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SIP: Session Initiation Protocol

Abstract

Many styles of multimedia conferencing are likely to co-exist on the Internet, and many of them share the need to invite users to participate. The Session Initiation Protocol (SIP) is a simple protocol designed to enable the invitation of users to participate in such multimedia sessions. It is not tied to any specific conference control scheme, providing support for either loosely or tightly controlled sessions. In particular, it aims to enable user mobility by relaying and redirecting invitations to a user’s current location.

This document is a product of the Multiparty Multimedia Session Control (MMUSIC) working group of the Internet Engineering Task Force. Comments are solicited and should be addressed to the working group’s mailing list at confctrl@isi.edu and/or the authors.

Authors’ Note

This document is the result of a merger of the Session Invitation Protocol (draft-ietf-mmusic-sip-00.txt) and the Simple Conference Invitation Protocol (draft-ietf-mmusic-scip-00.txt), and of an attempt to make SIP more generic and to fit into a more flexible infrastructure that includes companion protocols including SDP, HTTP and RTSP.

Changes

Since version -01, the following things have changed:

- CAPABILITIES to OPTIONS for closer alignment with HTTP and RTSP;
- Path to Via for closer alignment with HTTP and RTSP;
- Content type meta changed to application, since “meta” doesn’t exist as a top-level Internet media type.
- Formatting closer to HTTP and RTSP.

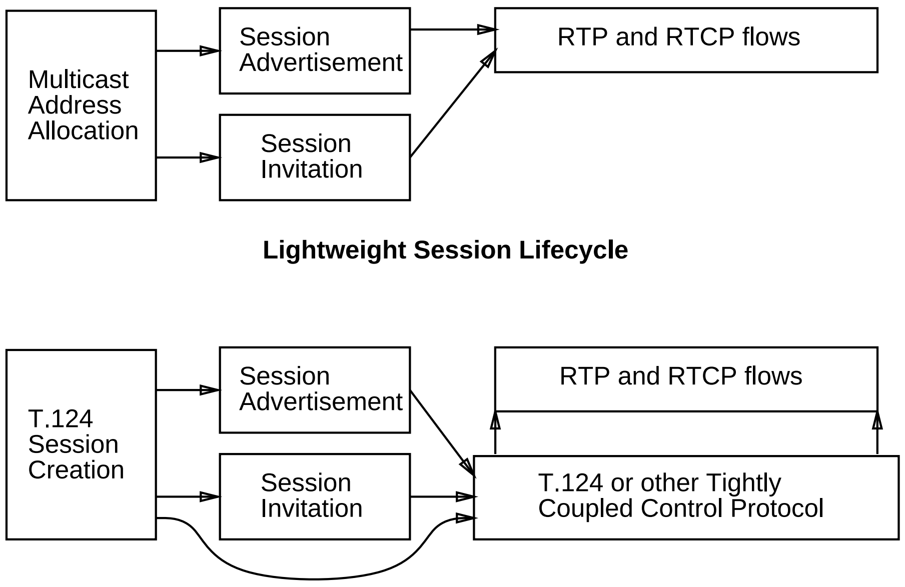
- Explain relationship to H.323.

1 Introduction

There are two basic ways to locate and to participate in a multimedia session:

- The session is advertised, users see the advertisement, then join the session address to participate.
- Users are invited to participate in a session, which may or may not already be advertised.

The Session Description Protocol (SDP) [1] together with the Session Announcement Protocol (SAP) [2], provide a mechanism for the former. This document presents the Session Initiation Protocol (SIP) to perform the latter. SIP MAY also use SDP to describe a session.



Tightly Coupled Session Lifecycle

Figure 1: Session Lifecycle

We make the design decision that how a user discovers that a session exists is orthogonal to a session's conference control model. Figure 1 shows a potential place for SIP in the lifecycle of both lightweight sessions and in more tightly-coupled conferencing. Note that the Session Initiation Protocol and the Session Announcement Protocol may be invoked or re-invoked at later stages in a session's lifecycle.

The Session Initiation Protocol is also intended to be used to invite servers into sessions. Examples might be where a recording server can be invited to participate in a live multimedia session to record that session, or a video-on-demand server can be invited to play a video stream into a live multimedia conference. In such cases we would like SIP to lead the server gracefully into the control protocol that controls the actual recording and playback.

We also make the design decision that inviting a user to participate in a session is independent of quality of service (QoS) guarantees for that session. Such QoS guarantees (if they are required) may be dependent on the full membership of the session, and this may or may not be known to the agent performing session invitation.

SIP offers some of the same functionality as H.323, but can also be used in conjunction with it. In this mode, SIP is used to locate the appropriate terminal, where the terminal is identified by its H.245 address [TBD: what does this look like?]. An H.323-capable terminal then proceeds with a normal H.323/H.245 invitation [3].

1.1 Requirements

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 RFC xxxx [4].

1.2 Terminology

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 HTTP.

Client: An application program that establishes connections for the purpose of sending requests. Clients may or may not interact directly with a human user.

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

Invitation: A request sent to attempt to contact a user (or service) to request that they participate in a session.

Invitee, Invited User: The person or service that the calling party is trying to invite to a conference.

Location server: A program that is contacted by a client and that returns one or more possible locations for the user or service without contacting that user or service directly.

Location service: A service used by a location server to obtain information about a user's possible location.

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 must interpret, and, if necessary, rewrite a request message before forwarding it.

Server: An application program that accepts connections in order to service requests by sending back responses. A server may be the called user agent, a proxy server, or a location server.

User Agent, Called User Agent: The server application which contacts the invitee to inform them of the invitation, and to return a reply.

Any given program may be capable of acting both as a client and a server. A typical multimedia conference controller would act as a client to initiate calls or to invite others to conferences and as a server to accept invitations.

1.3 General Requirements

SIP is a Session Initiation Protocol. It is not a conference control protocol. SIP can be used to perform a search for a user or service and to request that that user or service participate in a session.

Once SIP has been used to initiate a multimedia session SIP's task is finished. **There is no concept of a SIP session** (as opposed to a SIP search for a user or service). If whatever conference control mechanism is used in the session needs to add or remove a media stream, SIP may be used to perform this task, but again, once the information has been successfully conveyed to the participants, SIP is then no longer involved.

SIP must be able to utilize both UDP and TCP as transport protocols.

From a performance point of view, UDP is preferable as it allows the application to more carefully control the timing of messages, it allows parallel searches without requiring connection state for each outstanding request, and allows the use of multicast.

From a pragmatic point of view, TCP allows easier passage through existing firewalls, and with appropriate protocol design, allows common SIP, HTTP and RTSP servers.

When TCP is used, SIP can use either one or more than one connection to attempt to contact a user or to modify parameters of an existing session. The concept of a session is not implicitly bound to a TCP connection, so the initial SIP request and a subsequent SIP request may use different TCP connections or a single persistent connection as appropriate.

SIP is text based. This allows easy implementation in languages such as TCL and Perl, allows easy debugging, and most importantly, makes SIP flexible and extensible. As SIP is only used for session initiation, it is believed that the additional overhead of using a text-based protocol is not significant.

Unlike control protocols, there is minimal shared-state in SIP – in a minimal implementation the initiator maintains all the state about the current attempt to locate and contact a user or service - servers or proxies can be stateless (although they don't have to be). All the state needed to get a response back from a server to the initiator is carried in the SIP request itself - this is also necessary for loop prevention.

Whilst redesigning SIP, we have attempted to ensure that it has a clear interaction with the currently evolving Real-Time Stream Control Protocol.

1.4 Addressing

SIP is a protocol that exchanges messages between peer *user agents* or *proxies* for user agents. We assume the user agent is an application that acts on behalf of the user it represents (thus it is sometimes described as a *client* of the user) and that is co-resident with that user. A proxy for a user agent serves as a forwarding mechanism or bridge to the actual location of the user agent. We also refer to such proxies as *location server*.

In the computer realm, the equivalent of a personal telephone number combines the user's login id (`mjh`) with a machine host name (`metro.isi.edu`) or numeric network address (`128.16.64.78`). A user's location-specific 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.

However, users also publish several well-known addresses that are relatively location-independent, such as email or web home-page addresses. Rather than require that users provide their specific network locales, we can take advantage of email and web addresses as being (relatively) memorable, and also leverage off the Domain Name Service (DNS) to provide a first stage location mechanism. Note that an email address (`M.Handley@cs.ucl.ac.uk`) is usually different from the combination of a specific machine name and login name (`mjh@mercury.lcs.mit.edu`). SIP should allow both forms of addressing to be used, with the former requiring a location server to locate the user.

One perceived problem of email addressing is that it is possible to guess peoples' addresses and thus the system of unlisted (in the telephone directory) numbers is more of a problem. However, this really only provides security through obscurity, and real security is better provided through authentication and call screening.

1.5 Call Setup

Call setup is a multi-phase procedure. In the first phase, the requesting client tries to ascertain the address where it should contact the remote user agent or user agent proxy. The local client checks if the user address is location-specific. If so, then that is the address used for the remote user agent. If not, the requesting client looks up the domain part of the user address in the DNS. This provides one or more records giving IP addresses. If a new service (SRV) resource record [5] is returned giving a location server, then that is the address to contact next. If no relevant resource record is returned, but an A record is returned, then that is the address to contact next. If neither a resource record or an A record is returned, but an MX record is returned, then the mail host is the address to contact next.

Presuming an address for the invitee is found from the DNS, the second and subsequent phases basically implement a request-response protocol. A session description (typically using SDP format) is sent to the contact address with an invitation for the user to join the session.

This request may be sent over a TCP connection or as a single UDP datagram (the format of both is the same and is described later), and is sent to a well-known port.

If a user agent or conference server is listening on the relevant port, it can send one of the responses below. If no server or agent is listening, an ICMP port-unreachable response will be triggered which should cause the TCP connection setup to fail or cause a UDP send failure on retransmissions.

1.6 Locating a User

It is expected that a user is situated at one of several frequented locations. These locations can be dynamically registered with a location server for a site (for a local area network or organization), and incoming connections can be routed simultaneously to all of these locations if so desired. It is entirely up to the location server whether the server issues proxy requests for the requesting user, or if the server instructs the client to redirect the request.

In general a reply MUST be sent by the same mechanism that the request was sent by. Hence, if a request was unicast, then the reply MUST be unicast back to the requester; if the request was multicast, the reply MUST be multicast to the same group to which the request was sent; if the request was sent by TCP, the reply MUST be sent by TCP.

In all cases where a request is forwarded onwards, each host relaying the message SHOULD add its own address to the path of the message so that the replies can take the same path back, thus ensuring correct operation through compliant firewalls and loop-free requests. On the reply path, these routing headers MUST be removed as the reply retraces the path, so that routing internal to sites is hidden. When a multicast request is made, first the host making the request, then the multicast address itself are added to the path.

2 Notational Conventions and Generic Grammar

Since many of the definitions and syntax are identical to HTTP/1.1, this specification only points to the section where they are defined rather than copying it. For brevity, [HX.Y] is to be taken to refer to Section X.Y of the current HTTP/1.1 specification (RFC 2068).

All the mechanisms specified in this document are described in both prose and an augmented Backus-Naur form (BNF) similar to that used in RFC 2068 [H2.1]. It is described in detail in [6].

In this draft, we use indented and smaller-type paragraphs to provide background and motivation.

3 Protocol Parameters

3.1 SIP Version

[H3.1] applies, with HTTP replaced by SIP.

Applications sending Request or Response messages, as defined by this specification, MUST include an SIP-Version of ”SIP/2.0”. Use of this version number indicates that the sending application is at least conditionally compliant with this specification.

3.2 UCI: Universal Communication Identifier

[TBD: describe all legal address formats.]

4 SIP Message

All messages are text-based, using the conventions of HTTP/1.1 [H4.1], except for the additional ability of SIP to use UDP. When sent over TCP or UDP, multiple requests can be carried in a single TCP connection or UDP datagram. UDP Datagrams should not normally exceed the path MTU in size if it is known, or 1,000 bytes if the MTU is unknown.

4.1 Message Types

SIP messages consist of requests from client to server and responses from server to client.

```
SIP-message = Request | Response     ; HTTP/1.1 messages
```

Request (section 5) and response (section 6) messages use the generic message format of RFC 822 for transferring entities (the payload 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 CRLF) 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
```

In the interest of robustness, servers SHOULD ignore any empty line(s) received where a Request-Line is expected. In other words, if the server is reading the protocol stream at the beginning of a message and receives a CRLF first, it should ignore the CRLF.

4.2 Message Headers

HTTP header fields, which include general-header (section), request-header (section), response-header (section), fields, follow the same generic format as that given in Section 3.1 of RFC 822. 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 LWS, though a single SP is preferred. Header fields can be extended over multiple lines by preceding each extra line with at least one SP or HT. Applications SHOULD follow "common form" when generating HTTP 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 or combinations
                 of token, tspecials, and quoted-string>
```

The order in which header fields with differing field names are received is not significant.

Multiple message-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.

4.3 Message Body

The rules for when a message-body is allowed in a message differ for requests and responses.

The presence of a message-body in a request is signaled by the inclusion of a Content-Length or Transfer-Encoding header field in the request's message-headers. A message-body MAY be included in a request only when the request method allows an entity-body.

For response messages, whether or not a message-body is included with a message is dependent on both the request method and the response status code (section). All 1xx (informational) responses MUST NOT include a message-body. All other responses do include a message-body, although it may be of zero length.

4.4 Message Length

When a message-body is included with a message, the length of that body is determined by one of the following (in order of precedence):

1. Any response message which MUST NOT include a message-body (such as the 1xx responses) is always terminated by the first empty line after the header fields, regardless of the entity-header fields present in the message.
2. Otherwise, a Content-Length header MUST be present. (This requirement differs from HTTP/1.1.) Its value in bytes represents the length of the message-body.

The “chunked” transfer encoding of HTTP/1.1 MUST NOT be used for SIP.

4.5 General Header Fields

There are a few header fields which have general applicability for both request and response messages. These header fields apply only to the message being transmitted.

```
general-header = Date                       ; Section
               | Transfer-Encoding          ; Section
               | Via                        ; Section
```

General-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 general header fields if all parties in the communication recognize them to be general-header fields.

5 Request

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
```

The method may be either INVITE or CAPABILITY. The request ID may be any URL encoded string that can be guaranteed to be globally unique for the duration of the request. Using the initiator’s IP-address, process id, and instance (if more than one request is being made simultaneously) satisfies this requirement.

6 Response

[H6] applies except that `HTTP-Version` is replaced by `SIP-Version`. Also, SIP defines additional status codes and does not define some HTTP codes.

After receiving and interpreting a request message, the recipient responds with an SIP response message.

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

6.1 Status-Line

The first line of a Response message is the `Status-Line`, consisting of the protocol version followed by a numeric status code, the sequence number of the corresponding request and the textual phrase associated with the status code, with each element separated by `SP` characters. No `CR` or `LF` is allowed except in the final CRLF sequence. Note that the addition of a

```
Status-Line = SIP-Version SP Status-Code SP seq-no SP Reason-Phrase CRLF
```

6.1.1 Status Code and Reason Phrase

The Status-Code element is a 3-digit integer result code of the attempt to understand and satisfy the request. These codes are fully defined in section10. The `Reason-Phrase` is intended to give a short textual description of the Status-Code. The `Status-Code` is intended for use by automata and the Reason-Phrase is intended for the human user. The client is not required to examine or display the `Reason-Phrase`.

The first digit of the `Status-Code` defines the class of response. The last two digits do not have any categorization role. There are 5 values for the first digit:

- 1xx: Informational - Request received, continuing process
- 2xx: Success - The action was successfully received, understood, and accepted
- 3xx: Redirection - Further action must be taken in order to complete the request
- 4xx: Client Error - The request contains bad syntax or cannot be fulfilled
- 5xx: Server Error - The server failed to fulfill an apparently valid request

The individual values of the numeric status codes defined for SIP/2.0, and an example set of corresponding `Reason-Phrase`'s, are presented below. The reason phrases listed here are only recommended – they may be replaced by local equivalents without affecting the protocol. Note that SIP adopts many HTTP/1.1 status codes and adds SIP-specific status codes in the starting at 450 to avoid conflicts with newly defined HTTP status codes.

```
Status-Code     = "100"   ; Continue
                | "200"   ; OK
                | "300"   ; Multiple Choices
                | "301"   ; Moved Permanently
                | "302"   ; Moved Temporarily
                | "303"   ; See Other
```

```
                         | "305"   ; Use Proxy
                         | "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 Time-out
                         | "409"   ; Conflict
                         | "410"   ; Gone
                         | "411"   ; Length Required
                         | "412"   ; Precondition Failed
                         | "413"   ; Request Entity Too Large
                         | "414"   ; Request-URI Too Large
                         | "415"   ; Unsupported Media Type
                         | "500"   ; Internal Server Error
                         | "501"   ; Not Implemented
                         | "502"   ; Bad Gateway
                         | "503"   ; Service Unavailable
                         | "504"   ; Gateway Time-out
                         | "505"   ; HTTP Version not supported
                         | extension-code

   extension-code = 3DIGIT

   Reason-Phrase  = *<TEXT, excluding CR, LF>
```

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

6.1.2 Response Header Fields

The response-header fields allow the request recipient 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 = Location              ; Section
                  | Proxy-Authenticate    ; Section
                  | Public                ; Section
                  | Retry-After           ; Section
                  | Server                ; Section
                  | Vary                  ; Section
                  | WWW-Authenticate      ; Section
```

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.

7 SIP Message Body

The session description payload gives details of the session the user is being invited to join. Its Internet media type MUST be given by the "Content-type:" header field, and the payload length in bytes MUST be given by the Content-length header field. If the payload has undergone any encoding (such as compression) then this MUST be indicated by the Content-encoding: header field, otherwise Content-encoding: MUST be omitted.

The example below is a request message en route from initiator to invitee:

```
INVITE 128.16.64.19/65729 SIP/2.0
Via: SIP/2.0/UDP 239.128.16.254 16
Via: SIP/2.0/UDP 131.215.131.131
Via: SIP/2.0/UDP 128.16.64.19
From: mjh@isi.edu
To: schooler@cs.caltech.edu
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 first line above states that this is a SIP version 2.0 request.

The via fields give the hosts along the path from invitation initiator (the first element of the list) towards the invitee. 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 131.215.131.131.

The request header above states that the request was initiated by `mjh@isi.edu` (specifically it was initiated from 128.16.64.19, as can be seen from the Via header) and the user being invited is schooler@cs.caltech.edu.

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

The header is terminated by an empty line and is followed by the session description payload.

If required, the session description can be encrypted using public key cryptography, and then can also carry private session keys for the session. If this is the case, four random bytes are added to the beginning of the session description before encryption and are removed after decryption but before parsing.

8 Methods

The following methods are defined:

INVITE: The user or service is being invited to participate in the session. The session description given must be completely acceptable for a "200 OK" response to be given. This method MUST be supported by a SIP server.

OPTIONS: The user or service 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. Support of this method is OPTIONAL.

Methods that are not supported by a proxy server SHOULD be treated by that proxy as if they were an INVITE method, and relayed through unchanged or cause a redirection as appropriate.

Methods that are not supported by a user agent should cause a "501 Not Implemented" response to be returned.

9 Header Field Definitions

SIP header fields are similar to HTTP header fields in both syntax and semantics. In general the ordering of the header fields is not of importance (with the exception of Via fields, see below) but proxies MUST NOT reorder or otherwise modify header fields other than by adding a new Via field. This allows an authentication field to be added after the Via fields that will not be invalidated by proxies. Field names are not case-sensitive, although their values may be.

Content-Length, Content-Type, To, From header fields are compulsory. Other fields may be added as required. Header fields MUST be separated by a single linefeed character. The header MUST be separated from the payload by an empty line (two linefeed characters).

A compact form of these header fields is also defined in section 10.9 for use over UDP when the request has to fit into a single packet and size is an issue.

9.1 Accept

See [H14.1]. This header field is used only for the OPTIONS request to indicate what description formats are acceptable.

9.2 Accept-Language

See [H14.4]. The Accept-Language request header can be used to allow the client to indicate to the server in which language it would prefer to receive reason phrases. This may also be used as a hint by the proxy as to which destination to connect the call to (e.g., for selecting a human operator).

9.3 Authentication

Authentication fields provide a digital signature of the remaining fields for authentication purposes. They are *not yet defined*. The use of authentication headers is optional. If used, authentication headers MUST be added to the header after the Via fields and before the rest of the fields.

HS: Ordering and semantics needs work. Maybe we can recycle the S/MIME work?

9.4 Confirm

TBD.

9.5 Contact-Host

TBD.

9.6 From

The request header MUST contain a From request-header field, indicating the invitation initiator. The field MUST be machine-usable, as defined my *mailbox* in RFC 822 (as updated by RFC 1123). Only a single initiator and a single invited user are allowed to be specified in a single SIP request.

9.7 Retry-After

The Retry-After response-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) or 451* (Busy) response to indicate when the called party may be available again. The value of this field can be either an HTTP-date or an integer number of seconds (in decimal) after the time of the response.

```
Retry-After  = "Retry-After" ":" ( HTTP-date | delta-seconds )
```

Two examples of its use are

```
Retry-After: Fri, 31 Dec 1999 23:59:59 GMT
Retry-After: 120
```

In the latter example, the delay is 2 minutes.

9.8 Reason

TBD.

9.9 To

The To request-header field specifies the invited user, with the same syntax as the From field.

9.10 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. Initiators MUST add their own Path field to each request. This Path field MUST be the first field in the request. Subsequent proxies SHOULD each add their own additional Path field which MUST be added before any existing Path fields. When a reply passes through a proxy on the reverse path, that proxies Path field MUST be removed from the reply.

The format for a Via header is:

```
Via = "Via" ":" 1#( sent-protocol sent-by [ ttl ] [ comment ] )
sent-protocol     = [ protocol-name "/" ] protocol-version
                    [ "/" transport ]
protocol-name     = "SIP" | token
protocol-version  = token
transport         = "UDP" | "TCP"
sent-by           = host [ ":" port ]
ttl               = *DIGIT
```

TTL is included only if the address is a multicast address.

10 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. Response codes not defined by HTTP/1.1 are marked with an asterisk, and have codes x50 upwards to avoid clashes with future HTTP response codes, or 6xx which are not used by HTTP. The default behavior for unknown response codes is given for each category of codes.

10.1 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. If UDP transport is being used, the client SHOULD periodically re-send the request in case the final response is lost. Typically a server should send a "1xx" response if it expects to take more than one second to obtain a final reply.

10.1.1 100 Trying

Some further action is being taken (e.g., the request is being forwarded) but the user has not yet been located.

10.1.2 150 Ringing

The user agent or conference server has located a possible location where the user has been recently and is trying to alert them.

10.2 Successful 2xx

The request was successful and MUST terminate a search.

10.2.1 200 OK

The request was successful in contacting the user, and the user has agreed to participate.

10.3 Redirection 3xx

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

10.3.1 300 Multiple Choices

The requested resource corresponds to any one of a set of representations, each with its own specific location, and agent- driven negotiation information (section 13) is being provided so that the user (or user agent) can select a preferred representation 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. The entity format is specified by the media type given in the Content- Type header field. Depending upon the format and the capabilities of the user agent, selection of the most appropriate choice may be performed automatically. However, this specification does not define any standard for such automatic selection.

If the server has a preferred choice, it SHOULD include the specific URL for that representation in the Location field; user agents MAY use the Location field value for automatic redirection.

10.3.2 301 Moved Permanently

The requesting client should retry on the new address given by the Location: field because the user has permanently moved and the address this response is in reply to is no longer a current address for the user. A 301 response MUST NOT suggest any of the hosts in the request's path as the user's new location.

10.3.3 302 Moved Temporarily

The requesting client should retry on the new address(es) given by the Location header. A 302 response MUST NOT suggest any of the hosts in the request's path as the user's new location.

10.3.4 350* Alternative Service

The call was not successful, but alternative services are possible. The alternative services are described in the body of the reply.

10.4 Request Failure 4xx

4xx responses are definite failure responses that MUST terminate the existing search for a user or service. They SHOULD NOT be retried immediately without modification.

10.4.1 400 Bad Request

The request could not be understood due to malformed syntax.

10.4.2 401 Unauthorized

The request requires user authentication.

10.4.3 402 Payment Required

Reserved for future use.

10.4.4 403 Forbidden

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

10.4.5 404 Not Found

The server has definitive information that the user does not exist at the domain specified.

10.4.6 406 Not Acceptable

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

10.4.7 450* Decline

The user's machine was successfully contacted but the user explicitly does not wish to participate.

10.4.8 451* Busy

The user's machine was successfully contacted but the user is busy, or the user does not wish to participate (the ambiguity is intentional).

10.5 Server Failure 5xx

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

10.5.1 500 Server Internal Error

The server encountered an unexpected condition that prevented it from fulfilling the request.

10.5.2 501 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.

10.5.3 503 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 SHOULD handle the response as it would for a 500 response.

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

10.6 Search Responses 6xx

6xx responses are failure responses given whilst trying to locate the specified user or service. They are not definitive failures, and SHOULD NOT terminate the search if other possible locations remain untried.

10.6.1 600* Search Failure

The user agent or proxy server understood the user's address, but the request was unsuccessful in contacting the user. A proxy might return this error towards the initiator if an attempt to contact a server failed for an unknown reason.

10.6.2 601* Not known here

The call was unsuccessful because the user or service was not known at the address called. This is not a definitive failure; the address may be valid at another server.

10.6.3 602* Not currently here

The call was unsuccessful because although the the user or service was known at the address called, the user or service is not currently located at this address. This is not a definitive failure; the user may be contactable at another server.

10.6.4 603* Alternative Address

The call was unsuccessful because the user or service is not available at this location, but one or more alternative non-definitive locations are suggested to try in addition to any that may already be being tried. A 603 response MUST NOT suggest any of the hosts in the request's path as an alternative location.

10.7 Example: Normal Replies

An example reply is given below. The first line of the reply states the SIP version number, that it is a "200 OK" reply, which means the request was successful. The Via header are taken from the request, and entries are removed hop by hop as the reply retraces the request's path. A new authentication field is added by the invited user's agent if required. The session ID is taken directly from the original request, along with the request header. The original sense of From field is preserved (i.e, it's the session originator).

In addition, a Contact-host field is added giving details of the host the user was located on, or alternatively the relevant proxy contact point which should be reachable from the invitation initiator's host.

```
SIP/2.0 200 128.16.64.19/65729
Via: SIP/2.0/UDP 239.128.16.254 16
Via: SIP/2.0/UDP 131.215.131.131
Via: SIP/2.0/UDP 128.16.64.19 1
From: mjh@isi.edu
To: schooler@cs.caltech.edu
Contact-host: 131.215.131.147
```

This same format is used for replies for other categories of reply, except that some of then may require payloads to be carried.

If the invited user's agent requires confirmation of receipt of a "200 OK" reply, it may optionally add an additional Confirm: required header to the body of the message specifying that an acknowledgment is required. This is only permitted with category 2xx replies. An example is:

```
SIP/2.0 200 128.16.64.19/65729
Via: SIP/2.0/UDP 239.128.16.254 16
Via: SIP/2.0/UDP 131.215.131.131
Via: SIP/2.0/UDP 128.16.64.19
From: mjh@isi.edu
To: schooler@cs.caltech.edu
Contact-host: 131.215.131.147
Confirm: required
```

In response to such a request, the invitation initiators agent should retransmit its request with an additional Confirm header, with the value "true" or "false" stating whether the session still exists or no longer exists respectively (see section 7.1 for details). An example of an confirmation request is:

```
INVITE 128.16.64.19/65729 SIP/2.0
Via: SIP/2.0/UDP 239.128.16.254:70 16
Via: SIP/2.0/UDP 131.215.131.131
Via: SIP/2.0/UDP 128.16.64.19
From: mjh@isi.edu
To: schooler@cs.caltech.edu
Confirm: true
Content-type: application/sdp
Content-Length: 187

v=0
o=user1 2353655765 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
```

Such confirmations are still useful when TCP transport is used as they provide application level confirmation rather than transport level confirmation. If they are not used, it is possible that a “200 OK” response may be received after the application making the call has timed out the call and exited.

10.7.1 Redirects

“603 alternative address” replies and 301 and 302 moved replies should specify another location using the Location field.

An example of a “603 alternative address” reply is:

```
SIP/2.0 603 128.16.64.19/65729
Via: SIP/2.0/UDP 131.215.131.131 1
Via: SIP/2.0/UDP 128.16.64.19
From: mjh@isi.edu
To: schooler@cs.caltech.edu
Location: 239.128.16.254 16
Content-length:0
```

In this example, the proxy (131.215.131.131) is being advised to contact the multicast group 239.128.16.254 with a ttl of 16. In normal situations a server would not suggest a redirect to a local multicast group unless (as in the above situation) it knows that the previous proxy or client is within the scope of the local group.

For unicast 603 redirects, a proxy MAY query the suggested location itself or send MAY the redirect on back towards the client. For multicast 603 redirects, a proxy SHOULD query the multicast address itself rather than sending the redirect back towards the client as multicast may be scoped and this allows a proxy within the appropriate scope regions to make the query.

For 301 or 302 redirects, a proxy SHOULD send the redirect on back towards the client and terminate any other searches it is performing for the same request. Multicast 301 or 302 redirects MUST NOT be generated.

10.8 Alternative Services

An example of an “350 Alternative Service” reply is:

```
SIP/2.0 350 128.16.64.19/32492/2
Via: SIP/2.0/UDP 131.215.131.131
Via: SIP/2.0/UDP 128.16.64.19
From: mjh@isi.edu
To: schooler@cs.caltech.edu
Contact-host: IN IP4 131.215.131.131
```

```
Content-type: application/sdp
Content-length: 146

v=0
o=mm-server 2523535 0 IN IP4 131.215.131.131
s=Answering Machine
i=Leave an audio message
c=IN IP4 128.16.64.19
t=0 0
m=audio 12345 RTP/AVP 0
```

In this case, the answering server provides a session description that describes an "answering machine". If the invitation initiator decides to take advantage of this service, it should send an invitation request to the contact host (131.215.131.131) with the session description provided. This request should contain a different session id from the one in the original request. An example would be:

```
INVITE 128.16.64.19/32492/3 SIP/2.0
Via: SIP/2.0/UDP 128.16.64.19
From: mjh@isi.edu
To: schooler@cs.caltech.edu
Content-type: application/sdp
Content-length: 146

v=0
o=mm-server 2523535 0 IN IP4 128.16.5.31
s=Answering Machine
i=Leave an audio message
c=IN IP4 128.16.64.19
t=0 0
m=audio 12345 RTP/AVP PCMU
```

Invitation initiators can choose to treat a "350 Alternative Service" reply as a failure if they wish to do so.

10.8.1 Negotiation

A "406 Not Acceptable" reply means that the user wishes to communicate, but cannot support the session described adequately. The "406 Not Acceptable" reply contains a list of reasons why the session described cannot be supported. These reasons can be one or more of:

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

406.2 Incompatible Protocol: one or more protocols described in the request is not available.

406.3 Incompatible Format: one or more media formats described in the request is not available.

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

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

Other reasons are likely to be added later. It is hoped that negotiation will not frequently be needed, and when a new user is being invited to join a pre-existing lightweight session, negotiation may not be possible. If is up to the invitation initiator to decide whether or not to act on a "406 Not Acceptable" reply.

A complex example of a "406 Not Acceptable" reply is:

```
SIP/2.0 406 128.16.64.19/32492/5
From: mjh@isi.edu
To: schooler@cs.caltech.edu
Contact-host: 131.215.131.131
Reason: 406.1, 406.3, 406.4
Content-Type: meta/sdp
Content-Length: 50

v=0
s=Lets talk
b=CT:128
c=IN IP4 131.215.131.131
m=audio 3456 RTP/AVP 7 0 13
m=video 2232 RTP/AVP 31
```

In this example, the original request specified 256 kb/s total bandwidth, and the reply states that only 128 kb/s is available. The original request specified GSM audio, H.261 video, and WB whiteboard. The audio coding and whiteboard are not available, but the reply states that DVI, PCM or LPC audio could be supported in order of preference. The reply 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.

Invitation initiators MAY choose to treat "406 Not Acceptable" replies as a failure if they wish to do so.

10.9 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 reply is larger than the MTU (or default 1,000-byte limit if the MTU is not known). To reduce this problem, a more compact form of SIP is also defined by using alternative names for common header fields. These short forms are NOT abbreviations, they are field names. No other abbreviations are allowed.

short field name	long field name	note
a	Confirm	from “acknowledge”
c	Content-Type	
e	Content-Encoding	
f	From	
h	Contact-Host	
l	Content-Length	
m	Location	from ”moved”
r	Reason	
t	To	
v	Via	

Thus the header in section ?? could also be written:

```
INVITE 128.16.64.19/65729 SIP/2.0
p:IN IP4 UDP 239.128.16.254 1 16
p:IN IP4 UDP 131.215.131.131 1
p:IN IP4 UDP 128.16.64.19 1
f:mjh@isi.edu
t:schooler@cs.caltech.edu
c:application/sdp
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
```

Mixing short field names and long field names is allowed, but not recommended. Servers MUST accept both short and long field names for requests. Proxies MUST NOT translate a request between short and long forms if authentication fields are present.

11 SIP Transport

SIP is defined so it can use either UDP or TCP as a transport protocol.

UDP has advantages over TCP from a performance point of view, as the SIP application can keep control of the precise timing of retransmissions, and can also make simultaneous call attempts to many potential locations of many users without needing to keep TCP connection state for each connection.

TCP has the advantage that clients are simpler to implement because no retransmission timing code needs to be written and also that it is possible to have a single server serving SIP and HTTP with very little extra code.

With UDP, all the additional reliability code is in the client. It is recommended that servers SHOULD implement both TCP and UDP functionality as the additional server code required is very small.

Clients MAY implement either TCP or UDP transport or both as they see fit.

11.1 Reliability using UDP transport

The Session Invitation Protocol is straightforward in operation. Only the initiating client needs to keep any state regarding the current connection attempt. SIP assumes no additional reliability from IP. Requests or replies may be lost. A SIP client SHOULD simply retransmit a SIP request until it receives a reply, or until it has reached some maximum number of timeouts and retransmissions. If the reply is merely a 1xx Informational progress report, the initiating client SHOULD still continue retransmitting the request, albeit less frequently.

When the remote user agent or server sends a final 2xx or 4xx response (not a 1xx report), it cannot be sure the client has received the response, and thus SHOULD cache the results until a connection setup timeout has occurred to avoid having to contact the user again. The server MAY also choose to cache 3xx or 6xx responses if the cost of obtaining the response outweighs the cost of caching it.

It is possible that a user can be invited successfully, but that the reply that the user was successfully contacted may not reach the invitation initiator. If the session still exists but the initiator gives up on including the user, the contacted user has sufficient information to be able to join the session. However, if the session no longer exists because the invitation initiator "hung up" before the reply arrived and the session was to be two-way, the conferencing system should be prepared to deal with this circumstance.

One solution is for the initiator to acknowledge the invitee's "200 OK" reply. Although not required, in the case of a successful invitation the invited user's agent can make a confirmation request in its "200 OK" reply. In this case the initiator's agent sends a single request with a reply Confirm: true if the request was still valid or a reply Confirm: false if it was not so that a premature hang-up can be detected without a long timeout. Such a confirmation request may be retransmitted by the invited user's agent if it so desired. Confirmation requests can only be made with "200 OK" replies, and only the invitation initiator's agent may issue the actual confirmation.

Only a "200 OK" reply warrants such a confirmation handshake, because it is the only situation where user-relevant state may be instantiated anywhere other than at the initiator's client. In all other cases, it is not necessary that state is maintained. In particular, when a server makes multiple proxy requests, "5xx Server Error" and "6xx Search Response" replies do not immediately get passed back to the invitation initiator, and so no end-to-end acknowledgment of a failed request is possible.

11.2 Reliability using TCP transport

TCP is a reliable transport protocol, and so we do not need to define additional reliability mechanisms. However, we must define rules for connection closedown under normal operation.

The normal mode of operation is for the client (or proxy acting as a client) to make a TCP connection to the well-known port of a host housing a SIP server. The client then sends the SIP request to the server over this connection and waits for one or more replies. The client MAY close the connection at any time.

The server MAY send one or more 1xx Informational responses before sending a single 2xx, 3xx, 4xx, 5xx or 6xx reply. The server MUST NOT send more than one reply, with the exception of 1xx responses. 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).

The same application-level confirmation rules apply for TCP as for UDP.

12 Searching

A basic assumption of SIP is that a location server at the user's home site either knows where the user resides, knows how to locate the user, or at the very least knows another location server that possibly might have a better idea. How these servers get this information is outside the scope of SIP itself, but it is expected that many different user-location services will exist for some time. SIP is designed so that it does not care which location service SIP servers actually employ.

12.1 Proxy servers: Relaying and Redirection

If a proxy server receives a request for a user whose location it does not know, and for whom it has no better idea where the user might be, then the server should return a "601 Not Currently Here" reply message.

If the server does have an idea how to contact the user, it can either forward (relay) the request itself, or can redirect the invitation initiator to another client that is more likely to know by sending a 603, 301 or 302 response as appropriate. It can also gateway the request into some other form if some other invitation protocol is in use in a region containing the invited user, though in doing so the server is likely to give up being stateless.

Whether to relay the request or to redirect the request is up to the server itself. For example, if the server is on a firewall machine, then it will probably have to relay the request to servers inside the firewall. Additionally, if a local multicast group is to be used for user location, then the server is likely to relay the request. However, if the user is currently away from home, relaying the request makes little sense, and the server is more likely (though not compelled) to send a redirect reply. SIP is policy-free on this issue. In general, local searches are likely to be better performed by relaying whereas wide-area searches are likely to be better performed by redirection.

When SIP uses UDP transport, clients and servers can make multiple simultaneous requests to locate a particular user at low cost. This greatly speeds up any search for the user, and in most cases will only result in one successful response. Although several simultaneous paths may reach the same host, successful responses arriving from multiple paths will not confuse the client as they should all contain the same successful host address. However, this does imply that paths with many levels of relaying should be strongly discouraged as if the request is fanned out at each hop and relayed many times, request implosions could result. Thus servers that are not the first hop servers in a chain of servers SHOULD NOT make multiple parallel requests, but should send a redirection response with multiple alternatives. Thus a firewall host can still perform a parallel search but can control the fanout of the search.

12.2 Parallel Searches: Initiator Behavior

The session initiator may make a parallel search for a user. This can occur when DNS resolution results in multiple addresses, or when contacting a remote server results in a "603 Alternative Address" response containing multiple addresses to try. All such parallel searches for the same SIP request MUST contain the same SIP Id, though the sequence number (given in the Path field) SHOULD be different for each of the parallel searches.

Whilst performing a parallel search, different responses may result from different servers, and it is important for the initiating client to handle these responses correctly. In general, the following rules apply:

- If a 2xx response is received, the invitation was successful, the user should be informed and all pending requests should be terminated and/or ignored.
- If a 4xx response is received the invitation has definitively failed, the user should be informed, and all pending requests should be terminated and/or ignored.
- If a 3xx response is received, the search should be terminated and all pending requests should be terminated and/or ignored. However, further action MAY be taken depending on the actual reply without informing the user or alternatively the invitation MAY be regarded as having failed in which case the user MUST be informed.
- If a 5xx or 6xx response is received, the particular server responding is removed from the parallel search and the search continues. If a "603 Alternative Address" response is received, the search may be expanded to include those servers listed in the response that have not already responded. The user SHOULD NOT be informed unless there are no other servers left to try, in which case the user MUST be informed.
- If a 1xx response is received, the search continues. The user MAY be informed as deemed appropriate.

12.3 Parallel Searches: Proxy Behavior

In the same way that an Initiating Client can discover multiple addresses to try, a proxy server can also discover multiple addresses that it may try. For a proxy server to be stateless, it must not make multiple SIP requests because it would then be possible to return a 5xx or 6xx response to the Initiating Client and afterwards obtain a definitive answer. To be able to make multiple parallel SIP requests, it must keep state as to the replies it has already received and MUST NOT return any reply other than 1xx informational replies until it has received a definitive reply or has no further addresses to try.

Thus faced with DNS resolution giving multiple addresses, a proxy server that wishes to be stateless should only send a SIP request to the first address. Similarly a stateless proxy should not attempt to send SIP request to multiple addresses given in a "603 Alternative Address" response that is returned it it, but should forward such a response back towards the initiator.

Proxies that wish to keep state should follow the following rules regarding responses obtained during a parallel search:

- If a 2xx response is received, the invitation was successful, the 2xx response should be forwarded back towards the initiator, and all pending requests should be terminated and/or ignored.
- If a 4xx response is received the invitation has definitively failed, the 4xx response should be forwarded back towards the initiator, and all pending requests should be terminated and/or ignored.
- If a 3xx response is received the invitation is regarded by the proxy as having failed, the 3xx response should be forwarded back towards the initiator, the search should be terminated and all pending requests should be terminated and/or ignored.

- If a 5xx or 6xx response is received, the particular server responding is removed from the parallel search and the search continues. If a "603 Alternative Address" response is received, the search may be expanded to include those servers listed in the response that have not already responded. No response other than a periodic "100 Trying" response should be send towards the initiator unless there are no other servers left to try, in which case a response SHOULD be sent as described below.
- If a 1xx response is received, the search continues. The 1xx response MAY be forwarded towards the initiator as appropriate.

If a proxy had exhausted its search and still not obtained a definitive response (it received only 1xx, 5xx, and 6xx responses) the proxy should cache these responses and return the first response from the following ordered list:

1. 503 Service Unavailable;
2. 500 Server Internal Error;
3. 501 Not Implemented;
4. any other 5xx error not yet defined;
5. 600 Search Failure;
6. 602 Not Currently Here;
7. 601 Not Known Here;
8. any other 6xx error response not yet defined.

If a proxy has exhausted its search and the only response it has received has been "603 Alternative Address", then the proxy should send a "600 Search Failure" response if any connection attempt timed out or failed, or it should send "602 Not Currently Here" if two or more "603 Alternative Address" responses only provide references to each other.

12.4 Change of Transport at a Proxy

Editors note: this section is still incomplete. Several options exist for where the responsibility should lie for retransmissions from proxies between TCP and UDP transport. This section generally assumes local retransmission, but end-to-end transmission through a chain of proxies is also possible.

It is possible that a proxy server will receiver a request using TCP and relay it onwards using UDP or vice-versa. SIP does not assume end-to-end reliability even when the initiating client is using TCP, but a SIP client sending a request over TCP MAY assume that the request has been received by the server it sent the request to. Retransmission of the request is then not the responsibility of the client. However, a called user agent SHOULD NOT assume that a 2xx success response has been received by the invitation initiator, even if all the path fields in the request indicated TCP transport because it cannot be certain all those TCP connections still exist. If the called user agent requires knowledge that the response did reach the invitation initiator, it MAY add a Confirm: required field to the reply as it would if the response was sent using UDP.

In the following, the term "TCP-UDP proxy" is used to mean a proxy that received a request using TCP and relayed it using UDP. Similarly a "TCP-UDP proxy" receives a reply using UDP and should relay it using TCP.

12.4.1 Retransmission from a TCP-UDP Proxy

A proxy receiving a request with TCP transport and forwarding that request using UDP becomes responsible for retransmission of the request as required and for timing out the request if no answer is forthcoming.

12.4.2 Retransmissions arriving at a UDP-TCP Proxy

A proxy receiving a request using UDP transport and forwarding that request using TCP transport may have have SIP request state associated with that TCP connection or SIP response state associated with it.

If such a proxy receives a retransmission of the UDP request whilst in the state or awaiting a response (i.e, has *request* state), it SHOULD NOT forward the duplicate request into the TCP connection unless the request has been modified, but instead SHOULD respond with a "100 Trying" response sent back towards the initiator.

Note: This behavior is different from a UDP-UDP proxy which MUST forward the retransmitted request and MAY additionally respond with a "100 Trying" response sent back towards the initiator.

If such a proxy receives a retransmission of the UDP request in *response* state (i.e, it has already sent a definitive response) then the proxy MAY retransmit that response if it has cached it. Alternatively if it has not cached the response, it SHOULD resend the request towards the called user agent, either via an existing TCP connection if there is one or via a new TCP connection if there is not, to obtain a retransmission of the response. In the latter case, the proxy MAY additionally respond with a "100 Trying" response sent back towards the initiator.

Note: This behavior is the same as a UDP-to-UDP proxy in the same circumstances.

12.4.3 Confirmation arriving at a TCP-UDP Proxy

One possible event that may occur is that whilst performing a search using UDP, a response may arrive that should be relayed back towards the initiator using TCP, but the TCP connection has been terminated by the initiator. In this case the proxy MUST NOT attempt to relay the response (by opening a TCP connection) and should terminate any outstanding search. In this circumstance only, if the response was a "200 OK" response with a Confirm: required field, the proxy MAY resend the request to the Contact Host with a Confirm: false field to speed hang-up discovery at the called user agent.

12.4.4 Confirmation sent from a UDP-TCP Proxy

Normally a response that arrives at a proxy using TCP that should be sent back towards the initiator using UDP should be sent once, and should only be resent if the request is resent from the UDP proxy closer to the initiator. However, this does not allow for reliable confirmation.

13 Using Variants for Terminal Negotiation

Redirection allows the called party to indicate several communication alternatives to the caller using the 300 (Multiple Choices) response, all reachable using a single published communication identifier.

The Alternates header in the response contains the variant list. The response may contain an entity, typically of content type text/html, providing guidance to the user. The calling user agent is free to ignore this part and solely rely on the Alternates header.

```
SIP/2.0 300 Multiple Choices
Date: Thu, 06 Mar 1997 10:08:55 GMT
Alternates:
   {"hgs@erlang.cs.columbia.edu" 0.9 {mobility fixed} {class business}
     {service IP, voice-mail} {media all} {duplex full}},
   {"+12129397042" 0.8 {mobility fixed} {class business}
     {service POTS} {media audio} {duplex full}},
   {"+12129397000" 0.7 {mobility fixed} {class business}
     {service ISDN, attendant} {media audio} {duplex full}
     {language en, es, iw}},
   {"+12125551212" 0.6 {mobility mobile} {class personal}
     {service POTS} {media audio} {duplex full}}
  }
Content-Type: text/html
Content-Length: 283

<html>
You can reach <a href="http://www.cs.columbia.edu/˜doe">John Doe</a> at

<ul>
<li><a href="sip://hgs@erlang.cs.columbia.edu">Internet telephony</a>

<li><a href="phone://+1219397042">analog phone</a>

<li>...

</dl>
</html>
```

13.1 Variant Description

A variant can be described in a machine-readable way with a variant description [7].

```
variant-description =
  "{" <"> UCI <"> communication-quality *variant-attribute "}"

communications-quality = qvalue

variant-attribute = "{" "mobility"  ( "fixed" | "mobile" ) "}"
                  | "{" "class"     ( "personal" | "business" ) "}"
                  | "{" "language"  1#language-tag "}"
                  | "{" "service"   1#service-tag "}"
                  | "{" "media"     1#media-tag "}"
                  | "{" "features"  feature-list "}"
```

```
                    | "{" "description" quoted-string "}"
                    | "{" "duplex"    ( "full" | "half" | "receive-only" |
                                        "send-only" ) "}"
                    | extension-attribute

  extension-attribute = "{" extension-name extension-value "}"
  extension-name      = token
  extension-value     = *( token | quoted-string | LWS |
                           extension-specials )
  extension-specials  = <any element of tspecials except <"> and "}">

  language-tag        = <see [H3.10]>
  service-tag         = fax | IP | POTS | pager | voice-mail |
                        attendant
  media-tag           = <see SDP: audio | video | ... >
  feature-list        =
```

Attributes which are unknown should be omitted. New tags for class-tag and service-tag can be registered with IANA. The media tag uses Internet media types, e.g., audio, video, application/x-wb, etc. This is meant for indicating general communication capability, not the support for specific encodings. It should be sufficient to allow the caller to choose an appropriate communication address.

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