing request header fields in the original redirected request. As a general rule, if the header field can accept a comma-separated list of values, then the new header field value MAY be appended to any existing values in the original redirected request. If the header field does not accept multiple values, the value in the original redirected request MAY be overwritten by the header field value communicated in the contact address. For example, if a contact address is returned with the following value:

```
sip:user@host?Subject=foo&Call-Info=<http://www.foo.com>
```

Then any Subject header field in the original redirected request is overwritten, but the HTTP URL is merely appended to any existing Call-Info header field values.

It is RECOMMENDED that the UAC reuse the same To, From, and Call-ID used in the original redirected request, but the UAC MAY also choose to update the Call-ID header field value for new requests, for example.

Finally, once the new request has been constructed, it is sent using a new client transaction, and therefore MUST have a new branch ID in the top Via field as discussed in Section 8.1.1.7.

In all other respects, requests sent upon receipt of a redirect response SHOULD re-use the header fields and bodies of the original request.

In some instances, Contact header field values may be cached at UAC temporarily or permanently depending on the status code received and the presence of an expiration interval; see Sections 21.3.2 and 21.3.3.

#### 8.1.3.5 Processing 4xx Responses

Certain 4xx response codes require specific UA processing, independent of the method.

If a 401 (Unauthorized) or 407 (Proxy Authentication Required) response is received, the UAC SHOULD follow the authorization procedures of Section 22.2 and Section 22.3 to retry the request with credentials.

If a 413 (Request Entity Too Large) response is received (Section 21.4.11), the request contained a body that was longer than the UAS was willing to accept. If possible, the UAC SHOULD retry the request, either omitting the body or using one of a smaller length.

If a 415 (Unsupported Media Type) response is received (Section 21.4.13), the request contained media types not supported by the UAS. The UAC SHOULD retry sending the request, this time only using content with types listed in the Accept header field in the response, with encodings listed in the Accept-Encoding header field in the response, and with languages listed in the Accept-Language in the response.

If a 416 (Unsupported URI Scheme) response is received (Section 21.4.14), the Request-URI used a URI scheme not supported by the server. The client SHOULD retry the request, this time, using a SIP URI.

If a 420 (Bad Extension) response is received (Section 21.4.15), the request contained a Require or Proxy-Require header field listing an option-tag for a feature not supported by a proxy or UAS. The UAC SHOULD retry the request, this time omitting any extensions listed in the Unsupported header field in the response.

In all of the above cases, the request is retried by creating a new request with the appropriate modifications. This new request constitutes a new transaction and SHOULD have the same value of the Call-ID, To, and From of the previous request, but the CSeq should contain a new sequence number that is one higher than the previous.

With other 4xx responses, including those yet to be defined, a retry may or may not be possible depending on the method and the use case.

## 8.2 UAS Behavior

When a request outside of a dialog is processed by a UAS, there is a set of processing rules that are followed, independent of the method. Section 12 gives guidance on how a UAS can tell whether a request is inside or outside of a dialog.

Note that request processing is atomic. If a request is accepted, all state changes associated with it MUST be performed. If it is rejected, all state changes MUST NOT be performed.

UASs SHOULD process the requests in the order of the steps that follow in this section (that is, starting with authentication, then inspecting the method, the header fields, and so on throughout the remainder of this section).

### 8.2.1 Method Inspection

Once a request is authenticated (or authentication is skipped), the UAS MUST inspect the method of the request. If the UAS recognizes but does not support the method of a request, it MUST generate a 405 (Method Not Allowed) response. Procedures for generating responses are described in Section 8.2.6. The UAS MUST also add an Allow header field to the 405 (Method Not Allowed) response. The Allow header field MUST list the set of methods supported by the UAS generating the message. The Allow header field is presented in Section 20.5.

If the method is one supported by the server, processing continues.

### 8.2.2 Header Inspection

If a UAS does not understand a header field in a request (that is, the header field is not defined in this specification or in any supported extension), the server MUST ignore that header field and continue processing the message. A UAS SHOULD ignore any malformed header fields that are not necessary for processing requests.

#### 8.2.2.1 To and Request-URI

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. A UAS MAY apply any policy it wishes to determine whether to accept requests when the To header field is not the identity of the UAS. However, it is RECOMMENDED that a UAS accept requests even if they do not recognize the URI scheme (for example, a tel: URI) in the To header field, or if the To header field does not address a known or current user of this UAS. If, on the other hand, the UAS decides to reject the request, it SHOULD generate a response with a 403 (Forbidden) status code and pass it to the server transaction for transmission.

However, the Request-URI identifies the UAS that is to process the request. If the Request-URI uses a scheme not supported by the UAS, it SHOULD reject the request with a 416 (Unsupported URI Scheme) response. If the Request-URI does not identify an address that the UAS is willing to accept requests for, it SHOULD reject the request with a 404 (Not Found) response. Typically, a UA that uses the REGISTER method to bind its address-of-record to a specific contact address will see requests whose Request-URI equals that contact address. Other potential sources of received Request-URIs include the Contact header fields of requests and responses sent by the UA that establish or refresh dialogs.

#### 8.2.2.2 Merged Requests

If the request has no tag in the To header field, the UAS core MUST check the request against ongoing transactions. If the From tag, Call-ID, and CSeq exactly match those associated with an ongoing transaction, but the request does not match that transaction (based on the matching rules in Section 17.2.3), the UAS core SHOULD generate a 482 (Loop Detected) response and pass it to the server transaction.

> The same request has arrived at the UAS more than once, following different paths, most likely due to forking. The UAS processes the first such request received and responds with a 482 (Loop Detected) to the rest of them.

#### 8.2.2.3 Require

Assuming the UAS decides that it is the proper element to process the request, it examines the Require header field, if present.

The Require header field is used by a UAC to tell a UAS about SIP extensions that the UAC expects the UAS to support in order to process the request properly. Its format is described in Section 20.32. If a UAS does not understand an option-tag listed in a Require header field, it MUST respond by generating a response with status code 420 (Bad Extension). The UAS MUST add an Unsupported header field, and list in it those options it does not understand amongst those in the Require header field of the request.

Note that Require and Proxy-Require MUST NOT be used in a SIP CANCEL request, or in an ACK request sent for a non-2xx response. These header fields MUST be ignored if they are present in these requests.

An ACK request for a 2xx response MUST contain only those Require and Proxy-Require values that were present in the initial request.

Example:

```
UAC->UAS:   INVITE sip:watson@bell-telephone.com SIP/2.0
            Require: 100rel

UAS->UAC:   SIP/2.0 420 Bad Extension
            Unsupported: 100rel
```

This behavior ensures that the client-server interaction will proceed without delay when all options are understood by both sides, and only slow down if options are not understood (as in the example above). For a well-matched client-server pair, the interaction proceeds quickly, saving a round-trip often required by negotiation mechanisms. In addition, it also removes ambiguity when the client requires features that the server does not understand. Some features, such as call handling fields, are only of interest to end systems.

### 8.2.3 Content Processing

Assuming the UAS understands any extensions required by the client, the UAS examines the body of the message, and the header fields that describe it. If there are any bodies whose type (indicated by the Content-Type), language (indicated by the Content-Language) or encoding (indicated by the Content-Encoding) are not understood, and that body part is not optional (as indicated by the Content-Disposition header field), the UAS MUST reject the request with a 415 (Unsupported Media Type) response. The response MUST contain an Accept header field listing the types of all bodies it understands, in the event the request contained bodies of types not supported by the UAS. If the request contained content encodings not understood by the UAS, the response MUST contain an Accept-Encoding header field listing the encodings understood by the UAS. If the request contained content with languages not understood by the UAS, the response MUST contain an Accept-Language header field indicating the languages understood by the UAS. Beyond these checks, body handling depends on the method and type. For further information on the processing of content-specific header fields, see Section 7.4 as well as Section 20.11 through 20.15.

### 8.2.4 Applying Extensions

A UAS that wishes to apply some extension when generating the response MUST NOT do so unless support for that extension is indicated in the Supported header field in the request. If the desired extension is not supported, the server SHOULD rely only on baseline SIP and any other extensions supported by the client. In rare circumstances, where the server cannot process the request without the extension, the server MAY send a 421 (Extension Required) response. This response indicates that the proper response cannot be generated without support of a specific extension. The needed extension(s) MUST be included in a Require header field in the response. This behavior is NOT RECOMMENDED, as it will generally break interoperability.

Any extensions applied to a non-421 response MUST be listed in a Require header field included in the response. Of course, the server MUST NOT apply extensions not listed in the Supported header field in the request. As a result of this, the Require header field in a response will only ever contain option tags defined in standards-track RFCs.

### 8.2.5 Processing the Request

Assuming all of the checks in the previous subsections are passed, the UAS processing becomes method-specific. Section 10 covers the REGISTER request, Section 11 covers the OPTIONS request, Section 13 covers the INVITE request, and Section 15 covers the BYE request.

### 8.2.6 Generating the Response

When a UAS wishes to construct a response to a request, it follows the general procedures detailed in the following subsections. Additional behaviors specific to the response code in question, which are not detailed in this section, may also be required.

Once all procedures associated with the creation of a response have been completed, the UAS hands the response back to the server transaction from which it received the request.

#### 8.2.6.1 Sending a Provisional Response

One largely non-method-specific guideline for the generation of responses is that UASs SHOULD NOT issue a provisional response for a non-INVITE request. Rather, UASs SHOULD generate a final response to a non-INVITE request as soon as possible.

When a 100 (Trying) response is generated, any Timestamp header field present in the request MUST be copied into this 100 (Trying) response. If there is a delay in generating the response, the UAS SHOULD add a delay value into the Timestamp value in the response. This value MUST contain the difference between the time of sending of the response and receipt of the request, measured in seconds.

#### 8.2.6.2 Headers and Tags

The From field of the response MUST equal the From header field of the request. The Call-ID header field of the response MUST equal the Call-ID header field of the request. The CSeq header field of the response MUST equal the CSeq field of the request. The Via header field values in the response MUST equal the Via header field values in the request and MUST maintain the same ordering.

If a request contained a To tag in the request, the To header field in the response MUST equal that of the request. However, if the To header field in the request did not contain a tag, the URI in the To header field in the response MUST equal the URI in the To header field; additionally, the UAS MUST add a tag to the To header field in the response (with the exception of the 100 (Trying) response, in which a tag MAY be present). This serves to identify the UAS that is responding, possibly resulting in a component of a dialog ID. The same tag MUST be used for all responses to that request, both final and provisional (again excepting the 100 (Trying)). Procedures for the generation of tags are defined in Section 19.3.

### 8.2.7 Stateless UAS Behavior

A stateless UAS is a UAS that does not maintain transaction state. It replies to requests normally, but discards any state that would ordinarily be retained by a UAS after a response has been sent. If a stateless UAS receives a retransmission of a request, it regenerates the response and resends it, just as if it were replying to the first instance of the request. A UAS cannot be stateless unless the request processing for that method would always result in the same response if the requests are identical. This rules out stateless registrars, for example. Stateless UASs do not use a transaction layer; they receive requests directly from the transport layer and send responses directly to the transport layer.

The stateless UAS role is needed primarily to handle unauthenticated requests for which a challenge response is issued. If unauthenticated requests were handled statefully, then malicious floods of unauthenticated requests could create massive amounts of transaction state that might slow or completely halt call processing in a UAS, effectively creating a denial of service condition; for more information see Section 26.1.5.

The most important behaviors of a stateless UAS are the following:

- o A stateless UAS MUST NOT send provisional (1xx) responses.
- o A stateless UAS MUST NOT retransmit responses.
- o A stateless UAS MUST ignore ACK requests.
- o A stateless UAS MUST ignore CANCEL requests.
- o To header tags MUST be generated for responses in a stateless manner - in a manner that will generate the same tag for the same request consistently. For information on tag construction see Section 19.3.

In all other respects, a stateless UAS behaves in the same manner as a stateful UAS. A UAS can operate in either a stateful or stateless mode for each new request.

## 8.3 Redirect Servers

In some architectures it may be desirable to reduce the processing load on proxy servers that are responsible for routing requests, and improve signaling path robustness, by relying on redirection.

Redirection allows servers to push routing information for a request back in a response to the client, thereby taking themselves out of the loop of further messaging for this transaction while still aiding in locating the target of the request. When the originator of the request receives the redirection, it will send a new request based on the URI(s) it has received. By propagating URIs from the core of the network to its edges, redirection allows for considerable network scalability.

A redirect server is logically constituted of a server transaction layer and a transaction user that has access to a location service of some kind (see Section 10 for more on registrars and location services). This location service is effectively a database containing mappings between a single URI and a set of one or more alternative locations at which the target of that URI can be found.

A redirect server does not issue any SIP requests of its own. After receiving a request other than CANCEL, the server either refuses the request or gathers the list of alternative locations from the location service and returns a final response of class 3xx. For well-formed CANCEL requests, it SHOULD return a 2xx response. This response ends the SIP transaction. The redirect server maintains transaction state for an entire SIP transaction. It is the responsibility of clients to detect forwarding loops between redirect servers.

When a redirect server returns a 3xx response to a request, it populates the list of (one or more) alternative locations into the Contact header field. An "expires" parameter to the Contact header field values may also be supplied to indicate the lifetime of the Contact data.

The Contact header field contains URIs giving the new locations or user names to try, or may simply specify additional transport parameters. A 301 (Moved Permanently) or 302 (Moved Temporarily) response may also give the same location and username that was targeted by the initial request but specify additional transport parameters such as a different server or multicast address to try, or a change of SIP transport from UDP to TCP or vice versa.

However, redirect servers MUST NOT redirect a request to a URI equal to the one in the Request-URI; instead, provided that the URI does not point to itself, the server MAY proxy the request to the destination URI, or MAY reject it with a 404.

> If a client is using an outbound proxy, and that proxy actually redirects requests, a potential arises for infinite redirection loops.

Note that a Contact header field value MAY also refer to a different resource than the one originally called. For example, a SIP call connected to PSTN gateway may need to deliver a special informational announcement such as "The number you have dialed has been changed."

A Contact response header field can contain any suitable URI indicating where the called party can be reached, not limited to SIP URIs. For example, it could contain URIs for phones, fax, or irc (if they were defined) or a mailto: (RFC 2368 [32]) URL. Section 26.4.4 discusses implications and limitations of redirecting a SIPS URI to a non-SIPS URI.

The "expires" parameter of a Contact header field value indicates how long the URI is valid. The value of the parameter is a number indicating seconds. If this parameter is not provided, the value of the Expires header field determines how long the URI is valid. Malformed values SHOULD be treated as equivalent to 3600.

This provides a modest level of backwards compatibility with RFC 2543, which allowed absolute times in this header field. If an absolute time is received, it will be treated as malformed, and then default to 3600.

Redirect servers MUST ignore features that are not understood (including unrecognized header fields, any unknown option tags in Require, or even method names) and proceed with the redirection of the request in question.

# 9 Canceling a Request

The previous section has discussed general UA behavior for generating requests and processing responses for requests of all methods. In this section, we discuss a general purpose method, called CANCEL.

The CANCEL request, as the name implies, is used to cancel a previous request sent by a client. Specifically, it asks the UAS to cease processing the request and to generate an error response to that request. CANCEL has no effect on a request to which a UAS has already given a final response. Because of this, it is most useful to CANCEL requests to which it can take a server long time to respond. For this reason, CANCEL is best for INVITE requests, which can take a long time to generate a response. In that usage, a UAS that receives a CANCEL request for an INVITE, but has not yet sent a final response, would "stop ringing", and then respond to the INVITE with a specific error response (a 487).

CANCEL requests can be constructed and sent by both proxies and user agent clients. Section 15 discusses under what conditions a UAC would CANCEL an INVITE request, and Section 16.10 discusses proxy usage of CANCEL.

A stateful proxy responds to a CANCEL, rather than simply forwarding a response it would receive from a downstream element. For that reason, CANCEL is referred to as a "hop-by-hop" request, since it is responded to at each stateful proxy hop.

## 9.1 Client Behavior

A CANCEL request SHOULD NOT be sent to cancel a request other than INVITE.

Since requests other than INVITE are responded to immediately, sending a CANCEL for a non-INVITE request would always create a race condition.

The following procedures are used to construct a CANCEL request. The Request-URI, Call-ID, To, the numeric part of CSeq, and From header fields in the CANCEL request MUST be identical to those in the request being cancelled, including tags. A CANCEL constructed by a client MUST have only a single Via header field value matching the top Via value in the request being cancelled. Using the same values for these header fields allows the CANCEL to be matched with the request it cancels (Section 9.2 indicates how such matching occurs). However, the method part of the CSeq header field MUST have a value of CANCEL. This allows it to be identified and processed as a transaction in its own right (See Section 17).

If the request being cancelled contains a Route header field, the CANCEL request MUST include that Route header field's values.

> This is needed so that stateless proxies are able to route CANCEL requests properly.

The CANCEL request MUST NOT contain any Require or Proxy-Require header fields.

Once the CANCEL is constructed, the client SHOULD check whether it has received any response (provisional or final) for the request being cancelled (herein referred to as the "original request").

If no provisional response has been received, the CANCEL request MUST NOT be sent; rather, the client MUST wait for the arrival of a provisional response before sending the request. If the original request has generated a final response, the CANCEL SHOULD NOT be sent, as it is an effective no-op, since CANCEL has no effect on requests that have already generated a final response. When the client decides to send the CANCEL, it creates a client transaction for the CANCEL and passes it the CANCEL request along with the destination address, port, and transport. The destination address, port, and transport for the CANCEL MUST be identical to those used to send the original request.

> If it was allowed to send the CANCEL before receiving a response for the previous request, the server could receive the CANCEL before the original request.

Note that both the transaction corresponding to the original request and the CANCEL transaction will complete independently. However, a UAC canceling a request cannot rely on receiving a 487 (Request Terminated) response for the original request, as an RFC 2543-compliant UAS will not generate such a response. If there is no final response for the original request in 64*T1 seconds (T1 is defined in Section 17.1.1.1), the client SHOULD then consider the original transaction cancelled and SHOULD destroy the client transaction handling the original request.

## 9.2 Server Behavior

The CANCEL method requests that the TU at the server side cancel a pending transaction. The TU determines the transaction to be cancelled by taking the CANCEL request, and then assuming that the request method is anything but CANCEL or ACK and applying the transaction matching procedures of Section 17.2.3. The matching transaction is the one to be cancelled.

The processing of a CANCEL request at a server depends on the type of server. A stateless proxy will forward it, a stateful proxy might respond to it and generate some CANCEL requests of its own, and a UAS will respond to it. See Section 16.10 for proxy treatment of CANCEL.

A UAS first processes the CANCEL request according to the general UAS processing described in Section 8.2. However, since CANCEL requests are hop-by-hop and cannot be resubmitted, they cannot be challenged by the server in order to get proper credentials in an Authorization header field. Note also that CANCEL requests do not contain a Require header field.

If the UAS did not find a matching transaction for the CANCEL according to the procedure above, it SHOULD respond to the CANCEL with a 481 (Call Leg/Transaction Does Not Exist). If the transaction for the original request still exists, the behavior of the UAS on receiving a CANCEL request depends on whether it has already sent a final response for the original request. If it has, the CANCEL request has no effect on the processing of the original request, no effect on any session state, and no effect on the responses generated for the original request. If the UAS has not issued a final response for the original request, its behavior depends on the method of the original request. If the original request was an INVITE, the UAS SHOULD immediately respond to the INVITE with a 487 (Request Terminated). A CANCEL request has no impact on the processing of transactions with any other method defined in this specification.

Regardless of the method of the original request, as long as the CANCEL matched an existing transaction, the UAS answers the CANCEL request itself with a 200 (OK) response. This response is constructed following the procedures described in Section 8.2.6 noting that the To tag of the response to the CANCEL and the To tag in the response to the original request SHOULD be the same. The response to CANCEL is passed to the server transaction for transmission.

# 10 Registrations

## 10.1 Overview

SIP offers a discovery capability. If a user wants to initiate a session with another user, SIP must discover the current host(s) at which the destination user is reachable. This discovery process is frequently accomplished by SIP network elements such as proxy servers and redirect servers which are responsible for receiving a request, determining where to send it based on knowledge of the location of the user, and then sending it there. To do this, SIP network elements consult an abstract service known as a location service, which provides address bindings for a particular domain. These address bindings map an incoming SIP or SIPS URI, sip:bob@biloxi.com, for example, to one or more URIs that are somehow "closer" to the desired user, sip:bob@engineering.biloxi.com, for example. Ultimately, a proxy will consult a location service that maps a received URI to the user agent(s) at which the desired recipient is currently residing.

Registration creates bindings in a location service for a particular domain that associates an address-of-record URI with one or more contact addresses. Thus, when a proxy for that domain receives a request whose Request-URI matches the address-of-record, the proxy will forward the request to the contact addresses registered to that address-of-record. Generally, it only makes sense to register an address-of-record at a domain's location service when requests for that address-of-record would be routed to that domain. In most cases, this means that the domain of the registration will need to match the domain in the URI of the address-of-record.

There are many ways by which the contents of the location service can be established. One way is administratively. In the above example, Bob is known to be a member of the engineering department through access to a corporate database. However, SIP provides a mechanism for a UA to create a binding explicitly. This mechanism is known as registration.

Registration entails sending a REGISTER request to a special type of UAS known as a registrar. A registrar acts as the front end to the location service for a domain, reading and writing mappings based on the contents of REGISTER requests. This location service is then typically consulted by a proxy server that is responsible for routing requests for that domain.

An illustration of the overall registration process is given in Figure 2. Note that the registrar and proxy server are logical roles that can be played by a single device in a network; for purposes of clarity the two are separated in this illustration. Also note that UAs may send requests through a proxy server in order to reach a registrar if the two are separate elements.

SIP does not mandate a particular mechanism for implementing the location service. The only requirement is that a registrar for some domain MUST be able to read and write data to the location service, and a proxy or a redirect server for that domain MUST be capable of reading that same data. A registrar MAY be co-located with a particular SIP proxy server for the same domain.

### 10.2 Constructing the REGISTER Request

REGISTER requests add, remove, and query bindings. A REGISTER request can add a new binding between an address-of-record and one or more contact addresses. Registration on behalf of a particular address-of-record can be performed by a suitably authorized third party. A client can also remove previous bindings or query to determine which bindings are currently in place for an address-of-record.

Except as noted, the construction of the REGISTER request and the behavior of clients sending a REGISTER request is identical to the general UAC behavior described in Section 8.1 and Section 17.1.

A REGISTER request does not establish a dialog. A UAC MAY include a Route header field in a REGISTER request based on a pre-existing route set as described in Section 8.1. The Record-Route header field has no meaning in REGISTER requests or responses, and MUST be ignored if present. In particular, the UAC MUST NOT create a new route set based on the presence or absence of a Record-Route header field in any response to a REGISTER request.

The following header fields, except Contact, MUST be included in a REGISTER request. A Contact header field MAY be included:

Request-URI: The Request-URI names the domain of the location service for which the registration is meant (for example, "sip:chicago.com"). The "userinfo" and "@" components of the SIP URI MUST NOT be present.

To: The To header field contains the address of record whose registration is to be created, queried, or modified. The To header field and the Request-URI field typically differ, as the former contains a user name. This address-of-record MUST be a SIP URI or SIPS URI.

From: The From header field contains the address-of-record of the person responsible for the registration. The value is the same as the To header field unless the request is a third-party registration.

Call-ID: All registrations from a UAC SHOULD use the same Call-ID header field value for registrations sent to a particular registrar.

If the same client were to use different Call-ID values, a registrar could not detect whether a delayed REGISTER request might have arrived out of order.

CSeq: The CSeq value guarantees proper ordering of REGISTER requests. A UA MUST increment the CSeq value by one for each REGISTER request with the same Call-ID.

Contact: REGISTER requests MAY contain a Contact header field with zero or more values containing address bindings.

UAs MUST NOT send a new registration (that is, containing new Contact header field values, as opposed to a retransmission) until they have received a final response from the registrar for the previous one or the previous REGISTER request has timed out.

```
                                                 bob
                                               +----+
                                               | UA |
                                               |    |
                                               +----+
                                                  |
                                                  |3)INVITE
                                                  |   carol@chicago.com
         chicago.com         +--------+           V
         +---------+ 2)Store|Location|4)Query +-----+
         |Registrar|=======>| Service|<=======|Proxy|sip.chicago.com
         +---------+        +--------+=======>+-----+
               A                      5)Resp      |
               |                                  |
               |                                  |
     1)REGISTER|                                  |
               |                                  |
            +----+                                |
            | UA |<-------------------------------+
   cube2214a|    |                            6)INVITE
            +----+                    carol@cube2214a.chicago.com
             carol
```

Figure 2: REGISTER example

The following Contact header parameters have a special meaning in REGISTER requests:

action: The "action" parameter from RFC 2543 has been deprecated. UACs SHOULD NOT use the "action" parameter.

expires: The "expires" parameter indicates how long the UA would like the binding to be valid. The value is a number indicating seconds. If this parameter is not provided, the value of the Expires header field is used instead. Implementations MAY treat values larger than 2**32-1 (4294967295 seconds or 136 years) as equivalent to 2**32-1. Malformed values SHOULD be treated as equivalent to 3600.

### 10.2.1 Adding Bindings

The REGISTER request sent to a registrar includes the contact address(es) to which SIP requests for the address-of-record should be forwarded. The address-of-record is included in the To header field of the REGISTER request.

The Contact header field values of the request typically consist of SIP or SIPS URIs that identify particular SIP endpoints (for example, "sip:carol@cube2214a.chicago.com"), but they MAY use any URI scheme. A SIP UA can choose to register telephone numbers (with the tel URL, RFC 2806 [9]) or email addresses (with a mailto URL, RFC 2368 [32]) as Contacts for an address-of-record, for example.

For example, Carol, with address-of-record "sip:carol@chicago.com", would register with the SIP registrar of the domain chicago.com. Her registrations would then be used by a proxy server in the chicago.com domain to route requests for Carol's address-of-record to her SIP endpoint.

Once a client has established bindings at a registrar, it MAY send subsequent registrations containing new bindings or modifications to existing bindings as necessary. The 2xx response to the REGISTER request will contain, in a Contact header field, a complete list of bindings that have been registered for this address-of-record at this registrar.

If the address-of-record in the To header field of a REGISTER request is a SIPS URI, then any Contact header field values in the request SHOULD also be SIPS URIs. Clients should only register non-SIPS URIs under a SIPS address-of-record when the security of the resource represented by the contact address is guaranteed by other means. This may be applicable to URIs that invoke protocols other than SIP, or SIP devices secured by protocols other than TLS.

Registrations do not need to update all bindings. Typically, a UA only updates its own contact addresses.

#### 10.2.1.1 Setting the Expiration Interval of Contact Addresses

When a client sends a REGISTER request, it MAY suggest an expiration interval that indicates how long the client would like the registration to be valid. (As described in Section 10.3, the registrar selects the actual time interval based on its local policy.)

There are two ways in which a client can suggest an expiration interval for a binding: through an Expires header field or an "expires" Contact header parameter. The latter allows expiration intervals to be suggested on a per-binding basis when more than one binding is given in a single REGISTER request, whereas the former suggests an expiration interval for all Contact header field values that do not contain the "expires" parameter.

If neither mechanism for expressing a suggested expiration time is present in a REGISTER, the client is indicating its desire for the server to choose.

#### 10.2.1.2 Preferences among Contact Addresses

If more than one Contact is sent in a REGISTER request, the registering UA intends to associate all of the URIs in these Contact header field values with the address-of-record present in the To field. This list can be prioritized with the "q" parameter in the Contact header field. The "q" parameter indicates a relative preference for the particular Contact header field value compared to other bindings for this address-of-record. Section 16.6 describes how a proxy server uses this preference indication.

### 10.2.2 Removing Bindings

Registrations are soft state and expire unless refreshed, but can also be explicitly removed. A client can attempt to influence the expiration interval selected by the registrar as described in Section 10.2.1. A UA requests the immediate removal of a binding by specifying an expiration interval of "0" for that contact address in a REGISTER request. UAs SHOULD support this mechanism so that bindings can be removed before their expiration interval has passed.

The REGISTER-specific Contact header field value of "*" applies to all registrations, but it MUST NOT be used unless the Expires header field is present with a value of "0".

> Use of the "*" Contact header field value allows a registering UA to remove all bindings associated with an address-of-record without knowing their precise values.

### 10.2.3 Fetching Bindings

A success response to any REGISTER request contains the complete list of existing bindings, regardless of whether the request contained a Contact header field. If no Contact header field is present in a REGISTER request, the list of bindings is left unchanged.

### 10.2.4 Refreshing Bindings

Each UA is responsible for refreshing the bindings that it has previously established. A UA SHOULD NOT refresh bindings set up by other UAs.

The 200 (OK) response from the registrar contains a list of Contact fields enumerating all current bindings. The UA compares each contact address to see if it created the contact address, using comparison rules in Section 19.1.4. If so, it updates the expiration time interval according to the expires parameter or, if absent, the Expires field value. The UA then issues a REGISTER request for each of its bindings before the expiration interval has elapsed. It MAY combine several updates into one REGISTER request.

A UA SHOULD use the same Call-ID for all registrations during a single boot cycle. Registration refreshes SHOULD be sent to the same network address as the original registration, unless redirected.

### 10.2.5 Setting the Internal Clock

If the response for a REGISTER request contains a Date header field, the client MAY use this header field to learn the current time in order to set any internal clocks.

### 10.2.6 Discovering a Registrar

UAs can use three ways to determine the address to which to send registrations: by configuration, using the address-of-record, and multicast. A UA can be configured, in ways beyond the scope of this specification, with a registrar address. If there is no configured registrar address, the UA SHOULD use the host part of the address-of-record as the Request-URI and address the request there, using the normal SIP server location mechanisms [4]. For example, the UA for the user "sip:carol@chicago.com" addresses the REGISTER request to "sip:chicago.com".

Finally, a UA can be configured to use multicast. Multicast registrations are addressed to the well-known "all SIP servers" multicast address "sip.mcast.net" (224.0.1.75 for IPv4). No well-known IPv6 multicast address has been allocated; such an allocation will be documented separately when needed. SIP UAs MAY listen to that address and use it to become aware of the location of other local users (see [33]); however, they do not respond to the request.

> Multicast registration may be inappropriate in some environments, for example, if multiple businesses share the same local area network.

### 10.2.7 Transmitting a Request

Once the REGISTER method has been constructed, and the destination of the message identified, UACs follow the procedures described in Section 8.1.2 to hand off the REGISTER to the transaction layer.

If the transaction layer returns a timeout error because the REGISTER yielded no response, the UAC SHOULD NOT immediately re-attempt a registration to the same registrar.

> An immediate re-attempt is likely to also timeout. Waiting some reasonable time interval for the conditions causing the timeout to be corrected reduces unnecessary load on the network. No specific interval is mandated.

### 10.2.8 Error Responses

If a UA receives a 423 (Interval Too Brief) response, it MAY retry the registration after making the expiration interval of all contact addresses in the REGISTER request equal to or greater than the expiration interval within the Min-Expires header field of the 423 (Interval Too Brief) response.

## 10.3 Processing REGISTER Requests

A registrar is a UAS that responds to REGISTER requests and maintains a list of bindings that are accessible to proxy servers and redirect servers within its administrative domain. A registrar handles requests according to Section 8.2 and Section 17.2, but it accepts only REGISTER requests. A registrar MUST not generate 6xx responses.

A registrar MAY redirect REGISTER requests as appropriate. One common usage would be for a registrar listening on a multicast interface to redirect multicast REGISTER requests to its own unicast interface with a 302 (Moved Temporarily) response.

Registrars MUST ignore the Record-Route header field if it is included in a REGISTER request. Registrars MUST NOT include a Record-Route header field in any response to a REGISTER request.

> A registrar might receive a request that traversed a proxy which treats REGISTER as an unknown request and which added a Record-Route header field value.

A registrar has to know (for example, through configuration) the set of domain(s) for which it maintains bindings. REGISTER requests MUST be processed by a registrar in the order that they are received. REGISTER requests MUST also be processed atomically, meaning that a particular REGISTER request is either processed completely or not at all. Each REGISTER message MUST be processed independently of any other registration or binding changes.

When receiving a REGISTER request, a registrar follows these steps:

1. The registrar inspects the Request-URI to determine whether it has access to bindings for the domain identified in the Request-URI. If not, and if the server also acts as a proxy server, the server SHOULD forward the request to the addressed domain, following the general behavior for proxying messages described in Section 16.

2. To guarantee that the registrar supports any necessary extensions, the registrar MUST process the Require header field values as described for UASs in Section 8.2.2.

3. A registrar SHOULD authenticate the UAC. Mechanisms for the authentication of SIP user agents are described in Section 22. Registration behavior in no way overrides the generic authentication framework for SIP. If no authentication mechanism is available, the registrar MAY take the From address as the asserted identity of the originator of the request.

4. The registrar SHOULD determine if the authenticated user is authorized to modify registrations for this address-of-record. For example, a registrar might consult an authorization database that maps user names to a list of addresses-of-record for which that user has authorization to modify bindings. If the authenticated user is not authorized to modify bindings, the registrar MUST return a 403 (Forbidden) and skip the remaining steps.

   In architectures that support third-party registration, one entity may be responsible for updating the registrations associated with multiple addresses-of-record.

5. The registrar extracts the address-of-record from the To header field of the request. If the address-of-record is not valid for the domain in the Request-URI, the registrar MUST send a 404 (Not Found) response and skip the remaining steps. The URI MUST then be converted to a canonical form. To do that, all URI parameters MUST be removed (including the user-param), and any escaped characters MUST be converted to their unescaped form. The result serves as an index into the list of bindings.

6. The registrar checks whether the request contains the Contact header field. If not, it skips to the last step. If the Contact header field is present, the registrar checks if there is one Contact field value that contains the special value "*" and an Expires field. If the request has additional Contact fields or an expiration time other than zero, the request is invalid, and the server MUST return a 400 (Invalid Request) and skip the remaining steps. If not, the registrar checks whether the Call-ID agrees with the value stored for each binding. If not, it MUST remove the binding. If it does agree, it MUST remove the binding only if the CSeq in the request is higher than the value stored for that binding. Otherwise, the update MUST be aborted and the request fails.

7. The registrar now processes each contact address in the Contact header field in turn. For each address, it determines the expiration interval as follows:

   - If the field value has an "expires" parameter, that value MUST be taken as the requested expiration.

   - If there is no such parameter, but the request has an Expires header field, that value MUST be taken as the requested expiration.

   - If there is neither, a locally-configured default value MUST be taken as the requested expiration.

   The registrar MAY choose an expiration less than the requested expiration interval. If and only if the requested expiration interval is greater than zero AND smaller than one hour AND less than a registrar-configured minimum, the registrar MAY reject the registration with a response of 423 (Interval Too Brief). This response MUST contain a Min-Expires header field that states the minimum expiration interval the registrar is willing to honor. It then skips the remaining steps.

   Allowing the registrar to set the registration interval protects it against excessively frequent registration refreshes while limiting the state that it needs to maintain and decreasing the likelihood of registrations going stale. The expiration interval of a registration is frequently used in the creation of services. An example is a follow-me service, where the user may only be available at a terminal for a brief period. Therefore, registrars should accept brief registrations; a request should only be rejected if the interval is so short that the refreshes would degrade registrar performance.

For each address, the registrar then searches the list of current bindings using the URI comparison rules. If the binding does not exist, it is tentatively added. If the binding does exist, the registrar checks the Call-ID value. If the Call-ID value in the existing binding differs from the Call-ID value in the request, the binding MUST be removed if the expiration time is zero and updated otherwise. If they are the same, the registrar compares the CSeq value. If the value is higher than that of the existing binding, it MUST update or remove the binding as above. If not, the update MUST be aborted and the request fails.

This algorithm ensures that out-of-order requests from the same UA are ignored.

Each binding record records the Call-ID and CSeq values from the request.

The binding updates MUST be committed (that is, made visible to the proxy or redirect server) if and only if all binding updates and additions succeed. If any one of them fails (for example, because the back-end database commit failed), the request MUST fail with a 500 (Server Error) response and all tentative binding updates MUST be removed.

8. The registrar returns a 200 (OK) response. The response MUST contain Contact header field values enumerating all current bindings. Each Contact value MUST feature an "expires" parameter indicating its expiration interval chosen by the registrar. The response SHOULD include a Date header field.

# 11 Querying for Capabilities

The SIP method OPTIONS allows a UA to query another UA or a proxy server as to its capabilities. This allows a client to discover information about the supported methods, content types, extensions, codecs, etc. without "ringing" the other party. For example, before a client inserts a Require header field into an INVITE listing an option that it is not certain the destination UAS supports, the client can query the destination UAS with an OPTIONS to see if this option is returned in a Supported header field. All UAs MUST support the OPTIONS method.

The target of the OPTIONS request is identified by the Request-URI, which could identify another UA or a SIP server. If the OPTIONS is addressed to a proxy server, the Request-URI is set without a user part, similar to the way a Request-URI is set for a REGISTER request.

Alternatively, a server receiving an OPTIONS request with a Max-Forwards header field value of 0 MAY respond to the request regardless of the Request-URI.

> This behavior is common with HTTP/1.1. This behavior can be used as a "traceroute" functionality to check the capabilities of individual hop servers by sending a series of OPTIONS requests with incremented Max-Forwards values.

As is the case for general UA behavior, the transaction layer can return a timeout error if the OPTIONS yields no response. This may indicate that the target is unreachable and hence unavailable.

An OPTIONS request MAY be sent as part of an established dialog to query the peer on capabilities that may be utilized later in the dialog.

## 11.1 Construction of OPTIONS Request

An OPTIONS request is constructed using the standard rules for a SIP request as discussed in Section 8.1.1.

A Contact header field MAY be present in an OPTIONS.

An Accept header field SHOULD be included to indicate the type of message body the UAC wishes to receive in the response. Typically, this is set to a format that is used to describe the media capabilities of a UA, such as SDP (application/sdp).

The response to an OPTIONS request is assumed to be scoped to the Request-URI in the original request. However, only when an OPTIONS is sent as part of an established dialog is it guaranteed that future requests will be received by the server that generated the OPTIONS response.

Example OPTIONS request:

```
OPTIONS sip:carol@chicago.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKhjhs8ass877
Max-Forwards: 70
To: <sip:carol@chicago.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 63104 OPTIONS
Contact: <sip:alice@pc33.atlanta.com>
Accept: application/sdp
Content-Length: 0
```

## 11.2 Processing of OPTIONS Request

The response to an OPTIONS is constructed using the standard rules for a SIP response as discussed in Section 8.2.6. The response code chosen MUST be the same that would have been chosen had the request been an INVITE. That is, a 200 (OK) would be returned if the UAS is ready to accept a call, a 486 (Busy Here) would be returned if the UAS is busy, etc. This allows an OPTIONS request to be used to determine the basic state of a UAS, which can be an indication of whether the UAS will accept an INVITE request.

An OPTIONS request received within a dialog generates a 200 (OK) response that is identical to one constructed outside a dialog and does not have any impact on the dialog.

This use of OPTIONS has limitations due to the differences in proxy handling of OPTIONS and INVITE requests. While a forked INVITE can result in multiple 200 (OK) responses being returned, a forked OPTIONS will only result in a single 200 (OK) response, since it is treated by proxies using the non-INVITE handling. See Section 16.7 for the normative details.

If the response to an OPTIONS is generated by a proxy server, the proxy returns a 200 (OK), listing the capabilities of the server. The response does not contain a message body.

Allow, Accept, Accept-Encoding, Accept-Language, and Supported header fields SHOULD be present in a 200 (OK) response to an OPTIONS request. If the response is generated by a proxy, the Allow header field SHOULD be omitted as it is ambiguous since a proxy is method agnostic. Contact header fields MAY be present in a 200 (OK) response and have the same semantics as in a 3xx response. That is, they may list a set of alternative names and methods of reaching the user. A Warning header field MAY be present.

A message body MAY be sent, the type of which is determined by the Accept header field in the OPTIONS request (application/sdp is the default if the Accept header field is not present). If the types include one that can describe media capabilities, the UAS SHOULD include a body in the response for that purpose. Details on the construction of such a body in the case of application/sdp are described in [13].

Example OPTIONS response generated by a UAS (corresponding to the request in Section 11.1):

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKhjhs8ass877
 ;received=192.0.2.4
To: <sip:carol@chicago.com>;tag=93810874
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 63104 OPTIONS
Contact: <sip:carol@chicago.com>
Contact: <mailto:carol@chicago.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE
Accept: application/sdp
Accept-Encoding: gzip
Accept-Language: en
Supported: foo
Content-Type: application/sdp
Content-Length: 274

(SDP not shown)
```

# 12 Dialogs

A key concept for a user agent is that of a dialog. A dialog represents a peer-to-peer SIP relationship between two user agents that persists for some time. The dialog facilitates sequencing of messages between the user agents and proper routing of requests between both of them. The dialog represents a context in which to interpret SIP messages. Section 8 discussed method independent UA processing for requests and responses outside of a dialog. This section discusses how those requests and responses are used to construct a dialog, and then how subsequent requests and responses are sent within a dialog.

A dialog is identified at each UA with a dialog ID, which consists of a Call-ID value, a local tag and a remote tag. The dialog ID at each UA involved in the dialog is not the same. Specifically, the local tag at one UA is identical to the remote tag at the peer UA. The tags are opaque tokens that facilitate the generation of unique dialog IDs.

A dialog ID is also associated with all responses and with any request that contains a tag in the To field. The rules for computing the dialog ID of a message depend on whether the SIP element is a UAC or UAS. For a UAC, the Call-ID value of the dialog ID is set to the Call-ID of the message, the remote tag is set to the tag in the To field of the message, and the local tag is set to the tag in the From field of the message (these rules apply to both requests and responses). As one would expect for a UAS, the Call-ID value of the dialog ID is set to the Call-ID of the message, the remote tag is set to the tag in the From field of the message, and the local tag is set to the tag in the To field of the message.

A dialog contains certain pieces of state needed for further message transmissions within the dialog. This state consists of the dialog ID, a local sequence number (used to order requests from the UA to its peer), a remote sequence number (used to order requests from its peer to the UA), a local URI, a remote URI, remote target, a boolean flag called "secure", and a route set, which is an ordered list of URIs. The route set is the list of servers that need to be traversed to send a request to the peer. A dialog can also be in the "early" state, which occurs when it is created with a provisional response, and then transition to the "confirmed" state when a 2xx final response arrives. For other responses, or if no response arrives at all on that dialog, the early dialog terminates.

## 12.1 Creation of a Dialog

Dialogs are created through the generation of non-failure responses to requests with specific methods. Within this specification, only 2xx and 101-199 responses with a To tag, where the request was INVITE, will establish a dialog. A dialog established by a non-final response to a request is in the "early" state and it is called an early dialog. Extensions MAY define other means for creating dialogs. Section 13 gives more details that are specific to the INVITE method. Here, we describe the process for creation of dialog state that is not dependent on the method.

UAs MUST assign values to the dialog ID components as described below.

### 12.1.1 UAS behavior

When a UAS responds to a request with a response that establishes a dialog (such as a 2xx to INVITE), the UAS MUST copy all Record-Route header field values from the request into the response (including the URIs, URI parameters, and any Record-Route header field parameters, whether they are known or unknown to the UAS) and MUST maintain the order of those values. The UAS MUST add a Contact header field to the response. The Contact header field contains an address where the UAS would like to be contacted for subsequent requests in the dialog (which includes the ACK for a 2xx response in the case of an INVITE). Generally, the host portion of this URI is the IP address or FQDN of the host. The URI provided in the Contact header field MUST be a SIP or SIPS URI. If the request that initiated the dialog contained a

SIPS URI in the Request-URI or in the top Record-Route header field value, if there was any, or the Contact header field if there was no Record-Route header field, the Contact header field in the response MUST be a SIPS URI. The URI SHOULD have global scope (that is, the same URI can be used in messages outside this dialog). The same way, the scope of the URI in the Contact header field of the INVITE is not limited to this dialog either. It can therefore be used in messages to the UAC even outside this dialog.

The UAS then constructs the state of the dialog. This state MUST be maintained for the duration of the dialog.

If the request arrived over TLS, and the Request-URI contained a SIPS URI, the "secure" flag is set to TRUE.

The route set MUST be set to the list of URIs in the Record-Route header field from the request, taken in order and preserving all URI parameters. If no Record-Route header field is present in the request, the route set MUST be set to the empty set. This route set, even if empty, overrides any pre-existing route set for future requests in this dialog. The remote target MUST be set to the URI from the Contact header field of the request.

The remote sequence number MUST be set to the value of the sequence number in the CSeq header field of the request. The local sequence number MUST be empty. The call identifier component of the dialog ID MUST be set to the value of the Call-ID in the request. The local tag component of the dialog ID MUST be set to the tag in the To field in the response to the request (which always includes a tag), and the remote tag component of the dialog ID MUST be set to the tag from the From field in the request. A UAS MUST be prepared to receive a request without a tag in the From field, in which case the tag is considered to have a value of null.

> This is to maintain backwards compatibility with RFC 2543, which did not mandate From tags.

The remote URI MUST be set to the URI in the From field, and the local URI MUST be set to the URI in the To field.

### 12.1.2 UAC Behavior

When a UAC sends a request that can establish a dialog (such as an INVITE) it MUST provide a SIP or SIPS URI with global scope (i.e., the same SIP URI can be used in messages outside this dialog) in the Contact header field of the request. If the request has a Request-URI or a topmost Route header field value with a SIPS URI, the Contact header field MUST contain a SIPS URI.

When a UAC receives a response that establishes a dialog, it constructs the state of the dialog. This state MUST be maintained for the duration of the dialog.

If the request was sent over TLS, and the Request-URI contained a SIPS URI, the "secure" flag is set to TRUE.

The route set MUST be set to the list of URIs in the Record-Route header field from the response, taken in reverse order and preserving all URI parameters. If no Record-Route header field is present in the response, the route set MUST be set to the empty set. This route set, even if empty, overrides any pre-existing route set for future requests in this dialog. The remote target MUST be set to the URI from the Contact header field of the response.

The local sequence number MUST be set to the value of the sequence number in the CSeq header field of the request. The remote sequence number MUST be empty (it is established when the remote UA sends a request within the dialog). The call identifier component of the dialog ID MUST be set to the value of the Call-ID in the request. The local tag component of the dialog ID MUST be set to the tag in the From field in the request, and the remote tag component of the dialog ID MUST be set to the tag in the To field of the response. A UAC MUST be prepared to receive a response without a tag in the To field, in which case the tag is considered to have a value of null.

> This is to maintain backwards compatibility with RFC 2543, which did not mandate To tags.

The remote URI MUST be set to the URI in the To field, and the local URI MUST be set to the URI in the From field.

## 12.2 Requests within a Dialog

Once a dialog has been established between two UAs, either of them MAY initiate new transactions as needed within the dialog. The UA sending the request will take the UAC role for the transaction. The UA receiving the request will take the UAS role. Note that these may be different roles than the UAs held during the transaction that established the dialog.

Requests within a dialog MAY contain Record-Route and Contact header fields. However, these requests do not cause the dialog’s route set to be modified, although they may modify the remote target URI. Specifically, requests that are not target refresh requests do not modify the dialog’s remote target URI, and requests that are target refresh requests do. For dialogs that have been established with an

INVITE, the only target refresh request defined is re-INVITE (see Section 14). Other extensions may define different target refresh requests for dialogs established in other ways.

Note that an ACK is NOT a target refresh request.

Target refresh requests only update the dialog's remote target URI, and not the route set formed from the Record-Route. Updating the latter would introduce severe backwards compatibility problems with RFC 2543-compliant systems.

### 12.2.1 UAC Behavior

#### 12.2.1.1 Generating the Request

A request within a dialog is constructed by using many of the components of the state stored as part of the dialog.

The URI in the To field of the request MUST be set to the remote URI from the dialog state. The tag in the To header field of the request MUST be set to the remote tag of the dialog ID. The From URI of the request MUST be set to the local URI from the dialog state. The tag in the From header field of the request MUST be set to the local tag of the dialog ID. If the value of the remote or local tags is null, the tag parameter MUST be omitted from the To or From header fields, respectively.

Usage of the URI from the To and From fields in the original request within subsequent requests is done for backwards compatibility with RFC 2543, which used the URI for dialog identification. In this specification, only the tags are used for dialog identification. It is expected that mandatory reflection of the original To and From URI in mid-dialog requests will be deprecated in a subsequent revision of this specification.

The Call-ID of the request MUST be set to the Call-ID of the dialog. Requests within a dialog MUST contain strictly monotonically increasing and contiguous CSeq sequence numbers (increasing-by-one) in each direction (excepting ACK and CANCEL of course, whose numbers equal the requests being acknowledged or cancelled). Therefore, if the local sequence number is not empty, the value of the local sequence number MUST be incremented by one, and this value MUST be placed into the CSeq header field. If the local sequence number is empty, an initial value MUST be chosen using the guidelines of Section 8.1.1.5. The method field in the CSeq header field value MUST match the method of the request.

With a length of 32 bits, a client could generate, within a single call, one request a second for about 136 years before needing to wrap around. The initial value of the sequence number is chosen so that subsequent requests within the same call will not wrap around. A non-zero initial value allows clients to use a time-based initial sequence number. A client could, for example, choose the 31 most significant bits of a 32-bit second clock as an initial sequence number.

The UAC uses the remote target and route set to build the Request-URI and Route header field of the request.

If the route set is empty, the UAC MUST place the remote target URI into the Request-URI. The UAC MUST NOT add a Route header field to the request.

If the route set is not empty, and the first URI in the route set contains the lr parameter (see Section 19.1.1), the UAC MUST place the remote target URI into the Request-URI and MUST include a Route header field containing the route set values in order, including all parameters.

If the route set is not empty, and its first URI does not contain the lr parameter, the UAC MUST place the first URI from the route set into the Request-URI, stripping any parameters that are not allowed in a Request-URI. The UAC MUST add a Route header field containing the remainder of the route set values in order, including all parameters. The UAC MUST then place the remote target URI into the Route header field as the last value.

For example, if the remote target is sip:user@remoteua and the route set contains:

```
<sip:proxy1>,<sip:proxy2>,<sip:proxy3;lr>,<sip:proxy4>
```

The request will be formed with the following Request-URI and Route header field:

```
METHOD sip:proxy1
Route: <sip:proxy2>,<sip:proxy3;lr>,<sip:proxy4>,<sip:user@remoteua>
```

If the first URI of the route set does not contain the lr parameter, the proxy indicated does not understand the routing mechanisms described in this document and will act as specified in RFC 2543, replacing the Request-URI with the first Route header field value it receives while forwarding the message. Placing the Request-URI at the end of the Route header field preserves the information in that Request-URI across the strict router (it will be returned to the Request-URI when the request reaches a loose-router).

A UAC SHOULD include a Contact header field in any target refresh requests within a dialog, and unless there is a need to change it, the URI SHOULD be the same as used in previous requests within the dialog. If the "secure" flag is true, that URI MUST be a SIPS URI. As discussed in Section 12.2.2, a Contact header field in a target refresh request updates the remote target URI. This allows a UA to provide a new contact address, should its address change during the duration of the dialog.

However, requests that are not target refresh requests do not affect the remote target URI for the dialog.

The rest of the request is formed as described in Section 8.1.1.

Once the request has been constructed, the address of the server is computed and the request is sent, using the same procedures for requests outside of a dialog (Section 8.1.2).

The procedures in Section 8.1.2 will normally result in the request being sent to the address indicated by the topmost Route header field value or the Request-URI if no Route header field is present. Subject to certain restrictions, they allow the request to be sent to an alternate address (such as a default outbound proxy not represented in the route set).

#### 12.2.1.2 Processing the Responses

The UAC will receive responses to the request from the transaction layer. If the client transaction returns a timeout, this is treated as a 408 (Request Timeout) response.

The behavior of a UAC that receives a 3xx response for a request sent within a dialog is the same as if the request had been sent outside a dialog. This behavior is described in Section 8.1.3.4.

Note, however, that when the UAC tries alternative locations, it still uses the route set for the dialog to build the Route header of the request.

When a UAC receives a 2xx response to a target refresh request, it MUST replace the dialog's remote target URI with the URI from the Contact header field in that response, if present.

If the response for a request within a dialog is a 481 (Call/Transaction Does Not Exist) or a 408 (Request Timeout), the UAC SHOULD terminate the dialog. A UAC SHOULD also terminate a dialog if no response at all is received for the request (the client transaction would inform the TU about the timeout.)

For INVITE initiated dialogs, terminating the dialog consists of sending a BYE.

### 12.2.2 UAS Behavior

Requests sent within a dialog, as any other requests, are atomic. If a particular request is accepted by the UAS, all the state changes associated with it are performed. If the request is rejected, none of the state changes are performed.

Note that some requests, such as INVITEs, affect several pieces of state.

The UAS will receive the request from the transaction layer. If the request has a tag in the To header field, the UAS core computes the dialog identifier corresponding to the request and compares it with existing dialogs. If there is a match, this is a mid-dialog request. In that case, the UAS first applies the same processing rules for requests outside of a dialog, discussed in Section 8.2.

If the request has a tag in the To header field, but the dialog identifier does not match any existing dialogs, the UAS may have crashed and restarted, or it may have received a request for a different (possibly failed) UAS (the UASs can construct the To tags so that a UAS can identify that the tag was for a UAS for which it is providing recovery). Another possibility is that the incoming request has been simply misrouted. Based on the To tag, the UAS MAY either accept or reject the request. Accepting the request for acceptable To tags provides robustness, so that dialogs can persist even through crashes. UAs wishing to support this capability must take into consideration some issues such as choosing monotonically increasing CSeq sequence numbers even across reboots, reconstructing the route set, and accepting out-of-range RTP timestamps and sequence numbers.

If the UAS wishes to reject the request because it does not wish to recreate the dialog, it MUST respond to the request with a 481 (Call/Transaction Does Not Exist) status code and pass that to the server transaction.

Requests that do not change in any way the state of a dialog may be received within a dialog (for example, an OPTIONS request). They are processed as if they had been received outside the dialog.

If the remote sequence number is empty, it MUST be set to the value of the sequence number in the CSeq header field value in the request. If the remote sequence number was not empty, but the sequence number of the request is lower than the remote sequence number, the request is out of order and MUST be rejected with a 500 (Server Internal Error) response. If the remote sequence number was not empty, and the sequence number of the request is greater than the remote sequence number, the request is in order. It is possible for the CSeq sequence number to be higher than the remote sequence number by more than one. This is not an error condition, and a UAS SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. The UAS MUST then set the remote sequence number to the value of the sequence number in the CSeq header field value in the request.

> If a proxy challenges a request generated by the UAC, the UAC has to resubmit the request with credentials. The resubmitted request will have a new CSeq number. The UAS will never see the first request, and thus, it will notice a gap in the CSeq number space. Such a gap does not represent any error condition.

When a UAS receives a target refresh request, it MUST replace the dialog's remote target URI with the URI from the Contact header field in that request, if present.

### 12.3 Termination of a Dialog

Independent of the method, if a request outside of a dialog generates a non-2xx final response, any early dialogs created through provisional responses to that request are terminated. The mechanism for terminating confirmed dialogs is method specific. In this specification, the BYE method terminates a session and the dialog associated with it. See Section 15 for details.

## 13 Initiating a Session

### 13.1 Overview

When a user agent client desires to initiate a session (for example, audio, video, or a game), it formulates an INVITE request. The INVITE request asks a server to establish a session. This request may be forwarded by proxies, eventually arriving at one or more UAS that can potentially accept the invitation. These UASs will frequently need to query the user about whether to accept the invitation. After some time, those UASs can accept the invitation (meaning the session is to be established) by sending a 2xx response. If the invitation is not accepted, a 3xx, 4xx, 5xx or 6xx response is sent, depending on the reason for the rejection. Before sending a final response, the UAS can also send provisional responses (1xx) to advise the UAC of progress in contacting the called user.

After possibly receiving one or more provisional responses, the UAC will get one or more 2xx responses or one non-2xx final response. Because of the protracted amount of time it can take to receive final responses to INVITE, the reliability mechanisms for INVITE transactions differ from those of other requests (like OPTIONS). Once it receives a final response, the UAC needs to send an ACK for every final response it receives. The procedure for sending this ACK depends on the type of response. For final responses between 300 and 699, the ACK processing is done in the transaction layer and follows one set of rules (See Section 17). For 2xx responses, the ACK is generated by the UAC core.

A 2xx response to an INVITE establishes a session, and it also creates a dialog between the UA that issued the INVITE and the UA that generated the 2xx response. Therefore, when multiple 2xx responses are received from different remote UAs (because the INVITE forked), each 2xx establishes a different dialog. All these dialogs are part of the same call.

This section provides details on the establishment of a session using INVITE. A UA that supports INVITE MUST also support ACK, CANCEL and BYE.

## 13.2 UAC Processing

### 13.2.1 Creating the Initial INVITE

Since the initial INVITE represents a request outside of a dialog, its construction follows the procedures of Section 8.1.1. Additional processing is required for the specific case of INVITE.

An Allow header field (Section 20.5) SHOULD be present in the INVITE. It indicates what methods can be invoked within a dialog, on the UA sending the INVITE, for the duration of the dialog. For example, a UA capable of receiving INFO requests within a dialog [34] SHOULD include an Allow header field listing the INFO method.

A Supported header field (Section 20.37) SHOULD be present in the INVITE. It enumerates all the extensions understood by the UAC.

An Accept (Section 20.1) header field MAY be present in the INVITE. It indicates which Content-Types are acceptable to the UA, in both the response received by it, and in any subsequent requests sent to it within dialogs established by the INVITE. The Accept header field is especially useful for indicating support of various session description formats.

The UAC MAY add an Expires header field (Section 20.19) to limit the validity of the invitation. If the time indicated in the Expires header field is reached and no final answer for the INVITE has been received, the UAC core SHOULD generate a CANCEL request for the INVITE, as per Section 9.

A UAC MAY also find it useful to add, among others, Subject (Section 20.36), Organization (Section 20.25) and User-Agent (Section 20.41) header fields. They all contain information related to the INVITE.

The UAC MAY choose to add a message body to the INVITE. Section 8.1.1.10 deals with how to construct the header fields -- Content-Type among others -- needed to describe the message body.

There are special rules for message bodies that contain a session description - their corresponding Content-Disposition is "session". SIP uses an offer/answer model where one UA sends a session description, called the offer, which contains a proposed description of the session. The offer indicates the desired communications means (audio, video, games), parameters of those means (such as codec types) and addresses for receiving media from the answerer. The other UA responds with another session description, called the answer, which indicates which communications means are accepted, the parameters that apply to those means, and addresses for receiving media from the offerer. An offer/answer exchange is within the context of a dialog, so that if a SIP INVITE results in multiple dialogs, each is a separate offer/answer exchange. The offer/answer model defines restrictions on when offers and answers can be made (for example, you cannot make a new offer while one is in progress). This results in restrictions on where the offers and answers can appear in SIP messages. In this specification, offers and answers can only appear in INVITE requests and responses, and ACK. The usage of offers and answers is further restricted. For the initial INVITE transaction, the rules are:

- o The initial offer MUST be in either an INVITE or, if not there, in the first reliable non-failure message from the UAS back to the UAC. In this specification, that is the final 2xx response.

- o If the initial offer is in an INVITE, the answer MUST be in a reliable non-failure message from UAS back to UAC which is correlated to that INVITE. For this specification, that is only the final 2xx response to that INVITE. That same exact answer MAY also be placed in any provisional responses sent prior to the answer. The UAC MUST treat the first session description it receives as the answer, and MUST ignore any session descriptions in subsequent responses to the initial INVITE.

- o If the initial offer is in the first reliable non-failure message from the UAS back to UAC, the answer MUST be in the acknowledgement for that message (in this specification, ACK for a 2xx response).

- o After having sent or received an answer to the first offer, the UAC MAY generate subsequent offers in requests based on rules specified for that method, but only if it has received answers to any previous offers, and has not sent any offers to which it hasn't gotten an answer.

- o Once the UAS has sent or received an answer to the initial offer, it MUST NOT generate subsequent offers in any responses to the initial INVITE. This means that a UAS based on this specification alone can never generate subsequent offers until completion of the initial transaction.

Concretely, the above rules specify two exchanges for UAs compliant to this specification alone - the offer is in the INVITE, and the answer in the 2xx (and possibly in a 1xx as well, with the same value), or the offer is in the 2xx, and the answer is in the ACK. All user agents that support INVITE MUST support these two exchanges.

The Session Description Protocol (SDP) (RFC 2327 [1]) MUST be supported by all user agents as a means to describe sessions, and its usage for constructing offers and answers MUST follow the procedures defined in [13].

The restrictions of the offer-answer model just described only apply to bodies whose Content-Disposition header field value is "session". Therefore, it is possible that both the INVITE and the ACK contain a body message (for example, the INVITE carries a photo (Content-Disposition: render) and the ACK a session description (Content-Disposition: session)).

If the Content-Disposition header field is missing, bodies of Content-Type application/sdp imply the disposition "session", while other content types imply "render".

Once the INVITE has been created, the UAC follows the procedures defined for sending requests outside of a dialog (Section 8). This results in the construction of a client transaction that will ultimately send the request and deliver responses to the UAC.

### 13.2.2 Processing INVITE Responses

Once the INVITE has been passed to the INVITE client transaction, the UAC waits for responses for the INVITE. If the INVITE client transaction returns a timeout rather than a response the TU acts as if a 408 (Request Timeout) response had been received, as described in Section 8.1.3.

#### 13.2.2.1 1xx Responses

Zero, one or multiple provisional responses may arrive before one or more final responses are received. Provisional responses for an INVITE request can create "early dialogs". If a provisional response has a tag in the To field, and if the dialog ID of the response does not match an existing dialog, one is constructed using the procedures defined in Section 12.1.2.

The early dialog will only be needed if the UAC needs to send a request to its peer within the dialog before the initial INVITE transaction completes. Header fields present in a provisional response are applicable as long as the dialog is in the early state (for example, an Allow header field in a provisional response contains the methods that can be used in the dialog while this is in the early state).

#### 13.2.2.2 3xx Responses

A 3xx response may contain one or more Contact header field values providing new addresses where the callee might be reachable. Depending on the status code of the 3xx response (see Section 21.3), the UAC MAY choose to try those new addresses.

#### 13.2.2.3 4xx, 5xx and 6xx Responses

A single non-2xx final response may be received for the INVITE. 4xx, 5xx and 6xx responses may contain a Contact header field value indicating the location where additional information about the error can be found. Subsequent final responses (which would only arrive under error conditions) MUST be ignored.

All early dialogs are considered terminated upon reception of the non-2xx final response.

After having received the non-2xx final response the UAC core considers the INVITE transaction completed. The INVITE client transaction handles the generation of ACKs for the response (see Section 17).

#### 13.2.2.4 2xx Responses

Multiple 2xx responses may arrive at the UAC for a single INVITE request due to a forking proxy. Each response is distinguished by the tag parameter in the To header field, and each represents a distinct dialog, with a distinct dialog identifier.

If the dialog identifier in the 2xx response matches the dialog identifier of an existing dialog, the dialog MUST be transitioned to the "confirmed" state, and the route set for the dialog MUST be recomputed based on the 2xx response using the procedures of Section 12.2.1.2. Otherwise, a new dialog in the "confirmed" state MUST be constructed using the procedures of Section 12.1.2.

> Note that the only piece of state that is recomputed is the route set. Other pieces of state such as the highest sequence numbers (remote and local) sent within the dialog are not recomputed. The route set only is recomputed for backwards compatibility. RFC 2543 did not mandate mirroring of the Record-Route header field in a 1xx, only 2xx. However, we cannot update the entire state of the dialog, since mid-dialog requests may have been sent within the early dialog, modifying the sequence numbers, for example.

The UAC core MUST generate an ACK request for each 2xx received from the transaction layer. The header fields of the ACK are constructed in the same way as for any request sent within a dialog (see Section 12) with the exception of the CSeq and the header fields related to authentication. The sequence number of the CSeq header field MUST be the same as the INVITE being acknowledged, but the CSeq method MUST be ACK. The ACK MUST contain the same credentials as the INVITE. If the 2xx contains an offer (based on the rules above), the ACK MUST carry an answer in its body. If the offer in the 2xx response is not acceptable, the UAC core MUST generate a valid answer in the ACK and then send a BYE immediately.

Once the ACK has been constructed, the procedures of [4] are used to determine the destination address, port and transport. However, the request is passed to the transport layer directly for transmission, rather than a client transaction. This is because the UAC core handles retransmissions of the ACK, not the transaction layer. The ACK MUST be passed to the client transport every time a retransmission of the 2xx final response that triggered the ACK arrives.

The UAC core considers the INVITE transaction completed 64*T1 seconds after the reception of the first 2xx response. At this point all the early dialogs that have not transitioned to established dialogs are terminated. Once the INVITE transaction is considered completed by the UAC core, no more new 2xx responses are expected to arrive.

If, after acknowledging any 2xx response to an INVITE, the UAC does not want to continue with that dialog, then the UAC MUST terminate the dialog by sending a BYE request as described in Section 15.

## 13.3 UAS Processing

### 13.3.1 Processing of the INVITE

The UAS core will receive INVITE requests from the transaction layer. It first performs the request processing procedures of Section 8.2, which are applied for both requests inside and outside of a dialog.

Assuming these processing states are completed without generating a response, the UAS core performs the additional processing steps:

1. If the request is an INVITE that contains an Expires header field, the UAS core sets a timer for the number of seconds indicated in the header field value. When the timer fires, the invitation is considered to be expired. If the invitation expires before the UAS has generated a final response, a 487 (Request Terminated) response SHOULD be generated.

2. If the request is a mid-dialog request, the method-independent processing described in Section 12.2.2 is first applied. It might also modify the session; Section 14 provides details.

3. If the request has a tag in the To header field but the dialog identifier does not match any of the existing dialogs, the UAS may have crashed and restarted, or may have received a request for a different (possibly failed) UAS. Section 12.2.2 provides guidelines to achieve a robust behavior under such a situation.

Processing from here forward assumes that the INVITE is outside of a dialog, and is thus for the purposes of establishing a new session.

The INVITE may contain a session description, in which case the UAS is being presented with an offer for that session. It is possible that the user is already a participant in that session, even though the INVITE is outside of a dialog. This can happen when a user is invited to the same multicast conference by multiple other participants. If desired, the UAS MAY use identifiers within the session description to detect this duplication. For example, SDP contains a session id and version number in the origin (o) field. If the user is already a member of the session, and the session parameters contained in the session description have not changed, the UAS MAY silently accept the INVITE (that is, send a 2xx response without prompting the user).

If the INVITE does not contain a session description, the UAS is being asked to participate in a session, and the UAC has asked that the UAS provide the offer of the session. It MUST provide the offer in its first non-failure reliable message back to the UAC. In this specification, that is a 2xx response to the INVITE.

The UAS can indicate progress, accept, redirect, or reject the invitation. In all of these cases, it formulates a response using the procedures described in Section 8.2.6.

#### 13.3.1.1 Progress

If the UAS is not able to answer the invitation immediately, it can choose to indicate some kind of progress to the UAC (for example, an indication that a phone is ringing). This is accomplished with a provisional response between 101 and 199. These provisional responses establish early dialogs and therefore follow the procedures of Section 12.1.1 in addition to those of Section 8.2.6. A UAS MAY send as many provisional responses as it likes. Each of these MUST indicate the same dialog ID. However, these will not be delivered reliably.

If the UAS desires an extended period of time to answer the INVITE, it will need to ask for an "extension" in order to prevent proxies from canceling the transaction. A proxy has the option of canceling a transaction when there is a gap of 3 minutes between responses in a transaction. To prevent cancellation, the UAS MUST send a non-100 provisional response at every minute, to handle the possibility of lost provisional responses.

> An INVITE transaction can go on for extended durations when the user is placed on hold, or when interworking with PSTN systems which allow communications to take place without answering the call. The latter is common in Interactive Voice Response (IVR) systems.

#### 13.3.1.2 The INVITE is Redirected

If the UAS decides to redirect the call, a 3xx response is sent. A 300 (Multiple Choices), 301 (Moved Permanently) or 302 (Moved Temporarily) response SHOULD contain a Contact header field containing one or more URIs of new addresses to be tried. The response is passed to the INVITE server transaction, which will deal with its retransmissions.

#### 13.3.1.3 The INVITE is Rejected

A common scenario occurs when the callee is currently not willing or able to take additional calls at this end system. A 486 (Busy Here) SHOULD be returned in such a scenario. If the UAS knows that no other end system will be able to accept this call, a 600 (Busy Everywhere) response SHOULD be sent instead. However, it is unlikely that a UAS will be able to know this in general, and thus this response will not usually be used. The response is passed to the INVITE server transaction, which will deal with its retransmissions.

A UAS rejecting an offer contained in an INVITE SHOULD return a 488 (Not Acceptable Here) response. Such a response SHOULD include a Warning header field value explaining why the offer was rejected.

#### 13.3.1.4 The INVITE is Accepted

The UAS core generates a 2xx response. This response establishes a dialog, and therefore follows the procedures of Section 12.1.1 in addition to those of Section 8.2.6.

A 2xx response to an INVITE SHOULD contain the Allow header field and the Supported header field, and MAY contain the Accept header field. Including these header fields allows the UAC to determine the features and extensions supported by the UAS for the duration of the call, without probing.

If the INVITE request contained an offer, and the UAS had not yet sent an answer, the 2xx MUST contain an answer. If the INVITE did not contain an offer, the 2xx MUST contain an offer if the UAS had not yet sent an offer.

Once the response has been constructed, it is passed to the INVITE server transaction. Note, however, that the INVITE server transaction will be destroyed as soon as it receives this final response and passes it to the transport. Therefore, it is necessary to periodically pass the response directly to the transport until the ACK arrives. The 2xx response is passed to the transport with an interval that starts at T1 seconds and doubles for each retransmission until it reaches T2 seconds (T1 and T2 are defined in Section 17). Response retransmissions cease when an ACK request for the response is received. This is independent of whatever transport protocols are used to send the response.

Since 2xx is retransmitted end-to-end, there may be hops between UAS and UAC that are UDP. To ensure reliable delivery across these hops, the response is retransmitted periodically even if the transport at the UAS is reliable.

If the server retransmits the 2xx response for 64*T1 seconds without receiving an ACK, the dialog is confirmed, but the session SHOULD be terminated. This is accomplished with a BYE, as described in Section 15.

## 14 Modifying an Existing Session

A successful INVITE request (see Section 13) establishes both a dialog between two user agents and a session using the offer-answer model. Section 12 explains how to modify an existing dialog using a target refresh request (for example, changing the remote target URI of the dialog). This section describes how to modify the actual session. This modification can involve changing addresses or ports, adding a media stream, deleting a media stream, and so on. This is accomplished by sending a new INVITE request within the same dialog that established the session. An INVITE request sent within an existing dialog is known as a re-INVITE.

Note that a single re-INVITE can modify the dialog and the parameters of the session at the same time.

Either the caller or callee can modify an existing session.

The behavior of a UA on detection of media failure is a matter of local policy. However, automated generation of re-INVITE or BYE is NOT RECOMMENDED to avoid flooding the network with traffic when there is congestion. In any case, if these messages are sent automatically, they SHOULD be sent after some randomized interval.

Note that the paragraph above refers to automatically generated BYEs and re-INVITEs. If the user hangs up upon media failure, the UA would send a BYE request as usual.

### 14.1 UAC Behavior

The same offer-answer model that applies to session descriptions in INVITEs (Section 13.2.1) applies to re-INVITEs. As a result, a UAC that wants to add a media stream, for example, will create a new offer that contains this media stream, and send that in an INVITE request to its peer. It is important to note that the full description of the session, not just the change, is sent. This supports stateless session processing in various elements, and supports failover and recovery capabilities. Of course, a UAC MAY send a re-INVITE with no session description, in which case the first reliable non-failure response to the re-INVITE will contain the offer (in this specification, that is a 2xx response).

If the session description format has the capability for version numbers, the offerer SHOULD indicate that the version of the session description has changed.

The To, From, Call-ID, CSeq, and Request-URI of a re-INVITE are set following the same rules as for regular requests within an existing dialog, described in Section 12.

A UAC MAY choose not to add an Alert-Info header field or a body with Content-Disposition "alert" to re-INVITEs because UASs do not typically alert the user upon reception of a re-INVITE.

Unlike an INVITE, which can fork, a re-INVITE will never fork, and therefore, only ever generate a single final response. The reason a re-INVITE will never fork is that the Request-URI identifies the target as the UA instance it established the dialog with, rather than identifying an address-of-record for the user.

Note that a UAC MUST NOT initiate a new INVITE transaction within a dialog while another INVITE transaction is in progress in either direction.

1. If there is an ongoing INVITE client transaction, the TU MUST wait until the transaction reaches the completed or terminated state before initiating the new INVITE.

2. If there is an ongoing INVITE server transaction, the TU MUST wait until the transaction reaches the confirmed or terminated state before initiating the new INVITE.

However, a UA MAY initiate a regular transaction while an INVITE transaction is in progress. A UA MAY also initiate an INVITE transaction while a regular transaction is in progress.

If a UA receives a non-2xx final response to a re-INVITE, the session parameters MUST remain unchanged, as if no re-INVITE had been issued. Note that, as stated in Section 12.2.1.2, if the non-2xx final response is a 481 (Call/Transaction Does Not Exist), or a 408 (Request Timeout), or no response at all is received for the re-INVITE (that is, a timeout is returned by the INVITE client transaction), the UAC will terminate the dialog.

If a UAC receives a 491 response to a re-INVITE, it SHOULD start a timer with a value T chosen as follows:

1. If the UAC is the owner of the Call-ID of the dialog ID (meaning it generated the value), T has a randomly chosen value between 2.1 and 4 seconds in units of 10 ms.

2. If the UAC is not the owner of the Call-ID of the dialog ID, T has a randomly chosen value of between 0 and 2 seconds in units of 10 ms.

When the timer fires, the UAC SHOULD attempt the re-INVITE once more, if it still desires for that session modification to take place. For example, if the call was already hung up with a BYE, the re-INVITE would not take place.

The rules for transmitting a re-INVITE and for generating an ACK for a 2xx response to re-INVITE are the same as for the initial INVITE (Section 13.2.1).

## 14.2 UAS Behavior

Section 13.3.1 describes the procedure for distinguishing incoming re-INVITEs from incoming initial INVITEs and handling a re-INVITE for an existing dialog.

A UAS that receives a second INVITE before it sends the final response to a first INVITE with a lower CSeq sequence number on the same dialog MUST return a 500 (Server Internal Error) response to the second INVITE and MUST include a Retry-After header field with a randomly chosen value of between 0 and 10 seconds.

A UAS that receives an INVITE on a dialog while an INVITE it had sent on that dialog is in progress MUST return a 491 (Request Pending) response to the received INVITE.

If a UA receives a re-INVITE for an existing dialog, it MUST check any version identifiers in the session description or, if there are no version identifiers, the content of the session description to see if it has changed. If the session description has changed, the UAS MUST adjust the session parameters accordingly, possibly after asking the user for confirmation.

> Versioning of the session description can be used to accommodate the capabilities of new arrivals to a conference, add or delete media, or change from a unicast to a multicast conference.

If the new session description is not acceptable, the UAS can reject it by returning a 488 (Not Acceptable Here) response for the re-INVITE. This response SHOULD include a Warning header field.

If a UAS generates a 2xx response and never receives an ACK, it SHOULD generate a BYE to terminate the dialog.

A UAS MAY choose not to generate 180 (Ringing) responses for a re-INVITE because UACs do not typically render this information to the user. For the same reason, UASs MAY choose not to use an Alert-Info header field or a body with Content-Disposition "alert" in responses to a re-INVITE.

A UAS providing an offer in a 2xx (because the INVITE did not contain an offer) SHOULD construct the offer as if the UAS were making a brand new call, subject to the constraints of sending an offer that updates an existing session, as described in [13] in the case of SDP. Specifically, this means that it SHOULD include as many media formats and media types that the UA is willing to support. The UAS MUST ensure that the session description overlaps with its previous session description in media formats, transports, or other parameters that require support from the peer. This is to avoid the need for the peer to reject the session description. If, however, it is unacceptable to the UAC, the UAC SHOULD generate an answer with a valid session description, and then send a BYE to terminate the session.

# 15 Terminating a Session

This section describes the procedures for terminating a session established by SIP. The state of the session and the state of the dialog are very closely related. When a session is initiated with an INVITE, each 1xx or 2xx response from a distinct UAS creates a dialog, and if that response completes the offer/answer exchange, it also creates a session. As a result, each session is "associated" with a single dialog - the one which resulted in its creation. If an initial INVITE generates a non-2xx final response, that terminates all sessions (if any) and all dialogs (if any) that were created through responses to the request. By virtue of completing the transaction, a non-2xx final response also prevents further sessions from being created as a result of the INVITE. The BYE request is used to terminate a specific session or attempted session. In this case, the specific session is the one with the peer UA on the other side of the dialog. When a BYE is received on a dialog, any session associated with that dialog SHOULD terminate. A UA MUST NOT send a BYE outside of a dialog. The caller’s UA MAY send a BYE for either confirmed or early dialogs, and the callee’s UA MAY send a BYE on confirmed dialogs, but MUST NOT send a BYE on early dialogs.

However, the callee's UA MUST NOT send a BYE on a confirmed dialog until it has received an ACK for its 2xx response or until the server transaction times out. If no SIP extensions have defined other application layer states associated with the dialog, the BYE also terminates the dialog.

The impact of a non-2xx final response to INVITE on dialogs and sessions makes the use of CANCEL attractive. The CANCEL attempts to force a non-2xx response to the INVITE (in particular, a 487). Therefore, if a UAC wishes to give up on its call attempt entirely, it can send a CANCEL. If the INVITE results in 2xx final response(s) to the INVITE, this means that a UAS accepted the invitation while the CANCEL was in progress. The UAC MAY continue with the sessions established by any 2xx responses, or MAY terminate them with BYE.

> The notion of "hanging up" is not well defined within SIP. It is specific to a particular, albeit common, user interface. Typically, when the user hangs up, it indicates a desire to terminate the attempt to establish a session, and to terminate any sessions already created. For the caller's UA, this would imply a CANCEL request if the initial INVITE has not generated a final response, and a BYE to all confirmed dialogs after a final response. For the callee's UA, it would typically imply a BYE; presumably, when the user picked up the phone, a 2xx was generated, and so hanging up would result in a BYE after the ACK is received. This does not mean a user cannot hang up before receipt of the ACK, it just means that the software in his phone needs to maintain state for a short while in order to clean up properly. If the particular UI allows for the user to reject a call before its answered, a 403 (Forbidden) is a good way to express that. As per the rules above, a BYE can't be sent.

## 15.1 Terminating a Session with a BYE Request

### 15.1.1 UAC Behavior

A BYE request is constructed as would any other request within a dialog, as described in Section 12.

Once the BYE is constructed, the UAC core creates a new non-INVITE client transaction, and passes it the BYE request. The UAC MUST consider the session terminated (and therefore stop sending or listening for media) as soon as the BYE request is passed to the client transaction. If the response for the BYE is a 481 (Call/Transaction Does Not Exist) or a 408 (Request Timeout) or no response at all is received for the BYE (that is, a timeout is returned by the client transaction), the UAC MUST consider the session and the dialog terminated.

### 15.1.2 UAS Behavior

A UAS first processes the BYE request according to the general UAS processing described in Section 8.2. A UAS core receiving a BYE request checks if it matches an existing dialog. If the BYE does not match an existing dialog, the UAS core SHOULD generate a 481 (Call/Transaction Does Not Exist) response and pass that to the server transaction.

> This rule means that a BYE sent without tags by a UAC will be rejected. This is a change from RFC 2543, which allowed BYE without tags.

A UAS core receiving a BYE request for an existing dialog MUST follow the procedures of Section 12.2.2 to process the request. Once done, the UAS SHOULD terminate the session (and therefore stop sending and listening for media). The only case where it can elect not to are multicast sessions, where participation is possible even if the other participant in the dialog has terminated its involvement in the session. Whether or not it ends its participation on the session, the UAS core MUST generate a 2xx response to the BYE, and MUST pass that to the server transaction for transmission.

The UAS MUST still respond to any pending requests received for that dialog. It is RECOMMENDED that a 487 (Request Terminated) response be generated to those pending requests.

# 16 Proxy Behavior

## 16.1 Overview

SIP proxies are elements that route SIP requests to user agent servers and SIP responses to user agent clients. A request may traverse several proxies on its way to a UAS. Each will make routing decisions, modifying the request before forwarding it to the next element. Responses will route through the same set of proxies traversed by the request in the reverse order.

Being a proxy is a logical role for a SIP element. When a request arrives, an element that can play the role of a proxy first decides if it needs to respond to the request on its own. For instance, the request may be malformed or the element may need credentials from the client before acting as a proxy. The element MAY respond with any appropriate error code. When responding directly to a request, the element is playing the role of a UAS and MUST behave as described in Section 8.2.

A proxy can operate in either a stateful or stateless mode for each new request. When stateless, a proxy acts as a simple forwarding element. It forwards each request downstream to a single element determined by making a targeting and routing decision based on the request. It simply forwards every response it receives upstream. A stateless proxy discards information about a message once the message has been forwarded. A stateful proxy remembers information (specifically, transaction state) about each incoming request and any requests it sends as a result of processing the incoming request. It uses this information to affect the processing of future messages associated with that request. A stateful proxy MAY choose to "fork" a request, routing it to multiple destinations. Any request that is forwarded to more than one location MUST be handled statefully.

In some circumstances, a proxy MAY forward requests using stateful transports (such as TCP) without being transaction-stateful. For instance, a proxy MAY forward a request from one TCP connection to another transaction statelessly as long as it places enough information in the message to be able to forward the response down the same connection the request arrived on. Requests forwarded between different types of transports where the proxy's TU must take an active role in ensuring reliable delivery on one of the transports MUST be forwarded transaction statefully.

A stateful proxy MAY transition to stateless operation at any time during the processing of a request, so long as it did not do anything that would otherwise prevent it from being stateless initially (forking, for example, or generation of a 100 response). When performing such a transition, all state is simply discarded. The proxy SHOULD NOT initiate a CANCEL request.

Much of the processing involved when acting statelessly or statefully for a request is identical. The next several subsections are written from the point of view of a stateful proxy. The last section calls out those places where a stateless proxy behaves differently.

## 16.2 Stateful Proxy

When stateful, a proxy is purely a SIP transaction processing engine. Its behavior is modeled here in terms of the server and client transactions defined in Section 17. A stateful proxy has a server transaction associated with one or more client transactions by a higher layer proxy processing component (see figure 3), known as a proxy core. An incoming request is processed by a server transaction. Requests from the server transaction are passed to a proxy core. The proxy core determines where to route the request, choosing one or more next-hop locations. An outgoing request for each next-hop location is processed by its own associated client transaction. The proxy core collects the responses from the client transactions and uses them to send responses to the server transaction.

A stateful proxy creates a new server transaction for each new request received. Any retransmissions of the request will then be handled by that server transaction per Section 17. The proxy core MUST behave as a UAS with respect to sending an immediate provisional on that server transaction (such as 100 Trying) as described in Section 8.2.6. Thus, a stateful proxy SHOULD NOT generate 100 (Trying) responses to non-INVITE requests.

This is a model of proxy behavior, not of software. An implementation is free to take any approach that replicates the external behavior this model defines.

For all new requests, including any with unknown methods, an element intending to proxy the request MUST:

1. Validate the request (Section 16.3)

2. Preprocess routing information (Section 16.4)

3. Determine target(s) for the request (Section 16.5)

```
      +--------------------+
      |                    | +---+
      |                    | | C |
      |                    | | T |
      |                    | +---+
+---+ |       Proxy        | +---+   CT = Client Transaction
| S | |   "Higher" Layer   | | C |
| T | |                    | | T |   ST = Server Transaction
+---+ |                    | +---+
      |                    | +---+
      |                    | | C |
      |                    | | T |
      |                    | +---+
      +--------------------+
```

Figure 3: Stateful Proxy Model

4. Forward the request to each target (Section 16.6)

5. Process all responses (Section 16.7)

## 16.3 Request Validation

Before an element can proxy a request, it MUST verify the message's validity. A valid message must pass the following checks:

1. Reasonable Syntax

2. URI scheme

3. Max-Forwards

4. (Optional) Loop Detection

5. Proxy-Require

6. Proxy-Authorization

If any of these checks fail, the element MUST behave as a user agent server (see Section 8.2) and respond with an error code.

Notice that a proxy is not required to detect merged requests and MUST NOT treat merged requests as an error condition. The endpoints receiving the requests will resolve the merge as described in Section 8.2.2.2.

1. Reasonable syntax check

   The request MUST be well-formed enough to be handled with a server transaction. Any components involved in the remainder of these Request Validation steps or the Request Forwarding section MUST be well-formed. Any other components, well-formed or not, SHOULD be ignored and remain unchanged when the message is forwarded. For instance, an element would not reject a request because of a malformed Date header field. Likewise, a proxy would not remove a malformed Date header field before forwarding a request.

   This protocol is designed to be extended. Future extensions may define new methods and header fields at any time. An element MUST NOT refuse to proxy a request because it contains a method or header field it does not know about.

2. URI scheme check

   If the Request-URI has a URI whose scheme is not understood by the proxy, the proxy SHOULD reject the request with a 416 (Unsupported URI Scheme) response.

3. Max-Forwards check

   The Max-Forwards header field (Section 20.22) is used to limit the number of elements a SIP request can traverse.

   If the request does not contain a Max-Forwards header field, this check is passed.

   If the request contains a Max-Forwards header field with a field value greater than zero, the check is passed.

   If the request contains a Max-Forwards header field with a field value of zero (0), the element MUST NOT forward the request. If the request was for OPTIONS, the element MAY act as the final recipient and respond per Section 11. Otherwise, the element MUST return a 483 (Too many hops) response.

4. Optional Loop Detection check

   An element MAY check for forwarding loops before forwarding a request. If the request contains a Via header field with a sent-by value that equals a value placed into previous requests by the proxy, the request has been forwarded by this element before. The request has either looped or is legitimately spiraling through the element. To determine if the request has looped, the element MAY perform the branch parameter calculation described in Step 8 of Section 16.6 on this message and compare it to the parameter received in that Via header field. If the parameters match, the request has looped. If they differ, the request is spiraling, and processing continues. If a loop is detected, the element MAY return a 482 (Loop Detected) response.

5. Proxy-Require check

   Future extensions to this protocol may introduce features that require special handling by proxies. Endpoints will include a Proxy-Require header field in requests that use these features, telling the proxy not to process the request unless the feature is understood.

If the request contains a Proxy-Require header field (Section 20.29) with one or more option-tags this element does not understand, the element MUST return a 420 (Bad Extension) response. The response MUST include an Unsupported (Section 20.40) header field listing those option-tags the element did not understand.

6. Proxy-Authorization check

If an element requires credentials before forwarding a request, the request MUST be inspected as described in Section 22.3. That section also defines what the element must do if the inspection fails.

## 16.4 Route Information Preprocessing

The proxy MUST inspect the Request-URI of the request. If the Request-URI of the request contains a value this proxy previously placed into a Record-Route header field (see Section 16.6 item 4), the proxy MUST replace the Request-URI in the request with the last value from the Route header field, and remove that value from the Route header field. The proxy MUST then proceed as if it received this modified request.

This will only happen when the element sending the request to the proxy (which may have been an endpoint) is a strict router. This rewrite on receive is necessary to enable backwards compatibility with those elements. It also allows elements following this specification to preserve the Request-URI through strict-routing proxies (see Section 12.2.1.1).

This requirement does not obligate a proxy to keep state in order to detect URIs it previously placed in Record-Route header fields. Instead, a proxy need only place enough information in those URIs to recognize them as values it provided when they later appear.

If the Request-URI contains a maddr parameter, the proxy MUST check to see if its value is in the set of addresses or domains the proxy is configured to be responsible for. If the Request-URI has a maddr parameter with a value the proxy is responsible for, and the request was received using the port and transport indicated (explicitly or by default) in the Request-URI, the proxy MUST strip the maddr and any non-default port or transport parameter and continue processing as if those values had not been present in the request.

A request may arrive with a maddr matching the proxy, but on a port or transport different from that indicated in the URI. Such a request needs to be forwarded to the proxy using the indicated port and transport.

If the first value in the Route header field indicates this proxy, the proxy MUST remove that value from the request.

## 16.5 Determining Request Targets

Next, the proxy calculates the target(s) of the request. The set of targets will either be predetermined by the contents of the request or will be obtained from an abstract location service. Each target in the set is represented as a URI.

If the Request-URI of the request contains an maddr parameter, the Request-URI MUST be placed into the target set as the only target URI, and the proxy MUST proceed to Section 16.6.

If the domain of the Request-URI indicates a domain this element is not responsible for, the Request-URI MUST be placed into the target set as the only target, and the element MUST proceed to the task of Request Forwarding (Section 16.6).

There are many circumstances in which a proxy might receive a request for a domain it is not responsible for. A firewall proxy handling outgoing calls (the way HTTP proxies handle outgoing requests) is an example of where this is likely to occur.

If the target set for the request has not been predetermined as described above, this implies that the element is responsible for the domain in the Request-URI, and the element MAY use whatever mechanism it desires to determine where to send the request. Any of these mechanisms can be modeled as accessing an abstract Location Service. This may consist of obtaining information from a location service created by a SIP Registrar, reading a database, consulting a presence server, utilizing other protocols, or simply performing an algorithmic substitution on the Request-URI. When accessing the location service constructed by a registrar, the Request-URI MUST first be canonicalized as described in Section 10.3 before being used as an index. The output of these mechanisms is used to construct the target set.

If the Request-URI does not provide sufficient information for the proxy to determine the target set, it SHOULD return a 485 (Ambiguous) response. This response SHOULD contain a Contact header field containing URIs of new addresses to be tried. For example, an INVITE to sip:John.Smith@company.com may be ambiguous at a proxy whose location service has multiple John Smiths listed. See Section 21.4.23 for details.

Any information in or about the request or the current environment of the element MAY be used in the construction of the target set. For instance, different sets may be constructed depending on contents or the presence of header fields and bodies, the time of day of the request's arrival, the interface on which the request arrived, failure of previous requests, or even the element's current level of utilization.

As potential targets are located through these services, their URIs are added to the target set. Targets can only be placed in the target set once. If a target URI is already present in the set (based on the definition of equality for the URI type), it MUST NOT be added again.

A proxy MUST NOT add additional targets to the target set if the Request-URI of the original request does not indicate a resource this proxy is responsible for.

> A proxy can only change the Request-URI of a request during forwarding if it is responsible for that URI. If the proxy is not responsible for that URI, it will not recurse on 3xx or 416 responses as described below.

If the Request-URI of the original request indicates a resource this proxy is responsible for, the proxy MAY continue to add targets to the set after beginning Request Forwarding. It MAY use any information obtained during that processing to determine new targets. For instance, a proxy may choose to incorporate contacts obtained in a redirect response (3xx) into the target set. If a proxy uses a dynamic source of information while building the target set (for instance, if it consults a SIP Registrar), it SHOULD monitor that source for the duration of processing the request. New locations SHOULD be added to the target set as they become available. As above, any given URI MUST NOT be added to the set more than once.

> Allowing a URI to be added to the set only once reduces unnecessary network traffic, and in the case of incorporating contacts from redirect requests prevents infinite recursion.

For example, a trivial location service is a "no-op", where the target URI is equal to the incoming request URI. The request is sent to a specific next hop proxy for further processing. During request forwarding of Section 16.6, Item 6, the identity of that next hop, expressed as a SIP or SIPS URI, is inserted as the top-most Route header field value into the request.

If the Request-URI indicates a resource at this proxy that does not exist, the proxy MUST return a 404 (Not Found) response.

If the target set remains empty after applying all of the above, the proxy MUST return an error response, which SHOULD be the 480 (Temporarily Unavailable) response.

## 16.6 Request Forwarding

As soon as the target set is non-empty, a proxy MAY begin forwarding the request. A stateful proxy MAY process the set in any order. It MAY process multiple targets serially, allowing each client transaction to complete before starting the next. It MAY start client transactions with every target in parallel. It also MAY arbitrarily divide the set into groups, processing the groups serially and processing the targets in each group in parallel.

A common ordering mechanism is to use the qvalue parameter of targets obtained from Contact header fields (see Section 20.10). Targets are processed from highest qvalue to lowest. Targets with equal qvalues may be processed in parallel.

A stateful proxy must have a mechanism to maintain the target set as responses are received and associate the responses to each forwarded request with the original request. For the purposes of this model, this mechanism is a "response context" created by the proxy layer before forwarding the first request.

For each target, the proxy forwards the request following these steps:

1. Make a copy of the received request

2. Update the Request-URI

3. Update the Max-Forwards header field

4. Optionally add a Record-route header field value

5. Optionally add additional header fields

6. Postprocess routing information

7. Determine the next-hop address, port, and transport

8. Add a Via header field value

9. Add a Content-Length header field if necessary

10. Forward the new request

11. Set timer C

Each of these steps is detailed below:

1. Copy request

The proxy starts with a copy of the received request. The copy MUST initially contain all of the header fields from the received request. Fields not detailed in the processing described below MUST NOT be removed. The copy SHOULD maintain the ordering of the header fields as in the received request. The proxy MUST NOT reorder field values with a common field name (See Section 7.3.1). The proxy MUST NOT add to, modify, or remove the message body.

An actual implementation need not perform a copy; the primary requirement is that the processing for each next hop begin with the same request.

2. Request-URI

The Request-URI in the copy's start line MUST be replaced with the URI for this target. If the URI contains any parameters not allowed in a Request-URI, they MUST be removed.

This is the essence of a proxy's role. This is the mechanism through which a proxy routes a request toward its destination.

In some circumstances, the received Request-URI is placed into the target set without being modified. For that target, the replacement above is effectively a no-op.

3. Max-Forwards

If the copy contains a Max-Forwards header field, the proxy MUST decrement its value by one (1).

If the copy does not contain a Max-Forwards header field, the proxy MUST add one with a field value, which SHOULD be 70.

Some existing UAs will not provide a Max-Forwards header field in a request.

4. Record-Route

If this proxy wishes to remain on the path of future requests in a dialog created by this request (assuming the request creates a dialog), it MUST insert a Record-Route header field value into the copy before any existing Record-Route header field values, even if a Route header field is already present.

Requests establishing a dialog may contain a preloaded Route header field.

If this request is already part of a dialog, the proxy SHOULD insert a Record-Route header field value if it wishes to remain on the path of future requests in the dialog. In normal endpoint operation as described in Section 12, these Record-Route header field values will not have any effect on the route sets used by the endpoints.

The proxy will remain on the path if it chooses to not insert a Record-Route header field value into requests that are already part of a dialog. However, it would be removed from the path when an endpoint that has failed reconstitutes the dialog.

A proxy MAY insert a Record-Route header field value into any request. If the request does not initiate a dialog, the endpoints will ignore the value. See Section 12 for details on how endpoints use the Record-Route header field values to construct Route header fields.

Each proxy in the path of a request chooses whether to add a Record-Route header field value independently - the presence of a Record-Route header field in a request does not obligate this proxy to add a value.

The URI placed in the Record-Route header field value MUST be a SIP or SIPS URI. This URI MUST contain an lr parameter (see Section 19.1.1). This URI MAY be different for each destination the request is forwarded to. The URI SHOULD NOT contain the transport parameter unless the proxy has knowledge (such as in a private network) that the next downstream element that will be in the path of subsequent requests supports that transport.

The URI this proxy provides will be used by some other element to make a routing decision. This proxy, in general, has no way of knowing the capabilities of that element, so it must restrict itself to the mandatory elements of a SIP implementation: SIP URIs and either the TCP or UDP transports.

The URI placed in the Record-Route header field MUST resolve to the element inserting it (or a suitable stand-in) when the server location procedures of [4] are applied to it, so that subsequent requests reach the same SIP element. If the Request-URI contains a SIPS URI, or the topmost Route header field value (after the post processing of bullet 6) contains a SIPS URI, the URI placed into the Record-Route header field MUST be a SIPS URI. Furthermore, if the request was not received over TLS, the proxy MUST insert a Record-Route header field. In a similar fashion, a proxy that receives a request over TLS, but generates a request without a SIPS URI in the Request-URI or topmost Route header field value (after the post processing of bullet 6), MUST insert a Record-Route header field that is not a SIPS URI.

A proxy at a security perimeter must remain on the perimeter throughout the dialog.

If the URI placed in the Record-Route header field needs to be rewritten when it passes back through in a response, the URI MUST be distinct enough to locate at that time. (The request may spiral through this proxy, resulting in more than one Record-Route header field value being added). Item 8 of Section 16.7 recommends a mechanism to make the URI sufficiently distinct.

The proxy MAY include parameters in the Record-Route header field value. These will be echoed in some responses to the request such as the 200 (OK) responses to INVITE. Such parameters may be useful for keeping state in the message rather than the proxy.

If a proxy needs to be in the path of any type of dialog (such as one straddling a firewall), it SHOULD add a Record-Route header field value to every request with a method it does not understand since that method may have dialog semantics.

The URI a proxy places into a Record-Route header field is only valid for the lifetime of any dialog created by the transaction in which it occurs. A dialog-stateful proxy, for example, MAY refuse to accept future requests with that value in the Request-URI after the dialog has terminated. Non-dialog-stateful proxies, of course, have no concept of when the dialog has terminated, but they MAY encode enough information in the value to compare it against the dialog identifier of future requests and MAY reject requests not matching that information. Endpoints MUST NOT use a URI obtained from a Record-Route header field outside the dialog in which it was provided. See

Section 12 for more information on an endpoint's use of Record-Route header fields.

Record-routing may be required by certain services where the proxy needs to observe all messages in a dialog. However, it slows down processing and impairs scalability and thus proxies should only record-route if required for a particular service.

The Record-Route process is designed to work for any SIP request that initiates a dialog. INVITE is the only such request in this specification, but extensions to the protocol MAY define others.

5. Add Additional Header Fields

The proxy MAY add any other appropriate header fields to the copy at this point.

6. Postprocess routing information

A proxy MAY have a local policy that mandates that a request visit a specific set of proxies before being delivered to the destination. A proxy MUST ensure that all such proxies are loose routers. Generally, this can only be known with certainty if the proxies are within the same administrative domain. This set of proxies is represented by a set of URIs (each of which contains the lr parameter). This set MUST be pushed into the Route header field of the copy ahead of any existing values, if present. If the Route header field is absent, it MUST be added, containing that list of URIs.

If the proxy has a local policy that mandates that the request visit one specific proxy, an alternative to pushing a Route value into the Route header field is to bypass the forwarding logic of item 10 below, and instead just send the request to the address, port, and transport for that specific proxy. If the request has a Route header field, this alternative MUST NOT be used unless it is known that next hop proxy is a loose router. Otherwise, this approach MAY be used, but the Route insertion mechanism above is preferred for its robustness, flexibility, generality and consistency of operation. Furthermore, if the Request-URI contains a SIPS URI, TLS MUST be used to communicate with that proxy.

If the copy contains a Route header field, the proxy MUST inspect the URI in its first value. If that URI does not contain an lr parameter, the proxy MUST modify the copy as follows:

- The proxy MUST place the Request-URI into the Route header field as the last value.

- The proxy MUST then place the first Route header field value into the Request-URI and remove that value from the Route header field.

Appending the Request-URI to the Route header field is part of a mechanism used to pass the information in that Request-URI through strict-routing elements. "Popping" the first Route header field value into the Request-URI formats the message the way a strict-routing element expects to receive it (with its own URI in the Request-URI and the next location to visit in the first Route header field value).

7. Determine Next-Hop Address, Port, and Transport

The proxy MAY have a local policy to send the request to a specific IP address, port, and transport, independent of the values of the Route and Request-URI. Such a policy MUST NOT be used if the proxy is not certain that the IP address, port, and transport correspond to a server that is a loose router. However, this mechanism for sending the request through a specific next hop is NOT RECOMMENDED; instead a Route header field should be used for that purpose as described above.

In the absence of such an overriding mechanism, the proxy applies the procedures listed in [4] as follows to determine where to send the request. If the proxy has reformatted the request to send to a strict-routing element as described in step 6 above, the proxy MUST apply those procedures to the Request-URI of the request. Otherwise, the proxy MUST apply the procedures to the first value in the Route header field, if present, else the Request-URI. The procedures will produce an ordered set of (address, port, transport) tuples. Independently of which URI is being used as input to the procedures of [4], if the Request-URI specifies a SIPS resource, the proxy MUST follow the procedures of [4] as if the input URI were a SIPS URI.

As described in [4], the proxy MUST attempt to deliver the message to the first tuple in that set, and proceed through the set in order until the delivery attempt succeeds.

For each tuple attempted, the proxy MUST format the message as appropriate for the tuple and send the request using a new client transaction as detailed in steps 8 through 10.

Since each attempt uses a new client transaction, it represents a new branch. Thus, the branch parameter provided with the Via header field inserted in step 8 MUST be different for each attempt.

If the client transaction reports failure to send the request or a timeout from its state machine, the proxy continues to the next address in that ordered set. If the ordered set is exhausted, the request cannot be forwarded to this element in the target set. The proxy does not need to place anything in the response context, but otherwise acts as if this element of the target set returned a 408 (Request Timeout) final response.

8. Add a Via header field value

The proxy MUST insert a Via header field value into the copy before the existing Via header field values. The construction of this value follows the same guidelines of Section 8.1.1.7. This implies that the proxy will compute its own branch parameter, which will be globally unique for that branch, and contain the requisite magic cookie. Note that this implies that the branch parameter will be different for different instances of a spiraled or looped request through a proxy.

Proxies choosing to detect loops have an additional constraint in the value they use for construction of the branch parameter. A proxy choosing to detect loops SHOULD create a branch parameter separable into two parts by the implementation. The first part MUST satisfy the constraints of Section 8.1.1.7 as described above. The second is used to perform loop detection and distinguish loops from spirals.

Loop detection is performed by verifying that, when a request returns to a proxy, those fields having an impact on the processing of the request have not changed. The value placed in this part of the branch parameter SHOULD reflect all of those fields (including any Route, Proxy-Require and Proxy-Authorization header fields). This is to ensure that if the request is routed back to the proxy and one of those fields changes, it is treated as a spiral and not a loop (see Section 16.3). A common way to create this value is to compute a cryptographic hash of the To tag, From tag, Call-ID header field, the Request-URI of the request received (before translation), the topmost Via header, and the sequence number from the CSeq header field, in addition to any Proxy-Require and Proxy-Authorization header fields that may be present. The algorithm used to compute the hash is implementation-dependent, but MD5 (RFC 1321 [35]), expressed in hexadecimal, is a reasonable choice. (Base64 is not permissible for a token.)

If a proxy wishes to detect loops, the "branch" parameter it supplies MUST depend on all information affecting processing of a request, including the incoming Request-URI and any header fields affecting the request's admission or routing. This is necessary to distinguish looped requests from requests whose routing parameters have changed before returning to this server.

The request method MUST NOT be included in the calculation of the branch parameter. In particular, CANCEL and ACK requests (for non-2xx responses) MUST have the same branch value as the corresponding request they cancel or acknowledge. The branch parameter is used in correlating those requests at the server handling them (see Sections 17.2.3 and 9.2).

9. Add a Content-Length header field if necessary

If the request will be sent to the next hop using a stream-based transport and the copy contains no Content-Length header field, the proxy MUST insert one with the correct value for the body of the request (see Section 20.14).

10. Forward Request

A stateful proxy MUST create a new client transaction for this request as described in Section 17.1 and instructs the transaction to send the request using the address, port and transport determined in step 7.

11. Set timer C

In order to handle the case where an INVITE request never generates a final response, the TU uses a timer which is called timer C. Timer C MUST be set for each client transaction when an INVITE request is proxied. The timer MUST be larger than 3 minutes. Section 16.7 bullet 2 discusses how this timer is updated with provisional responses, and Section 16.8 discusses processing when it fires.

## 16.7 Response Processing

When a response is received by an element, it first tries to locate a client transaction (Section 17.1.3) matching the response. If none is found, the element MUST process the response (even if it is an informational response) as a stateless proxy (described below). If a match is found, the response is handed to the client transaction.

> Forwarding responses for which a client transaction (or more generally any knowledge of having sent an associated request) is not found improves robustness. In particular, it ensures that "late" 2xx responses to INVITE requests are forwarded properly.

As client transactions pass responses to the proxy layer, the following processing MUST take place:

1. Find the appropriate response context

2. Update timer C for provisional responses

3. Remove the topmost Via

4. Add the response to the response context

5. Check to see if this response should be forwarded immediately

6. When necessary, choose the best final response from the response context

If no final response has been forwarded after every client transaction associated with the response context has been terminated, the proxy must choose and forward the "best" response from those it has seen so far.

The following processing MUST be performed on each response that is forwarded. It is likely that more than one response to each request will be forwarded: at least each provisional and one final response.

7. Aggregate authorization header field values if necessary

8. Optionally rewrite Record-Route header field values

9. Forward the response

10. Generate any necessary CANCEL requests

Each of the above steps are detailed below:

1. Find Context

   The proxy locates the "response context" it created before forwarding the original request using the key described in Section 16.6. The remaining processing steps take place in this context.

2. Update timer C for provisional responses

   For an INVITE transaction, if the response is a provisional response with status codes 101 to 199 inclusive (i.e., anything but 100), the proxy MUST reset timer C for that client transaction. The timer MAY be reset to a different value, but this value MUST be greater than 3 minutes.

3. Via

   The proxy removes the topmost Via header field value from the response.

   If no Via header field values remain in the response, the response was meant for this element and MUST NOT be forwarded. The remainder of the processing described in this section is not performed on this message, the UAC processing rules described in Section 8.1.3 are followed instead (transport layer processing has already occurred).

   This will happen, for instance, when the element generates CANCEL requests as described in Section 10.

4. Add response to context

   Final responses received are stored in the response context until a final response is generated on the server transaction associated with this context. The response may be a candidate for the best final response to be returned on that server transaction. Information from this response may be needed in forming the best response, even if this response is not chosen.

   If the proxy chooses to recurse on any contacts in a 3xx response by adding them to the target set, it MUST remove them from the response before adding the response to the response context. However, a proxy SHOULD NOT recurse to a non-SIPS URI if the Request-URI of the original request was a SIPS URI. If the proxy recurses on all of the contacts in a 3xx response, the proxy SHOULD NOT add the resulting contactless response to the response context.

Removing the contact before adding the response to the response context prevents the next element upstream from retrying a location this proxy has already attempted.

3xx responses may contain a mixture of SIP, SIPS, and non-SIP URIs. A proxy may choose to recurse on the SIP and SIPS URIs and place the remainder into the response context to be returned, potentially in the final response.

If a proxy receives a 416 (Unsupported URI Scheme) response to a request whose Request-URI scheme was not SIP, but the scheme in the original received request was SIP or SIPS (that is, the proxy changed the scheme from SIP or SIPS to something else when it proxied a request), the proxy SHOULD add a new URI to the target set. This URI SHOULD be a SIP URI version of the non-SIP URI that was just tried. In the case of the tel URL, this is accomplished by placing the telephone-subscriber part of the tel URL into the user part of the SIP URI, and setting the hostpart to the domain where the prior request was sent. See Section 19.1.6 for more detail on forming SIP URIs from tel URLs.

As with a 3xx response, if a proxy "recurses" on the 416 by trying a SIP or SIPS URI instead, the 416 response SHOULD NOT be added to the response context.

5. Check response for forwarding

Until a final response has been sent on the server transaction, the following responses MUST be forwarded immediately:

- Any provisional response other than 100 (Trying)

- Any 2xx response

If a 6xx response is received, it is not immediately forwarded, but the stateful proxy SHOULD cancel all client pending transactions as described in Section 10, and it MUST NOT create any new branches in this context.

This is a change from RFC 2543, which mandated that the proxy was to forward the 6xx response immediately. For an INVITE transaction, this approach had the problem that a 2xx response could arrive on another branch, in which case the proxy would have to forward the 2xx. The result was that the UAC could receive a 6xx response followed by a 2xx response, which should never be allowed to happen. Under the new rules, upon receiving a 6xx, a proxy will issue a CANCEL request, which will generally result in 487 responses from all outstanding client transactions, and then at that point the 6xx is forwarded upstream.

After a final response has been sent on the server transaction, the following responses MUST be forwarded immediately:

- Any 2xx response to an INVITE request

A stateful proxy MUST NOT immediately forward any other responses. In particular, a stateful proxy MUST NOT forward any 100 (Trying) response. Those responses that are candidates for forwarding later as the "best" response have been gathered as described in step "Add Response to Context".

Any response chosen for immediate forwarding MUST be processed as described in steps "Aggregate Authorization Header Field Values" through "Record-Route".

This step, combined with the next, ensures that a stateful proxy will forward exactly one final response to a non-INVITE request, and either exactly one non-2xx response or one or more 2xx responses to an INVITE request.

6. Choosing the best response

A stateful proxy MUST send a final response to a response context’s server transaction if no final responses have been immediately forwarded by the above rules and all client transactions in this response context have been terminated.

The stateful proxy MUST choose the "best" final response among those received and stored in the response context.

If there are no final responses in the context, the proxy MUST send a 408 (Request Timeout) response to the server transaction.

Otherwise, the proxy MUST forward a response from the responses stored in the response context. It MUST choose from the 6xx class responses if any exist in the context. If no 6xx class responses are present, the proxy SHOULD choose from the lowest response class stored in the response context. The proxy MAY select any response within that chosen class. The proxy SHOULD give preference to responses that provide information affecting resubmission of this request, such as 401, 407, 415, 420, and 484 if the 4xx class is chosen.

A proxy which receives a 503 (Service Unavailable) response SHOULD NOT forward it upstream unless it can determine that any subsequent requests it might proxy will also generate a 503. In other words, forwarding a 503 means that the proxy knows it cannot service any requests, not just the one for the Request-URI in the request which generated the 503. If the only response that was received is a 503, the proxy SHOULD generate a 500 response and forward that upstream.

The forwarded response MUST be processed as described in steps "Aggregate Authorization Header Field Values" through "Record-Route".

For example, if a proxy forwarded a request to 4 locations, and received 503, 407, 501, and 404 responses, it may choose to forward the 407 (Proxy Authentication Required) response.

1xx and 2xx responses may be involved in the establishment of dialogs. When a request does not contain a To tag, the To tag in the response is used by the UAC to distinguish multiple responses to a dialog creating request. A proxy MUST NOT insert a tag into the To header field of a 1xx or 2xx response if the request did not contain one. A proxy MUST NOT modify the tag in the To header field of a 1xx or 2xx response.

Since a proxy may not insert a tag into the To header field of a 1xx response to a request that did not contain one, it cannot issue non-100 provisional responses on its own. However, it can branch the request to a UAS sharing the same element as the proxy. This UAS can return its own provisional responses, entering into an early dialog with the initiator of the request. The UAS does not have to be a discreet process from the proxy. It could be a virtual UAS implemented in the same code space as the proxy.

3-6xx responses are delivered hop-by-hop. When issuing a 3-6xx response, the element is effectively acting as a UAS, issuing its own response, usually based on the responses received from downstream elements. An element SHOULD preserve the To tag when simply forwarding a 3-6xx response to a request that did not contain a To tag.

A proxy MUST NOT modify the To tag in any forwarded response to a request that contains a To tag.

While it makes no difference to the upstream elements if the proxy replaced the To tag in a forwarded 3-6xx response, preserving the original tag may assist with debugging.

When the proxy is aggregating information from several responses, choosing a To tag from among them is arbitrary, and generating a new To tag may make debugging easier. This happens, for instance, when combining 401 (Unauthorized) and 407 (Proxy Authentication Required) challenges, or combining Contact values from unencrypted and unauthenticated 3xx responses.

7. Aggregate Authorization Header Field Values

If the selected response is a 401 (Unauthorized) or 407 (Proxy Authentication Required), the proxy MUST collect any WWW-Authenticate and Proxy-Authenticate header field values from all other 401 (Unauthorized) and 407 (Proxy Authentication Required) responses received so far in this response context and add them to this response without modification before forwarding. The resulting 401 (Unauthorized) or 407 (Proxy Authentication Required) response could have several WWW-Authenticate AND Proxy-Authenticate header field values.

This is necessary because any or all of the destinations the request was forwarded to may have requested credentials. The client needs to receive all of those challenges and supply credentials for each of them when it retries the request. Motivation for this behavior is provided in Section 26.

8. Record-Route

If the selected response contains a Record-Route header field value originally provided by this proxy, the proxy MAY choose to rewrite the value before forwarding the response. This allows the proxy to provide different URIs for itself to the next upstream and downstream elements. A proxy may choose to use this mechanism for any reason. For instance, it is useful for multi-homed hosts.

If the proxy received the request over TLS, and sent it out over a non-TLS connection, the proxy MUST rewrite the URI in the Record-Route header field to be a SIPS URI. If the proxy received the request over a non-TLS connection, and sent it out over TLS, the proxy MUST rewrite the URI in the Record-Route header field to be a SIP URI.

The new URI provided by the proxy MUST satisfy the same constraints on URIs placed in Record-Route header fields in requests (see Step 4 of Section 16.6) with the following modifications:

The URI SHOULD NOT contain the transport parameter unless the proxy has knowledge that the next upstream (as opposed to downstream) element that will be in the path of subsequent requests supports that transport.

When a proxy does decide to modify the Record-Route header field in the response, one of the operations it performs is locating the Record-Route value that it had inserted. If the request spiraled, and the proxy inserted a Record-Route value in each iteration of the spiral, locating the correct value in the response (which must be the proper iteration in the reverse direction) is tricky. The rules above recommend that a proxy wishing to rewrite Record-Route header field values insert sufficiently distinct URIs into the Record-Route header field so that the right one may be selected for rewriting. A RECOMMENDED mechanism to achieve this is for the proxy to append a unique identifier for the proxy instance to the user portion of the URI.

When the response arrives, the proxy modifies the first Record-Route whose identifier matches the proxy instance. The modification results in a URI without this piece of data appended to the user portion of the URI. Upon the next iteration, the same algorithm (find the topmost Record-Route header field value with the parameter) will correctly extract the next Record-Route header field value inserted by that proxy.

Not every response to a request to which a proxy adds a Record-Route header field value will contain a Record-Route header field. If the response does contain a Record-Route header field, it will contain the value the proxy added.

9. Forward response

After performing the processing described in steps "Aggregate Authorization Header Field Values" through "Record-Route", the proxy MAY perform any feature specific manipulations on the selected response. The proxy MUST NOT add to, modify, or remove the message body. Unless otherwise specified, the proxy MUST NOT remove any header field values other than the Via header field value discussed in Section 16.7 Item 3. In particular, the proxy MUST NOT remove any "received" parameter it may have added to the next Via header field value while processing the request associated with this response. The proxy MUST pass the response to the server transaction associated with the response context. This will result in the response being sent to the location now indicated in the topmost Via header field value. If the server transaction is no longer available to handle the transmission, the element MUST forward the response statelessly by sending it to the server transport. The server transaction might indicate failure to send the response or signal a timeout in its state machine. These errors would be logged for diagnostic purposes as appropriate, but the protocol requires no remedial action from the proxy.

The proxy MUST maintain the response context until all of its associated transactions have been terminated, even after forwarding a final response.

10. Generate CANCELs

If the forwarded response was a final response, the proxy MUST generate a CANCEL request for all pending client transactions associated with this response context. A proxy SHOULD also generate a CANCEL request for all pending client transactions associated with this response context when it receives a 6xx response. A pending client transaction is one that has received a provisional response, but no final response (it is in the proceeding state) and has not had an associated CANCEL generated for it. Generating CANCEL requests is described in Section 9.1.

The requirement to CANCEL pending client transactions upon forwarding a final response does not guarantee that an endpoint will not receive multiple 200 (OK) responses to an INVITE. 200 (OK) responses on more than one branch may be generated before the CANCEL requests can be sent and processed. Further, it is reasonable to expect that a future extension may override this requirement to issue CANCEL requests.

## 16.8 Processing Timer C

If timer C should fire, the proxy MUST either reset the timer with any value it chooses, or terminate the client transaction. If the client transaction has received a provisional response, the proxy MUST generate a CANCEL request matching that transaction. If the client transaction has not received a provisional response, the proxy MUST behave as if the transaction received a 408 (Request Timeout) response.

Allowing the proxy to reset the timer allows the proxy to dynamically extend the transaction's lifetime based on current conditions (such as utilization) when the timer fires.

## 16.9 Handling Transport Errors

If the transport layer notifies a proxy of an error when it tries to forward a request (see Section 18.4), the proxy MUST behave as if the forwarded request received a 503 (Service Unavailable) response.

If the proxy is notified of an error when forwarding a response, it drops the response. The proxy SHOULD NOT cancel any outstanding client transactions associated with this response context due to this notification.

> If a proxy cancels its outstanding client transactions, a single malicious or misbehaving client can cause all transactions to fail through its Via header field.

## 16.10 CANCEL Processing

A stateful proxy MAY generate a CANCEL to any other request it has generated at any time (subject to receiving a provisional response to that request as described in section 9.1). A proxy MUST cancel any pending client transactions associated with a response context when it receives a matching CANCEL request.

A stateful proxy MAY generate CANCEL requests for pending INVITE client transactions based on the period specified in the INVITE's Expires header field elapsing. However, this is generally unnecessary since the endpoints involved will take care of signaling the end of the transaction.

While a CANCEL request is handled in a stateful proxy by its own server transaction, a new response context is not created for it. Instead, the proxy layer searches its existing response contexts for the server transaction handling the request associated with this CANCEL. If a matching response context is found, the element MUST immediately return a 200 (OK) response to the CANCEL request. In this case, the element is acting as a user agent server as defined in Section 8.2. Furthermore, the element MUST generate CANCEL requests for all pending client transactions in the context as described in Section 16.7 step 10.

If a response context is not found, the element does not have any knowledge of the request to apply the CANCEL to. It MUST statelessly forward the CANCEL request (it may have statelessly forwarded the associated request previously).

## 16.11 Stateless Proxy

When acting statelessly, a proxy is a simple message forwarder. Much of the processing performed when acting statelessly is the same as when behaving statefully. The differences are detailed here.

A stateless proxy does not have any notion of a transaction, or of the response context used to describe stateful proxy behavior. Instead, the stateless proxy takes messages, both requests and responses, directly from the transport layer (See section 18). As a result, stateless proxies do not retransmit messages on their own. They do, however, forward all retransmissions they receive (they do not have the ability to distinguish a retransmission from the original message). Furthermore, when handling a request statelessly, an element MUST NOT generate its own 100 (Trying) or any other provisional response.

A stateless proxy MUST validate a request as described in Section 16.3

A stateless proxy MUST follow the request processing steps described in Sections 16.4 through 16.5 with the following exception:

- o A stateless proxy MUST choose one and only one target from the target set. This choice MUST only rely on fields in the message and time-invariant properties of the server. In particular, a retransmitted request MUST be forwarded to the same destination each time it is processed. Furthermore, CANCEL and non-Routed ACK requests MUST generate the same choice as their associated INVITE.

A stateless proxy MUST follow the request processing steps described in Section 16.6 with the following exceptions:

- o The requirement for unique branch IDs across space and time applies to stateless proxies as well. However, a stateless proxy cannot simply use a random number generator to compute the first component of the branch ID, as described in Section 16.6 bullet 8. This is because retransmissions of a request need to have the same value, and a stateless proxy cannot tell a retransmission from the original request. Therefore, the component of the branch parameter that makes it unique MUST be the same each time a retransmitted request is forwarded. Thus for a stateless proxy, the branch parameter MUST be computed as a combinatoric function of message parameters which are invariant on retransmission.

The stateless proxy MAY use any technique it likes to guarantee uniqueness of its branch IDs across transactions. However, the following procedure is RECOMMENDED. The proxy examines the branch ID in the topmost Via header field of the received request. If it begins with the magic cookie, the first component of the branch ID of the outgoing request is computed as a hash of the received branch ID. Otherwise, the first component of the branch ID is computed as a hash of the topmost Via, the tag in the To header field, the tag in the From header field, the Call-ID header field, the CSeq number (but not method), and the Request-URI from the received request. One of these fields will always vary across two different transactions.

- All other message transformations specified in Section 16.6 MUST result in the same transformation of a retransmitted request. In particular, if the proxy inserts a Record-Route value or pushes URIs into the Route header field, it MUST place the same values in retransmissions of the request. As for the Via branch parameter, this implies that the transformations MUST be based on time-invariant configuration or retransmission-invariant properties of the request.

- A stateless proxy determines where to forward the request as described for stateful proxies in Section 16.6 Item 10. The request is sent directly to the transport layer instead of through a client transaction.

  Since a stateless proxy must forward retransmitted requests to the same destination and add identical branch parameters to each of them, it can only use information from the message itself and time-invariant configuration data for those calculations. If the configuration state is not time-invariant (for example, if a routing table is updated) any requests that could be affected by the change may not be forwarded statelessly during an interval equal to the transaction timeout window before or after the change. The method of processing the affected requests in that interval is an implementation decision. A common solution is to forward them transaction statefully.

Stateless proxies MUST NOT perform special processing for CANCEL requests. They are processed by the above rules as any other requests. In particular, a stateless proxy applies the same Route header field processing to CANCEL requests that it applies to any other request.

Response processing as described in Section 16.7 does not apply to a proxy behaving statelessly. When a response arrives at a stateless proxy, the proxy MUST inspect the sent-by value in the first (topmost) Via header field value. If that address matches the proxy, (it equals a value this proxy has inserted into previous requests) the proxy MUST remove that header field value from the response and forward the result to the location indicated in the next Via header field value. The proxy MUST NOT add to, modify, or remove the message body. Unless specified otherwise, the proxy MUST NOT remove any other header field values. If the address does not match the proxy, the message MUST be silently discarded.

## 16.12 Summary of Proxy Route Processing

In the absence of local policy to the contrary, the processing a proxy performs on a request containing a Route header field can be summarized in the following steps.

1. The proxy will inspect the Request-URI. If it indicates a resource owned by this proxy, the proxy will replace it with the results of running a location service. Otherwise, the proxy will not change the Request-URI.

2. The proxy will inspect the URI in the topmost Route header field value. If it indicates this proxy, the proxy removes it from the Route header field (this route node has been reached).

3. The proxy will forward the request to the resource indicated by the URI in the topmost Route header field value or in the Request-URI if no Route header field is present. The proxy determines the address, port and transport to use when forwarding the request by applying the procedures in [4] to that URI.

If no strict-routing elements are encountered on the path of the request, the Request-URI will always indicate the target of the request.

### 16.12.1 Examples

#### 16.12.1.1 Basic SIP Trapezoid

This scenario is the basic SIP trapezoid, U1 -> P1 -> P2 -> U2, with both proxies record-routing. Here is the flow.

U1 sends:

```
INVITE sip:callee@domain.com SIP/2.0
Contact: sip:caller@u1.example.com
```

to P1. P1 is an outbound proxy. P1 is not responsible for domain.com, so it looks it up in DNS and sends it there. It also adds a Record-Route header field value:

```
INVITE sip:callee@domain.com SIP/2.0
Contact: sip:caller@u1.example.com
Record-Route: <sip:p1.example.com;lr>
```

P2 gets this. It is responsible for domain.com so it runs a location service and rewrites the Request-URI. It also adds a Record-Route header field value. There is no Route header field, so it resolves the new Request-URI to determine where to send the request:

```
INVITE sip:callee@u2.domain.com SIP/2.0
Contact: sip:caller@u1.example.com
Record-Route: <sip:p2.domain.com;lr>
Record-Route: <sip:p1.example.com;lr>
```

The callee at u2.domain.com gets this and responds with a 200 OK:

```
SIP/2.0 200 OK
Contact: sip:callee@u2.domain.com
Record-Route: <sip:p2.domain.com;lr>
Record-Route: <sip:p1.example.com;lr>
```

The callee at u2 also sets its dialog state’s remote target URI to sip:caller@u1.example.com and its route set to:

```
(<sip:p2.domain.com;lr>,<sip:p1.example.com;lr>)
```

This is forwarded by P2 to P1 to U1 as normal. Now, U1 sets its dialog state’s remote target URI to sip:callee@u2.domain.com and its route set to:

```
(<sip:p1.example.com;lr>,<sip:p2.domain.com;lr>)
```

Since all the route set elements contain the lr parameter, U1 constructs the following BYE request:

```
BYE sip:callee@u2.domain.com SIP/2.0
Route: <sip:p1.example.com;lr>,<sip:p2.domain.com;lr>
```

As any other element (including proxies) would do, it resolves the URI in the topmost Route header field value using DNS to determine where to send the request. This goes to P1. P1 notices that it is not responsible for the resource indicated in the Request-URI so it doesn't change it. It does see that it is the first value in the Route header field, so it removes that value, and forwards the request to P2:

```
BYE sip:callee@u2.domain.com SIP/2.0
Route: <sip:p2.domain.com;lr>
```

P2 also notices it is not responsible for the resource indicated by the Request-URI (it is responsible for domain.com, not u2.domain.com), so it doesn't change it. It does see itself in the first Route header field value, so it removes it and forwards the following to u2.domain.com based on a DNS lookup against the Request-URI:

```
BYE sip:callee@u2.domain.com SIP/2.0
```

16.12.1.2 Traversing a Strict-Routing Proxy

In this scenario, a dialog is established across four proxies, each of which adds Record-Route header field values. The third proxy implements the strict-routing procedures specified in RFC 2543 and many works in progress.

```
U1->P1->P2->P3->P4->U2
```

The INVITE arriving at U2 contains:

```
INVITE sip:callee@u2.domain.com SIP/2.0
Contact: sip:caller@u1.example.com
Record-Route: <sip:p4.domain.com;lr>
Record-Route: <sip:p3.middle.com>
Record-Route: <sip:p2.example.com;lr>
Record-Route: <sip:p1.example.com;lr>
```

Which U2 responds to with a 200 OK. Later, U2 sends the following BYE request to P4 based on the first Route header field value.

```
BYE sip:caller@u1.example.com SIP/2.0
Route: <sip:p4.domain.com;lr>
Route: <sip:p3.middle.com>
Route: <sip:p2.example.com;lr>
Route: <sip:p1.example.com;lr>
```

P4 is not responsible for the resource indicated in the Request-URI so it will leave it alone. It notices that it is the element in the first Route header field value so it removes it. It then prepares to send the request based on the now first Route header field value of sip:p3.middle.com, but it notices that this URI does not contain the lr parameter, so before sending, it reformats the request to be:

```
BYE sip:p3.middle.com SIP/2.0
Route: <sip:p2.example.com;lr>
Route: <sip:p1.example.com;lr>
Route: <sip:caller@u1.example.com>
```

P3 is a strict router, so it forwards the following to P2:

```
BYE sip:p2.example.com;lr SIP/2.0
Route: <sip:p1.example.com;lr>
Route: <sip:caller@u1.example.com>
```

P2 sees the request-URI is a value it placed into a Record-Route header field, so before further processing, it rewrites the request to be:

```
BYE sip:caller@u1.example.com SIP/2.0
Route: <sip:p1.example.com;lr>
```

P2 is not responsible for u1.example.com, so it sends the request to P1 based on the resolution of the Route header field value.

P1 notices itself in the topmost Route header field value, so it removes it, resulting in:

```
BYE sip:caller@u1.example.com SIP/2.0
```

Since P1 is not responsible for u1.example.com and there is no Route header field, P1 will forward the request to u1.example.com based on the Request-URI.

16.12.1.3 Rewriting Record-Route Header Field Values

In this scenario, U1 and U2 are in different private namespaces and they enter a dialog through a proxy P1, which acts as a gateway between the namespaces.

```
U1->P1->U2
```

U1 sends:

```
INVITE sip:callee@gateway.leftprivatespace.com SIP/2.0
Contact: <sip:caller@u1.leftprivatespace.com>
```

P1 uses its location service and sends the following to U2:

```
INVITE sip:callee@rightprivatespace.com SIP/2.0
Contact: <sip:caller@u1.leftprivatespace.com>
Record-Route: <sip:gateway.rightprivatespace.com;lr>
```

U2 sends this 200 (OK) back to P1:

```
SIP/2.0 200 OK
Contact: <sip:callee@u2.rightprivatespace.com>
Record-Route: <sip:gateway.rightprivatespace.com;lr>
```

P1 rewrites its Record-Route header parameter to provide a value that U1 will find useful, and sends the following to U1:

```
SIP/2.0 200 OK
Contact: <sip:callee@u2.rightprivatespace.com>
Record-Route: <sip:gateway.leftprivatespace.com;lr>
```

Later, U1 sends the following BYE request to P1:

```
BYE sip:callee@u2.rightprivatespace.com SIP/2.0
Route: <sip:gateway.leftprivatespace.com;lr>
```

which P1 forwards to U2 as:

```
BYE sip:callee@u2.rightprivatespace.com SIP/2.0
```

# 17 Transactions

SIP is a transactional protocol: interactions between components take place in a series of independent message exchanges. Specifically, a SIP transaction consists of a single request and any responses to that request, which include zero or more provisional responses and one or more final responses. In the case of a transaction where the request was an INVITE (known as an INVITE transaction), the transaction also includes the ACK only if the final response was not a 2xx response. If the response was a 2xx, the ACK is not considered part of the transaction.

The reason for this separation is rooted in the importance of delivering all 200 (OK) responses to an INVITE to the UAC. To deliver them all to the UAC, the UAS alone takes responsibility for retransmitting them (see Section 13.3.1.4), and the UAC alone takes responsibility for acknowledging them with ACK (see Section 13.2.2.4). Since this ACK is retransmitted only by the UAC, it is effectively considered its own transaction.

Transactions have a client side and a server side. The client side is known as a client transaction and the server side as a server transaction. The client transaction sends the request, and the server transaction sends the response. The client and server transactions are logical functions that are embedded in any number of elements. Specifically, they exist within user agents and stateful proxy servers. Consider the example in Section 4. In this example, the UAC executes the client transaction, and its outbound proxy executes the server transaction. The outbound proxy also executes a client transaction, which sends the request to a server transaction in the inbound proxy. That proxy also executes a client transaction, which in turn sends the request to a server transaction in the UAS. This is shown in Figure 4.

```
+---------+        +---------+        +---------+        +---------+
|      +-+|Request |+-+   +-+|Request |+-+   +-+|Request |+-+      |
|      |C||------->||S|   |C||------->||S|   |C||------->||S|      |
|      |l||        ||e|   |l||        ||e|   |l||        ||e|      |
|      |i||        ||r|   |i||        ||r|   |i||        ||r|      |
|      |e||        ||v|   |e||        ||v|   |e||        ||v|      |
|      |n||        ||e|   |n||        ||e|   |n||        ||e|      |
|      |t||        ||r|   |t||        ||r|   |t||        ||r|      |
|      | ||        || |   | ||        || |   | ||        || |      |
|      |T||        ||T|   |T||        ||T|   |T||        ||T|      |
|      |r||        ||r|   |r||        ||r|   |r||        ||r|      |
|      |a||        ||a|   |a||        ||a|   |a||        ||a|      |
|      |n||        ||n|   |n||        ||n|   |n||        ||n|      |
|      |s||Response||s|   |s||Response||s|   |s||Response||s|      |
|      +-+|<-------|+-+   +-+|<-------|+-+   +-+|<-------|+-+      |
+---------+        +---------+        +---------+        +---------+
   UAC               Outbound           Inbound              UAS
                     Proxy               Proxy
```

Figure 4: Transaction relationships

A stateless proxy does not contain a client or server transaction. The transaction exists between the UA or stateful proxy on one side, and the UA or stateful proxy on the other side. As far as SIP transactions are concerned, stateless proxies are effectively transparent. The purpose of the client transaction is to receive a request from the element in which the client is embedded (call this element the "Transaction User" or TU; it can be a UA or a stateful proxy), and reliably deliver the request to a server transaction.

The client transaction is also responsible for receiving responses and delivering them to the TU, filtering out any response retransmissions or disallowed responses (such as a response to ACK). Additionally, in the case of an INVITE request, the client transaction is responsible for generating the ACK request for any final response accepting a 2xx response.

Similarly, the purpose of the server transaction is to receive requests from the transport layer and deliver them to the TU. The server transaction filters any request retransmissions from the network. The server transaction accepts responses from the TU and delivers them to the transport layer for transmission over the network. In the case of an INVITE transaction, it absorbs the ACK request for any final response excepting a 2xx response.

The 2xx response and its ACK receive special treatment. This response is retransmitted only by a UAS, and its ACK generated only by the UAC. This end-to-end treatment is needed so that a caller knows the entire set of users that have accepted the call. Because of this special handling, retransmissions of the 2xx response are handled by the UA core, not the transaction layer. Similarly, generation of the ACK for the 2xx is handled by the UA core. Each proxy along the path merely forwards each 2xx response to INVITE and its corresponding ACK.

## 17.1 Client Transaction

The client transaction provides its functionality through the maintenance of a state machine.

The TU communicates with the client transaction through a simple interface. When the TU wishes to initiate a new transaction, it creates a client transaction and passes it the SIP request to send and an IP address, port, and transport to which to send it. The client transaction begins execution of its state machine. Valid responses are passed up to the TU from the client transaction.

There are two types of client transaction state machines, depending on the method of the request passed by the TU. One handles client transactions for INVITE requests. This type of machine is referred to as an INVITE client transaction. Another type handles client transactions for all requests except INVITE and ACK. This is referred to as a non-INVITE client transaction. There is no client transaction for ACK. If the TU wishes to send an ACK, it passes one directly to the transport layer for transmission.

The INVITE transaction is different from those of other methods because of its extended duration. Normally, human input is required in order to respond to an INVITE. The long delays expected for sending a response argue for a three-way handshake. On the other hand, requests of other methods are expected to complete rapidly. Because of the non-INVITE transaction's reliance on a two-way handshake, TUs SHOULD respond immediately to non-INVITE requests.

### 17.1.1 INVITE Client Transaction

#### 17.1.1.1 Overview of INVITE Transaction

The INVITE transaction consists of a three-way handshake. The client transaction sends an INVITE, the server transaction sends responses, and the client transaction sends an ACK. For unreliable transports (such as UDP), the client transaction retransmits requests at an interval that starts at T1 seconds and doubles after every retransmission. T1 is an estimate of the round-trip time (RTT), and it defaults to 500 ms. Nearly all of the transaction timers described here scale with T1, and changing T1 adjusts their values. The request is not retransmitted over reliable transports. After receiving a 1xx response, any retransmissions cease altogether, and the client waits for further responses. The server transaction can send additional 1xx responses, which are not transmitted reliably by the server transaction. Eventually, the server transaction decides to send a final response. For unreliable transports, that response is retransmitted periodically, and for reliable transports, it is sent once. For each final response that is received at the client transaction, the client transaction sends an ACK, the purpose of which is to quench retransmissions of the response.

#### 17.1.1.2 Formal Description

The state machine for the INVITE client transaction is shown in Figure 5. The initial state, "calling", MUST be entered when the TU initiates a new client transaction with an INVITE request. The client transaction MUST pass the request to the transport layer for transmission (see Section 18). If an unreliable transport is being used, the client transaction MUST start timer A with a value of T1. If a reliable transport is being used, the client transaction SHOULD NOT start timer A (Timer A controls request retransmissions). For any transport, the client transaction MUST start timer B with a value of 64*T1 seconds (Timer B controls transaction timeouts).

When timer A fires, the client transaction MUST retransmit the request by passing it to the transport layer, and MUST reset the timer with a value of 2*T1. The formal definition of retransmit within the context of the transaction layer is to take the message previously sent to the transport layer and pass it to the transport layer once more.

When timer A fires 2*T1 seconds later, the request MUST be retransmitted again (assuming the client transaction is still in this state). This process MUST continue so that the request is retransmitted with intervals that double after each transmission. These retransmissions SHOULD only be done while the client transaction is in the "calling" state.

The default value for T1 is 500 ms. T1 is an estimate of the RTT between the client and server transactions. Elements MAY (though it is NOT RECOMMENDED) use smaller values of T1 within closed, private networks that do not permit general Internet connection. T1 MAY be chosen larger, and this is RECOMMENDED if it is known in advance (such as on high latency access links) that the RTT is larger. Whatever the value of T1, the exponential backoffs on retransmissions described in this section MUST be used.

If the client transaction is still in the "Calling" state when timer B fires, the client transaction SHOULD inform the TU that a timeout has occurred. The client transaction MUST NOT generate an ACK. The value of 64*T1 is equal to the amount of time required to send seven requests in the case of an unreliable transport.

If the client transaction receives a provisional response while in the "Calling" state, it transitions to the "Proceeding" state. In the "Proceeding" state, the client transaction SHOULD NOT retransmit the request any longer. Furthermore, the provisional response MUST be passed to the TU. Any further provisional responses MUST be passed up to the TU while in the "Proceeding" state.

When in either the "Calling" or "Proceeding" states, reception of a response with status code from 300-699 MUST cause the client transaction to transition to "Completed". The client transaction MUST pass the received response up to the TU, and the client transaction MUST generate an ACK request, even if the transport is reliable (guidelines for constructing the ACK from the response are given in Section 17.1.1.3) and then pass the ACK to the transport layer for transmission. The ACK MUST be sent to the same address, port, and transport to which the original request was sent. The client transaction SHOULD start timer D when it enters the "Completed" state, with a value of at least 32 seconds for unreliable transports, and a value of zero seconds for reliable transports. Timer D reflects the amount of time that the server transaction can remain in the "Completed" state when unreliable transports are used. This is equal to Timer H in the INVITE server transaction, whose default is 64*T1. However, the client transaction does not know the value of T1 in use by the server transaction, so an absolute minimum of 32s is used instead of basing Timer D on T1.

Any retransmissions of the final response that are received while in the "Completed" state MUST cause the ACK to be re-passed to the transport layer for retransmission, but the newly received response MUST NOT be passed up to the TU. A retransmission of the response is defined as any response which would match the same client transaction based on the rules of Section 17.1.3.

```
                               |INVITE from TU
             Timer A fires     |INVITE sent
             Reset A,          V                      Timer B fires
             INVITE sent +-----------+                or Transport Err.
               +---------|           |---------------+inform TU
               |         |  Calling  |               |
               +-------->|           |-------------->|
                         +-----------+ 2xx           |
                            |  |       2xx to TU     |
                            |  |1xx                  |
    300-699 +---------------+  |1xx to TU            |
   ACK sent |                  |                     |
resp. to TU |  1xx             V                     |
            |  1xx to TU  -----------+               |
            |  +---------|           |               |
            |  |         |Proceeding |-------------->|
            |  +-------->|           | 2xx           |
            |            +-----------+ 2xx to TU     |
            |       300-699    |                     |
            |       ACK sent,  |                     |
            |       resp. to TU|                     |
            |                  |                     |      NOTE:
            |  300-699         V                     |
            |  ACK sent  +-----------+Transport Err. |  transitions
            |  +---------|           |Inform TU      |  labeled with
            |  |         | Completed |-------------->|  the event
            |  +-------->|           |               |  over the action
            |            +-----------+               |  to take
            |              ^   |                     |
            |              |   | Timer D fires       |
            +--------------+   | -                   |
                               |                     |
                               V                     |
                         +-----------+               |
                         |           |               |
                         | Terminated|<--------------+
                         |           |
                         +-----------+
```

Figure 5: INVITE client transaction

If timer D fires while the client transaction is in the "Completed" state, the client transaction MUST move to the terminated state.

When in either the "Calling" or "Proceeding" states, reception of a 2xx response MUST cause the client transaction to enter the "Terminated" state, and the response MUST be passed up to the TU. The handling of this response depends on whether the TU is a proxy core or a UAC core. A UAC core will handle generation of the ACK for this response, while a proxy core will always forward the 200 (OK) upstream. The differing treatment of 200 (OK) between proxy and UAC is the reason that handling of it does not take place in the transaction layer.

The client transaction MUST be destroyed the instant it enters the "Terminated" state. This is actually necessary to guarantee correct operation. The reason is that 2xx responses to an INVITE are treated differently; each one is forwarded by proxies, and the ACK handling in a UAC is different. Thus, each 2xx needs to be passed to a proxy core (so that it can be forwarded) and to a UAC core (so it can be acknowledged). No transaction layer processing takes place. Whenever a response is received by the transport, if the transport layer finds no matching client transaction (using the rules of Section 17.1.3), the response is passed directly to the core. Since the matching client transaction is destroyed by the first 2xx, subsequent 2xx will find no match and therefore be passed to the core.

#### 17.1.1.3 Construction of the ACK Request

This section specifies the construction of ACK requests sent within the client transaction. A UAC core that generates an ACK for 2xx MUST instead follow the rules described in Section 13.

The ACK request constructed by the client transaction MUST contain values for the Call-ID, From, and Request-URI that are equal to the values of those header fields in the request passed to the transport by the client transaction (call this the "original request"). The To header field in the ACK MUST equal the To header field in the response being acknowledged, and therefore will usually differ from the To header field in the original request by the addition of the tag parameter. The ACK MUST contain a single Via header field, and this MUST be equal to the top Via header field of the original request. The CSeq header field in the ACK MUST contain the same value for the sequence number as was present in the original request, but the method parameter MUST be equal to "ACK".

If the INVITE request whose response is being acknowledged had Route header fields, those header fields MUST appear in the ACK. This is to ensure that the ACK can be routed properly through any downstream stateless proxies.

Although any request MAY contain a body, a body in an ACK is special since the request cannot be rejected if the body is not understood. Therefore, placement of bodies in ACK for non-2xx is NOT RECOMMENDED, but if done, the body types are restricted to any that appeared in the INVITE, assuming that the response to the INVITE was not 415. If it was, the body in the ACK MAY be any type listed in the Accept header field in the 415.

For example, consider the following request:

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKkjshdyff
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=88sja8x
Max-Forwards: 70
Call-ID: 987asjd97y7atg
CSeq: 986759 INVITE
```

The ACK request for a non-2xx final response to this request would look like this:

```
ACK sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKkjshdyff
To: Bob <sip:bob@biloxi.com>;tag=99sa0xk
From: Alice <sip:alice@atlanta.com>;tag=88sja8x
Max-Forwards: 70
Call-ID: 987asjd97y7atg
CSeq: 986759 ACK
```

### 17.1.2 Non-INVITE Client Transaction

#### 17.1.2.1 Overview of the non-INVITE Transaction

Non-INVITE transactions do not make use of ACK. They are simple request-response interactions. For unreliable transports, requests are retransmitted at an interval which starts at T1 and doubles until it hits T2. If a provisional response is received, retransmissions continue for unreliable transports, but at an interval of T2. The server transaction retransmits the last response it sent, which can be a provisional or final response, only when a retransmission of the request is received. This is why request retransmissions need to continue even after a provisional response; they are to ensure reliable delivery of the final response.

Unlike an INVITE transaction, a non-INVITE transaction has no special handling for the 2xx response. The result is that only a single 2xx response to a non-INVITE is ever delivered to a UAC.

#### 17.1.2.2 Formal Description

The state machine for the non-INVITE client transaction is shown in Figure 6. It is very similar to the state machine for INVITE.

The "Trying" state is entered when the TU initiates a new client transaction with a request. When entering this state, the client transaction SHOULD set timer F to fire in 64*T1 seconds. The request MUST be passed to the transport layer for transmission. If an unreliable transport is in use, the client transaction MUST set timer E to fire in T1 seconds. If timer E fires while still in this state, the timer is reset, but this time with a value of MIN(2*T1, T2). When the timer fires again, it is reset to a MIN(4*T1, T2). This process continues so that retransmissions occur with an exponentially increasing interval that caps at T2. The default value of T2 is 4s, and it represents the amount of time a non-INVITE server transaction will take to respond to a request, if it does not respond immediately. For the default values of T1 and T2, this results in intervals of 500 ms, 1 s, 2 s, 4 s, 4 s, 4 s, etc.

If Timer F fires while the client transaction is still in the "Trying" state, the client transaction SHOULD inform the TU about the timeout, and then it SHOULD enter the "Terminated" state. If a provisional response is received while in the "Trying" state, the response MUST be passed to the TU, and then the client transaction SHOULD move to the "Proceeding" state. If a final response (status codes 200-699) is received while in the "Trying" state, the response MUST be passed to the TU, and the client transaction MUST transition to the "Completed" state.

If Timer E fires while in the "Proceeding" state, the request MUST be passed to the transport layer for retransmission, and Timer E MUST be reset with a value of T2 seconds. If timer F fires while in the "Proceeding" state, the TU MUST be informed of a timeout, and the client transaction MUST transition to the terminated state. If a final response (status codes 200-699) is received while in the "Proceeding" state, the response MUST be passed to the TU, and the client transaction MUST transition to the "Completed" state.

Once the client transaction enters the "Completed" state, it MUST set Timer K to fire in T4 seconds for unreliable transports, and zero seconds for reliable transports. The "Completed" state exists to buffer any additional response retransmissions that may be received (which is why the client transaction remains there only for unreliable transports). T4 represents the amount of time the network will take to clear messages between client and server transactions. The default value of T4 is 5s. A response is a retransmission when it matches the same transaction, using the rules specified in Section 17.1.3. If Timer K fires while in this state, the client transaction MUST transition to the "Terminated" state.

Once the transaction is in the terminated state, it MUST be destroyed immediately.

### 17.1.3 Matching Responses to Client Transactions

When the transport layer in the client receives a response, it has to determine which client transaction will handle the response, so that the processing of Sections 17.1.1 and 17.1.2 can take place. The branch parameter in the top Via header field is used for this purpose. A response matches a client transaction under two conditions:

1. If the response has the same value of the branch parameter in the top Via header field as the branch parameter in the top Via header field of the request that created the transaction.

2. If the method parameter in the CSeq header field matches the method of the request that created the transaction. The method is needed since a CANCEL request constitutes a different transaction, but shares the same value of the branch parameter.

If a request is sent via multicast, it is possible that it will generate multiple responses from different servers. These responses will all have the same branch parameter in the topmost Via, but vary in the To tag. The first response received, based on the rules above, will be used, and others will be viewed as retransmissions. That is not an error; multicast SIP provides only a rudimentary "single-hop-discovery-like" service that is limited to processing a single response. See Section 18.1.1 for details.

### 17.1.4 Handling Transport Errors

```
                                   |Request from TU
                                   |send request
               Timer E             V
               send request  +-----------+
                   +---------|           |-------------------+
                   |         |  Trying   |  Timer F          |
                   +-------->|           |  or Transport Err.|
                             +-----------+  inform TU        |
                200-699         |  |                         |
                resp. to TU     |  |1xx                      |
                +---------------+  |resp. to TU              |
                |                  |                         |
                |   Timer E        V       Timer F           |
                |   send req +-----------+ or Transport Err. |
                |  +---------|           | inform TU         |
                |  |         |Proceeding |------------------>|
                |  +-------->|           |-----+             |
                |            +-----------+     |1xx          |
                |              |      ^        |resp to TU   |
                | 200-699      |      +--------+             |
                | resp. to TU  |                             |
                |              |                             |
                |              V                             |
                |            +-----------+                   |
                |            |           |                   |
                |            | Completed |                   |
                |            |           |                   |
                |            +-----------+                   |
                |              ^   |                         |
                |              |   | Timer K                 |
                +--------------+   | -                       |
                                   |                         |
                                   V                         |
             NOTE:           +-----------+                   |
                             |           |                   |
         transitions         | Terminated|<------------------+
         labeled with        |           |
         the event           +-----------+
         over the action
         to take
```

Figure 6: non-INVITE client transaction

When the client transaction sends a request to the transport layer to be sent, the following procedures are followed if the transport layer indicates a failure.

The client transaction SHOULD inform the TU that a transport failure has occurred, and the client transaction SHOULD transition directly to the "Terminated" state. The TU will handle the failover mechanisms described in [4].

## 17.2 Server Transaction

The server transaction is responsible for the delivery of requests to the TU and the reliable transmission of responses. It accomplishes this through a state machine. Server transactions are created by the core when a request is received, and transaction handling is desired for that request (this is not always the case).

As with the client transactions, the state machine depends on whether the received request is an INVITE request.

### 17.2.1 INVITE Server Transaction

The state diagram for the INVITE server transaction is shown in Figure 7.

When a server transaction is constructed for a request, it enters the "Proceeding" state. The server transaction MUST generate a 100 (Trying) response unless it knows that the TU will generate a provisional or final response within 200 ms, in which case it MAY generate a 100 (Trying) response. This provisional response is needed to quench request retransmissions rapidly in order to avoid network congestion. The 100 (Trying) response is constructed according to the procedures in Section 8.2.6, except that the insertion of tags in the To header field of the response (when none was present in the request) is downgraded from MAY to SHOULD NOT. The request MUST be passed to the TU.

The TU passes any number of provisional responses to the server transaction. So long as the server transaction is in the "Proceeding" state, each of these MUST be passed to the transport layer for transmission. They are not sent reliably by the transaction layer (they are not retransmitted by it) and do not cause a change in the state of the server transaction. If a request retransmission is received while in the "Proceeding" state, the most recent provisional response that was received from the TU MUST be passed to the transport layer for retransmission. A request is a retransmission if it matches the same server transaction based on the rules of Section 17.2.3.

If, while in the "Proceeding" state, the TU passes a 2xx response to the server transaction, the server transaction MUST pass this response to the transport layer for transmission. It is not retransmitted by the server transaction; retransmissions of 2xx responses are handled by the TU. The server transaction MUST then transition to the "Terminated" state.

While in the "Proceeding" state, if the TU passes a response with status code from 300 to 699 to the server transaction, the response MUST be passed to the transport layer for transmission, and the state machine MUST enter the "Completed" state. For unreliable transports, timer G is set to fire in T1 seconds, and is not set to fire for reliable transports.

> This is a change from RFC 2543, where responses were always retransmitted, even over reliable transports.

When the "Completed" state is entered, timer H MUST be set to fire in 64*T1 seconds for all transports. Timer H determines when the server transaction abandons retransmitting the response. Its value is chosen to equal Timer B, the amount of time a client transaction will continue to retry sending a request. If timer G fires, the response is passed to the transport layer once more for retransmission, and timer G is set to fire in MIN(2*T1, T2) seconds. From then on, when timer G fires, the response is passed to the transport again for transmission, and timer G is reset with a value that doubles, unless that value exceeds T2, in which case it is reset with the value of T2. This is identical to the retransmit behavior for requests in the "Trying" state of the non-INVITE client transaction. Furthermore, while in the "Completed" state, if a request retransmission is received, the server SHOULD pass the response to the transport for retransmission.

If an ACK is received while the server transaction is in the "Completed" state, the server transaction MUST transition to the "Confirmed" state. As Timer G is ignored in this state, any retransmissions of the response will cease.

If timer H fires while in the "Completed" state, it implies that the ACK was never received. In this case, the server transaction MUST transition to the "Terminated" state, and MUST indicate to the TU that a transaction failure has occurred.

```
                               |INVITE
                               |pass INV to TU
            INVITE             V send 100 if TU won't in 200ms
            send response+-----------+
                +--------|           |--------+101-199 from TU
                |        | Proceeding|        |send response
                +------->|           |<-------+
                         |           |          Transport Err.
                         |           |          Inform TU
                         |           |--------------->+
                         +-----------+                |
            300-699 from TU |     |2xx from TU        |
            send response   |     |send response      |
                            |     +------------------>+
                            |                         |
            INVITE          V          Timer G fires  |
            send response+-----------+ send response  |
                +--------|           |--------+       |
                |        | Completed |        |       |
                +------->|           |<-------+       |
                         +-----------+                |
                            |     |                   |
                        ACK |     |                   |
                        -   |     +------------------>+
                            |        Timer H fires    |
                            V        or Transport Err.|
                         +-----------+  Inform TU     |
                         |           |                |
                         | Confirmed |                |
                         |           |                |
                         +-----------+                |
                               |                      |
                               |Timer I fires         |
                               |-                     |
                               |                      |
                               V                      |
                         +-----------+                |
                         |           |                |
                         | Terminated|<---------------+
                         |           |
                         +-----------+
```

Figure 7: INVITE server transaction

The purpose of the "Confirmed" state is to absorb any additional ACK messages that arrive, triggered from retransmissions of the final response. When this state is entered, timer I is set to fire in T4 seconds for unreliable transports, and zero seconds for reliable transports. Once timer I fires, the server MUST transition to the "Terminated" state.

Once the transaction is in the "Terminated" state, it MUST be destroyed immediately. As with client transactions, this is needed to ensure reliability of the 2xx responses to INVITE.

### 17.2.2 Non-INVITE Server Transaction

The state machine for the non-INVITE server transaction is shown in Figure 8.

The state machine is initialized in the "Trying" state and is passed a request other than INVITE or ACK when initialized. This request is passed up to the TU. Once in the "Trying" state, any further request retransmissions are discarded. A request is a retransmission if it matches the same server transaction, using the rules specified in Section 17.2.3.

While in the "Trying" state, if the TU passes a provisional response to the server transaction, the server transaction MUST enter the "Proceeding" state. The response MUST be passed to the transport layer for transmission. Any further provisional responses that are received from the TU while in the "Proceeding" state MUST be passed to the transport layer for transmission. If a retransmission of the request is received while in the "Proceeding" state, the most recently sent provisional response MUST be passed to the transport layer for retransmission. If the TU passes a final response (status codes 200-699) to the server while in the "Proceeding" state, the transaction MUST enter the "Completed" state, and the response MUST be passed to the transport layer for transmission.

When the server transaction enters the "Completed" state, it MUST set Timer J to fire in 64*T1 seconds for unreliable transports, and zero seconds for reliable transports. While in the "Completed" state, the server transaction MUST pass the final response to the transport layer for retransmission whenever a retransmission of the request is received. Any other final responses passed by the TU to the server transaction MUST be discarded while in the "Completed" state. The server transaction remains in this state until Timer J fires, at which point it MUST transition to the "Terminated" state.

The server transaction MUST be destroyed the instant it enters the "Terminated" state.

### 17.2.3 Matching Requests to Server Transactions

When a request is received from the network by the server, it has to be matched to an existing transaction. This is accomplished in the following manner.

The branch parameter in the topmost Via header field of the request is examined. If it is present and begins with the magic cookie "z9hG4bK", the request was generated by a client transaction compliant to this specification. Therefore, the branch parameter will be unique across all transactions sent by that client. The request matches a transaction if:

1. the branch parameter in the request is equal to the one in the top Via header field of the request that created the transaction, and

2. the sent-by value in the top Via of the request is equal to the one in the request that created the transaction, and

3. the method of the request matches the one that created the transaction, except for ACK, where the method of the request that created the transaction is INVITE.

This matching rule applies to both INVITE and non-INVITE transactions alike.

> The sent-by value is used as part of the matching process because there could be accidental or malicious duplication of branch parameters from different clients.

If the branch parameter in the top Via header field is not present, or does not contain the magic cookie, the following procedures are used. These exist to handle backwards compatibility with RFC 2543 compliant implementations.

The INVITE request matches a transaction if the Request-URI, To tag, From tag, Call-ID, CSeq, and top Via header field match those of the INVITE request which created the transaction. In this case, the INVITE is a retransmission of the original one that created the transaction. The ACK request matches a transaction if the Request-URI, From tag, Call-ID, CSeq number (not the method), and top Via header field match those of the INVITE request which created the transaction, and the To tag of the ACK matches the To tag of the response sent by the server transaction. Matching is done based on the matching rules defined for each of those header fields. Inclusion of the tag in the To header field in the ACK matching process helps disambiguate ACK for 2xx from ACK for other responses at a proxy, which may have forwarded both responses (This can occur in unusual conditions. Specifically, when a proxy forked a request, and then crashes, the responses may be delivered to another proxy, which might end up forwarding multiple responses upstream). An ACK request that matches an INVITE transaction matched by a previous ACK is considered a retransmission of that previous ACK.

```
                                  |Request received
                                  |pass to TU
                                  V
                            +-----------+
                            |           |
                            | Trying    |-------------+
                            |           |             |
                            +-----------+             |200-699 from TU
                                  |                   |send response
                                  |1xx from TU        |
                                  |send response      |
                                  |                   |
               Request            V      1xx from TU  |
               send response+-----------+send response|
                   +--------|           |--------+    |
                   |        | Proceeding|        |    |
                   +------->|           |<-------+    |
            +<--------------|           |             |
            |Trnsprt Err    +-----------+             |
            |Inform TU            |                   |
            |                     |                   |
            |                     |200-699 from TU    |
            |                     |send response      |
            |  Request            V                   |
            |  send response+-----------+             |
            |      +--------|           |             |
            |      |        | Completed |<------------+
            |      +------->|           |
            +<--------------|           |
            |Trnsprt Err    +-----------+
            |Inform TU            |
            |                     |Timer J fires
            |                     |-
            |                     |
            |                     V
            |               +-----------+
            |               |           |
            +-------------->| Terminated|
                            |           |
                            +-----------+
```

Figure 8: non-INVITE server transaction

For all other request methods, a request is matched to a transaction if the Request-URI, To tag, From tag, Call-ID, CSeq (including the method), and top Via header field match those of the request that created the transaction.  Matching is done based on the matching rules defined for each of those header fields. When a non-INVITE request matches an existing transaction, it is a retransmission of the request that created that transaction.

Because the matching rules include the Request-URI, the server cannot match a response to a transaction. When the TU passes a response to the server transaction, it must pass it to the specific server transaction for which the response is targeted.

### 17.2.4 Handling Transport Errors

When the server transaction sends a response to the transport layer to be sent, the following procedures are followed if the transport layer indicates a failure.

First, the procedures in [4] are followed, which attempt to deliver the response to a backup. If those should all fail, based on the definition of failure in [4], the server transaction SHOULD inform the TU that a failure has occurred, and SHOULD transition to the terminated state.

## 18 Transport

The transport layer is responsible for the actual transmission of requests and responses over network transports. This includes determination of the connection to use for a request or response in the case of connection-oriented transports.

The transport layer is responsible for managing persistent connections for transport protocols like TCP and SCTP, or TLS over those, including ones opened to the transport layer. This includes connections opened by the client or server transports, so that connections are shared between client and server transport functions. These connections are indexed by the tuple formed from the address, port, and transport protocol at the far end of the connection. When a connection is opened by the transport layer, this index is set to the destination IP, port and transport. When the connection is accepted by the transport layer, this index is set to the source IP address, port number, and transport. Note that, because the source port is often ephemeral, but it cannot be known whether it is ephemeral or selected through procedures in [4], connections accepted by the transport layer will frequently not be reused. The result is that two proxies in a "peering" relationship using a connection-oriented transport frequently will have two connections in use, one for transactions initiated in each direction.

It is RECOMMENDED that connections be kept open for some implementation-defined duration after the last message was sent or received over that connection. This duration SHOULD at least equal the longest amount of time the element would need in order to bring a transaction from instantiation to the terminated state. This is to make it likely that transactions are completed over the same connection on which they are initiated (for example, request, response, and in the case of INVITE, ACK for non-2xx responses). This usually means at least 64*T1 (see Section 17.1.1.1 for a definition of T1). However, it could be larger in an element that has a TU using a large value for timer C (bullet 11 of Section 16.6), for example.

All SIP elements MUST implement UDP and TCP. SIP elements MAY implement other protocols.

> Making TCP mandatory for the UA is a substantial change from RFC 2543. It has arisen out of the need to handle larger messages, which MUST use TCP, as discussed below. Thus, even if an element never sends large messages, it may receive one and needs to be able to handle them.

## 18.1 Clients

### 18.1.1 Sending Requests

The client side of the transport layer is responsible for sending the request and receiving responses. The user of the transport layer passes the client transport the request, an IP address, port, transport, and possibly TTL for multicast destinations.

If a request is within 200 bytes of the path MTU, or if it is larger than 1300 bytes and the path MTU is unknown, the request MUST be sent using an RFC 2914 [43] congestion controlled transport protocol, such as TCP. If this causes a change in the transport protocol from the one indicated in the top Via, the value in the top Via MUST be changed. This prevents fragmentation of messages over UDP and provides congestion control for larger messages. However, implementations MUST be able to handle messages up to the maximum datagram packet size. For UDP, this size is 65,535 bytes, including IP and UDP headers.

> The 200 byte "buffer" between the message size and the MTU accommodates the fact that the response in SIP can be larger than the request. This happens due to the addition of Record-Route header field values to the responses to INVITE, for example. With the extra buffer, the response can be about 170 bytes larger than the request, and still not be fragmented on IPv4 (about 30 bytes

is consumed by IP/UDP, assuming no IPSec). 1300 is chosen when path MTU is not known, based on the assumption of a 1500 byte Ethernet MTU.

If an element sends a request over TCP because of these message size constraints, and that request would have otherwise been sent over UDP, if the attempt to establish the connection generates either an ICMP Protocol Not Supported, or results in a TCP reset, the element SHOULD retry the request, using UDP. This is only to provide backwards compatibility with RFC 2543 compliant implementations that do not support TCP. It is anticipated that this behavior will be deprecated in a future revision of this specification.

A client that sends a request to a multicast address MUST add the "maddr" parameter to its Via header field value containing the destination multicast address, and for IPv4, SHOULD add the "ttl" parameter with a value of 1. Usage of IPv6 multicast is not defined in this specification, and will be a subject of future standardization when the need arises.

These rules result in a purposeful limitation of multicast in SIP. Its primary function is to provide a "single-hop-discovery-like" service, delivering a request to a group of homogeneous servers, where it is only required to process the response from any one of them. This functionality is most useful for registrations. In fact, based on the transaction processing rules in Section 17.1.3, the client transaction will accept the first response, and view any others as retransmissions because they all contain the same Via branch identifier.

Before a request is sent, the client transport MUST insert a value of the "sent-by" field into the Via header field. This field contains an IP address or host name, and port. The usage of an FQDN is RECOMMENDED. This field is used for sending responses under certain conditions, described below. If the port is absent, the default value depends on the transport. It is 5060 for UDP, TCP and SCTP, 5061 for TLS.

For reliable transports, the response is normally sent on the connection on which the request was received. Therefore, the client transport MUST be prepared to receive the response on the same connection used to send the request. Under error conditions, the server may attempt to open a new connection to send the response. To handle this case, the transport layer MUST also be prepared to receive an incoming connection on the source IP address from which the request was sent and port number in the "sent-by" field. It also

MUST be prepared to receive incoming connections on any address and port that would be selected by a server based on the procedures described in Section 5 of [4].

For unreliable unicast transports, the client transport MUST be prepared to receive responses on the source IP address from which the request is sent (as responses are sent back to the source address) and the port number in the "sent-by" field. Furthermore, as with reliable transports, in certain cases the response will be sent elsewhere. The client MUST be prepared to receive responses on any address and port that would be selected by a server based on the procedures described in Section 5 of [4].

For multicast, the client transport MUST be prepared to receive responses on the same multicast group and port to which the request is sent (that is, it needs to be a member of the multicast group it sent the request to.)

If a request is destined to an IP address, port, and transport to which an existing connection is open, it is RECOMMENDED that this connection be used to send the request, but another connection MAY be opened and used.

If a request is sent using multicast, it is sent to the group address, port, and TTL provided by the transport user. If a request is sent using unicast unreliable transports, it is sent to the IP address and port provided by the transport user.

### 18.1.2 Receiving Responses

When a response is received, the client transport examines the top Via header field value. If the value of the "sent-by" parameter in that header field value does not correspond to a value that the client transport is configured to insert into requests, the response MUST be silently discarded.

If there are any client transactions in existence, the client transport uses the matching procedures of Section 17.1.3 to attempt to match the response to an existing transaction. If there is a match, the response MUST be passed to that transaction. Otherwise, the response MUST be passed to the core (whether it be stateless proxy, stateful proxy, or UA) for further processing. Handling of these "stray" responses is dependent on the core (a proxy will forward them, while a UA will discard, for example).

## 18.2 Servers

### 18.2.1 Receiving Requests

A server SHOULD be prepared to receive requests on any IP address, port and transport combination that can be the result of a DNS lookup on a SIP or SIPS URI [4] that is handed out for the purposes of communicating with that server. In this context, "handing out" includes placing a URI in a Contact header field in a REGISTER request or a redirect response, or in a Record-Route header field in a request or response. A URI can also be "handed out" by placing it on a web page or business card. It is also RECOMMENDED that a server listen for requests on the default SIP ports (5060 for TCP and UDP, 5061 for TLS over TCP) on all public interfaces. The typical exception would be private networks, or when multiple server instances are running on the same host. For any port and interface that a server listens on for UDP, it MUST listen on that same port and interface for TCP. This is because a message may need to be sent using TCP, rather than UDP, if it is too large. As a result, the converse is not true. A server need not listen for UDP on a particular address and port just because it is listening on that same address and port for TCP. There may, of course, be other reasons why a server needs to listen for UDP on a particular address and port.

When the server transport receives a request over any transport, it MUST examine the value of the "sent-by" parameter in the top Via header field value. If the host portion of the "sent-by" parameter contains a domain name, or if it contains an IP address that differs from the packet source address, the server MUST add a "received" parameter to that Via header field value. This parameter MUST contain the source address from which the packet was received. This is to assist the server transport layer in sending the response, since it must be sent to the source IP address from which the request came.

Consider a request received by the server transport which looks like, in part:

```
INVITE sip:bob@Biloxi.com SIP/2.0
Via: SIP/2.0/UDP bobspc.biloxi.com:5060
```

The request is received with a source IP address of 192.0.2.4. Before passing the request up, the transport adds a "received" parameter, so that the request would look like, in part:

```
INVITE sip:bob@Biloxi.com SIP/2.0
Via: SIP/2.0/UDP bobspc.biloxi.com:5060;received=192.0.2.4
```

Next, the server transport attempts to match the request to a server transaction. It does so using the matching rules described in Section 17.2.3. If a matching server transaction is found, the request is passed to that transaction for processing. If no match is found, the request is passed to the core, which may decide to construct a new server transaction for that request. Note that when a UAS core sends a 2xx response to INVITE, the server transaction is destroyed. This means that when the ACK arrives, there will be no matching server transaction, and based on this rule, the ACK is passed to the UAS core, where it is processed.

### 18.2.2 Sending Responses

The server transport uses the value of the top Via header field in order to determine where to send a response. It MUST follow the following process:

- o If the "sent-protocol" is a reliable transport protocol such as TCP or SCTP, or TLS over those, the response MUST be sent using the existing connection to the source of the original request that created the transaction, if that connection is still open. This requires the server transport to maintain an association between server transactions and transport connections. If that connection is no longer open, the server SHOULD open a connection to the IP address in the "received" parameter, if present, using the port in the "sent-by" value, or the default port for that transport, if no port is specified. If that connection attempt fails, the server SHOULD use the procedures in [4] for servers in order to determine the IP address and port to open the connection and send the response to.

- o Otherwise, if the Via header field value contains a "maddr" parameter, the response MUST be forwarded to the address listed there, using the port indicated in "sent-by", or port 5060 if none is present. If the address is a multicast address, the response SHOULD be sent using the TTL indicated in the "ttl" parameter, or with a TTL of 1 if that parameter is not present.

- o Otherwise (for unreliable unicast transports), if the top Via has a "received" parameter, the response MUST be sent to the address in the "received" parameter, using the port indicated in the "sent-by" value, or using port 5060 if none is specified explicitly. If this fails, for example, elicits an ICMP "port unreachable" response, the procedures of Section 5 of [4] SHOULD be used to determine where to send the response.

- o Otherwise, if it is not receiver-tagged, the response MUST be sent to the address indicated by the "sent-by" value, using the procedures in Section 5 of [4].

## 18.3 Framing

In the case of message-oriented transports (such as UDP), if the message has a Content-Length header field, the message body is assumed to contain that many bytes. If there are additional bytes in the transport packet beyond the end of the body, they MUST be discarded. If the transport packet ends before the end of the message body, this is considered an error. If the message is a response, it MUST be discarded. If the message is a request, the element SHOULD generate a 400 (Bad Request) response. If the message has no Content-Length header field, the message body is assumed to end at the end of the transport packet.

In the case of stream-oriented transports such as TCP, the Content-Length header field indicates the size of the body. The Content-Length header field MUST be used with stream oriented transports.

## 18.4 Error Handling

Error handling is independent of whether the message was a request or response.

If the transport user asks for a message to be sent over an unreliable transport, and the result is an ICMP error, the behavior depends on the type of ICMP error. Host, network, port or protocol unreachable errors, or parameter problem errors SHOULD cause the transport layer to inform the transport user of a failure in sending. Source quench and TTL exceeded ICMP errors SHOULD be ignored.

If the transport user asks for a request to be sent over a reliable transport, and the result is a connection failure, the transport layer SHOULD inform the transport user of a failure in sending.

# 19 Common Message Components

There are certain components of SIP messages that appear in various places within SIP messages (and sometimes, outside of them) that merit separate discussion.

## 19.1 SIP and SIPS Uniform Resource Indicators

A SIP or SIPS URI identifies a communications resource. Like all URIs, SIP and SIPS URIs may be placed in web pages, email messages, or printed literature. They contain sufficient information to initiate and maintain a communication session with the resource.

Examples of communications resources include the following:

- o a user of an online service
- o an appearance on a multi-line phone
- o a mailbox on a messaging system
- o a PSTN number at a gateway service
- o a group (such as "sales" or "helpdesk") in an organization

A SIPS URI specifies that the resource be contacted securely. This means, in particular, that TLS is to be used between the UAC and the domain that owns the URI. From there, secure communications are used to reach the user, where the specific security mechanism depends on the policy of the domain. Any resource described by a SIP URI can be "upgraded" to a SIPS URI by just changing the scheme, if it is desired to communicate with that resource securely.

### 19.1.1 SIP and SIPS URI Components

The "sip:" and "sips:" schemes follow the guidelines in RFC 2396 [5]. They use a form similar to the mailto URL, allowing the specification of SIP request-header fields and the SIP message-body. This makes it possible to specify the subject, media type, or urgency of sessions initiated by using a URI on a web page or in an email message. The formal syntax for a SIP or SIPS URI is presented in Section 25. Its general form, in the case of a SIP URI, is:

```
sip:user:password@host:port;uri-parameters?headers
```

The format for a SIPS URI is the same, except that the scheme is "sips" instead of sip. These tokens, and some of the tokens in their expansions, have the following meanings:

user: The identifier of a particular resource at the host being addressed. The term "host" in this context frequently refers to a domain. The "userinfo" of a URI consists of this user field, the password field, and the @ sign following them. The userinfo part of a URI is optional and MAY be absent when the destination host does not have a notion of users or when the host itself is the resource being identified. If the @ sign is present in a SIP or SIPS URI, the user field MUST NOT be empty.

If the host being addressed can process telephone numbers, for instance, an Internet telephony gateway, a telephone-subscriber field defined in RFC 2806 [9] MAY be used to populate the user field. There are special escaping rules for encoding telephone-subscriber fields in SIP and SIPS URIs described in Section 19.1.2.

password: A password associated with the user. While the SIP and SIPS URI syntax allows this field to be present, its use is NOT RECOMMENDED, because the passing of authentication information in clear text (such as URIs) has proven to be a security risk in almost every case where it has been used. For instance, transporting a PIN number in this field exposes the PIN.

Note that the password field is just an extension of the user portion. Implementations not wishing to give special significance to the password portion of the field MAY simply treat "user:password" as a single string.

host: The host providing the SIP resource. The host part contains either a fully-qualified domain name or numeric IPv4 or IPv6 address. Using the fully-qualified domain name form is RECOMMENDED whenever possible.

port: The port number where the request is to be sent.

URI parameters: Parameters affecting a request constructed from the URI.

URI parameters are added after the hostport component and are separated by semi-colons.

URI parameters take the form:

```
parameter-name "=" parameter-value
```

Even though an arbitrary number of URI parameters may be included in a URI, any given parameter-name MUST NOT appear more than once.

This extensible mechanism includes the transport, maddr, ttl, user, method and lr parameters.

The transport parameter determines the transport mechanism to be used for sending SIP messages, as specified in [4]. SIP can use any network transport protocol. Parameter names are defined for UDP (RFC 768 [14]), TCP (RFC 761 [15]), and SCTP (RFC 2960 [16]). For a SIPS URI, the transport parameter MUST indicate a reliable transport.

The maddr parameter indicates the server address to be contacted for this user, overriding any address derived from the host field. When an maddr parameter is present, the port and transport components of the URI apply to the address indicated in the maddr parameter value. [4] describes the proper interpretation of the transport, maddr, and hostport in order to obtain the destination address, port, and transport for sending a request.

The maddr field has been used as a simple form of loose source routing. It allows a URI to specify a proxy that must be traversed en-route to the destination. Continuing to use the maddr parameter this way is strongly discouraged (the mechanisms that enable it are deprecated). Implementations should instead use the Route mechanism described in this document, establishing a pre-existing route set if necessary (see Section 8.1.1.1). This provides a full URI to describe the node to be traversed.

The ttl parameter determines the time-to-live value of the UDP multicast packet and MUST only be used if maddr is a multicast address and the transport protocol is UDP. For example, to specify a call to alice@atlanta.com using multicast to 239.255.255.1 with a ttl of 15, the following URI would be used:

```
sip:alice@atlanta.com;maddr=239.255.255.1;ttl=15
```

The set of valid telephone-subscriber strings is a subset of valid user strings. The user URI parameter exists to distinguish telephone numbers from user names that happen to look like telephone numbers. If the user string contains a telephone number formatted as a telephone-subscriber, the user parameter value "phone" SHOULD be present. Even without this parameter, recipients of SIP and SIPS URIs MAY interpret the pre-@ part as a telephone number if local restrictions on the name space for user name allow it.

The method of the SIP request constructed from the URI can be specified with the method parameter.

The lr parameter, when present, indicates that the element responsible for this resource implements the routing mechanisms specified in this document. This parameter will be used in the URIs proxies place into Record-Route header field values, and may appear in the URIs in a pre-existing route set.

This parameter is used to achieve backwards compatibility with systems implementing the strict-routing mechanisms of RFC 2543 and the rfc2543bis drafts up to bis-05. An element preparing to send a request based on a URI not containing this parameter can assume the receiving element implements strict-routing and reformat the message to preserve the information in the Request-URI.

Since the uri-parameter mechanism is extensible, SIP elements MUST silently ignore any uri-parameters that they do not understand.

Headers: Header fields to be included in a request constructed from the URI.

Headers fields in the SIP request can be specified with the "?" mechanism within a URI. The header names and values are encoded in ampersand separated hname = hvalue pairs. The special hname "body" indicates that the associated hvalue is the message-body of the SIP request.

Table 1 summarizes the use of SIP and SIPS URI components based on the context in which the URI appears. The external column describes URIs appearing anywhere outside of a SIP message, for instance on a web page or business card. Entries marked "m" are mandatory, those marked "o" are optional, and those marked "-" are not allowed. Elements processing URIs SHOULD ignore any disallowed components if they are present. The second column indicates the default value of an optional element if it is not present. "--" indicates that the element is either not optional, or has no default value.

URIs in Contact header fields have different restrictions depending on the context in which the header field appears. One set applies to messages that establish and maintain dialogs (INVITE and its 200 (OK) response). The other applies to registration and redirection messages (REGISTER, its 200 (OK) response, and 3xx class responses to any method).

### 19.1.2 Character Escaping Requirements

| | default | Req.-URI | To | From | reg./redir. Contact | dialog Contact/ R-R/Route | external |
|---|---|---|---|---|---|---|---|
| user | -- | o | o | o | o | o | o |
| password | -- | o | o | o | o | o | o |
| host | -- | m | m | m | m | m | m |
| port | (1) | o | - | - | o | o | o |
| user-param | ip | o | o | o | o | o | o |
| method | INVITE | - | - | - | - | - | o |
| maddr-param | -- | o | - | - | o | o | o |
| ttl-param | 1 | o | - | - | o | - | o |
| transp.-param | (2) | o | - | - | o | o | o |
| lr-param | -- | o | - | - | - | o | o |
| other-param | -- | o | o | o | o | o | o |
| headers | -- | - | - | - | o | - | o |

(1): The default port value is transport and scheme dependent. The default is 5060 for sip: using UDP, TCP, or SCTP. The default is 5061 for sip: using TLS over TCP and sips: over TCP.

(2): The default transport is scheme dependent. For sip:, it is UDP. For sips:, it is TCP.

Table 1: Use and default values of URI components for SIP header field values, Request-URI and references

SIP follows the requirements and guidelines of RFC 2396 [5] when defining the set of characters that must be escaped in a SIP URI, and uses its ""%" HEX HEX" mechanism for escaping. From RFC 2396 [5]:

> The set of characters actually reserved within any given URI component is defined by that component. In general, a character is reserved if the semantics of the URI changes if the character is replaced with its escaped US-ASCII encoding [5]. Excluded US-ASCII characters (RFC 2396 [5]), such as space and control characters and characters used as URI delimiters, also MUST be escaped. URIs MUST NOT contain unescaped space and control characters.

For each component, the set of valid BNF expansions defines exactly which characters may appear unescaped. All other characters MUST be escaped.

For example, "@" is not in the set of characters in the user component, so the user "j@s0n" must have at least the @ sign encoded, as in "j%40s0n".

Expanding the hname and hvalue tokens in Section 25 show that all URI reserved characters in header field names and values MUST be escaped.

The telephone-subscriber subset of the user component has special escaping considerations. The set of characters not reserved in the RFC 2806 [9] description of telephone-subscriber contains a number of characters in various syntax elements that need to be escaped when used in SIP URIs. Any characters occurring in a telephone-subscriber that do not appear in an expansion of the BNF for the user rule MUST be escaped.

Note that character escaping is not allowed in the host component of a SIP or SIPS URI (the % character is not valid in its expansion). This is likely to change in the future as requirements for Internationalized Domain Names are finalized. Current implementations MUST NOT attempt to improve robustness by treating received escaped characters in the host component as literally equivalent to their unescaped counterpart. The behavior required to meet the requirements of IDN may be significantly different.

### 19.1.3 Example SIP and SIPS URIs

```
sip:alice@atlanta.com
sip:alice:secretword@atlanta.com;transport=tcp
sips:alice@atlanta.com?subject=project%20x&priority=urgent
sip:+1-212-555-1212:1234@gateway.com;user=phone
sips:1212@gateway.com
sip:alice@192.0.2.4
sip:atlanta.com;method=REGISTER?to=alice%40atlanta.com
sip:alice;day=tuesday@atlanta.com
```

The last sample URI above has a user field value of "alice;day=tuesday". The escaping rules defined above allow a semicolon to appear unescaped in this field. For the purposes of this protocol, the field is opaque. The structure of that value is only useful to the SIP element responsible for the resource.

### 19.1.4 URI Comparison

Some operations in this specification require determining whether two SIP or SIPS URIs are equivalent. In this specification, registrars need to compare bindings in Contact URIs in REGISTER requests (see Section 10.3.). SIP and SIPS URIs are compared for equality according to the following rules:

- o A SIP and SIPS URI are never equivalent.

- o Comparison of the userinfo of SIP and SIPS URIs is case-sensitive. This includes userinfo containing passwords or formatted as telephone-subscribers. Comparison of all other components of the URI is case-insensitive unless explicitly defined otherwise.

- o The ordering of parameters and header fields is not significant in comparing SIP and SIPS URIs.

- o Characters other than those in the "reserved" set (see RFC 2396 [5]) are equivalent to their ""%" HEX HEX" encoding.

- o An IP address that is the result of a DNS lookup of a host name does not match that host name.

- o For two URIs to be equal, the user, password, host, and port components must match.

  A URI omitting the user component will not match a URI that includes one. A URI omitting the password component will not match a URI that includes one.

  A URI omitting any component with a default value will not match a URI explicitly containing that component with its default value. For instance, a URI omitting the optional port component will not match a URI explicitly declaring port 5060. The same is true for the transport-parameter, ttl-parameter, user-parameter, and method components.

  > Defining sip:user@host to not be equivalent to sip:user@host:5060 is a change from RFC 2543. When deriving addresses from URIs, equivalent addresses are expected from equivalent URIs. The URI sip:user@host:5060 will always resolve to port 5060. The URI sip:user@host may resolve to other ports through the DNS SRV mechanisms detailed in [4].

- o URI uri-parameter components are compared as follows:

  - Any uri-parameter appearing in both URIs must match.

  - A user, ttl, or method uri-parameter appearing in only one URI never matches, even if it contains the default value.

  - A URI that includes an maddr parameter will not match a URI that contains no maddr parameter.

  - All other uri-parameters appearing in only one URI are ignored when comparing the URIs.

o URI header components are never ignored. Any present header component MUST be present in both URIs and match for the URIs to match. The matching rules are defined for each header field in Section 20.

The URIs within each of the following sets are equivalent:

```
sip:%61lice@atlanta.com;transport=TCP
sip:alice@AtLanTa.CoM;Transport=tcp

sip:carol@chicago.com
sip:carol@chicago.com;newparam=5
sip:carol@chicago.com;security=on

sip:biloxi.com;transport=tcp;method=REGISTER?to=sip:bob%40biloxi.com
sip:biloxi.com;method=REGISTER;transport=tcp?to=sip:bob%40biloxi.com

sip:alice@atlanta.com?subject=project%20x&priority=urgent
sip:alice@atlanta.com?priority=urgent&subject=project%20x
```

The URIs within each of the following sets are not equivalent:

```
SIP:ALICE@AtLanTa.CoM;Transport=udp             (different usernames)
sip:alice@AtLanTa.CoM;Transport=UDP

sip:bob@biloxi.com                   (can resolve to different ports)
sip:bob@biloxi.com:5060

sip:bob@biloxi.com              (can resolve to different transports)
sip:bob@biloxi.com;transport=udp

sip:bob@biloxi.com     (can resolve to different port and transports)
sip:bob@biloxi.com:6000;transport=tcp

sip:carol@chicago.com                    (different header component)
sip:carol@chicago.com?Subject=next%20meeting

sip:bob@phone21.boxesbybob.com   (even though that's what
sip:bob@192.0.2.4                 phone21.boxesbybob.com resolves to)
```

Note that equality is not transitive:

o sip:carol@chicago.com and sip:carol@chicago.com;security=on are equivalent

o sip:carol@chicago.com and sip:carol@chicago.com;security=off are equivalent

- o sip:carol@chicago.com;security=on and sip:carol@chicago.com;security=off are not equivalent

### 19.1.5 Forming Requests from a URI

An implementation needs to take care when forming requests directly from a URI. URIs from business cards, web pages, and even from sources inside the protocol such as registered contacts may contain inappropriate header fields or body parts.

An implementation MUST include any provided transport, maddr, ttl, or user parameter in the Request-URI of the formed request. If the URI contains a method parameter, its value MUST be used as the method of the request. The method parameter MUST NOT be placed in the Request-URI. Unknown URI parameters MUST be placed in the message's Request-URI.

An implementation SHOULD treat the presence of any headers or body parts in the URI as a desire to include them in the message, and choose to honor the request on a per-component basis.

An implementation SHOULD NOT honor these obviously dangerous header fields: From, Call-ID, CSeq, Via, and Record-Route.

An implementation SHOULD NOT honor any requested Route header field values in order to not be used as an unwitting agent in malicious attacks.

An implementation SHOULD NOT honor requests to include header fields that may cause it to falsely advertise its location or capabilities. These include: Accept, Accept-Encoding, Accept-Language, Allow, Contact (in its dialog usage), Organization, Supported, and User-Agent.

An implementation SHOULD verify the accuracy of any requested descriptive header fields, including: Content-Disposition, Content-Encoding, Content-Language, Content-Length, Content-Type, Date, Mime-Version, and Timestamp.

If the request formed from constructing a message from a given URI is not a valid SIP request, the URI is invalid. An implementation MUST NOT proceed with transmitting the request. It should instead pursue the course of action due an invalid URI in the context it occurs.

> The constructed request can be invalid in many ways. These include, but are not limited to, syntax error in header fields, invalid combinations of URI parameters, or an incorrect description of the message body.

Sending a request formed from a given URI may require capabilities unavailable to the implementation. The URI might indicate use of an unimplemented transport or extension, for example. An implementation SHOULD refuse to send these requests rather than modifying them to match their capabilities. An implementation MUST NOT send a request requiring an extension that it does not support.

> For example, such a request can be formed through the presence of a Require header parameter or a method URI parameter with an unknown or explicitly unsupported value.

### 19.1.6 Relating SIP URIs and tel URLs

When a tel URL (RFC 2806 [9]) is converted to a SIP or SIPS URI, the entire telephone-subscriber portion of the tel URL, including any parameters, is placed into the userinfo part of the SIP or SIPS URI.

Thus, tel:+358-555-1234567;postd=pp22 becomes

```
sip:+358-555-1234567;postd=pp22@foo.com;user=phone
```

or

```
sips:+358-555-1234567;postd=pp22@foo.com;user=phone
```

not

```
sip:+358-555-1234567@foo.com;postd=pp22;user=phone
```

or

```
sips:+358-555-1234567@foo.com;postd=pp22;user=phone
```

In general, equivalent "tel" URLs converted to SIP or SIPS URIs in this fashion may not produce equivalent SIP or SIPS URIs. The userinfo of SIP and SIPS URIs are compared as a case-sensitive string. Variance in case-insensitive portions of tel URLs and reordering of tel URL parameters does not affect tel URL equivalence, but does affect the equivalence of SIP URIs formed from them.

For example,

```
tel:+358-555-1234567;postd=pp22
tel:+358-555-1234567;POSTD=PP22
```

are equivalent, while

```
sip:+358-555-1234567;postd=pp22@foo.com;user=phone
sip:+358-555-1234567;POSTD=PP22@foo.com;user=phone
```

are not.

Likewise,

```
tel:+358-555-1234567;postd=pp22;isub=1411
tel:+358-555-1234567;isub=1411;postd=pp22
```

are equivalent, while

```
sip:+358-555-1234567;postd=pp22;isub=1411@foo.com;user=phone
sip:+358-555-1234567;isub=1411;postd=pp22@foo.com;user=phone
```

are not.

To mitigate this problem, elements constructing telephone-subscriber fields to place in the userinfo part of a SIP or SIPS URI SHOULD fold any case-insensitive portion of telephone-subscriber to lower case, and order the telephone-subscriber parameters lexically by parameter name, excepting isdn-subaddress and post-dial, which occur first and in that order. (All components of a tel URL except for future-extension parameters are defined to be compared case-insensitive.)

Following this suggestion, both

```
tel:+358-555-1234567;postd=pp22
tel:+358-555-1234567;POSTD=PP22
```

become

```
sip:+358-555-1234567;postd=pp22@foo.com;user=phone
```

and both

```
tel:+358-555-1234567;tsp=a.b;phone-context=5
tel:+358-555-1234567;phone-context=5;tsp=a.b
```

become

```
sip:+358-555-1234567;phone-context=5;tsp=a.b@foo.com;user=phone
```

## 19.2 Option Tags

Option tags are unique identifiers used to designate new options (extensions) in SIP. These tags are used in Require (Section 20.32), Proxy-Require (Section 20.29), Supported (Section 20.37) and Unsupported (Section 20.40) header fields. Note that these options appear as parameters in those header fields in an option-tag = token form (see Section 25 for the definition of token).

Option tags are defined in standards track RFCs. This is a change from past practice, and is instituted to ensure continuing multi-vendor interoperability (see discussion in Section 20.32 and Section 20.37). An IANA registry of option tags is used to ensure easy reference.

## 19.3 Tags

The "tag" parameter is used in the To and From header fields of SIP messages. It serves as a general mechanism to identify a dialog, which is the combination of the Call-ID along with two tags, one from each participant in the dialog. When a UA sends a request outside of a dialog, it contains a From tag only, providing "half" of the dialog ID. The dialog is completed from the response(s), each of which contributes the second half in the To header field. The forking of SIP requests means that multiple dialogs can be established from a single request. This also explains the need for the two-sided dialog identifier; without a contribution from the recipients, the originator could not disambiguate the multiple dialogs established from a single request.

When a tag is generated by a UA for insertion into a request or response, it MUST be globally unique and cryptographically random with at least 32 bits of randomness. A property of this selection requirement is that a UA will place a different tag into the From header of an INVITE than it would place into the To header of the response to the same INVITE. This is needed in order for a UA to invite itself to a session, a common case for "hairpinning" of calls in PSTN gateways. Similarly, two INVITEs for different calls will have different From tags, and two responses for different calls will have different To tags.

Besides the requirement for global uniqueness, the algorithm for generating a tag is implementation-specific. Tags are helpful in fault tolerant systems, where a dialog is to be recovered on an alternate server after a failure. A UAS can select the tag in such a way that a backup can recognize a request as part of a dialog on the failed server, and therefore determine that it should attempt to recover the dialog and any other state associated with it.

# 20 Header Fields

The general syntax for header fields is covered in Section 7.3. This section lists the full set of header fields along with notes on syntax, meaning, and usage. Throughout this section, we use [HX.Y] to refer to Section X.Y of the current HTTP/1.1 specification RFC 2616 [8]. Examples of each header field are given.

Information about header fields in relation to methods and proxy processing is summarized in Tables 2 and 3.

The "where" column describes the request and response types in which the header field can be used. Values in this column are:

R: header field may only appear in requests;

r: header field may only appear in responses;

2xx, 4xx, etc.: A numerical value or range indicates response codes with which the header field can be used;

c: header field is copied from the request to the response.

An empty entry in the "where" column indicates that the header field may be present in all requests and responses.

The "proxy" column describes the operations a proxy may perform on a header field:

a: A proxy can add or concatenate the header field if not present.

m: A proxy can modify an existing header field value.

d: A proxy can delete a header field value.

r: A proxy must be able to read the header field, and thus this header field cannot be encrypted.

The next six columns relate to the presence of a header field in a method:

c: Conditional; requirements on the header field depend on the context of the message.

m: The header field is mandatory.

m*: The header field SHOULD be sent, but clients/servers need to be prepared to receive messages without that header field.

o: The header field is optional.

t: The header field SHOULD be sent, but clients/servers need to be prepared to receive messages without that header field.

If a stream-based protocol (such as TCP) is used as a transport, then the header field MUST be sent.

*: The header field is required if the message body is not empty. See Sections 20.14, 20.15 and 7.4 for details.

-: The header field is not applicable.

"Optional" means that an element MAY include the header field in a request or response, and a UA MAY ignore the header field if present in the request or response (The exception to this rule is the Require header field discussed in 20.32). A "mandatory" header field MUST be present in a request, and MUST be understood by the UAS receiving the request. A mandatory response header field MUST be present in the response, and the header field MUST be understood by the UAC processing the response. "Not applicable" means that the header field MUST NOT be present in a request. If one is placed in a request by mistake, it MUST be ignored by the UAS receiving the request. Similarly, a header field labeled "not applicable" for a response means that the UAS MUST NOT place the header field in the response, and the UAC MUST ignore the header field in the response.

A UA SHOULD ignore extension header parameters that are not understood.

A compact form of some common header field names is also defined for use when overall message size is an issue.

The Contact, From, and To header fields contain a URI. If the URI contains a comma, question mark or semicolon, the URI MUST be enclosed in angle brackets (< and >). Any URI parameters are contained within these brackets. If the URI is not enclosed in angle brackets, any semicolon-delimited parameters are header-parameters, not URI parameters.

## 20.1 Accept

The Accept header field follows the syntax defined in [H14.1]. The semantics are also identical, with the exception that if no Accept header field is present, the server SHOULD assume a default value of application/sdp.

An empty Accept header field means that no formats are acceptable.

Example:

| Header field | where | proxy | ACK | BYE | CAN | INV | OPT | REG |
|---|---|---|---|---|---|---|---|---|
| Accept | R | | - | o | - | o | m* | o |
| Accept | 2xx | | - | - | - | o | m* | o |
| Accept | 415 | | - | c | - | c | c | c |
| Accept-Encoding | R | | - | o | - | o | o | o |
| Accept-Encoding | 2xx | | - | - | - | o | m* | o |
| Accept-Encoding | 415 | | - | c | - | c | c | c |
| Accept-Language | R | | - | o | - | o | o | o |
| Accept-Language | 2xx | | - | - | - | o | m* | o |
| Accept-Language | 415 | | - | c | - | c | c | c |
| Alert-Info | R | ar | - | - | - | o | - | - |
| Alert-Info | 180 | ar | - | - | - | o | - | - |
| Allow | R | | - | o | - | o | o | o |
| Allow | 2xx | | - | o | - | m* | m* | o |
| Allow | r | | - | o | - | o | o | o |
| Allow | 405 | | - | m | - | m | m | m |
| Authentication-Info | 2xx | | - | o | - | o | o | o |
| Authorization | R | | o | o | o | o | o | o |
| Call-ID | c | r | m | m | m | m | m | m |
| Call-Info | | ar | - | - | - | o | o | o |
| Contact | R | | o | - | - | m | o | o |
| Contact | 1xx | | - | - | - | o | - | - |
| Contact | 2xx | | - | - | - | m | o | o |
| Contact | 3xx | d | - | o | - | o | o | o |
| Contact | 485 | | - | o | - | o | o | o |
| Content-Disposition | | | o | o | - | o | o | o |
| Content-Encoding | | | o | o | - | o | o | o |
| Content-Language | | | o | o | - | o | o | o |
| Content-Length | | ar | t | t | t | t | t | t |
| Content-Type | | | * | * | - | * | * | * |
| CSeq | c | r | m | m | m | m | m | m |
| Date | | a | o | o | o | o | o | o |
| Error-Info | 300-699 | a | - | o | o | o | o | o |
| Expires | | | - | - | - | o | - | o |
| From | c | r | m | m | m | m | m | m |
| In-Reply-To | R | | - | - | - | o | - | - |
| Max-Forwards | R | amr | m | m | m | m | m | m |
| Min-Expires | 423 | | - | - | - | - | - | m |
| MIME-Version | | | o | o | - | o | o | o |
| Organization | | ar | - | - | - | o | o | o |

Table 2: Summary of header fields, A--O

| Header field | where | proxy | ACK | BYE | CAN | INV | OPT | REG |
|---|---|---|---|---|---|---|---|---|
| Priority | R | ar | - | - | - | o | - | - |
| Proxy-Authenticate | 407 | ar | - | m | - | m | m | m |
| Proxy-Authenticate | 401 | ar | - | o | o | o | o | o |
| Proxy-Authorization | R | dr | o | o | - | o | o | o |
| Proxy-Require | R | ar | - | o | - | o | o | o |
| Record-Route | R | ar | o | o | o | o | o | - |
| Record-Route | 2xx,18x | mr | - | o | o | o | o | - |
| Reply-To | | | - | - | - | o | - | - |
| Require | | ar | - | c | - | c | c | c |
| Retry-After | 404,413,480,486 | | - | o | o | o | o | o |
| | 500,503 | | - | o | o | o | o | o |
| | 600,603 | | - | o | o | o | o | o |
| Route | R | adr | c | c | c | c | c | c |
| Server | r | | - | o | o | o | o | o |
| Subject | R | | - | - | - | o | - | - |
| Supported | R | | - | o | o | m* | o | o |
| Supported | 2xx | | - | o | o | m* | m* | o |
| Timestamp | | | o | o | o | o | o | o |
| To | c(1) | r | m | m | m | m | m | m |
| Unsupported | 420 | | - | m | - | m | m | m |
| User-Agent | | | o | o | o | o | o | o |
| Via | R | amr | m | m | m | m | m | m |
| Via | rc | dr | m | m | m | m | m | m |
| Warning | r | | - | o | o | o | o | o |
| WWW-Authenticate | 401 | ar | - | m | - | m | m | m |
| WWW-Authenticate | 407 | ar | - | o | - | o | o | o |

Table 3: Summary of header fields, P--Z; (1): copied with possible addition of tag

```
Accept: application/sdp;level=1, application/x-private, text/html
```

## 20.2 Accept-Encoding

The Accept-Encoding header field is similar to Accept, but restricts the content-codings [H3.5] that are acceptable in the response. See [H14.3]. The semantics in SIP are identical to those defined in [H14.3].

An empty Accept-Encoding header field is permissible. It is equivalent to Accept-Encoding: identity, that is, only the identity encoding, meaning no encoding, is permissible.

If no Accept-Encoding header field is present, the server SHOULD assume a default value of identity.

This differs slightly from the HTTP definition, which indicates that when not present, any encoding can be used, but the identity encoding is preferred.

Example:

```
Accept-Encoding: gzip
```

## 20.3 Accept-Language

The Accept-Language header field is used in requests to indicate the preferred languages for reason phrases, session descriptions, or status responses carried as message bodies in the response. If no Accept-Language header field is present, the server SHOULD assume all languages are acceptable to the client.

The Accept-Language header field follows the syntax defined in [H14.4]. The rules for ordering the languages based on the "q" parameter apply to SIP as well.

Example:

```
Accept-Language: da, en-gb;q=0.8, en;q=0.7
```

## 20.4 Alert-Info

When present in an INVITE request, the Alert-Info header field specifies an alternative ring tone to the UAS. When present in a 180 (Ringing) response, the Alert-Info header field specifies an alternative ringback tone to the UAC. A typical usage is for a proxy to insert this header field to provide a distinctive ring feature.

The Alert-Info header field can introduce security risks. These risks and the ways to handle them are discussed in Section 20.9, which discusses the Call-Info header field since the risks are identical.

In addition, a user SHOULD be able to disable this feature selectively.

This helps prevent disruptions that could result from the use of this header field by untrusted elements.

Example:

```
Alert-Info: <http://www.example.com/sounds/moo.wav>
```

## 20.5 Allow

The Allow header field lists the set of methods supported by the UA generating the message.

All methods, including ACK and CANCEL, understood by the UA MUST be included in the list of methods in the Allow header field, when present. The absence of an Allow header field MUST NOT be interpreted to mean that the UA sending the message supports no methods. Rather, it implies that the UA is not providing any information on what methods it supports.

Supplying an Allow header field in responses to methods other than OPTIONS reduces the number of messages needed.

Example:

```
Allow: INVITE, ACK, OPTIONS, CANCEL, BYE
```

## 20.6 Authentication-Info

The Authentication-Info header field provides for mutual authentication with HTTP Digest. A UAS MAY include this header field in a 2xx response to a request that was successfully authenticated using digest based on the Authorization header field.

Syntax and semantics follow those specified in RFC 2617 [17].

Example:

```
Authentication-Info: nextnonce="47364c23432d2e131a5fb210812c"
```

## 20.7 Authorization

The Authorization header field contains authentication credentials of a UA. Section 22.2 overviews the use of the Authorization header field, and Section 22.4 describes the syntax and semantics when used with HTTP authentication.

This header field, along with Proxy-Authorization, breaks the general rules about multiple header field values. Although not a comma-separated list, this header field name may be present multiple times, and MUST NOT be combined into a single header line using the usual rules described in Section 7.3.

In the example below, there are no quotes around the Digest parameter:

```
Authorization: Digest username="Alice", realm="atlanta.com",
 nonce="84a4cc6f3082121f32b42a2187831a9e",
 response="7587245234b3434cc3412213e5f113a5432"
```

## 20.8 Call-ID

The Call-ID header field uniquely identifies a particular invitation or all registrations of a particular client. A single multimedia conference can give rise to several calls with different Call-IDs, for example, if a user invites a single individual several times to the same (long-running) conference. Call-IDs are case-sensitive and are simply compared byte-by-byte.

The compact form of the Call-ID header field is i.

Examples:

```
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@biloxi.com
i:f81d4fae-7dec-11d0-a765-00a0c91e6bf6@192.0.2.4
```

## 20.9 Call-Info

The Call-Info header field provides additional information about the caller or callee, depending on whether it is found in a request or response. The purpose of the URI is described by the "purpose" parameter. The "icon" parameter designates an image suitable as an iconic representation of the caller or callee. The "info" parameter describes the caller or callee in general, for example, through a web page. The "card" parameter provides a business card, for example, in vCard [36] or LDIF [37] formats. Additional tokens can be registered using IANA and the procedures in Section 27.

Use of the Call-Info header field can pose a security risk. If a callee fetches the URIs provided by a malicious caller, the callee may be at risk for displaying inappropriate or offensive content, dangerous or illegal content, and so on. Therefore, it is RECOMMENDED that a UA only render the information in the Call-Info header field if it can verify the authenticity of the element that originated the header field and trusts that element. This need not be the peer UA; a proxy can insert this header field into requests.

Example:

```
Call-Info: <http://wwww.example.com/alice/photo.jpg> ;purpose=icon,
  <http://www.example.com/alice/> ;purpose=info
```

## 20.10 Contact

A Contact header field value provides a URI whose meaning depends on the type of request or response it is in.

A Contact header field value can contain a display name, a URI with URI parameters, and header parameters.

This document defines the Contact parameters "q" and "expires". These parameters are only used when the Contact is present in a REGISTER request or response, or in a 3xx response. Additional parameters may be defined in other specifications.

When the header field value contains a display name, the URI including all URI parameters is enclosed in "<" and ">". If no "<" and ">" are present, all parameters after the URI are header parameters, not URI parameters. The display name can be tokens, or a quoted string, if a larger character set is desired.

Even if the "display-name" is empty, the "name-addr" form MUST be used if the "addr-spec" contains a comma, semicolon, or question mark. There may or may not be LWS between the display-name and the "<".

These rules for parsing a display name, URI and URI parameters, and header parameters also apply for the header fields To and From.

> The Contact header field has a role similar to the Location header field in HTTP. However, the HTTP header field only allows one address, unquoted. Since URIs can contain commas and semicolons as reserved characters, they can be mistaken for header or parameter delimiters, respectively.

The compact form of the Contact header field is m (for "moved").

Examples:

```
Contact: "Mr. Watson" <sip:watson@worcester.bell-telephone.com>
   ;q=0.7; expires=3600,
   "Mr. Watson" <mailto:watson@bell-telephone.com> ;q=0.1
m: <sips:bob@192.0.2.4>;expires=60
```

## 20.11 Content-Disposition

The Content-Disposition header field describes how the message body or, for multipart messages, a message body part is to be interpreted by the UAC or UAS. This SIP header field extends the MIME Content-Type (RFC 2183 [18]).

Several new "disposition-types" of the Content-Disposition header are defined by SIP. The value "session" indicates that the body part describes a session, for either calls or early (pre-call) media. The value "render" indicates that the body part should be displayed or otherwise rendered to the user. Note that the value "render" is used rather than "inline" to avoid the connotation that the MIME body is displayed as a part of the rendering of the entire message (since the MIME bodies of SIP messages oftentimes are not displayed to users). For backward-compatibility, if the Content-Disposition header field is missing, the server SHOULD assume bodies of Content-Type application/sdp are the disposition "session", while other content types are "render".

The disposition type "icon" indicates that the body part contains an image suitable as an iconic representation of the caller or callee that could be rendered informationally by a user agent when a message has been received, or persistently while a dialog takes place. The value "alert" indicates that the body part contains information, such as an audio clip, that should be rendered by the user agent in an attempt to alert the user to the receipt of a request, generally a request that initiates a dialog; this alerting body could for example be rendered as a ring tone for a phone call after a 180 Ringing provisional response has been sent.

Any MIME body with a "disposition-type" that renders content to the user should only be processed when a message has been properly authenticated.

The handling parameter, handling-param, describes how the UAS should react if it receives a message body whose content type or disposition type it does not understand. The parameter has defined values of "optional" and "required". If the handling parameter is missing, the value "required" SHOULD be assumed. The handling parameter is described in RFC 3204 [19].

If this header field is missing, the MIME type determines the default content disposition. If there is none, "render" is assumed.

Example:

```
Content-Disposition: session
```

## 20.12 Content-Encoding

The Content-Encoding header field is used as a modifier to the "media-type". When present, its value indicates what additional content codings have been applied to the entity-body, and thus what decoding mechanisms MUST be applied in order to obtain the media-type referenced by the Content-Type header field. Content-Encoding is primarily used to allow a body to be compressed without losing the identity of its underlying media type.

If multiple encodings have been applied to an entity-body, the content codings MUST be listed in the order in which they were applied.

All content-coding values are case-insensitive. IANA acts as a registry for content-coding value tokens. See [H3.5] for a definition of the syntax for content-coding.

Clients MAY apply content encodings to the body in requests. A server MAY apply content encodings to the bodies in responses. The server MUST only use encodings listed in the Accept-Encoding header field in the request.

The compact form of the Content-Encoding header field is e. Examples:

```
Content-Encoding: gzip
e: tar
```

## 20.13 Content-Language

See [H14.12]. Example:

```
Content-Language: fr
```

## 20.14 Content-Length

The Content-Length header field indicates the size of the message-body, in decimal number of octets, sent to the recipient. Applications SHOULD use this field to indicate the size of the message-body to be transferred, regardless of the media type of the entity. If a stream-based protocol (such as TCP) is used as transport, the header field MUST be used.

The size of the message-body does not include the CRLF separating header fields and body. Any Content-Length greater than or equal to zero is a valid value. If no body is present in a message, then the Content-Length header field value MUST be set to zero.

The ability to omit Content-Length simplifies the creation of cgi-like scripts that dynamically generate responses.

The compact form of the header field is l.

Examples:

```
Content-Length: 349
l: 173
```

## 20.15 Content-Type

The Content-Type header field indicates the media type of the message-body sent to the recipient. The "media-type" element is defined in [H3.7]. The Content-Type header field MUST be present if the body is not empty. If the body is empty, and a Content-Type header field is present, it indicates that the body of the specific type has zero length (for example, an empty audio file).

The compact form of the header field is c.

Examples:

```
Content-Type: application/sdp
c: text/html; charset=ISO-8859-4
```

## 20.16 CSeq

A CSeq header field in a request contains a single decimal sequence number and the request method. The sequence number MUST be expressible as a 32-bit unsigned integer. The method part of CSeq is case-sensitive. The CSeq header field serves to order transactions within a dialog, to provide a means to uniquely identify transactions, and to differentiate between new requests and request retransmissions. Two CSeq header fields are considered equal if the sequence number and the request method are identical. Example:

```
CSeq: 4711 INVITE
```

## 20.17 Date

The Date header field contains the date and time. Unlike HTTP/1.1, SIP only supports the most recent RFC 1123 [20] format for dates. As in [H3.3], SIP restricts the time zone in SIP-date to "GMT", while RFC 1123 allows any time zone. An RFC 1123 date is case-sensitive.

The Date header field reflects the time when the request or response is first sent.

The Date header field can be used by simple end systems without a battery-backed clock to acquire a notion of current time. However, in its GMT form, it requires clients to know their offset from GMT.

Example:

```
Date: Sat, 13 Nov 2010 23:29:00 GMT
```

## 20.18 Error-Info

The Error-Info header field provides a pointer to additional information about the error status response.

SIP UACs have user interface capabilities ranging from pop-up windows and audio on PC softclients to audio-only on "black" phones or endpoints connected via gateways. Rather than forcing a server generating an error to choose between sending an error status code with a detailed reason phrase and playing an audio recording, the Error-Info header field allows both to be sent. The UAC then has the choice of which error indicator to render to the caller.

A UAC MAY treat a SIP or SIPS URI in an Error-Info header field as if it were a Contact in a redirect and generate a new INVITE, resulting in a recorded announcement session being established. A non-SIP URI MAY be rendered to the user.

Examples:

```
SIP/2.0 404 The number you have dialed is not in service
Error-Info: <sip:not-in-service-recording@atlanta.com>
```

## 20.19 Expires

The Expires header field gives the relative time after which the message (or content) expires.

The precise meaning of this is method dependent.

The expiration time in an INVITE does not affect the duration of the actual session that may result from the invitation. Session description protocols may offer the ability to express time limits on the session duration, however.

The value of this field is an integral number of seconds (in decimal) between 0 and (2**32)-1, measured from the receipt of the request.

Example:

```
Expires: 5
```

## 20.20 From

The From header field indicates the initiator of the request. This may be different from the initiator of the dialog. Requests sent by the callee to the caller use the callee's address in the From header field.

The optional "display-name" is meant to be rendered by a human user interface. A system SHOULD use the display name "Anonymous" if the identity of the client is to remain hidden. Even if the "display-name" is empty, the "name-addr" form MUST be used if the "addr-spec" contains a comma, question mark, or semicolon. Syntax issues are discussed in Section 7.3.1.

Two From header fields are equivalent if their URIs match, and their parameters match. Extension parameters in one header field, not present in the other are ignored for the purposes of comparison. This means that the display name and presence or absence of angle brackets do not affect matching.

See Section 20.10 for the rules for parsing a display name, URI and URI parameters, and header field parameters.

The compact form of the From header field is f.

Examples:

```
From: "A. G. Bell" <sip:agb@bell-telephone.com> ;tag=a48s
From: sip:+12125551212@server.phone2net.com;tag=887s
f: Anonymous <sip:c8oqz84zk7z@privacy.org>;tag=hyh8
```

## 20.21 In-Reply-To

The In-Reply-To header field enumerates the Call-IDs that this call references or returns. These Call-IDs may have been cached by the client then included in this header field in a return call.

> This allows automatic call distribution systems to route return calls to the originator of the first call. This also allows callees to filter calls, so that only return calls for calls they originated will be accepted. This field is not a substitute for request authentication.

Example:

```
In-Reply-To: 70710@saturn.bell-tel.com, 17320@saturn.bell-tel.com
```

## 20.22 Max-Forwards

The Max-Forwards header field must be used with any SIP method to limit the number of proxies or gateways that can forward the request to the next downstream server. This can also be useful when the client is attempting to trace a request chain that appears to be failing or looping in mid-chain.

The Max-Forwards value is an integer in the range 0-255 indicating the remaining number of times this request message is allowed to be forwarded. This count is decremented by each server that forwards the request. The recommended initial value is 70.

This header field should be inserted by elements that can not otherwise guarantee loop detection. For example, a B2BUA should insert a Max-Forwards header field.

Example:

```
Max-Forwards: 6
```

## 20.23 Min-Expires

The Min-Expires header field conveys the minimum refresh interval supported for soft-state elements managed by that server. This includes Contact header fields that are stored by a registrar. The header field contains a decimal integer number of seconds from 0 to (2**32)-1. The use of the header field in a 423 (Interval Too Brief) response is described in Sections 10.2.8, 10.3, and 21.4.17.

Example:

```
Min-Expires: 60
```

## 20.24 MIME-Version

See [H19.4.1].

Example:

```
MIME-Version: 1.0
```

## 20.25 Organization

The Organization header field conveys the name of the organization to which the SIP element issuing the request or response belongs.

> The field MAY be used by client software to filter calls.

Example:

```
Organization: Boxes by Bob
```

## 20.26 Priority

The Priority header field indicates the urgency of the request as perceived by the client. The Priority header field describes the priority that the SIP request should have to the receiving human or its agent. For example, it may be factored into decisions about call routing and acceptance. For these decisions, a message containing no Priority header field SHOULD be treated as if it specified a Priority of "normal". The Priority header field does not influence the use of communications resources such as packet forwarding priority in routers or access to circuits in PSTN gateways. The header field can have the values "non-urgent", "normal", "urgent", and "emergency", but additional values can be defined elsewhere. It is RECOMMENDED that the value of "emergency" only be used when life, limb, or property are in imminent danger. Otherwise, there are no semantics defined for this header field.

> These are the values of RFC 2076 [38], with the addition of "emergency".

Examples:

```
Subject: A tornado is heading our way!
Priority: emergency
```

or

```
Subject: Weekend plans
Priority: non-urgent
```

## 20.27 Proxy-Authenticate

A Proxy-Authenticate header field value contains an authentication challenge.

The use of this header field is defined in [H14.33]. See Section 22.3 for further details on its usage.

Example:

```
Proxy-Authenticate: Digest realm="atlanta.com",
 domain="sip:ss1.carrier.com", qop="auth",
 nonce="f84f1cec41e6cbe5aea9c8e88d359",
 opaque="", stale=FALSE, algorithm=MD5
```

## 20.28 Proxy-Authorization

The Proxy-Authorization header field allows the client to identify itself (or its user) to a proxy that requires authentication. A Proxy-Authorization field value consists of credentials containing the authentication information of the user agent for the proxy and/or realm of the resource being requested.

See Section 22.3 for a definition of the usage of this header field.

This header field, along with Authorization, breaks the general rules about multiple header field names. Although not a comma-separated list, this header field name may be present multiple times, and MUST NOT be combined into a single header line using the usual rules described in Section 7.3.1.

Example:

```
Proxy-Authorization: Digest username="Alice", realm="atlanta.com",
   nonce="c60f3082ee1212b402a21831ae",
   response="245f23415f11432b3434341c022"
```

## 20.29 Proxy-Require

The Proxy-Require header field is used to indicate proxy-sensitive features that must be supported by the proxy. See Section 20.32 for more details on the mechanics of this message and a usage example.

Example:

```
Proxy-Require: foo
```

## 20.30 Record-Route

The Record-Route header field is inserted by proxies in a request to force future requests in the dialog to be routed through the proxy.

Examples of its use with the Route header field are described in Sections 16.12.1.

Example:

```
Record-Route: <sip:server10.biloxi.com;lr>,
              <sip:bigbox3.site3.atlanta.com;lr>
```

## 20.31 Reply-To

The Reply-To header field contains a logical return URI that may be different from the From header field. For example, the URI MAY be used to return missed calls or unestablished sessions. If the user wished to remain anonymous, the header field SHOULD either be omitted from the request or populated in such a way that does not reveal any private information.

Even if the "display-name" is empty, the "name-addr" form MUST be used if the "addr-spec" contains a comma, question mark, or semicolon. Syntax issues are discussed in Section 7.3.1.

Example:

```
Reply-To: Bob <sip:bob@biloxi.com>
```

## 20.32 Require

The Require header field is used by UACs to tell UASs about options that the UAC expects the UAS to support in order to process the request. Although an optional header field, the Require MUST NOT be ignored if it is present.

The Require header field contains a list of option tags, described in Section 19.2. Each option tag defines a SIP extension that MUST be understood to process the request. Frequently, this is used to indicate that a specific set of extension header fields need to be understood. A UAC compliant to this specification MUST only include option tags corresponding to standards-track RFCs.

Example:

```
Require: 100rel
```

## 20.33 Retry-After

The Retry-After header field can be used with a 500 (Server Internal Error) or 503 (Service Unavailable) response to indicate how long the service is expected to be unavailable to the requesting client and with a 404 (Not Found), 413 (Request Entity Too Large), 480 (Temporarily Unavailable), 486 (Busy Here), 600 (Busy), or 603

(Decline) response to indicate when the called party anticipates being available again. The value of this field is a positive integer number of seconds (in decimal) after the time of the response.

An optional comment can be used to indicate additional information about the time of callback. An optional "duration" parameter indicates how long the called party will be reachable starting at the initial time of availability. If no duration parameter is given, the service is assumed to be available indefinitely.

Examples:

```
Retry-After: 18000;duration=3600
Retry-After: 120 (I'm in a meeting)
```

## 20.34 Route

The Route header field is used to force routing for a request through the listed set of proxies. Examples of the use of the Route header field are in Section 16.12.1.

Example:

```
Route: <sip:bigbox3.site3.atlanta.com;lr>,
       <sip:server10.biloxi.com;lr>
```

## 20.35 Server

The Server header field contains information about the software used by the UAS to handle the request.

Revealing the specific software version of the server might allow the server to become more vulnerable to attacks against software that is known to contain security holes. Implementers SHOULD make the Server header field a configurable option.

Example:

```
Server: HomeServer v2
```

## 20.36 Subject

The Subject header field provides a summary or indicates the nature of the call, allowing call filtering without having to parse the session description. The session description does not have to use the same subject indication as the invitation.

The compact form of the Subject header field is s.

Example:

```
Subject: Need more boxes
s: Tech Support
```

## 20.37 Supported

The Supported header field enumerates all the extensions supported by the UAC or UAS.

The Supported header field contains a list of option tags, described in Section 19.2, that are understood by the UAC or UAS. A UA compliant to this specification MUST only include option tags corresponding to standards-track RFCs. If empty, it means that no extensions are supported.

The compact form of the Supported header field is k.

Example:

```
Supported: 100rel
```

## 20.38 Timestamp

The Timestamp header field describes when the UAC sent the request to the UAS.

See Section 8.2.6 for details on how to generate a response to a request that contains the header field. Although there is no normative behavior defined here that makes use of the header, it allows for extensions or SIP applications to obtain RTT estimates.

Example:

```
Timestamp: 54
```

## 20.39 To

The To header field specifies the logical recipient of the request.

The optional "display-name" is meant to be rendered by a human-user interface. The "tag" parameter serves as a general mechanism for dialog identification.

See Section 19.3 for details of the "tag" parameter.

Comparison of To header fields for equality is identical to comparison of From header fields. See Section 20.10 for the rules for parsing a display name, URI and URI parameters, and header field parameters.

The compact form of the To header field is t.

The following are examples of valid To header fields:

```
To: The Operator <sip:operator@cs.columbia.edu>;tag=287447
t: sip:+12125551212@server.phone2net.com
```

## 20.40 Unsupported

The Unsupported header field lists the features not supported by the UAS. See Section 20.32 for motivation.

Example:

```
Unsupported: foo
```

## 20.41 User-Agent

The User-Agent header field contains information about the UAC originating the request. The semantics of this header field are defined in [H14.43].

Revealing the specific software version of the user agent might allow the user agent to become more vulnerable to attacks against software that is known to contain security holes. Implementers SHOULD make the User-Agent header field a configurable option.

Example:

```
User-Agent: Softphone Beta1.5
```

## 20.42 Via

The Via header field indicates the path taken by the request so far and indicates the path that should be followed in routing responses. The branch ID parameter in the Via header field values serves as a transaction identifier, and is used by proxies to detect loops.

A Via header field value contains the transport protocol used to send the message, the client’s host name or network address, and possibly the port number at which it wishes to receive responses. A Via header field value can also contain parameters such as "maddr", "ttl", "received", and "branch", whose meaning and use are described in other sections. For implementations compliant to this specification, the value of the branch parameter MUST start with the magic cookie "z9hG4bK", as discussed in Section 8.1.1.7.

Transport protocols defined here are "UDP", "TCP", "TLS", and "SCTP". "TLS" means TLS over TCP. When a request is sent to a SIPS URI, the protocol still indicates "SIP", and the transport protocol is TLS.

```
Via: SIP/2.0/UDP erlang.bell-telephone.com:5060;branch=z9hG4bK87asdks7
Via: SIP/2.0/UDP 192.0.2.1:5060 ;received=192.0.2.207
     ;branch=z9hG4bK77asjd
```

The compact form of the Via header field is v.

In this example, the message originated from a multi-homed host with two addresses, 192.0.2.1 and 192.0.2.207. The sender guessed wrong as to which network interface would be used. Erlang.bell-telephone.com noticed the mismatch and added a parameter to the previous hop's Via header field value, containing the address that the packet actually came from.

The host or network address and port number are not required to follow the SIP URI syntax. Specifically, LWS on either side of the ":" or "/" is allowed, as shown here:

```
Via: SIP / 2.0 / UDP first.example.com: 4000;ttl=16
  ;maddr=224.2.0.1 ;branch=z9hG4bKa7c6a8dlze.1
```

Even though this specification mandates that the branch parameter be present in all requests, the BNF for the header field indicates that it is optional. This allows interoperation with RFC 2543 elements, which did not have to insert the branch parameter.

Two Via header fields are equal if their sent-protocol and sent-by fields are equal, both have the same set of parameters, and the values of all parameters are equal.

## 20.43 Warning

The Warning header field is used to carry additional information about the status of a response. Warning header field values are sent with responses and contain a three-digit warning code, host name, and warning text.

The "warn-text" should be in a natural language that is most likely to be intelligible to the human user receiving the response. This decision can be based on any available knowledge, such as the location of the user, the Accept-Language field in a request, or the

Content-Language field in a response. The default language is i-default [21].

The currently-defined "warn-code"s are listed below, with a recommended warn-text in English and a description of their meaning. These warnings describe failures induced by the session description. The first digit of warning codes beginning with "3" indicates warnings specific to SIP. Warnings 300 through 329 are reserved for indicating problems with keywords in the session description, 330 through 339 are warnings related to basic network services requested in the session description, 370 through 379 are warnings related to quantitative QoS parameters requested in the session description, and 390 through 399 are miscellaneous warnings that do not fall into one of the above categories.

300 Incompatible network protocol: One or more network protocols contained in the session description are not available.

301 Incompatible network address formats: One or more network address formats contained in the session description are not available.

302 Incompatible transport protocol: One or more transport protocols described in the session description are not available.

303 Incompatible bandwidth units: One or more bandwidth measurement units contained in the session description were not understood.

304 Media type not available: One or more media types contained in the session description are not available.

305 Incompatible media format: One or more media formats contained in the session description are not available.

306 Attribute not understood: One or more of the media attributes in the session description are not supported.

307 Session description parameter not understood: A parameter other than those listed above was not understood.

330 Multicast not available: The site where the user is located does not support multicast.

331 Unicast not available: The site where the user is located does not support unicast communication (usually due to the presence of a firewall).

370 Insufficient bandwidth: The bandwidth specified in the session description or defined by the media exceeds that known to be available.

399 Miscellaneous warning: The warning text can include arbitrary information to be presented to a human user or logged. A system receiving this warning MUST NOT take any automated action.

1xx and 2xx have been taken by HTTP/1.1.

Additional "warn-code"s can be defined through IANA, as defined in Section 27.2.

Examples:

```
Warning: 307 isi.edu "Session parameter 'foo' not understood"
Warning: 301 isi.edu "Incompatible network address type 'E.164'"
```

## 20.44 WWW-Authenticate

A WWW-Authenticate header field value contains an authentication challenge. See Section 22.2 for further details on its usage.

Example:

```
WWW-Authenticate: Digest realm="atlanta.com",
  domain="sip:boxesbybob.com", qop="auth",
  nonce="f84f1cec41e6cbe5aea9c8e88d359",
  opaque="", stale=FALSE, algorithm=MD5
```

# 21 Response Codes

The response codes are consistent with, and extend, HTTP/1.1 response codes. Not all HTTP/1.1 response codes are appropriate, and only those that are appropriate are given here. Other HTTP/1.1 response codes SHOULD NOT be used. Also, SIP defines a new class, 6xx.

## 21.1 Provisional 1xx

Provisional responses, also known as informational responses, indicate that the server contacted is performing some further action and does not yet have a definitive response. A server sends a 1xx response if it expects to take more than 200 ms to obtain a final response. Note that 1xx responses are not transmitted reliably. They never cause the client to send an ACK. Provisional (1xx) responses MAY contain message bodies, including session descriptions.

### 21.1.1 100 Trying

This response indicates that the request has been received by the next-hop server and that some unspecified action is being taken on behalf of this call (for example, a database is being consulted). This response, like all other provisional responses, stops retransmissions of an INVITE by a UAC. The 100 (Trying) response is different from other provisional responses, in that it is never forwarded upstream by a stateful proxy.

### 21.1.2 180 Ringing

The UA receiving the INVITE is trying to alert the user. This response MAY be used to initiate local ringback.

### 21.1.3 181 Call Is Being Forwarded

A server MAY use this status code to indicate that the call is being forwarded to a different set of destinations.

### 21.1.4 182 Queued

The called party is temporarily unavailable, but the server has decided to queue the call rather than reject it. When the callee becomes available, it will return the appropriate final status response. The reason phrase MAY give further details about the status of the call, for example, "5 calls queued; expected waiting time is 15 minutes". The server MAY issue several 182 (Queued) responses to update the caller about the status of the queued call.

### 21.1.5 183 Session Progress

The 183 (Session Progress) response is used to convey information about the progress of the call that is not otherwise classified. The Reason-Phrase, header fields, or message body MAY be used to convey more details about the call progress.

## 21.2 Successful 2xx

The request was successful.

### 21.2.1 200 OK

The request has succeeded. The information returned with the response depends on the method used in the request.

### 21.3 Redirection 3xx

3xx responses give information about the user's new location, or about alternative services that might be able to satisfy the call.

#### 21.3.1 300 Multiple Choices

The address in the request resolved to several choices, each with its own specific location, and the user (or UA) can select a preferred communication end point and redirect its request to that location.

The response MAY include a message body containing a list of resource characteristics and location(s) from which the user or UA can choose the one most appropriate, if allowed by the Accept request header field. However, no MIME types have been defined for this message body.

The choices SHOULD also be listed as Contact fields (Section 20.10). Unlike HTTP, the SIP response MAY contain several Contact fields or a list of addresses in a Contact field. UAs MAY use the Contact header field value for automatic redirection or MAY ask the user to confirm a choice. However, this specification does not define any standard for such automatic selection.

> This status response is appropriate if the callee can be reached at several different locations and the server cannot or prefers not to proxy the request.

#### 21.3.2 301 Moved Permanently

The user can no longer be found at the address in the Request-URI, and the requesting client SHOULD retry at the new address given by the Contact header field (Section 20.10). The requestor SHOULD update any local directories, address books, and user location caches with this new value and redirect future requests to the address(es) listed.

#### 21.3.3 302 Moved Temporarily

The requesting client SHOULD retry the request at the new address(es) given by the Contact header field (Section 20.10). The Request-URI of the new request uses the value of the Contact header field in the response.

The duration of the validity of the Contact URI can be indicated through an Expires (Section 20.19) header field or an expires parameter in the Contact header field. Both proxies and UAs MAY cache this URI for the duration of the expiration time. If there is no explicit expiration time, the address is only valid once for recursing, and MUST NOT be cached for future transactions.

If the URI cached from the Contact header field fails, the Request-URI from the redirected request MAY be tried again a single time.

> The temporary URI may have become out-of-date sooner than the expiration time, and a new temporary URI may be available.

### 21.3.4 305 Use Proxy

The requested resource MUST be accessed through the proxy given by the Contact field. The Contact field gives the URI of the proxy. The recipient is expected to repeat this single request via the proxy. 305 (Use Proxy) responses MUST only be generated by UASs.

### 21.3.5 380 Alternative Service

The call was not successful, but alternative services are possible.

The alternative services are described in the message body of the response. Formats for such bodies are not defined here, and may be the subject of future standardization.

## 21.4 Request Failure 4xx

4xx responses are definite failure responses from a particular server. The client SHOULD NOT retry the same request without modification (for example, adding appropriate authorization). However, the same request to a different server might be successful.

### 21.4.1 400 Bad Request

The request could not be understood due to malformed syntax. The Reason-Phrase SHOULD identify the syntax problem in more detail, for example, "Missing Call-ID header field".

### 21.4.2 401 Unauthorized

The request requires user authentication. This response is issued by UASs and registrars, while 407 (Proxy Authentication Required) is used by proxy servers.

#### 21.4.3 402 Payment Required

Reserved for future use.

#### 21.4.4 403 Forbidden

The server understood the request, but is refusing to fulfill it. Authorization will not help, and the request SHOULD NOT be repeated.

#### 21.4.5 404 Not Found

The server has definitive information that the user does not exist at the domain specified in the Request-URI. This status is also returned if the domain in the Request-URI does not match any of the domains handled by the recipient of the request.

#### 21.4.6 405 Method Not Allowed

The method specified in the Request-Line is understood, but not allowed for the address identified by the Request-URI.

The response MUST include an Allow header field containing a list of valid methods for the indicated address.

#### 21.4.7 406 Not Acceptable

The resource identified by the request is only capable of generating response entities that have content characteristics not acceptable according to the Accept header field sent in the request.

#### 21.4.8 407 Proxy Authentication Required

This code is similar to 401 (Unauthorized), but indicates that the client MUST first authenticate itself with the proxy. SIP access authentication is explained in Sections 26 and 22.3.

This status code can be used for applications where access to the communication channel (for example, a telephony gateway) rather than the callee requires authentication.

#### 21.4.9 408 Request Timeout

The server could not produce a response within a suitable amount of time, for example, if it could not determine the location of the user in time. The client MAY repeat the request without modifications at any later time.

#### 21.4.10 410 Gone

The requested resource is no longer available at the server and no forwarding address is known. This condition is expected to be considered permanent. If the server does not know, or has no facility to determine, whether or not the condition is permanent, the status code 404 (Not Found) SHOULD be used instead.

#### 21.4.11 413 Request Entity Too Large

The server is refusing to process a request because the request entity-body is larger than the server is willing or able to process. The server MAY close the connection to prevent the client from continuing the request.

If the condition is temporary, the server SHOULD include a Retry-After header field to indicate that it is temporary and after what time the client MAY try again.

#### 21.4.12 414 Request-URI Too Long

The server is refusing to service the request because the Request-URI is longer than the server is willing to interpret.

#### 21.4.13 415 Unsupported Media Type

The server is refusing to service the request because the message body of the request is in a format not supported by the server for the requested method. The server MUST return a list of acceptable formats using the Accept, Accept-Encoding, or Accept-Language header field, depending on the specific problem with the content. UAC processing of this response is described in Section 8.1.3.5.

#### 21.4.14 416 Unsupported URI Scheme

The server cannot process the request because the scheme of the URI in the Request-URI is unknown to the server. Client processing of this response is described in Section 8.1.3.5.

#### 21.4.15 420 Bad Extension

The server did not understand the protocol extension specified in a Proxy-Require (Section 20.29) or Require (Section 20.32) header field. The server MUST include a list of the unsupported extensions in an Unsupported header field in the response. UAC processing of this response is described in Section 8.1.3.5.

### 21.4.16 421 Extension Required

The UAS needs a particular extension to process the request, but this extension is not listed in a Supported header field in the request. Responses with this status code MUST contain a Require header field listing the required extensions.

A UAS SHOULD NOT use this response unless it truly cannot provide any useful service to the client. Instead, if a desirable extension is not listed in the Supported header field, servers SHOULD process the request using baseline SIP capabilities and any extensions supported by the client.

### 21.4.17 423 Interval Too Brief

The server is rejecting the request because the expiration time of the resource refreshed by the request is too short. This response can be used by a registrar to reject a registration whose Contact header field expiration time was too small. The use of this response and the related Min-Expires header field are described in Sections 10.2.8, 10.3, and 20.23.

### 21.4.18 480 Temporarily Unavailable

The callee’s end system was contacted successfully but the callee is currently unavailable (for example, is not logged in, logged in but in a state that precludes communication with the callee, or has activated the "do not disturb" feature). The response MAY indicate a better time to call in the Retry-After header field. The user could also be available elsewhere (unbeknownst to this server). The reason phrase SHOULD indicate a more precise cause as to why the callee is unavailable. This value SHOULD be settable by the UA. Status 486 (Busy Here) MAY be used to more precisely indicate a particular reason for the call failure.

This status is also returned by a redirect or proxy server that recognizes the user identified by the Request-URI, but does not currently have a valid forwarding location for that user.

### 21.4.19 481 Call/Transaction Does Not Exist

This status indicates that the UAS received a request that does not match any existing dialog or transaction.

### 21.4.20 482 Loop Detected

The server has detected a loop (Section 16.3 Item 4).

#### 21.4.21 483 Too Many Hops

The server received a request that contains a Max-Forwards (Section 20.22) header field with the value zero.

#### 21.4.22 484 Address Incomplete

The server received a request with a Request-URI that was incomplete. Additional information SHOULD be provided in the reason phrase.

> This status code allows overlapped dialing. With overlapped dialing, the client does not know the length of the dialing string. It sends strings of increasing lengths, prompting the user for more input, until it no longer receives a 484 (Address Incomplete) status response.

#### 21.4.23 485 Ambiguous

The Request-URI was ambiguous. The response MAY contain a listing of possible unambiguous addresses in Contact header fields. Revealing alternatives can infringe on privacy of the user or the organization. It MUST be possible to configure a server to respond with status 404 (Not Found) or to suppress the listing of possible choices for ambiguous Request-URIs.

Example response to a request with the Request-URI sip:lee@example.com:

```
SIP/2.0 485 Ambiguous
Contact: Carol Lee <sip:carol.lee@example.com>
Contact: Ping Lee <sip:p.lee@example.com>
Contact: Lee M. Foote <sips:lee.foote@example.com>
```

> Some email and voice mail systems provide this functionality. A status code separate from 3xx is used since the semantics are different: for 300, it is assumed that the same person or service will be reached by the choices provided. While an automated choice or sequential search makes sense for a 3xx response, user intervention is required for a 485 (Ambiguous) response.

#### 21.4.24 486 Busy Here

The callee's end system was contacted successfully, but the callee is currently not willing or able to take additional calls at this end system. The response MAY indicate a better time to call in the Retry-After header field. The user could also be available elsewhere, such as through a voice mail service. Status 600 (Busy Everywhere) SHOULD be used if the client knows that no other end system will be able to accept this call.

#### 21.4.25 487 Request Terminated

The request was terminated by a BYE or CANCEL request. This response is never returned for a CANCEL request itself.

#### 21.4.26 488 Not Acceptable Here

The response has the same meaning as 606 (Not Acceptable), but only applies to the specific resource addressed by the Request-URI and the request may succeed elsewhere.

A message body containing a description of media capabilities MAY be present in the response, which is formatted according to the Accept header field in the INVITE (or application/sdp if not present), the same as a message body in a 200 (OK) response to an OPTIONS request.

#### 21.4.27 491 Request Pending

The request was received by a UAS that had a pending request within the same dialog. Section 14.2 describes how such "glare" situations are resolved.

#### 21.4.28 493 Undecipherable

The request was received by a UAS that contained an encrypted MIME body for which the recipient does not possess or will not provide an appropriate decryption key. This response MAY have a single body containing an appropriate public key that should be used to encrypt MIME bodies sent to this UA. Details of the usage of this response code can be found in Section 23.2.

### 21.5 Server Failure 5xx

5xx responses are failure responses given when a server itself has erred.

#### 21.5.1 500 Server Internal Error

The server encountered an unexpected condition that prevented it from fulfilling the request. The client MAY display the specific error condition and MAY retry the request after several seconds.

If the condition is temporary, the server MAY indicate when the client may retry the request using the Retry-After header field.

### 21.5.2 501 Not Implemented

The server does not support the functionality required to fulfill the request. This is the appropriate response when a UAS does not recognize the request method and is not capable of supporting it for any user. (Proxies forward all requests regardless of method.)

Note that a 405 (Method Not Allowed) is sent when the server recognizes the request method, but that method is not allowed or supported.

### 21.5.3 502 Bad Gateway

The server, while acting as a gateway or proxy, received an invalid response from the downstream server it accessed in attempting to fulfill the request.

### 21.5.4 503 Service Unavailable

The server is temporarily unable to process the request due to a temporary overloading or maintenance of the server. The server MAY indicate when the client should retry the request in a Retry-After header field. If no Retry-After is given, the client MUST act as if it had received a 500 (Server Internal Error) response.

A client (proxy or UAC) receiving a 503 (Service Unavailable) SHOULD attempt to forward the request to an alternate server. It SHOULD NOT forward any other requests to that server for the duration specified in the Retry-After header field, if present.

Servers MAY refuse the connection or drop the request instead of responding with 503 (Service Unavailable).

### 21.5.5 504 Server Time-out

The server did not receive a timely response from an external server it accessed in attempting to process the request. 408 (Request Timeout) should be used instead if there was no response within the period specified in the Expires header field from the upstream server.

### 21.5.6 505 Version Not Supported

The server does not support, or refuses to support, the SIP protocol version that was used in the request. The server is indicating that it is unable or unwilling to complete the request using the same major version as the client, other than with this error message.

### 21.5.7 513 Message Too Large

The server was unable to process the request since the message length exceeded its capabilities.

## 21.6 Global Failures 6xx

6xx responses indicate that a server has definitive information about a particular user, not just the particular instance indicated in the Request-URI.

### 21.6.1 600 Busy Everywhere

The callee's end system was contacted successfully but the callee is busy and does not wish to take the call at this time. The response MAY indicate a better time to call in the Retry-After header field. If the callee does not wish to reveal the reason for declining the call, the callee uses status code 603 (Decline) instead. This status response is returned only if the client knows that no other end point (such as a voice mail system) will answer the request. Otherwise, 486 (Busy Here) should be returned.

### 21.6.2 603 Decline

The callee's machine was successfully contacted but the user explicitly does not wish to or cannot participate. The response MAY indicate a better time to call in the Retry-After header field. This status response is returned only if the client knows that no other end point will answer the request.

### 21.6.3 604 Does Not Exist Anywhere

The server has authoritative information that the user indicated in the Request-URI does not exist anywhere.

### 21.6.4 606 Not Acceptable

The user's agent was contacted successfully but some aspects of the session description such as the requested media, bandwidth, or addressing style were not acceptable.

A 606 (Not Acceptable) response means that the user wishes to communicate, but cannot adequately support the session described. The 606 (Not Acceptable) response MAY contain a list of reasons in a Warning header field describing why the session described cannot be supported. Warning reason codes are listed in Section 20.43.

A message body containing a description of media capabilities MAY be present in the response, which is formatted according to the Accept header field in the INVITE (or application/sdp if not present), the same as a message body in a 200 (OK) response to an OPTIONS request.

It is hoped that negotiation will not frequently be needed, and when a new user is being invited to join an already existing conference, negotiation may not be possible. It is up to the invitation initiator to decide whether or not to act on a 606 (Not Acceptable) response.

This status response is returned only if the client knows that no other end point will answer the request.

## 22 Usage of HTTP Authentication

SIP provides a stateless, challenge-based mechanism for authentication that is based on authentication in HTTP. Any time that a proxy server or UA receives a request (with the exceptions given in Section 22.1), it MAY challenge the initiator of the request to provide assurance of its identity. Once the originator has been identified, the recipient of the request SHOULD ascertain whether or not this user is authorized to make the request in question. No authorization systems are recommended or discussed in this document.

The "Digest" authentication mechanism described in this section provides message authentication and replay protection only, without message integrity or confidentiality. Protective measures above and beyond those provided by Digest need to be taken to prevent active attackers from modifying SIP requests and responses.

Note that due to its weak security, the usage of "Basic" authentication has been deprecated. Servers MUST NOT accept credentials using the "Basic" authorization scheme, and servers also MUST NOT challenge with "Basic". This is a change from RFC 2543.

### 22.1 Framework

The framework for SIP authentication closely parallels that of HTTP (RFC 2617 [17]). In particular, the BNF for auth-scheme, auth-param, challenge, realm, realm-value, and credentials is identical (although the usage of "Basic" as a scheme is not permitted). In SIP, a UAS uses the 401 (Unauthorized) response to challenge the identity of a UAC. Additionally, registrars and redirect servers MAY make use of 401 (Unauthorized) responses for authentication, but proxies MUST NOT, and instead MAY use the 407 (Proxy Authentication Required) response. The requirements for inclusion of the Proxy-Authenticate, Proxy-Authorization, WWW-Authenticate, and Authorization in the various messages are identical to those described in RFC 2617 [17].

Since SIP does not have the concept of a canonical root URL, the notion of protection spaces is interpreted differently in SIP. The realm string alone defines the protection domain. This is a change from RFC 2543, in which the Request-URI and the realm together defined the protection domain.

> This previous definition of protection domain caused some amount of confusion since the Request-URI sent by the UAC and the Request-URI received by the challenging server might be different, and indeed the final form of the Request-URI might not be known to the UAC. Also, the previous definition depended on the presence of a SIP URI in the Request-URI and seemed to rule out alternative URI schemes (for example, the tel URL).

Operators of user agents or proxy servers that will authenticate received requests MUST adhere to the following guidelines for creation of a realm string for their server:

- o Realm strings MUST be globally unique. It is RECOMMENDED that a realm string contain a hostname or domain name, following the recommendation in Section 3.2.1 of RFC 2617 [17].

- o Realm strings SHOULD present a human-readable identifier that can be rendered to a user.

For example:

```
INVITE sip:bob@biloxi.com SIP/2.0
Authorization: Digest realm="biloxi.com", <...>
```

Generally, SIP authentication is meaningful for a specific realm, a protection domain. Thus, for Digest authentication, each such protection domain has its own set of usernames and passwords. If a server does not require authentication for a particular request, it MAY accept a default username, "anonymous", which has no password (password of ""). Similarly, UACs representing many users, such as PSTN gateways, MAY have their own device-specific username and password, rather than accounts for particular users, for their realm.

While a server can legitimately challenge most SIP requests, there are two requests defined by this document that require special handling for authentication: ACK and CANCEL.

Under an authentication scheme that uses responses to carry values used to compute nonces (such as Digest), some problems come up for any requests that take no response, including ACK. For this reason, any credentials in the INVITE that were accepted by a server MUST be accepted by that server for the ACK. UACs creating an ACK message will duplicate all of the Authorization and Proxy-Authorization header field values that appeared in the INVITE to which the ACK corresponds. Servers MUST NOT attempt to challenge an ACK.

Although the CANCEL method does take a response (a 2xx), servers MUST NOT attempt to challenge CANCEL requests since these requests cannot be resubmitted. Generally, a CANCEL request SHOULD be accepted by a server if it comes from the same hop that sent the request being canceled (provided that some sort of transport or network layer security association, as described in Section 26.2.1, is in place).

When a UAC receives a challenge, it SHOULD render to the user the contents of the "realm" parameter in the challenge (which appears in either a WWW-Authenticate header field or Proxy-Authenticate header field) if the UAC device does not already know of a credential for the realm in question. A service provider that pre-configures UAs with credentials for its realm should be aware that users will not have the opportunity to present their own credentials for this realm when challenged at a pre-configured device.

Finally, note that even if a UAC can locate credentials that are associated with the proper realm, the potential exists that these credentials may no longer be valid or that the challenging server will not accept these credentials for whatever reason (especially when "anonymous" with no password is submitted). In this instance a server may repeat its challenge, or it may respond with a 403 Forbidden. A UAC MUST NOT re-attempt requests with the credentials that have just been rejected (though the request may be retried if the nonce was stale).

## 22.2 User-to-User Authentication

When a UAS receives a request from a UAC, the UAS MAY authenticate the originator before the request is processed. If no credentials (in the Authorization header field) are provided in the request, the UAS can challenge the originator to provide credentials by rejecting the request with a 401 (Unauthorized) status code.

The WWW-Authenticate response-header field MUST be included in 401 (Unauthorized) response messages. The field value consists of at least one challenge that indicates the authentication scheme(s) and parameters applicable to the realm.

An example of the WWW-Authenticate header field in a 401 challenge is:

```
WWW-Authenticate: Digest
        realm="biloxi.com",
        qop="auth,auth-int",
        nonce="dcd98b7102dd2f0e8b11d0f600bfb0c093",
        opaque="5ccc069c403ebaf9f0171e9517f40e41"
```

When the originating UAC receives the 401 (Unauthorized), it SHOULD, if it is able, re-originate the request with the proper credentials. The UAC may require input from the originating user before proceeding. Once authentication credentials have been supplied (either directly by the user, or discovered in an internal keyring), UAs SHOULD cache the credentials for a given value of the To header field and "realm" and attempt to re-use these values on the next request for that destination. UAs MAY cache credentials in any way they would like.

If no credentials for a realm can be located, UACs MAY attempt to retry the request with a username of "anonymous" and no password (a password of "").

Once credentials have been located, any UA that wishes to authenticate itself with a UAS or registrar -- usually, but not necessarily, after receiving a 401 (Unauthorized) response -- MAY do so by including an Authorization header field with the request. The Authorization field value consists of credentials containing the authentication information of the UA for the realm of the resource being requested as well as parameters required in support of authentication and replay protection.

An example of the Authorization header field is:

```
Authorization: Digest username="bob",
        realm="biloxi.com",
        nonce="dcd98b7102dd2f0e8b11d0f600bfb0c093",
        uri="sip:bob@biloxi.com",
        qop=auth,
        nc=00000001,
        cnonce="0a4f113b",
        response="6629fae49393a05397450978507c4ef1",
        opaque="5ccc069c403ebaf9f0171e9517f40e41"
```

When a UAC resubmits a request with its credentials after receiving a 401 (Unauthorized) or 407 (Proxy Authentication Required) response, it MUST increment the CSeq header field value as it would normally when sending an updated request.

## 22.3 Proxy-to-User Authentication

Similarly, when a UAC sends a request to a proxy server, the proxy server MAY authenticate the originator before the request is processed. If no credentials (in the Proxy-Authorization header field) are provided in the request, the proxy can challenge the originator to provide credentials by rejecting the request with a 407 (Proxy Authentication Required) status code. The proxy MUST populate the 407 (Proxy Authentication Required) message with a Proxy-Authenticate header field value applicable to the proxy for the requested resource.

The use of Proxy-Authenticate and Proxy-Authorization parallel that described in [17], with one difference. Proxies MUST NOT add values to the Proxy-Authorization header field. All 407 (Proxy Authentication Required) responses MUST be forwarded upstream toward the UAC following the procedures for any other response. It is the UAC's responsibility to add the Proxy-Authorization header field value containing credentials for the realm of the proxy that has asked for authentication.

> If a proxy were to resubmit a request adding a Proxy-Authorization header field value, it would need to increment the CSeq in the new request. However, this would cause the UAC that submitted the original request to discard a response from the UAS, as the CSeq value would be different.

When the originating UAC receives the 407 (Proxy Authentication Required) it SHOULD, if it is able, re-originate the request with the proper credentials. It should follow the same procedures for the display of the "realm" parameter that are given above for responding to 401.

If no credentials for a realm can be located, UACs MAY attempt to retry the request with a username of "anonymous" and no password (a password of "").

The UAC SHOULD also cache the credentials used in the re-originated request.

The following rule is RECOMMENDED for proxy credential caching:

If a UA receives a Proxy-Authenticate header field value in a 401/407 response to a request with a particular Call-ID, it should incorporate credentials for that realm in all subsequent requests that contain the same Call-ID. These credentials MUST NOT be cached across dialogs; however, if a UA is configured with the realm of its local outbound proxy, when one exists, then the UA MAY cache credentials for that realm across dialogs. Note that this does mean a future request in a dialog could contain credentials that are not needed by any proxy along the Route header path.

Any UA that wishes to authenticate itself to a proxy server -- usually, but not necessarily, after receiving a 407 (Proxy Authentication Required) response -- MAY do so by including a Proxy-Authorization header field value with the request. The Proxy-Authorization request-header field allows the client to identify itself (or its user) to a proxy that requires authentication. The Proxy-Authorization header field value consists of credentials containing the authentication information of the UA for the proxy and/or realm of the resource being requested.

A Proxy-Authorization header field value applies only to the proxy whose realm is identified in the "realm" parameter (this proxy may previously have demanded authentication using the Proxy-Authenticate field). When multiple proxies are used in a chain, a Proxy-Authorization header field value MUST NOT be consumed by any proxy whose realm does not match the "realm" parameter specified in that value.

Note that if an authentication scheme that does not support realms is used in the Proxy-Authorization header field, a proxy server MUST attempt to parse all Proxy-Authorization header field values to determine whether one of them has what the proxy server considers to be valid credentials. Because this is potentially very time-consuming in large networks, proxy servers SHOULD use an authentication scheme that supports realms in the Proxy-Authorization header field.

If a request is forked (as described in Section 16.7), various proxy servers and/or UAs may wish to challenge the UAC. In this case, the forking proxy server is responsible for aggregating these challenges into a single response. Each WWW-Authenticate and Proxy-Authenticate value received in responses to the forked request MUST be placed into the single response that is sent by the forking proxy to the UA; the ordering of these header field values is not significant.

> When a proxy server issues a challenge in response to a request, it will not proxy the request until the UAC has retried the request with valid credentials. A forking proxy may forward a request simultaneously to multiple proxy servers that require authentication, each of which in turn will not forward the request until the originating UAC has authenticated itself in their respective realm. If the UAC does not provide credentials for

each challenge, the proxy servers that issued the challenges will not forward requests to the UA where the destination user might be located, and therefore, the virtues of forking are largely lost.

When resubmitting its request in response to a 401 (Unauthorized) or 407 (Proxy Authentication Required) that contains multiple challenges, a UAC MAY include an Authorization value for each WWW-Authenticate value and a Proxy-Authorization value for each Proxy-Authenticate value for which the UAC wishes to supply a credential. As noted above, multiple credentials in a request SHOULD be differentiated by the "realm" parameter.

It is possible for multiple challenges associated with the same realm to appear in the same 401 (Unauthorized) or 407 (Proxy Authentication Required). This can occur, for example, when multiple proxies within the same administrative domain, which use a common realm, are reached by a forking request. When it retries a request, a UAC MAY therefore supply multiple credentials in Authorization or Proxy-Authorization header fields with the same "realm" parameter value. The same credentials SHOULD be used for the same realm.

## 22.4 The Digest Authentication Scheme

This section describes the modifications and clarifications required to apply the HTTP Digest authentication scheme to SIP. The SIP scheme usage is almost completely identical to that for HTTP [17].

Since RFC 2543 is based on HTTP Digest as defined in RFC 2069 [39], SIP servers supporting RFC 2617 MUST ensure they are backwards compatible with RFC 2069. Procedures for this backwards compatibility are specified in RFC 2617. Note, however, that SIP servers MUST NOT accept or request Basic authentication.

The rules for Digest authentication follow those defined in [17], with "HTTP/1.1" replaced by "SIP/2.0" in addition to the following differences:

1. The URI included in the challenge has the following BNF:

   ```
   URI  =  SIP-URI / SIPS-URI
   ```

2. The BNF in RFC 2617 has an error in that the 'uri' parameter of the Authorization header field for HTTP Digest

authentication is not enclosed in quotation marks. (The example in Section 3.5 of RFC 2617 is correct.) For SIP, the 'uri' MUST be enclosed in quotation marks.

3. The BNF for digest-uri-value is:

```
digest-uri-value  =  Request-URI ; as defined in Section 25
```

4. The example procedure for choosing a nonce based on Etag does not work for SIP.

5. The text in RFC 2617 [17] regarding cache operation does not apply to SIP.

6. RFC 2617 [17] requires that a server check that the URI in the request line and the URI included in the Authorization header field point to the same resource. In a SIP context, these two URIs may refer to different users, due to forwarding at some proxy. Therefore, in SIP, a server MAY check that the Request-URI in the Authorization header field value corresponds to a user for whom the server is willing to accept forwarded or direct requests, but it is not necessarily a failure if the two fields are not equivalent.

7. As a clarification to the calculation of the A2 value for message integrity assurance in the Digest authentication scheme, implementers should assume, when the entity-body is empty (that is, when SIP messages have no body) that the hash of the entity-body resolves to the MD5 hash of an empty string, or:

```
   H(entity-body) = MD5("") =
"d41d8cd98f00b204e9800998ecf8427e"
```

8. RFC 2617 notes that a cnonce value MUST NOT be sent in an Authorization (and by extension Proxy-Authorization) header field if no qop directive has been sent. Therefore, any algorithms that have a dependency on the cnonce (including "MD5-Sess") require that the qop directive be sent. Use of the "qop" parameter is optional in RFC 2617 for the purposes of backwards compatibility with RFC 2069; since RFC 2543 was based on RFC 2069, the "qop" parameter must unfortunately remain optional for clients and servers to receive. However, servers MUST always send a "qop" parameter in WWW-Authenticate and Proxy-Authenticate header field values. If a client receives a "qop" parameter in a challenge header field, it MUST send the "qop" parameter in any resulting authorization header field.

RFC 2543 did not allow usage of the Authentication-Info header field (it effectively used RFC 2069). However, we now allow usage of this header field, since it provides integrity checks over the bodies and provides mutual authentication. RFC 2617 [17] defines mechanisms for backwards compatibility using the qop attribute in the request. These mechanisms MUST be used by a server to determine if the client supports the new mechanisms in RFC 2617 that were not specified in RFC 2069.

# 23 S/MIME

SIP messages carry MIME bodies and the MIME standard includes mechanisms for securing MIME contents to ensure both integrity and confidentiality (including the 'multipart/signed' and 'application/pkcs7-mime' MIME types, see RFC 1847 [22], RFC 2630 [23] and RFC 2633 [24]). Implementers should note, however, that there may be rare network intermediaries (not typical proxy servers) that rely on viewing or modifying the bodies of SIP messages (especially SDP), and that secure MIME may prevent these sorts of intermediaries from functioning.

> This applies particularly to certain types of firewalls.

> The PGP mechanism for encrypting the header fields and bodies of SIP messages described in RFC 2543 has been deprecated.

## 23.1 S/MIME Certificates

The certificates that are used to identify an end-user for the purposes of S/MIME differ from those used by servers in one important respect - rather than asserting that the identity of the holder corresponds to a particular hostname, these certificates assert that the holder is identified by an end-user address. This address is composed of the concatenation of the "userinfo" "@" and "domainname" portions of a SIP or SIPS URI (in other words, an email address of the form "bob@biloxi.com"), most commonly corresponding to a user's address-of-record.

These certificates are also associated with keys that are used to sign or encrypt bodies of SIP messages. Bodies are signed with the private key of the sender (who may include their public key with the message as appropriate), but bodies are encrypted with the public key of the intended recipient. Obviously, senders must have foreknowledge of the public key of recipients in order to encrypt message bodies. Public keys can be stored within a UA on a virtual keyring.

Each user agent that supports S/MIME MUST contain a keyring specifically for end-users' certificates. This keyring should map between addresses of record and corresponding certificates. Over time, users SHOULD use the same certificate when they populate the originating URI of signaling (the From header field) with the same address-of-record.

Any mechanisms depending on the existence of end-user certificates are seriously limited in that there is virtually no consolidated authority today that provides certificates for end-user applications. However, users SHOULD acquire certificates from known public certificate authorities. As an alternative, users MAY create self-signed certificates. The implications of self-signed certificates are explored further in Section 26.4.2. Implementations may also use pre-configured certificates in deployments in which a previous trust relationship exists between all SIP entities.

Above and beyond the problem of acquiring an end-user certificate, there are few well-known centralized directories that distribute end-user certificates. However, the holder of a certificate SHOULD publish their certificate in any public directories as appropriate. Similarly, UACs SHOULD support a mechanism for importing (manually or automatically) certificates discovered in public directories corresponding to the target URIs of SIP requests.

## 23.2 S/MIME Key Exchange

SIP itself can also be used as a means to distribute public keys in the following manner.

Whenever the CMS SignedData message is used in S/MIME for SIP, it MUST contain the certificate bearing the public key necessary to verify the signature.

When a UAC sends a request containing an S/MIME body that initiates a dialog, or sends a non-INVITE request outside the context of a dialog, the UAC SHOULD structure the body as an S/MIME 'multipart/signed' CMS SignedData body. If the desired CMS service is EnvelopedData (and the public key of the target user is known), the UAC SHOULD send the EnvelopedData message encapsulated within a SignedData message.

When a UAS receives a request containing an S/MIME CMS body that includes a certificate, the UAS SHOULD first validate the certificate, if possible, with any available root certificates for certificate authorities. The UAS SHOULD also determine the subject of the certificate (for S/MIME, the SubjectAltName will contain the appropriate identity) and compare this value to the From header field of the request. If the certificate cannot be verified, because it is self-signed, or signed by no known authority, or if it is verifiable but its subject does not correspond to the From header field of request, the UAS MUST notify its user of the status of the certificate (including the subject of the certificate, its signer, and any key fingerprint information) and request explicit permission before proceeding. If the certificate was successfully verified and the subject of the certificate corresponds to the From header field of the SIP request, or if the user (after notification) explicitly authorizes the use of the certificate, the UAS SHOULD add this certificate to a local keyring, indexed by the address-of-record of the holder of the certificate.

When a UAS sends a response containing an S/MIME body that answers the first request in a dialog, or a response to a non-INVITE request outside the context of a dialog, the UAS SHOULD structure the body as an S/MIME 'multipart/signed' CMS SignedData body. If the desired CMS service is EnvelopedData, the UAS SHOULD send the EnvelopedData message encapsulated within a SignedData message.

When a UAC receives a response containing an S/MIME CMS body that includes a certificate, the UAC SHOULD first validate the certificate, if possible, with any appropriate root certificate. The UAC SHOULD also determine the subject of the certificate and compare this value to the To field of the response; although the two may very well be different, and this is not necessarily indicative of a security breach. If the certificate cannot be verified because it is self-signed, or signed by no known authority, the UAC MUST notify its user of the status of the certificate (including the subject of the certificate, its signator, and any key fingerprint information) and request explicit permission before proceeding. If the certificate was successfully verified, and the subject of the certificate corresponds to the To header field in the response, or if the user (after notification) explicitly authorizes the use of the certificate, the UAC SHOULD add this certificate to a local keyring, indexed by the address-of-record of the holder of the certificate. If the UAC had not transmitted its own certificate to the UAS in any previous transaction, it SHOULD use a CMS SignedData body for its next request or response.

On future occasions, when the UA receives requests or responses that contain a From header field corresponding to a value in its keyring, the UA SHOULD compare the certificate offered in these messages with the existing certificate in its keyring. If there is a discrepancy, the UA MUST notify its user of a change of the certificate (preferably in terms that indicate that this is a potential security breach) and acquire the user's permission before continuing to process the signaling. If the user authorizes this certificate, it SHOULD be added to the keyring alongside any previous value(s) for this address-of-record.

Note well however, that this key exchange mechanism does not guarantee the secure exchange of keys when self-signed certificates, or certificates signed by an obscure authority, are used - it is vulnerable to well-known attacks. In the opinion of the authors, however, the security it provides is proverbially better than nothing; it is in fact comparable to the widely used SSH application. These limitations are explored in greater detail in Section 26.4.2.

If a UA receives an S/MIME body that has been encrypted with a public key unknown to the recipient, it MUST reject the request with a 493 (Undecipherable) response. This response SHOULD contain a valid certificate for the respondent (corresponding, if possible, to any address of record given in the To header field of the rejected request) within a MIME body with a 'certs-only' "smime-type" parameter.

A 493 (Undecipherable) sent without any certificate indicates that the respondent cannot or will not utilize S/MIME encrypted messages, though they may still support S/MIME signatures.

Note that a user agent that receives a request containing an S/MIME body that is not optional (with a Content-Disposition header "handling" parameter of "required") MUST reject the request with a 415 Unsupported Media Type response if the MIME type is not understood. A user agent that receives such a response when S/MIME is sent SHOULD notify its user that the remote device does not support S/MIME, and it MAY subsequently resend the request without S/MIME, if appropriate; however, this 415 response may constitute a downgrade attack.

If a user agent sends an S/MIME body in a request, but receives a response that contains a MIME body that is not secured, the UAC SHOULD notify its user that the session could not be secured. However, if a user agent that supports S/MIME receives a request with an unsecured body, it SHOULD NOT respond with a secured body, but if it expects S/MIME from the sender (for example, because the sender's From header field value corresponds to an identity on its keychain), the UAS SHOULD notify its user that the session could not be secured.

A number of conditions that arise in the previous text call for the notification of the user when an anomalous certificate-management event occurs. Users might well ask what they should do under these circumstances. First and foremost, an unexpected change in a certificate, or an absence of security when security is expected, are causes for caution but not necessarily indications that an attack is in progress. Users might abort any connection attempt or refuse a connection request they have received; in telephony parlance, they could hang up and call back. Users may wish to find an alternate means to contact the other party and confirm that their key has legitimately changed. Note that users are sometimes compelled to change their certificates, for example when they suspect that the secrecy of their private key has been compromised. When their private key is no longer private, users must legitimately generate a new key and re-establish trust with any users that held their old key.

Finally, if during the course of a dialog a UA receives a certificate in a CMS SignedData message that does not correspond with the certificates previously exchanged during a dialog, the UA MUST notify its user of the change, preferably in terms that indicate that this is a potential security breach.

## 23.3 Securing MIME bodies

There are two types of secure MIME bodies that are of interest to SIP: use of these bodies should follow the S/MIME specification [24] with a few variations.

- o "multipart/signed" MUST be used only with CMS detached signatures.

  This allows backwards compatibility with non-S/MIME-compliant recipients.

- o S/MIME bodies SHOULD have a Content-Disposition header field, and the value of the "handling" parameter SHOULD be "required."

- o If a UAC has no certificate on its keyring associated with the address-of-record to which it wants to send a request, it cannot send an encrypted "application/pkcs7-mime" MIME message. UACs MAY send an initial request such as an OPTIONS message with a CMS detached signature in order to solicit the certificate of the remote side (the signature SHOULD be over a "message/sip" body of the type described in Section 23.4).

  Note that future standardization work on S/MIME may define non-certificate based keys.

- o Senders of S/MIME bodies SHOULD use the "SMIMECapabilities" (see Section 2.5.2 of [24]) attribute to express their capabilities and preferences for further communications. Note especially that senders MAY use the "preferSignedData" capability to encourage receivers to respond with CMS SignedData messages (for example, when sending an OPTIONS request as described above).

- S/MIME implementations MUST at a minimum support SHA1 as a digital signature algorithm, and 3DES as an encryption algorithm. All other signature and encryption algorithms MAY be supported. Implementations can negotiate support for these algorithms with the "SMIMECapabilities" attribute.

- Each S/MIME body in a SIP message SHOULD be signed with only one certificate. If a UA receives a message with multiple signatures, the outermost signature should be treated as the single certificate for this body. Parallel signatures SHOULD NOT be used.

  The following is an example of an encrypted S/MIME SDP body within a SIP message:

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/pkcs7-mime; smime-type=enveloped-data;
     name=smime.p7m
Content-Disposition: attachment; filename=smime.p7m
   handling=required

*******************************************************
* Content-Type: application/sdp                       *
*                                                     *
* v=0                                                 *
* o=alice 53655765 2353687637 IN IP4 pc33.atlanta.com *
* s=-                                                 *
* t=0 0                                               *
* c=IN IP4 pc33.atlanta.com                           *
* m=audio 3456 RTP/AVP 0 1 3 99                       *
* a=rtpmap:0 PCMU/8000                                *
*******************************************************
```

## 23.4 SIP Header Privacy and Integrity using S/MIME: Tunneling SIP

As a means of providing some degree of end-to-end authentication, integrity or confidentiality for SIP header fields, S/MIME can encapsulate entire SIP messages within MIME bodies of type "message/sip" and then apply MIME security to these bodies in the same manner as typical SIP bodies. These encapsulated SIP requests and responses do not constitute a separate dialog or transaction, they are a copy of the "outer" message that is used to verify integrity or to supply additional information.

If a UAS receives a request that contains a tunneled "message/sip" S/MIME body, it SHOULD include a tunneled "message/sip" body in the response with the same smime-type.

Any traditional MIME bodies (such as SDP) SHOULD be attached to the "inner" message so that they can also benefit from S/MIME security. Note that "message/sip" bodies can be sent as a part of a MIME "multipart/mixed" body if any unsecured MIME types should also be transmitted in a request.

### 23.4.1 Integrity and Confidentiality Properties of SIP Headers

When the S/MIME integrity or confidentiality mechanisms are used, there may be discrepancies between the values in the "inner" message and values in the "outer" message. The rules for handling any such differences for all of the header fields described in this document are given in this section.

Note that for the purposes of loose timestamping, all SIP messages that tunnel "message/sip" SHOULD contain a Date header in both the "inner" and "outer" headers.

#### 23.4.1.1 Integrity

Whenever integrity checks are performed, the integrity of a header field should be determined by matching the value of the header field in the signed body with that in the "outer" messages using the comparison rules of SIP as described in 20.

Header fields that can be legitimately modified by proxy servers are: Request-URI, Via, Record-Route, Route, Max-Forwards, and Proxy-Authorization. If these header fields are not intact end-to-end, implementations SHOULD NOT consider this a breach of security. Changes to any other header fields defined in this document constitute an integrity violation; users MUST be notified of a discrepancy.

#### 23.4.1.2 Confidentiality

When messages are encrypted, header fields may be included in the encrypted body that are not present in the "outer" message.

Some header fields must always have a plaintext version because they are required header fields in requests and responses - these include:

To, From, Call-ID, CSeq, Contact. While it is probably not useful to provide an encrypted alternative for the Call-ID, CSeq, or Contact, providing an alternative to the information in the "outer" To or From is permitted. Note that the values in an encrypted body are not used for the purposes of identifying transactions or dialogs - they are merely informational. If the From header field in an encrypted body differs from the value in the "outer" message, the value within the encrypted body SHOULD be displayed to the user, but MUST NOT be used in the "outer" header fields of any future messages.

Primarily, a user agent will want to encrypt header fields that have an end-to-end semantic, including: Subject, Reply-To, Organization, Accept, Accept-Encoding, Accept-Language, Alert-Info, Error-Info, Authentication-Info, Expires, In-Reply-To, Require, Supported, Unsupported, Retry-After, User-Agent, Server, and Warning. If any of these header fields are present in an encrypted body, they should be used instead of any "outer" header fields, whether this entails displaying the header field values to users or setting internal states in the UA. They SHOULD NOT however be used in the "outer" headers of any future messages.

If present, the Date header field MUST always be the same in the "inner" and "outer" headers.

Since MIME bodies are attached to the "inner" message, implementations will usually encrypt MIME-specific header fields, including: MIME-Version, Content-Type, Content-Length, Content-Language, Content-Encoding and Content-Disposition. The "outer" message will have the proper MIME header fields for S/MIME bodies. These header fields (and any MIME bodies they preface) should be treated as normal MIME header fields and bodies received in a SIP message.

It is not particularly useful to encrypt the following header fields: Min-Expires, Timestamp, Authorization, Priority, and WWW-Authenticate. This category also includes those header fields that can be changed by proxy servers (described in the preceding section). UAs SHOULD never include these in an "inner" message if they are not included in the "outer" message. UAs that receive any of these header fields in an encrypted body SHOULD ignore the encrypted values.

Note that extensions to SIP may define additional header fields; the authors of these extensions should describe the integrity and confidentiality properties of such header fields. If a SIP UA encounters an unknown header field with an integrity violation, it MUST ignore the header field.

### 23.4.2 Tunneling Integrity and Authentication

Tunneling SIP messages within S/MIME bodies can provide integrity for SIP header fields if the header fields that the sender wishes to secure are replicated in a "message/sip" MIME body signed with a CMS detached signature.

Provided that the "message/sip" body contains at least the fundamental dialog identifiers (To, From, Call-ID, CSeq), then a signed MIME body can provide limited authentication. At the very least, if the certificate used to sign the body is unknown to the recipient and cannot be verified, the signature can be used to ascertain that a later request in a dialog was transmitted by the same certificate-holder that initiated the dialog. If the recipient of the signed MIME body has some stronger incentive to trust the certificate (they were able to validate it, they acquired it from a trusted repository, or they have used it frequently) then the signature can be taken as a stronger assertion of the identity of the subject of the certificate.

In order to eliminate possible confusions about the addition or subtraction of entire header fields, senders SHOULD replicate all header fields from the request within the signed body. Any message bodies that require integrity protection MUST be attached to the "inner" message.

If a Date header is present in a message with a signed body, the recipient SHOULD compare the header field value with its own internal clock, if applicable. If a significant time discrepancy is detected (on the order of an hour or more), the user agent SHOULD alert the user to the anomaly, and note that it is a potential security breach.

If an integrity violation in a message is detected by its recipient, the message MAY be rejected with a 403 (Forbidden) response if it is a request, or any existing dialog MAY be terminated. UAs SHOULD notify users of this circumstance and request explicit guidance on how to proceed.

The following is an example of the use of a tunneled "message/sip" body:

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Date: Thu, 21 Feb 2002 13:02:03 GMT
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: multipart/signed;
  protocol="application/pkcs7-signature";
  micalg=sha1; boundary=boundary42
Content-Length: 568

--boundary42
Content-Type: message/sip

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <bob@biloxi.com>
From: Alice <alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Date: Thu, 21 Feb 2002 13:02:03 GMT
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

--boundary42
Content-Type: application/pkcs7-signature; name=smime.p7s
Content-Transfer-Encoding: base64
Content-Disposition: attachment; filename=smime.p7s;
   handling=required
```

ghyHhHUujhJhjH77n8HHGTrfvbnj756tbB9HG4VQpfyF467GhIGfHfYT6
4VQpfyF467GhIGfHfYT6jH77n8HHGghyHhHUujhJh756tbB9HGTrfvbnj
n8HHGTrfvhJhjH776tbB9HG4VQbnj7567GhIGfHfYT6ghyHhHUujpfyF4
7GhIGfHfYT64VQbnj756

```
--boundary42-
```

### 23.4.3 Tunneling Encryption

It may also be desirable to use this mechanism to encrypt a "message/sip" MIME body within a CMS EnvelopedData message S/MIME body, but in practice, most header fields are of at least some use to the network; the general use of encryption with S/MIME is to secure message bodies like SDP rather than message headers. Some informational header fields, such as the Subject or Organization could perhaps warrant end-to-end security. Headers defined by future SIP applications might also require obfuscation.

Another possible application of encrypting header fields is selective anonymity. A request could be constructed with a From header field that contains no personal information (for example, sip:anonymous@anonymizer.invalid). However, a second From header field containing the genuine address-of-record of the originator could be encrypted within a "message/sip" MIME body where it will only be visible to the endpoints of a dialog.

> Note that if this mechanism is used for anonymity, the From header field will no longer be usable by the recipient of a message as an index to their certificate keychain for retrieving the proper S/MIME key to associated with the sender. The message must first be decrypted, and the "inner" From header field MUST be used as an index.

In order to provide end-to-end integrity, encrypted "message/sip" MIME bodies SHOULD be signed by the sender. This creates a "multipart/signed" MIME body that contains an encrypted body and a signature, both of type "application/pkcs7-mime".

In the following example, of an encrypted and signed message, the text boxed in asterisks ("*") is encrypted:

```
     INVITE sip:bob@biloxi.com SIP/2.0
     Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
     To: Bob <sip:bob@biloxi.com>
     From: Anonymous <sip:anonymous@atlanta.com>;tag=1928301774
     Call-ID: a84b4c76e66710
     CSeq: 314159 INVITE
     Max-Forwards: 70
     Date: Thu, 21 Feb 2002 13:02:03 GMT
     Contact: <sip:pc33.atlanta.com>
     Content-Type: multipart/signed;
       protocol="application/pkcs7-signature";
       micalg=sha1; boundary=boundary42
     Content-Length: 568

     --boundary42
     Content-Type: application/pkcs7-mime; smime-type=enveloped-data;
          name=smime.p7m
     Content-Transfer-Encoding: base64
     Content-Disposition: attachment; filename=smime.p7m
        handling=required
     Content-Length: 231

   ***********************************************************
   * Content-Type: message/sip                               *
   *                                                         *
   * INVITE sip:bob@biloxi.com SIP/2.0                       *
   * Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 *
   * To: Bob <bob@biloxi.com>                                *
   * From: Alice <alice@atlanta.com>;tag=1928301774          *
   * Call-ID: a84b4c76e66710                                 *
   * CSeq: 314159 INVITE                                     *
   * Max-Forwards: 70                                        *
   * Date: Thu, 21 Feb 2002 13:02:03 GMT                     *
   * Contact: <sip:alice@pc33.atlanta.com>                   *
   *                                                         *
   * Content-Type: application/sdp                           *
   *                                                         *
   * v=0                                                     *
   * o=alice 53655765 2353687637 IN IP4 pc33.atlanta.com     *
   * s=Session SDP                                           *
   * t=0 0                                                   *
   * c=IN IP4 pc33.atlanta.com                               *
   * m=audio 3456 RTP/AVP 0 1 3 99                           *
   * a=rtpmap:0 PCMU/8000                                    *
   ***********************************************************
```

```
--boundary42
Content-Type: application/pkcs7-signature; name=smime.p7s
Content-Transfer-Encoding: base64
Content-Disposition: attachment; filename=smime.p7s;
   handling=required
```


```
ghyHhHUujhJhjH77n8HHGTrfvbnj756tbB9HG4VQpfyF467GhIGfHfYT6
4VQpfyF467GhIGfHfYT6jH77n8HHGghyHhHUujhJh756tbB9HGTrfvbnj
n8HHGTrfvhJhjH776tbB9HG4VQbnj7567GhIGfHfYT6ghyHhHUujpfyF4
7GhIGfHfYT64VQbnj756
```


```
--boundary42-
```

## 24 Examples

In the following examples, we often omit the message body and the corresponding Content-Length and Content-Type header fields for brevity.

### 24.1 Registration

Bob registers on start-up. The message flow is shown in Figure 9. Note that the authentication usually required for registration is not shown for simplicity.

```
biloxi.com         Bob's
 registrar       softphone
    |                |
    |   REGISTER F1  |
    |<---------------|
    |    200 OK F2   |
    |--------------->|
```

Figure 9: SIP Registration Example

F1 REGISTER Bob -> Registrar

```
REGISTER sip:registrar.biloxi.com SIP/2.0
Via: SIP/2.0/UDP bobspc.biloxi.com:5060;branch=z9hG4bKnashds7
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Bob <sip:bob@biloxi.com>;tag=456248
Call-ID: 843817637684230@998sdasdh09
CSeq: 1826 REGISTER
Contact: <sip:bob@192.0.2.4>
Expires: 7200
Content-Length: 0
```

The registration expires after two hours. The registrar responds with a 200 OK:

F2 200 OK Registrar -> Bob

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bobspc.biloxi.com:5060;branch=z9hG4bKnashds7
 ;received=192.0.2.4
To: Bob <sip:bob@biloxi.com>;tag=2493k59kd
From: Bob <sip:bob@biloxi.com>;tag=456248
Call-ID: 843817637684230@998sdasdh09
CSeq: 1826 REGISTER
Contact: <sip:bob@192.0.2.4>
Expires: 7200
Content-Length: 0
```

## 24.2 Session Setup

This example contains the full details of the example session setup in Section 4. The message flow is shown in Figure 1. Note that these flows show the minimum required set of header fields - some other header fields such as Allow and Supported would normally be present.

F1 INVITE Alice -> atlanta.com proxy

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142
```

(Alice’s SDP not shown)

F2 100 Trying atlanta.com proxy -> Alice

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
 ;received=192.0.2.1
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Content-Length: 0
```

F3 INVITE atlanta.com proxy -> biloxi.com proxy

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com;branch=z9hG4bK77ef4c2312983.1
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
 ;received=192.0.2.1
Max-Forwards: 69
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

(Alice’s SDP not shown)
```

F4 100 Trying biloxi.com proxy -> atlanta.com proxy

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com;branch=z9hG4bK77ef4c2312983.1
 ;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
 ;received=192.0.2.1
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Content-Length: 0
```

F5 INVITE biloxi.com proxy -> Bob

```
INVITE sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP server10.biloxi.com;branch=z9hG4bK4b43c2ff8.1
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com;branch=z9hG4bK77ef4c2312983.1
 ;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
 ;received=192.0.2.1
Max-Forwards: 68
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

(Alice’s SDP not shown)
```

F6 180 Ringing Bob -> biloxi.com proxy

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP server10.biloxi.com;branch=z9hG4bK4b43c2ff8.1
 ;received=192.0.2.3
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com;branch=z9hG4bK77ef4c2312983.1
 ;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
 ;received=192.0.2.1
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
Contact: <sip:bob@192.0.2.4>
CSeq: 314159 INVITE
Content-Length: 0
```

F7 180 Ringing biloxi.com proxy -> atlanta.com proxy

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com;branch=z9hG4bK77ef4c2312983.1
 ;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
 ;received=192.0.2.1
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
Contact: <sip:bob@192.0.2.4>
CSeq: 314159 INVITE
Content-Length: 0
```

F8 180 Ringing atlanta.com proxy -> Alice

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
 ;received=192.0.2.1
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
Contact: <sip:bob@192.0.2.4>
CSeq: 314159 INVITE
Content-Length: 0
```

F9 200 OK Bob -> biloxi.com proxy

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP server10.biloxi.com;branch=z9hG4bK4b43c2ff8.1
 ;received=192.0.2.3
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com;branch=z9hG4bK77ef4c2312983.1
 ;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
 ;received=192.0.2.1
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131
```

(Bob’s SDP not shown)

F10 200 OK biloxi.com proxy -> atlanta.com proxy

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com;branch=z9hG4bK77ef4c2312983.1
 ;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
 ;received=192.0.2.1
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131
```

(Bob’s SDP not shown)

F11 200 OK atlanta.com proxy -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
 ;received=192.0.2.1
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131

(Bob's SDP not shown)
```

F12 ACK Alice -> Bob

```
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds9
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0
```

The media session between Alice and Bob is now established.

Bob hangs up first. Note that Bob's SIP phone maintains its own CSeq numbering space, which, in this example, begins with 231. Since Bob is making the request, the To and From URIs and tags have been swapped.

F13 BYE Bob -> Alice

```
BYE sip:alice@pc33.atlanta.com SIP/2.0
Via: SIP/2.0/UDP 192.0.2.4;branch=z9hG4bKnashds10
Max-Forwards: 70
From: Bob <sip:bob@biloxi.com>;tag=a6c85cf
To: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 231 BYE
Content-Length: 0
```

F14 200 OK Alice -> Bob

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.4;branch=z9hG4bKnashds10
From: Bob <sip:bob@biloxi.com>;tag=a6c85cf
To: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 231 BYE
Content-Length: 0
```

The SIP Call Flows document [40] contains further examples of SIP messages.

# 25 Augmented BNF for the SIP Protocol

All of the mechanisms specified in this document are described in both prose and an augmented Backus-Naur Form (BNF) defined in RFC 2234 [10]. Section 6.1 of RFC 2234 defines a set of core rules that are used by this specification, and not repeated here. Implementers need to be familiar with the notation and content of RFC 2234 in order to understand this specification. Certain basic rules are in uppercase, such as SP, LWS, HTAB, CRLF, DIGIT, ALPHA, etc. Angle brackets are used within definitions to clarify the use of rule names.

The use of square brackets is redundant syntactically. It is used as a semantic hint that the specific parameter is optional to use.

## 25.1 Basic Rules

The following rules are used throughout this specification to describe basic parsing constructs. The US-ASCII coded character set is defined by ANSI X3.4-1986.

```
alphanum  =  ALPHA / DIGIT
```

Several rules are incorporated from RFC 2396 [5] but are updated to make them compliant with RFC 2234 [10]. These include:

```
reserved    =  ";" / "/" / "?" / ":" / "@" / "&" / "=" / "+"
               / "$" / ","
unreserved  =  alphanum / mark
mark        =  "-" / "_" / "." / "!" / "~" / "*" / "'"
               / "(" / ")"
escaped     =  "%" HEXDIG HEXDIG
```

SIP header field values can be folded onto multiple lines if the continuation line begins with a space or horizontal tab. All linear white space, including folding, has the same semantics as SP. A recipient MAY replace any linear white space with a single SP before interpreting the field value or forwarding the message downstream. This is intended to behave exactly as HTTP/1.1 as described in RFC 2616 [8]. The SWS construct is used when linear white space is optional, generally between tokens and separators.

```
LWS  =  [*WSP CRLF] 1*WSP ; linear whitespace
SWS  =  [LWS] ; sep whitespace
```

To separate the header name from the rest of value, a colon is used, which, by the above rule, allows whitespace before, but no line break, and whitespace after, including a linebreak. The HCOLON defines this construct.

```
HCOLON  =  *( SP / HTAB ) ":" SWS
```

The TEXT-UTF8 rule is only used for descriptive field contents and values that are not intended to be interpreted by the message parser. Words of *TEXT-UTF8 contain characters from the UTF-8 charset (RFC 2279 [7]). The TEXT-UTF8-TRIM rule is used for descriptive field contents that are n t quoted strings, where leading and trailing LWS is not meaningful. In this regard, SIP differs from HTTP, which uses the ISO 8859-1 character set.

```
TEXT-UTF8-TRIM  =  1*TEXT-UTF8char *(*LWS TEXT-UTF8char)
TEXT-UTF8char   =  %x21-7E / UTF8-NONASCII
UTF8-NONASCII   =  %xC0-DF 1UTF8-CONT
                /  %xE0-EF 2UTF8-CONT
                /  %xF0-F7 3UTF8-CONT
                /  %xF8-Fb 4UTF8-CONT
                /  %xFC-FD 5UTF8-CONT
UTF8-CONT       =  %x80-BF
```

A CRLF is allowed in the definition of TEXT-UTF8-TRIM only as part of a header field continuation. It is expected that the folding LWS will be replaced with a single SP before interpretation of the TEXT-UTF8-TRIM value.

Hexadecimal numeric characters are used in several protocol elements. Some elements (authentication) force hex alphas to be lower case.

```
LHEX  =  DIGIT / %x61-66 ;lowercase a-f
```

Many SIP header field values consist of words separated by LWS or special characters. Unless otherwise stated, tokens are case-insensitive. These special characters MUST be in a quoted string to be used within a parameter value. The word construct is used in Call-ID to allow most separators to be used.

```
token       =  1*(alphanum / "-" / "." / "!" / "%" / "*"
               / "_" / "+" / "`" / "'" / "~" )
separators  =  "(" / ")" / "<" / ">" / "@" /
               "," / ";" / ":" / "\" / DQUOTE /
               "/" / "[" / "]" / "?" / "=" /
               "{" / "}" / SP / HTAB
word        =  1*(alphanum / "-" / "." / "!" / "%" / "*" /
               "_" / "+" / "`" / "'" / "~" /
               "(" / ")" / "<" / ">" /
               ":" / "\" / DQUOTE /
               "/" / "[" / "]" / "?" /
               "{" / "}" )
```

When tokens are used or separators are used between elements, whitespace is often allowed before or after these characters:

```
STAR    =  SWS "*" SWS ; asterisk
SLASH   =  SWS "/" SWS ; slash
EQUAL   =  SWS "=" SWS ; equal
LPAREN  =  SWS "(" SWS ; left parenthesis
RPAREN  =  SWS ")" SWS ; right parenthesis
RAQUOT  =  ">" SWS ; right angle quote
LAQUOT  =  SWS "<"; left angle quote
COMMA   =  SWS "," SWS ; comma
SEMI    =  SWS ";" SWS ; semicolon
COLON   =  SWS ":" SWS ; colon
LDQUOT  =  SWS DQUOTE; open double quotation mark
RDQUOT  =  DQUOTE SWS ; close double quotation mark
```

Comments can be included in some SIP header fields by surrounding the comment text with parentheses. Comments are only allowed in fields containing "comment" as part of their field value definition. In all other fields, parentheses are considered part of the field value.

```
comment  =  LPAREN *(ctext / quoted-pair / comment) RPAREN
ctext    =  %x21-27 / %x2A-5B / %x5D-7E / UTF8-NONASCII
            / LWS
```

ctext includes all chars except left and right parens and backslash. A string of text is parsed as a single word if it is quoted using double-quote marks. In quoted strings, quotation marks (") and backslashes (\) need to be escaped.

```
quoted-string  =  SWS DQUOTE *(qdtext / quoted-pair ) DQUOTE
qdtext         =  LWS / %x21 / %x23-5B / %x5D-7E
                  / UTF8-NONASCII
```

The backslash character ("\") MAY be used as a single-character quoting mechanism only within quoted-string and comment constructs. Unlike HTTP/1.1, the characters CR and LF cannot be escaped by this mechanism to avoid conflict with line folding and header separation.

```
quoted-pair  =  "\" (%x00-09 / %x0B-0C
                / %x0E-7F)

SIP-URI          =  "sip:" [ userinfo ] hostport
                    uri-parameters [ headers ]
SIPS-URI         =  "sips:" [ userinfo ] hostport
                    uri-parameters [ headers ]
userinfo         =  ( user / telephone-subscriber ) [ ":" password ] "@"
user             =  1*( unreserved / escaped / user-unreserved )
user-unreserved  =  "&" / "=" / "+" / "$" / "," / ";" / "?" / "/"
password         =  *( unreserved / escaped /
                    "&" / "=" / "+" / "$" / "," )
hostport         =  host [ ":" port ]
host             =  hostname / IPv4address / IPv6reference
hostname         =  *( domainlabel "." ) toplabel [ "." ]
domainlabel      =  alphanum
                    / alphanum *( alphanum / "-" ) alphanum
toplabel         =  ALPHA / ALPHA *( alphanum / "-" ) alphanum
```

```
IPv4address    =  1*3DIGIT "." 1*3DIGIT "." 1*3DIGIT "." 1*3DIGIT
IPv6reference  =  "[" IPv6address "]"
IPv6address    =  hexpart [ ":" IPv4address ]
hexpart        =  hexseq / hexseq "::" [ hexseq ] / "::" [ hexseq ]
hexseq         =  hex4 *( ":" hex4)
hex4           =  1*4HEXDIG
port           =  1*DIGIT
```

The BNF for telephone-subscriber can be found in RFC 2806 [9]. Note, however, that any characters allowed there that are not allowed in the user part of the SIP URI MUST be escaped.

```
uri-parameters    =  *( ";" uri-parameter)
uri-parameter     =  transport-param / user-param / method-param
                     / ttl-param / maddr-param / lr-param / other-param
transport-param   =  "transport="
                     ( "udp" / "tcp" / "sctp" / "tls"
                     / other-transport)
other-transport   =  token
user-param        =  "user=" ( "phone" / "ip" / other-user)
other-user        =  token
method-param      =  "method=" Method
ttl-param         =  "ttl=" ttl
maddr-param       =  "maddr=" host
lr-param          =  "lr"
other-param       =  pname [ "=" pvalue ]
pname             =  1*paramchar
pvalue            =  1*paramchar
paramchar         =  param-unreserved / unreserved / escaped
param-unreserved  =  "[" / "]" / "/" / ":" / "&" / "+" / "$"

headers         =  "?" header *( "&" header )
header          =  hname "=" hvalue
hname           =  1*( hnv-unreserved / unreserved / escaped )
hvalue          =  *( hnv-unreserved / unreserved / escaped )
hnv-unreserved  =  "[" / "]" / "/" / "?" / ":" / "+" / "$"

SIP-message    =  Request / Response
Request        =  Request-Line
                  *( message-header )
                  CRLF
                  [ message-body ]
Request-Line   =  Method SP Request-URI SP SIP-Version CRLF
Request-URI    =  SIP-URI / SIPS-URI / absoluteURI
absoluteURI    =  scheme ":" ( hier-part / opaque-part )
hier-part      =  ( net-path / abs-path ) [ "?" query ]
net-path       =  "//" authority [ abs-path ]
abs-path       =  "/" path-segments
```

```
opaque-part    =  uric-no-slash *uric
uric           =  reserved / unreserved / escaped
uric-no-slash  =  unreserved / escaped / ";" / "?" / ":" / "@"
                  / "&" / "=" / "+" / "$" / ","
path-segments  =  segment *( "/" segment )
segment        =  *pchar *( ";" param )
param          =  *pchar
pchar          =  unreserved / escaped /
                  ":" / "@" / "&" / "=" / "+" / "$" / ","
scheme         =  ALPHA *( ALPHA / DIGIT / "+" / "-" / "." )
authority      =  srvr / reg-name
srvr           =  [ [ userinfo "@" ] hostport ]
reg-name       =  1*( unreserved / escaped / "$" / ","
                  / ";" / ":" / "@" / "&" / "=" / "+" )
query          =  *uric
SIP-Version    =  "SIP" "/" 1*DIGIT "." 1*DIGIT

message-header  =  (Accept
                /  Accept-Encoding
                /  Accept-Language
                /  Alert-Info
                /  Allow
                /  Authentication-Info
                /  Authorization
                /  Call-ID
                /  Call-Info
                /  Contact
                /  Content-Disposition
                /  Content-Encoding
                /  Content-Language
                /  Content-Length
                /  Content-Type
                /  CSeq
                /  Date
                /  Error-Info
                /  Expires
                /  From
                /  In-Reply-To
                /  Max-Forwards
                /  MIME-Version
                /  Min-Expires
                /  Organization
                /  Priority
                /  Proxy-Authenticate
                /  Proxy-Authorization
                /  Proxy-Require
                /  Record-Route
                /  Reply-To
```

```
                /  Require
                /  Retry-After
                /  Route
                /  Server
                /  Subject
                /  Supported
                /  Timestamp
                /  To
                /  Unsupported
                /  User-Agent
                /  Via
                /  Warning
                /  WWW-Authenticate
                /  extension-header) CRLF

INVITEm           =  %x49.4E.56.49.54.45 ; INVITE in caps
ACKm              =  %x41.43.4B ; ACK in caps
OPTIONSm          =  %x4F.50.54.49.4F.4E.53 ; OPTIONS in caps
BYEm              =  %x42.59.45 ; BYE in caps
CANCELm           =  %x43.41.4E.43.45.4C ; CANCEL in caps
REGISTERm         =  %x52.45.47.49.53.54.45.52 ; REGISTER in caps
Method            =  INVITEm / ACKm / OPTIONSm / BYEm
                     / CANCELm / REGISTERm
                     / extension-method
extension-method  =  token
Response          =  Status-Line
                     *( message-header )
                     CRLF
                     [ message-body ]

Status-Line     =  SIP-Version SP Status-Code SP Reason-Phrase CRLF
Status-Code     =  Informational
               /   Redirection
               /   Success
               /   Client-Error
               /   Server-Error
               /   Global-Failure
               /   extension-code
extension-code  =  3DIGIT
Reason-Phrase   =  *(reserved / unreserved / escaped
                   / UTF8-NONASCII / UTF8-CONT / SP / HTAB)

Informational  =  "100"  ;  Trying
              /   "180"  ;  Ringing
              /   "181"  ;  Call Is Being Forwarded
              /   "182"  ;  Queued
              /   "183"  ;  Session Progress
```

```
Success  =  "200"  ;  OK

Redirection  =  "300"  ;  Multiple Choices
             /   "301"  ;  Moved Permanently
             /   "302"  ;  Moved Temporarily
             /   "305"  ;  Use Proxy
             /   "380"  ;  Alternative Service

Client-Error  =  "400"  ;  Bad Request
              /   "401"  ;  Unauthorized
              /   "402"  ;  Payment Required
              /   "403"  ;  Forbidden
              /   "404"  ;  Not Found
              /   "405"  ;  Method Not Allowed
              /   "406"  ;  Not Acceptable
              /   "407"  ;  Proxy Authentication Required
              /   "408"  ;  Request Timeout
              /   "410"  ;  Gone
              /   "413"  ;  Request Entity Too Large
              /   "414"  ;  Request-URI Too Large
              /   "415"  ;  Unsupported Media Type
              /   "416"  ;  Unsupported URI Scheme
              /   "420"  ;  Bad Extension
              /   "421"  ;  Extension Required
              /   "423"  ;  Interval Too Brief
              /   "480"  ;  Temporarily not available
              /   "481"  ;  Call Leg/Transaction Does Not Exist
              /   "482"  ;  Loop Detected
              /   "483"  ;  Too Many Hops
              /   "484"  ;  Address Incomplete
              /   "485"  ;  Ambiguous
              /   "486"  ;  Busy Here
              /   "487"  ;  Request Terminated
              /   "488"  ;  Not Acceptable Here
              /   "491"  ;  Request Pending
              /   "493"  ;  Undecipherable

Server-Error  =  "500"  ;  Internal Server Error
              /   "501"  ;  Not Implemented
              /   "502"  ;  Bad Gateway
              /   "503"  ;  Service Unavailable
              /   "504"  ;  Server Time-out
              /   "505"  ;  SIP Version not supported
              /   "513"  ;  Message Too Large
```

```
Global-Failure  =  "600"  ;  Busy Everywhere
               /   "603"  ;  Decline
               /   "604"  ;  Does not exist anywhere
               /   "606"  ;  Not Acceptable

Accept         =  "Accept" HCOLON
                   [ accept-range *(COMMA accept-range) ]
accept-range   =  media-range *(SEMI accept-param)
media-range    =  ( "*/*"
                  / ( m-type SLASH "*" )
                  / ( m-type SLASH m-subtype )
                  ) *( SEMI m-parameter )
accept-param   =  ("q" EQUAL qvalue) / generic-param
qvalue         =  ( "0" [ "." 0*3DIGIT ] )
                  / ( "1" [ "." 0*3("0") ] )
generic-param  =  token [ EQUAL gen-value ]
gen-value      =  token / host / quoted-string

Accept-Encoding  =  "Accept-Encoding" HCOLON
                     [ encoding *(COMMA encoding) ]
encoding         =  codings *(SEMI accept-param)
codings          =  content-coding / "*"
content-coding   =  token

Accept-Language  =  "Accept-Language" HCOLON
                     [ language *(COMMA language) ]
language         =  language-range *(SEMI accept-param)
language-range   =  ( ( 1*8ALPHA *( "-" 1*8ALPHA ) ) / "*" )

Alert-Info   =  "Alert-Info" HCOLON alert-param *(COMMA alert-param)
alert-param  =  LAQUOT absoluteURI RAQUOT *( SEMI generic-param )

Allow  =  "Allow" HCOLON [Method *(COMMA Method)]

Authorization     =  "Authorization" HCOLON credentials
credentials       =  ("Digest" LWS digest-response)
                     / other-response
digest-response   =  dig-resp *(COMMA dig-resp)
dig-resp          =  username / realm / nonce / digest-uri
                      / dresponse / algorithm / cnonce
                      / opaque / message-qop
                      / nonce-count / auth-param
username          =  "username" EQUAL username-value
username-value    =  quoted-string
digest-uri        =  "uri" EQUAL LDQUOT digest-uri-value RDQUOT
digest-uri-value  =  rquest-uri ; Equal to request-uri as specified
                     by HTTP/1.1
message-qop       =  "qop" EQUAL qop-value
```

```
cnonce            =  "cnonce" EQUAL cnonce-value
cnonce-value      =  nonce-value
nonce-count       =  "nc" EQUAL nc-value
nc-value          =  8LHEX
dresponse         =  "response" EQUAL request-digest
request-digest    =  LDQUOT 32LHEX RDQUOT
auth-param        =  auth-param-name EQUAL
                     ( token / quoted-string )
auth-param-name   =  token
other-response    =  auth-scheme LWS auth-param
                     *(COMMA auth-param)
auth-scheme       =  token

Authentication-Info  =  "Authentication-Info" HCOLON ainfo
                        *(COMMA ainfo)
ainfo                =  nextnonce / message-qop
                         / response-auth / cnonce
                         / nonce-count
nextnonce            =  "nextnonce" EQUAL nonce-value
response-auth        =  "rspauth" EQUAL response-digest
response-digest      =  LDQUOT *LHEX RDQUOT

Call-ID  =  ( "Call-ID" / "i" ) HCOLON callid
callid   =  word [ "@" word ]

Call-Info   =  "Call-Info" HCOLON info *(COMMA info)
info        =  LAQUOT absoluteURI RAQUOT *( SEMI info-param)
info-param  =  ( "purpose" EQUAL ( "icon" / "info"
               / "card" / token ) ) / generic-param

Contact        =  ("Contact" / "m" ) HCOLON
                  ( STAR / (contact-param *(COMMA contact-param)))
contact-param  =  (name-addr / addr-spec) *(SEMI contact-params)
name-addr      =  [ display-name ] LAQUOT addr-spec RAQUOT
addr-spec      =  SIP-URI / SIPS-URI / absoluteURI
display-name   =  *(token LWS)/ quoted-string

contact-params     =  c-p-q / c-p-expires
                      / contact-extension
c-p-q              =  "q" EQUAL qvalue
c-p-expires        =  "expires" EQUAL delta-seconds
contact-extension  =  generic-param
delta-seconds      =  1*DIGIT

Content-Disposition   =  "Content-Disposition" HCOLON
                         disp-type *( SEMI disp-param )
disp-type             =  "render" / "session" / "icon" / "alert"
                         / disp-extension-token
```

```
disp-param            =  handling-param / generic-param
handling-param        =  "handling" EQUAL
                         ( "optional" / "required"
                         / other-handling )
other-handling        =  token
disp-extension-token  =  token

Content-Encoding  =  ( "Content-Encoding" / "e" ) HCOLON
                     content-coding *(COMMA content-coding)

Content-Language  =  "Content-Language" HCOLON
                     language-tag *(COMMA language-tag)
language-tag      =  primary-tag *( "-" subtag )
primary-tag       =  1*8ALPHA
subtag            =  1*8ALPHA

Content-Length  =  ( "Content-Length" / "l" ) HCOLON 1*DIGIT
Content-Type     =  ( "Content-Type" / "c" ) HCOLON media-type
media-type       =  m-type SLASH m-subtype *(SEMI m-parameter)
m-type           =  discrete-type / composite-type
discrete-type    =  "text" / "image" / "audio" / "video"
                    / "application" / extension-token
composite-type   =  "message" / "multipart" / extension-token
extension-token  =  ietf-token / x-token
ietf-token       =  token
x-token          =  "x-" token
m-subtype        =  extension-token / iana-token
iana-token       =  token
m-parameter      =  m-attribute EQUAL m-value
m-attribute      =  token
m-value          =  token / quoted-string

CSeq  =  "CSeq" HCOLON 1*DIGIT LWS Method

Date          =  "Date" HCOLON SIP-date
SIP-date      =  rfc1123-date
rfc1123-date  =  wkday "," SP date1 SP time SP "GMT"
date1         =  2DIGIT SP month SP 4DIGIT
                 ; day month year (e.g., 02 Jun 1982)
time          =  2DIGIT ":" 2DIGIT ":" 2DIGIT
                 ; 00:00:00 - 23:59:59
wkday         =  "Mon" / "Tue" / "Wed"
                 / "Thu" / "Fri" / "Sat" / "Sun"
month         =  "Jan" / "Feb" / "Mar" / "Apr"
                 / "May" / "Jun" / "Jul" / "Aug"
                 / "Sep" / "Oct" / "Nov" / "Dec"

Error-Info  =  "Error-Info" HCOLON error-uri *(COMMA error-uri)
```

```
error-uri   =  LAQUOT absoluteURI RAQUOT *( SEMI generic-param )

Expires     =  "Expires" HCOLON delta-seconds
From        =  ( "From" / "f" ) HCOLON from-spec
from-spec   =  ( name-addr / addr-spec )
               *( SEMI from-param )
from-param  =  tag-param / generic-param
tag-param   =  "tag" EQUAL token

In-Reply-To  =  "In-Reply-To" HCOLON callid *(COMMA callid)

Max-Forwards  =  "Max-Forwards" HCOLON 1*DIGIT

MIME-Version  =  "MIME-Version" HCOLON 1*DIGIT "." 1*DIGIT

Min-Expires  =  "Min-Expires" HCOLON delta-seconds

Organization  =  "Organization" HCOLON [TEXT-UTF8-TRIM]

Priority        =  "Priority" HCOLON priority-value
priority-value  =  "emergency" / "urgent" / "normal"
                   / "non-urgent" / other-priority
other-priority  =  token

Proxy-Authenticate  =  "Proxy-Authenticate" HCOLON challenge
challenge           =  ("Digest" LWS digest-cln *(COMMA digest-cln))
                       / other-challenge
other-challenge     =  auth-scheme LWS auth-param
                       *(COMMA auth-param)
digest-cln          =  realm / domain / nonce
                        / opaque / stale / algorithm
                        / qop-options / auth-param
realm               =  "realm" EQUAL realm-value
realm-value         =  quoted-string
domain              =  "domain" EQUAL LDQUOT URI
                       *( 1*SP URI ) RDQUOT
URI                 =  absoluteURI / abs-path
nonce               =  "nonce" EQUAL nonce-value
nonce-value         =  quoted-string
opaque              =  "opaque" EQUAL quoted-string
stale               =  "stale" EQUAL ( "true" / "false" )
algorithm           =  "algorithm" EQUAL ( "MD5" / "MD5-sess"
                       / token )
qop-options         =  "qop" EQUAL LDQUOT qop-value
                       *("," qop-value) RDQUOT
qop-value           =  "auth" / "auth-int" / token

Proxy-Authorization  =  "Proxy-Authorization" HCOLON credentials
```

```
Proxy-Require  =  "Proxy-Require" HCOLON option-tag
                  *(COMMA option-tag)
option-tag     =  token

Record-Route  =  "Record-Route" HCOLON rec-route *(COMMA rec-route)
rec-route     =  name-addr *( SEMI rr-param )
rr-param      =  generic-param

Reply-To      =  "Reply-To" HCOLON rplyto-spec
rplyto-spec   =  ( name-addr / addr-spec )
                 *( SEMI rplyto-param )
rplyto-param  =  generic-param
Require       =  "Require" HCOLON option-tag *(COMMA option-tag)

Retry-After  =  "Retry-After" HCOLON delta-seconds
                [ comment ] *( SEMI retry-param )

retry-param  =  ("duration" EQUAL delta-seconds)
                / generic-param

Route        =  "Route" HCOLON route-param *(COMMA route-param)
route-param  =  name-addr *( SEMI rr-param )

Server           =  "Server" HCOLON server-val *(LWS server-val)
server-val       =  product / comment
product          =  token [SLASH product-version]
product-version  =  token

Subject  =  ( "Subject" / "s" ) HCOLON [TEXT-UTF8-TRIM]

Supported  =  ( "Supported" / "k" ) HCOLON
              [option-tag *(COMMA option-tag)]

Timestamp  =  "Timestamp" HCOLON 1*(DIGIT)
               [ "." *(DIGIT) ] [ LWS delay ]
delay      =  *(DIGIT) [ "." *(DIGIT) ]

To        =  ( "To" / "t" ) HCOLON ( name-addr
             / addr-spec ) *( SEMI to-param )
to-param  =  tag-param / generic-param

Unsupported  =  "Unsupported" HCOLON option-tag *(COMMA option-tag)
User-Agent  =  "User-Agent" HCOLON server-val *(LWS server-val)
```

```
Via               =  ( "Via" / "v" ) HCOLON via-parm *(COMMA via-parm)
via-parm          =  sent-protocol LWS sent-by *( SEMI via-params )
via-params        =  via-ttl / via-maddr
                     / via-received / via-branch
                     / via-extension
via-ttl           =  "ttl" EQUAL ttl
via-maddr         =  "maddr" EQUAL host
via-received      =  "received" EQUAL (IPv4address / IPv6address)
via-branch        =  "branch" EQUAL token
via-extension     =  generic-param
sent-protocol     =  protocol-name SLASH protocol-version
                     SLASH transport
protocol-name     =  "SIP" / token
protocol-version  =  token
transport         =  "UDP" / "TCP" / "TLS" / "SCTP"
                     / other-transport
sent-by           =  host [ COLON port ]
ttl               =  1*3DIGIT ; 0 to 255

Warning        =  "Warning" HCOLON warning-value *(COMMA warning-value)
warning-value  =  warn-code SP warn-agent SP warn-text
warn-code      =  3DIGIT
warn-agent     =  hostport / pseudonym
                  ;  the name or pseudonym of the server adding
                  ;  the Warning header, for use in debugging
warn-text      =  quoted-string
pseudonym      =  token

WWW-Authenticate  =  "WWW-Authenticate" HCOLON challenge

extension-header  =  header-name HCOLON header-value
header-name       =  token
header-value      =  *(TEXT-UTF8char / UTF8-CONT / LWS)
message-body  =  *OCTET
```

## 26 Security Considerations: Threat Model and Security Usage Recommendations

SIP is not an easy protocol to secure. Its use of intermediaries, its multi-faceted trust relationships, its expected usage between elements with no trust at all, and its user-to-user operation make security far from trivial. Security solutions are needed that are deployable today, without extensive coordination, in a wide variety of environments and usages. In order to meet these diverse needs, several distinct mechanisms applicable to different aspects and usages of SIP will be required.

Note that the security of SIP signaling itself has no bearing on the security of protocols used in concert with SIP such as RTP, or with the security implications of any specific bodies SIP might carry (although MIME security plays a substantial role in securing SIP). Any media associated with a session can be encrypted end-to-end independently of any associated SIP signaling. Media encryption is outside the scope of this document.

The considerations that follow first examine a set of classic threat models that broadly identify the security needs of SIP. The set of security services required to address these threats is then detailed, followed by an explanation of several security mechanisms that can be used to provide these services. Next, the requirements for implementers of SIP are enumerated, along with exemplary deployments in which these security mechanisms could be used to improve the security of SIP. Some notes on privacy conclude this section.

## 26.1 Attacks and Threat Models

This section details some threats that should be common to most deployments of SIP. These threats have been chosen specifically to illustrate each of the security services that SIP requires.

The following examples by no means provide an exhaustive list of the threats against SIP; rather, these are "classic" threats that demonstrate the need for particular security services that can potentially prevent whole categories of threats.

These attacks assume an environment in which attackers can potentially read any packet on the network - it is anticipated that SIP will frequently be used on the public Internet. Attackers on the network may be able to modify packets (perhaps at some compromised intermediary). Attackers may wish to steal services, eavesdrop on communications, or disrupt sessions.

### 26.1.1 Registration Hijacking

The SIP registration mechanism allows a user agent to identify itself to a registrar as a device at which a user (designated by an address of record) is located. A registrar assesses the identity asserted in the From header field of a REGISTER message to determine whether this request can modify the contact addresses associated with the address-of-record in the To header field. While these two fields are frequently the same, there are many valid deployments in which a third-party may register contacts on a user's behalf.

The From header field of a SIP request, however, can be modified arbitrarily by the owner of a UA, and this opens the door to malicious registrations. An attacker that successfully impersonates a party authorized to change contacts associated with an address-of-record could, for example, de-register all existing contacts for a URI and then register their own device as the appropriate contact address, thereby directing all requests for the affected user to the attacker’s device.

This threat belongs to a family of threats that rely on the absence of cryptographic assurance of a request’s originator. Any SIP UAS that represents a valuable service (a gateway that interworks SIP requests with traditional telephone calls, for example) might want to control access to its resources by authenticating requests that it receives. Even end-user UAs, for example SIP phones, have an interest in ascertaining the identities of originators of requests.

This threat demonstrates the need for security services that enable SIP entities to authenticate the originators of requests.

### 26.1.2 Impersonating a Server

The domain to which a request is destined is generally specified in the Request-URI. UAs commonly contact a server in this domain directly in order to deliver a request. However, there is always a possibility that an attacker could impersonate the remote server, and that the UA’s request could be intercepted by some other party.

For example, consider a case in which a redirect server at one domain, chicago.com, impersonates a redirect server at another domain, biloxi.com. A user agent sends a request to biloxi.com, but the redirect server at chicago.com answers with a forged response that has appropriate SIP header fields for a response from biloxi.com. The forged contact addresses in the redirection response could direct the originating UA to inappropriate or insecure resources, or simply prevent requests for biloxi.com from succeeding.

This family of threats has a vast membership, many of which are critical. As a converse to the registration hijacking threat, consider the case in which a registration sent to biloxi.com is intercepted by chicago.com, which replies to the intercepted registration with a forged 301 (Moved Permanently) response. This response might seem to come from biloxi.com yet designate chicago.com as the appropriate registrar. All future REGISTER requests from the originating UA would then go to chicago.com.

Prevention of this threat requires a means by which UAs can authenticate the servers to whom they send requests.

### 26.1.3 Tampering with Message Bodies

As a matter of course, SIP UAs route requests through trusted proxy servers. Regardless of how that trust is established (authentication of proxies is discussed elsewhere in this section), a UA may trust a proxy server to route a request, but not to inspect or possibly modify the bodies contained in that request.

Consider a UA that is using SIP message bodies to communicate session encryption keys for a media session. Although it trusts the proxy server of the domain it is contacting to deliver signaling properly, it may not want the administrators of that domain to be capable of decrypting any subsequent media session. Worse yet, if the proxy server were actively malicious, it could modify the session key, either acting as a man-in-the-middle, or perhaps changing the security characteristics requested by the originating UA.

This family of threats applies not only to session keys, but to most conceivable forms of content carried end-to-end in SIP. These might include MIME bodies that should be rendered to the user, SDP, or encapsulated telephony signals, among others. Attackers might attempt to modify SDP bodies, for example, in order to point RTP media streams to a wiretapping device in order to eavesdrop on subsequent voice communications.

Also note that some header fields in SIP are meaningful end-to-end, for example, Subject. UAs might be protective of these header fields as well as bodies (a malicious intermediary changing the Subject header field might make an important request appear to be spam, for example). However, since many header fields are legitimately inspected or altered by proxy servers as a request is routed, not all header fields should be secured end-to-end.

For these reasons, the UA might want to secure SIP message bodies, and in some limited cases header fields, end-to-end. The security services required for bodies include confidentiality, integrity, and authentication. These end-to-end services should be independent of the means used to secure interactions with intermediaries such as proxy servers.

### 26.1.4 Tearing Down Sessions

Once a dialog has been established by initial messaging, subsequent requests can be sent that modify the state of the dialog and/or session. It is critical that principals in a session can be certain that such requests are not forged by attackers.

Consider a case in which a third-party attacker captures some initial messages in a dialog shared by two parties in order to learn the parameters of the session (To tag, From tag, and so forth) and then inserts a BYE request into the session. The attacker could opt to forge the request such that it seemed to come from either participant. Once the BYE is received by its target, the session will be torn down prematurely.

Similar mid-session threats include the transmission of forged re-INVITEs that alter the session (possibly to reduce session security or redirect media streams as part of a wiretapping attack).

The most effective countermeasure to this threat is the authentication of the sender of the BYE. In this instance, the recipient needs only know that the BYE came from the same party with whom the corresponding dialog was established (as opposed to ascertaining the absolute identity of the sender). Also, if the attacker is unable to learn the parameters of the session due to confidentiality, it would not be possible to forge the BYE. However, some intermediaries (like proxy servers) will need to inspect those parameters as the session is established.

### 26.1.5 Denial of Service and Amplification

Denial-of-service attacks focus on rendering a particular network element unavailable, usually by directing an excessive amount of network traffic at its interfaces. A distributed denial-of-service attack allows one network user to cause multiple network hosts to flood a target host with a large amount of network traffic.

In many architectures, SIP proxy servers face the public Internet in order to accept requests from worldwide IP endpoints. SIP creates a number of potential opportunities for distributed denial-of-service attacks that must be recognized and addressed by the implementers and operators of SIP systems.

Attackers can create bogus requests that contain a falsified source IP address and a corresponding Via header field that identify a targeted host as the originator of the request and then send this request to a large number of SIP network elements, thereby using hapless SIP UAs or proxies to generate denial-of-service traffic aimed at the target.

Similarly, attackers might use falsified Route header field values in a request that identify the target host and then send such messages to forking proxies that will amplify messaging sent to the target.

Record-Route could be used to similar effect when the attacker is certain that the SIP dialog initiated by the request will result in numerous transactions originating in the backwards direction.

A number of denial-of-service attacks open up if REGISTER requests are not properly authenticated and authorized by registrars. Attackers could de-register some or all users in an administrative domain, thereby preventing these users from being invited to new sessions. An attacker could also register a large number of contacts designating the same host for a given address-of-record in order to use the registrar and any associated proxy servers as amplifiers in a denial-of-service attack. Attackers might also attempt to deplete available memory and disk resources of a registrar by registering huge numbers of bindings.

The use of multicast to transmit SIP requests can greatly increase the potential for denial-of-service attacks.

These problems demonstrate a general need to define architectures that minimize the risks of denial-of-service, and the need to be mindful in recommendations for security mechanisms of this class of attacks.

## 26.2 Security Mechanisms

From the threats described above, we gather that the fundamental security services required for the SIP protocol are: preserving the confidentiality and integrity of messaging, preventing replay attacks or message spoofing, providing for the authentication and privacy of the participants in a session, and preventing denial-of-service attacks. Bodies within SIP messages separately require the security services of confidentiality, integrity, and authentication.

Rather than defining new security mechanisms specific to SIP, SIP reuses wherever possible existing security models derived from the HTTP and SMTP space.

Full encryption of messages provides the best means to preserve the confidentiality of signaling - it can also guarantee that messages are not modified by any malicious intermediaries. However, SIP requests and responses cannot be naively encrypted end-to-end in their entirety because message fields such as the Request-URI, Route, and Via need to be visible to proxies in most network architectures so that SIP requests are routed correctly. Note that proxy servers need to modify some features of messages as well (such as adding Via header field values) in order for SIP to function. Proxy servers must therefore be trusted, to some degree, by SIP UAs. To this purpose, low-layer security mechanisms for SIP are recommended, which encrypt the entire SIP requests or responses on the wire on a hop-by-hop basis, and that allow endpoints to verify the identity of proxy servers to whom they send requests.

SIP entities also have a need to identify one another in a secure fashion. When a SIP endpoint asserts the identity of its user to a peer UA or to a proxy server, that identity should in some way be verifiable. A cryptographic authentication mechanism is provided in SIP to address this requirement.

An independent security mechanism for SIP message bodies supplies an alternative means of end-to-end mutual authentication, as well as providing a limit on the degree to which user agents must trust intermediaries.

### 26.2.1 Transport and Network Layer Security

Transport or network layer security encrypts signaling traffic, guaranteeing message confidentiality and integrity.

Oftentimes, certificates are used in the establishment of lower-layer security, and these certificates can also be used to provide a means of authentication in many architectures.

Two popular alternatives for providing security at the transport and network layer are, respectively, TLS [25] and IPSec [26].

IPSec is a set of network-layer protocol tools that collectively can be used as a secure replacement for traditional IP (Internet Protocol). IPSec is most commonly used in architectures in which a set of hosts or administrative domains have an existing trust relationship with one another. IPSec is usually implemented at the operating system level in a host, or on a security gateway that provides confidentiality and integrity for all traffic it receives from a particular interface (as in a VPN architecture). IPSec can also be used on a hop-by-hop basis.

In many architectures IPSec does not require integration with SIP applications; IPSec is perhaps best suited to deployments in which adding security directly to SIP hosts would be arduous. UAs that have a pre-shared keying relationship with their first-hop proxy server are also good candidates to use IPSec. Any deployment of IPSec for SIP would require an IPSec profile describing the protocol tools that would be required to secure SIP. No such profile is given in this document.

TLS provides transport-layer security over connection-oriented protocols (for the purposes of this document, TCP); "tls" (signifying TLS over TCP) can be specified as the desired transport protocol within a Via header field value or a SIP-URI. TLS is most suited to architectures in which hop-by-hop security is required between hosts with no pre-existing trust association. For example, Alice trusts her local proxy server, which after a certificate exchange decides to trust Bob's local proxy server, which Bob trusts, hence Bob and Alice can communicate securely.

TLS must be tightly coupled with a SIP application. Note that transport mechanisms are specified on a hop-by-hop basis in SIP, thus a UA that sends requests over TLS to a proxy server has no assurance that TLS will be used end-to-end.

The TLS_RSA_WITH_AES_128_CBC_SHA ciphersuite [6] MUST be supported at a minimum by implementers when TLS is used in a SIP application. For purposes of backwards compatibility, proxy servers, redirect servers, and registrars SHOULD support TLS_RSA_WITH_3DES_EDE_CBC_SHA. Implementers MAY also support any other ciphersuite.

### 26.2.2 SIPS URI Scheme

The SIPS URI scheme adheres to the syntax of the SIP URI (described in 19), although the scheme string is "sips" rather than "sip". The semantics of SIPS are very different from the SIP URI, however. SIPS allows resources to specify that they should be reached securely.

A SIPS URI can be used as an address-of-record for a particular user - the URI by which the user is canonically known (on their business cards, in the From header field of their requests, in the To header field of REGISTER requests). When used as the Request-URI of a request, the SIPS scheme signifies that each hop over which the request is forwarded, until the request reaches the SIP entity responsible for the domain portion of the Request-URI, must be secured with TLS; once it reaches the domain in question it is handled in accordance with local security and routing policy, quite possibly using TLS for any last hop to a UAS. When used by the originator of a request (as would be the case if they employed a SIPS URI as the address-of-record of the target), SIPS dictates that the entire request path to the target domain be so secured.

The SIPS scheme is applicable to many of the other ways in which SIP URIs are used in SIP today in addition to the Request-URI, including in addresses-of-record, contact addresses (the contents of Contact headers, including those of REGISTER methods), and Route headers. In each instance, the SIPS URI scheme allows these existing fields to designate secure resources. The manner in which a SIPS URI is dereferenced in any of these contexts has its own security properties which are detailed in [4].

The use of SIPS in particular entails that mutual TLS authentication SHOULD be employed, as SHOULD the ciphersuite TLS_RSA_WITH_AES_128_CBC_SHA. Certificates received in the authentication process SHOULD be validated with root certificates held by the client; failure to validate a certificate SHOULD result in the failure of the request.

> Note that in the SIPS URI scheme, transport is independent of TLS, and thus "sips:alice@atlanta.com;transport=tcp" and "sips:alice@atlanta.com;transport=sctp" are both valid (although note that UDP is not a valid transport for SIPS). The use of "transport=tls" has consequently been deprecated, partly because it was specific to a single hop of the request. This is a change since RFC 2543.

Users that distribute a SIPS URI as an address-of-record may elect to operate devices that refuse requests over insecure transports.

### 26.2.3 HTTP Authentication

SIP provides a challenge capability, based on HTTP authentication, that relies on the 401 and 407 response codes as well as header fields for carrying challenges and credentials. Without significant modification, the reuse of the HTTP Digest authentication scheme in SIP allows for replay protection and one-way authentication.

The usage of Digest authentication in SIP is detailed in Section 22.

### 26.2.4 S/MIME

As is discussed above, encrypting entire SIP messages end-to-end for the purpose of confidentiality is not appropriate because network intermediaries (like proxy servers) need to view certain header fields in order to route messages correctly, and if these intermediaries are excluded from security associations, then SIP messages will essentially be non-routable.

However, S/MIME allows SIP UAs to encrypt MIME bodies within SIP, securing these bodies end-to-end without affecting message headers. S/MIME can provide end-to-end confidentiality and integrity for message bodies, as well as mutual authentication. It is also possible to use S/MIME to provide a form of integrity and confidentiality for SIP header fields through SIP message tunneling.

The usage of S/MIME in SIP is detailed in Section 23.

## 26.3 Implementing Security Mechanisms

### 26.3.1 Requirements for Implementers of SIP

Proxy servers, redirect servers, and registrars MUST implement TLS, and MUST support both mutual and one-way authentication. It is strongly RECOMMENDED that UAs be capable initiating TLS; UAs MAY also be capable of acting as a TLS server. Proxy servers, redirect servers, and registrars SHOULD possess a site certificate whose subject corresponds to their canonical hostname. UAs MAY have certificates of their own for mutual authentication with TLS, but no provisions are set forth in this document for their use. All SIP elements that support TLS MUST have a mechanism for validating certificates received during TLS negotiation; this entails possession of one or more root certificates issued by certificate authorities (preferably well-known distributors of site certificates comparable to those that issue root certificates for web browsers).

All SIP elements that support TLS MUST also support the SIPS URI scheme.

Proxy servers, redirect servers, registrars, and UAs MAY also implement IPSec or other lower-layer security protocols.

When a UA attempts to contact a proxy server, redirect server, or registrar, the UAC SHOULD initiate a TLS connection over which it will send SIP messages. In some architectures, UASs MAY receive requests over such TLS connections as well.

Proxy servers, redirect servers, registrars, and UAs MUST implement Digest Authorization, encompassing all of the aspects required in 22. Proxy servers, redirect servers, and registrars SHOULD be configured with at least one Digest realm, and at least one "realm" string supported by a given server SHOULD correspond to the server’s hostname or domainname.

UAs MAY support the signing and encrypting of MIME bodies, and transference of credentials with S/MIME as described in Section 23. If a UA holds one or more root certificates of certificate authorities in order to validate certificates for TLS or IPSec, it SHOULD be capable of reusing these to verify S/MIME certificates, as appropriate. A UA MAY hold root certificates specifically for validating S/MIME certificates.

Note that is it anticipated that future security extensions may upgrade the normative strength associated with S/MIME as S/MIME implementations appear and the problem space becomes better understood.

### 26.3.2 Security Solutions

The operation of these security mechanisms in concert can follow the existing web and email security models to some degree. At a high level, UAs authenticate themselves to servers (proxy servers, redirect servers, and registrars) with a Digest username and password; servers authenticate themselves to UAs one hop away, or to another server one hop away (and vice versa), with a site certificate delivered by TLS.

On a peer-to-peer level, UAs trust the network to authenticate one another ordinarily; however, S/MIME can also be used to provide direct authentication when the network does not, or if the network itself is not trusted.

The following is an illustrative example in which these security mechanisms are used by various UAs and servers to prevent the sorts of threats described in Section 26.1. While implementers and network administrators MAY follow the normative guidelines given in the remainder of this section, these are provided only as example implementations.

#### 26.3.2.1 Registration

When a UA comes online and registers with its local administrative domain, it SHOULD establish a TLS connection with its registrar (Section 10 describes how the UA reaches its registrar). The registrar SHOULD offer a certificate to the UA, and the site identified by the certificate MUST correspond with the domain in which the UA intends to register; for example, if the UA intends to register the address-of-record ’alice@atlanta.com’, the site certificate must identify a host within the atlanta.com domain (such as sip.atlanta.com). When it receives the TLS Certificate message, the UA SHOULD verify the certificate and inspect the site identified by the certificate. If the certificate is invalid, revoked, or if it does not identify the appropriate party, the UA MUST NOT send the REGISTER message and otherwise proceed with the registration.

When a valid certificate has been provided by the registrar, the UA knows that the registrar is not an attacker who might redirect the UA, steal passwords, or attempt any similar attacks.

The UA then creates a REGISTER request that SHOULD be addressed to a Request-URI corresponding to the site certificate received from the registrar. When the UA sends the REGISTER request over the existing TLS connection, the registrar SHOULD challenge the request with a 401 (Proxy Authentication Required) response. The "realm" parameter within the Proxy-Authenticate header field of the response SHOULD correspond to the domain previously given by the site certificate. When the UAC receives the challenge, it SHOULD either prompt the user for credentials or take an appropriate credential from a keyring corresponding to the "realm" parameter in the challenge. The username of this credential SHOULD correspond with the "userinfo" portion of the URI in the To header field of the REGISTER request. Once the Digest credentials have been inserted into an appropriate Proxy-Authorization header field, the REGISTER should be resubmitted to the registrar.

> Since the registrar requires the user agent to authenticate itself, it would be difficult for an attacker to forge REGISTER requests for the user's address-of-record. Also note that since the REGISTER is sent over a confidential TLS connection, attackers will not be able to intercept the REGISTER to record credentials for any possible replay attack.

Once the registration has been accepted by the registrar, the UA SHOULD leave this TLS connection open provided that the registrar also acts as the proxy server to which requests are sent for users in this administrative domain. The existing TLS connection will be reused to deliver incoming requests to the UA that has just completed registration.

> Because the UA has already authenticated the server on the other side of the TLS connection, all requests that come over this connection are known to have passed through the proxy server - attackers cannot create spoofed requests that appear to have been sent through that proxy server.

#### 26.3.2.2 Interdomain Requests

Now let's say that Alice's UA would like to initiate a session with a user in a remote administrative domain, namely "bob@biloxi.com". We will also say that the local administrative domain (atlanta.com) has a local outbound proxy.

The proxy server that handles inbound requests for an administrative domain MAY also act as a local outbound proxy; for simplicity's sake we'll assume this to be the case for atlanta.com (otherwise the user agent would initiate a new TLS connection to a separate server at this point). Assuming that the client has completed the registration process described in the preceding section, it SHOULD reuse the TLS connection to the local proxy server when it sends an INVITE request to another user. The UA SHOULD reuse cached credentials in the INVITE to avoid prompting the user unnecessarily.

When the local outbound proxy server has validated the credentials presented by the UA in the INVITE, it SHOULD inspect the Request-URI to determine how the message should be routed (see [4]). If the "domainname" portion of the Request-URI had corresponded to the local domain (atlanta.com) rather than biloxi.com, then the proxy server would have consulted its location service to determine how best to reach the requested user.

> Had "alice@atlanta.com" been attempting to contact, say, "alex@atlanta.com", the local proxy would have proxied to the request to the TLS connection Alex had established with the registrar when he registered. Since Alex would receive this request over his authenticated channel, he would be assured that Alice’s request had been authorized by the proxy server of the local administrative domain.

However, in this instance the Request-URI designates a remote domain. The local outbound proxy server at atlanta.com SHOULD therefore establish a TLS connection with the remote proxy server at biloxi.com. Since both of the participants in this TLS connection are servers that possess site certificates, mutual TLS authentication SHOULD occur. Each side of the connection SHOULD verify and inspect the certificate of the other, noting the domain name that appears in the certificate for comparison with the header fields of SIP messages. The atlanta.com proxy server, for example, SHOULD verify at this stage that the certificate received from the remote side corresponds with the biloxi.com domain. Once it has done so, and TLS negotiation has completed, resulting in a secure channel between the two proxies, the atlanta.com proxy can forward the INVITE request to biloxi.com.

The proxy server at biloxi.com SHOULD inspect the certificate of the proxy server at atlanta.com in turn and compare the domain asserted by the certificate with the "domainname" portion of the From header field in the INVITE request. The biloxi proxy MAY have a strict security policy that requires it to reject requests that do not match the administrative domain from which they have been proxied.

> Such security policies could be instituted to prevent the SIP equivalent of SMTP ’open relays’ that are frequently exploited to generate spam.

This policy, however, only guarantees that the request came from the domain it ascribes to itself; it does not allow biloxi.com to ascertain how atlanta.com authenticated Alice. Only if biloxi.com has some other way of knowing atlanta.com's authentication policies could it possibly ascertain how Alice proved her identity. biloxi.com might then institute an even stricter policy that forbids requests that come from domains that are not known administratively to share a common authentication policy with biloxi.com.

Once the INVITE has been approved by the biloxi proxy, the proxy server SHOULD identify the existing TLS channel, if any, associated with the user targeted by this request (in this case "bob@biloxi.com"). The INVITE should be proxied through this channel to Bob. Since the request is received over a TLS connection that had previously been authenticated as the biloxi proxy, Bob knows that the From header field was not tampered with and that atlanta.com has validated Alice, although not necessarily whether or not to trust Alice's identity.

Before they forward the request, both proxy servers SHOULD add a Record-Route header field to the request so that all future requests in this dialog will pass through the proxy servers. The proxy servers can thereby continue to provide security services for the lifetime of this dialog. If the proxy servers do not add themselves to the Record-Route, future messages will pass directly end-to-end between Alice and Bob without any security services (unless the two parties agree on some independent end-to-end security such as S/MIME). In this respect the SIP trapezoid model can provide a nice structure where conventions of agreement between the site proxies can provide a reasonably secure channel between Alice and Bob.

> An attacker preying on this architecture would, for example, be unable to forge a BYE request and insert it into the signaling stream between Bob and Alice because the attacker has no way of ascertaining the parameters of the session and also because the integrity mechanism transitively protects the traffic between Alice and Bob.

#### 26.3.2.3 Peer-to-Peer Requests

Alternatively, consider a UA asserting the identity "carol@chicago.com" that has no local outbound proxy. When Carol wishes to send an INVITE to "bob@biloxi.com", her UA SHOULD initiate a TLS connection with the biloxi proxy directly (using the mechanism described in [4] to determine how to best to reach the given Request-URI). When her UA receives a certificate from the biloxi proxy, it SHOULD be verified normally before she passes her INVITE across the TLS connection. However, Carol has no means of proving her identity to the biloxi proxy, but she does have a CMS-detached signature over a "message/sip" body in the INVITE. It is unlikely in this instance that Carol would have any credentials in the biloxi.com realm, since she has no formal association with biloxi.com. The biloxi proxy MAY also have a strict policy that precludes it from even bothering to challenge requests that do not have biloxi.com in the "domainname" portion of the From header field - it treats these users as unauthenticated.

The biloxi proxy has a policy for Bob that all non-authenticated requests should be redirected to the appropriate contact address registered against 'bob@biloxi.com', namely <sip:bob@192.0.2.4>. Carol receives the redirection response over the TLS connection she established with the biloxi proxy, so she trusts the veracity of the contact address.

Carol SHOULD then establish a TCP connection with the designated address and send a new INVITE with a Request-URI containing the received contact address (recomputing the signature in the body as the request is readied). Bob receives this INVITE on an insecure interface, but his UA inspects and, in this instance, recognizes the From header field of the request and subsequently matches a locally cached certificate with the one presented in the signature of the body of the INVITE. He replies in similar fashion, authenticating himself to Carol, and a secure dialog begins.

> Sometimes firewalls or NATs in an administrative domain could preclude the establishment of a direct TCP connection to a UA. In these cases, proxy servers could also potentially relay requests to UAs in a way that has no trust implications (for example, forgoing an existing TLS connection and forwarding the request over cleartext TCP) as local policy dictates.

#### 26.3.2.4 DoS Protection

In order to minimize the risk of a denial-of-service attack against architectures using these security solutions, implementers should take note of the following guidelines.

When the host on which a SIP proxy server is operating is routable from the public Internet, it SHOULD be deployed in an administrative domain with defensive operational policies (blocking source-routed traffic, preferably filtering ping traffic). Both TLS and IPSec can also make use of bastion hosts at the edges of administrative domains that participate in the security associations to aggregate secure tunnels and sockets. These bastion hosts can also take the brunt of denial-of-service attacks, ensuring that SIP hosts within the administrative domain are not encumbered with superfluous messaging.

No matter what security solutions are deployed, floods of messages directed at proxy servers can lock up proxy server resources and prevent desirable traffic from reaching its destination. There is a computational expense associated with processing a SIP transaction at a proxy server, and that expense is greater for stateful proxy servers than it is for stateless proxy servers. Therefore, stateful proxies are more susceptible to flooding than stateless proxy servers.

UAs and proxy servers SHOULD challenge questionable requests with only a single 401 (Unauthorized) or 407 (Proxy Authentication Required), forgoing the normal response retransmission algorithm, and thus behaving statelessly towards unauthenticated requests.

> Retransmitting the 401 (Unauthorized) or 407 (Proxy Authentication Required) status response amplifies the problem of an attacker using a falsified header field value (such as Via) to direct traffic to a third party.

In summary, the mutual authentication of proxy servers through mechanisms such as TLS significantly reduces the potential for rogue intermediaries to introduce falsified requests or responses that can deny service. This commensurately makes it harder for attackers to make innocent SIP nodes into agents of amplification.

## 26.4 Limitations

Although these security mechanisms, when applied in a judicious manner, can thwart many threats, there are limitations in the scope of the mechanisms that must be understood by implementers and network operators.

### 26.4.1 HTTP Digest

One of the primary limitations of using HTTP Digest in SIP is that the integrity mechanisms in Digest do not work very well for SIP. Specifically, they offer protection of the Request-URI and the method of a message, but not for any of the header fields that UAs would most likely wish to secure.

The existing replay protection mechanisms described in RFC 2617 also have some limitations for SIP. The next-nonce mechanism, for example, does not support pipelined requests. The nonce-count mechanism should be used for replay protection.

Another limitation of HTTP Digest is the scope of realms. Digest is valuable when a user wants to authenticate themselves to a resource with which they have a pre-existing association, like a service provider of which the user is a customer (which is quite a common scenario and thus Digest provides an extremely useful function). By way of contrast, the scope of TLS is interdomain or multirealm, since certificates are often globally verifiable, so that the UA can authenticate the server with no pre-existing association.

### 26.4.2 S/MIME

The largest outstanding defect with the S/MIME mechanism is the lack of a prevalent public key infrastructure for end users. If self-signed certificates (or certificates that cannot be verified by one of the participants in a dialog) are used, the SIP-based key exchange mechanism described in Section 23.2 is susceptible to a man-in-the-middle attack with which an attacker can potentially inspect and modify S/MIME bodies. The attacker needs to intercept the first exchange of keys between the two parties in a dialog, remove the existing CMS-detached signatures from the request and response, and insert a different CMS-detached signature containing a certificate supplied by the attacker (but which seems to be a certificate for the proper address-of-record). Each party will think they have exchanged keys with the other, when in fact each has the public key of the attacker.

It is important to note that the attacker can only leverage this vulnerability on the first exchange of keys between two parties - on subsequent occasions, the alteration of the key would be noticeable to the UAs. It would also be difficult for the attacker to remain in the path of all future dialogs between the two parties over time (as potentially days, weeks, or years pass).

SSH is susceptible to the same man-in-the-middle attack on the first exchange of keys; however, it is widely acknowledged that while SSH is not perfect, it does improve the security of connections. The use of key fingerprints could provide some assistance to SIP, just as it does for SSH. For example, if two parties use SIP to establish a voice communications session, each could read off the fingerprint of the key they received from the other, which could be compared against the original. It would certainly be more difficult for the man-in-the-middle to emulate the voices of the participants than their signaling (a practice that was used with the Clipper chip-based secure telephone).

The S/MIME mechanism allows UAs to send encrypted requests without preamble if they possess a certificate for the destination address-of-record on their keyring. However, it is possible that any particular device registered for an address-of-record will not hold the certificate that has been previously employed by the device's current user, and that it will therefore be unable to process an encrypted request properly, which could lead to some avoidable error signaling. This is especially likely when an encrypted request is forked.

The keys associated with S/MIME are most useful when associated with a particular user (an address-of-record) rather than a device (a UA). When users move between devices, it may be difficult to transport private keys securely between UAs; how such keys might be acquired by a device is outside the scope of this document.

Another, more prosaic difficulty with the S/MIME mechanism is that it can result in very large messages, especially when the SIP tunneling mechanism described in Section 23.4 is used. For that reason, it is RECOMMENDED that TCP should be used as a transport protocol when S/MIME tunneling is employed.

#### 26.4.3 TLS

The most commonly voiced concern about TLS is that it cannot run over UDP; TLS requires a connection-oriented underlying transport protocol, which for the purposes of this document means TCP.

It may also be arduous for a local outbound proxy server and/or registrar to maintain many simultaneous long-lived TLS connections with numerous UAs. This introduces some valid scalability concerns, especially for intensive ciphersuites. Maintaining redundancy of long-lived TLS connections, especially when a UA is solely responsible for their establishment, could also be cumbersome.

TLS only allows SIP entities to authenticate servers to which they are adjacent; TLS offers strictly hop-by-hop security. Neither TLS, nor any other mechanism specified in this document, allows clients to authenticate proxy servers to whom they cannot form a direct TCP connection.

#### 26.4.4 SIPS URIs

Actually using TLS on every segment of a request path entails that the terminating UAS must be reachable over TLS (perhaps registering with a SIPS URI as a contact address). This is the preferred use of SIPS. Many valid architectures, however, use TLS to secure part of the request path, but rely on some other mechanism for the final hop to a UAS, for example. Thus SIPS cannot guarantee that TLS usage will be truly end-to-end. Note that since many UAs will not accept incoming TLS connections, even those UAs that do support TLS may be required to maintain persistent TLS connections as described in the TLS limitations section above in order to receive requests over TLS as a UAS.

Location services are not required to provide a SIPS binding for a SIPS Request-URI. Although location services are commonly populated by user registrations (as described in Section 10.2.1), various other protocols and interfaces could conceivably supply contact addresses for an AOR, and these tools are free to map SIPS URIs to SIP URIs as appropriate. When queried for bindings, a location service returns its contact addresses without regard for whether it received a request with a SIPS Request-URI. If a redirect server is accessing the location service, it is up to the entity that processes the Contact header field of a redirection to determine the propriety of the contact addresses.

Ensuring that TLS will be used for all of the request segments up to the target domain is somewhat complex. It is possible that cryptographically authenticated proxy servers along the way that are non-compliant or compromised may choose to disregard the forwarding rules associated with SIPS (and the general forwarding rules in Section 16.6). Such malicious intermediaries could, for example, retarget a request from a SIPS URI to a SIP URI in an attempt to downgrade security.

Alternatively, an intermediary might legitimately retarget a request from a SIP to a SIPS URI. Recipients of a request whose Request-URI uses the SIPS URI scheme thus cannot assume on the basis of the Request-URI alone that SIPS was used for the entire request path (from the client onwards).

To address these concerns, it is RECOMMENDED that recipients of a request whose Request-URI contains a SIP or SIPS URI inspect the To header field value to see if it contains a SIPS URI (though note that it does not constitute a breach of security if this URI has the same scheme but is not equivalent to the URI in the To header field). Although clients may choose to populate the Request-URI and To header field of a request differently, when SIPS is used this disparity could be interpreted as a possible security violation, and the request could consequently be rejected by its recipient. Recipients MAY also inspect the Via header chain in order to double-check whether or not TLS was used for the entire request path until the local administrative domain was reached. S/MIME may also be used by the originating UAC to help ensure that the original form of the To header field is carried end-to-end.

If the UAS has reason to believe that the scheme of the Request-URI has been improperly modified in transit, the UA SHOULD notify its user of a potential security breach.

As a further measure to prevent downgrade attacks, entities that accept only SIPS requests MAY also refuse connections on insecure ports.

End users will undoubtedly discern the difference between SIPS and SIP URIs, and they may manually edit them in response to stimuli. This can either benefit or degrade security. For example, if an attacker corrupts a DNS cache, inserting a fake record set that effectively removes all SIPS records for a proxy server, then any SIPS requests that traverse this proxy server may fail. When a user, however, sees that repeated calls to a SIPS AOR are failing, they could on some devices manually convert the scheme from SIPS to SIP and retry. Of course, there are some safeguards against this (if the destination UA is truly paranoid it could refuse all non-SIPS requests), but it is a limitation worth noting. On the bright side, users might also divine that 'SIPS' would be valid even when they are presented only with a SIP URI.

## 26.5 Privacy

SIP messages frequently contain sensitive information about their senders - not just what they have to say, but with whom they communicate, when they communicate and for how long, and from where they participate in sessions. Many applications and their users require that this sort of private information be hidden from any parties that do not need to know it.

Note that there are also less direct ways in which private information can be divulged. If a user or service chooses to be reachable at an address that is guessable from the person's name and organizational affiliation (which describes most addresses-of-record), the traditional method of ensuring privacy by having an unlisted "phone number" is compromised. A user location service can infringe on the privacy of the recipient of a session invitation by divulging their specific whereabouts to the caller; an implementation consequently SHOULD be able to restrict, on a per-user basis, what kind of location and availability information is given out to certain classes of callers. This is a whole class of problem that is expected to be studied further in ongoing SIP work.

In some cases, users may want to conceal personal information in header fields that convey identity. This can apply not only to the From and related headers representing the originator of the request, but also the To - it may not be appropriate to convey to the final destination a speed-dialing nickname, or an unexpanded identifier for a group of targets, either of which would be removed from the Request-URI as the request is routed, but not changed in the To header field if the two were initially identical. Thus it MAY be desirable for privacy reasons to create a To header field that differs from the Request-URI.

# 27 IANA Considerations

All method names, header field names, status codes, and option tags used in SIP applications are registered with IANA through instructions in an IANA Considerations section in an RFC.

The specification instructs the IANA to create four new sub-registries under http://www.iana.org/assignments/sip-parameters: Option Tags, Warning Codes (warn-codes), Methods and Response Codes, added to the sub-registry of Header Fields that is already present there.

## 27.1 Option Tags

This specification establishes the Option Tags sub-registry under http://www.iana.org/assignments/sip-parameters.

Option tags are used in header fields such as Require, Supported, Proxy-Require, and Unsupported in support of SIP compatibility mechanisms for extensions (Section 19.2). The option tag itself is a string that is associated with a particular SIP option (that is, an extension). It identifies the option to SIP endpoints.

Option tags are registered by the IANA when they are published in standards track RFCs. The IANA Considerations section of the RFC must include the following information, which appears in the IANA registry along with the RFC number of the publication.

- o Name of the option tag. The name MAY be of any length, but SHOULD be no more than twenty characters long. The name MUST consist of alphanum (Section 25) characters only.

- o Descriptive text that describes the extension.

## 27.2 Warn-Codes

This specification establishes the Warn-codes sub-registry under http://www.iana.org/assignments/sip-parameters and initiates its population with the warn-codes listed in Section 20.43. Additional warn-codes are registered by RFC publication.

The descriptive text for the table of warn-codes is:

Warning codes provide information supplemental to the status code in SIP response messages when the failure of the transaction results from a Session Description Protocol (SDP) (RFC 2327 [1]) problem.

The "warn-code" consists of three digits. A first digit of "3" indicates warnings specific to SIP. Until a future specification describes uses of warn-codes other than 3xx, only 3xx warn-codes may be registered.

Warnings 300 through 329 are reserved for indicating problems with keywords in the session description, 330 through 339 are warnings related to basic network services requested in the session description, 370 through 379 are warnings related to quantitative QoS parameters requested in the session description, and 390 through 399 are miscellaneous warnings that do not fall into one of the above categories.

## 27.3 Header Field Names

This obsoletes the IANA instructions about the header sub-registry under http://www.iana.org/assignments/sip-parameters.

The following information needs to be provided in an RFC publication in order to register a new header field name:

- o The RFC number in which the header is registered;
- o the name of the header field being registered;
- o a compact form version for that header field, if one is defined;

Some common and widely used header fields MAY be assigned one-letter compact forms (Section 7.3.3). Compact forms can only be assigned after SIP working group review, followed by RFC publication.

## 27.4 Method and Response Codes

This specification establishes the Method and Response-Code sub-registries under http://www.iana.org/assignments/sip-parameters and initiates their population as follows. The initial Methods table is:

| | |
|---|---|
| INVITE | [RFC3261] |
| ACK | [RFC3261] |
| BYE | [RFC3261] |
| CANCEL | [RFC3261] |
| REGISTER | [RFC3261] |
| OPTIONS | [RFC3261] |
| INFO | [RFC2976] |

The response code table is initially populated from Section 21, the portions labeled Informational, Success, Redirection, Client-Error, Server-Error, and Global-Failure. The table has the following format:

```
   Type (e.g., Informational)
         Number    Default Reason Phrase         [RFC3261]
```

The following information needs to be provided in an RFC publication in order to register a new response code or method:

- o The RFC number in which the method or response code is registered;
- o the number of the response code or name of the method being registered;
- o the default reason phrase for that response code, if applicable;

## 27.5 The "message/sip" MIME type.

This document registers the "message/sip" MIME media type in order to allow SIP messages to be tunneled as bodies within SIP, primarily for end-to-end security purposes. This media type is defined by the following information:

Media type name: message
Media subtype name: sip
Required parameters: none

Optional parameters: version
version: The SIP-Version number of the enclosed message (e.g., "2.0"). If not present, the version defaults to "2.0".
Encoding scheme: SIP messages consist of an 8-bit header optionally followed by a binary MIME data object. As such, SIP messages must be treated as binary. Under normal circumstances SIP messages are transported over binary-capable transports, no special encodings are needed.

Security considerations: see below
Motivation and examples of this usage as a security mechanism in concert with S/MIME are given in 23.4.

## 27.6 New Content-Disposition Parameter Registrations

This document also registers four new Content-Disposition header "disposition-types": alert, icon, session and render. The authors request that these values be recorded in the IANA registry for Content-Dispositions.

Descriptions of these "disposition-types", including motivation and examples, are given in Section 20.11.

Short descriptions suitable for the IANA registry are:

| | |
|---|---|
| alert | the body is a custom ring tone to alert the user |
| icon | the body is displayed as an icon to the user |
| render | the body should be displayed to the user |
| session | the body describes a communications session, for example, as RFC 2327 SDP body |

# 28 Changes From RFC 2543

This RFC revises RFC 2543. It is mostly backwards compatible with RFC 2543. The changes described here fix many errors discovered in RFC 2543 and provide information on scenarios not detailed in RFC 2543. The protocol has been presented in a more cleanly layered model here.

We break the differences into functional behavior that is a substantial change from RFC 2543, which has impact on interoperability or correct operation in some cases, and functional behavior that is different from RFC 2543 but not a potential source of interoperability problems. There have been countless clarifications as well, which are not documented here.

## 28.1 Major Functional Changes

- o When a UAC wishes to terminate a call before it has been answered, it sends CANCEL. If the original INVITE still returns a 2xx, the UAC then sends BYE. BYE can only be sent on an existing call leg (now called a dialog in this RFC), whereas it could be sent at any time in RFC 2543.

- o The SIP BNF was converted to be RFC 2234 compliant.

- o SIP URL BNF was made more general, allowing a greater set of characters in the user part. Furthermore, comparison rules were simplified to be primarily case-insensitive, and detailed handling of comparison in the presence of parameters was described. The most substantial change is that a URI with a parameter with the default value does not match a URI without that parameter.

- o Removed Via hiding. It had serious trust issues, since it relied on the next hop to perform the obfuscation process. Instead, Via hiding can be done as a local implementation choice in stateful proxies, and thus is no longer documented.

- o In RFC 2543, CANCEL and INVITE transactions were intermingled. They are separated now. When a user sends an INVITE and then a CANCEL, the INVITE transaction still terminates normally. A UAS needs to respond to the original INVITE request with a 487 response.

- o Similarly, CANCEL and BYE transactions were intermingled; RFC 2543 allowed the UAS not to send a response to INVITE when a BYE was received. That is disallowed here. The original INVITE needs a response.

- o In RFC 2543, UAs needed to support only UDP. In this RFC, UAs need to support both UDP and TCP.

- o In RFC 2543, a forking proxy only passed up one challenge from downstream elements in the event of multiple challenges. In this RFC, proxies are supposed to collect all challenges and place them into the forwarded response.

- o In Digest credentials, the URI needs to be quoted; this is unclear from RFC 2617 and RFC 2069 which are both inconsistent on it.

- o SDP processing has been split off into a separate specification [13], and more fully specified as a formal offer/answer exchange process that is effectively tunneled through SIP. SDP is allowed in INVITE/200 or 200/ACK for baseline SIP implementations; RFC 2543 alluded to the ability to use it in INVITE, 200, and ACK in a single transaction, but this was not well specified. More complex SDP usages are allowed in extensions.

- o Added full support for IPv6 in URIs and in the Via header field. Support for IPv6 in Via has required that its header field parameters allow the square bracket and colon characters. These characters were previously not permitted. In theory, this could cause interop problems with older implementations. However, we have observed that most implementations accept any non-control ASCII character in these parameters.

- o DNS SRV procedure is now documented in a separate specification [4]. This procedure uses both SRV and NAPTR resource records and no longer combines data from across SRV records as described in RFC 2543.

- o Loop detection has been made optional, supplanted by a mandatory usage of Max-Forwards. The loop detection procedure in RFC 2543 had a serious bug which would report "spirals" as an error condition when it was not. The optional loop detection procedure is more fully and correctly specified here.

- o Usage of tags is now mandatory (they were optional in RFC 2543), as they are now the fundamental building blocks of dialog identification.

- o Added the Supported header field, allowing for clients to indicate what extensions are supported to a server, which can apply those extensions to the response, and indicate their usage with a Require in the response.

- o Extension parameters were missing from the BNF for several header fields, and they have been added.

- o Handling of Route and Record-Route construction was very underspecified in RFC 2543, and also not the right approach. It has been substantially reworked in this specification (and made vastly simpler), and this is arguably the largest change. Backwards compatibility is still provided for deployments that do not use "pre-loaded routes", where the initial request has a set of Route header field values obtained in some way outside of Record-Route. In those situations, the new mechanism is not interoperable.

- o In RFC 2543, lines in a message could be terminated with CR, LF, or CRLF. This specification only allows CRLF.

- o Usage of Route in CANCEL and ACK was not well defined in RFC 2543. It is now well specified; if a request had a Route header field, its CANCEL or ACK for a non-2xx response to the request need to carry the same Route header field values. ACKs for 2xx responses use the Route values learned from the Record-Route of the 2xx responses.

- o RFC 2543 allowed multiple requests in a single UDP packet. This usage has been removed.

- o Usage of absolute time in the Expires header field and parameter has been removed. It caused interoperability problems in elements that were not time synchronized, a common occurrence. Relative times are used instead.

- o The branch parameter of the Via header field value is now mandatory for all elements to use. It now plays the role of a unique transaction identifier. This avoids the complex and bug-laden transaction identification rules from RFC 2543. A magic cookie is used in the parameter value to determine if the previous hop has made the parameter globally unique, and comparison falls back to the old rules when it is not present. Thus, interoperability is assured.

- o In RFC 2543, closure of a TCP connection was made equivalent to a CANCEL. This was nearly impossible to implement (and wrong) for TCP connections between proxies. This has been eliminated, so that there is no coupling between TCP connection state and SIP processing.

- o RFC 2543 was silent on whether a UA could initiate a new transaction to a peer while another was in progress. That is now specified here. It is allowed for non-INVITE requests, disallowed for INVITE.

- o PGP was removed. It was not sufficiently specified, and not compatible with the more complete PGP MIME. It was replaced with S/MIME.

- o Added the "sips" URI scheme for end-to-end TLS. This scheme is not backwards compatible with RFC 2543. Existing elements that receive a request with a SIPS URI scheme in the Request-URI will likely reject the request. This is actually a feature; it ensures that a call to a SIPS URI is only delivered if all path hops can be secured.

- o Additional security features were added with TLS, and these are described in a much larger and complete security considerations section.

- o In RFC 2543, a proxy was not required to forward provisional responses from 101 to 199 upstream. This was changed to MUST. This is important, since many subsequent features depend on delivery of all provisional responses from 101 to 199.

- o Little was said about the 503 response code in RFC 2543. It has since found substantial use in indicating failure or overload conditions in proxies. This requires somewhat special treatment. Specifically, receipt of a 503 should trigger an attempt to contact the next element in the result of a DNS SRV lookup. Also, 503 response is only forwarded upstream by a proxy under certain conditions.

- o RFC 2543 defined, but did no sufficiently specify, a mechanism for UA authentication of a server. That has been removed. Instead, the mutual authentication procedures of RFC 2617 are allowed.

- o A UA cannot send a BYE for a call until it has received an ACK for the initial INVITE. This was allowed in RFC 2543 but leads to a potential race condition.

- o A UA or proxy cannot send CANCEL for a transaction until it gets a provisional response for the request. This was allowed in RFC 2543 but leads to potential race conditions.

- o The action parameter in registrations has been deprecated. It was insufficient for any useful services, and caused conflicts when application processing was applied in proxies.

- o RFC 2543 had a number of special cases for multicast. For example, certain responses were suppressed, timers were adjusted, and so on. Multicast now plays a more limited role, and the protocol operation is unaffected by usage of multicast as opposed to unicast. The limitations as a result of that are documented.

- o Basic authentication has been removed entirely and its usage forbidden.

- o Proxies no longer forward a 6xx immediately on receiving it. Instead, they CANCEL pending branches immediately. This avoids a potential race condition that would result in a UAC getting a 6xx followed by a 2xx. In all cases except this race condition, the result will be the same - the 6xx is forwarded upstream.

- o RFC 2543 did not address the problem of request merging. This occurs when a request forks at a proxy and later rejoins at an element. Handling of merging is done only at a UA, and procedures are defined for rejecting all but the first request.

## 28.2 Minor Functional Changes

- o Added the Alert-Info, Error-Info, and Call-Info header fields for optional content presentation to users.

- o Added the Content-Language, Content-Disposition and MIME-Version header fields.

- o Added a "glare handling" mechanism to deal with the case where both parties send each other a re-INVITE simultaneously. It uses the new 491 (Request Pending) error code.

- o Added the In-Reply-To and Reply-To header fields for supporting the return of missed calls or messages at a later time.

- o Added TLS and SCTP as valid SIP transports.

- o There were a variety of mechanisms described for handling failures at any time during a call; those are now generally unified. BYE is sent to terminate.

- o RFC 2543 mandated retransmission of INVITE responses over TCP, but noted it was really only needed for 2xx. That was an artifact of insufficient protocol layering. With a more coherent transaction layer defined here, that is no longer needed. Only 2xx responses to INVITEs are retransmitted over TCP.

- o Client and server transaction machines are now driven based on timeouts rather than retransmit counts. This allows the state machines to be properly specified for TCP and UDP.

- o The Date header field is used in REGISTER responses to provide a simple means for auto-configuration of dates in user agents.

- o Allowed a registrar to reject registrations with expirations that are too short in duration. Defined the 423 response code and the Min-Expires for this purpose.

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# A Table of Timer Values

Table 4 summarizes the meaning and defaults of the various timers used by this specification.

| Timer | Value | Section | Meaning |
|---|---|---|---|
| T1 | 500ms default | Section 17.1.1.1 | RTT Estimate |
| T2 | 4s | Section 17.1.2.2 | The maximum retransmit interval for non-INVITE requests and INVITE responses |
| T4 | 5s | Section 17.1.2.2 | Maximum duration a message will remain in the network |
| Timer A | initially T1 | Section 17.1.1.2 | INVITE request retransmit interval, for UDP only |
| Timer B | 64*T1 | Section 17.1.1.2 | INVITE transaction timeout timer |
| Timer C | > 3min | Section 16.6 bullet 11 | proxy INVITE transaction timeout |
| Timer D | > 32s for UDP<br>0s for TCP/SCTP | Section 17.1.1.2 | Wait time for response retransmits |
| Timer E | initially T1 | Section 17.1.2.2 | non-INVITE request retransmit interval, UDP only |
| Timer F | 64*T1 | Section 17.1.2.2 | non-INVITE transaction timeout timer |
| Timer G | initially T1 | Section 17.2.1 | INVITE response retransmit interval |
| Timer H | 64*T1 | Section 17.2.1 | Wait time for ACK receipt |
| Timer I | T4 for UDP<br>0s for TCP/SCTP | Section 17.2.1 | Wait time for ACK retransmits |
| Timer J | 64*T1 for UDP<br>0s for TCP/SCTP | Section 17.2.2 | Wait time for non-INVITE request retransmits |
| Timer K | T4 for UDP<br>0s for TCP/SCTP | Section 17.1.2.2 | Wait time for response retransmits |

Table 4: Summary of timers

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# Diversion Indication in SIP

## Abstract

This RFC, which contains the text of an Internet Draft that was submitted originally to the SIP Working Group, is being published now for the historical record and to provide a reference for later Informational RFCs.  The original Abstract follows.

This document proposes an extension to the Session Initiation Protocol (SIP).  This extension provides the ability for the called SIP user agent to identify from whom the call was diverted and why the call was diverted.  The extension defines a general header, Diversion, which conveys the diversion information from other SIP user agents and proxies to the called user agent.

This extension allows enhanced support for various features, including Unified Messaging, Third-Party Voicemail, and Automatic Call Distribution (ACD).  SIP user agents and SIP proxies that receive diversion information may use this as supplemental information for feature invocation decisions.

## Status of This Memo

## IESG Note

This document contains an early proposal to the IETF SIP Working Group that was not chosen for standardization. Discussions on the topic resulted in the informational RFC 3325, "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks", and the standard solution that was chosen can be found in RFC 4244, "An Extension to the Session Initiation Protocol (SIP) for Request History Information".

## Copyright Notice

Table of Contents

## 1. Introduction

This RFC, which contains the text of an Internet Draft that was submitted originally to the SIP Working Group, is being published now for the historical record and to provide a reference for later Informational RFCs.

In the legacy telephony network, redirection information is passed through the network in ISDN/ISUP (ISDN User Part) signaling messages. This information is used by various service providers and business applications to support enhanced features for the end user.

An analogous mechanism of providing redirection information would enable such enhanced features for SIP users.

The Diversion header allows implementation of feature logic based on from whom the call was diverted.

## 2. Terminology

### 2.1. Requirements Language

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119].

### 2.2. Definitions

diversion:

A change to the ultimate destination endpoint of a request. A change in the Request-URI of a request that was not caused by a routing decision. This is also sometimes called a deflection or redirection.

A diversion can occur when the "user" portion of the Request-URI is changed for a reason other than expansion or translation.

A diversion can occur when only the "host" portion of the Request-URI has changed if the change was due to a non-routing decision.

divertor:

The entity that diverted the call.

recursing:

A SIP proxy or user agent that handles a received or internally generated 3xx response by forking new request(s) itself.

non-recursing:

A SIP proxy or user agent that handles a received or internally generated 3xx response by forwarding it upstream.

### 2.3. Abbreviations

CFUNC: Call Forward Unconditional

CFTOD: Call Forward Time-of-Day

CFB: Call Forward on Busy

CFNA: Call Forward on No Answer

CFUNV: Call Forward Unavailable

ACD: Automatic Call Distribution

## 3. Overview

In order to implement certain third-party features such as Third-Party Voicemail and Automatic Call Distribution (ACD) applications, diversion information needs to be given to the called third party so that he may respond to the caller intelligently. In these situations, the party receiving a diverted call needs answers for two questions:

Question 1: From whom was the request diverted?

Question 2: Why was the request diverted?

This document proposes usage of the Diversion header to answer these questions for the party receiving the diverted call.

Insertion of the previous Request-URI (before the diversion occurred) into the Diversion header answers question 1.

Insertion of the "reason" tag into the Diversion header (by the divertor) answers question 2.

### 3.1. When Is the Diversion Header Used?

The Diversion header SHOULD be added when a SIP proxy server, SIP redirect server, or SIP user agent changes the ultimate endpoint that will receive the call.

Diversion information SHOULD NOT be added for normal call routing changes to the Request-URI. Thus, the Diversion header is not added when features such as speed dial change the Request-URI.

When a diversion occurs, a Diversion header SHOULD be added to the forwarded request or forwarded 3xx response. The Diversion header MUST contain the Request-URI of the request prior to the diversion. The Diversion header SHOULD contain a reason that the diversion occurred.

Existing Diversion headers received in an incoming request MUST NOT be removed or changed in forwarded requests.

Existing Diversion headers received in an incoming response MUST NOT be removed or changed in the forwarded response.

A Diversion header is added when features such as call forwarding or call deflection change the Request-URI.

## 4. Extension Syntax

The syntax of the Diversion header is:

```
Diversion = "Diversion" ":" 1# (name-addr *( ";" diversion_params ))
diversion-params = diversion-reason | diversion-counter |
                   diversion-limit | diversion-privacy |
                   diversion-screen | diversion-extension
diversion-reason = "reason" "="
                ( "unknown" | "user-busy" | "no-answer" |
                  "unavailable" | "unconditional" |
                  "time-of-day" | "do-not-disturb" |
                  "deflection" | "follow-me" |
                  "out-of-service" | "away" |
                  token | quoted-string )
diversion-counter = "counter" "=" 1*2DIGIT
diversion-limit = "limit" "=" 1*2DIGIT
diversion-privacy = "privacy" "=" ( "full" | "name" |
                    "uri" | "off" | token | quoted-string )
diversion-screen = "screen" "=" ( "yes" | "no" | token |
                                  quoted-string )
diversion-extension = token ["=" (token | quoted-string)]
```

The following is an extension of tables 4 and 5 in [RFC3261] for the Diversion header:

| | where | enc. | e-e | ACK | BYE | CAN | INV | OPT | REG |
|---|---|---|---|---|---|---|---|---|---|
| Diversion | R | | h | - | - | - | o | - | - |
| Diversion | 3xx | | h | - | - | - | o | - | - |

## 5. Detailed Semantics

### 5.1. UAS Behavior

A SIP User Agent Service (UAS) that receives a request and returns a 3xx SHOULD add a Diversion header containing the previous Request-URI and the reason for the diversion.

### 5.2. UAC Behavior

A SIP UAC that receives a 3xx containing a Diversion header SHOULD copy the Diversion header into each downstream forked request that resulted from the 3xx.

### 5.3. Redirect Server Behavior

A SIP redirect server that receives a request and returns a 3xx containing a Contact that diverts the request to a different endpoint SHOULD add a Diversion header containing the Request-URI from the incoming request and the reason for the diversion.

### 5.4. Proxy Server Behavior

A non-recursing SIP proxy that receives a 3xx containing a Diversion header SHOULD forward the 3xx containing the Diversion header upstream unchanged.

A SIP proxy that receives a request and invokes a feature that changes the Request-URI of the forwarded request in order to divert the request to a different endpoint SHOULD add a Diversion header containing the Request-URI from the incoming request and the reason for the diversion.

A SIP proxy that receives a request and returns a 3xx containing a Contact that diverts the request to a different endpoint SHOULD add a Diversion header containing the Request-URI from the incoming request and the reason for the diversion.

#### 5.4.1. Proxy Logic for Diversion Header

```
if (pdu.is_request()) {
    if (request-URI is changed due to a called feature) {
        if (proxy.is_recursing()) {
            Add the Diversion header (indicating the reason
              that the call has been diverted) to
              the downstream forwarded request(s).
        } else {
            Add the Diversion header (indicating the reason
              that the call has been diverted) to
              the upstream forwarded 3xx response.
        }
    }
} else if (pdu.is_response()) {
    if (pdu.is_3xx()) {
        if (proxy.is_recursing()) {
          Copy Diversion header into forwarded INVITE(s).
        } else {
          Forward response upstream.
        }
    }
}
```

## 6. Examples Using Diversion Header

There are several implementations of call forwarding features that can be implemented by either recursing or non-recursing SIP proxies or SIP user agents.

A SIP proxy or user agent that generates or forwards 3xxs upstream is non-recursing. A SIP proxy or user agent that handles received (or internally generated) 3xxs itself is recursing.

The following examples illustrate usage of the Diversion header for some of the variants of recursing and non-recursing proxies and user agents.

### 6.1. Call Forward Unconditional

Usage of the Diversion header is shown below for several variant implementations of Call Forward Unconditional.

#### 6.1.1. Network Call Forward Unconditional (P2 Recursing)

In this message flow, the call would normally be routed to Bob@B. However, Proxy 2 (P2) recursively implements Call Forward Unconditional (CFUNC) to Carol@C.

```
                         +------------------------+
                         | Bob@P2: CFUNC->Carol@C |
                         +-------+----------------+
                                  \
                                   \
A              P1               P2             B          C
                              recursing
|              |              |              |           |
|--INV Bob@P1->|              |              |           |
|              |              |              |           |
|              |--INV Bob@P2->|              |           |
|              |              |              |           |
|              |              |--INVITE Carol@C------->|
|              |              |  Diversion: Bob@P2     |
|              |              |    ;reason=unconditional
|              |              |              |           |
|              |              |<-200-------------------|
|              |              |              |           |
|              |<-200---------|              |           |
|              |              |              |           |
|<-200---------|              |              |           |
|              |              |              |           |
|--ACK------------------------------------------------->|
|              |              |              |           |
|              |              |              |           |
```

### 6.1.2. Network Call Forward Unconditional (P1 Non-Recursing, P2 Non-Recursing)

In this message flow, Proxy 2 (P2) non-recursively implements Call Forward Unconditional (CFUNC) to Carol@C. Proxy 1 (P1) is non-recursing.

```
                     +------------------------+
                     | Bob@P2: CFUNC->Carol@C |
                     +------+-----------------+
                             \
                              \
A              P1             P2            B          C
            non-recursing  non-recursing
|              |              |             |          |
|--INV Bob@P1->|              |             |          |
|              |              |             |          |
|              |--INV Bob@P2->|             |          |
|              |              |             |          |
|              |<-302---------|             |          |
|              |  Contact: Carol@C          |          |
|              |  Diversion: Bob@P2         |          |
|              |    ;reason=unconditional   |          |
|              |              |             |          |
|              |--ACK-------->|             |          |
|              |              |             |          |
|<-302---------|              |             |          |
|  Contact: Carol@C           |             |          |
|  Diversion: Bob@P2          |             |          |
|    ;reason=unconditional    |             |          |
|              |              |             |          |
|--ACK-------->|              |             |          |
|              |              |             |          |
|--INVITE Carol@C------------------------------------->|
|  Diversion: Bob@P2          |             |          |
|    ;reason=unconditional    |             |          |
|              |              |             |          |
|<-200-------------------------------------------------|
|              |              |             |          |
|--ACK------------------------------------------------>|
|              |              |             |          |
|              |              |             |          |
```

### 6.1.3. Network Call Forward Unconditional (P1 Recursing, P2 Non-Recursing)

In this message flow, Proxy 2 (P2) non-recursively implements Call Forward Unconditional (CFUNC) to Carol@C.  Proxy 1 (P1) is recursing.

```
                     +------------------------+
                     | Bob@P2: CFUNC->Carol@C |
                     +------+------------------+
                             \
                              \
A              P1             P2            B          C
            recursing      non-recursing
|              |              |             |          |
|--INV Bob@P1->|              |             |          |
|              |              |             |          |
|              |--INV Bob@P2->|             |          |
|              |              |             |          |
|              |<-302---------|             |          |
|              |  Contact: Carol@C          |          |
|              |  Diversion: Bob@P2         |          |
|              |    ;reason=unconditional   |          |
|              |              |             |          |
|              |--ACK-------->|             |          |
|              |              |             |          |
|              |--INVITE Carol@C---------------------->|
|              |  Diversion: Bob@P2         |          |
|              |    ;reason=unconditional   |          |
|              |              |             |          |
|              |<-200----------------------------------|
|              |              |             |          |
|<-200---------|              |             |          |
|              |              |             |          |
|--ACK------------------------------------------------>|
|              |              |             |          |
|              |              |             |          |
```

#### 6.1.4. Endpoint Call Forward Unconditional (P1 Recursing, P2 Non-Recursing)

In this message flow, user agent server B (B) non-recursively implements Call Forward Unconditional (CFUNC) to Carol@C. Proxy 2 (P2) is non-recursing. Proxy 1 (P1) is recursing.

```
                                    +-----------------------+
                                    | Bob@B: CFUNC->Carol@C |
                                    +------+----------------+
                                            \
                                             \
A              P1             P2              B          C
            recursing      non-recursing
|              |              |               |          |
|--INV Bob@P1->|              |               |          |
|              |              |               |          |
|              |--INV Bob@P2->|               |          |
|              |              |--INV Bob@B--->|          |
|              |              |               |          |
|              |              |<-302----------|          |
|              |              |  Contact: Carol@C        |
|              |              |  Diversion: Bob@B        |
|              |              |    ;reason=unconditional
|              |              |               |          |
|              |              |--ACK--------->|          |
|              |              |               |          |
|              |<-302---------|               |          |
|              |  Contact: Carol@C            |          |
|              |  Diversion: Bob@B            |          |
|              |    ;reason=unconditional     |          |
|              |              |               |          |
|              |--ACK-------->|               |          |
|              |              |               |          |
|              |--INVITE Carol@C------------------------>|
|              |  Diversion: Bob@B            |          |
|              |    ;reason=unconditional     |          |
|              |              |               |          |
|              |<-200------------------------------------|
|              |              |               |          |
|<-200---------|              |               |          |
|              |              |               |          |
|--ACK-------------------------------------------------->|
|              |              |               |          |
|              |              |               |          |
```

### 6.2. Call Forward on Busy

Usage of the Diversion header is shown below for several variant implementations of Call Forward on Busy.

#### 6.2.1. Network Call Forward on Busy (P2 Recursing)

In this message flow, Proxy 2 (P2) recursively implements Call Forward on Busy (CFB) to Carol@C.

```
                     +----------------------+
                     | Bob@P2: CFB->Carol@C |
                     +------+---------------+
                             \
                              \
A              P1             P2            B          C
                           recursing
|              |              |             |          |
|--INV Bob@P1->|              |             |          |
|              |              |             |          |
|              |--INV Bob@P2->|             |          |
|              |              |             |          |
|              |              |--INV Bob@B->|          |
|              |              |             |          |
|              |              |<-486--------|          |
|              |              |             |          |
|              |              |--ACK------->|          |
|              |              |             |          |
|              |              |--INVITE Carol@C------->|
|              |              |  Diversion: Bob@P2     |
|              |              |    ;reason=user-busy   |
|              |              |             |          |
|              |              |<-200-------------------|
|              |              |             |          |
|              |<-200---------|             |          |
|              |              |             |          |
|<-200---------|              |             |          |
|              |              |             |          |
|--ACK------------------------------------------------>|
|              |              |             |          |
|              |              |             |          |
```

### 6.2.2. Network Call Forward on Busy (P1 Non-Recursing, P2 Non-Recursing)

In this message flow, Proxy 2 (P2) non-recursively implements Call Forward on Busy (CFB) to Carol@C. Proxy 1 (P1) is non-recursing.

```
                     +----------------------+
                     | Bob@P2: CFB->Carol@C |
                     +------+---------------+
                             \
                              \
A              P1             P2            B          C
            non-recursing  non-recursing
|              |              |             |          |
|--INV Bob@P1->|              |             |          |
|              |              |             |          |
|              |--INV Bob@P2->|             |          |
|              |              |             |          |
|              |              |--INV Bob@B->|          |
|              |              |             |          |
|              |              |<-486--------|          |
|              |              |             |          |
|              |              |--ACK------->|          |
|              |              |             |          |
|              |<-302---------|             |          |
|              |  Contact: Carol@C          |          |
|              |  Diversion: Bob@P2         |          |
|              |    ;reason=user-busy       |          |
|              |              |             |          |
|              |--ACK-------->|             |          |
|              |              |             |          |
|<-302---------|              |             |          |
|  Contact: Carol@C           |             |          |
|  Diversion: Bob@P2          |             |          |
|    ;reason=user-busy        |             |          |
|              |              |             |          |
|--ACK-------->|              |             |          |
|              |              |             |          |
|--INVITE Carol@C------------------------------------->|
|  Diversion: Bob@P2          |             |          |
|    ;reason=user-busy        |             |          |
|              |              |             |          |
|<-200-------------------------------------------------|
|              |              |             |          |
|--ACK------------------------------------------------>|
|              |              |             |          |
|              |              |             |          |
```

### 6.2.3. Network Call Forward on Busy (P1 Recursing, P2 Non-Recursing)

In this message flow, Proxy 2 (P2) non-recursively implements Call Forward on Busy (CFB) to Carol@C.  Proxy 1 (P1) is recursing.

```
                     +----------------------+
                     | Bob@P2: CFB->Carol@C |
                     +-------+--------------+
                             \
                              \
A              P1             P2            B          C
            recursing      non-recursing
|              |              |             |          |
|--INV Bob@P1->|              |             |          |
|              |              |             |          |
|              |--INV Bob@P2->|             |          |
|              |              |             |          |
|              |              |--INV Bob@B->|          |
|              |              |             |          |
|              |              |<-486--------|          |
|              |              |             |          |
|              |              |--ACK------->|          |
|              |              |             |          |
|              |<-302---------|             |          |
|              |  Contact: Carol@C          |          |
|              |  Diversion: Bob@P2         |          |
|              |    ;reason=user-busy       |          |
|              |              |             |          |
|              |--ACK-------->|             |          |
|              |              |             |          |
|              |--INVITE Carol@C---------------------->|
|              |  Diversion: Bob@P2         |          |
|              |    ;reason=user-busy       |          |
|              |              |             |          |
|              |<-200----------------------------------|
|              |              |             |          |
|<-200---------|              |             |          |
|              |              |             |          |
|--ACK-------------------------------------------------->|
|              |              |             |          |
|              |              |             |          |
```

### 6.2.4. Endpoint Call Forward on Busy (P1 Recursing, P2 Non-Recursing)

In this message flow, user agent server B (B) non-recursively implements Call Forward on Busy (CFB) to Carol@C. Proxy 2 (P2) is non-recursing. Proxy 1 (P1) is recursing.

```
                               +---------------------+
                               | Bob@B: CFB->Carol@C |
                               +------+--------------+
                                          \
                                           \
A              P1             P2            B          C
            recursing      non-recursing
|              |              |             |          |
|--INV Bob@P1->|              |             |          |
|              |              |             |          |
|              |--INV Bob@P2->|             |          |
|              |              |             |          |
|              |              |--INV Bob@B->|          |
|              |              |             |          |
|              |              |<-302--------|          |
|              |              |  Contact: Carol@C      |
|              |              |  Diversion: Bob@B      |
|              |              |    ;reason=user-busy   |
|              |              |             |          |
|              |              |--ACK------->|          |
|              |              |             |          |
|              |<-302---------|             |          |
|              |  Contact: Carol@C          |          |
|              |  Diversion: Bob@B          |          |
|              |    ;reason=user-busy       |          |
|              |              |             |          |
|              |--ACK-------->|             |          |
|              |              |             |          |
|              |--INVITE Carol@C---------------------->|
|              |  Diversion: Bob@B          |          |
|              |    ;reason-user-busy       |          |
|              |              |             |          |
|              |<-200----------------------------------|
|              |              |             |          |
|<-200---------|              |             |          |
|              |              |             |          |
|--ACK------------------------------------------------>|
|              |              |             |          |
|              |              |             |          |
```

### 6.3. Call Forward on No-Answer

Usage of the Diversion header is shown below for several variant implementations of Call Forward on No-Answer.

#### 6.3.1. Network Call Forward on No-Answer (P2 Recursing)

In this message flow, Proxy 2 (P2) recursively implements Call Forward on No Answer (CFNA) to Carol@C.

```
                     +-----------------------+
                     | Bob@P2: CFNA->Carol@C |
                     +------+----------------+
                             \
                              \
A              P1             P2            B          C
                           recursing
|              |              |             |          |
|--INV Bob@P1->|              |             |          |
|              |              |             |          |
|              |--INV Bob@P2->|             |          |
|              |              |             |          |
|              |              |--INV Bob@B->|          |
|              |              |             |          |
|              |              |<-180--------|          |
|              |              |             |          |
|              |              |             |          |
|              |              |             |          |
|              |           timeout          |          |
|              |              |--INVITE Carol@C------->|
|              |              |  Diversion: Bob@P2     |
|              |              |    ;reason=no-answer   |
|              |              |             |          |
|              |              |<-200-------------------|
|              |              |             |          |
|              |<-200---------|             |          |
|              |              |             |          |
|<-200---------|              |             |          |
|              |              |             |          |
|--ACK------------------------------------------------>|
|              |              |             |          |
|              |              |             |          |
```

### 6.3.2. Network Call Forward on No-Answer (P1 Non-Recursing, P2 Non-Recursing)

In this message flow, Proxy 2 (P2) non-recursively implements Call Forward on No Answer (CFNA) to Carol@C. Proxy 1 (P1) is non-recursing.

```
                        +------------------------+
                        | Bob@P2: CFNA->Carol@C |
                        +------+----------------+
                                \
                                 \
A              P1              P2             B           C
            non-recursing  non-recursing
|               |               |              |           |
|--INV Bob@P1->|               |              |           |
|               |               |              |           |
|               |--INV Bob@P2->|              |           |
|               |               |              |           |
|               |               |--INV Bob@B->|           |
|               |               |              |           |
|               |               |<-180--------|           |
|               |               |              |           |
|               |               |              |           |
|               |               |              |           |
|               |            timeout           |           |
|               |<-302---------|              |           |
|               |  Contact: Carol@C            |           |
|               |  Diversion: Bob@P2           |           |
|               |    ;reason=no-answer         |           |
|               |               |              |           |
|               |--ACK-------->|              |           |
|               |               |              |           |
|<-302---------|               |              |           |
|  Contact: Carol@C            |              |           |
|  Diversion: Bob@P2           |              |           |
|    ;reason=no-answer         |              |           |
|               |               |              |           |
|--ACK-------->|               |              |           |
|               |               |              |           |
|--INVITE Carol@C--------------------------------------->|
|  Diversion: Bob@P2           |              |           |
|    ;reason=no-answer         |              |           |
|               |               |              |           |
|<-200---------------------------------------------------|
|               |               |              |           |
|--ACK-------------------------------------------------->|
|               |               |              |           |
```

### 6.3.3. Network Call Forward on No Answer (P1 Recursing, P2 Non-Recursing)

In this message flow, Proxy 2 (P2) non-recursively implements Call Forward on No Answer (CFNA) to Carol@C. Proxy 1 (P1) is recursing.

```
                     +-----------------------+
                     | Bob@P2: CFNA->Carol@C |
                     +------+----------------+
                             \
                              \
A              P1             P2            B          C
            recursing      non-recursing
|              |              |             |          |
|--INV Bob@P1->|              |             |          |
|              |              |             |          |
|              |--INV Bob@P2->|             |          |
|              |              |             |          |
|              |              |--INV Bob@B->|          |
|              |              |             |          |
|              |              |<-180--------|          |
|              |              |             |          |
|              |              |             |          |
|              |              |             |          |
|              |            timeout         |          |
|              |<-302---------|             |          |
|              |  Contact: Carol@C          |          |
|              |  Diversion: Bob@P2         |          |
|              |    ;reason=no-answer       |          |
|              |              |             |          |
|              |--ACK-------->|             |          |
|              |              |             |          |
|              |--INVITE Carol@C---------------------->|
|              |  Diversion: Bob@P2         |          |
|              |    ;reason=no-answer       |          |
|              |              |             |          |
|              |<-200----------------------------------|
|              |              |             |          |
|<-200---------|              |             |          |
|              |              |             |          |
|--ACK------------------------------------------------>|
|              |              |             |          |
|              |              |             |          |
```

### 6.3.4. Endpoint Call Forward on No-Answer (P1 Recursing, P2 Non-Recursing, B Non-Recursing)

In this message flow, user agent server B (B) non-recursively implements Call Forward on No Answer (CFNA) to Carol@C. Proxy 2 (P2) is non-recursing. Proxy 1 (P1) is recursing.

```
                                  +----------------------+
                                  | Bob@B: CFNA->Carol@C |
                                  +------+---------------+
                                          \
                                           \
A              P1             P2            B          C
            recursing      non-recursing
|              |              |             |          |
|--INV Bob@P1->|              |             |          |
|              |              |             |          |
|              |--INV Bob@P2->|             |          |
|              |              |             |          |
|              |              |--INV Bob@B->|          |
|              |              |             |          |
|              |              |             |          |
|              |              |             |          |
|              |              |          timeout       |
|              |              |<-302--------|          |
|              |              |  Contact: Carol@C      |
|              |              |  Diversion: Bob@B      |
|              |              |    ;reason=no-answer   |
|              |              |             |          |
|              |              |--ACK------->|          |
|              |              |             |          |
|              |<-302---------|             |          |
|              |  Contact: Carol@C          |          |
|              |  Diversion: Bob@B          |          |
|              |    ;reason=no-answer       |          |
|              |              |             |          |
|              |--ACK-------->|             |          |
|              |              |             |          |
|              |--INVITE Carol@C---------------------->|
|              |  Diversion: Bob@B          |          |
|              |    ;reason-no-answer       |          |
|              |              |             |          |
|              |<-200----------------------------------|
|              |              |             |          |
|<-200---------|              |             |          |
|              |              |             |          |
|--ACK------------------------------------------------>|
|              |              |             |          |
```

### 6.4. Call Forward on Unavailable

Usage of the Diversion header is shown below for several variant implementations of Call Forward on Unavailable.

#### 6.4.1. Network Call Forward on Unavailable (P2 Recursing)

In this message flow, Proxy 2 (P2) recursively implements Call Forward on Unavailable (CFUNV) to Carol@C.

```
                     +------------------------+
                     | Bob@P2: CFUNV->Carol@C |
                     +------+-----------------+
                             \
                              \
A              P1             P2            B          C
                           recursing
|              |              |             |          |
|--INV Bob@P1->|              |             |          |
|<-100---------|              |             |          |
|              |--INV Bob@P2->|             |          |
|              |<-100---------|             |          |
|              |              |--INV Bob@B->|          |
|              |              |--INV Bob@B->|          |
|              |              |--INV Bob@B->|          |
|              |              |  ...        |          |
|              |              |--INV Bob@B->|          |
|              |           timeout          |          |
|              |              |--INVITE Carol@C------->|
|              |              |  Diversion: Bob@P2     |
|              |              |    ;reason=unavailable
|              |              |             |          |
|              |              |<-200-------------------|
|              |              |             |          |
|              |<-200---------|             |          |
|              |              |             |          |
|<-200---------|              |             |          |
|              |              |             |          |
|--ACK------------------------------------------------>|
|              |              |             |          |
|              |              |             |          |
```

#### 6.4.2. Network Call Forward on Unavailable (P1 Non-Recursing, P2 Non-Recursing)

In this message flow, Proxy 2 (P2) non-recursively implements Call Forward on Unavailable (CFUNV) to Carol@C. Proxy 1 (P1) is non-recursing.

```
                      +-------------------------+
                      | Bob@P2: CFUNV->Carol@C |
                      +-------+-----------------+
                               \
                                \
A               P1              P2             B           C
            non-recursing  non-recursing
|               |               |              |           |
|--INV Bob@P1->|               |              |           |
|               |               |              |           |
|               |--INV Bob@P2->|              |           |
|               |               |              |           |
|               |<-100---------|              |           |
|               |               |              |           |
|<-100---------|               |              |           |
|               |               |--INV Bob@B->|           |
|               |               |--INV Bob@B->|           |
|               |               |--INV Bob@B->|           |
|               |               |  ...        |           |
|               |               |--INV Bob@B->|           |
|               |            timeout           |           |
|               |<-302---------|              |           |
|               |  Contact: Carol@C            |           |
|               |  Diversion: Bob@P2           |           |
|               |    ;reason=unavailable       |           |
|               |               |              |           |
|               |--ACK-------->|              |           |
|               |               |              |           |
|<-302---------|               |              |           |
|  Contact: Carol@C            |              |           |
|  Diversion: Bob@P2           |              |           |
|    ;reason=unavailable       |              |           |
|               |               |              |           |
|--ACK-------->|               |              |           |
|               |               |              |           |
|--INVITE Carol@C------------------------------------->|
|  Diversion: Bob@P2           |              |           |
|    ;reason=unavailable       |              |           |
|               |               |              |           |
|<-200--------------------------------------------------|
|               |               |              |           |
|--ACK------------------------------------------------->|
|               |               |              |           |
|               |               |              |           |
```

### 6.4.3. Network Call Forward on Unavailable (P1 Recursing, P2 Non-Recursing)

In this message flow, Proxy 2 (P2) non-recursively implements Call Forward on Unavailable (CFUNV) to Carol@C. Proxy 1 (P1) is recursing.

```
                     +------------------------+
                     | Bob@P2: CFUNV->Carol@C |
                     +------+-----------------+
                             \
                              \
A              P1             P2            B          C
            recursing      non-recursing
|              |              |             |          |
|--INV Bob@P1->|              |             |          |
|<-100---------|              |             |          |
|              |--INV Bob@P2->|             |          |
|              |<-100---------|             |          |
|              |              |--INV Bob@B->|          |
|              |              |--INV Bob@B->|          |
|              |              |--INV Bob@B->|          |
|              |              |  ...        |          |
|              |              |--INV Bob@B->|          |
|              |           timeout          |          |
|              |<-302---------|             |          |
|              |  Contact: Carol@C          |          |
|              |  Diversion: Bob@P2         |          |
|              |    ;reason=unavailable     |          |
|              |              |             |          |
|              |--ACK-------->|             |          |
|              |              |             |          |
|              |--INVITE Carol@C---------------------->|
|              |  Diversion: Bob@P2         |          |
|              |    ;reason=unavailable     |          |
|              |              |             |          |
|              |<-200----------------------------------|
|              |              |             |          |
|<-200---------|              |             |          |
|              |              |             |          |
|--ACK------------------------------------------------>|
|              |              |             |          |
|              |              |             |          |
```

### 6.5. Multiple Diversions

Usage of the Diversion header when multiple diversions occur are shown the following two examples.

#### 6.5.1. Call Forward Unconditional and Call Forward Busy

In this message flow, Proxy 2 (P2) implements Call Forward Unconditional (CFUNC) to Carol@C. C then implements Call Forward on Busy (CFB) to 5551234@D. P2 is non-recursing. P1 is recursing. C is non-recursing.

```
            +------------------------+     +-------------------------+
            | Bob@P2: CFUNC->Carol@C |     | Carol@C: CFB->5551234@D |
            +---------------+--------+     +--------+----------------+
                             \                       \
                              \                       \
A              P1             P2            B          C          D
            recursing      non-recursing          non-recursing
|              |              |             |          |          |
|--INV Bob@P1->|              |             |          |          |
|              |              |             |          |          |
|              |--INV Bob@P2->|             |          |          |
|              |              |             |          |          |
|              |<-302---------|             |          |          |
|              |  Contact: Carol@C          |          |          |
|              |  Diversion: Bob@P2         |          |          |
|              |    ;reason=unconditional   |          |          |
|              |              |             |          |          |
|              |--ACK-------->|             |          |          |
|              |              |             |          |          |
|              |--INVITE Carol@C---------------------->|          |
|              |  Diversion: Bob@P2         |          |          |
|              |    ;reason=unconditional   |          |          |
|              |              |             |          |          |
|              |<-302----------------------------------|          |
|              |  Contact: 5551234@D        |          |          |
|              |  Diversion: Carol@C        |          |          |
|              |    ;reason=user-busy       |          |          |
|              |    ;privacy="full"         |          |          |
|              |  Diversion: Bob@P2         |          |          |
|              |    ;reason=unconditional   |          |          |
|              |              |             |          |          |
|              |--ACK--------------------------------->|          |
|              |              |             |          |          |
|              |--INVITE 5551234@D------------------------------->|
|              |  Diversion: Carol@C        |          |          |
|              |    ;reason=user-busy       |          |          |
|              |    ;privacy="full"         |          |          |
|              |  Diversion: Bob@P2         |          |          |
|              |    ;reason=unconditional   |          |          |
|              |              |             |          |          |
|              |<-200---------------------------------------------|
|              |              |             |          |          |
|<-200---------|              |             |          |          |
|              |              |             |          |          |
|--ACK----------------------------------------------------------->|
|              |              |             |          |          |
|              |              |             |          |          |
```

### 6.5.2. Call Forward Unconditional and Call Forward No Answer

In this message flow, Proxy 2 (P2) implements Call Forward Unconditional (CFUNC) to Carol@C. (P2 would normally have routed the call to B). C then implements Call Forward on No Answer (CFNA) to 5551234@D. P2 is recursing. C is recursing.

```
         +------------------------+  +--------------------------+
         | Bob@P2: CFUNC->Carol@C |  | Carol@C: CFNA->5551234@D |
         +------------------+-----+  +-----+--------------------+
                             \              \
                              \              \
A              P1             P2       B      C                 D
                            recursing       recursing
|              |              |        |      |                 |
|--INV Bob@P1->|              |        |      |                 |
|              |              |        |      |                 |
|              |--INV Bob@P2->|        |      |                 |
|              |              |        |      |                 |
|              |              |--INV Carol@C->|                 |
|              |              |  Diversion: Bob@P2              |
|              |              |    ;reason=unconditional        |
|              |              |        |      |                 |
|              |              |<--180---------|                 |
|              |              |        |      |                 |
|              |<-180---------|        |      |                 |
|              |              |        |      |                 |
|<-180---------|              |        |      |                 |
|              |              |        |      |                 |
|              |              |        |      |                 |
|              |              |        |    timeout             |
|              |              |        |      |--INV 5551234@D->|
|              |              |        |      |Diversion: Carol@C
|              |              |        |      |  ;reason=no-answer
|              |              |        |      |  ;privacy="full"
|              |              |        |      |Diversion: Bob@P2
|              |              |        |      |  ;reason= unconditional
|              |              |        |      |                 |
|              |              |        |      |<-200------------|
|              |              |        |      |         |       |
|              |              |<-200----------|         |       |
|              |              |        |      |         |       |
|              |<-200---------|        |      |         |       |
|              |              |        |      |         |       |
|<-200---------|              |        |      |         |       |
|              |              |        |      |         |       |
|--ACK--------------------------------------------------------->|
|              |              |        |      |         |       |
```

## 7. Security Considerations

There are some privacy considerations when using the Diversion header. Usage of the Diversion header implies that the diverting UAS trusts the diverted-to UAS. Usage of the Diversion header by SIP proxies or SIP user agents can cause information leakage of route information and called information to untrusted SIP proxies and untrusted callers in upstream 3xxs. Leakage of this information can be mitigated by having a recursing trusted upstream proxy server. For a SIP network architecture where all proxies are required to be non-recursive, Diversion header hiding may be considered necessary in order to prevent leakage of route information to the caller. To accomplish Diversion header hiding, a trusted upstream proxy would add a Record-Route header and use a secret key to encrypt the contents of the Diversion header in 3xxs that are forwarded upstream. On receipt of re-INVITEs, the proxy would decrypt the contents of the Diversion header (using its secret key) and forward the INVITE. There is no currently defined interaction of the Diversion and Hide headers. Question: Should there be?

## 8. Further Examples

Only the relevant headers have been included in the following examples. The contents of the Session Description Protocols (SDPs) have also been omitted.

### 8.1. Night Service/Automatic Call Distribution (ACD) Using Diversion Header

In the following two message flows, two separate companies, WeSellPizza.com and WeSellFlowers.com, have contracted with a third company, NightService.com to provide nighttime support for their incoming voice calls.

In the first flow, Alice calls out for pizza. In the second flow, Alice calls for roses. In both instances, the same night service company (and receptionist, Carol) answers the call. However, because the Diversion header is used, Carol is able to customize her greeting to the caller.

```
                +--------------------------------------+
                | WeSellPizza@P2: CFTOD->nightserv@P3 |
                +------------+-------------------------+
                              \
                               \
UAC             P1              P2                   P3               UAS1
                        (WeSellPizza.com)  (NightService.com)
                                                (ACD)
| [1]           |               |                   |                   |
|-INV pizza@P1->|               |                   |                   |
|               |               |                   |                   |
|               |               |                   |                   |
|               | [2] INVITE WeSellPizza@P2          |                   |
|               |------------->|                    |                   |
|               |               |                   |                   |
|               |               | [3]               |                   |
|               |               |-INV nightserv@P3->|                   |
|               |               | Diversion: WeSellPizza@P2              |
|               |               |   ;reason=time-of-day                  |
|               |               |                   |                   |
|               |               |                   | [4]               |
|               |               |                   |-INV Carol@uas1-->|
|               |               |          Diversion: WeSellPizza@P2 |
|               |               |              ;reason=time-of-day   |
|               |               |                   |                   |
|               |               |                   |<-[5] 200---------|
|               |               |<-[6] 200----------|                   |
|               |<-[7] 200-----|                   |                   |
|<-[8] 200------|               |                   |                   |
|               |               |                   |                   |
|--[9] ACK------------------------------------------------------------->|
|               |               |                   |                   |
|<==========================================="Hello, WeSellPizza"======|
|               |               |                   |                   |
|               |               |                   |                   |
```

Alice calls for pizza.

[1] SIP UAC to SIP proxy server 1:

```
INVITE sip:pizza@p1.isp.com SIP/2.0
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:pizza@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

The ISP's originating proxy translated the keyword pizza to the company WeSellPizza.com

[2] SIP proxy server 1 to SIP proxy server 2 (WeSellPizza.com):

```
INVITE sip:WeSellPizza@p2.isp.com SIP/2.0
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:pizza@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

It's after midnight and the pizza people are in bed. Fortunately, WeSellPizza.com has contracted with NightService.com to answer their nighttime calls. Thus, P2 implements CFTOD to NightService.com.

[3] SIP proxy server 2 (WeSellPizza.com) to
SIP proxy server 3 (NightService.com):

```
INVITE sip:NightService@p3.isp.com SIP/2.0
Via: SIP/2.0/UDP p2.isp.com
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:pizza@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Diversion: <sip:WeSellPizza@p2.isp.com>
  ;reason=time-of-day
Content-Type: application/sdp
```

Carol is available to receive the incoming call.

[4] SIP proxy server 3 (NightService.com) to UAS1 (Carol):

```
INVITE sip:carol@uas1.nightservice.com SIP/2.0
Via: SIP/2.0/UDP p3.isp.com
Via: SIP/2.0/UDP p2.isp.com
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:pizza@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Diversion: <sip:WeSellPizza@p2.isp.com>
  ;reason=time-of-day
Content-Type: application/sdp
```

The ACD keys off the Diversion header to pull up the WeSellPizza FAQ on Carol’s web browser.

[5] UAS1 to SIP proxy server 3:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP p3.isp.com
Via: SIP/2.0/UDP p2.isp.com
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: carol@uas1.nightservice.com
From: sip:alice@isp.com
To: <sip:pizza@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

[6] SIP proxy server 3 to SIP proxy server 2:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP p2.isp.com
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: carol@uas1.nightservice.com
From: sip:alice@isp.com
To: <sip:pizza@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

[7] SIP proxy server 2 to SIP proxy server 1:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: carol@uas1.nightservice.com
From: sip:alice@isp.com
To: <sip:pizza@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

[8] SIP proxy server 1 to UAC

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: carol@uas1.nightservice.com
From: sip:alice@isp.com
To: <sip:pizza@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

[9] SIP UAC to UAS1:

```
ACK sip:uas1.nightservice.com SIP/2.0
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: <sip:pizza@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
```

The RTP flows begin and Carol answers "Hello, WeSellPizza. How may I help you?"

```
                   +-----------------------------------------+
                   | WeSellFlowers@P4: CFTOD->nightserv@P3 |
                   +-------------+-------------------------+
                                  \
                                   \
UAC             P1                P4                  P3              UAS1
                          (WeSellFlowers.com)  (NightService.com)
                                                     (ACD)
| [1]           |                 |                   |                |
|-INV roses@P1->|                 |                   |                |
|               |                 |                   |                |
|               | [2] INVITE WeSellFlowers@P4         |                |
|               |---------------->|                   |                |
|               |                 |                   |                |
|               | [3]             |                   |                |
|               |<-302------------|                   |                |
|               |  Contact: nightservice@P3           |                |
|               |  Diversion: WeSellFlowers@P4        |                |
|               |    ;reason=time-of-day              |                |
|               |                 |                   |                |
|               |--[4] ACK------->|                   |                |
|               |                 |                   |                |
|               | [5]             |                   |                |
|               |-INVITE nightservice@P3------------->|                |
|               | Diversion: WeSellFlowers@P4         |                |
|               |   ;reason=time-of-day               |                |
|               |                 |                   |                |
|               |                 |                   | [6]            |
|               |                 |               -INV Carol@uas1----->|
|               |                 |          Diversion: WeSellFlowers@P4
|               |                 |            ;reason=time-of-day
|               |                 |                                    |
|               |                 |               |<-[7] 200-----------|
|               |<-[8] 200----------------------------|                |
|<-[9] 200------|                 |                   |                |
|               |                 |                   |                |
|--[10] ACK----------------------------------------------------------->|
|               |                 |                   |                |
|<========================================"Hello, WeSellFlowers"======|
|               |                 |                   |                |
|               |                 |                   |                |
```

Alice calls for roses.

[1] SIP UAC to SIP proxy server 1:

```
INVITE sip:roses@p1.isp.com SIP/2.0
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:roses@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

The ISP's originating proxy translated the keyword roses to the company WeSellFlowers.com

[2] SIP proxy server 1 to SIP proxy server 4 (WeSellFlowers.com):

```
INVITE sip:WeSellFlowers@p4.isp.com SIP/2.0
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:roses@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

It's now 1 a.m. and the florists are also in bed. Fortunately, WeSellFlowers.com has contracted with NightService.com to answer their nighttime calls, too. Thus, P4 implements CFTOD to NightService.com.

[3] SIP proxy server 4 (WeSellFlowers.com) to
SIP proxy server 1 (NightService.com):

```
SIP/2.0 302 Moved Temporarily
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: NightService@p3.isp.com
From: sip:alice@isp.com
To: <sip:roses@p1.isp.com>;tag=p4
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Diversion: <sip:WeSellFlowers@p4.isp.com>
  ;reason=time-of-day
```

[4] SIP proxy server 1 to SIP proxy server 4 (WeSellFlowers.com):

```
ACK sip:uas1.nightservice.com SIP/2.0
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: <sip:roses@p1.isp.com>;tag=p4
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
```

[5] SIP proxy server 1 (WeSellFlowers.com) to
SIP proxy server 3 (NightService.com):

```
INVITE sip:NightService@p3.isp.com SIP/2.0
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:roses@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Diversion: <sip:WeSellFlowers@p4.isp.com>
  ;reason=time-of-day
Content-Type: application/sdp
```

Carol is available to receive the incoming call.

[6] SIP proxy server 3 (NightService.com) to UAS1 (Carol):

```
INVITE sip:carol@uas1.nightservice.com SIP/2.0
Via: SIP/2.0/UDP p3.isp.com
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:roses@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Diversion: <sip:WeSellFlowers@p4.isp.com>
  ;reason=time-of-day
Content-Type: application/sdp
```

The ACD keys off the Diversion header to pull up the WeSellFlowers FAQ on Carol’s web browser.

[7] SIP UAS1 to SIP proxy server 3:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP p3.isp.com
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: carol@uas1.nightservice.com
From: sip:alice@isp.com
To: <sip:roses@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

[8] SIP proxy server 3 to SIP proxy server 1:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: carol@uas1.nightservice.com
From: sip:alice@isp.com
To: <sip:roses@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

[9] SIP proxy server 1 to UAC

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: carol@uas1.nightservice.com
From: sip:alice@isp.com
To: <sip:roses@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

[10] SIP UAC to SIP UAS1:

```
ACK sip:uas1.nightservice.com SIP/2.0
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: <sip:roses@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
```

The RTP flows begin and Carol answers "Hello, WeSellFlowers. How may I help you?"

### 8.2. Voicemail Service Using Diversion Header

Bob has contracted his Voicemail to a third-party company, Voicemail.com. In this message flow, Bob has hit the Do-Not-Disturb button on his phone. The Do-Not-Disturb functionality of Bob's phone is configured to CFUNC (Call Forward Unconditional) to voicemail@isp.com. Because the Diversion header is used, Voicemail.com is able to place the incoming call into Bob's voice mailbox.

```
                                +----------------------------------------------+
                                | Bob@UAS1: CFDoNotDisturb->voicemail@isp.com |
                                +---------------------------------------+------+
                                                                         \
                                                                          \
UAC1                  P1                  P2             UAS1        UAS2
                                                                   Voicemail.com
|                     |                    |               |           |
|--[1] INV Bob@P1->|                    |               |           |
|                     |                    |               |           |
|                     |--[2] INV Bob@P2->|               |           |
|                     |                    |               |           |
|                     |                 [3] INV Bob@uas1->|           |
|                     |                    |               |           |
|                     |                  [4] <- 302-------|           |
|                     |                 Contact: voicemail@isp.com    |
|                     |                 Diversion: Bob@uas1           |
|                     |                   ;reason=do-not-disturb      |
|                     |                    |               |           |
|                     |                    |[5] ACK------>|           |
|                     |                    |               |           |
|                     |<-[6] 302---------|               |           |
|                     |  Contact: voicemail@isp.com        |           |
|                     |  Diversion: Bob@uas1               |           |
|                     |    ;reason=do-not-disturb          |           |
|                     |                    |               |           |
|                     |--[7] ACK-------->|               |           |
|<-[8] 302---------|                    |               |           |
|  Contact: voicemail@isp.com             |               |           |
|  Diversion: Bob@uas1                    |               |           |
|    ;reason=do-not-disturb               |               |           |
|                     |                    |               |           |
|--[9] ACK-------->|                    |               |           |
|                     |                    |               |           |
|--[10] INVITE voicemail@isp.com------------------------------->|
|  Diversion: Bob@uas1                    |               |           |
|    ;reason=do-not-disturb               |               |           |
|                     |                    |               |           |
|<--[11] 200-----------------------------------------------------|
|                     |                    |               |           |
|---[12] ACK---------------------------------------------------->|
|                     |                    |               |           |
|                     |                    |               |           |
```

Alice calls Bob.

[1] SIP UAC to SIP proxy server 1:

```
INVITE sip:Bob@p1.isp.com SIP/2.0
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:Bob@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

The ISP's originating proxy routes the request to proxy 2 (P2).

[2] SIP proxy server 1 to SIP proxy server 2:

```
INVITE sip:Bob@p2.isp.com SIP/2.0
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:Bob@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

[3] SIP proxy server 2 to UAS1 (Bob's SIP phone):

```
INVITE sip:Bob@uas1.isp.com SIP/2.0
Via: SIP/2.0/UDP p2.isp.com
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:Bob@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

Since Bob had hit the Do-Not-Disturb button on his SIP phone, Bob's phone forwards the call to his voicemail service.

[4] User agent server 1 (UAS1) to SIP proxy server 2 (P2)

```
SIP/2.0 302 Moved Temporarily
Via: SIP/2.0/UDP p2.isp.com
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: Voicemail@isp.com
From: sip:alice@isp.com
To: <sip:Bob@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Diversion: <sip:Bob@uas1.isp.com>
  ;reason=do-not-disturb
```

[5] SIP proxy server 2 to UAS1 (Bob's SIP phone):

```
ACK sip:Bob@uas1.isp.com SIP/2.0
Via: SIP/2.0/UDP p2.isp.com
From: sip:alice@isp.com
To: <sip:Bob@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
```

[6] SIP proxy server 2 (P2) to SIP proxy server 1 (P1):

```
SIP/2.0 302 Moved Temporarily
Via: SIP/2.0/UDP p1.isp.com
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: Voicemail@isp.com
From: sip:alice@isp.com
To: <sip:Bob@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Diversion: <sip:Bob@uas1.isp.com>
  ;reason=do-not-disturb
```

[7] SIP proxy server 1 to SIP proxy server 2:

```
ACK sip:Bob@p2.isp.com SIP/2.0
Via: SIP/2.0/UDP p1.isp.com
From: sip:alice@isp.com
To: <sip:Bob@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
```

[8] SIP proxy server 1 (P1) to UAC (alice-pc):

```
SIP/2.0 302 Moved Temporarily
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: Voicemail@isp.com
From: sip:alice@isp.com
To: <sip:Bob@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Diversion: <sip:Bob@uas1.isp.com>
  ;reason=do-not-disturb
```

[9] SIP UAC to SIP proxy server 1:

```
ACK sip:Bob@p1.isp.com SIP/2.0
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: <sip:Bob@p1.isp.com>;tag=uas1
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
```

[10] SIP UAC (alice-pc) to Voicemail server.

```
INVITE sip:Voicemail@isp.com SIP/2.0
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: sip:Bob@p1.isp.com
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Diversion: <sip:Bob@uas1.isp.com>
  ;reason=do-not-disturb
Content-Type: application/sdp
```

[11] Voicemail server to SIP UAC (alice-pc):

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP alice-pc.isp.com
Contact: Voicemail@isp.com
From: sip:alice@isp.com
To: <sip:Bob@p1.isp.com>;tag=uas2
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
Content-Type: application/sdp
```

[12] SIP UAC to Voicemail server:

```
ACK sip:Voicemail@isp.com SIP/2.0
Via: SIP/2.0/UDP alice-pc.isp.com
From: sip:alice@isp.com
To: <sip:Bob@p1.isp.com>;tag=uas2
Call-ID: 12345600@alice-pc.isp.com
CSeq: 1 INVITE
```

Because the Diversion header is present, the Voicemail server is able to place Alice’s message into Bob’s voice mailbox.

### 8.3. Questions and Answers on Alternative Approaches

Question 1:

Why do we need the Diversion header when we can see the To: header?

Answer:

a) The To: header is not guaranteed to have significance to the called party.

For example, the To: header may contain a locally significant URL (to the caller) such as a private numbering plan, speed dial digits, telephony escape digits, or telephony prefix digits.

Without a Diversion header, enumerating all possible locally significant To: headers that anyone might use to contact Bob@uas1.isp.com becomes a configuration problem at Voicemail@isp.com and is prone to namespace collision.

Support for Diversion headers enables Bob to contract a third-party service (Voicemail@isp.com) with a single globally significant URL for his voice mailbox (Bob@uas1.isp.com).

b) Given a set of multiple diversions, there is a policy decision of which Diversion header takes precedence for service logic.

Different services (or even different users for the same service) may want to configure this policy differently (first, last, second to last, etc.).

Question 2:

Why do we need the Diversion header when we can see the Via: header?

Answer:

The Via header does not contain information about servers whom have deflected the call (using a 3xx).

## 9. Mapping ISUP/ISDN Redirection Information to SIP Diversion Header

The discussions below regarding ISUP/ISDN reflect generic elements in ISUP/ISDN. In some variations of ISUP/ISDN, the information elements are represented differently. Regardless of the ISUP/ISDN variant, translation should be performed for the "first redirecting number" and the "last redirecting number".

In order to prevent ambiguity, it is important to highlight a terminology mismatch between ISUP/ISDN and SIP. In SIP, a "redirect" indicates the act of returning a 3xx response. In ISUP/ISDN, a "redirection" is diversion of a call by a network entity. In ISUP/ISDN, a call may also be deflected (by an endpoint). Diversion is the more generic term that refers to either the act of an network redirection or endpoint deflection.

In SIP, Diversion can be implemented as either an upstream 3xx (non-recursive) or an additionally forked downstream request (recursive). In the following text, a lowercase "redirect" indicates the SIP usage, while an uppercase "Redirect" indicates ISUP usage.

### 9.1. Mapping ISUP/ISDN Diversion Reason Codes

ISUP and ISDN define the following diversion reasons:

```
0000 = Unknown
0001 = Call forwarding busy or called DTE busy
0010 = Call forwarding no reply
1111 = Call forwarding unconditional or systematic
       call redirection
1010 = Call deflection or call forwarding by the called DTE
1001 = Call forwarding DTE out of order
```

Mapping of ISUP/ISDN reason codes to Diversion reason codes is performed as follows:

| ISUP/ISDN reason code | Diversion reason code |
|---|---|
| 0001 | "user-busy" |
| 0010 | "no-answer" |
| 1111 | "unconditional" |
| 1010 | "deflection" |
| 1001 | "unavailable" |
| 0000 | all others |

### 9.2. Mapping ISUP Redirection Information to SIP Diversion Header

This section describes how generic ISUP diversion information elements may be translated across an ISUP/SIP gateway.

#### 9.2.1. ISUP Definitions

| | |
|---|---|
| Called Party Number | The number of the party to which the call is currently being routed. |
| Redirecting Number | The number to which the call was being routed when the last diversion occurred. |
| Redirecting Reason | The reason that the last diversion occurred. |
| Original Called Number | The number to which the call was being routed when the first diversion occurred. |
| Original Redirecting Reason | The reason that the first diversion occurred. |
| Redirection Counter | The count of the total number of diversions that have occurred. |
| Address Presentation | Indication of whether presentation is allowed or restricted. |

### 9.2.2. ISUP Parameters

When a SIP call transits a SIP/ISUP gateway, the following information in the ISUP message should be examined/set when translating SIP Diversion headers to ISUP diversion information:

1) Redirecting Number

2) Redirecting Reason

3) Redirecting Address Presentation

4) Original Called Number

5) Original Redirecting Reason

6) Original Address Presentation

7) Redirection Counter

An ISUP message contains information on the first and last diversions that occurred. The Redirection number is the number to which the call was being routed when the last diversion occurred. The Redirecting Reason is the reason that the last diversion occurred.

The Original Called Number is the number to which the call was being routed when the first diversion occurred. The Original Redirecting Reason is the reason that the first diversion occurred.

When only one Diversion has occurred, the number to which the call was being routed when the diversion occurred is in the Redirecting Number and the reason for that diversion is carried in the Redirect Reason.

### 9.2.3. ISUP to SIP Translation

The ISUP Redirecting Number SHOULD be used to set the value of the name-addr of the top-most Diversion header. The ISUP Redirecting Number address presentation SHOULD be used to set the value of the diversion-privacy of the top-most Diversion header. The ISUP Redirecting Reason SHOULD be used to set the value of the diversion-reason of the top-most Diversion header. When present, the Original Called Number SHOULD be used to set the name-addr of the bottom-most Diversion header. When present, the Original Redirecting Reason SHOULD be used to set the diversion-reason of the bottom-most Diversion header. When present, the Original Address Presentation SHOULD be used to set the diversion-privacy of the bottom-most Diversion header.

The Redirection Counter value minus 1 SHOULD be stored in the diversion- counter associated with the top-most Diversion header. Presence of the diversion-counter for the bottom-most Diversion header is optional. If present, the diversion-counter of the bottom-most Diversion header SHOULD be 1.

#### 9.2.4. SIP to ISUP Translation

The name-addr of the top-most Diversion header SHOULD be used to set the ISUP Redirecting Number. The diversion-reason of the top-most Diversion header SHOULD be used to set the ISUP Redirecting Reason. The diversion-privacy of the top-most Diversion header SHOULD be used to set the ISUP Redirecting Address Presentation.

When multiple Diversion headers are present, the name-addr of the bottom- most Diversion header SHOULD be used to set the ISUP Original Redirecting Number. When multiple Diversion headers are present, the diversion-reason of the bottom-most Diversion header SHOULD be used to set the ISUP Original Redirecting Reason. When multiple Diversion headers are present, the diversion-privacy of the bottom-most Diversion header SHOULD be used to set the ISUP Original Redirecting Address Presentation.

The ISUP Redirection Counter SHOULD be set equal to the sum of the counters of all Diversion headers in the SIP message. A Diversion header that does not explicitly specify a diversion-counter tag counts as 1.

### 9.2.5. Example of ISUP to SIP Translation

```
                                    ISUP/SIP GW
                                        |
--IAM---------------------------------->|
  Called Party Number    =+19195551004 |
  Redirecting Number     =+19195551002 |
    Address Presentation =presentation restricted
  Original Called Number =+19195551001 |
  RedirectionInformation:              |
    Original Redirecting Reason = Unconditional (1111)
    Redirecting Reason = User busy (0001)
    Redirection Counter = 5            |
                                        |
                                        |--INVITE +19195551004------>
                                        |  Diversion: <tel:+19195551002>
                                        |    ;reason=user-busy
                                        |    ;privacy="full"
                                        |    ;counter=4
                                        |  Diversion: <tel:+19195551001>
                                        |    ;reason=unconditional
                                        |    ;counter=1
                                        |
                                        |
```

### 9.2.6. Example of SIP to ISUP Translation

```
                                   ISUP/SIP GW
                                       |
                                       |<--INVITE +19195551004------
                                       |  Diversion: <tel:+19195551002>
                                       |    ;reason=user-busy
                                       |    ;privacy="full"
                                       |    ;counter=4
                                       |  Diversion: <tel:+19195551001>
                                       |    ;reason=unconditional
                                       |    ;counter=1
                                       |
                                       |
                                       |
<--IAM---------------------------------|
  Called Party Number    =+19195551004 |
  Redirecting Number     =+19195551002 |
    Address Presentation =presentation restricted
  Original Called Number =+19195551001 |
  RedirectionInformation:              |
    Original Redirecting Reason = Unconditional (1111)
    Redirecting Reason = User busy (0001)
    Redirection Counter = 5            |
```

## 9.3. Mapping ISDN Redirection Information to SIP Diversion Header

An ISDN message can contain up to two instances of a Redirecting Number information element. When a diversion occurs, an additional Redirection number information element is added. When a third (or greater) diversion occurs, the new Redirecting Number information element replaces the bottom-most Redirection number information element.

### 9.3.1. ISDN Definitions

| | |
|---|---|
| Called Party Number | The number of the party to which the call is currently being routed. |
| Redirecting Number information element | Aggregate information element that contains Redirecting number and Reason for diversion. |

| | |
|---|---|
| Redirecting Number | The number to which the call was being routed when the last diversion occurred. |
| Reason for Diversion | The reason that the last diversion occurred. |
| Origin of Number | Indicates whether the number is user provided and screened or network provided. |
| Presentation Status | Indicates if presentation is allowed or prohibited. |

### 9.3.2. ISDN Parameters

When a SIP call transits a SIP/ISDN gateway, the following information in the ISDN message should be examined/set when translating SIP Diversion headers to ISDN diversion information:

1) Redirecting Number of the top-most Redirecting Number information element

2) Reason for diversion of the top-most Redirection number information element

3) Origin of Number and Presentation Status of the top-most Redirection number information element

4) Redirection number of the bottom-most Redirection number information element

5) Reason for diversion of the bottom-most Redirection number information element

6) Origin of Number and Presentation Status of the bottom-most Redirection number information element

An ISDN message contains information on the first and last diversions that occurred. The top-most Redirection number information element contains information (including the Redirecting Number, Origin of Number, Presentation Status, and Reason for diversion) about the last diversion that occurred. The bottom-most Redirection number information element contains information (including the Redirecting Number, Origin of Number, Presentation Status, and Reason for diversion) about the first diversion that occurred.

If only one Diversion has occurred, only one Redirection number information element is present.

The Redirecting Number information element has the same Type of Number/Numbering Plan, and Digits as the Calling Party Number information element.

There is no Redirection Counter associated with this ISDN information element.

Notice that the order of the Redirection number information elements in an ISDN message (top=first, bottom=last) is reversed from the order of Diversion headers in a SIP message (top=last, bottom=first).

#### 9.3.3. ISDN to SIP Translation

The Redirecting Number of the top-most ISDN Redirecting Number information element SHOULD be used to set the value of the name-addr of the bottom-most Diversion header. The Reason for Diversion of the top-most ISDN Redirecting Number information element SHOULD be used to set the value of the diversion-reason of the bottom-most Diversion header.

The Origin of Number of the top-most ISDN Redirecting Number information element SHOULD be used to set the value of the diversion-screen of the bottom-most Diversion header. The Presentation Status of the top-most ISDN Redirecting Number information element SHOULD be used to set the value of the diversion-privacy of the bottom-most Diversion header.

The Redirecting Number of the bottom-most ISDN Redirecting Number information element SHOULD be used to set the value of the name-addr of the top-most Diversion header. The Reason for Diversion of the bottom-most ISDN Redirecting Number information element SHOULD be used to set the value of the diversion-reason of the top-most Diversion header.

The Origin of Number of the bottom-most ISDN Redirecting Number information element SHOULD be used to set the value of the diversion-screen of the top-most Diversion header. The Presentation Status of the bottom-most ISDN Redirecting Number information element SHOULD be used to set the value of the diversion-privacy of the top-most Diversion header.

Presence of the diversion-counter in each of the Diversion headers is optional. If present, the diversion-counter of each Diversion header SHOULD be 1.

#### 9.3.4. SIP to ISDN Translation

The name-addr of the top-most Diversion header SHOULD be used to set the Redirecting Number of the bottom-most ISDN Redirecting Number information element.

The diversion-reason of the top-most Diversion header SHOULD be used to set the Reason for Diversion of the bottom-most ISDN Redirecting Number information element.

The diversion-screen of the top-most Diversion header SHOULD be used to set the Origin of Number of the bottom-most ISDN Redirecting Number information element.

The diversion-privacy of the top-most Diversion header SHOULD be used to set the Presentation Status of the bottom-most ISDN Redirecting Number information element.

The name-addr of the bottom-most Diversion header SHOULD be used to set the Redirecting Number of the top-most ISDN Redirecting Number information element.

The diversion-reason of the bottom-most Diversion header SHOULD be used to set the Reason for Diversion of the top-most ISDN Redirecting Number information element.

The diversion-screen of the bottom-most Diversion header SHOULD be used to set the Origin of Number of the top-most ISDN Redirecting Number information element.

The diversion-privacy of the bottom-most Diversion header SHOULD be used to set the Presentation Status of the top-most ISDN Redirecting Number information element.

9.3.5.  Example of ISDN to SIP Translation

```
                        ISDN/SIP GW
                                       |
--Setup------------------------------->|
  Called party number      =+19195551004
  Redirecting Number information element:
    Redirecting Number     =+19195551001
    Reason for redirection = Unconditional (1111)
    Origin of Number       = passed network screening
    Presentation Status    = presentation allowed
  Redirecting Number information element:
    Redirecting Number     =+19195551002
    Reason for redirection = User busy (0001)
    Origin of Number       = passed network screening
    Presentation Status    = presentation prohibited
                                       |
                                       |--INVITE tel:+19195551004---->
                                       |  Diversion: <tel:+19195551002>
                                       |    ;reason=user-busy
                                       |    ;screen="yes"
                                       |    ;privacy="off"
                                       |  Diversion: <tel:+19195551001>
                                       |    ;reason=unconditional
                                       |    ;screen="yes"
                                       |    ;privacy="full"
                                       |
                                       |
```

### 9.3.6. Example of SIP to ISDN Translation

```
                                   ISDN/SIP GW
                                       |
<--Setup-------------------------------|
  Called party number      =+19195551004
  Redirecting Number information element:
    Redirecting Number     =+19195551001
    Reason for redirection = Unconditional (1111)
    Origin of Number       = passed network screening
    Presentation Status    = presentation allowed
  Redirecting Number information element:
    Redirecting Number     =+19195551002
    Reason for redirection = User busy (0001)
    Origin of Number       = passed network screening
    Presentation Status    = presentation prohibited
                                       |
                                       |<--INVITE tel:+19195551004----
                                       |  Diversion: <tel:+19195551002>
                                       |    ;reason=user-busy
                                       |    ;screen="yes"
                                       |    ;privacy="off
                                       |  Diversion: <tel:+19195551001>
                                       |    ;reason=unconditional
                                       |    ;screen="yes"
                                       |    ;privacy="full"
                                       |
```

## 9.4. Information Loss in SIP to ISUP/ISDN Translation

Because ISUP and ISDN only support a subset of the information in a SIP Diversion header, information loss occurs during translation at a SIP/ISUP or SIP/ISDN boundary.

### 9.4.1. Loss of Diversion URI Information

Because ISUP and ISDN only support a subset of URI types (specifically tel: URIs and sip:x@y;user=phone URIs), diversion information occurring for other URI types may be lost when crossing from SIP to ISDN or ISUP.

### 9.4.2. Loss of Diversion Reason Information

Because ISUP and ISDN only support a subset of the reason codes supported by the Diversion header, specific reason code information may be lost when crossing from SIP to ISDN or ISUP.

### 9.4.3. Loss of Diversion Counter Information

Because ISDN does not support a counter field (indicating the number of diversions that have occurred), counter information may be lost when crossing from SIP to ISDN.

## 10. Contributors

Special acknowledgement to both Bryan Byerly and JR Yang. As original authors of this document, both were instrumental is getting this document written.

## 11. Acknowledgements

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## 12. Normative References

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# The Session Initiation Protocol (SIP) "Replaces" Header

## Status of this Memo

This document specifies an Internet standards track protocol for the Internet community, and requests discussion and suggestions for improvements. Please refer to the current edition of the "Internet Official Protocol Standards" (STD 1) for the standardization state and status of this protocol. Distribution of this memo is unlimited.

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## Abstract

This document defines a new header for use with Session Initiation Protocol (SIP) multi-party applications and call control. The Replaces header is used to logically replace an existing SIP dialog with a new SIP dialog. This primitive can be used to enable a variety of features, for example: "Attended Transfer" and "Call Pickup". Note that the definition of these example features is non-normative.

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## 1. Overview

This document describes a SIP [1] extension header field as part of the SIP multiparty applications architecture framework [10]. The Replaces header is used to logically replace an existing SIP dialog with a new SIP dialog. This is especially useful in peer-to-peer call control environments.

One use of the "Replaces" header is to replace one participant with another in a multimedia conversation. While this functionality is already available using 3rd party call control [11] style call control, the 3pcc model requires a central point of control which may not be desirable in many environments. As such, a method of performing these same call control primitives in a distributed, peer-to-peer fashion is very desirable.

Use of a new INVITE with a new header for dialog matching was chosen over making implicit associations in an incoming INVITE based on call-id or other fields for the following reasons:

- o An INVITE already has the correct semantics for a new call

- o Using an explicit Replaces header in a new request makes the intent of the request obvious.

o A unique call-id may be given to the replacement call. This avoids dialog matching problems in any of the related User Agents.

o There are no adverse effects if the header is unsupported.

The Replaces header enables services such as attended call transfer, retrieve from park, and transition from locally mixed conferences to two party calls in a distributed peer-to-peer way. This list of services is not exhaustive. Although the Replaces header is frequently used in combination with the REFER [8] method as used in a Transfer [12], they may be used independently.

For example, Alice is talking to Bob from phone1. She transfers Bob to a Parking Place while she goes to the lab. When she gets there she retrieves the "parked" call from phone2 by sending an INVITE with a Replaces header field to Bob with the dialog information Bob shared with the Parking Place. Alice got this information using some out of band mechanism. Perhaps she subscribed to this information from the Parking Place (using the session dialog package [13]), or went to a website and clicked on a URI. A short call flow for this example follows. (Via and Max-Forwards headers are omitted for clarity.)

```
     Alice          Alice                            Parking
     phone1         phone2             Bob             Place
     |               |                  |                   |
     |<=============================>|                   |
     |               |                  |                   |
     |        Alice transfers Bob to Parking Place          |
     |               |                  |                   |
     |------------REFER/200----------->|     *1    *2      |
     |<--NOTIFY/200 (trying)-----------|--INVITE/200/ACK-->|
     |<--NOTIFY/200 (success)----------|<================>|
     |------------BYE/200------------->|                   |
     |               |                  |                   |
     |               |                  |                   |
     |  Alice later retrieves call from another phone       |
     |               |                  |                   |
     |            *3 |-INV w/Replaces->|                   |
     |               |<--200-----------|                   |
     |               |---ACK---------->|----BYE/200------->|
     |               |<===============>|                   |
     |               |                  |                   |
```

Message *1: Bob-> Parking Place

```
INVITE sip:parkingplace@example.org SIP/2.0
To: <sip:parkingplace@example.org>
From: <sip:bob@example.org>;tag=7743
Call-ID: 425928@bobster.example.org
CSeq: 1 INVITE
Contact: <sip:bob@bobster.example.org>
Referred-By: <sip:alice@phone1.example.org>
```

Message *2: Parking Place -> Bob

```
SIP/2.0 200 OK
To: <sip:parkingplace@example.org>;tag=6472
From: <sip:bob@example.org>;tag=7743
Call-ID: 425928@bobster.example.org
CSeq: 1 INVITE
Contact: <sip:parkplace@monopoly.example.org>
```

Message *3: Alice@phone2 -> Bob

```
INVITE sip:bob@bobster.example.org
To: <sip:bob@example.org>
From: <sip:alice@phone2.example.org>;tag=8983
Call-ID: 09870@phone2.example.org
CSeq: 1 INVITE
Contact: <sip:alice@phone2.example.org>
Require: replaces
Replaces: 425928@bobster.example.org;to-tag=7743;from-tag=6472
```

## 2. Conventions

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14, RFC 2119 [2].

This document refers frequently to the terms "confirmed dialog" and "early dialog". These are defined in Section 12 of SIP [1].

## 3. User Agent Server Behavior: Receiving a Replaces Header

The Replaces header contains information used to match an existing SIP dialog (call-id, to-tag, and from-tag). Upon receiving an INVITE with a Replaces header, the User Agent (UA) attempts to match this information with a confirmed or early dialog. The User Agent Server (UAS) matches the to-tag and from-tag parameters as if they were tags present in an incoming request. In other words, the to-tag parameter is compared to the local tag, and the from-tag parameter is compared to the remote tag.

If more than one Replaces header field is present in an INVITE, or if a Replaces header field is present in a request other than INVITE, the UAS MUST reject the request with a 400 Bad Request response.

The Replaces header has specific call control semantics. If both a Replaces header field and another header field with contradictory semantics are present in a request, the request MUST be rejected with a 400 "Bad Request" response.

If the Replaces header field matches more than one dialog, the UA MUST act as if no match is found.

If no match is found, the UAS rejects the INVITE and returns a 481 Call/Transaction Does Not Exist response. Likewise, if the Replaces header field matches a dialog which was not created with an INVITE, the UAS MUST reject the request with a 481 response.

If the Replaces header field matches a dialog which has already terminated, the UA SHOULD decline the request with a 603 Declined response. (If the matched invitation was just terminated, the replacement request should fail as well. Declining the request with a 600-class response prevents an irritating race-condition where the UA rings or alerts for a replacement call which is not wanted.)

If the Replaces header field matches an active dialog, the UA MUST verify that the initiator of the new INVITE is authorized to replace the matched dialog. If the initiator of the new INVITE has been successfully authenticated as equivalent to the user who is being replaced, then the replacement is authorized. For example, if the user being replaced and the initiator of the replacement dialog share the same credentials for Digest authentication [6], or they sign the replacement request with S/MIME [7] with the same private key and present the (same) corresponding certificate used in the original dialog, then the replacement is authorized.

Alternatively, the Referred-By mechanism [4] defines a mechanism that the UAS can use to verify that a replacement request was sent on behalf of the other participant in the matched dialog (in this case, triggered by a REFER request). If the replacement request contains a Referred-By header that corresponds to the user being replaced, the UA SHOULD treat the replacement as if the replacement was authorized by the replaced party. The Referred-By header SHOULD reference a corresponding, valid Refererred-By Authenticated Identity Body [5].

The UA MAY apply other local policy to authorize the remainder of the request. In other words, the UAS may apply a different policy to the replacement dialog than was applied to the replaced dialog.

In addition, the UA MAY use other authorization mechanisms defined for this purpose in standards track extensions. Extensions could define other mechanisms for transitively asserting authorization of a replacement.

If authorization is successful, the UA attempts to accept the new INVITE, reassign the user interface and other resources of the matched dialog to the new INVITE, and shut down the replaced dialog. If the UA cannot accept the new INVITE (for example: it cannot establish required QoS or keying, or it has incompatible media), the UA MUST return an appropriate error response and MUST leave the matched dialog unchanged.

If the Replaces header field matches a confirmed dialog, it checks for the presence of the "early-only" flag in the Replaces header field. (This flag allows the UAC to prevent a potentially undesirable race condition described in Section 7.1.) If the flag is present, the UA rejects the request with a 486 Busy response. Otherwise, it accepts the new INVITE by sending a 200-class response, and shuts down the replaced dialog by sending a BYE. If the Replaces header field matches an early dialog that was initiated by the UA, it accepts the new INVITE by sending a 200-class response, and shuts down the replaced dialog by sending a CANCEL.

If the Replaces header field matches an early dialog that was not initiated by this UA, it returns a 481 (Call/Transaction Does Not Exist) response to the new INVITE, and leaves the matched dialog unchanged. Note that since Replaces matches only a single dialog, the replacement dialog will not be retargeted according to the same forking logic as the original request which created the early dialog.

(Currently, no use cases have been identified for replacing just a single dialog in this circumstance.)

## 4. User Agent Client Behavior: Sending a Replaces Header

A User Agent that wishes to replace a single existing early or confirmed dialog with a new dialog of its own, MAY send the target User Agent an INVITE request containing a Replaces header field. The User Agent Client (UAC) places the Call-ID, to-tag, and from-tag information for the target dialog in a single Replaces header field and sends the new INVITE to the target. If the user agent only wishes to replace an early dialog (as in the Call Pickup example in Section 7.1), the UAC MAY also include the "early-only" parameter in the Replaces header field. A UAC MUST NOT send an INVITE with a Replaces header field that attempts to replace an early dialog which was not originated by the target of the INVITE with a Replaces header field.

Note that use of this mechanism does not provide a way to match multiple dialogs, nor does it provide a way to match an entire call, an entire transaction, or to follow a chain of proxy forking logic. For example, if Alice replaces Cathy in an early dialog with Bob, but Bob does not answer, Alice’s replacement request will not match other dialogs to which Bob’s UA redirects, nor other branches to which his proxy forwards. Although this specification takes reasonable precautions to prevent unexpected behavior in the face of forking, implementations SHOULD only address replacement requests (i.e., set the Request-URI of the replacement request) to the SIP Contact URI of the target.

## 5. Proxy behavior

Proxy Servers do not require any new behavior to support this extension. They simply pass the Replaces header field transparently as described in the SIP specification.

Note that it is possible for a proxy (especially when forking based on some application layer logic, such as caller screening or time-of-day routing) to forward an INVITE request containing a Replaces header field to a completely orthogonal set of Contacts other than the original request it was intended to replace. In this case, the INVITE request with the Replaces header field will fail.

## 6. Syntax

### 6.1. The Replaces Header

The Replaces header field indicates that a single dialog identified by the header field is to be shut down and logically replaced by the incoming INVITE in which it is contained. It is a request header only, and defined only for INVITE requests. The Replaces header field MAY be encrypted as part of end-to-end encryption. Only a single Replaces header field value may be present in a SIP request.

This document adds the following entry to Table 2 of [1]. Additions to this table are also provided for extension methods defined at the time of publication of this document. This is provided as a courtesy to the reader and is not normative in any way. MESSAGE, SUBSCRIBE and NOTIFY, REFER, INFO, UPDATE, PRACK, and PUBLISH are defined respectively in [15], [16], [8], [17], [18], [19], and [20].

| Header field | where | proxy | ACK | BYE | CAN | INV | OPT | REG | MSG |
|---|---|---|---|---|---|---|---|---|---|
| Replaces | R | | - | - | - | o | - | - | - |

| | | | SUB | NOT | REF | INF | UPD | PRA | PUB |
|---|---|---|---|---|---|---|---|---|---|
| Replaces | R | | - | - | - | - | - | - | - |

The following syntax specification uses the augmented Backus-Naur Form (BNF) as described in RFC 2234 [3]. The syntax below relies on a number of productions from SIP [1].

```
Replaces        = "Replaces" HCOLON callid *(SEMI replaces-param)
replaces-param  = to-tag / from-tag / early-flag / generic-param
to-tag          = "to-tag" EQUAL token
from-tag        = "from-tag" EQUAL token
early-flag      = "early-only"
```

A Replaces header field MUST contain exactly one to-tag and exactly one from-tag, as they are required for unique dialog matching. For compatibility with dialogs initiated by RFC 2543 [9] compliant UAs, a tag of zero matches both tags of zero and null. A Replaces header field MAY contain the early-flag.

Examples:

```
Replaces: 98732@sip.example.com
          ;from-tag=r33th4x0r
          ;to-tag=ff87ff

Replaces: 12adf2f34456gs5;to-tag=12345;from-tag=54321;early-only

Replaces: 87134@171.161.34.23;to-tag=24796;from-tag=0
```

## 6.2. New Option Tag for Require and Supported Headers

This specification defines a new Require/Supported header option tag "replaces". UAs which support the Replaces header MUST include the "replaces" option tag in a Supported header field. UAs that want explicit failure notification if Replaces is not supported MAY include the "replaces" option in a Require header field.

Example:

```
Require: replaces, 100rel
```

## 7. Usage Examples

The following non-normative examples are not intended to enumerate all the possibilities for the usage of this extension, but rather to provide examples or ideas only. For more examples, please see SIP Service Examples [14]. Via and Max-Forwards headers are omitted for clarity and brevity.

### 7.1. Replacing an Early Dialog at the Originator

In this example, Bob just arrived in the lab and hasn't registered there yet. He hears his desk phone ring. He quickly logs into a software UA on a nearby computer. Among other things, the software UA has access to the dialog state of his desk phone. When it notices that his phone is ringing, it offers him the choice of taking the call there. The software UA sends an INVITE with Replaces to Alice. When Alice's UA receives this new INVITE, it CANCELs her original INVITE and connects Alice to Bob.

```
                           Bob                      Bob
    Alice                  desk                     lab
     |                       |                        |
  *1 |-----INVITE----------->|                        |
  *2 |<----180---------------|  Bob hears desk phone  |
     |                       |  ringing from lab but  |
     |                       |  isn't REGISTERed yet  |
     |                       |                        |
     |                       |<--fetch dialog state --|
     |                       |---response ----------->|
*3/4 |<-----INVITE with Replaces/200/ACK--------------|
*5/6 |------CANCEL/200------>|                        |
*7   |<-----487--------------|                        |
     |------ACK------------->|                        |
     |                       |                        |
     |                       |                        |
```

Message *1: Alice -> Bob's desk phone

```
INVITE sip:bob@example.org SIP/2.0
To: <sip:bob@example.org>
From: <sip:alice@example.org>;tag=7743
Call-ID: 425928@phone.example.org
CSeq: 1 INVITE
Contact: <sip:alice@phone.example.org>
```

Message *2: Bob’s desk phone -> Alice

```
SIP/2.0 180 Ringing
To: <sip:bob@example.org>;tag=6472
From: <sip:alice@example.org>;tag=7743
Call-ID: 425928@phone.example.org
CSeq: 1 INVITE
Contact: <sip:bob@bobster.example.org>
```

Message *3: Bob in lab -> Alice

```
INVITE sip:alice@phone.example.org
To: <sip:alice@example.org>
From: <sip:bob@example.org>;tag=8983
Call-ID: 09870@labpc.example.org
CSeq: 1 INVITE
Contact: <sip:bob@labpc.example.org>
Replaces: 425928@phone.example.org
 ;to-tag=7743;from-tag=6472;early-only
```

Message *4: Alice -> Bob in lab

```
SIP/2.0 200 OK
To: <sip:alice@example.org>;tag=9232
From: <sip:bob@example.org>;tag=8983
Call-ID: 09870@labpc.example.org
CSeq: 1 INVITE
Contact: <sip:alice@phone.example.org>
```

Message *5: Alice -> Bob’s desk

```
CANCEL sip:bob@example.org SIP/2.0
To: <sip:bob@example.org>
From: <sip:alice@example.org>;tag=7743
Call-ID: 425928@phone.example.org
CSeq: 1 CANCEL
Contact: <sip:alice@phone.example.org>
```

Message *6: Bob’s desk -> Alice

```
SIP/2.0 200 OK
To: <sip:bob@example.org>
From: <sip:alice@example.org>;tag=7743
Call-ID: 425928@phone.example.org
CSeq: 1 CANCEL
Contact: <sip:bob@bobster.example.org>
```

Message *7: Bob’s desk -> Alice

```
SIP/2.0 487 Request Terminated
To: <sip:bob@example.org>;tag=6472
From: <sip:alice@example.org>;tag=7743
Call-ID: 425928@phone.example.org
CSeq: 1 INVITE
```

## 8. Security Considerations

The extension specified in this document significantly changes the relative security of SIP devices. Currently in SIP, even if an eavesdropper learns the Call-ID, To, and From headers of a dialog, they cannot easily modify or destroy that dialog if Digest authentication or end-to-end message integrity are used.

This extension can be used to disconnect participants or replace participants in a multimedia conversation. As such, invitations with the Replaces header MUST only be accepted if the peer requesting replacement has been properly authenticated using a standard SIP mechanism (Digest or S/MIME), and authorized to request a replacement of the target dialog. All SIP implementations are already required to support Digest Authentication. In addition, implementations which support the Replaces header SHOULD also implement the Referred-By mechanism.

How a User Agent determines which requests are legitimately authorized to make dialog replacements is non-trivial and depends on a considerable amount of local policy configuration. In general, there are four cases when an authorization for a replacement is reasonable or warranted.

1. Replacement made by a party considered equivalent to the replaced party

2. Replacement made on behalf of the replaced party (perhaps transitively)

3. Replacement made by a former participant

4. Replacement made by a specifically authorized party

Starting with #1 for example, if an executive and an assistant both receive requests for a shared address-of-record, if so configured, either should be able to replace dialogs of the other for the shared identity. Both could even share the same keying material (Digest or S/MIME), or one could hold an authorization document signed by the other expressing this relationship. Likewise, in a call center environment, each call center agent could possess credentials to which supervisors also have access.

The most common use case of a replacement is on the request of the replaced participant (who no longer wants to be involved). This is the case in many features, such as completing an Attended Transfer and converting a 3-way call to a point-to-point call. Such replacements are typically triggered by a REFER [8] request from the replaced participant. The Referred-By [4] mechanism defines one way to identify the apparent original requester and can point to a SIP Authenticated Identity Body [5] (an S/MIME-based signed assertion) to secure this information.

In the example in section 1, Alice sends an INVITE with Replaces to Bob. Alice was a former participant in the conversation and had a previous dialog relationship with Bob. Alice can use the same Digest or S/MIME credentials she used to authenticate with Bob during the original call to prove that she was a former participant. Note that this justification for replacing calls is more dangerous than the others, and in most cases is another way to authorize that the replacing participant is available. Implementations SHOULD NOT rely on this method as an authorization mechanism.

The last scenario is the easiest to secure but the least likely to be useful in practice. It is unlikely that an arbitrary host in the Internet is aware of any special authorization relationship between the replaced and the replacing parties. However, this use case may be useful in some environments. Since this usage does not effectively degrade the security of the solution, it is still allowed.

Some mechanisms for obtaining the dialog information needed by the Replaces header (Call-ID, to-tag, and from-tag) include URIs on a web page, subscriptions to an appropriate event package, and notifications after a REFER request. Since manipulating this dialog information could cause User Agents to replace the wrong dialog, use of message integrity protection for this information is STRONGLY RECOMMENDED. Use of end-to-end security mechanisms to encrypt this information is also RECOMMENDED.

This extension was designed to take advantage of future signature or authorization schemes defined in standards track extensions. In general, call control features benefit considerably from such work.

## 9. IANA Considerations

### 9.1. Registration of "Replaces" SIP header

Name of Header: Replaces

Short form: none

Normative description: section 6.1 of this document

### 9.2. Registration of "replaces" SIP Option-tag

Name of option: replaces

Description: Support for the SIP Replaces header

SIP headers defined: Replaces

Normative description: This document

## 10. Acknowledgments

Thanks to Robert Sparks, Alan Johnston, Dan Petrie, Ben Campbell, and many other members of the SIP WG for their continued support of the cause of distributed call control in SIP.

## 11. References

### 11.1. Normative References

[1] Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J., Sparks, R., Handley, M., and E. Schooler, "SIP: Session Initiation Protocol", RFC 3261, June 2002.

[2] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.

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## 13. Full Copyright Statement



## Intellectual Property



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# Session Initiation Protocol Service Examples

## Status of This Memo

This document specifies an Internet Best Current Practices for the Internet Community, and requests discussion and suggestions for improvements. Distribution of this memo is unlimited.

## Abstract

This document gives examples of Session Initiation Protocol (SIP) services. This covers most features offered in so-called IP Centrex offerings from local exchange carriers and PBX (Private Branch Exchange) features. Most of the services shown in this document are implemented in the SIP user agents, although some require the assistance of a SIP proxy. Some require some extensions to SIP including the REFER, SUBSCRIBE, and NOTIFY methods and the Replaces and Join header fields. These features are not intended to be an exhaustive set, but rather show implementations of common features likely to be implemented on SIP IP telephones in a business environment.

Table of Contents

# 1. Overview

This document provides example call flows detailing a SIP implementation of the following traditional telephony services:

| | |
|---|---|
| Call Hold | 3-Way Conference |
| Consultation Hold | Find-Me |
| Music on Hold | Incoming Call Screening |
| Unattended Transfer | Outgoing Call Screening |
| Attended Transfer | Call Park |
| Instant Messaging Transfer | Call Pickup |
| Unconditional Call Forwarding | Automatic Redial |
| Call Forwarding on Busy | Click to Dial |
| Call Forwarding on No Answer | |

Note that the Single Line Extension call flow has been removed from this document and will be covered in a separate document.

The call flows shown in this document were developed in the design of a SIP IP communications network. They represent an example set of so-called IP Centrex services or PBX services.

It is the hope of the authors that this document will be useful for SIP implementers, designers, and protocol researchers alike and will help further the goal of a standard implementation of RFC 3261 [RFC3261] and some of its extensions.

These flows represent carefully checked and working group reviewed scenarios of SIP service examples as a companion to the specifications.

These call flows are based on the current version 2.0 of SIP in RFC 3261 [RFC3261] with Session Description Protocol (SDP) usage described in RFC 3264 [RFC3264]. Other RFCs also form part of the SIP standard and are used and referenced in these call flows.

The SIP specification and the other referenced documents are definitive as far as protocol issues are concerned. Also, these flows do not represent the only way to implement these services -- other approaches such as 3pcc (Third Party Call Control) [RFC3725] or Back-to-Back User Agents (B2BUAs) can be used. This specification does not preclude these or other approaches for implementing such services. The peer-to-peer design and principles of these service examples are described in the Multiparty Framework document [FRAMEWORK].

These flows assume the functionality described in the SIP Call Flow Examples document [RFC3665], which explores basic SIP behavior. Some of the scenarios described herein make use of the SIP method extension REFER [RFC3515], the SIP header extension Replaces [RFC3891], and the SIP header extension Join [RFC3911]. The SIP Events document [RFC3265] describes the use of SUBSCRIBE and NOTIFY, while the SIP Dialog Event Package document [RFC4235] describes the dialog event package. Some examples make use of the GRUU (Globally Routable User Agent URI) extension [GRUU].

These flows were prepared assuming a network of proxies, registrars, and other SIP servers. The use of Secure SIP URIs (sips) is shown throughout this document, implying TLS transport on each hop with assumed certificate validation. However, other security approaches can be used. The use of Digest authentication is shown in some examples.

The emphasis in these call flows is the SIP signaling exchange. As a result, only very simple SDP offer/answer exchanges are shown with audio media. These flows apply equally well for other media and multimedia sessions. For more advanced examples of SDP offer/answer exchanges, refer to [RFC4317].

Each call flow is presented with a textual description of the scenario, a message flow diagram showing the messages exchanged between separate network elements, and the detailed contents of each message shown in the diagram.

For simplicity in reading and editing the document, there are a number of differences between some of the examples and actual SIP messages. For example, the HTTP Digest responses are not actual MD5 encodings. Call-IDs are often repeated, and CSeq counts often begin at 1. Header fields are usually shown in the same order. Usually only the minimum required header field set is shown. Also, message body content lengths are often not calculated, but instead shown as "..." where the actual octet count would be.

## 1.1. Legend for Message Flows

Dashed lines (---) represent control messages that are mandatory to the call scenario. These control messages can be SIP signaling.

Double dashed lines (===) represent media paths between network elements.

Messages with parentheses around the name represent optional control messages.

Messages are identified in the figures as F1, F2, etc. This references the message details in the table that follows the figure.

Lines longer than 72 characters are handled using the <allOneLine> convention defined in Section 2.1 of RFC 4475 [RFC4475].

Comments in the message details are shown in the following form:

/* Comments. */

## 2. Service Examples

### 2.1. Call Hold

```
            Alice           Proxy            Bob
             |    INVITE F1   |              |
             |--------------->|              |
             |                |   INVITE F2  |
             |(100 Trying) F3 |------------->|
             |<---------------|              |
             |                |180 Ringing F4|
             | 180 Ringing F5 |<-------------|
             |<---------------|              |
             |                |  200 OK F6   |
             |    200 OK F7   |<-------------|
             |<---------------|              |
             |     ACK F8     |              |
             |--------------->|    ACK F9    |
             |                |------------->|
             |    Both way RTP Established   |
             |<=============================>|
             |                |INVITE(hold) F10
             |INVITE(hold) F11|<-------------|
             |<---------------|              |
             |    200 OK F12  |              |
             |--------------->|   200 OK F13 |
             |                |------------->|
             |                |     ACK F14  |
             |     ACK F15    |<-------------|
             |<---------------|              |
             |           No RTP Sent!        |
             |                |  INVITE F16  |
             |   INVITE F17   |<-------------|
             |<---------------|              |
             |   200 OK F18   |              |
             |--------------->|  200 OK F19  |
             |                |------------->|
             |                |    ACK F20   |
             |     ACK F21    |<-------------|
             |<---------------|              |
             |    Both way RTP Established   |
             |<=============================>|
             |     BYE F22    |              |
             |--------------->|   BYE F23    |
             |                |------------->|
             |                |   200 OK F24 |
             |    200 OK F25  |<-------------|
             |<---------------|              |
```

In this scenario, Alice calls Bob, then Bob places the call on hold. Bob then takes the call off hold, then Alice hangs up the call. Note that hold is unidirectional in nature. However, a UA that places the other party on hold will generally also stop sending media, resulting in no media exchange between the UAs. Older UAs may set the connection address to 0.0.0.0 when initiating hold. However, this behavior has been deprecated in favor or using the a=inactive SDP attribute if no media is sent, or the a=sendonly attribute if media is still sent.

Also note the use of the rendering feature tag defined in RFC 4235 [RFC4235] used in F10 and F11 to indicate that Bob's UA is no longer rendering media to Bob, i.e., that Bob has placed the call on hold.

Message Details

F1 INVITE Alice -> Proxy 1

```
INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345601@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F2 INVITE Proxy 1 -> Bob

```
INVITE sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
Via: SIP/2.0/TLS client.atlanta.example.com:5061
```

```
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345601@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F3 (100 Trying) Proxy 1 -> Alice

```
SIP/2.0 100 Trying
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345601@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0
```

F4 180 Ringing Bob -> Proxy 1

```
SIP/2.0 180 Ringing
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
```

```
Call-ID: 12345601@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Content Length:0


F5 180 Ringing Proxy 1 -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345601@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Content Length: 0


F6 200 OK Bob -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345601@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

```
F7 200 OK Proxy 1 -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345601@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F8 ACK Alice -> Proxy 1

ACK sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf92
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345601@atlanta.example.com
CSeq: 1 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: 0


F9 ACK Proxy 1 -> Bob

ACK sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK837492.1
Via: SIP/2.0/TLS client.atlanta.example.com:5061
```

```
 ;branch=z9hG4bK74bf92
 ;received=192.0.2.103
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345601@atlanta.example.com
CSeq: 1 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: 0


/* Bob places Alice on hold.  Note that the version is
   incremented in the o= field of the SDP. */

F10 INVITE Bob -> Proxy 1

INVITE sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>;+sip.rendering="no"
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844528 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=sendonly


F11 INVITE Proxy 1 -> Alice

INVITE sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
Via: SIP/2.0/TLS client.biloxi.example.com:5061
```

```
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.105
Record-Route: <sips:ss1.example.com;lr>
Max-Forwards: 69
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>;+sip.rendering="no"
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844528 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=sendonly


/* Alice replies to hold. */

F12 200 OK Alice -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.105
Record-Route: <sips:ss1.example.com;lr>
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...
```

```
v=0
o=alice 2890844526 2890844527 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=recvonly
```

F13 200 OK Proxy 1 -> Bob

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.105
Record-Route: <sips:ss1.example.com;lr>
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844527 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=recvonly
```

F14 ACK Bob -> Proxy 1

```
ACK sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds72
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 1 ACK
```

```
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: 0


F15 ACK Proxy 1 -> Alice

ACK sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds72
 ;received=192.0.2.105
Max-Forwards: 69
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 1 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: 0


/* Bob takes the call off hold. */

F16 INVITE Bob -> Proxy 1

INVITE sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds73
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 2 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844529 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
```

```
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F17 INVITE Proxy 1 -> Alice

INVITE sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK837493.1
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds73
 ;received=192.0.2.105
Record-Route: <sips:ss1.example.com;lr>
Max-Forwards: 69
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 2 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844529 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F18 200 OK Alice -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK837493.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds73
 ;received=192.0.2.105
Record-Route: <sips:ss1.example.com;lr>
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 2 INVITE
```

```
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844528 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F19 200 OK Proxy 1 -> Bob

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds73
 ;received=192.0.2.105
Record-Route: <sips:ss1.example.com;lr>
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 2 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844528 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F20 ACK Bob -> Proxy 1

ACK sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds74
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
```

```
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 2 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: 0
```

F21 ACK Proxy 1 -> Alice

```
ACK sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK837494.1
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds74
 ;received=192.0.2.105
Max-Forwards: 69
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 2 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: 0
```

/* RTP Media stream re-established.  Alice disconnects. */

F22 BYE Alice -> Proxy 1

```
BYE sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf97
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345601@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0
```

F23 BYE Proxy 1 -> Bob

```
BYE sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK837497.1
```

```
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf97
 ;received=192.0.2.103
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345601@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0
```

F24 200 OK Bob -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK837497.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf97
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345601@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0
```

F25 200 OK Proxy 1 -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf97
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345601@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0
```

## 2.2. Consultation Hold

```
           Alice            Proxy           Bob                 Carol
             |                |              |                   |
             |    INVITE F1   |              |                   |
             |--------------->|   INVITE F2  |                   |
             |                |------------->|                   |
             |(100 Trying) F3 |              |                   |
             |<---------------|180 Ringing F4|                   |
             |                |<-------------|                   |
             | 180 Ringing F5 |              |                   |
             |<---------------|  200 OK F6   |                   |
             |                |<-------------|                   |
             |    200 OK F7   |              |                   |
             |<---------------|              |                   |
             |     ACK F8     |              |                   |
             |--------------->|     ACK F9   |                   |
             |                |------------->|                   |
             |     Both way RTP Established  |                   |
             |<=============================>|                   |
             |                |INVITE(hold) F10                  |
             |INVITE(hold) F11|<-------------|                   |
             |<---------------|              |                   |
             |    200 OK F12  |              |                   |
             |--------------->|   200 OK F13 |                   |
             |                |------------->|                   |
             |                |     ACK F14  |                   |
             |                |<-------------|                   |
             |    ACK F15     |              |                   |
             |<---------------|              |                   |
             |         No RTP Sent!          |                   |
             |                |   INVITE F16 |                   |
             |                |<-------------|                   |
             |                |              |    INVITE F17     |
             |                |--------------------------------->|
             |                |(100 Trying) F18                  |
             |                |------------->|                   |
             |                |              | 180 Ringing F19   |
             |                |<---------------------------------|
             |                | 180 Ringing F20                  |
             |                |------------->|                   |
             |                |              |    200 OK F21     |
             |                |<---------------------------------|
             |                |   200 OK F22 |                   |
             |                |------------->|                   |
             |                |    ACK F23   |                   |
             |                |<-------------|                   |
             |                |              |      ACK F24      |
```

```
          |                 |---------------------------------->|
          |                 |         Both way RTP Established  |
          |                 |              |<==================>|
          |                 |   BYE F25    |                    |
          |                 |<-------------|                    |
          |                 |              |       BYE F26      |
          |                 |---------------------------------->|
          |                 |              |    200 OK F27      |
          |                 |<----------------------------------|
          |                 |   200 OK F28 |                    |
          |                 |------------->|                    |
          |                 |   INVITE F29 |                    |
          |    INVITE F30   |<-------------|                    |
          |<----------------|              |                    |
          |    200 OK F31   |              |                    |
          |---------------->|   200 OK F32 |                    |
          |                 |------------->|                    |
          |                 |      ACK F33 |                    |
          |                 |<-------------|                    |
          |    ACK F34      |              |                    |
          |<----------------|              |                    |
          |      Both way RTP Established  |                    |
          |<==============================>|                    |
          |    BYE F35      |              |                    |
          |---------------->|    BYE F36   |                    |
          |                 |------------->|                    |
          |                 |   200 OK F37 |                    |
          |                 |<-------------|                    |
          |  200 OK F38     |              |                    |
          |<----------------|              |                    |
          |                 |              |                    |
```

In this scenario, Alice calls Bob. Bob places call on hold. Bob calls Carol. Bob then disconnects with Carol, then takes the call with Alice off hold. The call ends when Alice hangs up.

Also note the use of the rendering feature tag defined in RFC 4235 [RFC4235] used in F10 to indicate that Bob's UA is no longer rendering media to Bob, i.e., that Bob has placed the call on hold.

Message Details

F1 INVITE Alice -> Proxy 1

```
INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
Max-Forwards: 70
```

```
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F2 INVITE Proxy 1 -> Bob

```
INVITE sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

```
F3 (100 Trying) Proxy 1 -> Alice

SIP/2.0 100 Trying
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0


F4 180 Ringing Bob -> Proxy 1
SIP/2.0 180 Ringing
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Content Length:0


F5 180 Ringing Proxy 1 -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Content Length: 0
```

F6 200 OK Bob -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F7 200 OK Proxy 1 -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...
```

```
v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F8 ACK Alice -> Proxy 1

```
ACK sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf45
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: 0
```

F9 ACK Proxy 1 -> Bob

```
ACK sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK837494.1
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf45
 ;received=192.0.2.103
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: 0
```

```
/* Bob places Alice on hold. */

F10 INVITE Bob -> Proxy 1

INVITE sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>;+sip.rendering="no"
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844528 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=sendonly


F11 INVITE Proxy 1 -> Alice

INVITE sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK837497.1
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.105
Record-Route: <sips:ss1.example.com;lr>
Max-Forwards: 69
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>;+sip.rendering="no"
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...
```

```
v=0
o=bob 2890844527 2890844528 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=sendonly
```

F12 200 OK Alice -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK837497.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.105
Record-Route: <sips:ss1.example.com;lr>
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844527 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=recvonly
```

F13 200 OK Proxy 1 -> Bob

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.105
Record-Route: <sips:ss1.example.com;lr>
From: Bob <sips:bob@biloxi.example.com>;tag=314159
```

```
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844527 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=recvonly


F14 ACK Bob -> Proxy 1

ACK sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashdsg
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: 0


F15 ACK Proxy 1 -> Alice

ACK sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK8374.1
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashdsg
 ;received=192.0.2.105
Max-Forwards: 69
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
```

```
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0
```

F16 INVITE Bob -> Proxy 1

```
INVITE sips:carol@chicago.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds22
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>
Call-ID: 9876543210@biloxi.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844834 2890844834 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 50170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F17 INVITE Proxy 1 -> Carol

```
INVITE sips:carol@client.chicago.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749a.1
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds22
 ;received=192.0.2.105
Record-Route: <sips:ss1.example.com;lr>
Max-Forwards: 69
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>
Call-ID: 9876543210@biloxi.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
```

```
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844834 2890844834 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 50170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F18 (100 Trying) Proxy 1 -> Bob

SIP/2.0 100 Trying
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds22
 ;received=192.0.2.105
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>
Call-ID: 9876543210@biloxi.example.com
CSeq: 1 INVITE
Content-Length: 0


F19 180 Ringing Carol -> Proxy 1
SIP/2.0 180 Ringing
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749a.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds22
 ;received=192.0.2.105
Record-Route: <sips:ss1.example.com;lr>
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=456654
Call-ID: 9876543210@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:carol@client.chicago.example.com>
Content Length:0


F20 180 Ringing Proxy 1 -> Bob

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds22
```

```
 ;received=client.chicago.example.com
Record-Route: <sips:ss1.example.com;lr>
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=456654
Call-ID: 9876543210@biloxi.example.com
CSeq: 1 INVITE
Contact: <sips:carol@client.chicago.example.com>
Content Length: 0


F21 200 OK Carol -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749a.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds22
 ;received=192.0.2.105
Record-Route: <sips:ss1.example.com;lr>
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=456654
Call-ID: 9876543210@biloxi.example.com
CSeq: 1 INVITE
Contact: <sips:carol@client.chicago.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=carol 2890844922 2890844922 IN IP4 client.chicago.example.com
s=
c=IN IP4 client.chicago.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F22 200 OK Proxy 1 -> Bob

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds22
 ;received=192.0.2.105
Record-Route: <sips:ss1.example.com;lr>
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=456654
```

```
Call-ID: 9876543210@biloxi.example.com
CSeq: 1 INVITE
Contact: <sips:carol@client.chicago.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=carol 2890844922 2890844922 IN IP4 client.chicago.example.com
s=
c=IN IP4 client.chicago.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F23 ACK Bob -> Proxy 1

```
ACK sips:carol@client.chicago.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds24
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=456654
Call-ID: 9876543210@biloxi.example.com
CSeq: 1 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: 0
```

F24 ACK Proxy 1 -> Carol

```
ACK sips:carol@client.chicago.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749b.1
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds24
 ;received=192.0.2.105
Max-Forwards: 69
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=456654
Call-ID: 9876543210@biloxi.example.com
CSeq: 1 ACK
```

```
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: 0


F25 BYE Bob -> Proxy 1

BYE sips:carol@client.chicago.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7j
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=456654
Call-ID: 9876543210@biloxi.example.com
CSeq: 2 BYE
Content-Length: 0


F26 BYE Proxy 1 -> Carol

BYE sips:carol@client.chicago.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749k.1
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7j
 ;received=192.0.2.105
Max-Forwards: 69
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=456654
Call-ID: 9876543210@biloxi.example.com
CSeq: 2 BYE
Content-Length: 0


F27 200 OK Carol -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749k.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7j
 ;received=192.0.2.105
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=456654
```

```
Call-ID: 9876543210@biloxi.example.com
CSeq: 2 BYE
Content-Length: 0


F28 200 OK Proxy 1 -> Bob

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7j
 ;received=192.0.2.105
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=456654
Call-ID: 9876543210@biloxi.example.com
CSeq: 2 BYE
Content-Length: 0


/* Bob takes the call off hold. */

F29 INVITE Bob -> Proxy 1

INVITE sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7b
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 2 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844529 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F30 INVITE Proxy 1 -> Alice

```
INVITE sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749q.1
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7b
 ;received=192.0.2.105
Record-Route: <sips:ss1.example.com;lr>
Max-Forwards: 69
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 2 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844529 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F31 200 OK Alice -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749q.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7b
 ;received=192.0.2.105
Record-Route: <sips:ss1.example.com;lr>
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 2 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...
```

```
v=0
o=alice 2890844526 2890844528 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F32 200 OK Proxy 1 -> Bob

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7b
 ;received=192.0.2.105
Record-Route: <sips:ss1.example.com;lr>
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 2 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844528 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F33 ACK Bob -> Proxy 1

ACK sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7d7
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 2 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
```

```
Content-Length: 0

F34 ACK Proxy 1 -> Alice

ACK sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK8374.1
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7d7
 ;received=192.0.2.105
Max-Forwards: 69
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 2 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: 0


F35 BYE Alice -> Proxy 1

BYE sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf10
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345600@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0


F36 BYE Proxy 1 -> Bob

BYE sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK8379.1
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf10
 ;received=192.0.2.103
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345600@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0
```

F37 200 OK Bob -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK8379.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf10
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345600@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0
```

F38 200 OK Proxy 1 -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf10
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345600@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0
```

## 2.3. Music on Hold

```
           Alice             Bob       Music Server
             |                |              |
             |    INVITE F1   |              |
             |--------------->|              |
             | 180 Ringing F2 |              |
             |<---------------|              |
             |    200 OK F3   |              |
             |<---------------|              |
             |     ACK F4     |              |
             |--------------->|              |
             |       RTP      |              |
             |<==============>|              |
             |                |              |
             |   Bob places Alice on hold    |
             |                |              |
             | INVITE (hold) F5              |
             |<---------------|              |
             |    200 OK F6   |              |
             |--------------->|              |
             |     ACK F7     |              |
             |<---------------|              |
             |    no RTP      |              |
             |                |              |
             |  Bob initiates music on hold  |
             |                |              |
             |                |   REFER Refer-To: A F8
             |                |------------->|
             |                |    202 F9    |
             |                |<-------------|
             |                |   NOTIFY F10 |
             |                |<-------------|
             |                |    200 F11   |
             |                |------------->|
             |  INVITE F12 Replaces: B       |
             |<------------------------------|
             |          200 OK F13           |
             |------------------------------>|
             |           ACK F14             |
             |<------------------------------|
             |           RTP Music           |
             |<==============================|
             |     BYE F15    |              |
             |--------------->|  NOTIFY F17  |
             |  200 OK F16    |<-------------|
             |<---------------|  200 OK F18  |
             |                |------------->|
```

```
            |                 |              |
            | The music on hold is complete |
            |                 |              |
            |    Bob takes Alice off hold    |
            |                 |              |
            |  INVITE Replaces: M F19        |
            |<----------------|              |
            |    200 OK F20   |              |
            |---------------->|              |
            |     ACK F21     |              |
            |<----------------|              |
            |       RTP       |              |
            |<===============>|              |
            |            BYE F22             |
            |------------------------------->|
            |           200 OK F23           |
            |<-------------------------------|
```

In this flow, Bob places Alice on hold with music. This is performed by Bob sending a REFER to a Music Server that sends an INVITE with Replaces to Alice. The Music Server then sends RTP music to Alice. Bob picks the call up from hold by sending an INVITE with Replaces to Alice.

Note the use of the rendering feature tag defined in RFC 4235 [RFC4235] used in F5 to indicate that Bob's UA is no longer rendering media to Bob, i.e., that Bob has placed the call on hold. Feature tags are also used in F12 with the automaton (defined in RFC 3840 [RFC3840]) and byeless feature tags (defined in RFC 4235 [RFC4235]) to describe the capabilities of the Music Server.

Should Alice not wish to receive music on hold, her UA could refuse F12 and she will remain on hold with Bob, but in silence.

Message Details

F1 INVITE Alice -> Bob

```
INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:a8342043f@atlanta.example.com;gr>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
```

```
Supported: replaces, gruu
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F2 180 Ringing Bob -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Content-Length: 0


F3 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
```

```
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F4 ACK Alice -> Bob

ACK sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bfd
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: 0


/* Bob places Alice on hold. */

F5 INVITE Bob -> Alice

INVITE sips:a8342043f@atlanta.example.com;gr SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bK874bk
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
From: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: 12345600@atlanta.example.com
CSeq: 712 INVITE
Contact: <sips:bob@client.biloxi.example.com>;+sip.rendering="no"
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844528 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=sendonly
```

F6 200 OK Alice -> Bob

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bK874bk
 ;received=192.0.2.105
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
From: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: 12345600@atlanta.example.com
CSeq: 712 INVITE
Contact: <sips:a8342043f@atlanta.example.com;gr>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844527 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=recvonly
```

F7 ACK Bob -> Alice

```
ACK sips:a8342043f@atlanta.example.com;gr SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKq874b
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
From: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: 12345600@atlanta.example.com
CSeq: 712 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: 0
```

/* Bob REFERs Music Server to establish session with Alice
   which replaces the established session between Alice and Bob. */

F8 REFER Bob -> Music Server

```
REFER sips:music@server.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds9
```

```
     Max-Forwards: 70
     From: Bob <sips:bob@biloxi.example.com>;tag=02134
     To: Music Server <sips:music@server.example.com>
     Call-ID: 4802029847@biloxi.example.com
     CSeq: 1 REFER
   <allOneLine>
     Refer-To: <sips:a8342043f@atlanta.example.com;gr?Replaces=
     12345600%40atlanta.example.com%3Bfrom-tag%3D23431
     %3Bto-tag%3D1234567&Require=replaces>
   </allOneLine>
     Referred-By: <sips:bob@biloxi.example.com>
     Contact: <sips:bob@client.biloxi.example.com>
     Content-Length: 0


     F9 202 Accepted Music Server -> Bob

     SIP/2.0 202 Accepted
     Via: SIP/2.0/TLS client.biloxi.example.com:5061
      ;branch=z9hG4bKnashds9
      ;received=192.0.2.105
     From: Bob <sips:bob@biloxi.example.com>;tag=02134
     To: Music Server <sips:music@server.example.com>;tag=56323
     Call-ID: 4802029847@biloxi.example.com
     Contact: <sips:music@server.example.com>
     CSeq: 1 REFER
     Content-Length: 0


     F10 NOTIFY Music Server -> Bob

     NOTIFY sips:bob@client.biloxi.example.com SIP/2.0
     Via: SIP/2.0/TLS server.example.com:5061
      ;branch=z9hG4bK74bT6
     To: Bob <sips:bob@biloxi.example.com>;tag=02134
     Max-Forwards: 70
     From: Music Server <sips:music@server.example.com>;tag=56323
     Call-ID: 4802029847@biloxi.example.com
     CSeq: 1 NOTIFY
     Event: refer
     Subscription-State: active;expires=60
     Contact: <sips:music@server.example.com>
     Content-Type: message/sipfrag
     Content-Length: ...

     SIP/2.0 100 Trying
```

```
F11 200 OK Bob -> Music Server

SIP/2.0 200 OK
Via: SIP/2.0/TLS server.example.com:5061
 ;branch=z9hG4bK74bT6
 ;received=192.0.2.103
To: Bob <sips:bob@biloxi.example.com>;tag=02134
From: Music Server <sips:music@server.example.com>;tag=56323
Call-ID: 4802029847@biloxi.example.com
CSeq: 1 NOTIFY
Content-Length: 0


/* Music Server places call to Alice to replace session
   between Alice and Bob. */

F12 INVITE Music Server -> Alice

INVITE sips:a8342043f@atlanta.example.com;gr SIP/2.0
Via: SIP/2.0/TLS server.example.com:5061
 ;branch=z9hG4bK74rf
Max-Forwards: 70
From: <sips:music@server.example.com>;tag=0111
To: <sips:a8342043f@atlanta.example.com;gr>
Call-ID: a5-75-34-12-76@server.example.com
CSeq: 1 INVITE
Referred-By: <sips:bob@biloxi.example.com>
Contact: <sips:music@server.example.com>;automaton
 ;+sip.byeless;+sip.rendering="no"
Require: replaces
Replaces: 12345600@atlanta.example.com
 ;from-tag=23431;to-tag=1234567
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=MusicServer 2890844576 2890844576 IN IP4 server.example.com
s=
c=IN IP4 server.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=sendonly
```

F13 200 OK Alice -> Music Server

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS server.example.com:5061
 ;branch=z9hG4bK74rf
 ;received=192.0.2.103
From: <sips:music@server.example.com>;tag=0111
To: <sips:a8342043f@atlanta.example.com;gr>;tag=098594
Call-ID: a5-75-34-12-76@server.example.com
CSeq: 1 INVITE
Contact: <sips:a8342043f@atlanta.example.com;gr>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces, gruu
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=recvonly
```

F14 ACK Music Server -> Alice

```
ACK sips:a8342043f@atlanta.example.com;gr SIP/2.0
Via: SIP/2.0/TLS server.example.com:5061
 ;branch=z9hG4bK7rfF
Max-Forwards: 70
From: <sips:music@server.example.com>;tag=0111
To: <sips:a8342043f@atlanta.example.com;gr>;tag=098594
Call-ID: a5-75-34-12-76@server.example.com
CSeq: 1 ACK
Content-Length: 0
```

F15 BYE Alice -> Bob

```
BYE sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bKnashds7
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: 12345600@atlanta.example.com
```

```
CSeq: 2 BYE
Content-Length: 0


F16 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.105
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: 12345600@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0


/* Music Server reports success back to Bob by returning
   a 200 OK response.  Bob obtains the dialog identifiers
   from the headers included in the response. */

F17 NOTIFY Music Server -> Bob

NOTIFY sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS server.example.com:5061
 ;branch=z9hG4bK74bf9
To: Bob <sips:bob@biloxi.example.com>;tag=02134
Max-Forwards: 70
From: Music Server <sips:music@server.example.com>;tag=56323
Call-ID: 4802029847@biloxi.example.com
CSeq: 2 NOTIFY
Event: refer
Subscription-State: terminated;reason=noresource
Contact: <sips:music@server.example.com>
Content-Type: message/sipfrag
Content-Length: ...

SIP/2.0 200 OK
Via: SIP/2.0/TLS server.example.com:5061
 ;branch=z9hG4bK74rf
 ;received=192.0.2.103
From: <sips:music@server.example.com>;tag=0111
To: <sips:a8342043f@atlanta.example.com;gr>;tag=098594
Call-ID: a5-75-34-12-76@server.example.com
CSeq: 1 INVITE
Contact: <sips:a8342043f@atlanta.example.com;gr>
```

F18 200 OK Bob -> Music Server

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS server.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
To: Bob <sips:bob@biloxi.example.com>;tag=02134
From: Music Server <sips:music@server.example.com>;tag=56323
Call-ID: 4802029847@biloxi.example.com
CSeq: 2 NOTIFY
Content-Length: 0
```

/* Alice is now parked at the Music Server. */

/* Bob picks up the call by sending an INVITE to Alice, who
   replaces the existing session with the Music Server. */

F19 INVITE Bob -> Alice

```
INVITE sips:a8342043f@atlanta.example.com;gr SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bK74bf9
From: Bob <sips:bob@biloxi.example.com>;tag=4i323pr
To: Alice <sips:a8342043f@atlanta.example.com;gr>
Call-ID: uioewrjk2k2were
CSeq: 42121 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER,
 SUBSCRIBE, NOTIFY
Replaces: a5-75-34-12-76@server.example.com
 ;to-tag=098594;from-tag=0111
Contact: <sips:bob@client.biloxi.example.com>
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844631 2890844631 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3458 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=sendrecv
```

F20 200 OK Alice -> Bob

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.105
From: Bob <sips:bob@biloxi.example.com>;tag=4i323pr
To: Alice <sips:a8342043f@atlanta.example.com;gr>;tag=6654323
Call-ID: uioewrjk2k2were
CSeq: 42121 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER,
 SUBSCRIBE, NOTIFY
Contact: <sips:alice@client.atlanta.example.com>
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844576 2890844576 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=sendrecv
```

F21 200 ACK Bob -> Alice

```
ACK sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKj974bf9
From: Bob <sips:bob@biloxi.example.com>;tag=4i323pr
To: Alice <sips:a8342043f@atlanta.example.com;gr>;tag=6654323
Call-ID: uioewrjk2k2were
CSeq: 42121 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER,
 SUBSCRIBE, NOTIFY
Supported: replaces
Content-Length: 0
```

F22 BYE Alice -> Music Server

```
BYE sips:music@server.example.com SIP/2.0
Max-Forwards: 70
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74rf
```

```
To: <sips:music@server.example.com>;tag=0111
From: <sips:a8342043f@atlanta.example.com;gr>;tag=098594
Call-ID: a5-75-34-12-76@server.example.com
CSeq: 15 BYE
Content-Length: 0


F23 200 OK Music Server -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74rf
 ;received=192.0.2.103
To: <sips:music@server.example.com>;tag=0111
From: <sips:a8342043f@atlanta.example.com;gr>;tag=098594
Call-ID: a5-75-34-12-76@server.example.com
CSeq: 15 BYE
Content-Length: 0


/* Normal media session between Alice and Bob is resumed. */
```

## 2.4. Transfer - Unattended

```
          Alice                   Bob                  Carol
            |       INVITE F1      |                     |
            |<---------------------|                     |
            |    180 Ringing F2    |                     |
            |--------------------->|                     |
            |       200 OK F3      |                     |
            |--------------------->|                     |
            |         ACK F4       |                     |
            |<---------------------|                     |
            |         RTP          |                     |
            |<====================>|                     |
            |                      |                     |
            |  Alice performs unattended transfer        |
            |                      |                     |
            | REFER Refer-To:C F5  |                     |
            |--------------------->|                     |
            |  202 Accepted F6     |                     |
            |<---------------------|                     |
            |       NOTIFY F7      |                     |
            |<---------------------|                     |
            |       200 OK F8      |                     |
            |--------------------->|                     |
            |        BYE F9        |                     |
            |--------------------->|                     |
            |      200 OK F10      |                     |
            |<---------------------|                     |
            |    No RTP Session    | INVITE Referred-By: A F11
            |                      |-------------------->|
            |                      |   180 Ringing F12   |
            |                      |<--------------------|
            |                      |      200 OK F13     |
            |                      |<--------------------|
            |                      |        ACK F14      |
            |                      |-------------------->|
            |                      |          RTP        |
            |                      |<===================>|
            |       NOTIFY F15     |                     |
            |<---------------------|                     |
            |       200 OK F16     |                     |
            |--------------------->|                     |
            |                      |                     |
```

In this scenario, Bob calls Alice. Alice then transfers Bob to Carol, then Alice disconnects with Bob. Bob establishes the session to Carol then reports the success back to Alice in the NOTIFY in F15. If the transfer fails, Bob can send a new INVITE back to Alice to re-establish the session.

Despite the BYE sent by Alice in F9, the dialog between Alice and Bob still exists until the subscription created by the REFER has terminated (either due to a NOTIFY containing a Subscription-State: terminated;reason=noresource header field, as in F15, or a 481 response to a NOTIFY).

For more about call transfer, see the transfer document [TRANSFER].

Message Details

F1 INVITE Bob -> Alice

```
INVITE sips:alice@atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>
Call-ID: 12345601@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F2 180 Ringing Alice -> Bob

```
SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.113
From: Bob <sips:bob@biloxi.example.com>;tag=314159
```

```
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Content-Length: 0


F3 200 OK Alice -> Bob

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.113
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F4 ACK Bob -> Alice

ACK sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds2
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 1 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: 0


/* Session is established between Alice and Bob. */
```

```
/* Alice performs unattended transfer of Bob to Carol. */


F5 REFER Alice -> Bob

REFER sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds8
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345601@atlanta.example.com
CSeq: 101 REFER
Refer-To: <sips:carol@chicago.example.com>
Referred-By: <alice@atlanta.example.com>
Contact: <sips:alice@client.atlanta.example.com>
Content-Length: 0


F6 202 Accepted Bob -> Alice

SIP/2.0 202 Accepted
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds8
 ;received=192.0.2.105
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345601@atlanta.example.com
Contact: <sips:bob@client.biloxi.example.com>
CSeq: 101 REFER
Content-Length: 0


F7 NOTIFY Bob -> Alice

NOTIFY sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds32
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 2 NOTIFY
Event: refer
Subscription-State: active;expires=60
Contact: <sips:bob@client.biloxi.example.com>
Content-Type: message/sipfrag
Content-Length: ...
```

```
SIP/2.0 100 Trying


F8 200 OK Alice -> Bob

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds32
 ;received=192.0.2.113
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 2 NOTIFY
Content-Length: 0


/* Alice now disconnects with Bob. */


F9 BYE Alice -> Bob

BYE sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds43
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345601@atlanta.example.com
CSeq: 102 BYE
Content-Length: 0


F10 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds43
 ;received=192.0.2.105
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345601@atlanta.example.com
CSeq: 102 BYE
Content-Length: 0

/* Bob attempts the transfer to Carol. */
```

F11 INVITE Bob -> Carol

```
INVITE sips:carol@chicago.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds1
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>
Call-ID: 7436222@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Referred-By: <alice@atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844539 2890844539 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3458 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F12 180 Ringing Carol -> Bob

```
SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds1
 ;received=192.0.2.113
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=928287
Call-ID: 7436222@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:carol@client.chicago.example.com>
Content-Length: 0
```

F13 200 OK Carol -> Bob

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds1
 ;received=192.0.2.113
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=928287
```

```
Call-ID: 7436222@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:carol@client.chicago.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=carol 2890944542 2890844542 IN IP4 client.chicago.example.com
s=
c=IN IP4 client.chicago.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F14 ACK Bob -> Carol

ACK sips:carol@client.chicago.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds9
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=928287
Call-ID: 7436222@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0


/* Bob and Carol now have established a session.  Bob reports
   success to Alice, which Alice probably ignores. */


F15 NOTIFY Bob -> Alice

NOTIFY sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds67
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 3 NOTIFY
Event: refer
Subscription-State: terminated;reason=noresource
Contact: <sips:bob@client.biloxi.example.com>
Content-Type: message/sipfrag
```

```
Content-Length: ...

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds1
 ;received=192.0.2.113
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=928287
Call-ID: 7436222@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:carol@client.chicago.example.com>
Content-Type: application/sdp
Content-Length: ...


F16 200 OK Alice -> Bob

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds6
 ;received=192.0.2.113
From: Bob <sips:bob@biloxi.example.com>;tag=314159
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345601@atlanta.example.com
CSeq: 3 NOTIFY
Content-Length: 0
```

## 2.5. Transfer - Attended

```
           Alice             Bob          Carol
             |                |              |
             |    INVITE F1   |              |
             |--------------->|              |
             | 180 Ringing F2 |              |
             |<---------------|              |
             |    200 OK F3   |              |
             |<---------------|              |
             |     ACK F4     |              |
             |--------------->|              |
             |       RTP      |              |
             |<==============>|              |
             |INVITE (hold) F5|              |
             |<---------------|              |
             |    200 OK F6   |              |
             |--------------->|              |
             |     ACK F7     |              |
             |<---------------|              |
             |     No RTP     |              |
             |                |  INVITE F8   |
             |                |------------->|
             |                | 180 Ringing F9
             |                |<-------------|
             |                |  200 OK F10  |
             |                |    ACK F11   |
             |                |------------->|
             |                |     RTP      |
             |                |<============>|
             |                |INVITE (hold) F12
             |                |------------->|
             |                | 200 OK F13   |
             |                |<-------------|
             |                |    ACK F14   |
             |                |------------->|
             |                |     No RTP   |
             | REFER Refer-To: C F15         |
             |<---------------|              |
             |202 Accepted F16|              |
             |--------------->|              |
             |   NOTIFY F17   |              |
             |--------------->|              |
             |   200 OK F18   |              |
             |<---------------|              |
             |     INVITE Replaces: B F19    |
             |----------------------------->|
```

```
             |            200 OK F20           |
             |<--------------------------------|
             |              ACK F21            |
             |-------------------------------->|
             |                RTP              |
             |<===============================>|
             |                |    BYE F22     |
             |                |<---------------|
             |                |  200 OK F23    |
             |                |--------------->|
             |   NOTIFY F24   |                |
             |--------------->|                |
             |   200 OK F25   |                |
             |<---------------|                |
             |    BYE F26     |                |
             |<---------------|                |
             |   200 OK F27   |                |
             |--------------->|                |
```

In this scenario, Alice calls Bob. Bob puts Alice on hold then calls Carol to announce transfer, then places Carol on hold. Bob transfers Alice to Carol, which replaces the session between Bob and Carol. Carol then disconnects session with Bob. Alice reports success of transfer to Bob, who then disconnects with Alice. In this example, the Replaces header field [RFC3891] is inserted into the Refer-To URI by Bob. Note that the Refer-To URI is the Contact URI returned by Carol in the 200 OK response F10. This ensures that only the correct instance of Carol is reached. The presence of the gr URI parameter in the Contact URI in message F10 indicates that the Contact URI is a GRUU [GRUU] and will be globally routable outside of the dialog. Without knowing the Contact URI is a gruu, Bob must be prepared, if the triggered INVITE had failed, to retry the REFER with a Refer-To URI of the URI used to reach Carol but with a Require: replaces header escaped in the Refer-To header field, as discussed in the transfer document [TRANSFER].

Message Details

F1 INVITE Alice -> Bob

```
INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
```

```
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F2 180 Ringing Bob -> Alice

```
SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Content-Length: 0
```

F3 200 OK Bob -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...
```

```
v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F4 ACK Alice -> Bob

ACK sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: 0


/* Alice and Bob have established a session.
   Bob puts Alice on hold. */


F5 INVITE Bob -> Alice

INVITE sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=23431
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1024 INVITE
Contact: <sips:bob@client.biloxi.example.com>;+sip.rendering="no"
Content-Type: application/sdp
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: ...

v=0
o=bob 2890844527 2890844528 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
```

```
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=sendonly


F6 200 OK Alice -> Bob

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.113
From: Bob <sips:bob@biloxi.example.com>;tag=23431
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1024 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844527 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=recvonly


F7 ACK Bob -> Alice

ACK sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds3
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=23431
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1024 ACK
Content-Length: 0

/* Bob calls Carol. */
```

F8 INVITE Bob -> Carol

```
INVITE sips:carol@chicago.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnash
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>
Call-ID: sdjfdjfskdf@biloxi.example.com
CSeq: 42 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 28908445834 2890844834 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3458 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F9 180 Ringing Carol -> Bob

```
SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnash
 ;received=192.0.2.113
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=5f35a3
Call-ID: sdjfdjfskdf@biloxi.example.com
CSeq: 42 INVITE
Contact: <sips:39itp34klkd@chicago.example.com>
Content-Length: 0
```

F10 200 OK Carol -> Bob

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnash
 ;received=192.0.2.113
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=5f35a3
Call-ID: sdjfdjfskdf@biloxi.example.com
```

```
CSeq: 42 INVITE
Contact: <sips:39itp34klkd@chicago.example.com;gr>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu
Content-Type: application/sdp
Content-Length: ...

v=0
o=carol 2890844922 2890844922 IN IP4 client.chicago.example.com
s=
c=IN IP4 client.chicago.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F11 ACK Bob -> Carol

ACK sips:39itp34klkd@chicago.example.com;gr SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashd5
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=5f35a3
Call-ID: sdjfdjfskdf@biloxi.example.com
CSeq: 42 ACK
Content-Length: 0

/* Bob puts Carol on hold. */


F12 INVITE Bob -> Carol

INVITE sips:39itp34klkd@chicago.example.com;gr SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds0
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=5f35a3
Call-ID: sdjfdjfskdf@biloxi.example.com
CSeq: 43 INVITE
Contact: <sips:bob@client.biloxi.example.com>;+sip.rendering="no"
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...
```

```
v=0
o=bob 289084834 2890844835 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3458 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=sendonly


F13 200 OK Carol -> Bob

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds0
 ;received=192.0.2.113
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=5f35a3
Call-ID: sdjfdjfskdf@biloxi.example.com
CSeq: 43 INVITE
Contact: <sips:39itp34klkd@chicago.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu
Content-Type: application/sdp
Content-Length: ...

v=0
o=carol 2890844922 2890844923 IN IP4 client.chicago.example.com
s=
c=IN IP4 client.chicago.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=recvonly


F14 ACK Bob -> Carol

ACK sips:39itp34klkd@chicago.example.com;gr SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnash334
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=5f35a3
Call-ID: sdjfdjfskdf@biloxi.example.com
```

```
   CSeq: 43 ACK
   Content-Length: 0

   /* Bob transfers Alice to Carol. */


   F15 REFER Bob -> Alice

   REFER sips:alice@client.atlanta.example.com SIP/2.0
   Via: SIP/2.0/TLS client.biloxi.example.com:5061
    ;branch=z9hG4bKnashds2g
   Max-Forwards: 70
   From: Bob <sips:bob@biloxi.example.com>;tag=23431
   To: Alice <sips:alice@atlanta.example.com>;tag=1234567
   Call-ID: 12345600@atlanta.example.com
   CSeq: 1025 REFER
 <allOneLine>
   Refer-To: <sips:39itp34klkd@chicago.example.com?Replaces=
   sdjfdjfskdf%40biloxi.example.com%3Bto-tag%3D5f35a3
   %3Bfrom-tag%3D8675309&Require=replaces>
 </allOneLine>
   Referred-By: <sips:bob@biloxi.example.com>
   Contact: <sips:bob@client.biloxi.example.com>
   Content-Length: 0


   F16 202 Accepted Alice -> Bob

   SIP/2.0 202 Accepted
   Via: SIP/2.0/TLS client.biloxi.example.com:5061
    ;branch=z9hG4bKnashds2g
    ;received=192.0.2.113
   From: Bob <sips:bob@biloxi.example.com>;tag=23431
   To: Alice <sips:alice@atlanta.example.com>;tag=1234567
   Call-ID: 12345600@atlanta.example.com
   Contact: <sips:alice@client.atlanta.example.com>
   CSeq: 1025 REFER
   Content-Length: 0


   F17 NOTIFY Alice -> Bob

   NOTIFY sips:bob@client.biloxi.example.com SIP/2.0
   Via: SIP/2.0/TLS client.atlanta.example.com:5061
    ;branch=z9hG4bK74bfK
   Max-Forwards: 70
   From: Alice <sips:alice@atlanta.example.com>;tag=1234567
   To: Bob <sips:bob@biloxi.example.com>;tag=23431
```

```
   Call-ID: 12345600@atlanta.example.com
   CSeq: 2 NOTIFY
   Contact: <sips:alice@client.atlanta.example.com>
   Event: refer
   Subscription-State: active;expires=60
   Content-Type: message/sipfrag
   Content-Length: ...

   SIP/2.0 100 Trying


   F18 200 OK Bob -> Alice

   SIP/2.0 200 OK
   Via: SIP/2.0/TLS client.atlanta.example.com:5061
    ;branch=z9hG4bK74bfK
    ;received=192.0.2.103
   From: Alice <sips:alice@atlanta.example.com>;tag=1234567
   To: Bob <sips:bob@biloxi.example.com>;tag=23431
   Call-ID: 12345600@atlanta.example.com
   CSeq: 2 NOTIFY
   Content-Length: 0

   /* Alice establishes session with Carol, which replaces the
      session between Bob and Carol. */


   F19 INVITE Alice -> Carol

   INVITE sips:39itp34klkd@chicago.example.com;gr SIP/2.0
   Via: SIP/2.0/TLS chicago.example.com:5061
    ;branch=z9hG4bKadfe4ko
   To: Carol <sips:39itp34klkd@chicago.example.com>
   Max-Forwards: 70
   From: Alice <sips:alice@atlanta.example.com>;tag=3461
   Call-ID: 9435674543@atlanta.example.com
   CSeq: 1 INVITE
   Require: replaces
   Referred-By: <sips:bob@biloxi.example.com>
   Replaces: sdjfdjfskdf@biloxi.example.com
    ;to-tag=5f35a3;from-tag=8675309
   Contact: <sips:alice@client.atlanta.example.com>
   Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
   Supported: replaces
   Content-Type: application/sdp
   Content-Length: ...
```

```
v=0
o=alice 2890844989 2890844989 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 3458 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F20 200 OK Carol -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/TLS chicago.example.com:5061
 ;branch=z9hG4bKadfe4ko
 ;received=192.0.2.103
To: Carol <sips:39itp34klkd@chicago.example.com>;tag=ff3a
From: Alice <sips:alice@atlanta.example.com>;tag=3461
Call-ID: 9435674543@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:39itp34klkd@chicago.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu
Content-Type: application/sdp
Content-Length: ...

v=0
o=carol 2890844221 2890844221 IN IP4 client.chicago.example.com
s=
c=IN IP4 client.chicago.example.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F21 ACK Alice -> Carol

ACK sips:39itp34klkd@chicago.example.com;gr SIP/2.0
Via: SIP/2.0/TLS chicago.example.com:5061
 ;branch=z9hG4bKadfe4kU3
To: Carol <sips:39itp34klkd@chicago.example.com>;tag=ff3a
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=3461
Call-ID: 9435674543@atlanta.example.com
CSeq: 1 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
```

```
   Supported: replaces
   Content-Length: 0

   /* Carol then disconnects from Bob. */


   F22 BYE Carol -> Bob

   BYE sips:bob@client.biloxi.example.com SIP/2.0
   Via: SIP/2.0/TLS client.atlanta.example.com:5061
    ;branch=z9hG4bK74bfE
   To: Bob <sips:bob@biloxi.example.com>;tag=8675309
   Max-Forwards: 70
   From: Carol <sips:carol@chicago.example.com>;tag=5f35a3
   Call-ID: sdjfdjfskdf@biloxi.example.com
   CSeq: 1 BYE
   Content-Length: 0


   F23 200 OK Bob -> Carol

   SIP/2.0 200 OK
   Via: SIP/2.0/TLS client.atlanta.example.com:5061
    ;branch=z9hG4bK74bfE
    ;received=192.0.2.123
   To: Bob <sips:bob@biloxi.example.com>;tag=8675309
   From: Carol <sips:carol@chicago.example.com>;tag=5f35a3
   Call-ID: sdjfdjfskdf@biloxi.example.com
   CSeq: 1 BYE
   Content-Length: 0


   /* Alice tells Bob that the call has been
      successfully transferred. */

   F24 NOTIFY Alice -> Bob

   NOTIFY sips:bob@client.biloxi.example.com SIP/2.0
   Via: SIP/2.0/TLS client.atlanta.example.com:5061
    ;branch=z9hG4bK74bf2N
   Max-Forwards: 70
   From: Alice <sips:alice@atlanta.example.com>;tag=1234567
   To: Bob <sips:bob@biloxi.example.com>;tag=23431
   Call-ID: 12345600@atlanta.example.com
   CSeq: 3 NOTIFY
   Event: refer
   Subscription-State: terminated;reason=noresource
```

```
Contact: <sips:alice@client.atlanta.example.com>
Content-Type: message/sipfrag
Content-Length: ...

SIP/2.0 200 OK
Via: SIP/2.0/TLS chicago.example.com:5061
 ;branch=z9hG4bKadfe4ko
 ;received=192.0.2.103
To: Carol <sips:39itp34klkd@chicago.example.com>;tag=ff3a
From: Alice <sips:alice@atlanta.example.com>;tag=3461
Call-ID: 9435674543@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:39itp34klkd@chicago.example.com>


F25 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: 12345600@atlanta.example.com
CSeq: 3 NOTIFY
Content-Length: 0

/* Bob disconnects with Alice. */


F26 BYE Bob -> Alice

BYE sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7P
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=23431
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1026 BYE
Content-Length: 0


F27 200 OK Alice -> Bob

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7P
```

```
        ;received=192.0.2.113
       From: Bob <sips:bob@biloxi.example.com>;tag=23431
       To: Alice <sips:alice@atlanta.example.com>;tag=1234567
       Call-ID: 12345600@atlanta.example.com
       CSeq: 1026 BYE
       Content-Length: 0
```

## 2.6. Transfer - Instant Messaging

```
             Alice          Bob                Carol
               |             |                   |
               |   INVITE F1 |                   |
               |------------->|                  |
               | 180 Ringing F2                  |
               |<-------------|                  |
               |  200 OK F3  |                   |
               |<-------------|                  |
               |    ACK F4   |                   |
               |------------->|                  |
               |  Both way RTP Established       |
               |<============>|                  |
               |             |     MESSAGE F5    |
               |             |------------------>|
               |             |     200 OK F6     |
               |             |<------------------|
               |             |                   |
               |       INVITE Replaces:A-B F7    |
               |<--------------------------------|
               |            200 OK F8            |
               |-------------------------------->|
               |               ACK F9            |
               |<--------------------------------|
               |   Both way RTP Established      |
               |<===============================>|
               |    BYE F10  |                   |
               |------------->|                  |
               |  200 OK F11 |                   |
               |<-------------|                  |
               |                                 |
```

In this scenario, Alice and Bob establish a session between them. Bob wants Carol to take the call and so sends an Instant Message (IM) to Carol containing Alice’s URI and an embedded Replaces header field. If Carol clicks on the URI, Carol’s SIP UA sends an INVITE to Alice, which replaces the session with Bob.

This scenario shows the use of the SIP MESSAGE [RFC3428] method to pass the URI. However, another IM protocol or other method could have been used to pass the URI from Bob to Carol.

Message Details

```
F1 INVITE Alice -> Bob
INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:a8342043f@atlanta.example.com;gr>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F2 180 Ringing Bob -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=3145678
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Content-Length: 0
```

F3 200 OK Bob -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=3145678
Call-ID: 12345600@atlanta.example.com
Contact: <sips:bob@client.biloxi.example.com>
CSeq: 1 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F4 ACK Alice -> Bob

```
ACK sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74r
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=3145678
Call-ID: 12345600@atlanta.example.com
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
CSeq: 1 ACK
Content-Length: 0
```

/* Bob IMs Carol. */

F5 MESSAGE Bob -> Carol

```
MESSAGE sips:carol@chicago.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnash
Max-Forwards: 70
```

```
      From: Bob <sips:bob@biloxi.example.com>;tag=8675309
      To: Carol <sips:carol@chicago.example.com>
      Call-ID: sdjfdjfskdf@biloxi.example.com
      CSeq: 42 MESSAGE
      Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE
      Supported: replaces
      Content-Type: text/html
      Content-Length: ...

      <HTML>Do you want to take this call from
    <allOneLine>
      <A HREF="sips:a8342043f@atlanta.example.com;gr?Replaces=
      12345600@atlanta.example.com%3Bto-tag%3D3145678
      %3Bfrom-tag%3D1234567&Require=replaces">
      Alice</A>?
    </allOneLine>
       </HTML>


      F6 200 OK Carol -> Bob

      SIP/2.0 200 OK
      Via: SIP/2.0/TLS client.biloxi.example.com:5061
       ;branch=z9hG4bKnash
       ;received=192.0.2.113
      From: Bob <sips:bob@biloxi.example.com>;tag=8675309
      To: Carol <sips:carol@chicago.example.com>;tag=5f35a3
      Call-ID: sdjfdjfskdf@biloxi.example.com
      CSeq: 42 MESSAGE
      Contact: <sips:carol@client.chicago.example.com>
      Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE
      Supported: replaces
      Content-Length: 0

      /* Carol takes the call from Bob. */


      F7 INVITE  Carol -> Alice

      INVITE sips:a8342043f@atlanta.example.com;gr SIP/2.0
      Via: SIP/2.0/TLS pc.biloxi.example.com:5061
       ;branch=z9hG4bK74HH
      Max-Forwards: 70
      From: Carol <sips:carol@chicago.example.com>;tag=8675310
      To: Alice <sips:a8342043f@atlanta.example.com;gr>
      Call-ID: 563456212@b2.chicago.example.com
      CSeq: 1 INVITE
      Require: replaces
```

```
Replaces: 12345600@atlanta.example.com
 ;to-tag=3145678;from-tag=1234567
Contact: <sips:carol@client.chicago.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=carol 2890843122 2890843122 IN IP4 client.chicago.example.com
s=
c=IN IP4 client.chicago.example.com
t=0 0
m=audio 5342 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* Alice matches the dialog information in the
   Replaces header and accepts the INVITE. */


F8 200 OK Alice -> Carol

SIP/2.0 200 OK
Via: SIP/2.0/TLS pc.biloxi.example.com:5061
 ;branch=z9hG4bK74HH
 ;received=192.0.2.114
From: Carol <sips:carol@chicago.example.com>;tag=8675310
To: Alice <sips:a8342043f@atlanta.example.com;gr>;tag=131256
Call-ID: 563456212@b2.chicago.example.com
CSeq: 1 INVITE
Contact: <sips:a8342043f@atlanta.example.com;gr>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 289084543 289084543 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

```
F9 ACK  Carol -> Alice

ACK sips:a8342043f@atlanta.example.com;gr SIP/2.0
Via: SIP/2.0/TLS b2.biloxi.example.com:5061
 ;branch=z9hG4bK7435
Max-Forwards: 70
From: Carol <sips:carol@chicago.example.com>;tag=8675310
To: Alice <sips:a8342043f@atlanta.example.com;gr>;tag=131256
Call-ID: 563456212@b2.chicago.example.com
CSeq: 1 ACK
Content-Length: 0


/* RTP streams are established between Alice and Carol.
   Alice hangs up with Bob due to the Replaces header field. */


F10 BYE Alice -> Bob

BYE sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=3145678
Call-ID: 12345600@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0


F11 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=3145678
Call-ID: 12345600@atlanta.example.com
CSeq:  2 BYE
Content-Length: 0
```

## 2.7. Call Forwarding Unconditional

```
            Alice           Proxy          Gateway
              |                |              |
              |    INVITE F1   |              |
              |--------------->|              |
              |(100 Trying) F2 |              |
              |<---------------|              |
              | (181 Call Is Being Forwarded) F3
              |<---------------|   INVITE F4  |
              |                |------------->|
              |                |180 Ringing F5|
              | 180 Ringing F6 |<-------------|
              |<---------------|  200 OK F7   |
              |    200 OK F8   |<-------------|
              |<---------------|              |
              |     ACK F9     |              |
              |--------------->|    ACK F10   |
              |                |------------->|
              |    Both way RTP Established   |
              |<=============================>|
              |     BYE F11    |              |
              |--------------->|   BYE F12    |
              |                |------------->|
              |                |   200 OK F13 |
              |    200 OK F14  |<-------------|
              |<---------------|              |
              |                |              |
```

Bob wants all calls forwarded to the Public Switched Telephone Network (PSTN) (which is just another URI to the proxy server). Alice calls Bob. The proxy server rewrites the Request URI, and forwards the INVITE to a Gateway. Details of messaging behind the Gateway are not shown.

Note that the 181 Call is Being Forwarded response is shown as sent by the proxy. Strictly speaking, the proxy is behaving as a user agent in this case as a proxy cannot generate non-100 provisional responses.

Note also that forwarding could be accomplished using a redirect (302 Moved Temporarily response).

Message Details

```
F1 INVITE Alice -> Proxy
INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F2 (100 Trying) Proxy -> Alice

SIP/2.0 100 Trying
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0


F3 (181 Call is Being Forwarded) Proxy -> Alice

SIP/2.0 181 Call is Being Forwarded
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=9214d
```

```
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0

/* Proxy forwards call by rewriting Request-URI. */


F4 INVITE Proxy -> Gateway

INVITE sips:+19727293660@gw1.example.com;user=phone SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F5 180 Ringing Gateway -> Proxy

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345600@atlanta.example.com
```

```
   CSeq: 1 INVITE
   Contact: <sips:+19727293660@gw1.example.com;user=phone>
   Content Length:0


   F6 180 Ringing Proxy -> Alice

   SIP/2.0 180 Ringing
   Via: SIP/2.0/TLS client.atlanta.example.com:5061
    ;branch=z9hG4bK74bf9
    ;received=192.0.2.103
   Record-Route: <sips:ss1.example.com;lr>
   From: Alice <sips:alice@atlanta.example.com>;tag=1234567
   To: Bob <sips:bob@biloxi.example.com>;tag=314159
   Call-ID: 12345600@atlanta.example.com
   CSeq: 1 INVITE
   Contact: <sips:+19727293660@gw1.example.com;user=phone>
   Content Length: 0


   F7 200 OK Gateway -> Proxy

   SIP/2.0 200 OK
   Via: SIP/2.0/TLS ss1.example.com:5061
    ;branch=z9hG4bK83749.1
    ;received=192.0.2.54
   Via: SIP/2.0/TLS client.atlanta.example.com:5061
    ;branch=z9hG4bK74bf9
    ;received=192.0.2.103
   Record-Route: <sips:ss1.example.com;lr>
   From: Alice <sips:alice@atlanta.example.com>;tag=1234567
   To: Bob <sips:bob@biloxi.example.com>;tag=314159
   Call-ID: 12345600@atlanta.example.com
   CSeq: 1 INVITE
   Contact: <sips:+19727293660@gw1.example.com;user=phone>
   Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
   Content-Type: application/sdp
   Content-Length: ...

   v=0
   o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.example.com
   s=
   c=IN IP4 gatewayone.example.com
   t=0 0
   m=audio 3456 RTP/AVP 0
   a=rtpmap:0 PCMU/8000
```

F8 200 OK Proxy -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:+19727293660@gw1.example.com;user=phone>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: ...

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.example.com
s=
c=IN IP4 gatewayone.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F9 ACK Alice -> Proxy

```
ACK sips:+19727293660@gw1.example.com;user=phone SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf31
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0
```

F10 ACK Proxy -> Gateway

```
ACK sips:+19727293660@gw1.example.com;user=phone SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749ws.1
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf31
 ;received=192.0.2.103
Max-Forwards: 69
```

```
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0


F11 BYE Alice -> Proxy

BYE sips:+19727293660@gw1.example.com;user=phone SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bfJe
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345600@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0


F12 BYE Proxy -> Gateway

BYE sips:+19727293660@gw1.example.com;user=phone SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749G1
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bfJe
 ;received=192.0.2.103
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345600@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0


F13 200 OK Gateway -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749G1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bfJe
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
```

```
Call-ID: 12345600@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0


F14 200 OK Proxy -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bfJe
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345600@atlanta.example.com
CSeq:  2 BYE
Content-Length: 0
```

## 2.8. Call Forwarding - Busy

```
           Alice            Proxy          User B1             User B2
             |                |              |                   |
             |    INVITE F1   |              |                   |
             |--------------->|   INVITE F2  |                   |
             |                |------------->|                   |
             |(100 Trying) F3 |              |                   |
             |<---------------|  486 Busy F4 |                   |
             |                |<-------------|                   |
             |                |     ACK F5   |                   |
             |                |------------->|                   |
             |(181 Call is Being Forwarded) F6                   |
             |<---------------|              |    INVITE F7      |
             |                |--------------------------------->|
             |                |              | 180 Ringing F8    |
             |                |<---------------------------------|
             | 180 Ringing F9 |              |                   |
             |<---------------|              |      200 OK F10   |
             |                |<---------------------------------|
             |    200 OK F11  |              |                   |
             |<---------------|              |                   |
             |     ACK F12    |              |                   |
             |--------------->|              |     ACK F13       |
             |                |--------------------------------->|
             |               Both way RTP Established            |
             |<=================================================>|
             |    BYE F14     |              |                   |
             |--------------->|              |      BYE F15      |
             |                |--------------------------------->|
             |                |              |    200 OK F16     |
             |  200 OK F17    |<---------------------------------|
             |<---------------|              |                   |
             |                |              |                   |
```

Bob wants calls to B1 forwarded to B2 if B1 is busy (this information is known to the proxy). Alice calls B1, B1 is busy, the proxy server places call to B2.

Message Details

F1 INVITE Alice -> Proxy

```
INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
Max-Forwards: 70
```

```
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F2 INVITE Proxy -> B1

```
INVITE sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F3 (100 Trying) Proxy -> Alice

```
SIP/2.0 100 Trying
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0
```

F4 486 Busy Here B1 -> Proxy

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=765432
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0
```

F5 ACK Proxy -> B1

```
ACK sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=765432
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0
```

F6 (181 Call is Being Forwarded) Proxy -> Alice

```
SIP/2.0 181 Call is Being Forwarded
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
```

```
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=9214d
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0

/* The proxy now forwards the call to B2. */


F7 INVITE Proxy -> B2

INVITE sips:bob@client2.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.2
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F8 180 Ringing B2 -> Proxy

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.2
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
```

```
   To: Bob <sips:bob@biloxi.example.com>;tag=7654321
   Call-ID: 12345600@atlanta.example.com
   CSeq: 1 INVITE
   Contact: <sips:bob@client2.biloxi.example.com>
   Content-Length: 0


   F9 180 Ringing Proxy -> Alice

   SIP/2.0 180 Ringing
   Via: SIP/2.0/TLS client.atlanta.example.com:5061
    ;branch=z9hG4bK74bf9
    ;received=192.0.2.103
   Record-Route: <sips:ss1.example.com;lr>
   From: Alice <sips:alice@atlanta.example.com>;tag=1234567
   To: Bob <sips:bob@biloxi.example.com>;tag=7654321
   Call-ID: 12345600@atlanta.example.com
   CSeq: 1 INVITE
   Contact: <sips:bob@client2.biloxi.example.com>
   Content-Length: 0


   F10 200 OK B2 -> Proxy

   SIP/2.0 200 OK
   Via: SIP/2.0/TLS ss1.example.com:5061
    ;branch=z9hG4bK83749.2
    ;received=192.0.2.54
   Via: SIP/2.0/TLS client.atlanta.example.com:5061
    ;branch=z9hG4bK74bf9
    ;received=192.0.2.103
   Record-Route: <sips:ss1.example.com;lr>
   From: Alice <sips:alice@atlanta.example.com>;tag=1234567
   To: Bob <sips:bob@biloxi.example.com>;tag=7654321
   Call-ID: 12345600@atlanta.example.com
   CSeq: 1 INVITE
   Contact: <sips:bob@client2.biloxi.example.com>
   Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
   Content-Type: application/sdp
   Content-Length: ...

   v=0
   o=bob 2890844527 2890844527 IN IP4 client2.biloxi.example.com
   s=
   c=IN IP4 client2.biloxi.example.com
   t=0 0
   m=audio 3456 RTP/AVP 0
   a=rtpmap:0 PCMU/8000
```

F11 200 OK Proxy -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=7654321
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client2.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844527 IN IP4 client2.biloxi.example.com
s=
c=IN IP4 client2.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F12 ACK Alice -> Proxy

```
ACK sips:bob@client2.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bfX
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=7654321
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0
```

F13 ACK Proxy -> B2

```
ACK sips:bob@client2.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83731
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bfX
 ;received=192.0.2.103
Max-Forwards: 69
```

```
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=7654321
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between Alice and B2. */

/* Alice eventually hangs up with User B2. */


F14 BYE Alice -> Proxy

BYE sips:bob@client2.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bW4
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=7654321
Call-ID: 12345600@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0


F15 BYE Proxy -> B2

BYE sips:bob@client2.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK837493
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bW4
 ;received=192.0.2.103
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=7654321
Call-ID: 12345600@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0


F16 200 OK B2 -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK837493
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
```

```
 ;branch=z9hG4bK74bW4
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=7654321
Call-ID: 12345600@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0
```

F17 200 OK Proxy -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bW4
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=7654321
Call-ID: 12345600@atlanta.example.com
CSeq:  2 BYE
Content-Length: 0
```

## 2.9. Call Forwarding - No Answer

```
           Alice            Proxy         User B1           User B2
             |                 |              |                   |
             |    INVITE F1    |              |                   |
             |---------------->|   INVITE F2  |                   |
             |(100 Trying) F3  |------------->|                   |
             |<----------------|180 Ringing F4|                   |
             | 180 Ringing F5  |<-------------|                   |
             |<----------------|              |                   |
             |                  Request Timeout                   |
             |                 |              |                   |
             |                 |   CANCEL F6  |                   |
             |                 |------------->|                   |
             |                 |   200 OK F7  |                   |
             |                 |<-------------|                   |
             |                 |     487 F8   |                   |
             |                 |<-------------|                   |
             |                 |     ACK F9   |                   |
             |                 |------------->|                   |
             |(181 Call is Being Forwarded) F10                   |
             |<----------------|              |    INVITE F11     |
             |                 |--------------------------------->|
             |                 |              | 180 Ringing F12   |
             | 180 Ringing F13|<----------------------------------|
             |<----------------|              |     200 OK F14    |
             |                 |<---------------------------------|
             |   200 OK F15    |              |                   |
             |<----------------|              |                   |
             |     ACK F16     |              |                   |
             |---------------->|              |    ACK F17        |
             |                 |--------------------------------->|
             |               Both way RTP Established             |
             |<==================================================>|
             |     BYE F18     |              |                   |
             |---------------->|              |     BYE F19       |
             |                 |--------------------------------->|
             |                 |              |    200 OK F20     |
             |   200 OK F21    |<---------------------------------|
             |<----------------|              |                   |
             |                 |              |                   |
```

Bob wants calls to B1 forwarded to B2 if B1 is not answered (information is known to the proxy server). Alice calls B1 and no one answers. The proxy server then places the call to B2.

Message Details

F1 INVITE Alice -> Proxy

```
INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F2 INVITE Proxy -> B1

```
INVITE sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
```

```
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F3 (100 Trying) Proxy -> Alice

SIP/2.0 100 Trying
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0


F4 180 Ringing B1 -> Proxy

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=3145678
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Content-Length: 0


F5 180 Ringing Proxy -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=3145678
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
```

```
Contact: <sips:bob@client.biloxi.example.com>
Content-Length: 0

/* B1 rings until a configurable timer expires in the proxy.  The
   proxy sends Cancel and proceeds down the list of routes. */


F6 CANCEL Proxy -> B1

CANCEL sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 CANCEL
Content-Length: 0


F7 200 OK B1 -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
 ;received=192.0.2.54
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=329d823
Call-ID: 12345600@atlanta.example.com
CSeq: 1 CANCEL
Content-Length: 0


F8 487 Request Terminated B1 -> Proxy

SIP/2.0 487 Request Terminated
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=3145678
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0
```

F9 ACK Proxy -> B1

```
ACK sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=3145678
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0
```

F10 (181 Call is Being Forwarded) Proxy -> Alice

```
SIP/2.0 181 Call is Being Forwarded
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=9214d
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0
```

F11 INVITE Proxy -> B2

```
INVITE sips:bob@client2.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.2
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
```

```
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F12 180 Ringing B2 -> Proxy

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.2
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=765432
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client2.biloxi.example.com>
Content-Length: 0


F13 180 Proxy -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=765432
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client2.biloxi.example.com>
Content-Length: 0


F14 200 OK B2 -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.2
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
```

```
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=765432
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client2.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844527 IN IP4 client2.biloxi.example.com
s=
c=IN IP4 client2.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F15 200 OK Proxy -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=765432
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client2.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844527 IN IP4 client2.biloxi.example.com
s=
c=IN IP4 client2.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F16 ACK Alice -> Proxy

```
ACK sips:bob@client2.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
```

```
 ;branch=z9hG4bK74bf3
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=765432
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0


F17 ACK Proxy -> B2

ACK sips:bob@client2.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK8374.1
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf3
 ;received=192.0.2.103
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=765432
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between Alice and B2.
   Alice hangs up with User B2. */


F18 BYE Alice -> Proxy

BYE sips:bob@client2.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74b3f
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=765432
Call-ID: 12345600@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0


F19 BYE Proxy -> B2

BYE sips:bob@client2.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK837.1
```

```
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74b3f
 ;received=192.0.2.103
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=765432
Call-ID: 12345600@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0
```

F20 200 OK B2 -> Proxy

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK837.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74b3f
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=765432
Call-ID: 12345600@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0
```

F21 200 OK Proxy -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74b3f
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=765432
Call-ID: 12345600@atlanta.example.com
CSeq:  2 BYE
Content-Length: 0
```

## 2.10. 3-Way Conference - Third Party Is Added

```
            Alice           Bob             Carol
             |    INVITE F1   |               |
             |--------------->|               |
             | 180 Ringing F2 |               |
             |<---------------|               |
             |    200 OK F3   |               |
             |<---------------|               |
             |     ACK F4     |               |
             |--------------->|               |
             |       RTP      |               |
             |<==============>|               |
             |   INVITE F5    |               |
             |<---------------|               |
             |    200 OK F6   |               |
             |--------------->|               |
             |     ACK F7     |               |
             |<---------------|  INVITE F8    |
             |                |-------------->|
             |                |    180 F9     |
             |                |<--------------|
             |                |  200 OK F10   |
             |                |<--------------|
             |                |  ACK F11      |
             |                |-------------->|
             |                |       RTP     |
             |                |<=============>|
```

In this scenario, Alice and Bob are in a 2-party call (session) when Bob wishes to add Carol into the conversation. Bob is capable of media mixing in a 3-party call. Bob first sends a re-INVITE to Alice, changing Contact URIs to one that indicates Bob's mixer and acts like a focus. As a result, Bob includes the "isfocus" feature tag [RFC3840] as described in [RFC4579]. Bob then INVITEs Carol using the same Contact URI. Note that Bob could wait to re-INVITE Alice until after Carol has answered. Bob could also put Alice on hold before calling Carol.

Message Details

F1 INVITE Alice -> Bob

```
INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
```

```
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F2 180 Ringing Bob -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:b54gh42f5@biloxi.example.com>
Content-Length: 0


F3 200 OK Bob -> Alice

SIP/2.0 200 OK

Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:b54gh42f5@biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
```

```
Supported: replaces, gruu
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F4 ACK Alice -> Bob

ACK sips:b54gh42f5@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bfL
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

/* Alice and Bob have established a session.
   Bob re-INVITEs, changing Contact URIs. */


F5 INVITE Bob -> Alice

INVITE sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=23431
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1024 INVITE
Contact: <sips:bob-Mixer@client.biloxi.example.com>;isfocus
Content-Type: application/sdp
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu
Content-Length: ...

v=0
o=bob 2890844527 2890844528 IN IP4 client.biloxi.example.com
s=
```

```
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F6 200 OK Alice -> Bob

SIP/2.0 200 OK

Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.113
From: Bob <sips:bob@biloxi.example.com>;tag=23431
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1024 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F7 ACK Bob -> Alice

ACK sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnash3G
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=23431
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1024 ACK
Content-Length: 0

/* Bob calls Carol. */
```

```
F8 INVITE Bob -> Carol

INVITE sips:carol@chicago.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashJfd
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>
Call-ID: sdjfdjfskdf@biloxi.example.com
CSeq: 42 INVITE
Contact: <sips:bob-Mixer@client.biloxi.example.com>;isfocus
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 28908445834 2890844834 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 48174 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F9 180 Ringing Carol -> Bob

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashJfd
 ;received=192.0.2.113
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=341313
Call-ID: sdjfdjfskdf@biloxi.example.com
CSeq: 42 INVITE
Contact: <sips:carol@client.chicago.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: 0


F10 200 OK Carol -> Bob

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashJfd
 ;received=192.0.2.113
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
```

```
To: Carol <sips:carol@chicago.example.com>;tag=341313
Call-ID: sdjfdjfskdf@biloxi.example.com
CSeq: 42 INVITE
Contact: <sips:carol@client.chicago.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=carol 2890844922 2890844922 IN IP4 client.chicago.example.com
s=
c=IN IP4 client.chicago.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F11 ACK Bob -> Carol

```
ACK sips:carol@client.chicago.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnash431
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=341313
Call-ID: sdjfdjfskdf@biloxi.example.com
CSeq: 42 ACK
Content-Length: 0
```

```
/* Bob's mixer now mixes media from both Alice and Carol
   to create the 3-way conference. */
```

## 2.11. 3-Way Conference - Third Party Joins

```
          Alice           Bob            Carol
            |    INVITE F1   |              |
            |--------------->|              |
            | 180 Ringing F2 |              |
            |<---------------|              |
            |    200 OK F3   |              |
            |<---------------|              |
            |     ACK F4     |              |
            |--------------->|              |
            |       RTP      |              |
            |<==============>|  INVITE Join:A-B F5
            |                |<-------------|
            |                |    180 F6    |
            |                |------------->|
            |   INVITE F7    |              |
            |<---------------|              |
            |    200 OK F8   |              |
            |--------------->|              |
            |     ACK F9     |              |
            |<---------------|              |
            |                |  200 OK F10  |
            |                |------------->|
            |                |  ACK F11     |
            |                |<-------------|
            |                |      RTP     |
            |                |<============>|
```

In this scenario, Alice and Bob are in a 2-party call and Carol wishes to join, resulting in a 3-party call. Carol could have learned Bob's dialog identifier using some non-SIP means, or possibly from a NOTIFY with the dialog package sent by Bob. Carol sends an INVITE to Bob containing a Join header identifying the dialog between Alice and Bob. Bob re-INVITEs Alice to switch to focus mode and includes the "isfocus" feature tag [RFC3840] as described in [RFC4579]. Bob then accepts the INVITE from Carol, resulting in the 3-way call.

Message Details

F1 INVITE Alice -> Bob

```
INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
```

```
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F2 180 Ringing Bob -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:b54gh42f5@biloxi.example.com>
Content-Length: 0


F3 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:b54gh42f5@biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
```

```
Supported: replaces, join, gruu
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F4 ACK Alice -> Bob

ACK sips:b54gh42f5@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf6
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

/* Alice and Bob have established a session.
   Carol requests to join the session. */


F5 INVITE Carol -> Bob

INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS chicago.example.com:5061
 ;branch=z9hG4bKnashds7
Max-Forwards: 70
From: Carol <sips:carol@chicago.example.com>;tag=8675309
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 452k499sk@chicago.example.com
CSeq: 99 INVITE
Contact: <sips:carol@client.chicago.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, join
Join: 12345600@atlanta.example.com;from-tag=1234567;to-tag=23431
Content-Type: application/sdp
Content-Length: ...
```

```
v=0
o=carol 2890844922 2890844922 IN IP4 client.chicago.example.com
s=
c=IN IP4 client.chicago.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F6 180 Ringing Bob -> Carol

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS chicago.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=120.
From: Carol <sips:carol@chicago.example.com>;tag=8675309
To: Bob <sips:bob@biloxi.example.com>;tag=0982
Call-ID: 452k499sk@chicago.example.com
CSeq: 99 INVITE
Contact: <sips:bob-Mixer@client.biloxi.example.com>;isfocus
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Length: 0


F7 INVITE Bob -> Alice

INVITE sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashdyKL
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=23431
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1024 INVITE
Contact: <sips:bob-Mixer@client.biloxi.example.com>;isfocus
Content-Type: application/sdp
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, join, gruu
Content-Length: ...

v=0
o=bob 2890844527 2890844528 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

```
F8 200 OK Alice -> Bob
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashdyKL
 ;received=192.0.2.113
From: Bob <sips:bob@biloxi.example.com>;tag=23431
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1024 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F9 ACK Bob -> Alice

ACK sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnash3g
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=23431
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1024 ACK
Content-Length: 0


F10 200 OK Bob -> Carol

SIP/2.0 200 OK
Via: SIP/2.0/TLS chicago.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=120.
From: Carol <sips:carol@chicago.example.com>;tag=8675309
To: Bob <sips:bob@biloxi.example.com>;tag=0982
Call-ID: 452k499sk@chicago.example.com
CSeq: 99 INVITE
Contact: <sips:bob-Mixer@client.biloxi.example.com>;isfocus
```

```
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, join, gruu
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 28908445834 2890844834 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 48174 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F11 ACK OK Carol -> Bob

```
ACK sips:bob-Mixer@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS chicago.example.com:5061
 ;branch=z9hG4bKnash4Gf
Max-Forwards: 70
From: Carol <sips:carol@chicago.example.com>;tag=8675309
To: Bob <sips:bob@biloxi.example.com>;tag=0982
Call-ID: 452k499sk@chicago.example.com
CSeq: 99 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, join
Content-Length:0
```

### 2.12. Find-Me

```
             Alice           Proxy           User   User   User   User
                                              B1     B2     B3     B4
             |                |              |      |      |      |
             |    INVITE F1   |              |      |      |      |
             |--------------->|   INVITE F2  |      |      |      |
             |                |------------->|      |      |      |
             |(100 Trying) F3 |              |      |      |      |
             |<---------------|180 Ringing F4|      |      |      |
             |                |<-------------|      |      |      |
             | 180 Ringing F5 |              |      |      |      |
             |<---------------|              |      |      |      |
             |                |    Timeout   |      |      |      |
             |                |              |      |      |      |
             |                |   CANCEL F6  |      |      |      |
             |                |------------->|      |      |      |
             |                |   200 OK F7  |      |      |      |
             |                |<-------------|      |      |      |
             |                |     487 F8   |      |      |      |
             |                |<-------------|      |      |      |
             |                |     ACK F9   |      |      |      |
             |                |------------->|      |      |      |
             |                |     INVITE F10      |      |      |
             |                |-------------------->|      |      |
             |                |480 Not Logged In F11|      |      |
             |                |<--------------------|      |      |
             |                |              ACK F12|      |      |
             |                |-------------------->|      |      |
             |                |          INVITE F13        |      |
             |                |--------------------------->|      |
             |                |      486 Busy Here F14     |      |
             |                |<---------------------------|      |
             |                |            ACK F15         |      |
             |                |--------------------------->|      |
             |                |              INVITE F16           |
             |                |---------------------------------->|
             |                |            180 Ringing F17        |
             |     180 F18    |<----------------------------------|
             |<---------------|               200 OK F19          |
             |   200 OK F20   |<----------------------------------|
             |<---------------|                                   |
             |     ACK F21    |                                   |
             |--------------->|                 ACK F22           |
             |                |---------------------------------->|
             |               Both way RTP Established             |
             |<==================================================>|
```

```
             |                |                BYE F23           |
             |     BYE F24    |<---------------------------------|
             |<---------------|                                  |
             |    200 OK F25  |                                  |
             |--------------->|              200 OK F26          |
             |                |--------------------------------->|
             |                |                                  |
```

Alice’s call to Bob will result in an attempt to locate Bob by calling locations from a list of contacts. The location to answer the call becomes the active set; no other sets may join the call.

While this flow shows a sequential search, the search could be accomplished using parallel forking.

Message Details

F1 INVITE Alice -> Proxy

```
INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t= 0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F2 INVITE Proxy -> B1

```
INVITE sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
```

```
Record-Route: <sips:ss1.example.com;lr>
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t= 0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F3 (100 Trying) Proxy -> Alice

SIP/2.0 100 Trying
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0


F4 180 Ringing B1 -> Proxy

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=765432
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Content-Length: 0
```

F5 180 Ringing Proxy -> Alice

```
SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=765432
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Content-Length: 0
```

/* B1 rings until a configurable timer in the proxy expires. The proxy then sends Cancel and proceeds down the list of routes. */

F6 CANCEL Proxy -> B1

```
CANCEL sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 CANCEL
Content-Length: 0
```

F7 200 OK B1 -> Proxy

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
 ;received=192.0.2.54
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=765432
Call-ID: 12345600@atlanta.example.com
CSeq: 1 CANCEL
Content-Length: 0
```

F8 487 Request Terminated B1 -> Proxy

```
SIP/2.0 487 Request Terminated
```

```
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=765432
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0
```

F9 ACK Proxy -> B1

```
ACK sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.1
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=765432
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0
```

F10 INVITE Proxy -> B2

```
INVITE sips:bob@client2.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.2
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
```

```
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F11 480 Not Logged In B2 -> Proxy

SIP/2.0 480 Not Logged In
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.2
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314756
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0


F12 ACK Proxy -> B2

ACK sips:bob@client2.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.2
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314756
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0


F13 INVITE Proxy -> B3

INVITE sips:bob@client3.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.3
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
```

```
Contact: <sips:alice@client.atlanta.example.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F14 486 Busy Here B3 -> Proxy

SIP/2.0 486 Busy Here
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.3
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=7654321
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0


F15 ACK Proxy -> B3

ACK sips:bob@client3.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.3
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=7654321
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0


F16 INVITE Proxy -> B4

INVITE sips:bob@client4.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.4
Via: SIP/2.0/TLS client.atlanta.example.com:5061
```

```
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F17 180 Ringing B4 -> Proxy

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.4
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=7137136
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client4.biloxi.example.com>
Content-Length: 0


F18 180 Ringing Proxy -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=7137136
```

```
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client4.biloxi.example.com>
Content-Length: 0


F19 200 OK B4 -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83749.4
 ;received=192.0.2.54
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=7137136
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client4.biloxi.example.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844527 IN IP4 client4.biloxi.example.com
s=
c=IN IP4 client4.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F20 200 OK Proxy -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
Record-Route: <sips:ss1.example.com;lr>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=7137136
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client4.biloxi.example.com>
Content-Type: application/sdp
Content-Length: ...
```

```
v=0
o=bob 2890844527 2890844527 IN IP4 client4.biloxi.example.com
s=
c=IN IP4 client4.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F21 ACK Alice -> Proxy

ACK sips:bob@client4.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=7137136
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0


F22 ACK Proxy -> B4

ACK sips:bob@client4.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK8374
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf
 ;received=192.0.2.103
Max-Forwards: 69
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=7137136
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between Alice and B4. */

/* User B4 hangs up with Alice. */


F23 BYE B4 -> Proxy

BYE sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client4.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
```

```
Route: <sips:ss1.example.com;lr>
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=7137136
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1 BYE
Content-Length: 0
```

F24 BYE Proxy -> Alice

```
BYE sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83754
Via: SIP/2.0/TLS client4.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.105
Max-Forwards: 69
From: Bob <sips:bob@biloxi.example.com>;tag=7137136
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1 BYE
Content-Length: 0
```

F25 200 OK Alice -> Proxy

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.example.com:5061
 ;branch=z9hG4bK83754
 ;received=192.0.2.54
Via: SIP/2.0/TLS client4.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.105
From: Bob <sips:bob@biloxi.example.com>;tag=7137136
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1 BYE
Content-Length: 0
```

F26 200 OK Proxy -> B4

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client4.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.105
From: Bob <sips:bob@biloxi.example.com>;tag=7137136
```

```
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 1 BYE
Content-Length: 0
```

## 2.13. Call Management (Incoming Call Screening)

```
                                                  Announcement
           Alice            Proxy             Bob           Server

             |                |               |               |
             |    INVITE F1   |               |               |
             |------------------------------->|               |
             | 305 Use Proxy F2               |               |
             |<-------------------------------|               |
             |     ACK F3     |               |               |
             |------------------------------->|               |
             |    INVITE F4   |               |               |
             |--------------->|               |               |
             | 407 Proxy Authentication F5    |               |
             |<---------------|               |               |
             |     ACK F6     |               |               |
             |--------------->|               |               |
             |    INVITE F7   |               |               |
             |--------------->|               |               |
             | 403 Screening Failure (Terminating) Error-Info: URI F8
             |<---------------|               |               |
             |     ACK F9     |               |               |
             |--------------->|               |               |
             |    INVITE F10  |               |               |
             |----------------------------------------------->|
             |    200 OK F11  |               |               |
             |----------------------------------------------->|
             |     ACK F12    |                               |
             |----------------------------------------------->|
             |          Announcement Played to Caller         |
             |<===============================================|
             |                                     BYE F13    |
             |<-----------------------------------------------|
             |     200 OK F14                                 |
             |----------------------------------------------->|
             |                                                |
             |                                                |
             |                                                |
```

Bob has an incoming call screening list; Alice is included on the list of addresses from which Bob will not accept calls. Alice attempts to call Bob. Messages F1, F2, and F3 are included to show that Bob does not accept INVITEs that have not been screened by the proxy.

Note that call screening cannot be done using the From header -- instead some form of authentication credentials must be used.

The screening proxy inserts an announcement URI in an Error-Info header field, which Alice accesses by sending an INVITE to listen to the Announcement. The Announcement Server uses the automaton and rendering feature tags in F12 and F13 to indicate that it is a media server only capable of playing announcements.

Message Details

F1 INVITE Alice -> Bob

```
INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* Bob only accepts INVITEs that have been screened
   by the proxy. */

F2 305 Use Proxy Bob -> Alice

```
SIP/2.0 305 Use Proxy
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=342123
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:ss1.example.com>
Content-Length: 0
```

F3 ACK Alice -> Bob

```
ACK sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=342123
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0
```

/* A retries the call through the proxy. */

F4 INVITE Alice -> Proxy 1

```
INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf0
Max-Forwards: 70
Route: <sips:ss1.example.com>
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 2 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* Proxy 1 challenges Alice for authentication. */

F5 407 Proxy Authentication Required Proxy 1 -> Alice

```
SIP/2.0 407 Proxy Authentication Required
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf0
 ;received=192.0.2.103
```

```
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=7886765
Call-ID: 12345600@atlanta.example.com
CSeq: 2 INVITE
Proxy-Authenticate: Digest realm="example.com",
   nonce="ea9c8e88df84f1cec4341ae6cbe5a359",
   qop="auth", nc=00000001, cnonce="0a4f113b",
   opaque="", stale=FALSE, algorithm=MD5
Content-Length: 0


F6 ACK Alice -> Proxy 1

ACK sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf0
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=7886765
Call-ID: 12345600@atlanta.example.com
CSeq: 2 ACK
Content-Length: 0

/* Alice responds by sending an INVITE with authentication
   credentials in it. */


F7 INVITE Alice -> Proxy 1

INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf2
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 3 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Proxy-Authorization: Digest username="alice",
   realm="example.com", qop=auth,
   nc=00000001, cnonce="4gr84543ft2",
   nonce="ae9137be1c87d175c2dd63302a0d6e0a",
   opaque="", uri="sips:bob@biloxi.example.com",
   response="bbaec39f943bdcb3620d90afc548a45c"
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: ...
```

```
v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F8 403 Screening Failure (Terminating) Proxy 1 -> Alice

SIP/2.0 403 Screening Failure (Terminating)
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf2
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=ffe254
Call-ID: 12345600@atlanta.example.com
CSeq: 3 INVITE
Error-Info: <sips:screen-fail-term-ann@ms.biloxi.example.com>
Content-Length: 0


F9 ACK Alice -> Proxy 1

ACK sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf2
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=ffe254
Call-ID: 12345600@atlanta.example.com
Proxy-Authorization: Digest username="alice",
 realm="example.com", nonce="ae9137be1c87d175c2dd63302a0d6e0a",
 opaque="", uri="sips:bob@biloxi.example.com",
 response="bbaec39f943bdcb3620d90afc548a45c"
CSeq: 3 ACK
Content-Length: 0

/* To hear the recording, Alice connects to the Error-Info URI. */


F10 INVITE Alice -> Proxy 1

INVITE sips:screen-fail-term-ann@ms.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bfj
Max-Forwards: 70
```

```
   From: Alice <sips:alice@atlanta.example.com>;tag=1234567
   To: Bob <sips:bob@biloxi.example.com>
   Call-ID: 12345600@atlanta.example.com
   CSeq: 4 INVITE
   Contact: <sips:alice@client.atlanta.example.com>
   Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
   Content-Type: application/sdp
   Content-Length: ...

   v=0
   o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
   s=
   c=IN IP4 client.atlanta.example.com
   t=0 0
   m=audio 49170 RTP/AVP 0
   a=rtpmap:0 PCMU/8000
```

F11 200 OK Announcement Server -> Proxy 1

```
   SIP/2.0 200 OK
   Via: SIP/2.0/TLS client.atlanta.example.com:5061
    ;branch=z9hG4bK74bfj
    ;received=192.0.2.103
   From: Alice <sips:alice@atlanta.example.com>;tag=1234567
   To: Bob <sips:bob@biloxi.example.com>;tag=234934
   Call-ID: 12345600@atlanta.example.com
   CSeq: 4 INVITE
   Contact: <sips:ms.biloxi.example.com>
    ;automaton;+sip.rendering="no"
   Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
   Content-Type: application/sdp
   Content-Length: ...

   v=0
   o=annc 2890844543 2890844543 IN IP4 announce.biloxi.example.com
   s=
   c=IN IP4 announce.biloxi.example.com
   t=0 0
   m=audio 49174 RTP/AVP 0
   a=rtpmap:0 PCMU/8000
```

F12 ACK Alice -> Announcement Server

```
   ACK sips:ms.biloxi.example.com SIP/2.0
   Via: SIP/2.0/TLS client.atlanta.example.com:5061
    ;branch=z9hG4bK74b32
```

```
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=234934
Call-ID: 12345600@atlanta.example.com
CSeq: 4 ACK
Content-Length: 0

/* Announcement Server plays announcement then disconnects. */


F13 BYE Announcement Server -> Alice

BYE sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS announcement.example.com:5061
 ;branch=z9hG4bK74bKS
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=234934
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 2334 BYE
Content-Length: 0


F14 200 OK Alice -> Announcement Server

SIP/2.0 200 OK
Via: SIP/2.0/TLS announcement.example.com:5061
 ;branch=z9hG4bK74bKS
 ;received=192.0.2.103
From: Bob <sips:bob@biloxi.example.com>;tag=234934
To: Alice <sips:alice@atlanta.example.com>;tag=1234567
Call-ID: 12345600@atlanta.example.com
CSeq: 2334 BYE
Content-Length: 0
```

## 2.14. Call Management (Outgoing Call Screening)

```
          Alice            Proxy           Bob
            |                 |              |
            |    INVITE F1    |              |
            |---------------->|              |
            | 407 Proxy Authentication F2    |
            |<----------------|              |
            |      ACK F3     |              |
            |---------------->|              |
            |    INVITE F4    |              |
            |---------------->|              |
            | 403 Screening Failure (Originating) F5
            |<----------------|              |
            |      ACK F6     |              |
            |---------------->|              |
            |                 |              |
```

Alice has an outgoing call screening list; Bob is included on the list of addresses to which Alice will not be able to place a call. Alice attempts to call Bob.

Alice could establish a session to listen to the announcement in the Error-Info header field.

Message Details

F1 INVITE Alice -> Proxy 1

```
INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
```

```
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* Proxy 1 challenges Alice for authentication. */


F2 407 Proxy Authentication Required Proxy 1 -> Alice

SIP/2.0 407 Proxy Authentication Required
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=90210
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Proxy-Authenticate: Digest realm="example.com",
 nonce="ea9c8e88df84f1cec4341ae6cbe5a359",
 qop="auth", nc=00000001, cnonce="0a4f113b",
 opaque="", stale=FALSE, algorithm=MD5
Content-Length: 0


F3 ACK Alice -> Proxy 1

ACK sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=90210
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

/* Alice responds be sending an INVITE with authentication
   credentials in it. */


F4 INVITE Alice -> Proxy 1

INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74b4
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
```

```
CSeq: 2 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Proxy-Authorization: Digest username="alice", realm="example.com",
 nonce="cb360afc54bbaec39f943bd820d9a45c", opaque="",
 uri="sips:bob@biloxi.example.com",
 response="b9d2e5bcdec9f69ab2a9b44f270285a6"
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F5 403 Screening Failure (Originating) Proxy 1 -> Alice

```
SIP/2.0 403 Screening Failure (Originating)
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74b4
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=18017
Call-ID: 12345600@atlanta.example.com
CSeq: 2 INVITE
Error-Info: <sips:screen-fail-orig-ann@announcement.example.com>
Content-Length: 0
```

F6 ACK Alice -> Proxy 1

```
ACK sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74b4
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=18017
Call-ID: 12345600@atlanta.example.com
CSeq: 2 ACK
Proxy-Authorization: Digest username="alice", realm="example.com",
 nonce="cb360afc54bbaec39f943bd820d9a45c", opaque="",
 uri="sips:bob@biloxi.example.com",
 response="b9d2e5bcdec9f69ab2a9b44f270285a6"
Content-Length: 0
```

## 2.15.  Call Park

```
           Alice           Bob        Park Server       Carol
             |              |              |              |
             |   INVITE F1  |              |              |
             |------------->|              |              |
             |180 Ringing F2|              |              |
             |<-------------|              |              |
             |  200 OK F3   |              |              |
             |<-------------|              |              |
             |    ACK F4    |              |              |
             |------------->|              |              |
             |  RTP Media   |              |              |
             |<============>|              |              |
             |      Bob Parks Call         |              |
             |              |   REFER Refer-To: A F5      |
             |              |------------->|              |
             |              |    202 F6    |              |
             |              |<-------------|              |
             |              |   NOTIFY F7  |              |
             |              |<-------------|              |
             |              |    200 F8    |              |
             |              |------------->|              |
             |  INVITE F9 Replaces: B      |              |
             |<----------------------------|              |
             |          200 OK F10         |              |
             |---------------------------->|              |
             |           ACK F11           |              |
             |<----------------------------|              |
             |           RTP Music         |              |
             |<===========================>|              |
             |     BYE F12  |              |              |
             |------------->|  NOTIFY F14  |              |
             |  200 OK F13  |<-------------|              |
             |<-------------|  200 OK F15  |              |
             |              |------------->|              |
             |              |   Carol picks up the call   |
             |              |              |              |
             |              |              | SUBSCRIBE F16|
             |              |              |<-------------|
             |              |              |  200 OK F17  |
             |              |              |------------->|
             |              |              |  NOTIFY F18  |
             |              |              |------------->|
             |              |              |  200 OK F19  |
             |              |              |<-------------|
             |           INVITE Replaces: Park Server F20 |
             |<-------------------------------------------|
```

```
     |              |              |   200 F21    |
     |-------------------------------------------->|
     |              |              |    ACK F22   |
     |<--------------------------------------------|
     |                   RTP Media                  |
     |<===========================================>|
     |           BYE F23           |              |
     |---------------------------->|              |
     |          200 OK F24         |              |
     |<----------------------------|              |
     |       No more RTP Music     |              |
```

In this example, Alice calls Bob. Bob then parks the call at the Park Server by sending a REFER to the Park Server. The server sends an INVITE to Alice, which replaces the session between Alice and Bob. The Park Server utilizes the automaton, rendering, and byeless feature tags in F9 to indicate its capabilities to Alice. The call is accepted by Alice and causes Alice to send a BYE to Bob. Bob receives notification of the successful park, and also receives the dialog identifiers in the application/sip body of the NOTIFY response.

Carol wishes to retrieve the call, so she sends an INVITE containing the dialog identifiers to Alice, which replaces the session with the Park Server. Alice accepts the call and sends a BYE to the Park Server. Carol obtains the dialog identifiers from a NOTIFY from the Park Server.

Note that this call flow is a special case of call transfer.

Note also that this flow could also be used for Music on Hold.

Message Details

F1 INVITE Alice -> Bob

```
INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.alice.example.com:5061
 ;branch=z9hG4bKnashds7
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345601@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:a8342043f@atlanta.example.com;gr>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces, gruu
Content-Type: application/sdp
```

```
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F2 180 Ringing Bob -> Alice

```
SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.105
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345601@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Content-Length: 0
```

F3 200 OK Bob -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.105
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345601@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F4 ACK Alice -> Bob

```
ACK sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bKnashds7
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345601@atlanta.example.com
CSeq: 1 ACK
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Content-Length: 0
```

```
/* Bob REFERs Park Server to establish session with Alice,
   which replaces the established session between Alice and Bob.
   Note that there is no session established between Bob
   and the Park Server. */
```

F5 REFER Bob -> Park Server

```
REFER sips:park@server.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds9
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=02134
To: Park Server <sips:park@server.example.com>
Call-ID: 4802029847@biloxi.example.com
CSeq: 1 REFER
<allOneLine>
Refer-To: <sips:a8342043f@atlanta.example.com;gr?Replaces=
12345601%40atlanta.example.com%3Bfrom-tag%3D314159
%3Bto-tag%3D1234567&Require=replaces>
</allOneLine>
Referred-By: <sips:bob@biloxi.example.com>
Contact: <sips:bob@client.biloxi.example.com>
Content-Length: 0
```

F6 202 Accepted Park Server -> Bob

```
SIP/2.0 202 Accepted
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds9
 ;received=192.0.2.105
From: Bob <sips:bob@biloxi.example.com>;tag=02134
To: Park Server <sips:park@server.example.com>;tag=56323
```

```
   Call-ID: 4802029847@biloxi.example.com
   Contact: <sips:park@server.example.com>
   CSeq: 1 REFER
   Content-Length: 0


   F7 NOTIFY Park Server -> Bob

   NOTIFY sips:bob@client.biloxi.example.com SIP/2.0
   Via: SIP/2.0/TLS server.example.com:5061
    ;branch=z9hG4bK74bT6
   To: Bob <sips:bob@biloxi.example.com>;tag=02134
   Max-Forwards: 70
   From: Park Server <sips:park@server.example.com>;tag=56323
   Call-ID: 4802029847@biloxi.example.com
   CSeq: 1 NOTIFY
   Event: refer
   Contact: <sips:park@server.example.com>
   Subscription-State: active;expires=60
   Content-Type: message/sipfrag
   Content-Length: ...

   SIP/2.0 100 Trying


   F8 200 OK Bob -> Park Server

   SIP/2.0 200 OK
   Via: SIP/2.0/TLS server.example.com:5061
    ;branch=z9hG4bK74bT6
    ;received=192.0.2.103
   To: Bob <sips:bob@biloxi.example.com>;tag=02134
   From: Park Server <sips:park@server.example.com>;tag=56323
   Call-ID: 4802029847@biloxi.example.com
   CSeq: 1 NOTIFY
   Content-Length: 0

   /* Park Server places call to Alice to replace session
      between Alice and Bob. */


   F9 INVITE Park Server -> Alice

   INVITE sips:a8342043f@atlanta.example.com;gr SIP/2.0
   Via: SIP/2.0/TLS server.example.com:5061
    ;branch=z9hG4bK74rf
   Max-Forwards: 70
   From: <sips:park@server.example.com>;tag=0111
```

```
To: <sips:a8342043f@atlanta.example.com;gr>
Call-ID: a5-75-34-12-76@server.example.com
CSeq: 1 INVITE
Referred-By: <sips:bob@biloxi.example.com>
Contact: <sips:park@server.example.com>;automaton
    ;+sip.byeless;+sip.rendering="no"
Require: replaces
Replaces: 12345601@atlanta.example.com
 ;from-tag=314159;to-tag=1234567
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=ParkServer 2890844576 2890844576 IN IP4 Park.server.example.com
s=
c=IN IP4 server.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F10 200 OK Alice -> Park Server

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS server.example.com:5061
 ;branch=z9hG4bK74rf
 ;received=192.0.2.103
From: <sips:park@server.example.com>;tag=0111
To: <sips:a8342043f@atlanta.example.com;gr>;tag=098594
Call-ID: a5-75-34-12-76@server.example.com
CSeq: 1 INVITE
Contact: <sips:a8342043f@atlanta.example.com;gr>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces, gruu
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=recvonly
```

F11 ACK Park Server -> Alice

```
ACK sips:a8342043f@atlanta.example.com;gr SIP/2.0
Via: SIP/2.0/TLS server.example.com:5061
 ;branch=z9hG4bK7rfF
Max-Forwards: 70
From: <sips:park@server.example.com>;tag=0111
To: <sips:a8342043f@atlanta.example.com;gr>;tag=098594
Call-ID: a5-75-34-12-76@server.example.com
CSeq: 1 ACK
Content-Length: 0
```

F12 BYE Alice -> Bob

```
BYE sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bKnashds7
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345601@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0
```

F13 200 OK Bob -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.105
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 12345601@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0

/* Park Server reports success back to Bob by returning
   a 200 OK response.  Bob obtains the dialog identifiers
   from the headers included in the response. */
```

F14 NOTIFY Park Server -> Bob

```
NOTIFY sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS server.example.com:5061
 ;branch=z9hG4bK74bf9
```

```
To: Bob <sips:bob@biloxi.example.com>;tag=02134
Max-Forwards: 70
From: Park Server <sips:park@server.example.com>;tag=56323
Call-ID: 4802029847@biloxi.example.com
CSeq: 2 NOTIFY
Event: refer
Subscription-State: terminated;reason=noresource
Contact: <sips:park@server.example.com>;automaton
    ;+sip.byeless;+sip.rendering="no"
Content-Type: message/sipfrag
Content-Length: ...

SIP/2.0 200 OK
Via: SIP/2.0/TLS server.example.com:5061
 ;branch=z9hG4bK74rf
 ;received=192.0.2.103
From: <sips:park@server.example.com>;tag=0111
To: <sips:a8342043f@atlanta.example.com;gr>;tag=098594
Call-ID: a5-75-34-12-76@server.example.com
CSeq: 1 INVITE
Contact: <sips:a8342043f@atlanta.example.com;gr>


F15 200 OK Bob -> Park Server

SIP/2.0 200 OK
Via: SIP/2.0/TLS server.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
To: Bob <sips:bob@biloxi.example.com>;tag=02134
From: Park Server <sips:park@server.example.com>;tag=56323
Call-ID: 4802029847@biloxi.example.com
CSeq: 2 NOTIFY
Content-Length: 0

/* Alice is now parked at the Park Server. */

/* Carol picks up the call by sending an INVITE to A, which
   replaces the existing session with the Park Server.
   Carol needs to know the dialog information to construct
   the Replaces header. */


F16 SUBSCRIBE  Carol -> Park Server

SUBSCRIBE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.chicago.example.com:5061
 ;branch=z9hG4bK74b232
```

```
Max-Forwards: 70
From: Carol <sips:carol@chicago.example.com>;tag=158x93461
To: <sips:park@server.example.com>
Call-ID: 2d6485356dfaj34dsf
CSeq: 1 SUBSCRIBE
Contact: <sips:carol@client.chicago.example.com>
Event: dialog
Expires: 0
Accept: application/dialog-info+xml
Content-Length: 0
```

F17 200 OK Park Server -> Carol

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.chicago.example.com:5061
 ;branch=z9hG4bK74b232
 ;received=192.0.2.105
From: Carol <sips:carol@chicago.example.com>;tag=158x93461
To: <sips:park@server.example.com>;tag=3213j
Call-ID: 2d6485356dfaj34dsf
CSeq: 1 SUBSCRIBE
Contact: <sips:park@server.example.com>;automaton
    ;+sip.byeless;+sip.rendering="no"
Content-Length: 0
```

F18 NOTIFY  Park Server -> Carol

```
NOTIFY sips:carol@client.example.com SIP/2.0
Via: SIP/2.0/TLS server.example.com:5061
 ;branch=z9hG4bK74b8skd
Max-Forwards: 70
To: Carol <sips:carol@chicago.example.com>;tag=158x93461
From: <sips:park@server.example.com>;tag=3213j
Call-ID: 2d6485356dfaj34dsf
CSeq: 1 NOTIFY
Contact: <sips:park@server.example.com>;automaton
    ;+sip.byeless;+sip.rendering="no"
Event: dialog
Subscription-State: terminated;reason=timeout
Content-Type: application/dialog-info+xml
Content-Length: ...
<?xml version="1.0"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info"
      version="0" state="full" entity="sips:park@server.example.com">
   <dialog id="439920143524"
      call-id="a5-75-34-12-76@server.example.com"
```

```
         local-tag="0111" remote-tag="098594" direction="initiator">
        <duration>1</duration>
        <local>
           <target>sips:park@server.example.com</target>
        </local>
        <remote>
          <target>sips:a8342043f@atlanta.example.com;gr</target>
        </remote>
        <state>confirmed</state>
      </dialog>
    </dialog-info>
```


F19 200 OK Carol -> Park Server

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS server.example.com:5061
 ;branch=z9hG4bK74b8skd
 ;received=192.0.2.103
To: Carol <sips:carol@chicago.example.com>;tag=158x93461
From: <sips:park@server.example.com>;tag=3213j
Call-ID: 2d6485356dfaj34dsf
CSeq: 1 NOTIFY
Contact: <sips:carol@client.chicago.example.com>
Content-Length: 0
```

F20 INVITE Carol -> Alice

```
INVITE sips:alice@atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.chicago.example.com:5061
 ;branch=z9hG4bK74bQ2
Max-Forwards: 70
From: Carol <sips:carol@chicago.example.com>;tag=5893461
To: Alice <sips:alice@atlanta.example.com>
Call-ID: 6485356@chicago.example.com
CSeq: 1 INVITE
Contact: <sips:carol@client.chicago.example.com>
Require: replaces
Replaces: a5-75-34-12-76@server.example.com
 ;to-tag=098594;from-tag=0111
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER,
 SUBSCRIBE, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
```

```
o=carol 2890844922 2890844922 IN IP4 client.chicago.example.com
s=
c=IN IP4 client.chicago.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F21 200 OK Alice -> Carol

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.chicago.example.com:5061
 ;branch=z9hG4bK74bQ2
 ;received=192.0.2.105
From: Carol <sips:carol@chicago.example.com>;tag=5893461
To: Alice <sips:alice@atlanta.example.com>;tag=222
Call-ID: 6485356@chicago.example.com
CSeq: 1 INVITE
Contact: <sips:a8342043f@atlanta.example.com;gr>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces, gruu
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844527 2890844527 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F22 ACK Carol -> Alice

ACK sips:a8342043f@atlanta.example.com;gr SIP/2.0
Via: SIP/2.0/TLS client.chicago.example.com:5061
 ;branch=z9hG4bK74bJ0
Max-Forwards: 70
From: Carol <sips:carol@chicago.example.com>;tag=5893461
To: Alice <sips:alice@atlanta.example.com>;tag=222
Call-ID: 6485356@chicago.example.com
CSeq: 1 ACK
Content-Length: 0

/* A replaces the session to the Park Server with the new
   session with C and generates a BYE to disconnect the
   Park Server. */
```

F23 BYE Alice -> Park Server

```
BYE sips:park@server.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74b4N
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=098594
To: <sips:park@server.example.com>;tag=0111
Call-ID: a5-75-34-12-76@server.example.com
CSeq: 1 BYE
Content-Length: 0
```

F24 200 OK Park Server -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74b4N
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=098594
To: <sips:park@server.example.com>;tag=0111
Call-ID: a5-75-34-12-76@server.example.com
CSeq: 1 BYE
Content-Length: 0
```

## 2.16. Call Pickup

```
            Alice          Bob                 Bill
             |              |                   |
             |   INVITE F1  |                   |
             |------------->|                   |
             |180 Ringing F2|                   |
             |<-------------|                   |
             |              |   SUBSCRIBE F3    |
             |              |<------------------|
             |              |     200 OK F4     |
             |              |------------------>|
             |              |     NOTIFY F5     |
             |              |------------------>|
             |              |     200 OK F6     |
             |              |<------------------|
             |          INVITE Replaces:Bob  F7 |
             |<---------------------------------|
             |              |     200 OK F8     |
             |--------------------------------->|
             |   CANCEL F9  |                   |
             |------------->|                   |
             |  200 OK F10  |                   |
             |<-------------|                   |
             |    487 F11   |                   |
             |<-------------|                   |
             |    ACK F12   |                   |
             |------------->|                   |
             |                    ACK F13       |
             |<---------------------------------|
             |                                  |
             |    Two-Way RTP Established       |
             |<================================>|
             |                     BYE F14      |
             |--------------------------------->|
             |                   200 OK F15     |
             |<---------------------------------|
             |                                  |
```

Bob and Bill are part of a work group at example.com that can pick up each other's calls. Alice calls Bob, who does not answer. Bill wishes to pick up the call and sends a SUBSCRIBE to Bob to retrieve the dialog information. Bill then generates an INVITE with a Replaces to Alice. Alice answers the INVITE and sends a CANCEL to stop Bob's phone ringing. Note that the relative order of the 487/ACK sequence (F11/F12) and the 200 OK to the CANCEL (F10) is not deterministic.

This call flow shows the use of the "early-only" parameter [RFC3891] in the Replaces header field of F7. This parameter prevents Alice from accepting the INVITE if Bob has already accepted the INVITE. If Bill had wished to "take" the call from Bob regardless of whether he had answered, the parameter would not have been present in F7.

Also note that the subscription between Bob and Carol could have been established prior to Alice's call.

Message Details

F1 INVITE Alice -> Bob

```
INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:a8342043f@atlanta.example.com;gr>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F2 180 Ringing Bob -> Alice

```
SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=3145678
Call-ID: 12345600@atlanta.example.com
```

```
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Content-Length: 0

/* Bill decides to pick up the call. */


F3 SUBSCRIBE  Bill -> Bob

SUBSCRIBE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS pc.biloxi.example.com:5061
 ;branch=z9hG4bK74bf
Max-Forwards: 70
From: Bill <sips:bill@biloxi.example.com>;tag=8675309
To: Bob <sips:bob@biloxi.example.com>
Call-ID: rt4353gs2egg@pc.biloxi.example.com
CSeq: 1 SUBSCRIBE
Contact: <sips:bill@pc.biloxi.example.com>
Event: dialog
Expires: 0
Accept: application/dialog-info+xml
Content-Length: 0


F4 200 OK Bob -> Bill

SIP/2.0 200 OK
Via: SIP/2.0/TLS pc.biloxi.example.com:5061
 ;branch=z9hG4bK74bf
 ;received=192.0.2.114
Max-Forwards: 70
From: Bill <sips:bill@biloxi.example.com>;tag=8675309
To: Bob <sips:bob@biloxi.example.com>;tag=31451098
Call-ID: rt4353gs2egg@pc.biloxi.example.com
CSeq: 1 SUBSCRIBE
Content-Length: 0


F5 NOTIFY  Bob -> Bill

NOTIFY sips:bill@pc.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bK74br
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=31451098
To: Bill <sips:bill@biloxi.example.com>;tag=8675309
Call-ID: rt4353gs2egg@pc.biloxi.example.com
CSeq: 1 NOTIFY
```

```
   Contact: <sips:bob@client.biloxi.example.com>
   Event: dialog
   Subscription-State: terminated;reason=timeout
   Content-Type: application/dialog-info+xml
   Content-Length: ...
```


```
   <?xml version="1.0"?>
   <dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info"
          version="0" state="full" entity="sips:bob@biloxi.example.com">
      <dialog id="94992014524" call-id="12345600@atlanta.example.com"
          local-tag="3145678" remote-tag="1234567" direction="recipient">
        <duration>1</duration>
        <local>
          <identity display="Bob">sips:bob@biloxi.example.com</identity>
          <target>sips:bob@client.biloxi.example.com</target>
        </local>
        <remote>
          <identity display="Alice">sips:alice@atlanta.example.com
                                                         </identity>
          <target>sips:a8342043@atlanta.example.com;gr</target>
        </remote>
        <state>early</state>
      </dialog>
    </dialog-info>
```


F6 200 OK  Bill -> Bob

```
   SIP/2.0 200 OK
   Via: SIP/2.0/TLS client.biloxi.example.com:5061
    ;branch=z9hG4bK74br
    ;received=192.0.2.105
   From: Bob <sips:bob@biloxi.example.com>;tag=31451098
   To: Bill <sips:bill@biloxi.example.com>;tag=8675309
   Call-ID: rt4353gs2egg@pc.biloxi.example.com
   CSeq: 1 NOTIFY
   Contact: <sips:bill@pc.biloxi.example.com>
   Content-Length: 0
```

F7 INVITE  Bill -> Alice

```
   INVITE sips:a8342043f@atlanta.example.com;gr SIP/2.0
   Via: SIP/2.0/TLS pc.biloxi.example.com:5061
    ;branch=z9hG4bK74HH
   Max-Forwards: 70
   From: Bill <sips:bill@biloxi.example.com>;tag=8675310
   To: Alice <sips:alice@atlanta.example.com>
```

```
Call-ID: 563456212@b2.biloxi.example.com
CSeq: 1 INVITE
Require: replaces
Replaces: 12345600@atlanta.example.com
 ;from-tag=314578;to-tag=1234567;early-only
Contact: <sips:bill@pc.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bill 2890843122 2890843122 IN IP4 pc.biloxi.example.com
s=
c=IN IP4 pc.biloxi.example.com
t=0 0
m=audio 5342 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* Alice matches the dialog information in the Replaces header
   and accepts the INVITE. */


F8 200 OK  Alice -> Bill

SIP/2.0 200 OK
Via: SIP/2.0/TLS pc.biloxi.example.com:5061
 ;branch=z9hG4bK74HH
 ;received=192.0.2.114
From: Bill <sips:bill@biloxi.example.com>;tag=8675310
To: Alice <sips:alice@atlanta.example.com>;tag=131256
Call-ID: 563456212@b2.biloxi.example.com
CSeq: 1 INVITE
Contact: <sips:a8342043f@atlanta.example.com;gr>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces, gruu
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 289084543 289084543 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* Alice stops Bob’s phone from ringing by sending a CANCEL. */
```

F9 CANCEL Alice -> Bob

```
CANCEL sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 CANCEL
Content-Length: 0
```

F10 200 OK Bob -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=3145678
Call-ID: 12345600@atlanta.example.com
CSeq: 1 CANCEL
Content-Length: 0
```

F11 487 Request Terminated Bob -> Alice

```
SIP/2.0 487 Request Terminated
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=3145678
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0
```

F12 ACK Alice -> Bob

```
ACK sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK83749.1
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=3145678
```

```
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0
```

F13 ACK  Bill -> Alice

```
ACK sips:a8342043f@atlanta.example.com;gr SIP/2.0
Via: SIP/2.0/TLS pc.biloxi.example.com:5061
 ;branch=z9hG4bK7435
Max-Forwards: 70
From: Bill <sips:bill@biloxi.example.com>;tag=8675310
To: Alice <sips:alice@atlanta.example.com>;tag=131256
Call-ID: 563456212@b2.biloxi.example.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between Alice and Bill.
   Later, Alice hangs up with Bill. */
```

F14 BYE Alice -> Bill

```
BYE sips:bill@pc.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf2
Max-Forwards: 70
To: Bill <sips:bill@biloxi.example.com>;tag=8675310
From: Alice <sips:alice@atlanta.example.com>;tag=131256
Call-ID: 563456212@b2.biloxi.example.com
CSeq: 1 BYE
Content-Length: 0
```

F15 200 OK Bill -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf2
 ;received=192.0.2.105
To: Bill <sips:bill@biloxi.example.com>;tag=8675310
From: Alice <sips:alice@atlanta.example.com>;tag=131256
Call-ID: 563456212@b2.biloxi.example.com
CSeq:  1 BYE
Content-Length: 0
```

## 2.17. Automatic Redial

```
           Alice             Bob
             |                |
             |    INVITE F1   |
             |--------------->|
             |486 Busy Here F2|
             |<---------------|    Bob is busy
             |     ACK F3     |
             |--------------->|
             |  SUBSCRIBE F4  |
             |--------------->|
             |   200 OK F5    |
             |<---------------|
             |    NOTIFY F6   |
             |<---------------|
             |   200 OK F7    |
             |--------------->|
             |                |
             |    NOTIFY F8   |    Bob is now available
             |<---------------|
             |   200 OK F9    |
             |--------------->|
             |                |
             |   INVITE F10   |
             |--------------->|    Session setup successful
             | 180 Ringing F11|
             |<---------------|
             |   200 OK F12   |
             |<---------------|
             |     ACK F13    |
             |--------------->|
             |  Media Session |
             |<==============>|
             |                |
             |   NOTIFY F14   |
             |<---------------|
             |   200 OK F15   |
             |--------------->|
             |                |
             |  SUBSCRIBE F16 | Alice terminates subscription
             |--------------->|
             |   200 OK F17   |
             |<---------------|
             |   NOTIFY F18   |
             |<---------------|
             |   200 OK F19   |
             |--------------->|
             |                |
```

Bob is initially busy when Alice calls. Alice subscribes to Bob's call state using a SUBSCRIBE F4. Bob sends a NOTIFY F8 when Bob is available. Alice is alerted, then Alice sends an INVITE to Bob to establish the session. The subscription is terminated using SUBSCRIBE F16.

Message Details

F1 INVITE Alice -> Bob

```
INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F2 486 Busy Here

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=982039i4
Call-ID: 12345600@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0
```

F3 ACK Alice -> Bob

```
ACK sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=982039i4
Call-ID: 12345600@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0
```

F4 SUBSCRIBE Alice -> Bob

```
SUBSCRIBE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74b8G
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=837348234
To: Bob <sips:bob@biloxi.example.com>
Call-ID: 4524526232@atlanta.example.com
CSeq: 1 SUBSCRIBE
Contact: sips:alice@client.atlanta.example.com
Event: dialog
Accept: application/dialog-info+xml
Content-Length: 0
```

F5 200 OK Bob -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74b8G
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=837348234
To: Bob <sips:bob@biloxi.example.com>;tag=341123
Call-ID: 4524526232@atlanta.example.com
Expires: 60
CSeq: 1 SUBSCRIBE
Contact: sips:bob@client.biloxi.example.com
Content-Length: 0
```

F6 NOTIFY Bob -> Alice

```
NOTIFY sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bK74bn2
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=341123
To: Alice <sips:alice@atlanta.example.com>;tag=837348234
Call-ID: 4524526232@atlanta.example.com
CSeq: 1 NOTIFY
Contact: <sips:bob@client.biloxi.example.com>
Event: dialog
Subscription-State: active;expires=59
Content-Type: application/dialog-info+xml
Content-Length: ...

<?xml version="1.0"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info"
       version="0" state="full" entity="sips:bob@biloxi.example.com">
   <dialog id="562623442g3">
      <duration>1</duration>
     <state>confirmed</state>
   </dialog>
 </dialog-info>
```

F7 200 OK Alice -> Bob

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bK74bn2
 ;received=192.0.2.105
From: Bob <sips:bob@biloxi.example.com>;tag=341123
To: Alice <sips:alice@atlanta.example.com>;tag=837348234
Call-ID: 4524526232@atlanta.example.com
CSeq: 1 NOTIFY
Content-Length: 0

/* Bob is now available. */
```

F8 NOTIFY Bob -> Alice

```
NOTIFY sips:alice@atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bK74bVi
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=341123
```

```
To: Alice <sips:alice@atlanta.example.com>;tag=837348234
Call-ID: 4524526232@atlanta.example.com
CSeq: 2 NOTIFY
Event: dialog
Subscription-State: active;expires=27
Contact: <sips:bob@client.biloxi.example.com>
Content-Type: application/dialog-info+xml
Content-Length: ...

<?xml version="1.0"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info"
       version="0" state="full" entity="sips:bob@biloxi.example.com">
   <dialog id="562623442g3">
     <state>terminated</state>
   </dialog>
</dialog-info>
```

F9 200 OK Alice -> Bob

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bK74bVi
 ;received=192.0.2.105
From: Bob <sips:bob@biloxi.example.com>;tag=341123
To: Alice <sips:alice@atlanta.example.com>;tag=837348234
Call-ID: 4524526232@atlanta.example.com
CSeq: 2 NOTIFY
Content-Length: 0
```

F10 INVITE Alice -> Bob

```
INVITE sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bfq
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=f23fkg14k
To: Bob <sips:bob@biloxi.example.com>
Call-ID: aoij4i9okitr@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:alice@client.atlanta.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...
```

```
v=0
o=alice 2890844826 2890844826 IN IP4 client.atlanta.example.com
s=
c=IN IP4 client.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F11 180 Ringing Bob -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bfq
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=f23fkg14k
To: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: aoij4i9okitr@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Content-Length: 0


F12 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bfq
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=f23fkg14k
To: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: aoij4i9okitr@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890854527 2890854527 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F13 ACK Alice -> Bob

```
ACK sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK74bLBJ
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=f23fkg14k
To: Bob <sips:bob@biloxi.example.com>;tag=23431
Call-ID: aoij4i9okitr@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0
```

F14 NOTIFY Bob -> Alice

```
NOTIFY sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bK4bnd2
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=341123
To: Alice <sips:alice@atlanta.example.com>;tag=837348234
Call-ID: 4524526232@atlanta.example.com
CSeq: 3 NOTIFY
Contact: <sips:bob@client.biloxi.example.com>
Event: dialog
Subscription-State: active;expires=15
Content-Type: application/dialog-info+xml
Content-Length: ...

<?xml version="1.0"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info"
       version="0" state="full" entity="sips:bob@biloxi.example.com">
   <dialog id="62d2623442g3">
      <duration>1</duration>
     <state>confirmed</state>
   </dialog>
 </dialog-info>
```

F15 200 OK Alice -> Bob

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bK4bnd2
 ;received=192.0.2.105
From: Bob <sips:bob@biloxi.example.com>;tag=341123
To: Alice <sips:alice@atlanta.example.com>;tag=837348234
```

```
Call-ID: 4524526232@atlanta.example.com
CSeq: 3 NOTIFY
Content-Length: 0

/* Alice terminates the subscription. */


F16 SUBSCRIBE Alice -> Bob

SUBSCRIBE sips:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK474b8
Max-Forwards: 70
From: Alice <sips:alice@atlanta.example.com>;tag=837348234
To: Alice <sips:alice@atlanta.example.com>;tag=837348234
Call-ID: 4524526232@atlanta.example.com
CSeq: 2 SUBSCRIBE
Contact: sips:alice@client.atlanta.example.com
Event: dialog
Expires: 0
Accept: application/dialog-info+xml
Content-Length: 0


F17 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.example.com:5061
 ;branch=z9hG4bK474b8
 ;received=192.0.2.103
From: Alice <sips:alice@atlanta.example.com>;tag=837348234
To: Bob <sips:bob@biloxi.example.com>;tag=341123
Call-ID: 4524526232@atlanta.example.com
Expires: 0
CSeq: 2 SUBSCRIBE
Contact: sips:bob@client.biloxi.example.com
Content-Length: 0


F18 NOTIFY Bob -> Alice

NOTIFY sips:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKb5n2j
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=341123
To: Alice <sips:alice@atlanta.example.com>;tag=837348234
Call-ID: 4524526232@atlanta.example.com
```

```
   CSeq: 4 NOTIFY
   Contact: <sips:bob@client.biloxi.example.com>
   Event: dialog
   Subscription-State: terminated;reason=noresource
   Content-Type: application/dialog-info+xml
   Content-Length: ...

   <?xml version="1.0"?>
   <dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info"
          version="0" state="full" entity="sips:bob@biloxi.example.com">
      <dialog id="62d2623442g3">
         <duration>3</duration>
        <state>confirmed</state>
      </dialog>
    </dialog-info>


   F19 200 OK Alice -> Bob

   SIP/2.0 200 OK
   Via: SIP/2.0/TLS client.biloxi.example.com:5061
    ;branch=z9hG4bKb5n2j
    ;received=192.0.2.105
   From: Bob <sips:bob@biloxi.example.com>;tag=341123
   To: Alice <sips:alice@atlanta.example.com>;tag=837348234
   Call-ID: 4524526232@atlanta.example.com
   CSeq: 4 NOTIFY
   Content-Length: 0
```

## 2.18. Click to Dial

```
 Bob's PC                Bob               Carol
     | REFER Refer-To:Carol F1                |
     |------------------->|                   |
     |  202 Accepted F2   |                   |
     |<-------------------|                   |
     |                    |     INVITE F3     |
     |                    |------------------>|
     |                    |   180 Ringing F4  |
     |                    |<------------------|
     |                    |     200 OK F5     |
     |                    |<------------------|
     |                    |       ACK F6      |
     |                    |------------------>|
     |                    |        RTP        |
     |                    |<=================>|
     |                    |                   |
```

In this example, while browsing the web on his PC, Bob clicks on Carol's SIP URI, intending to establish a session with Carol. Bob's web browser passes the SIP URI to the SIP client on Bob's PC. The PC client is configured with the URI of Bob's SIP phone. A REFER is sent to the SIP phone, which results in the establishment of the session between Bob and Carol.

Note that Bob's PC requests that no REFER dialog be established by the use of the Refer-Sub: false header field [RFC4488].

This flow is preferable to the 3pcc flow because the end-to-end SIP signaling is not interrupted by the 3pcc controller, and because Bob's experience of the call will not be marred by the lack of ringback tone or possible clipping. Suitable authorization of the REFER and explicit authorization of the triggered INVITE by Bob are necessary.

Message Details

/* Bob's PC SIP client sends a REFER to Bob's SIP phone. */

F1 REFER PC -> Bob

```
REFER sips:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/TLS pc.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
Max-Forwards: 70
From: <sips:pc.biloxi.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>
```

```
Call-ID: 1234560183434
CSeq: 1 REFER
Refer-To: <sips:carol@chicago.example.com>
Refer-Sub: false
Contact: <sips:pc.biloxi.example.com>
Content-Length: 0


F2 202 Accepted Bob -> PC

SIP/2.0 202 Accepted
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashds7
 ;received=192.0.2.103
From: <sips:pc.biloxi.example.com>;tag=1234567
To: Bob <sips:bob@biloxi.example.com>;tag=314159
Call-ID: 1234560183434
Contact: <sips:bob@client.biloxi.example.com>
CSeq: 1 REFER
Refer-Sub: false
Content-Length: 0


F3 INVITE Bob -> Carol

INVITE sips:carol@chicago.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashdK9
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>
Call-ID: 7436222@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:bob@client.biloxi.example.com>
Referred-By: <sips:pc.biloxi.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=bob 2890844539 2890844539 IN IP4 client.biloxi.example.com
s=
c=IN IP4 client.biloxi.example.com
t=0 0
m=audio 3458 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F4 180 Ringing Carol -> Bob

```
SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashdK9
 ;received=192.0.2.113
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=928287
Call-ID: 7436222@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:carol@client.chicago.example.com>
Content-Length: 0
```

F5 200 OK Carol -> Bob

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashdK9
 ;received=192.0.2.113
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=928287
Call-ID: 7436222@atlanta.example.com
CSeq: 1 INVITE
Contact: <sips:carol@client.chicago.example.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: ...

v=0
o=carol 2890844527 2890844527 IN IP4 client.chicago.example.com
s=
c=IN IP4 client.chicago.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F6 ACK Bob -> Carol

```
ACK sips:carol@client.chicago.example.com SIP/2.0
Via: SIP/2.0/TLS client.biloxi.example.com:5061
 ;branch=z9hG4bKnashd43
Max-Forwards: 70
From: Bob <sips:bob@biloxi.example.com>;tag=8675309
To: Carol <sips:carol@chicago.example.com>;tag=928287
```

```
Call-ID: 7436222@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

/* Bob and Carol now have established a session. */
```

## 3. Security Considerations

Since many of the examples in this document involve SIP call control, either peer-to-peer or 3pcc, the security considerations in the Multiparty Framework document [FRAMEWORK] apply.

Many of the services shown in this document rely on a particular user agent being part of a group. Members of a group could be, for example, employees within a particular department, a set of home phone extensions, members of a call center, etc. As such, user agents that are part of the group permit other group members special privileges and features. For example, while a user agent may not in general allow another user agent to learn detailed dialog information, this information might be shared with another group member in order to facilitate a service such as call pickup. Group members must be authenticated using normal SIP means such as certificates or shared secrets.

The service examples in this document make extensive use of the SIP call control primitives REFER, Replaces, Join, and the dialog package. The security considerations associated with each of these extensions [RFC3515], [RFC3891], [RFC3911], [RFC4235] apply to the scenarios in this document.

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