

TTC STANDARD

JJ-90.24

Technical Specifications on Basic Call Interface for SIP Terminals Connecting with Provider's SIP Network

(English Edition)

Version 2.0

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THE TELECOMMUNICATION TECHNOLOGY COMMITTEE



Introduction

This document provides the TTC original Standard formulated by the TTC Signaling working group.

The working group translated JJ-90.24 Japanese Version 2.0 (June 1,2006) into English, and issued JJ-90.24 English Version on Sep. 2, 2009.

In case of dispute, the original to be referred is the Japanese Edition of the text.

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TTC Signaling Working Group

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<Reference>

1. Relation to International Recommendations

There are no international recommendations related to this standard.

2. Revision History

Version No.	Date	Description
Version 1.0	June 2, 2005	Initial publication (revision of TS-1006, Version 1.0) Revisions to TS-1006 are as follows. <ul style="list-style-type: none">•Changes to main document:<ul style="list-style-type: none">- Add description of Expires header in REGISTER method in Section 4.- In Table 4-3, change handling of userinfo to "m".- In Section 7.1, add description of maddr parameter.- In Section 10.2.1, add a postscript stating that the value used for fmt list in support of G.711 is 0.- In Section 10.2.2, change description of "o=" line version value.•Add Appendix 2•Add Appendix 3•Correct typographical mistakes
Version 1.1	August 25, 2005	Reference document is modified because "session timer" has been approved to be assigned an official RFC number in IETF
Version 2.0	Sep 2, 2009	Revisions to version 1.1 are as follows. <ul style="list-style-type: none">•Add descriptions for wideband codecs.<ul style="list-style-type: none">- In Section 10.2.1, add description of Offer/Answer for wideband codecs.- In Appendix ii, add optional items.- Add appendix iv as examples of SDP setting of wideband codecs.•Correct typographical mistakes

3. References

3.1 Normative References

- [1] "SIP: Session Initiation Protocol," TTC Standard JF-IETF-RFC3261, Ver. 1, The Telecommunication Technology Committee (TTC), June 2005.
- [2] "Reliability of Provisional Responses in SIP," TTC Standard JF-IETF-RFC3262, Ver. 1, The Telecommunication Technology Committee (TTC), June 2005.
- [3] "An Offer/Answer model with Session Description Protocol (SDP)," TTC Standard JF-IETF-RFC3264, Ver. 1, The Telecommunication Technology Committee (TTC), June 2005.
- [4] "The Session Initiation Protocol UPDATE Method," TTC Standard JF-IETF-RFC3311, Ver. 1, The Telecommunication Technology Committee (TTC), June 2005.
- [5] "A Privacy Mechanism for the Session Initiation Protocol (SIP)," TTC Standard JF-IETF-RFC3323, Ver. 1, The Telecommunication Technology Committee (TTC), June 2005.
- [6] "Short Term Requirements for Network Asserted Identity," TTC Standard JF-IETF-RFC3324, Ver. 1, The Telecommunication Technology Committee (TTC), June 2005.
- [7] "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Domains," TTC Standard JF-IETF-RFC3325, Ver. 1, The Telecommunication Technology Committee (TTC), June 2005.
- [8] "The tel URI for Telephone Numbers," TTC Standard JF-IETF-RFC3966, Ver. 1, The Telecommunication Technology Committee (TTC), June 2005.
- [9] "SDP: Session Description Protocol," TTC Standard JF-IETF-RFC2327, Ver. 1, The Telecommunication

Technology Committee (TTC), June 2005.

- [10] “RTP: A Transport Protocol for Real-Time Applications,” TTC Standard JF-IETF-STD64, The Telecommunication Technology Committee (TTC), June 2005.
- [11] “RTP Profile for Audio and Video Conferences with Minimal Control,” TTC Standard JF-IETF-STD65, The Telecommunication Technology Committee (TTC), June 2005.
- [12] “The International Public Telecommunications Numbering Plan,” ITU-T Recommendation E.164, ITU-T, 1997.
- [13] “Pulse Code Modulation (PCM) of Voice Frequencies,” ITU-T Recommendation G.711, ITU-T, 1998.
- [14] “Technical Specification of the Framework on Provider’s SIP Networks,” TTC Standard JJ-90.21, The Telecommunication Technology Committee (TTC), June 2005.
- [15] “Technical Specification on Network Asserted User Identity Information Transferring through Provider’s SIP Networks,” JJ-90.22, The Telecommunication Technology Committee (TTC), June 2005.
- [16] “Session Timers in the Session Initiation Protocol (SIP)”, JF-IETF-RFC4028, The Telecommunication Technologies Committee (TTC), August 2005.
- [17] “RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals”, TTC Standard JF-IETF-RFC2833, The Telecommunication Technology Committee (TTC), June 2006.

3.2 Informative References

- [18] “Technology Report on Session Initiation Protocol (SIP),” TR-1007, The Telecommunication Technology Committee (TTC), March 2003.
- [19] “7KHZ AUDIO-CODING WITHIN 64 KBPS”, ITU-T Recommendation G.722, ITU-T, 1988.
- [20] “Low-complexity coding at 24 and 32 kbit/s for hands-free operation in systems with low frame loss”, ITU-T Recommendation G.722.1, ITU-T, 2005.

4. Industrial Property Rights

Information regarding submittal of TTC’s “The Policy for the Handling of Industrial Property Rights” is available on TTC’s home page.

5. Responsible working group

Signaling Working Group

1. Overview

1.1. Scope

This standard describes recommended specifications for the processing and behavior of SIP terminals connected to a provider's SIP network via a user call interface (Interface B) for the network-connection architecture specified in JJ-90.21 [14].

This standard does not prevent terminals connecting via Interface B from having capabilities exceeding the scope of this standard. However, connectivity with terminals conforming to this standard should always be taken into account.

1.2. Objectives and Provisions of this Standard

In conjunction with the implementation of SIP terminals, this standard aims to be:

- A implementation viable standard by providing a uniform interpretation of the provisions covering connection conditions;
- A standard that can be commonly applied to connections with a comprehensive provider SIP network that connects to diverse communication business formats; and
- A standard that gives full consideration to connectivity with SIP user agents (UA) that either beyond the scope of these provisions or do not strictly observe them.

To meet the above objectives, the following provisions are specified:

- Items covering the usage of SIP and its extended provisions as specified in JF-IETF-RFC3261 [1] as call-control-signal conditions and registration-signal conditions;
- Items covering media-capability conditions, SDP, and RTP as voice media conditions¹; and
- Items covering behaviour of SIP terminals in relation to other establishment of a call and registration behaviour.

1.3. Contents of this Standard

The main body of this standard:

- Specifies behaviours at time of location registration (Section 4), at time of outgoing (Section 5), at time of incoming (Section 6), and at time of call release (Section 11) as provisions for connecting to a provider's SIP network, and specifies processing of record routes (Section 7) where needed;
- Specifies detailed behaviours in relation to 100rel (Section 8) and session timer (Section 10) as function extensions whose implementation may be necessary;
- Specifies behaviours needed for the offer/answer model to establish a voice call (Section 10);
- Specifies behaviours for caller number display when making and receiving a call as a provision for additional services (Section 12); and
- Specifies message format (Section 13).

Appendix i : Provides sequence examples

Appendix ii : Lists setting options in table form.

Appendix iii : Describes the background to the making of this standard and its role.

¹ Connectivity with SIP UA supporting media other than voice must also be taken into account.

Appendix iv: Describes SDP setting examples for the use of wideband codec, etc.

1.4. Terminology

Terms used in this standard are defined below.

AoR	:	Address of Record. A character string in SIP Uniform Resource Identifier (URI) format assigned to a user by the provider of the SIP network and bound to a telephone number.
re-INVITE	:	An INVITE request message sent during an existing dialog.
Offer	:	An SDP Offer in the SDP offer/answer model specified in JF-IETF-RFC3264 [3].
Answer	:	An SDP Answer in the SDP offer/answer model specified in JF-IETF-RFC3264 [3].
SIP proxy server address	:	Address of a SIP proxy server that sends and receives SIP messages so that SIP terminals can establish a session; given in Fully Qualified Domain Name (FQDN) format or IPv4 address format.
SIP registrar address	:	Address of a SIP registrar that enables the provider's SIP network to dynamically accept a SIP terminal's Register registration of its location so that an Initial INVITE request can be forwarded with respect to a specific AoR; given in FQDN format or IPv4 address format.
SIP terminal	:	The generic name for terminal facilities equipped with a SIP user agent for connecting with the provider's SIP network and communicating by voice. It is the target of this standard.
SIP UA	:	An ordinary SIP user agent existing either inside or outside a provider's SIP network. SIP UA also represents SIP terminal.
Authentication request	:	A request that includes an Authorization or Proxy-Authorization header set with authentication credentials based on the HTTP Digest authentication scheme.
Password	:	A character string associated with a user name and used when performing authentication by the HTTP Digest scheme.
User	:	The generic name for the SIP-terminal user (a human being or other terminal facility connected to a telephone), or an upper-layer application that takes control of a SIP UA.
User name	:	A character string used when performing authentication by the HTTP Digest scheme.
User resources	:	The generic name for resources provided within a SIP terminal to enable a user to perform call control or the sending/receiving of voice media in SIP terminal (for example, a 2W analog interface in a home gateway (GW) connected to a 2W analog telephone).
Location information	:	Position of a SIP UA on an IP network with respect to a specific AoR at some instant in time. It is expressed in SIP URI format and includes IP address and port number which are used for SIP message destination

2. Various Definitions

2.1. Table-related Definitions

Table 2-1 defines common codes used in the tables of this standard to indicate provision type.

Table 2-1/ JJ-90.24: Definition of provision types appearing in tables

Code	Code Name	Meaning
m	mandatory	This function must be supported. Here, “mandatory” means that the referenced provision is available as a function. It does not mean, however, that it must always be expressed. It means, rather, that it must be present if deemed necessary by the behaviour specified in the referenced document. If, for example, this code were to be specified for a transmission parameter, it does not mean that that parameter must always be included, but that it must be included as needed in accordance with the specifications of the referenced document (dynamic application requirements).
o	optional	Support of the capability in question depends on the implementation; it may or may not be supported.
n/a	not applicable	Conditions for support of this function are meaningless. (Shows standard prescribing this function.) Support string may be omitted.
x	prohibited (excluded)	This function must not be used.
c <integer>	conditional	Conditions for support of this function (m, o, n/a, x) depends on other selectable conditions (indicated by c<integer>).
o.<integer>	qualified optional	Indicates an option that may be exclusively selected from among that indicated by <integer>.
i	irrelevant	Indicates a provision outside the scope of this standard; may prove useful if base specifications become the base for another standard.

3. Lower Layer

The sending and receiving of SIP messages is performed over UDP/IPv4. This provision does not preclude the use of TCP, TLS, IPv6, etc.

4. Location Registration

This section specifies the behaviours for registering the location of a SIP terminal by the REGISTER method ².

4.1. Sending a REGISTER Request

4.1.1. Basic behaviour

A SIP terminal sends a REGISTER request addressed to a SIP registrar in the provider’s SIP network in order to register with the network its location with respect to the AoR that it possesses. The request is sent directly to the SIP registrar or via a SIP proxy server in the provider’s SIP network. The registered location is used to determine location of the SIP terminal when a call is received at that AoR and to forward an Initial INVITE to the address indicated by location information.

In the event that a 401 (Unauthorized) response is received in response to a REGISTER request, authentication processing MUST be performed as described in Section 4.2.

4.1.2. Request-URI

The Request-URI of a REGISTER request is configured in SIP URI format. The configuration of a Request-URI in other than SIP URI format is beyond the scope of this standard.

² The specifications of this section do not apply when connecting to a provider’s SIP network that does not use the REGISTER method to determine location.

The Request-URI is configured as follows.

```
Request-uri = "sip:" host [ :port ] [ uri-parameters ]
```

Table 4-1/JJ-90.24: Configuration of Request-URI when sending a REGISTER request

Element	Reference	Type (caller)	Remarks
host	[1]	m	Generally, the SIP proxy server address (may be different than the AoR host character string)
port	[1]	c1	
uri-parameters	[1]	o	May be omitted. Detail specifications lie beyond the scope of this standard.
c1: "o" if using 5060; "m" otherwise ³			

4.1.3. REGISTER Request Header

4.1.3.1. Setting of Essential Header Fields

Table 4-2 describes the configurations for a SIP terminal of essential header fields in a REGISTER request as found in Table 2 and Table 3 of Section 20 in JF-IETF-RFC3261 [1].

³ The port number used may be specified by the provider's SIP network.

Table 4-2/JJ-90.24: Setting of essential header fields when sending a REGISTER request

Field	Value	Remarks
To	Specifies AoR (SIP URI) targeted by location registration.	
From	Specifies AoR (SIP URL) targeted by location registration (in this case, same as To header field).	A function could be provided that changes this setting and sets a third-party AoR different from the AoR targeted by location registration ⁴ .
Call-ID	<Global-Unique-ID>	When performing location registration using the same Call-ID as that used by a past location registration, it must be kept in mind that a REGISTER request that does not set CSeq to a value larger than its value at that time is not allowed. (For example, a REGISTER request made after rebooting a SIP terminal would not be allowed at a SIP registrar if the same Call-ID was specified without incrementing CSeq.)
Via	<via-param>	Conforms to JF-IETF-RFC3261 [1].
CSeq	"integer value" LWS Method	Conforms to JF-IETF-RFC3261 [1]; a value that can be expressible as a 32-bit unsigned integer
Max-Forwards	70	Conforms to JF-IETF-RFC3261 [1].

4.1.3.2. Setting of Contact Header Fields

Table 4-3 describes the configurations of Contact header fields in a REGISTER request as location information. (Note that use of "*" for complete deletion of all registered location information is described in Section 4.1.5 and that processing for confirmation of location information is described in Section 4.1.7.)

For reasons described below, it is desirable that the userinfo field of the Contact header be specified as a random alphanumeric character string not easily inferred by another person as prescribed in Section 13.2.2 ⁵.

- The possibility of sustaining damage from a prank call due to a fraudulent Initial INVITE request not coming from the provider's SIP network is high if that value can be easily inferred by another person such as when using the user's name or telephone number (judging the validity of a received Initial INVITE request is described in Section 6.1).
- In the case of a SIP terminal that automatically obtains an IP address as in DHCP or PPPoE, the unexpected occurrence of a fault (such as a power failure) would prevent the sending of a REGISTER request for deleting location information, which would mean that the location information would continue to be stored on the SIP registrar side. If the IP address in question were then to be assigned to another SIP terminal, the Initial INVITE request that should have been received by the SIP terminal that had the unexpected fault could possibly be received by the SIP terminal that obtained the same IP address. This ostensibly erroneous receive behaviour can be prevented by checking the userinfo field for consistency when receiving an Initial INVITE request.

⁴ The provider's SIP network policy may or may not allow a third-party AoR different from the AoR targeted by location registration.

⁵ The necessity and manner of setting the userinfo field in the Contact header may be defined outside the scope of this standard in a provider's SIP network.

Table 4-3/JJ-90.24: Setting of contact header parameters in a REGISTER request

Parameter	Value	Type (caller)	Remarks
displayname	None	o	
addr-spec	SIP-URI	m	Formats other than SIPS-URI are outside the scope of this standard.
userinfo	character string	m	Any character string may be set in accordance with JF-IETF-RFC3261 [1].
host	IPv4address format	m	Formats other than IPv4address (such as IPv6reference and hostname) are outside the scope of this standard.
port	Omitted or set to a numerical value	c1	If omitted, the port for receiving Initial INVITE request is 5060.
user-param	Omitted or "phone" or "ip"	o	If set, the parameter MAY be ignored by the SIP registrar.
transport-param	Omitted	o	A value other than UDP is outside the scope of this standard. If set, the parameter MAY be ignored by the SIP registrar.
maddr-param	Omitted	x	
headers	Omitted	x	
other URI parameter	Omitted	x	
q parameter (c-p-q)	Omitted or a character string representing a number between 0 and 1 to the third decimal place	o	Usage depends on the provider's SIP network
expires parameter ⁶ (c-p-expires)	Omitted or a number between 0 and 4294967295	o	If set in a format other than that allowed, message may not be allowed or parameter may be ignored by the SIP registrar. If omitted, the value MAY differ according to the policy of the SIP registrar ⁷ .
contact-extension	Omitted	n/a	
c1: "o" if using 5060; "m" otherwise			

4.1.4. Registration

A SIP terminal wishing to register location information against its own AoR sends a REGISTER request to the address of a SIP registrar after setting the Contact header in that request. Any setting of the Expires parameter or Expires header shall conform to Table 4-3.

4.1.5. Deletion

A SIP terminal sends a delete REGISTER message to the SIP registrar to delete location information previously registered with the server. This may be deletion of only specific location information (specific deletion) or deletion of all location information (total deletion).

Considering that a SIP terminal may sometimes experience a power cutoff or a sequence anomaly during the shutdown process, the SIP terminal SHOULD delete all location information registered against its own AoR after startup and before registering any new location information. A total deletion SHOULD also be performed in the event that the deletion of certain previously registered location information cannot, for some reason, be guaranteed.

⁶ Content of Expires header is the same as this standard.

⁷ A value may be recommended by the provider's SIP network.

4.1.5.1. Specific Deletion

To delete specific location information bound to its own AoR, a SIP terminal sends a REGISTER request to the SIP registrar server with the Contact header of that request set with that location information and the expires parameter or Expires header field of the Contact header set to 0. Then, out of all location information bound to the AoR in question, the SIP registrar server unbinds and deletes only the location information specified in the Contact header.

4.1.5.2. Total Deletion

To delete all location information bound to its own AoR, a SIP terminal sends a REGISTER request to the SIP registrar server with the Contact header of that request set to "*" and the Expires header set to 0. The SIP registrar server then deletes all location information bound to the AoR in question.

4.1.5.3. Cautionary Notes on Deletion of Location Information

A SIP terminal is usually able to manage the location information that it has registered with a SIP registrar server with respect to the AoR that it manages. Such a terminal, however, may sometimes be operating in an environment featuring dynamic IP address allocation, which means that the terminal's own IP address may change due, for example, to the termination of an application, a reboot, or the expiration of the IP address's retention period. In such a case, the SIP terminal may not be able to manage correct location information at the SIP registrar server unless it actively deletes or updates that information at the server. This is why a SIP terminal SHOULD delete or update its location information in such a situation.

4.1.6. Update

Given that a SIP terminal has registered location information with a SIP registrar server with respect to its AoR, it may wish to update that location information and its previously set retention period before that period expires. To do so, the SIP terminal sends an update REGISTER request with the Contact header set to the previously registered location information to the SIP registrar server. If setting the expires parameter or the Expires header field at this time, setting shall conform to Table 4-3.

Update timing (T s) MUST be set so that it does not exceed the retention period (Z s), so that frequent REGISTER request submissions do not occur (such as by setting it to 0.x times⁸ the retention period (Z s)), and, in consideration of resend time, so that the remaining time in the retention period is much more than Timer F (=32 s) specified in JF-IETF-RFC3261 0.

4.1.7. Confirmation

To check the location information and associated retention periods, if any, it has registered with the SIP registrar server with respect to its own AoR, a SIP terminal MAY send a confirmation REGISTER request without setting the Contact header to the SIP registrar server.

4.2. Authentication

If a SIP terminal receives a 401 (Unauthorized) response from the SIP registrar server after sending it a REGISTER request, an authentication procedure shall be performed according to the authentication scheme specified in the WWW-Authenticate header of that response. This scheme makes use of HTTP Digest.

⁸ This value may be specified by the provider's SIP network.

4.2.1. Receiving a 401 (Unauthorized) Response

If, after sending a REGISTER request to the SIP registrar server, a SIP terminal receives a 401 (Unauthorized) response from the server, an authentication procedure shall be performed according to the authentication scheme specified in the WWW-Authenticate header of that response. Table 4-4 summarizes the handling of a received WWW-Authenticate header.

Table 4-4/JJ-90.24: Handling of WWW-Authenticate header received in a 401 (Unauthorized) response

Directive	Value	Type (callee)	Remarks
nonce	<nonce>	m	
realm	<realm>	m	realm character string
domain	<domain>	m	domain character string
opaque	<opaque>	m	Null character only in double quotes or a random string of 32 characters
stale	TRUE or FALSE	m	<p>【TRUE】 The value of the nonce parameter in the Authorization header included in the sent REGISTER request has been found by the SIP registrar server to have expired or to be malformed. The authentication procedure is to be performed again using the nonce value received in the 401 (Unauthorized) response.</p> <p>【FALSE】 If an Authorization header was not included in the sent REGISTER request (which implies a new authentication), an authentication REGISTER request is to be sent according to the specified authentication procedure. If an Authorization header was included, the SIP registrar server has judged the value of the username or response parameter in the Authentication header to be invalid and will notify the user of such in some way.</p>
algorithm	MD5	m	
qop-options	auth, auth-int	m	
other parameter	-	o	Ignored. This need not apply when used in applications.

* Items included in angled brackets (“<” and “>”) indicates a value set by the SIP registrar server to convey a certain condition or setting.

4.2.2. Sending an Authentication REGISTER Request

After receiving a 401 (Unauthorized) response, the SIP terminal sends the SIP registrar server a REGISTER request that includes an Authorization header specifying user name and password according to the specified authentication procedure. Table 4-5 summarizes the handling of the Authorization header.

Note that the value of Call-ID set in this authentication REGISTER request shall be the same as that received in the 401 (Unauthorized) response and that Cseq shall be incremented by 1.

Table 4-5/JJ-90.24: Handling of Authorization header to be sent with an Authentication REGISTER Request

Directive	Value	Type (caller)	Remarks
username	<username>	m	Username used in authentication
nonce	Copied from received WWW-Authenticate header	m	Same value as received in 401 (Unauthorized) response
realm	Copied from received WWW-Authenticate header	m	Same value as received in 401 (Unauthorized) response
uri	Same as Request-URI	m	
response	<response>	m	Value computed from password using algorithm set in WWW-Authenticate header of 401 (Unauthorized) response
opaque	Copied from received WWW-Authenticate header	m	Same value as received in 401 (Unauthorized) response
algorithm	MD5	m	Same value as received in 401 (Unauthorized) response
cnonce	Arbitrary value created by terminal	m	Set only when qop-options is set in the WWW-Authenticate header received in the 401 (Unauthorized) response
nonce-count	Number of authentication-request submissions in hexadecimal	m	Set only when qop-options is set in the WWW-Authenticate header received in the 401 (Unauthorized) response
message-qop		m	Set only when qop-options is set in the WWW-Authenticate header received in the 401 (Unauthorized) response. (*1) If set, the same value as received in 401 (Unauthorized) response.
other parameter	-	o	Not set. This need not apply when used in applications.

(*1) The message-qop directive MUST NOT be set if qop-options has not been set in the WWW-Authenticate header received in the 401 (Unauthorized) response.

4.3. Receiving a 2xx Response

4.3.1. Basic Behavior

Receiving a 2xx response after sending a REGISTER request indicates completion of a registration, deletion, update, or confirmation behavior as requested by the REGISTER request with respect to location information. The action to be taken on receiving a 2xx response for each of these behaviours is described below.

4.3.2. Receiving a 2xx Response for Registration

On receiving a 200 (OK) response from the SIP registrar server indicating completion of registration, the SIP terminal records the location information requested for addition and included in that Contact header plus the retention period (Z s) specified in the Expires header field.

Cautionary notes on comparing the location information sent by the SIP terminal to the SIP registrar server with the registered URI in the Contact header of the 200 (OK) response received from the SIP registrar server are given in Section 4.3.6.

4.3.3. Receiving a 2xx Response for Deletion

On receiving a 200 (OK) response from the SIP registrar server indicating completion of registration, the SIP terminal considers that the location information in question has been deleted at the SIP registrar server.

4.3.4. Receiving a 2xx Response for Update

On receiving a 200 (OK) response from the SIP registrar server indicating completion of an update behavior, the SIP

terminal records the location information requested for update and included in that Contact header plus the retention period (Z s) specified by the expires parameter or in the Expires header field.

Cautionary notes on comparing the location information sent by the SIP terminal to the SIP registrar server with the registered URI in the Contact header of the 200 (OK) response received from the SIP registrar server are given in Section 4.3.6.

4.3.5. Receiving a 2xx Response for Confirmation

On receiving a 200 (OK) response from the SIP registrar server indicating completion of confirmation, the SIP terminal can now examine each piece of location information included in the Contact header and the associated retention period specified by the expires parameter or in the Expires header field. Any behaviors performed after checking this information is outside the scope of this standard.

4.3.6. Cautionary Notes on Checking the Location Information for Validity

When having the SIP terminal check the location information included in the Contact header of a submitted REGISTER request against the location information included in the Contact header of a 200 (OK) response received from the SIP registrar server, it must be kept in mind that URI parameters not recognized by the SIP registrar server may not be included and maintained with location information, which means that they SHOULD NOT be targets of comparison. This warning need not apply to parameters that are especially deemed necessary such as those used in applications.

4.4. Receiving a 3xx/4xx/5xx/6xx Response

4.4.1. Basic behaviour

The SIP terminal processes a received response of this type in accordance with Table 13-4. A retry as described in Section 4.5.1 shall be performed as needed.

4.5. Considerations on Congestion Control

4.5.1. Cautionary Notes on Retry When Location Registration Fails

After submitting a REGISTER request, a SIP terminal may receive an error response that includes a Retry-After header (in particular, a 4xx-6xx response, which, in JF-IETF-RFC3261 [1], means a 404 (Not Found) response, 413 (Request Entity Too Large) response, 480 (Temporarily Unavailable) response, 486 (Busy Here) response, 500 (Server Internal Error) response, 503 (Service Unavailable) response, a 600 (Busy Everywhere) response, and a 603 (Decline) response).

In such a situation, the provider's SIP network may have some kind of problems such as congestion of the SIP registrar server, etc. To therefore avoid any further congestion, it is RECOMMENDED that the request be resent after the time interval specified in the Retry-After header (keeping in mind that an error response may be received again even after resending the request after the specified time). If an error response is received without a Retry-After header, the request SHOULD nevertheless be resent after an appropriate period of time for the same reason (excluding the reception of a 401 (Unauthorized) response).

Similarly, for the occurrence of an error in a layer below the SIP application layer (e.g., ICMP error notification, message resend timeout), a location-information registration retry SHOULD only be attempted after an appropriate period of time.

4.5.2. Cautionary Notes on Multiple REGISTER Requests

A SIP terminal might find it desirable to send multiple REGISTER requests at the same time such as when managing multiple AoRs or when registering multiple location-information entries at a SIP registrar server. To therefore avoid unnecessary congestion at a SIP registrar server due to location-information registration behaviors, consideration SHOULD be given to restricting the submission of consecutive REGISTER requests within a short time.

In addition, when receiving a 401 (Unauthorized) response from the SIP registrar server after sending a REGISTER request containing an Authentication header, the SIP terminal SHOULD refrain from resending a REGISTER request using the same user name and password (excluding the case in which the value of the stale parameter in the WWW-Authenticate header is TRUE) so as to avoid the submission of unnecessary REGISTER requests.

4.5.3. Cautionary Notes on the Occurrence of Temporary Faults

If a SIP terminal detects that it cannot send or receive SIP messages for some reason but then removes the source of the problem and returns to a state in which it can, it SHOULD immediately check whether its location information has been changed or whether binding has been updated regardless of the registration retention period (and, if registration has been deleted, proceed immediately with registration processing).

However, to avoid processing congestion at the SIP registrar server due to simultaneous location registration behaviors caused by simultaneous terminal recoveries following a wide-area failure in the access network, and to avoid unnecessary repetition of location registration behaviors due to intermittent temporary faults, the submission of REGISTER requests following fault recovery SHOULD be made only at statistically uniform time intervals within an appropriate period of time.

5. Basic Call Control Behaviors for a Outgoing Call

5.1. Sending an Initial INVITE Request

5.1.1. Basic Behaviour

To issue an outgoing call request from the user, the SIP terminal sends out an Initial INVITE request addressed to a SIP proxy server. Note here that it SHOULD be possible for an Initial INVITE request to be sent out independent of any existing SIP sequences even when an outgoing call request is made before the normal completion of a SIP transaction or dialog associated with a previous call.

An Initial INVITE request MUST include an offer. In addition, if a 407 (Proxy Authentication Required) response is received after sending out an Initial INVITE request, authentication processing MUST be performed as described in Section 5.1.4.

The use of SIP extensions as specified by option-tag involves the setting of extension character strings (such as “100rel”) in the Supported header. Note that the use of the Require header or Proxy-Require header here MAY prevent establishment of a call from being made for some or all call-establishment patterns. In this specification, the “100rel” and “timer” extension functions are included. Details on using each of these extensions are described in Section 8 and Section 9, respectively.

5.1.2. Request-URI

The Request-URI of an Initial INVITE request is configured in SIP URI format. The configuration of a Request-URI in other than SIP URI format is beyond the scope of this standard.

The Request-URI is configured as follows.

Request-uri = "sip:" user"@" host [:port] [uri-parameters]

Table 5-1/JJ-90.24: Configuration of Request-URI when sending an Initial INVITE request

Element	Value	Type (caller)	Remarks
user	DIGIT	m	In general, the character string corresponding to the user's dial number
host	Ipv4address or hostname format	m	In general, the SIP proxy server address (may be different than the AoR host character string)
port		c1	
uri-parameters		o	
c1: "o" if using 5060; "m" otherwise ⁹			

5.1.3. AoR (From header)

An AoR or anonymous URI shall be set in addr-spec of the From header. Section 12.1 gives guidelines on which to select. The user part of AoR may be set with any character string within the conditions described in Section 13.2.1¹⁰.

5.1.4. Authentication

If a SIP terminal receives a 407 (Proxy Authentication Required) response from the SIP proxy server after sending it an Initial INVITE request, an authentication procedure shall be performed according to the authentication scheme specified in the Proxy-Authenticate header of that response. This scheme makes use of HTTP Digest.

5.1.4.1. Receiving a 407 (Proxy Authentication Required) Response

On receiving a 407 (Proxy Authentication Required) response from a SIP proxy server after sending it an Initial INVITE request, the SIP terminal shall perform an authentication procedure according to the authentication scheme specified in the Proxy-Authenticate header of that response. Table 5-2 summarizes the handling of a received Proxy-Authenticate header.

⁹ The port number used may be specified by the provider's SIP network.

¹⁰ Note that the AoR user section supposes a character string assigned by the provider's SIP network based on policy and is not limited to "telephone-subscriber."

Table 5-2/JJ-90.24: Handling of Proxy-Authenticate header received in a 407 (Proxy Authentication Required) response

Directive	Value	Type (callee)	Remarks
nonce	<nonce>	m	
realm	<realm>	m	realm character string
domain	<domain>	m	domain character string
opaque	<opaque>	m	Null character only in double quotes or a random string of 32 characters
stale	TRUE、もしくは FALSE	m	[TRUE] The value of the nonce parameter in the Proxy-Authorization header included in the sent Initial INVITE request has been found by the SIP registrar server to have expired or to be malformed. The authentication procedure is to be performed again using the nonce value received in the 407 (Proxy Authentication Required) response. [FALSE] If a Proxy-Authorization header was not included in the sent Initial INVITE request (which implies a new authentication), an authentication Initial INVITE request is to be sent according to the specified authentication procedure. If a Proxy-Authorization header was included, the SIP registrar server has judged the value of the username or response parameter in the Proxy-Authentication header to be invalid and will notify the user of such in some way.
algorithm	MD5	m	
qop-options	auth, auth-int	m	
other parameter	-	o	Ignored. This need not apply when used in applications.

* Items included in angled brackets (“<” and “>”) indicates a value set by the SIP proxy server depending on a certain condition or setting.

5.1.4.2. Sending an Authentication Initial INVITE Request

After receiving a 407 (Proxy Authentication Required) response, the SIP terminal sends the SIP proxy server an Initial INVITE request that includes a Proxy-Authorization header specifying user name and password according to the specified authentication procedure. Table 5-3 summarizes the handling of this Proxy-Authorization header

The value of Call-ID set in this authentication Initial INVITE request shall be the same as that received in the 407 (Proxy Authentication Required) response and Cseq shall be incremented by 1.

Table 5-3/JJ-90.24: Handling of Proxy-Authorization header to be sent with an Authentication Initial INVITE Request

Directive	Value	Type (caller)	Remarks
username	<username>	m	Username used in authentication
nonce	Copied from received Proxy-Authenticate header	m	Same value as received in 407 (Proxy Authentication Required) response
realm	Copied from received Proxy-Authenticate header	m	Same value as received in 407 (Proxy Authentication Required) response
uri	Same as Request-URI	m	
response	<response>	m	Value computed from password using algorithm set in Proxy-Authenticate header of 407 (Proxy Authentication Required) response
opaque	Copied from received Proxy-Authenticate header	m	Same value as received in 407 (Proxy Authentication Required) response
algorithm	MD5	m	Same value as received in 407 (Proxy Authentication Required) response
cnonce	Arbitrary value created by terminal	m	Set only when qop-options is set in the Proxy-Authenticate header received in the 407 (Proxy Authentication Required) response
nonce-count	Number of authentication-request submissions in hexadecimal	m	Set only when qop-options is set in the Proxy-Authenticate header received in the 407 (Proxy Authentication Required) response
message-qop		m	Set only when qop-options is set in the Proxy-Authenticate header received in the 407 (Proxy Authentication Required) response. (*1) If set, the same value as received in the 407 (Proxy Authentication Required) response.
other parameter	-	o	Not set. This need not apply when used in applications.

(*1) The message-qop directive MUST NOT be set if qop-options has not been set in the Proxy-Authenticate header received in the 407 (Proxy Authentication Required) response.

5.2. Receiving a 1xx Response

5.2.1. Basic Behaviour

Between the time of sending out an Initial INVITE request and receiving a final response with respect to that request, the SIP terminal may receive no 1xx response or one or several 1xx responses. The SIP terminal will, in any case, perform processing based on the SIP response received. If a 1xx response should be received, the SIP terminal MUST stop the resending of Initial INVITE requests and MUST wait for the final response.

The case of SDP included in a 1xx response and its relation with the receiving of a 180 (Ringing) response is described in Section 5.2.2.

On receiving a 1xx (excluding 100 (Trying)) response in which the tag parameter is set in the To header, an Early dialog shall be established. Processing MUST be performed according to the received SIP response signal even if the tag parameter is not included in the To header.

A Require header may be set in a 1xx response (excluding 100 (Trying)) in accordance with the content of the Supported header in the sent Initial INVITE request. Here, it must be kept in mind that an option-tag set in the Supported header MAY not necessarily be set in the Require header and that the Require header itself MAY not exist. In such a case, it MUST still be possible to establish the session without the use of the extensions in question.

In this specification, the “100rel” extension function is included. Details on using this extension are described in Section 8.

5.2.2. Ring-tone Generation and Early Media

5.2.2.1. Ring-tone Generation

If a 180 (Ringing) response without SDP is received before receiving any 1xx (excluding 100 (Trying)) response with SDP, the SIP terminal **MUST** generate a ring tone using its own sound source at that time. Then, within the same dialog, the ring tone **MUST** continue to be generated as long as any subsequently received 1xx (excluding 100 (Trying)) response does not include SDP (in other words, the ring tone must not be undone). However, if SDP is included in a received 1xx (excluding 100 (Trying)) response, a media path **MUST** be connected as described in Section 5.2.2.2 and a sound source played back from the callee side.

5.2.2.2. Early Media

5.2.2.2.1. Basic Behavior

On receiving a 1xx (excluding 100 (Trying)) response with SDP set, a SIP terminal **MUST** interpret that response as an answer regardless of whether the 100rel extension is applied and **MUST** be able to establish Early media by connecting a path. This action takes into account the fact that all 1xx (excluding 100 (Trying)) responses cannot be transferred reliably (the 100rel extension is not applied to all 1xx (excluding 100 (Trying)) responses transferred by the provider's SIP network). By therefore interpreting the first 1xx (excluding 100 (Trying)) response to be received with SDP (regardless of whether that response is reliable) as an answer, session-creation processing can be initiated and a voice path connected in both directions. Then, for the same dialog, the generation of receive media **MUST** be continued without relation to whether SDP exists in any subsequently received 1xx (excluding 100 (Trying)) response (that is to say, reprocessing of that media **MUST NOT** take place).

5.2.2.2.2. Session Modification by the UPDATE Method

Given that UPDATE was included in the Allow header of the Initial INVITE request and that an answer was received through a 1xx (excluding 100 (Trying)) response deemed reliable by application of the 100rel extension (following completion of the initial offer/answer), the possibility of receiving an UPDATE request including an offer from the callee before receiving the final response must be considered. Then, if a change to session contents as specified by an offer from the callee can be affected on one's own side through the UPDATE method, those changes **SHOULD** be made and a 200 (OK) response including an appropriate answer **SHOULD** be returned. On the other hand, if the specified session changes are not possible, a 488 (Not Acceptable Here) response **MUST** be returned. In this case, termination of the existing session would not be performed and the SIP UA that made the media-change request would be allowed to decide on how the session should then be handled.

5.2.3. Management of Multiple Dialogs and Media

In the event that an Initial INVITE request has been subjected to forking by the provider's SIP network, the caller SIP terminal may receive multiple 1xx (excluding 100 (Trying)) responses from callee SIP terminals. Thus, in addition to any existing dialog (or dialogs), a SIP terminal **MUST** be able to behave as a user agent client (UAC) and accommodate multiple dialogs that result from receiving a response with a To-tag different from that of other responses that have so far been received with respect to the sending of one Initial INVITE request.

A SIP terminal **MUST** also be able to accommodate multiple dialogs using different media.

Table 5-4 summarizes the minimum and desired range of implementation with respect to caller SIP terminal processing taking the above discussion into account.

Table 5-4/JJ-90.24: Management of Multiple Dialogs and Media (Caller SIP terminal)

	Existing Dialog	New Dialog	Processing
①	Early dialog	Early dialog	<ul style="list-style-type: none"> • Policies can be established that dictate which dialog is given priority in user-interface processing under certain conditions such as the presence of SDP, the content of SDP, etc. If using 100rel, however, a 2xx response may be received without an answer, in which case it is RECOMMENDED that all session information be saved or a BYE request be sent and the Early dialog explicitly terminated. In particular, if there are no conditions that can be used as criteria, it is RECOMMENDED that the new dialog be given priority (taking into account transferring at time of no-reply, etc.)

5.3. Receiving a 2xx Response

5.3.1. Basic Behaviour

The SIP terminal establishes a confirmed dialog according to the first 2xx response it receives. If tag is not included in the To header of that 2xx response, it MUST be able to establish a confirmed dialog treating tag of the To header as null (See Section 5.3.3 in the event that a 1xx (excluding 100 (Trying)) response including tag in the To header has been previously received).

For the case that an offer was included in the Initial INVITE request and an answer was received by a message belonging to the same dialog before the SIP terminal received the 2xx response, the content of the SDP contained in the 2xx response is expected to be the same as the previously received answer and is therefore ignored. If an answer was not received before receiving the 2xx response, the session shall be established according to the answer contained in the 2xx response.

5.3.2. Sending an ACK Request

The SIP terminal sends an ACK request in reply to a 2xx response that has been received. If the received 2xx response includes a Record-Route header, any request messages sent out by the SIP terminal from this ACK request on in the same dialog MUST involve Record-Routing processing included in the Route header described in Section 7.

If no Record-Route exists, the SIP terminal sends an ACK request addressed to the remote target host set in the Contact header of the received 2xx response.

5.3.3. Management of Multiple Dialogs and Media

A SIP terminal may receive 2xx responses from multiple callee SIP UAs if the Initial INVITE request has been forked by the provider's SIP network. Thus, in addition to any existing dialog (or dialogs), a SIP terminal MUST be able to behave as a UAC and accommodate multiple dialogs that result from receiving a response with a To-tag different from that of other responses that have so far been received with respect to the sending of one Initial INVITE request.

A SIP terminal MUST also be able to accommodate multiple dialogs using different media.

Table 5-5 summarizes the minimum and desired range of implementation with respect to caller SIP terminal processing taking the above discussion into account.

Table 5-5/JJ-90.24: Management of Multiple Dialogs and Media (Caller SIP terminal)

	Existing Dialog	New Dialog	Processing
①	Early dialog	Confirmed dialog	•Changes the session according to the content of the confirmed dialog. The Early dialog may be explicitly terminated by sending a BYE request or its content destroyed after $64 \times T1$.
②	Confirmed dialog	Confirmed dialog	•Policies MAY be established that dictate which dialog is to be given priority (or if both are to be maintained simultaneously) based on certain conditions such as SDP content. Regardless of which is given priority, the other SHOULD be explicitly released by sending a BYE request (the failure to return ACK requests will simply generate more 2xx responses).

5.4. Receiving a 3xx Response

5.4.1. Basic Behaviour

If, on receiving a 3xx response, the SIP terminal cannot confirm that the response comes from the same provider's SIP network and that there is no security risk, Initial INVITE request MUST NOT be transferred to the address specified in the Contact header included in the message. At this time, outgoing call processing shall be terminated while informing the user with a Busy Tone or the like that the call cannot be made.

5.4.2. Sending an ACK Request

The same processing shall be performed as when sending an ACK request in reply to an ordinary error response as specified in JF-IETF-RFC3261 [1].

5.5. Receiving a 4xx/5xx/6xx Response

5.5.1. Basic Behaviour

The SIP terminal processes the receiving of a response of this type according to the information given in Table 13-2. It passes the received response code, reason phrase, and header types to the upper-layer application, returns an ACK request in accordance with JF-IETF-RFC3261 [1], and completes the Initial INVITE request transaction.

- On receiving a 413 (Request Entity Too Large) response, 415 (Unsupported Media Type) response, 416 (Unsupported URI Scheme) response, or 420 (Bad Extension) response, a function that can correct the request and perform a retry according to Section 8.1.3.5 of JF-IETF-RFC3261 [1] MAY be advantageous.
- On receiving a 422 (Session Interval Too Small) response, the behaviors specified in Section 9 should be observed.

5.5.2. Sending an ACK Request

The same processing shall be performed as when sending an ACK request in reply to an ordinary error response as specified in JF-IETF-RFC3261 [1].

5.6. A Call Abortion

5.6.1. Sending a CANCEL Request

An SIP terminal may wish to abort a call after receiving a 1xx response and before receiving a final response due, for example, to a disconnect request from the user (at the time of an on-hook behavior, application termination, etc.) or a calling timeout. To abandon a call, the SIP terminal sends a CANCEL request with respect to an Initial INVITE request in accordance with JF-IETF-RFC3261 [1].

Note that a transaction will continue to be maintained and processed within a timer as long as a 2xx response to the

CANCEL request has not been received or a final response (ordinary 487 (Request Terminated) response) to the Initial INVITE request has not been received after receiving a 2xx response to the CANCEL request. During this time, it MUST be possible for the SIP terminal to send out a new Initial INVITE request to create the next INVITE transaction if a new call request has been issued by the user.

Furthermore, for the situation in which a 1xx response has not yet been received with respect to an Initial INVITE request but the user-resource side detects that the call in question is being aborted, no CANCEL request can be sent out according to JF-IETF-RFC3261 [1]. To therefore stop Initial INVITE resend processing, the Initial INVITE transaction can be saved until transaction timeout. Later, on receiving a 1xx response with respect to the Initial INVITE request, the CANCEL request can be submitted. In the event that an error response is received with respect to an INVITE request, processing shall conform to Section 5.3.

5.6.2. Cautionary Notes on Signal/Process Conflicts

If, after sending a CANCEL request, a 2xx response is received with respect to that Initial INVITE request, the SIP terminal MUST first send an ACK request to establish a dialog and then send a BYE request to terminate that dialog.

A variety of maintained states can be considered in SIP message sequences, such as the resending of CANCEL requests or the resending of BYE requests, the latter of which can result from the receiving of a 200 (OK) response while not receiving a final response to an Initial INVITE request. In any case, it MUST be possible for the SIP terminal to send or receive a new Initial INVITE request in conjunction with the outgoing or incoming of a new call in parallel with such maintained states.

5.7. Other Cautionary Notes

5.7.1. Considerations on Congestion Control

To prevent unnecessary congestion at a SIP proxy server due to the sending of INVITE requests from a SIP terminal, a function SHOULD be provided at the SIP terminal to limit the sending of consecutive requests within a short time interval. In addition, requests SHOULD NOT be resent using the same user name and password excluding the case when the stale parameter of the Proxy-Authenticate header of a 407 (Proxy Authentication Required) response received from the SIP proxy server is TRUE, or the case in which a WWW-Authenticate header or Proxy-Authenticate header exists set with a realm parameter not previously received.

5.7.2. Contact Header Values in a Request

Considering that the value of the local target host field in the Contact header of an INVITE request turns out to be the address of a request sent to oneself (the SIP terminal sending the INVITE request) within the same dialog, it is RECOMMENDED that a unique userinfo part not easily inferred by a third party be created. In addition, whenever the SIP terminal in question registers location information, the userinfo part of the Contact header in the REGISTER request SHOULD be different from one REGISTER request to another. However, its value should not change in subsequent transactions within the same dialog.

Within a dialog, if the Request-URI in a received request has a value different from that expected by the SIP terminal, that request SHOULD be destroyed save for any special circumstances.

5.7.3. INVITE Request Retry

A SIP proxy server may return an error response containing a Retry-After header in reply to a received INVITE request. In this case, the sending of an INVITE request to the same Request-URI SHOULD NOT be retried before the time specified in the Retry-After header to prevent the occurrence of congestion or other problems in the provider's SIP

network.

6. Basic Call Control Behaviors for Receiving a Call

6.1. Receiving an Initial INVITE Request

6.1.1. Basic Behaviour

A SIP terminal judges a received INVITE request with no tag parameter in the To header as an Initial INVITE request. Then, on determining that the Initial INVITE request is addressed to oneself by checking the Request-URI as described in Section 6.1.2, the SIP terminal returns a 100 (Trying) response and commences the receiving process.

If the SIP terminal is in a state in which a SIP transaction or dialog in relation to a previous call has not yet terminated normally, it SHOULD be possible to receive and process an Initial INVITE request independent of any existing processes.

6.1.2. Request-URI

Based on the conditions listed below, the SIP terminal compares the Request-URI in the received Initial INVITE request with the location information specified in the Contact header of the REGISTER request submitted when registering one's own location information. The SIP terminal judges the Initial INVITE request to be an incoming call that should be processed accordingly only if the Request-URI is the same as that location information ¹¹.

- The URI userinfo part and host part in the Contact header set by the SIP terminal at the time of location registration must have the same values as that in the received Initial INVITE request.
- With regard to URI parameters not recognized by the SIP registrar server, it must be kept in mind that they may not be included and maintained with location information and therefore SHOULD NOT be targets of comparison.
- Even if a port number has been specified for hostport, that port may not necessarily be specified in host port in the Request-URI of the Initial INVITE request. In short, the port specified in hostport of the Request-URI SHOULD NOT be targeted for comparison.

6.2. Sending 1xx Response

6.2.1. Basic Behaviour

In the event that a 1xx response (excluding 100 (Trying)) or a final response cannot be sent out immediately after receiving an Initial INVITE request, the SIP terminal SHOULD send out 100 (Trying) response to prevent unnecessary resends of the Initial INVITE request.

When sending a 1xx response other than a 100 (Trying) response, an Early dialog MUST be established by creating a tag parameter in accordance with Section 13.2.2 and entering that tag in the To header. Any subsequent 1xx responses MUST enter a tag parameter in the To header with the same value as that created here.

Also, when sending one or more 1xx responses other than 100 (Trying) responses, and when supporting the SIP extension set in option-tag included in the Supported header of the received Initial INVITE request, that option-tag shall be set in the Require header of all 1xx responses to be sent out. If the Supported header itself does not exist in the recognition Initial INVITE request or if a supported option-tag is not set in the Supported header, the above Require header MUST NOT be set and it MUST be possible to establish a session without using the SIP extension in question. This specification prescribes the use of "100rel," which is described in detail in Section 8.

¹¹ This limitation may not hold true depending on conditions in the provider's SIP network.

6.2.2. 180 (Ringing) Response

Considering that reception of a 180 (Ringing) response by the SIP terminal on the caller requires special processing as described in Section 5.2.2, 180 (Ringing) response MUST NOT be sent until the user calling state can be ascertained (e.g., up until an extension-designation receive-completion signal is received from the user (such as a PBX) assuming that the user resource is a 2W analog interface and that a dial-in sequence is used, or up until a receive-completion signal is received from an information-receiving terminal in the case of a number-display sequence). In other words, a 180 (Ringing) response MUST be sent as soon as the user calling state can be ascertained.

6.2.3. Early Media

If the caller SIP UA happens to be a PSTN gateway in the provider's SIP network, Early media does not connect with the call channel on the PSTN side at the PSTN gateway, which means that the callee SIP terminal SHOULD NOT, in general, include SDP in a 1xx response in this case.

6.3. Sending a 2xx Response

6.3.1. Basic Behaviour

The sending of a 2xx response on receiving a reply from the user signifies that a confirmed dialog is to be established. At this time, if tag was previously set in the To header of a 1xx (excluding 100 (Trying)) response and an Early dialog established, the same tag MUST be set in the To header of that 2xx response. Likewise, if Early media was previously established when sending a 1xx (excluding 100 (Trying)) response (not recommended for the reason given in Section 6.2.3) and SDP is included in the 2xx response, that SDP shall be the same as the SDP used when establishing that Early media. If Early media has not been established, the 2xx response shall contain an answer.

If the callee SIP terminal supports the SIP extension specified by option-tag included in the Supported header of the received Initial INVITE request, it shall set that option-tag corresponding to the SIP extension to be used after session establishment in the Require header of the 2xx response to be returned. If a Supported header does not exist in the recognition Initial INVITE request, or if a supported option-tag is not specified in the Supported header, it MUST be possible to establish a session without setting the above Require header and without using that SIP extension. This specification prescribes the use of "session timer," which is described in detail in Section 9.

6.4. Sending a 3xx Response

A 3xx response MUST NOT be transmitted for security reasons.

6.5. Sending a 4xx/5xx/6xx Response

On determining that session-establishment processing in response to a received Initial INVITE request cannot be continued for some reason such as user line busy or no available lines, the SIP terminal sends out an error response with an appropriate response code from among 4xx/5xx/6xx responses. At this time, if the user resource should be in a state in which it can process new incoming or outgoing calls, the SIP terminal MUST be able to process the sending or receiving of an Initial INVITE request even if it is still in the ACK-request-wait state.

6.6. Receiving an ACK Request in Reply to an Error Response

On receiving an ACK request in reply to the sending of an error response, one way for the SIP terminal to perform transaction matching is to compare the Request-URI in the ACK request with the Request-URI in the Initial INVITE request. Here, however, it is RECOMMENDED that URI parameters not be targeted for comparison considering signal

recognizability and connectivity.

6.7. A Call Abortion

With respect to a received Initial INVITE request, a SIP terminal MAY receive a CANCEL request against that Initial INVITE request or a BYE request corresponding to an established Early dialog after sending a 1xx response but before receiving an ACK request corresponding to the sending of a final response (the caller SIP UA is aborting the call).

6.7.1. Receiving a CANCEL/BYE Request

If a final response to an Initial INVITE request has not yet been received on receiving a CANCEL request or a BYE request, a 2xx response corresponding to a CANCEL or BYE request shall be returned together with a 487 (Request Terminated) response with respect to the Initial INVITE request and processing for stopping or releasing user-resource calling shall be performed. Furthermore, if an Early dialog had previously been established by sending a 1xx (excluding 100 (Trying)) response, the SIP response message sent here MUST be set with the same To-tag parameter as that set in that 1xx response. Also, if a 487 (Request Terminated) response is in the process of being resent due to the non-receipt of an ACK request, the SIP terminal MUST still be able to perform, in parallel, the sending or receiving of an Initial INVITE request due to a new outgoing or incoming call.

6.7.2. Cautionary Notes on Signal/Process Conflicts

Up to the time that an ACK request is received after the callee SIP terminal sends a 2xx response in reply to an Initial INVITE request, a CANCEL request may be received with respect to that INVITE transaction or dialog. If so, the SIP terminal will enter into a receive-wait state in SIP for the ACK request from the caller SIP UA and for the subsequent BYE request. At this time, the callee SIP terminal SHOULD use this receipt of a CANCEL request as an opportunity to send a Busy Tone (or to perform an equivalent behavior) with respect to the callee user resource so as to notify it that a disconnect has occurred on the caller.

On receiving a CANCEL request after sending a 200 (OK) response as described above, the callee SIP terminal may enter into a state in which 200 (OK) responses are being resent due to the non-receipt of an ACK request or in which a BYE request has not yet been received after receiving the ACK request. In this state, the SIP terminal MUST still be able to perform, in parallel, the sending or receiving of an Initial INVITE request due to a new outgoing or incoming call

6.8. Other Cautionary Notes

6.8.1. Contact Header Values in a Response

Considering that the value of the local target host field in the Contact header of a 1xx (excluding 100 (Trying)) response and 2xx response with respect to an INVITE request turns out to be the address of a request sent to oneself (the SIP terminal receiving the INVITE request) within the same dialog, it is RECOMMENDED that a unique userinfo part not easily inferred by a third party be created. In addition, whenever the SIP terminal in question registers location information, the userinfo part of the Contact header in the REGISTER request SHOULD be different from one REGISTER request to another. However, its value should not change in subsequent transactions within the same dialog.

Within a dialog, if the Request-URI in a received request has a value different from that expected by the SIP terminal, that request SHOULD be destroyed save for any special circumstances.

7. Route Set and Remote Target

This section describes items associated with route set and remote target that must be taken into account when configuring SIP request messages to be sent out during a dialog.

7.1. Basic Behavior

If the first uri of a route set includes the lr parameter, the SIP terminal configures the Request-URI and Route header of SIP request messages to be sent during a dialog as follows.

- Sets a remote target in Request-URI.
- Sets an ordered route set in the Route header.
- Sets multiple Route headers if setting multiple route sets.

SIP request messages configured in this way are sent out addressed to an adjacent route (the Route header set to the top level). Note here that SIP proxy occasionally sets its own address by itself in the maddr parameter in the Record-Route header. In this case, the route set and the destination of a SIP request message including the maddr parameter **MUST** be managed.

If the first uri of a route set does not include the lr parameter, the SIP terminal configures the Request-URI and Route header of SIP request messages to be sent during a dialog as follows.

- Sets the first route set in Request-URI.
- Sets a route set other than the one set in Request-URI in a Route header in the order to be followed.
- Sets a remote target at the end of a Route header.
- Sets multiple Route headers if setting multiple route sets.

Note here that SIP proxy occasionally sets its own address by itself in the maddr parameter in the Record-Route header. In this case, the route set and the destination of a SIP request message including the maddr parameter **MUST** be managed.

7.2. Caller Route Set

7.2.1. When Sending an Initial INVITE Request

No route set is considered to exist when sending out an Initial INVITE request; a pre-existing route **MUST NOT** be set.

7.2.2. When Establishing an Early Dialog

A route set is determined from the Record-Route header received in a 1xx response after sending out a SIP request (PRACK request) during an Early dialog. If multiple Early dialogs are to be established, a route set shall be managed and used for each dialog.

7.2.3. When Establishing a Confirmed Dialog

In the case of a Confirmed dialog, a route set is determined from the Record-Route header received in a 200 (OK) response; the route set **MUST NOT** be changed during that dialog. (If, within the same Confirmed dialog, a different Record-Route header happens to be set in a 200 (OK) response received after sending a SIP request message or in a received SIP request message, it shall be ignored.)

7.3. Callee Route Set

The recipient SIP terminal determines a route set from the Record-Route header on receiving an Initial INVITE request.

7.4. Caller Remote Target

The caller remote target is the same as the URI in the Contact header received from the other party's SIP UA in an Initial INVITE request. If a different URI is subsequently received after setting up a dialog, the remote target for that

dialog shall be rewritten with the new URI.

7.5. Callee Remote Target

The callee remote target is the same as the URI in the Contact header received from the other party's SIP UA in a 1xx response, UPDATE request, and 200 (OK) response. If a different URI is subsequently received after setting up a dialog, the remote target for that dialog shall be rewritten with the new URI.

8. 100rel Extension

The 100rel extension specified in JF-IETF-RFC3262 [2] MUST be implemented to guarantee the delivery of a 1xx response. It MUST be possible to stop the use of this function by making an appropriate setting ¹².

8.1. Exchanging Capabilities

When sending an Initial INVITE request, the caller SIP terminal sets 100rel in the Supported header to inform the callee SIP UA that it supports this function. Whether or not the 100rel extension is to be used depends on the callee SIP UA (the Require header MUST NOT be set in the Initial INVITE request).

The callee SIP terminal can perform 100rel extension behaviors only if 100rel is set in the Supported header of a received Initial INVITE request.

8.2. Sending a 1xx response

Provided that 100rel has been set in the Supported header of a received Initial INVITE request, the callee SIP terminal sets 100rel in the Require header of all 1xx responses (excluding 100 (Trying)) to be sent and also sets the RSeq header. The callee SIP terminal resends a 1xx response (excluding 100 (Trying)) according to JF-IETF-RFC3262 [2] until it receives a PRACK request. On receiving a PRACK request, the callee SIP terminal checks the RACK header method, CSeq-num, and response-num to verify whether the received PRACK request corresponds to the sent 1xx response (excluding 100 (Trying)). If the PRACK request checks out, the callee SIP terminal MUST send a 200 (OK) response in reply to that PRACK request. If an expected PRACK request is not received before the 1xx response (excluding 100 (Trying)) resend timer times out, the callee SIP terminal sends a 500 (Server Internal Error) response against the Initial INVITE request and terminates the Initial INVITE request transaction.

If the callee SIP terminal needs to send multiple 1xx responses (excluding 100 (Trying)), it SHOULD NOT send another 1xx response (excluding 100 (Trying)) until the PRACK request for the previously sent 1xx response (excluding 100 (Trying)) is received.

8.3. Sending a PRACK Request

On receiving a 1xx response (excluding 100 (Trying)) with 100rel set in the Require header, the caller SIP terminal returns a PRACK request with the RACK header set. The caller SIP terminal resends the PRACK request according to JF-IETF-RFC3262 [2] until it receives a final response. Also, if a 1xx response (excluding 100 (Trying)) with RSeq incremented is received before receiving a final response to that PRACK request, a 100rel extension behavior shall be performed as a new 1xx response.

9. Session Timer Extension

The session timer extension specified in JF-IETF-RFC4028 [16] MUST be implemented to detect a session release at

¹² The provider's SIP network may specify that the use of this function be stopped.

a exceptional procedure time. It MUST be possible to stop the use of this function by making an appropriate setting ¹³.

9.1. Exchanging Capabilities

When sending an Initial INVITE request, the caller SIP terminal sets timer in the Supported header to inform the callee SIP UA that it supports this function. Whether or not the timer extension is to be used depends on the callee SIP UA (the Require header MUST NOT be set in the Initial INVITE request). More specifically, whether or not the session timer has been started depends on whether “timer” has been set in the Require header of a 200 (OK) response. If this response is not received, the session timer has not been started. The callee SIP terminal starts the session timer if the Session-Expires header is set in the received Initial INVITE request. It sets timer in the Require header of a 200 (OK) response to be sent only if timer has been set in the Supported header. The above also holds in a session-refresh request transaction.

9.2. Session-refresh Request

A re-INVITE request or an UPDATE request can be used as a session-refresh request. Here, a function enabling the sending and receiving of UPDATE requests must be implemented in the SIP terminal. Specifically, a SIP terminal MUST have the capability of setting “UPDATE” in an Allow header that it sends, and if it itself is to be the refresher, refresh behaviors MUST be possible based on an exchange of capabilities with the other party’s SIP UA via an Allow header when establishing a session as described in Sections 9.2.1 and 9.2.2. It also MUST be possible by making a setting to exchange capabilities only through a re-INVITE request as a means of making a session-refresh request without the use of UPDATE requests ¹⁴.

9.2.1. re-INVITE Request

In the case that UPDATE requests cannot be used based on Allow-header exchange with the other party’s SIP UA when establishing a session (i.e., the case in which UPDATE is not included in the Allow header of an Initial INVITE request or 200 (OK) response), a re-INVITE request may be used. Note here that SDP MUST be set in the re-INVITE request and 200 (OK) response.

9.2.2. UPDATE Request

In the case that UPDATE requests can be used based on Allow-header exchange with the other party’s SIP UA when establishing a session (i.e., the case in which UPDATE is included in the Allow header of an Initial INVITE request or 200 (OK) response), an UPDATE request will be used. It is RECOMMENDED that an UPDATE request/response having the purpose of a session refresh not be set with SDP. Consideration MUST also be given to the receiving of a session-refresh request using a re-INVITE even if UPDATE happens to be set in an Allow header received from the other party’s SIP UA.

9.3. Determining the Refresher

9.3.1. At Session Establishment Time

9.3.1.1. Caller SIP Terminal

It is RECOMMENDED that the caller SIP terminal not set the refresher parameter in the Session-Expires header of an Initial INVITE request and that selection of a refresher be left to processing by the callee SIP terminal. In particular, the

¹³ The provider’s SIP network may specify that the use of this function be stopped. If so instructed, a SIP terminal does not perform the operations specified in this section.

¹⁴ The provider’s SIP network may specify that the use of UPDATE requests be stopped.

caller SIP takes the setting of `refresher=uac` in the `Session-Expires` header of a received 200 (OK) response to mean that it will send the next session-refresh request (refresher), and it takes the setting of `refresher=uas` to mean that it will wait for a session-refresh request up until the session timer expires (non-refresher). Also, if the caller SIP terminal should set the refresher parameter in the `Session-Expires` header of an Initial INVITE and the value of the refresher parameter set in the `Session-Expires` header of a received 200 (OK) response is different, behaviors MUST be performed with the received value.

9.3.1.2. Callee SIP Terminal

The callee SIP terminal takes the setting of `refresher=uas` in the `Session-Expires` header of a received Initial INVITE request to mean that it will send the next session-refresh request (refresher), and it takes the setting of `refresher=uac` to mean that it will wait for a session-refresh request up until the session timer expires (non-refresher). The value of the received refresher parameter is copied directly into the `Session-Expires` header of the 200 (OK) response to be returned.

If a `Supported` header exists in an Initial INVITE request received by the callee SIP terminal and “timer” is included while no refresher parameter exists in the `Session-Expires` header, the callee SIP terminal sets either `refresher=uas` or `refresher=uac` in the `Session-Expires` header of the 200 (OK) response to be returned.

If no `Supported` header exists in an Initial INVITE request received by the callee SIP terminal or “timer” is not included while no refresher parameter exists in the `Session-Expires` header, the callee SIP terminal sets `refresher=uas` in the `Session-Expires` header of the 200 (OK) response to be returned meaning that it will act as the refresher.

9.3.2. At Session Update Time

9.3.2.1. Refresher Side

If a SIP terminal establishes a session with itself as refresher, it sends a session-refresh request with the `Session-Expires` header set to `refresher=uac`.

9.3.2.2. Non-refresher Side

The non-refresher SIP terminal takes the setting of `refresher=uas` in the `Session-Expires` header of a received session-refresh request to mean that it will send the next session-refresh request (refresher), and it takes the setting of `refresher=uac` to mean that it will wait for a session-refresh request up until the session timer expires (non-refresher). The value of the received refresher parameter is copied directly into the `Session-Expires` header of the 200 (OK) response to be returned.

If a `Supported` header exists in the received session-refresh request and “timer” is included while no refresher parameter exists in the `Session-Expires` header, the `Session-Expires` header of the 200 (OK) response to be returned is set with either `refresher=uas` or `refresher=uac`.

9.4. Session Timer Value and Min-SE Value

9.4.1. Caller/Refresher-side

An implementation that allows the value set in “delta-seconds” of the `Session-Expires` header of an Initial INVITE request or session-refresh request to be changed is RECOMMENDED¹⁵. Actually, the value set here is sometimes changed at a SIP proxy server as long as that change does not contradict the setting of `Min-SE` header. While the `Min-SE` header SHOULD NOT be set, an implementation that would allow that value to be changed even if `Min-SE` header is set would be RECOMMENDED¹⁶.

¹⁵ The prescribed value may depend on the provider’s SIP network.

¹⁶ The set value must be less than the value prescribed by the provider’s SIP network.

If a 422 (Session Interval Too Small) response with Min-SE header set is received in reply to an Initial INVITE request or session-refresh request, session-establishment behaviors shall continue without affecting the user resource using the received Min-SE header and the Initial INVITE request with the value of the received Min-SE header newly set in the Session-Expires header.

Then, if a 422 (Session Interval Too Small) is again received in reply to an Initial-INVITE-request or session-refresh-request retry, a retry will again be performed in the same way. At this time, the largest Min-SE header value that has so far been sent/received shall be recorded for use as a Min-SE value within a dialog when establishing a dialog.

Finally, using the “delta-seconds” value in the Session-Expires header received in a 200 (OK) response, the caller sets and starts the session timer.

9.4.2. Callee/Non-refresher-side

The value of the Session-Expires header set in the received Initial INVITE request or session-refresh request is set in the Session-Expires header of the 200 (OK) response. However, for the case that timer is set in the Supported header of the received Initial INVITE request and the value of the Session-Expires header happens to be too small as a session-timer value that can be set by a SIP terminal on it's own, a 422 (Session Interval Too Small) response is returned with the smallest value that can be set as a session-timer value on one's own set in the Min-SE header. Note, however, that a 422 (Session Interval Too Small) response MUST NOT be returned with a session-timer value greater than that prescribed by the provider's SIP network.

The callee finally sends a 200 (OK) response. At this time, it also sets and starts the session timer using the value set in the Session-Expires header, and if a Min-SE header was included in the received Initial INVITE request, it records the largest Min-SE header value that has so far been sent/received for use as a Min-SE value within a dialog.

9.5. Session Refresh

9.5.1. Refresher Side

The refresher side sends a session-refresh request when the session timer for the dialog in question reaches half of the session-timer value determined at the immediately previous session-refresh request (if no session-refresh request has yet been sent/received in that dialog, the session-timer value determined at session establishment is used). Then, after receiving a 200 (OK) response, the refresher side again sets and starts the session timer using the value received in the Session-Expires header.

Also, if the refresher side should receive a re-INVITE request or UPDATE request containing a Session-Expires header, it MUST recognize such a request as a session-refresh request and process it in the same way as a non-refresher case.

9.5.2. Non-refresher Side

On the non-refresher side, if an UPDATE request or re-INVITE request containing a Session-Expires header for a certain dialog is received before the session timer for that dialog expires, that request shall be judged to be a session-refresh request. The non-refresher side then sends a 200 (OK) response and again sets and starts the session timer using the value set in the Session-Expires header.

9.5.3. Sending a Request in Conjunction with a Media Change

If a session timer is being used, the sending of a re-INVITE request or UPDATE request in conjunction with a media change after the establishment of a Confirmed dialog MUST also be treated as the sending of a session-refresh request.

9.6. Session Refresh Failure

If a session-refresh-request resend timeout occurs or if a 408 (Request Timeout) response or 481 (Call/Transaction Does Not Exist) response is received, the refresher-side SIP terminal sends a BYE request and releases the call as prescribed in Section 12.2.1.2 of JF-IETF-RFC3261 [1]. If an error response other than a 408 (Request Timeout) response or 481 (Call/Transaction Does Not Exist) response is received, the refresher-side SIP terminal behaves as prescribed in JF-IETF-RFC3261 [1] for various response codes. However, when receiving a 422 (Session Interval Too Small) response, the refresher-side SIP terminal behaves in the same way as when receiving the same response in reply to an Initial INVITE request as described in Section 9.4.1.

Alternatively, if a session-refresh-request resend timeout occurs or if a response other than a 422 (Session Interval Too Small) response is received, the refresher-side SIP terminal MAY continue the session up to 10 seconds before the session timer expires as long as no disconnect instruction is received from the user resource, and then forcibly send a BYE request with respect to that dialog and send a Busy Tone to the user resource (or perform equivalent processing).

On the non-refresher side, if a session-refresh request has not been received by 32 seconds before the session timer expires (or if one third of the session-time value is less than 32 seconds, by that value before the session timer expires), the non-refresher SIP terminal forcibly sends a BYE request with respect to that dialog at that time and sends a Busy Tone to the user resource (or performs equivalent processing).

This behavior may also be done 10 seconds before the session timer expires (or if one third of the session-timer value is less than 10 seconds, by that value before the session timer expires).

10. Media Establishment

10.1. Codec

ITU-T Recommendation G.711 μ -Law (64 kbit/s) MUST be supported. It is also RECOMMENDED that the Packet Loss Concealment (PLC) function corresponding to G.711 Appendix I be provided. If other codecs should be supported, it SHOULD be possible to specify which has priority, to select a particular codec, etc., by means of settings¹⁷.

10.2. SDP

10.2.1. Media Establishment by Offer/Answer

Media is established by an offer/answer exchange based on JF-IETF-RFC3264 [3]. With regard to SDP line-feed code, it MUST be possible to perform processing by receiving only LF at receiving time without having to set CRLF at sending time.

An offer made at the time of session establishment is sent together with an Initial INVITE request, and an answer within the same dialog is made through SDP received by a 1xx (excluding 100 (Trying)) response or through SDP received by a 200 (OK) response, whichever is received first. Also, it MUST be possible to process SDP received in a 1xx (excluding 100 (Trying)) response as an answer even if the 100rel extension is not operating. See Section 5.3.3 for processing performed when receiving SDP any number of times in 1xx/2xx responses in multiple dialogs.

Because of losses along SIP-response-message paths or differences between the routes used by SIP messages and RTP packets, a caller SIP terminal may receive RTP from the other party before receiving an answer. In this case, the jitter buffer MAY be adjusted and a path connection made to the user resource, for example, if possible and as needed. For the same reasons, a SIP terminal that has sent an answer may receive ICMP Unreachable with respect to the sent RTP packet, but it should be ignored and dialog/media continued.

¹⁷ Which codec to treat as the one with highest priority is defined by the provider's SIP network.

Media is sent and received by RTP/RTCP based on “c=”, “m=”, and “a=” lines in SDP exchange. However, disconnect processing MUST NOT be performed simply because RTCP cannot be received (the existence of other-party SIP UAs that do not send RTCP must be considered).

If no acceptable codecs are set in the received SDP, a SIP terminal returns 488 (Not Acceptable Here) response in accordance with JF-IETF-RFC3261[1]. A SIP terminal can set 304 to the Warning header in 488 response and also can set its acceptable codecs to the message body.

An offer included in an Initial INVITE request is configured as follows.

- ◆ The “m=” line sets the defined priority of supported codecs¹⁸ or a particular codec in fmt list. The fmt list value corresponding to G.711 is 0. The number of the codecs which can be set in the “m=” line may be specified by the provider’s SIP network.
- ◆ The attributes a=inactive, a=sendonly, and a=recvonly are not set. If setting a=ptime in G.711, a value of 20 ms is RECOMMENDED¹⁹.
- ◆ Media Description part is not repeated.
- ◆ Lines that may be omitted are omitted.

An answer is configured as follows.

- ◆ On the “m=” line, for the case that multiple codecs are given on the “m=” line of the received offer, those codecs and those that this SIP terminal supports are compared, and the codecs to be set in fmt list are determined by a defined selection scheme. Various selection schemes can be considered such as one based on the priority order received in the offer or one based on codec priority defined by this SIP terminal²⁰. One codec is selected according to this priority and is set as an answer. If telephone-event which is defined in JF-IETF-RFC2833[17] is used for sending DTMF, telephone-event is also set other than the selected codec.
- ◆ If receiving an offer with multiple “m=” lines, received “m=” lines MUST NOT be deleted but for codecs that cannot be used or will not be used, “0” MUST be set in port of “m=” line shown in the fmt list. Also, in the event that a re-INVITE request must be sent as a session-refresh request, an answer configured in the same way as above MUST be sent.
- ◆ If the attributes a=inactive, a=sendonly, and a=recvonly happen to be received in the offer, “a=” lines shall be set as specified. If these attributes are not supported, a 488 (Not Acceptable Here) response shall be returned. At this time, however, termination processing for the existing session shall not be performed; rather, the handling of that response shall be entrusted to the SIP UA that issued the media-change request. Also, if a=ptime is set, the value in the offer, if given, shall be used, and if not, a value of 20 ms shall be set.
- ◆ If a codec is specified using dynamic payload type in the “m=” line of the offer and the relevant codec is selected as an answer, the provider’s SIP network may specify the setting of the same dynamic payload type as the offer to the “m=” line of the answer. Also for the setting to the “a=” line, the provider’s SIP network may specify the setting of the same value as the offer.
- ◆ Lines that may be omitted are omitted.

¹⁸ Priority in the case of multiple codec functions is specified by the provider’s SIP network.

¹⁹ The provider’s SIP network may allow other values.

²⁰ Priority in the case of multiple codec functions is specified by the provider’s SIP network.

10.2.2. Receiving a Media Change Request

If receiving a re-INVITE request or UPDATE request containing SDP, the value of version on the “o=” line of the offer shall be compared with the value of version on the “o=” line of the offer or answer made in relation to the previously completed media-establishment or media-change transaction. If the values are different, the received request shall be judged to be a new media-change request. In this case, an answer shall be configured the same as for media establishment and media shall be reestablished in accordance with the changed specified.

If no changes have been made with respect to an offer or answer that a SIP terminal has previously sent, that SIP terminal MUST NOT change version on the “o=” line.

If the changes specified in a media-change request cannot be carried out, a 488 (Not Acceptable Here) response shall be returned. At this time, termination processing for the existing session shall not be performed and the handling of that response shall be entrusted to the SIP UA that issued the media-change request. Also, if receiving a media-change request involving the changing of IP addresses or port numbers (or both), the SIP terminal in question MUST be equipped with the capability of making those changes.

10.2.3. Sending a Media-change Request

When making a media-change request ²¹, the SIP UA in question MUST increment by 1 the value of version on the “o=” line in the previously sent SDP and MUST set the incremented value in a re-INVITE request or UPDATE request used to make an offer. If a dialog has already been established and the session timer extension is being used, behaviors MUST follow those described in Section 9.

If, in a state in which a dialog has been established, the UAS determines that the changes specified by a media-change request cannot be executed, a 488 (Not Acceptable Here) response may be received. If the session would like to be continued and the session timer extension is operating in that dialog, behaviors MUST follow those described in Section 9.

10.2.4. Cautionary Notes on re-INVITE Transactions

The following provides cautionary notes on the sending and receiving of offers/answers in conjunction with media changes using re-INVITE transactions as part of session refresh behaviors using the session timer extension (see Section 9).

When an offer supporting multiple codecs has been included in an Initial INVITE request, fmt list on the “m=” line in a subsequent offer made in a re-INVITE transaction sent by that SIP UA or on the “m=” line in the associated answer shall include only the codecs selected in the answer received at the time of media establishment. Here, though SDP content has changed, it does not really involve a media-change request as far as the other party’s SIP UA is concerned. For this reason, the value of version on the “o=” line MUST NOT be changed.

Again, for the case that an offer supporting multiple codecs has been received by an Initial INVITE request, an offer made in a re-INVITE transaction received from that SIP UA and the associated answer shall be judged to be a successfully completed offer/answer even under any of the following conditions and the session refresh shall be considered to have completed normally.

- ◆ The value of version in the “o=” line has been incremented and fmt list on the “m=” line has been changed only to the codecs selected in the answer received at the time of media establishment.
- ◆ The offer is exactly the same as that received in the Initial INVITE request (multiple codecs are included in fmt list on the “m=” line).

²¹ The acceptability of making media-change requests is defined by the provider’s SIP network.

- ◆ The value of version in the “o=” line has not been incremented and firt list on the “m=” line has been changed only to the codecs selected in the answer received at the time of media establishment.

11. Call Release

11.1. Sending a BYE Request

On detecting the end of a call on the user resource, the SIP terminal sends a BYE request for the dialog in question if a BYE request has not yet been received and releases the dialog, media, and user resource. Regardless of the BYE transaction state (such as a BYE-request-resend state or error-response-receive state), it shall be possible to send or receive an Initial INVITE request for a new outgoing or incoming call.

11.2. Receiving a BYE Request

On receiving a BYE request while a call is in progress for the dialog in question, the SIP terminal sends a 200 (OK) response and MUST release the dialog and media. The receiving of this BYE request is used as an opportunity to send the user resource a Busy Tone (or to perform an equivalent behavior).

12. Additional Functions

12.1. Number-Notification/Number-Notification Restriction

12.1.1. Setting Rules for Initial INVITE Request

To control number notification, a SIP terminal must make appropriate settings when sending an Initial INVITE request. Various schemes can be used: the provider’s SIP network can be notified using a scheme prescribed by JF-IETF-RFC3323 [5] or JF-IETF-RFC3325 [7] (scheme 1 or scheme 2) or the user’s dialed number can be directly set in the Initial INVITE request (scheme 3 or scheme 4). Configurations conditions for Initial INVITE request using schemes 1, 2, 3, and 4 are listed in Tables 12-1, 12-2, 12-3 and 12-4, respectively. It SHOULD be possible to select one of these four schemes by making an appropriate configuration.

Table 12-1/JJ-90.24: Header configuration conditions for number notification by scheme 1

Field	Notification	Notification Restriction	Remarks
Privacy header	none	id	
P-Preferred-Identity header	<AoR>	<AoR>	MAY not be set for notification
From header name-addr	<AoR>	<sip:anonymous@anonymous.invalid>	For notification restriction setting, a function shall be provided that allows for free entering of character strings, and character strings other than the one shown to the left MUST be possible.
Request-URI userinfo part	telephone number	telephone number	184/186 not set
To header userinfo part	Same as Request-URI	Same as Request-URI	184/186 not set

Table 12-2/JJ-90.24: Header configuration conditions for number notification by scheme 2

Field	Notification	Notification Restriction	Remarks
Privacy header	none	id	
P-Preferred-Identity header	<AoR>	<AoR>	MAY not be set for notification
From header name-addr	<AoR>	<sip:anonymous@anonymous.invalid>	For notification restriction setting, a function shall be provided that allows for free entering of character strings, and character strings other than the one shown to the left shall be possible.
Request-URI userinfo part	(186) telephone number	(184) telephone number	184/186 shall be set before the telephone number when dialed by the user.
To header userinfo part	Same as Request-URI	Same as Request-URI	184/186 shall be set before the telephone number when dialed by the user.

Table 12-3/JJ-90.24: Header configuration conditions for number notification by scheme 3

Field	Notification	Notification Restriction	Remarks
Privacy header	-	-	MUST NOT be set
P-Preferred-Identity header	-	-	MUST NOT be set
From header name-addr	<AoR>	<sip:anonymous@anonymous.invalid>	For notification restriction setting, a function shall be provided that allows for free entering of character strings, and character strings other than the one shown to the left shall be possible.
Request-URI userinfo part	(186) telephone number	(184) telephone number	184/186 shall be set before the telephone number when dialed by the user.
To header userinfo part	Same as Request-URI	Same as Request-URI	184/186 shall be set before the telephone number when dialed by the user.

Table 12-4/JJ-90.24: Header configuration conditions for number notification by scheme 4

Field	Notification	Notification Restriction	Remarks
Privacy header	-	-	MUST NOT be set
P-Preferred-Identity header	-	-	MUST NOT be set
From header name-addr	<AoR>	<AoR >	
Request-URI userinfo part	(186) telephone number	(184) telephone number	184/186 shall be set before the telephone number when dialed by the user.
To header userinfo part	Same as Request-URI	Same as Request-URI	184/186 shall be set before the telephone number when dialed by the user.

12.1.2. Cautionary Notes on Header Transparent Transfer

Because of the possibility of transferring unclear header information to a callee SIP UA by a SIP proxy server, character strings corresponding to user information MUST NOT be set in SIP header fields other than those indicated in Section 12.1.1 (excluding username in the Proxy-Authorization header) and in SDP component elements ²².

12.2. Caller Number Display

In accordance with the settings of the P-Asserted-Identity, Privacy, and From headers in the received Initial INVITE request, the following logic is used to decide what type of caller information to display to the callee user ²³.

(1) DISPLAYNAME of sip URI in P-Asserted-Identity header

-- If any one of the conditions in Table 12-5 is satisfied, display the corresponding notification restriction reason in the “Display Content (Reason)” column of Table 12-5.

--If none of the conditions in Table 12-5 are satisfied, go to (2).

(2) DISPLAYNAME of tel_URI in P-Asserted-Identity header

-- If any one of the conditions in Table 12-5 is satisfied, display the corresponding notification restriction reason in the “Display Content (Reason)” column of Table 12-5.

--If none of the conditions in Table 12-5 are satisfied, go to (3).

(3) Privacy header

-- If id is set, go to (6)

-- If id is not set or if header is not set, go to (4).

(4) userinfo part in From header

-- If any one of the conditions in Table 12-5 is satisfied, display the corresponding notification restriction reason in the “Display Content (Reason)” column of Table 12-5.

--If none of the conditions in Table 12-5 are satisfied, go to (5).

²² Information indicating the caller may be set in header fields shown in Section 12.1.1.

²³ A different setting method may be specified by the provider’s SIP network.

(5) From DISPLAYNAME

-- If any one of the conditions in Table 12-5 is satisfied, display the corresponding notification restriction reason in the "Display Content (Meaning)" column of Table 12-5.

-- If any one of the conditions in Table 12-6 is satisfied, display the caller number using the corresponding format in the "Display Content (Meaning)" column of Table 12-6.

-- If none of the conditions in Table 12-5 and Table 12-6 are satisfied, display the notification restriction reason corresponding to "Unavailable."

(6) From DISPLAYNAME

-- If any one of the conditions in Table 12-6 is satisfied, display the caller number using the corresponding format in the "Display Content (Meaning)" column of Table 12-6.

-- If none of the conditions in Table 12-6 are satisfied, display the notification restriction reason corresponding to "Unavailable."

Table 12-5/JJ-90.24: Notification restriction reason display

Receive Content (*1)(*2)	Display Content (Meaning)
Anonymous	No caller ID: rejected by user
Coin line/payphone	No caller ID: call from public telephone
Interaction with other service	No caller ID: service conflict
Unavailable	No caller ID: service unavailable

(*1) MAY be enclosed in double quotes.

(*2) The character strings shown in this table MAY be followed by arbitrary character strings.

Table 12-6/JJ-90.24: Caller telephone number display

Receive Content (*1)	Display Content (Meaning)
Numbers only	Received numeric string
Begins with "+81"; only numbers after "+"	Numeric string with "+81" deleted but prefixed with "0"
Begins with "+"; only numbers after "+" (no "81" after "+")	Numeric string with "+" deleted but prefixed with "010"

(*1) MAY be enclosed in double quotes when used as DISPLAYNAME.

13. SIP-related Signal Formats

This section lists SIP headers and other component elements related to SIP request/response messages in table format.

13.1. Basic Signal Formats

13.1.1. SIP Request Signals

Table 13-1 lists the SIP request signals covered in this standard.

Table 13-1/JJ-90.24: List of SIP request signals

SIP Request Signal	Ref.	Type (caller)	Type (callee)	Remarks
ACK	[1]	m	m	
BYE	[1]	m	m	
CANCEL	[1]	m	m	
Initial INVITE	[1]	m	m	
PRACK	[2]	c1	m	
re-INVITE	[1]	m	m	
REGISTER	[1]	c1	i	
UPDATE	[4]	c2	m	
Other requests		i	i	
c1: "m" if 100rel is set in Supported header of Initial INVITE; "x" otherwise ²⁴ c2: "m" if location registration behaviors are necessary; "x" otherwise ²⁵ c3: "m" if UPDATE is used for session update; "x" otherwise ²⁶				

13.1.2. SIP Response Signals

The following tables list SIP response signals covered in this standard with respect to each of the SIP request signals listed in Table 13-1. The minimum requirements prescribed in Section 8.1.3.2 of JF-IETF-RFC3261 [1] MUST be supported when receiving a SIP response signal.

²⁴ Although this function must be implemented, its use may be stopped by the provider's SIP network.

²⁵ Although this function must be implemented, its use may be stopped by the provider's SIP network.

²⁶ Although this function must be implemented, its use may be stopped by the provider's SIP network.

Table 13-2/JJ-90.24: List of SIP response signals for INVITE request

SIP Response Signal			Ref. Ref.	Type (caller)	Type (callee)	Remarks	
Type	Code	Phrase					
1xx	100	Trying	[1][4]	x	o		
1xx	100	Trying	[1][4]	mx	mo		
	180	Ringings Being Forwarded	[1][4]	mx	mo		
	182	Call Being Forwarded	[1][4]	ox	oo		
	183	Session Progress	[1][4]	ox	oo		
	Other	Session Progress	[1][4]	ox	mo		
2xx	200	OK	[1][4]	im	om		
2xx	200	OK	[1][4]	mi	mo		
3xx	300	Multiple Choices	[1][4]	ix	oi		
3xx	300	Multiple Choices	[1][4]	xx	c1i		
	302	Moved Permanently	[1][4]	xx	c1i		
	305	Moved Temporarily	[1][4]	xx	c1i		
	308	Use Alternative Service	[1][4]	xx	c1i		
	Other	Alternative Service	[1][4]	xx	c1i		
4xx	400	Bad Request	[1][4]	io	io		
4xx	400	Bad Request	[1][4]	oo	oo		
	401	Unauthorized	[1][4]	xo	oo		
	402	Forbidden	[1][4]	oo	oo		
	403	Forbidden	[1][4]	oo	oo		
	404	Not Found	[1][4]	oo	oo		
	405	Method Not Allowed	[1][4]	oo	oo		
	406	Not Acceptable	[1][4]	xo	oo		
	407	Proxy Authentication Required	[1]	x	m		
	408	Request Timeout	[1][4]	o	o		
	408	Request Timeout	[1][4]	oo	oo		
	413	Request Entity Too Large	[1][4]	oo	oo		
	413	Request Entity Too Large	[1][4]	oo	oo		
	415	Unsupported Media Type	[1][4]	oo	oo		
	416	Unsupported Media Type	[1][4]	oo	oo		
	420	Unsupported URI Scheme	[1][4]	om	oo		
	421	Extension Required	[1][4]	mx	oo		
	422	Session Interval Too Small	[4][16]	xm	om		
	422	Session Interval Too Small	[16]	m	m		
	423	Interval Too Brief	[1][4]	n/a	o		
	423	Interval Too Brief	[1][4]	n/a	oo		
	480	Temporarily Unavailable	[1][4]	oo	oo		
	480	Temporarily Unavailable	[1][4]	oo	oo		
	481	Call/Transaction Does Not Exist	[1]	o	o		
	482	Not Detected	[1][4]	o	o		
	483	Too Many Requests	[1][4]	oo	oo		
	484	Address Incomplete	[1][4]	oo	oo		
	485	Address Incomplete	[1][4]	oo	oo		
	486	Busy	[1][4]	oo	oo		
	487	Request Terminated	[1][4]	mo	mo		
	488	Request Not Allowed	[1][4]	mm	mo		
	489	Request Pending	[1][4]	mo	mo		
	490	Request Pending	[1][4]	oo	oo		
	Other	Undecipherable	[1][4]	oi	oo		
	5xx	500	Server Internal Error	[1][4]	io	oo	
	5xx	500	Server Internal Error	[1][4]	oo	oo	
		501	Not Implemented	[1][4]	oo	oo	
		502	Bad Gateway	[1][4]	oo	oo	
		503	Service Unavailable	[1][4]	oo	oo	
		504	Service Unavailable	[1][4]	oo	oo	
		505	Version Not Supported	[1][4]	oo	oo	
Other		Message Too Large	[1][4]	oi	oo		
6xx	600	Busy Everywhere	[1][4]	io	oo		
6xx	600	Busy Everywhere	[1][4]	oo	oo		
	604	Does Not Exist Anywhere	[1][4]	oo	oo		
	606	Does Not Exist Anywhere	[1][4]	oo	oo		
	Other	Not Acceptable	[1][4]	oi	oo		
Other		[1]	i	o			

c1: "m" if that response can be confirmed as coming from the SIP proxy server that one belongs to and that it presents no security risk; "x" otherwise

Table 13-3/JJ-90.24: List of SIP response signals for UPDATE request

Type	SIP Response Signal		Ref.	Type (caller)	Type (callee)	Remarks
	Code	Phrase				
1xx	100	Trying	[1]	n/a	o	
	180	Ringling	[1]	n/a	o	
	181	Call Is Being Forwarded	[1]	n/a	o	
	182	Queued	[1]	n/a	o	
	183	Session Progress	[1]	n/a	o	
	Other		[1]	n/a	o	
2xx	200	OK	[1]	n/a	m	
	Other		[1]	n/a	o	
3xx	300	Multiple Choices	[1]	n/a	i	
	301	Moved Permanently	[1]	n/a	i	
	302	Moved Temporarily	[1]	n/a	i	
	305	Use Proxy	[1]	n/a	i	
	380	Alternative Service	[1]	n/a	i	
	Other		[1]	n/a	i	
4xx	400	Bad Request	[1]	n/a	o	
	401	Unauthorized	[1]	n/a	m	
	402	Payment Required	[1]	n/a	o	
	403	Forbidden	[1]	n/a	o	
	404	Not Found	[1]	n/a	o	
	405	Method Not Allowed	[1]	n/a	o	
	406	Not Acceptable	[1]	n/a	o	
	407	Proxy Authentication Required	[1]	n/a	o	
	408	Request Timeout	[1]	n/a	o	
	410	Gone	[1]	n/a	o	
	413	Request Entity Too Large	[1]	n/a	o	
	414	Request-URI Too Long	[1]	n/a	o	
	415	Unsupported Media Type	[1]	n/a	o	
	416	Unsupported URI Scheme	[1]	n/a	o	
	420	Bad Extension	[1]	n/a	o	
	421	Extension Required	[1]	n/a	o	
	422	Session Interval Too Small	[16]	n/a	n/a	
	423	Interval Too Brief	[1]	n/a	m	
	480	Temporarily Unavailable	[1]	n/a	o	
	481	Call/Transaction Does Not Exist	[1]	n/a	o	
	482	Loop Detected	[1]	n/a	o	
	483	Too Many Hops	[1]	n/a	o	
	484	Address Incomplete	[1]	n/a	o	
	485	Ambiguous	[1]	n/a	o	
	486	Busy Here	[1]	n/a	o	
	487	Request Terminated	[1]	n/a	o	
	488	Not Acceptable Here	[1]	n/a	o	
	491	Request Pending	[1]	n/a	o	
	493	Undecipherable	[1]	n/a	o	
	Other		[1]	n/a	o	
5xx	500	Server Internal Error	[1]	n/a	o	
	501	Not Implemented	[1]	n/a	o	
	502	Bad Gateway	[1]	n/a	o	
	503	Service Unavailable	[1]	n/a	o	
	504	Server Time-out	[1]	n/a	o	
	505	Version Not Supported	[1]	n/a	o	
	513	Message Too Large	[1]	n/a	o	
	Other		[1]	n/a	o	
6xx	600	Busy Everywhere	[1]	n/a	n/a	
	603	Decline	[1]	n/a	n/a	
	604	Does Not Exist Anywhere	[1]	n/a	n/a	
	606	Not Acceptable	[1]	n/a	n/a	
	Other		[1]	n/a	n/a	

Table 13-4/JJ-90.24: List of SIP response signals for REGISTER request

Table 13-5/JJ-90.24: List of SIP response signals for CANCEL, PRACK, and BYE requests

13.1.3. SDP

SIP Response Signal			Ref.	Type (caller)	Type (callee)	Remarks
Type	Code	Phrase				
1xx	100	Trying	[1] [2]	x	o	
	180	Ringling	[1] [2]	x	o	
	181	Call Is Being Forwarded	[1] [2]	x	o	
	182	Queued	[1] [2]	x	o	
	183	Session Progress	[1] [2]	x	o	
	Other		[1] [2]	i	o	
2xx	200	OK	[1] [2]	m	m	
	Other		[1] [2]	i	o	
3xx	300	Multiple Choices	[1] [2]	x	i	
	301	Moved Permanently	[1] [2]	x	i	
	302	Moved Temporarily	[1] [2]	x	i	
	305	Use Proxy	[1] [2]	x	i	
	380	Alternative Service	[1] [2]	x	i	
	Other		[1] [2]	i	i	
4xx	400	Bad Request	[1] [2]	o	o	
	401	Unauthorized	[1] [2]	o	o	
	402	Payment Required	[1] [2]	o	o	
	403	Forbidden	[1] [2]	o	o	
	404	Not Found	[1] [2]	o	o	
	405	Method Not Allowed	[1] [2]	o	o	
	406	Not Acceptable	[1] [2]	o	o	
	407	Proxy Authentication Required	[1] [2]	x	o	
	408	Request Timeout	[1] [2]	o	o	
	410	Gone	[1] [2]	o	o	
	413	Request Entity Too Large	[1] [2]	o	o	
	414	Request-URI Too Long	[1] [2]	o	o	
	415	Unsupported Media Type	[1] [2]	o	o	
	416	Unsupported URI Scheme	[1] [2]	o	o	
	420	Bad Extension	[1] [2]	m	o	
	421	Extension Required	[1] [2]	o	o	
	422	Session Interval Too Small	[16] [2]	n/a	n/a	
	423	Interval Too Brief	[1] [2]	n/a	o	
	480	Temporarily Unavailable	[1] [2]	o	o	
	481	Call/Transaction Does Not Exist	[1] [2]	o	o	
	482	Loop Detected	[1] [2]	o	o	
	483	Too Many Hops	[1] [2]	o	o	
	484	Address Incomplete	[1] [2]	o	o	
	485	Ambiguous	[1] [2]	o	o	
	486	Busy Here	[1] [2]	o	o	
	487	Request Terminated	[1] [2]	o	o	
	488	Not Acceptable Here	[1] [2]	o	o	
	491	Request Pending	[1] [2]	o	o	
	493	Undecipherable	[1] [2]	o	o	
	Other		[1] [2]	i	o	
5xx	500	Server Internal Error	[1] [2]	o	o	
	501	Not Implemented	[1] [2]	o	o	
	502	Bad Gateway	[1] [2]	o	o	
	503	Service Unavailable	[1] [2]	o	o	
	504	Server Time-out	[1] [2]	o	o	
	505	Version Not Supported	[1] [2]	o	o	
	513	Message Too Large	[1] [2]	o	o	
	Other		[1] [2]	i	o	
6xx	600	Busy Everywhere	[1] [2]	o	o	
	603	Decline	[1] [2]	o	o	
	604	Does Not Exist Anywhere	[1] [2]	o	o	
	606	Not Acceptable	[1] [2]	o	o	
	Other		[1] [2]	i	o	

Table 13-6 lists each SDP field covered in this standard.

Table 13-6/JJ-90.24: SDP

Item	Ref.	Type (caller)	Type (callee)	Remarks (referenced section, etc.)	
Session description	v=	[9]	m	m	Fixed to "0"
	o=	[9]	m	m	
	s=	[9]	m	m	
	i=	[9]	o	o	
	u=	[9]	o	o	
	e=	[9]	o	o	
	p=	[9]	o	o	
	c=	[9]	m	m	
	b=	[9]	o	o	
	z=	[9]	o	o	
	k=	[9]	o	o	
a=	[9]	o	m		
Time description	t=	[9]	m	m	
	r=	[9]	o	o	
Media description	m=	[9]	m	m	
	j=	[9]	o	o	
	c=	[9]	o	o	
	b=	[9]	o	o	
	k=	[9]	o	o	
a=	[9]	m	m		

13.2. Dynamic Variables: Common Conditions

13.2.1. Implementation Values for SIP Terminal Settings and Saved Information

To connect to a provider's SIP network, a SIP terminal sets and saves various types of user information such as the address of servers targeted for connection and one's own AoR. Table 13-7 lists implementation requirements for setting and saving information (such as variable lengths) with regard to minimally required information.

Table 13-7/JJ-90.24: Implementation requirements for settings and saved information

Set Item	Minimum Implementation Requirements	Remarks
Character length of SIP proxy server address	<ul style="list-style-type: none"> IPv4 address format FQDN format: must support a minimum of 32 characters 	
Character length of SIP registrar server address	<ul style="list-style-type: none"> IPv4 address format FQDN format: must support a minimum of 32 characters 	Since the SIP proxy server as destination of INVITE requests and the SIP registrar server as destination of REGISTER requests may be physically different, the SIP proxy server address is given its own field.
Character length of AoR	<ul style="list-style-type: none"> user part: alphanumeric character string; must support a minimum of 32 characters host part: alphanumeric character string; must support a minimum of 32 characters 	
Character length of user name	Alphanumeric character string: must support a minimum of 32 characters	User name used in HTTP Digest authentication
Character length of password	Alphanumeric character string: must support a minimum of 32 characters	Password used in HTTP Digest authentication

13.2.2. Requirements for SIP Terminal Variable Values

There are variable elements such as dynamically set character strings whose configuration conditions vary depending on the implementation of an SIP terminal. Table 13-8 lists the maximum number of characters for various variable elements at the time of creation by a SIP terminal and the minimum number of characters for the same variable elements that, though not an object of creation, must be received by a SIP terminal from another party under normal connection processing.

Table 13-8/JJ-90.24: Requirements for SIP terminal variable values

Variable Element having Variable Send/Receive Requirements	Requirements at Time of Creation	Requirements at Time of Receive Processing	Remarks
Character length of one SIP header line	Maximum 255 bytes (CRLF included)	Minimum 255 bytes	If character length per SIP header line should exceed 255 bytes due to multiple uri's at time of creation, the header is divided into multiple lines. Example: Record-Route, Route, Via
Character length of realm	-	Minimum 64 bytes	
Number of via's	—	Minimum 10 hops	Usually "1" when sending a SIP request message
Character length of Via-branch	Maximum 32 bytes excluding z9hG4bK	Minimum 128 bytes	
Character length of local uri	—	Minimum 128 bytes	uri set in From header of Initial INVITE request
Character length of remote uri	—	Minimum 128 bytes	uri set in To header of Initial INVITE request
Character length of To/From tag	Maximum 32 bytes	Minimum 128 bytes	
Character length of Call-ID	Maximum 64 bytes	Minimum 128 bytes	
Sequence number of CSeq header	As a default value, a number from 1 to 999900	Minimum representation range for a 32-bit unsigned integer	
Sequence number of RSeq header	As a default value, a number no greater than 999900	Minimum representation range for a 32-bit unsigned integer	
Character length of location information	user part: maximum 32 bytes Total: maximum 64 bytes	-	SIP URI set in Contact header of REGISTER request
Character length of remote target	user part: maximum 32 bytes Total: maximum 64 bytes	Minimum 128 bytes	uri set in Contact header of own Initial INVITE request or in 1xx/2xx response (same requirements for uri set in Contact header of REGISTER request)
Number of route sets	—	Minimum 10 hops	Sending requirements follow receiving requirements.
Character length of one rec-route	—	Minimum 128 bytes	Sending requirements follow receiving requirements.

Variable Element having Variable Send/Receive Requirements	Requirements at Time of Creation	Requirements at Time of Receive Processing	Remarks
Character length of username on SDP "o=" line	Maximum 10 bytes	Minimum 64 bytes	
Character length of session id on SDP "o=" line	As a default value, a number no greater than 999900	Minimum representation range for a 64-bit unsigned integer	
Value of version on SDP "o=" line	As a default value, a number no greater than 999900	Minimum representation range for a 64-bit unsigned integer	
Character length of session name on SDP "s=" line	Maximum 10 bytes	Minimum 64 bytes	

13.3. SIP Request Signals

This section lists each header for each of the SIP request signals (see Table 13-1) covered by this standard.

13.3.1. ACK Request (at time of 2xx response)

Table 13-9 lists each header of an ACK request when the final response to an Initial INVITE request or re-INVITE request is a 2xx response.

Table 13-9 /JJ-90.24: ACK request settings (2xx response)

Item		Ref.	Type (caller)	Type (callee)	Remarks (referenced section in this document, etc.)
Request Line	Method	[1]	m	m	
	Request-URI	[1]	m	m	Sections 5.1.2 and 6.1.2
	SIP-Version	[1]	m	m	Fixed to "SIP/2.0"
Message Header	Authorization	[1]	x	o	
	Call-ID	[1]	m	m	
	Contact	[1]	o	o	
	Content-Disposition	[1]	x	o	
	Content-Encoding	[1]	x	o	
	Content-Language	[1]	x	o	
	Content-Length	[1]	m	o	
	Content-Type	[1]	x	o	
	CSeq	[1]	m	m	
	Date	[1]	o	o	
	From	[1]	m	m	
	Max-Forwards	[1]	m	o	
	MIME-Version	[1]	x	o	
	Privacy	[5]	x	o	
	Proxy-Authorization	[1]	o	o	
	Record-Route	[1]	x	o	
	Route	[1]	m	o	
	Timestamp	[1]	o	o	
	To	[1]	m	m	
	User-Agent	[1]	o	o	
Via	[1]	m	m		
Other			i	o	
SDP		[9]	x	o	

13.3.2. ACK Request (at time of 3xx/4xx/5xx/6xx responses)

Table 13-10 /JJ-90.24: ACK request settings (3xx/4xx/5xx/6xx responses)

Item		Ref.	Type (caller)	Type (callee)	Remarks (referenced section, etc.)
Request Line	Method	[1]	m	m	
	Request-URI	[1]	m	m	Same value as INVITE Request-URI
	SIP-Version	[1]	m	m	Fixed to "SIP/2.0"
Message Header	Authorization	[1]	x	o	
	Call-ID	[1]	m	m	
	Contact	[1]	o	o	
	Content-Disposition	[1]	x	o	
	Content-Encoding	[1]	x	o	
	Content-Language	[1]	x	o	
	Content-Length	[1]	m	o	
	Content-Type	[1]	x	o	
	CSeq	[1]	m	m	
	Date	[1]	o	o	
	From	[1]	m	m	
	Max-Forwards	[1]	m	o	
	MIME-Version	[1]	x	o	
	Privacy	[5]	x	o	
	Proxy-Authorization	[1]	o	o	
	Record-Route	[1]	x	o	
	Route	[1]	x	o	
	Timestamp	[1]	o	o	
	To	[1]	m	m	
	User-Agent	[1]	o	o	
Via	[1]	m	m	Branch-id is same value as that of INVITE.	
Other			o	o	
SDP		[9]	x	o	

13.3.3. BYE Request

Table 13-11 /JJ-90.24: BYE request settings

	Item	Ref.	Type (caller)	Type (callee)	Remarks (referenced section, etc.)
Request Line	Method	[1]	m	m	
	Request-URI	[1]	m	m	Follows Record-Routing behaviors (see Section 7)
	SIP-Version	[1]	m	m	Fixed to "SIP/2.0"
Message Header	Accept	[1]	o	o	
	Accept-Encoding	[1]	o	o	
	Accept-Language	[1]	o	o	
	Allow	[1]	o	o	
	Authorization	[1]	x	o	
	Call-ID	[1]	m	m	
	Content-Disposition	[1]	x	o	
	Content-Encoding	[1]	x	o	
	Content-Language	[1]	x	o	
	Content-Length	[1]	m	o	
	Content-Type	[1]	x	o	
	CSeq	[1]	m	m	
	Date	[1]	o	o	
	From	[1]	m	m	
	Max-Forwards	[1]	m	o	
	MIME-Version	[1]	x	o	
	P-Asserted-Identity	[7]	x	o	
	P-Preferred-Identity	[7]	x	o	
	Privacy	[5]	x	o	
	Proxy-Authorization	[1]	o	o	
	Proxy-Require	[1]	x	o	
	Record-Route	[1]	x	o	
	Require	[1]	x	o	
	Route	[1]	m	o	
	Supported	[1]	o	o	
	Timestamp	[1]	o	o	
	To	[1]	m	m	
	User-Agent	[1]	o	o	
	Via	[1]	m	m	
Other			i	o	
SDP		[9]	x	o	

13.3.4. CANCEL Request

Table 13-12 /JJ-90.24: CANCEL request settings

Item		Ref.	Type (caller)	Type (callee)	Remarks (referenced section, etc.)
Request Line	Method	[1]	m	m	
	Request-URI	[1]	m	m	Set to same value as that of INVITE
	SIP-Version	[1]	m	m	Fixed to "SIP/2.0"
Message Header	Authorization	[1]	x	o	
	Call-ID	[1]	m	m	
	Content-Length	[1]	m	o	
	CSeq	[1]	m	m	
	Date	[1]	o	o	
	From	[1]	m	m	
	Max-Forwards	[1]	m	o	
	Privacy	[5]	x	o	
	Record-Route	[1]	x	o	
	Route	[1]	x	o	
	Supported	[1]	o	o	
	Timestamp	[1]	o	o	
	To	[1]	m	m	
	User-Agent	[1]	o	o	
	Via	[1]	m	m	
	Other			i	o
SDP		[9]	x	o	

13.3.5. Initial INVITE Request

Table 13-13 /JJ-90.24: Initial INVITE request settings

	Item	Ref.	Type (caller)	Type (callee)	Remarks (referenced section, etc.)
Request Line	Method	[1]	m	m	
	Request-URI	[1]	m	m	
	SIP-Version	[1]	m	m	Fixed to "SIP/2.0"
Message Header	Accept	[1]	o	o	
	Accept-Encoding	[1]	o	o	
	Accept-Language	[1]	o	o	
	Alert-Info	[1]	o	o	
	Allow	[1]	m	c1	At sending, insert "PRACK" if using the 100rel extension and "UPDATE" if using the UPDATE method ²⁷ .
	Authorization	[1]	o	o	
	Call-ID	[1]	m	m	
	Call-Info	[1]	o	o	
	Contact	[1]	m	m	
	Content-Disposition	[1]	x	o	
	Content-Encoding	[1]	x	o	
	Content-Language	[1]	x	o	
	Content-Length	[1]	m	o	
	Content-Type	[1]	m	m	
	CSeq	[1]	m	m	
	Date	[1]	o	o	
	Expires	[1]	o	o	
	From	[1]	m	m	
	In-Reply-To	[1]	o	o	
	Max-Forwards	[1]	m	o	
	MIME-Version	[1]	x	o	
	Min-SE	[16]	c2	c3	
	Organization	[1]	o	o	
	P-Asserted-Identity	[7]	x	m	
	P-Preferred-Identity	[7]	o	o	
	Priority	[1]	o	o	
	Privacy	[5][7]	c4	m	
Proxy-Authorization	[1]	m	o		
Proxy-Require	[1]	x	o		
Record-Route	[1]	x	m		
Reply-To	[1]	o	o		
Require	[1]	x	m		
Route	[1]	x	o		
Session-Expires	[16]	c5	c6		

²⁷ Although these functions (100rel extension and UPDATE method) must be implemented, their use may be stopped by the provider's SIP network.

Subject	[1]	o	o	
Supported	[1][2] [16]	c7	c7	
Timestamp	[1]	o	o	
To	[1]	m	m	
User-Agent	[1]	o	o	
Via	[1]	m	m	
Other		i	o	
SDP	[9]	m	m	

c1: "m" if using the UPDATE method; "o" otherwise ²⁸
c2: "m" if using the session timer extension and a 422 (Session Interval Too Small) response has been previously received by the same calling request (same Call-ID); "o" if using the session timer extension and a 422 (Session Interval Too Small) response has not been previously received by the same calling request (same Call-ID); "x" if not using the session timer extension" ²⁹
c3: "m" if using the session timer extension; "o" otherwise ³⁰
c4: "m" if using Table 12-1 or 12-2; "x" if using Table 12-3 or 12-4
c5: "m" if using the session timer extension; "x" otherwise ³¹
c6: "m" if using the session timer extension; "o" otherwise ³²
c7: "m" if using the 100rel or session timer extension; "i" otherwise ³³

²⁸ Although this function must be implemented, its use may be stopped by the provider's SIP network.

²⁹ Although this function must be implemented, its use may be stopped by the provider's SIP network.

³⁰ Although this function must be implemented, its use may be stopped by the provider's SIP network.

³¹ Although this function must be implemented, its use may be stopped by the provider's SIP network.

³² Although this function must be implemented, its use may be stopped by the provider's SIP network.

³³ Although these functions (100rel and session timer extensions) must be implemented, their use may be stopped by the provider's SIP network.

13.3.6. PRACK Request

Table 13-14 /JJ-90.24: PRACK request settings

	Item	Ref.	Type (caller)	Type (callee)	Remarks (referenced section, etc.)
Request Line	Method	[1] [2]	m	m	
	Request-URI	[1] [2]	m	m	Follows Record-Routing behaviors (see Section 7)
	SIP-Version	[1] [2]	m	m	Fixed to "SIP/2.0"
Message Header	Accept	[1] [2]	o	o	
	Accept-Encoding	[1] [2]	o	o	
	Accept-Language	[1] [2]	o	o	
	Allow	[1] [2]	o	o	
	Authorization	[1] [2]	x	o	
	Call-ID	[1] [2]	m	m	
	Content-Disposition	[1] [2]	x	o	
	Content-Encoding	[1] [2]	x	o	
	Content-Language	[1] [2]	x	o	
	Content-Length	[1] [2]	m	o	
	Content-Type	[1] [2]	x	o	
	CSeq	[1] [2]	m	m	
	Date	[1] [2]	o	o	
	From	[1] [2]	m	m	
	Max-Forwards	[1] [2]	m	o	
	MIME-Version	[1] [2]	x	o	
	Privacy	[2] [5]	x	o	
	Proxy-Authorization	[1] [2]	o	o	
	Proxy-Require	[1] [2]	x	o	
	Record-Route	[1] [2]	x	o	
	Require	[1] [2]	x	o	
	Route	[1] [2]	m	o	
	RAck	[1] [2]	m	m	
	Supported	[1] [2]	o	o	
	Timestamp	[1] [2]	o	o	
	To	[1] [2]	m	m	
User-Agent	[1] [2]	o	o		
Via	[1] [2]	m	m		
Other			i	o	
SDP		[9]	x	o	

13.3.7. re-INVITE Request

Table 13-15 /JJ-90.24: re-INVITE request settings

	Item	Ref.	Type (caller)	Type (callee)	Remarks (referenced section, etc.)
Request Line	Method	[1]	m	m	
	Request-URI	[1]	m	m	Follows Record-Routing behaviors (see Section 7)
	SIP-Version	[1]	m	m	Fixed to "SIP/2.0"
Message Header	Accept	[1]	o	o	
	Accept-Encoding	[1]	o	o	
	Accept-Language	[1]	o	o	
	Alert-Info	[1]	o	o	
	Allow	[1]	m	o	At sending, insert "UPDATE" if using the UPDATE method. ³⁴
	Authorization	[1]	x	o	
	Call-ID	[1]	m	m	
	Call-Info	[1]	o	o	
	Contact	[1]	m	m	
	Content-Disposition	[1]	x	o	
	Content-Encoding	[1]	x	o	
	Content-Language	[1]	x	o	
	Content-Length	[1]	m	o	
	Content-Type	[1]	m	m	
	CSeq	[1]	m	m	
	Date	[1]	o	o	
	Expires	[1]	o	o	
	From	[1]	m	m	
	In-Reply-To	[1]	o	o	
	Max-Forwards	[1]	m	o	
	MIME-Version	[1]	x	o	
	Min-SE	[16]	c2	c3	
	Organization	[1]	o	o	
	P-Asserted-Identity	[7]	x	o	
	P-Preferred-Identity	[7]	x	o	
	Priority	[1]	o	o	
	Privacy	[5]	x	o	
	Proxy-Authorization	[1]	o	o	
	Proxy-Require	[1]	x	o	
	Record-Route	[1]	x	o	
Reply-To	[1]	o	o		
Require	[1]	x	o		
Route	[1]	m	o		

³⁴ Although this function must be implemented, its use may be stopped by the provider's SIP network.

Session-Expires	[16]	c4	c3	
Subject	[1]	o	o	
Supported	[1] [16]	c5	c5	
Timestamp	[1]	o	o	
To	[1]	m	m	
User-Agent	[1]	o	o	
Via	[1]	m	m	
Other		i	o	
SDP	[9]	m	m	See Section 10 of this document.

c1: "m" if using the UPDATE method; "o" otherwise ³⁵
c2: "m" if using the session timer extension and a Min-SE header has been previously received in a 422 (Session Interval Too Small) response or in a session-refresh request (INVITE/UPDATE) within the same dialog; "o" if using the session timer extension and a Min-SE header has not been previously received in a 422 (Session Interval Too Small) response or in a session-refresh request (INVITE/UPDATE) within the same dialog; "x" if not using the session timer extension ³⁶
c3: "m" if using the session timer extension; "o" otherwise ³⁷
c4: "m" if using the session timer extension; "x" otherwise ³⁸
c5: "m" if using the session timer extension; "i" otherwise ³⁹

³⁵ Although this function must be implemented, its use may be stopped by the provider's SIP network.

³⁶ Although this function must be implemented, its use may be stopped by the provider's SIP network.

³⁷ Although this function must be implemented, its use may be stopped by the provider's SIP network.

³⁸ Although this function must be implemented, its use may be stopped by the provider's SIP network.

³⁹ Although these functions (100rel and session timer extensions) must be implemented, their use may be stopped by the provider's SIP network.

13.3.8. REGISTER Request

Table 13-16 /JJ-90.24: REGISTER request settings

	Item	Ref.	Type (caller)	Type (callee)	Remarks (referenced section, etc.)
Request Line	Method	[1]	m	n/a	
	Request-URI	[1]	m	n/a	
	SIP-Version	[1]	m	n/a	Fixed to "SIP/2.0"
Message Header	Accept	[1]	o	n/a	
	Accept-Encoding	[1]	o	n/a	
	Accept-Language	[1]	o	n/a	
	Allow	[1]	o	n/a	
	Authorization	[1]	c1	n/a	
	Call-ID	[1]	m	n/a	
	Call-Info	[1]	o	n/a	
	Contact	[1]	c2	n/a	
	Content-Disposition	[1]	x	n/a	
	Content-Encoding	[1]	x	n/a	
	Content-Language	[1]	x	n/a	
	Content-Length	[1]	m	n/a	
	Content-Type	[1]	x	n/a	
	CSeq	[1]	m	n/a	
	Date	[1]	o	n/a	
	Expires	[1]	o	n/a	
	From	[1]	m	n/a	
	Max-Forwards	[1]	m	n/a	
	MIME-Version	[1]	x	n/a	
	Organization	[1]	o	n/a	
	Privacy	[5]	x	n/a	
	Proxy-Authorization	[1]	x	n/a	
	Proxy-Require	[1]	x	n/a	
	Require	[1]	x	n/a	
	Route	[1]	x	n/a	
	Supported	[1]	o	n/a	
	Timestamp	[1]	o	n/a	
To	[1]	m	n/a		
User-Agent	[1]	o	n/a		
Via	[1]	m	n/a		
Other	[1]	i	n/a		
SDP		[9]	x	n/a	
<p>c1: "m" if sending a request after receiving an authentication request by a 401 (Unauthorized) response; "x" otherwise</p> <p>c2: "x" if confirming location information as described in Section 4.1.7; "m" otherwise</p>					

Table 13-17 /JJ-90.24: UPDATE request settings

	Item	Ref.	Type (caller)	Type (callee)	Remarks (referenced section, etc.)
Request Line	Method	[1] [4]	m	m	
	Request-URI	[1] [4]	m	m	Follows Record-Routing behaviors (see Section 7)
	SIP-Version	[1] [4]	m	m	Fixed to "SIP/2.0"
Message Header	Accept	[1] [4]	o	o	
	Accept-Encoding	[1] [4]	o	o	
	Accept-Language	[1] [4]	o	o	
	Allow	[1] [4]	m	o	
	Authorization	[1] [4]	x	o	
	Call-ID	[1] [4]	m	m	
	Call-Info	[1] [4]	o	o	
	Contact	[1] [4]	m	m	
	Content-Disposition	[1] [4]	x	o	
	Content-Encoding	[1] [4]	x	o	
	Content-Language	[1] [4]	x	o	
	Content-Length	[1] [4]	m	o	
	Content-Type	[1] [4]	o	o	
	CSeq	[1] [4]	m	m	
	Date	[1] [4]	o	o	
	From	[1] [4]	m	m	
	Max-Forwards	[1] [4]	m	o	
	MIME-Version	[1] [4]	x	o	
	Min-SE	[4] [16]	c1	c2	
	Organization	[1] [4]	o	o	
	Privacy	[4] [5]	x	o	
	Proxy-Authorization	[1] [4]	o	o	
	Proxy-Require	[1] [4]	x	o	
	Record-Route	[1] [4]	x	o	
	Require	[1] [4]	x	o	
	Route	[1] [4]	m	o	
	Session-Expires	[4] [16]	c3	c2	
	Supported	[1] [4] [16]	c4	c4	
	Timestamp	[1] [4]	o	o	
	To	[1] [4]	m	m	
User-Agent	[1] [4]	o	o		
Via	[1] [4]	m	m		
Other			i	o	
SDP		[9]	c5	m	

- c1: “m” if using the session timer extension and a Min-SE header has been previously received in a 422 (Session Interval Too Small) response or in a session-refresh request (INVITE/UPDATE) within the same dialog; “o” if using the session timer extension and a Min-SE header has not been previously received in a 422 (Session Interval Too Small) response or in a session-refresh request (INVITE/UPDATE) within the same dialog; “x” if not using the session timer extension ⁴⁰
- c2: “m” if using the session timer extension; “o” otherwise ⁴¹
- c3: “m” if using the session timer extension; “x” otherwise ⁴²
- c4: “m” if using the session timer extension; “i” otherwise ⁴³
- c5: “x” if using only for session refresh as part of the session timer extension; “o” otherwise

13.4. SIP Response Signals

This section lists the headers in each of the response signals associated with each of the SIP request signals (see Table 13-1) covered by this standard.

13.4.1. Responses to BYE Request

⁴⁰ Although this function must be implemented, its use may be stopped by the provider’s SIP network.

⁴¹ Although this function must be implemented, its use may be stopped by the provider’s SIP network.

⁴² Although this function must be implemented, its use may be stopped by the provider’s SIP network.

⁴³ Although this function must be implemented, its use may be stopped by the provider’s SIP network.

Table 13-18/JJ-90.24: Response settings to BYE request

Item	status-Code	R ef.	Type (caller)	Type (callee)	Remarks (referenced section, etc.)	
status-Line	SIP-Version	all codes	[1]	m	m	Fixed to "SIP/2.0"
	Status-Code	all codes	[1]	m	m	
	Reason-Phrase	all codes	[1]	m	o	
Message Header	Accept	415	[1]	m	o	
	Accept-Encoding	415	[1]	m	o	
	Accept-Language	415	[1]	m	o	
	Allow	1xx,2xx	[1]	o	o	
	Allow	300-699	[1]	o	o	
	Allow	405	[1]	m	o	
	Authentication-Info	2xx	[1]	o	o	
	Call-ID	all codes	[1]	m	m	
	Contact	3xx 485	[1]	o	o	
	Content-Disposition	all codes	[1]	x	o	
	Content-Encoding	all codes	[1]	x	o	
	Content-Language	all codes	[1]	x	o	
	Content-Length	all codes	[1]	m	o	
	Content-Type	all codes	[1]	x	o	
	CSeq	all codes	[1]	m	m	
	Date	all codes	[1]	o	o	
	Error-Info	300-699	[1]	o	o	
	From	all codes	[1]	m	m	
	MIME-Version	all codes	[1]	x	o	
	P-Asserted-Identity	all codes	[7]	x	o	
	P-Preferred-Identity	all codes	[7]	x	o	
	Privacy		[5]	x	o	
	Proxy-Authenticate	401	[1]	x	o	
	Proxy-Authenticate	407	[1]	x	m	
	Record-Route	2xx, 18x	[1]	x	o	
	Require	all codes	[1]	x	o	
	Retry-After	404,413, 480,486, 500,503, 600,603	[1]	o	o	
	Server	all codes	[1]	o	o	
	Supported	2xx	[1]	o	o	
	Timestamp	all codes	[1]	o	o	
To	all codes	[1]	m	m		
Unsupported	420	[1]	m	o		
User-Agent	all codes	[1]	o	o		
Via	all codes	[1]	m	m		
Warning	all codes	[1]	o	o		

	www-Authenticate	401 407	[1]	x	o	
	Other			i	o	
SDP			[9]	x	o	

13.4.2. Responses to CANCEL Request

Table 13-19/JJ-90.24: Response settings to CANCEL request

	Item	status-Code	Ref.	Type (caller)	Type (callee)	Remarks (referenced section, etc.)
status-Line	SIP-Version	all codes	[1]	m	m	Fixed to "SIP/2.0"
	Status-Code	all codes	[1]	m	m	
	Reason-Phrase	all codes	[1]	m	o	
Message Header	Call-ID	all codes	[1]	m	m	
	Content-Length	all codes	[1]	m	o	
	CSeq	all codes	[1]	m	m	
	Date	all codes	[1]	o	o	
	Error-Info	300-699	[1]	o	o	
	From	all codes	[1]	m	m	
	Proxy-Authenticate	401	[1]	x	o	
	Privacy	all codes	[5]	x	o	
	Record-Route	2xx, 18x	[1]	x	o	
	Retry-After	404,413,480,486,500,503,600,603	[1]	o	o	
	Server	all codes	[1]	o	o	
	Supported	2xx	[1]	o	o	
	Timestamp	all codes	[1]	o	o	
	To	all codes	[1]	m	m	
	User-Agent	all codes	[1]	o	o	
	Via	all codes	[1]	m	m	
	Warning	all codes	[1]	o	o	
Other				i	o	
SDP			[9]	x	o	

13.4.3. Responses to Initial INVITE Request

Table 13-20/JJ-90.24: Response settings to Initial INVITE request

	Item	status-Code	Ref.	Type (caller)	Type (callee)	Remarks (referenced section, etc.)
status-Line	SIP-Version	all codes	[1]	m	m	Fixed to "SIP/2.0"
	Status-Code	all codes	[1]	m	m	
	Reason-Phrase	all codes	[1]	m	o	
Message Header	Accept	2xx	[1]	o	o	
	Accept	415	[1]	m	o	
	Accept-Encoding	2xx	[1]	o	o	
	Accept-Encoding	415	[1]	m	o	
	Accept-Language	2xx	[1]	o	o	
	Accept-Language	415	[1]	m	o	
	Alert-Info	180	[1]	o	o	
	Allow	1xx,2xx	[1]	m	c1	At sending, insert "PRACK" if using the 100rel extension and "UPDATE" if using the UPDATE method ⁴⁴ .
	Allow	300-699	[1]	o	o	
	Allow	405	[1]	m	o	
	Authentication-Info	2xx	[1]	o	o	
	Call-ID	all codes	[1]	m	m	
	Call-Info	all codes	[1]	o	o	
	Contact	1xx,	[1]	m	c2	
		2xx	[1]	m	m	
	Contact	300-699	[1]	o	o	
	Contact	485	[1]	o	o	
	Content-Disposition	all codes	[1]	x	o	
	Content-Encoding	all codes	[1]	x	o	
	Content-Language	all codes	[1]	x	o	
	Content-Length	all codes	[1]	m	o	
	Content-Type	1xx	[1]	x	m	
	Content-Type	2xx	[1]	m	m	
	Content-Type	300-699	[1]	x	o	
	CSeq	all codes	[1]	m	m	
	Date	all codes	[1]	o	o	
	Error-Info	300-699	[1]	o	o	
	Expires	all codes	[1]	o	o	
	From	all codes	[1]	m	m	
	MIME-Version	all codes	[1]	x	o	
Min-SE	422	[16]	m	m		
Organization	all codes	[1]	o	o		
P-Asserted-Identity		[7]	x	o		

⁴⁴ Although these functions (100rel extension and UPDATE method) must be implemented, their use may be stopped by the provider's SIP network.

P-Preferred-Identity		[7]	x	o		
Privacy		[5][7]	x	o		
Proxy-Authenticate	401	[1]	x	o		
Proxy-Authenticate	407	[1]	x	m		
Record-Route	2xx,18x	[1]	m	m		
Reply-To	all codes	[1]	o	o		
Require	1xx	[1] [2]	c3	c4		
Require	2xx	[1] [16]	c5	c6		
Require	300-699	[1]	x	o		
Retry-After	404,413, 480,486, 500,503, 600,603	[1]	o	o		
RSeq	1xx	[1] [2]	c3	c4		
Server	all codes	[1]	o	o		
Session-Expires	2xx	[16]	c7	c6		
Supported	2xx	[1]	c6	i		
Timestamp	all codes	[1]	o	o		
To	all codes	[1]	m	m		
User-Agent	all codes	[1]	o	o		
Unsupported	420	[1]	m	o		
Via	all codes	[1]	m	m		
Warning	all codes	[1]	o	o		
WWW-Authenticate	401 407	[1]	x	o		
Other	all codes		i	o		
SDP		1xx	[9]	x	m	
		2xx	[9]	m	m	
		3xx/4xx/5 xx/6xx	[9]	x	o	

c1: "m" if using the 100rel extension or the UPDATE method; "o" otherwise ⁴⁵
c2: "m" if using the 100rel extension; "o" otherwise ⁴⁶
c3: "m" if using the 100rel extension and "100rel" is set in the Supported header of a received INVITE request; "x" otherwise ⁴⁷
c4: "m" if using the 100rel extension; "i" otherwise ⁴⁸
c5: "m" if using the session timer extension and "timer" is set in the Supported header of a received INVITE request; "x" otherwise ⁴⁹
c6: "m" if using the session timer extension; "i" otherwise ⁵⁰
c7: "m" if using the session timer extension and the Session-Expires header is set in a received INVITE request; "x" otherwise ⁵¹

⁴⁵ Although these functions (100rel extension and UPDATE method) must be implemented, their use may be stopped by the provider's SIP network.

⁴⁶ Although this function must be implemented, its use may be stopped by the provider's SIP network.

⁴⁷ Although this function must be implemented, its use may be stopped by the provider's SIP network.

⁴⁸ Although this function must be implemented, its use may be stopped by the provider's SIP network.

⁴⁹ Although this function must be implemented, its use may be stopped by the provider's SIP network.

⁵⁰ Although this function must be implemented, its use may be stopped by the provider's SIP network.

⁵¹ Although this function must be implemented, its use may be stopped by the provider's SIP network.

Table 13-21/JJ-90.24: Response settings to PRACK request

Item	status-Code	Ref.	Type (caller)	Type (callee)	Remarks (referenced section, etc.)	
status-Line	SIP-Version	all codes	[1] [2]	m	m	Fixed to "SIP/2.0"
	Status-Code	all codes	[1] [2]	m	m	
	Reason-Phrase	all codes	[1] [2]	m	o	
Message Header	Accept	415	[1] [2]	m	o	
	Accept-Encoding	415	[1] [2]	m	o	
	Accept-Language	415	[1] [2]	m	o	
	Allow	all codes	[1] [2]	o	o	
	Authentication-Info	2xx	[1] [2]	o	o	
	Call-ID	all codes	[1] [2]	m	m	
	Contact	3xx 485	[1] [2]	o	o	
	Content-Disposition	all codes	[1] [2]	x	o	
	Content-Encoding	all codes	[1] [2]	x	o	
	Content-Language	all codes	[1] [2]	x	o	
	Content-Length	all codes	[1] [2]	m	o	
	Content-Type	all codes	[1] [2]	x	o	
	CSeq	all codes	[1] [2]	m	m	
	Date	all codes	[1] [2]	o	o	
	Error-Info	300-699	[1] [2]	o	o	
	From	all codes	[1] [2]	m	m	
	MIME-Version	all codes	[1] [2]	x	o	
	Privacy	all codes	[5]	x	o	
	Proxy-Authenticate	401	[1] [2]	x	o	
	Proxy-Authenticate	407	[1] [2]	x	m	
	Record-Route	2xx,18x	[1] [2]	x	o	
	Require	all codes	[1] [2]	x	o	
	Retry-After	404,413,480,486,500,503,600,603	[1] [2]	o	o	
	Server	all codes	[1] [2]	o	o	
	Supported	2xx	[1] [2]	o	o	
	Timestamp	all codes	[1] [2]	o	o	
To	all codes	[1] [2]	m	m		
Unsupported	420	[1] [2]	m	o		
User-Agent	all codes	[1] [2]	o	o		

	Via	all codes	[1] [2]	m	m	
	WWW-Authenticate	401	[1] [2]	x	o	
	Other	all codes		i	o	
SDP		all codes	[9]	x	o	

13.4.5. Responses to re-INVITE Request

Table 13-22/JJ-90.24: Response settings to re-INVITE request

	Item	status-Code	Ref.	Type (caller)	Type (callee)	Remarks (referenced section, etc.)
status-Line	SIP-Version	all codes	[1]	m	m	Fixed to "SIP/2.0"
	Status-Code	all codes	[1]	m	m	
	Reason-Phrase	all codes	[1]	m	o	
Message Header	Accept	2xx	[1]	o	o	
	Accept	415	[1]	m	o	
	Accept-Encoding	2xx	[1]	o	o	
	Accept-Encoding	415	[1]	m	o	
	Accept-Language	2xx	[1]	o	o	
	Accept-Language	415	[1]	m	o	
	Alert-Info	180	[1]	o	o	
	Allow	2xx	[1]	m	c1	At sending, insert "UPDATE" if using the UPDATE method ⁵² .
	Allow	300-699	[1]	o	o	
	Allow	405	[1]	m	o	
	Authentication-Info	2xx	[1]	o	o	
	Call-ID	all codes	[1]	m	m	
	Call-Info	all codes	[1]	o	o	
	Contact	2xx	[1]	m	m	
	Contact	300-699	[1]	o	o	
	Contact	485	[1]	o	o	
	Content-Disposition	all codes	[1]	x	o	
	Content-Encoding	all codes	[1]	x	o	
	Content-Language	all codes	[1]	x	o	
	Content-Length	all codes	[1]	m	o	
	Content-Type	2xx	[1]	m	m	
	Content-Type	300-699	[1]	x	o	
	CSeq	all codes	[1]	m	m	
	Date	all codes	[1]	o	o	
	Error-Info	300-699	[1]	o	o	
	Expires	all codes	[1]	o	o	
	From	all codes	[1]	m	m	
	MIME-Version	all codes	[1]	x	o	
	Min-SE	422	[16]	m	m	
	Organization	all codes	[1]	o	o	
P-Asserted-Identity	all codes	[7]	x	o		

⁵² Although this function must be implemented, its use may be stopped by the provider's SIP network.

P-Preferred-Identity	all codes	[7]	x	o	
Privacy	all codes	[5]	x	o	
Proxy-Authenticate	401	[1]	x	o	
Proxy-Authenticate	407	[1]	x	m	
Record-Route	2xx	[1]	x	o	
Reply-To	all codes	[1]	o	o	
Require	2xx	[1] [16]	c2	c3	
Require	300-699	[1]	x	o	
Retry-After	404,413, 480,486, 500,503, 600,603	[1]	o	o	
Server	all codes	[1]	o	o	
Session-Expires	2xx	[1] [16]	c4	c3	
Supported	2xx	[1] [16]	c3	o	
Timestamp	all codes	[1]	o	o	
To	all codes	[1]	m	m	
User-Agent	all codes	[1]	o	o	
Unsupported	420	[1]	m	o	
Via	all codes	[1]	m	m	
Warning	all codes	[1]	o	o	
WWW-Authenticate	401 407	[1]	x	o	
Other	all codes		i	o	
SDP	2xx	[9]	m	m	
	3xx/4xx/ 5xx/6xx	[9]	x	o	

c1: "m" if using the UPDATE method; "o" otherwise ⁵³
c2: "m" if using the session timer extension and "timer" is set in the Supported header of a received re-INVITE request; "i" otherwise ⁵⁴
c3: "m" if using the session timer extension; "i" otherwise ⁵⁵
c4: "m" if using the session timer extension and the Session-Expires header is set in a received re-INVITE request; "x" otherwise ⁵⁶

⁵³ Although this function must be implemented, its use may be stopped by the provider's SIP network.

⁵⁴ Although this function must be implemented, its use may be stopped by the provider's SIP network.

⁵⁵ Although this function must be implemented, its use may be stopped by the provider's SIP network.

⁵⁶ Although this function must be implemented, its use may be stopped by the provider's SIP network.

Table 13-23/JJ-90.24: Response settings to REGISTER request

	Item	status-Code	Ref.	Type (caller)	Type (callee)	Remarks (referenced section, etc.)
status-Line	SIP-Version	all codes	[1]	n/a	m	Fixed to "SIP/2.0"
	Status-Code	all codes	[1]	n/a	m	
	Reason-Phrase	all codes	[1]	n/a	o	
Message Header	Accept	2xx	[1]	n/a	o	
	Accept	415	[1]	n/a	o	
	Accept-Encoding	2xx	[1]	n/a	o	
	Accept-Encoding	415	[1]	n/a	o	
	Accept-Language	2xx	[1]	n/a	o	
	Accept-Language	415	[1]	n/a	o	
	Allow	all codes	[1]	n/a	o	
	Authentication-Info	2xx	[1]	n/a	o	
	Call-ID	all codes	[1]	n/a	m	
	Call-Info	all codes	[1]	n/a	o	
	Contact	2xx,	[1]	n/a	m	
	Contact	3xx	[1]	n/a	o	
	Contact	485	[1]	n/a	o	
	Content-Disposition	all codes	[1]	n/a	o	
	Content-Encoding	all codes	[1]	n/a	o	
	Content-Language	all codes	[1]	n/a	o	
	Content-Length	all codes	[1]	n/a	o	
	Content-Type	all codes	[1]	n/a	o	
	CSeq	all codes	[1]	n/a	m	
	Date	all codes	[1]	n/a	o	
	Error-Info	300-699	[1]	n/a	o	
	Expires	all codes	[1]	n/a	m	
	From	all codes	[1]	n/a	m	
	Min-Expires	423	[16]	n/a	m	
	MIME-Version	all codes	[1]	n/a	o	
	Organization	all codes	[1]	n/a	o	
Privacy		[5]	n/a	o		

Proxy-Authenticate	401 407	[1]	n/a	x	
Require	all codes	[1]	n/a	o	
Retry-After	404,413,480,486,500,503,600,603	[1]	n/a	o	
Server	all codes	[1]	n/a	o	
Supported	2xx	[1]	n/a	o	
Timestamp	all codes	[1]	n/a	o	
To	all codes	[1]	n/a	m	
Unsupported	420	[1]	n/a	o	
User-Agent	all codes	[1]	n/a	o	
Via	all codes	[1]	n/a	m	
Warning	all codes	[1]	n/a	o	
WWW-Authenticate	401	[1]	n/a	m	
WWW-Authenticate	407	[1]	n/a	x	
Other	all codes		n/a	o	
SDP	all codes	[9]	n/a	o	

13.4.7. Responses to UPDATE Request

Table 13-24/JJ-90.24: Response settings to UPDATE request

	Item	status-Code	Ref.	Type (caller)	Type (callee)	Remarks (referenced section, etc.)
status-Line	SIP-Version	all codes	[1] [4]	m	m	Fixed to "SIP/2.0"
	Status-Code	all codes	[1] [4]	m	m	
	Reason-Phrase	all codes	[1] [4]	m	o	
Message Header	Accept	2xx	[1] [4]	o	o	
	Accept	415	[1] [4]	m	o	
	Accept-Encoding	2xx	[1] [4]	o	o	
	Accept-Encoding	415	[1] [4]	m	o	
	Accept-Language	2xx	[1] [4]	o	o	
	Accept-Language	415	[1] [4]	m	o	
	Allow	2xx	[1] [4]	m	c1	
	Allow	300-699	[1] [4]	o	o	
	Allow	405	[1] [4]	m	o	
	Authentication-Info	2xx	[1] [4]	o	o	
	Call-ID	all codes	[1] [4]	m	m	
	Call-Info	all codes	[1] [4]	o	o	
	Contact	2xx	[1] [4]	m	m	
	Contact	300-699	[1] [4]	o	o	
	Contact	485	[1] [4]	o	o	
	Content-Disposition	all codes	[1] [4]	x	o	
	Content-Encoding	all codes	[1] [4]	x	o	
	Content-Language	all codes	[1] [4]	x	o	
	Content-Length	all codes	[1] [4]	m	o	
	Content-Type	2xx	[1] [4]	c2	c3	
	Content-Type	300-699	[1] [4]	x	o	
	CSeq	all codes	[1] [4]	m	m	
	Date	all codes	[1] [4]	o	o	
	Error-Info	300-699	[1] [4]	o	o	
	From	all codes	[1] [4]	m	m	
	MIME-Version	all codes	[1] [4]	x	o	
	Min-SE	422	[16]	m	m	
	Organization	all codes	[1] [4]	o	o	
	Privacy	all codes	[5]	x	o	
	Proxy-Authenticate	401	[1] [4]	o	x	
	Proxy-Authenticate	407	[1] [4]	x	m	
	Record-Route	2xx	[1] [4]	x	o	
Reply-To	all codes	[1] [4]	o	o		
Require	2xx	[1] [4] [16]	c4	c5		
Require	300-699	[1] [4]	x	o		

Retry-After	404,413, 480,486, 500,503, 600,603	[1] [4]	o	o	
Server	all codes	[1] [4]	o	o	
Session-Expires	2xx	[16]	c6	c5	
Supported	2xx	[1] [4]	c5	o	
Timestamp	all codes	[1] [4]	o	o	
To	all codes	[1] [4]	m	m	
User-Agent	all codes	[1] [4]	o	o	
Unsupported	420	[1] [4]	m	o	
Via	all codes	[1] [4]	m	m	
Warning	all codes	[1] [4]	o	o	
WWW-Authenticate	401 407	[1] [4]	x	o	
Other	all codes		i	o	
SDP	2xx	[9]	c1	c7	
	3xx/4xx/ 5xx/6xx	[9]	x	o	

c1: "m" if using the UPDATE method; "o" otherwise ⁵⁷
c2: "m" if SDP is included in a received UPDATE request; "x" otherwise
c3: "m" if SDP is included in a sent UPDATE request; "i" otherwise
c4: "m" if using the session timer extension and "timer" is set in the Supported header of a received UPDATE request; "i" otherwise ⁵⁸
c5: "m" if using the session timer extension; "i" otherwise ⁵⁹
c6: "m" if using the session timer extension and the Session-Expires header is set in a received UPDATE request; "i" otherwise
c7: "m" if SDP is included in a sent UPDATE request; "x" otherwise

⁵⁷ Although this function must be implemented, its use may be stopped by the provider's SIP network.

⁵⁸ Although this function must be implemented, its use may be stopped by the provider's SIP network.

⁵⁹ Although this function must be implemented, its use may be stopped by the provider's SIP network.

Appendix i: Sequence Examples

This appendix provides examples of establishment of a call sequences corresponding to typical send/receive behaviors in SIP call establishment.

Please note that the sequence examples listed here are meant only as reference material for system implementation and may be changed as needed due to actual service contents and/or terminal functions in the provider's SIP network. Note also that the content of these sequence examples do not guarantee call connectivity or quality.

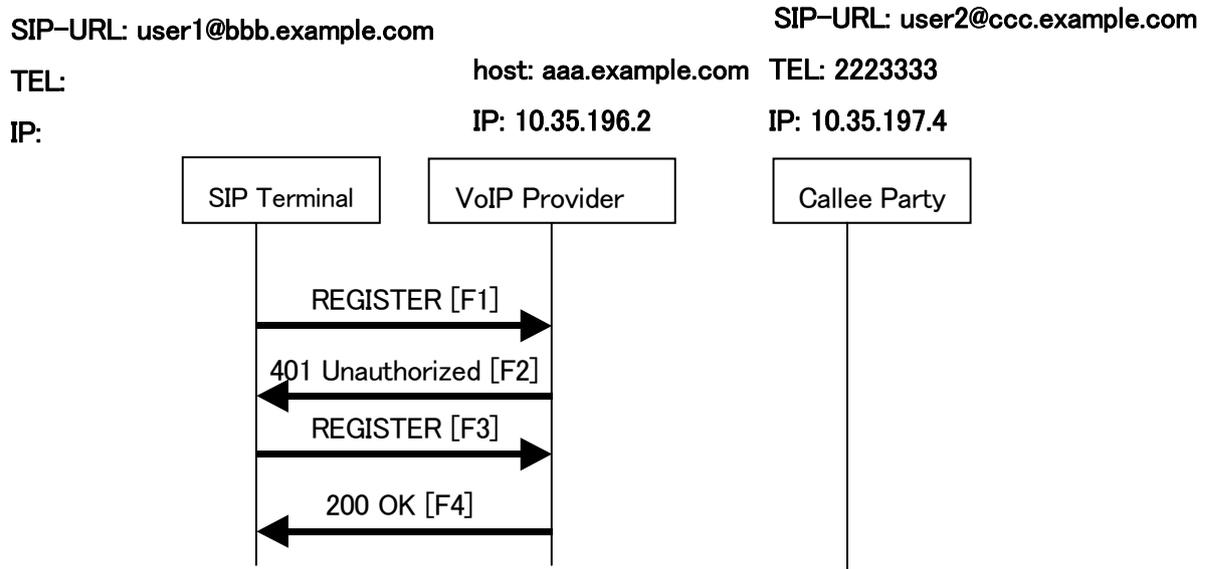
Appendix Table 1-1/JJ-90.24: List of sequence examples

No	Sequence Name	Remarks
1	REGISTER Registration	Appendix Figure 1
2	REGISTER Specific Deletion	Appendix Figure 2
3	REGISTER Total Deletion	Appendix Figure 3
4	Connection Sequence 1: (call/receive 100rel used, call/receive timer used, call/receive UPDATE used)	Appendix Figure 4
5	Connection Sequence 2: (call/receive 100rel not used, call/receive timer used, call/receive UPDATE not used)	Appendix Figure 5
6	Connection Sequence 3: (call/receive 100rel used, call/receive timer used, call UPDATE not used, receive UPDATE used)	Appendix Figure 6
7	Connection Sequence 4: (call/receive 100rel used, call/receive timer used, call UPDATE used, receive UPDATE not used)	Appendix Figure 7
8	Connection Sequence 5: (call/receive 100rel used, call timer not used, receive timer used)	Appendix Figure 8
9	Connection Sequence 6: (call/receive 100rel used, call timer used, receive timer not used)	Appendix Figure 9
10	Connection Sequence 7: (call 100rel not used)	Appendix Figure 10
11	Connection Sequence 8: (call 100rel used, receive 100rel not used)	Appendix Figure 11
12	Call Abortion	Appendix Figure 12

i.1 REGISTER Registration

Appendix Figure 1-1 shows the sequence for REGISTER Registration.

Registration and server functions in this sequence are incorporated in SIP proxy servers. Message examples follow the sequence.



Appendix Figure 1-1/JJ-90.24: REGISTER Registration

Message Details:

F1 REGISTER 2221111 → aaa.example.com

```
REGISTER sip:aaa.example.com SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: <sip:user1@bbb.example.com>;tag=1234567
To: <sip:user1@bbb.example.com>
Call-ID: b84c4d76f6@10.35.195.2
CSeq: 1 REGISTER
Contact: <sip:y3a6sn@10.35.195.2>
Expires: 3600
Content-Length: 0
```

F2 aaa.example.com → 2221111

SIP/2.0 401 Unauthorized

Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK74bf9

From: <sip:user1@bbb.example.com>;tag=1234567

To: <sip:user1@bbb.example.com>;tag=567890

Call-ID: b84c4d76f6@10.35.195.2

CSeq: 1 REGISTER

WWW-Authenticate: Digest realm="aaa.example.com", nonce="ae9137be",
domain="sip:aaa.example.com",algorithm=MD5,opaque="", stale=false

Content-Length: 0

F3 REGISTER 2221111 → aaa.example.com

REGISTER sip:aaa.example.com SIP/2.0

Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK74bfa

Max-Forwards: 70

From: <sip:user1@bbb.example.com>;tag=1234568

To: <sip:user1@bbb.example.com>

Call-ID: b84c4d76f6@10.35.195.2

CSeq: 2 REGISTER

Contact: <sip:y3a6sn@10.35.195.2>

Authorization: Digest realm="aaa.example.com",nonce="ae9137be",username="bob",
uri="sip:aaa.example.com",response="6iib19cef56c9a0a3i5aieff23a234",algorithm=MD5,opaque=""

Expires: 3600

Content-Length: 0

F4 aaa.example.com → 2221111

SIP/2.0 200 OK

Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK74bfa

From: <sip:user1@bbb.example.com>;tag=1234568

To: <sip:user1@bbb.example.com>;tag=567891

Call-ID: b84c4d76f6@10.35.195.2

CSeq: 2 REGISTER

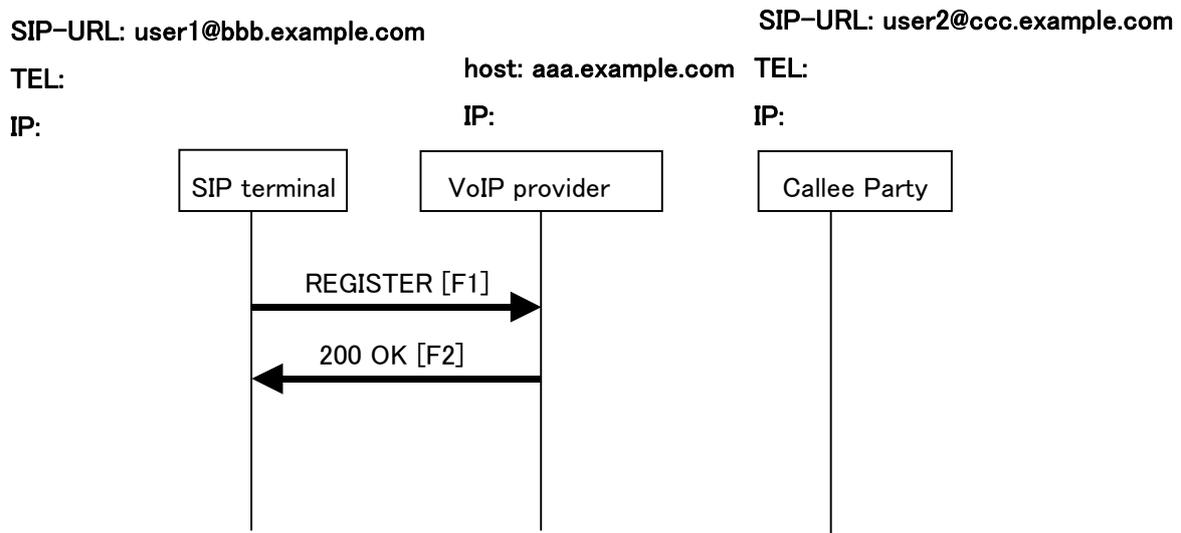
Contact: <sip:y3a6sn@10.35.195.2>;expires=3600

Content-Length: 0

i.2 REGISTER Specific Deletion

Appendix Figure 1-2 shows the sequence for deleting specific contact information previously registered by REGISTER (authentication sequence is omitted).

Registration and server functions in this sequence are incorporated in SIP proxy servers. Message examples follow the sequence.



Appendix Figure 1-2/JJ-90.24: REGISTER Specific Deletion

Message Details:

F1 REGISTER 2221111 → aaa.example.com

```
REGISTER sip:aaa.example.com SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK74bff
Max-Forwards: 70
From: <sip:user1@bbb.example.com>;tag=1234566
To: <sip:user1@bbb.example.com>
Call-ID: b84c4d76f6@10.35.195.2
CSeq: 3 REGISTER
Contact: <sip:y3a6sn@10.35.195.2:5060>
Expires: 0
Content-Length: 0
```

F2 aaa.example.com → 2221111

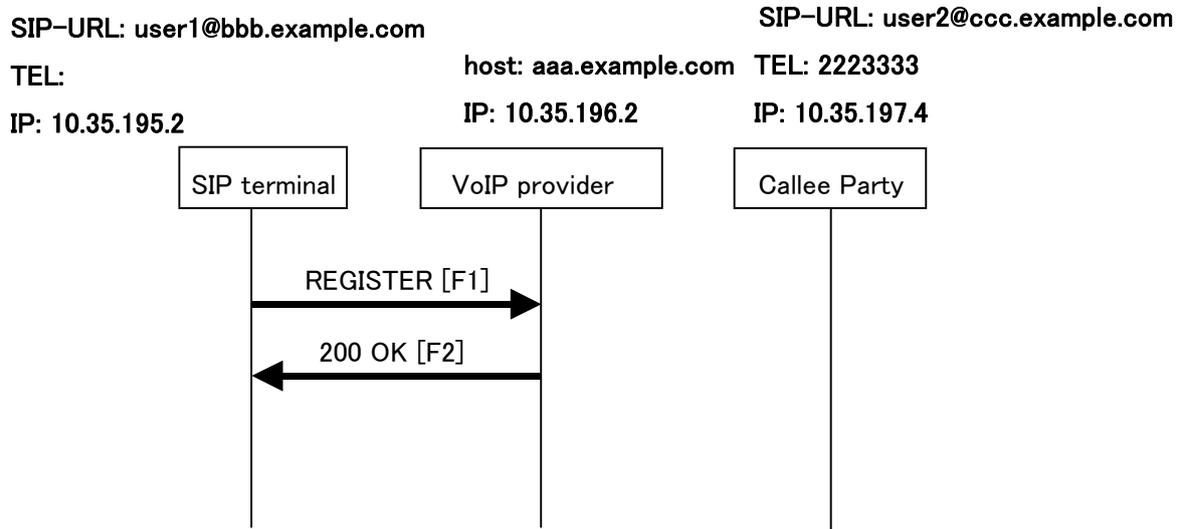
```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK74bff
```

From: <sip:user1@bbb.example.com>;tag=1234566
To: <sip:user1@bbb.example.com>;tag=567898
Call-ID: b84c4d76f6@10.35.195.2
CSeq: 3 REGISTER
Contact: <sip:y3a6sn@10.35.195.2:5060>;expires=0
Content-Length: 0

i.3 REGISTER Total Deletion

Appendix Figure 1-3 shows the sequence for deleting all contact information previously registered by REGISTER (authentication sequence is omitted).

Registration and server functions in this sequence are incorporated in SIP proxy servers. Message examples follow the sequence.



Appendix Figure 1-3/JJ-90.24: REGISTER Total Deletion

Message Details:

F1 REGISTER 2221111 → aaa.example.com

REGISTER sip:aaa.example.com SIP/2.0

Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK74bfg

Max-Forwards: 70

From: <sip:user1@bbb.example.com>;tag=123456d

To: <sip:user1@bbb.example.com>

Call-ID: b84c4d76f6@10.35.195.2

CSeq: 4 REGISTER

Contact: *

Expires: 0

Content-Length: 0

F2 aaa.example.com → 2221111

SIP/2.0 200 OK

Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK74bfg

From: <sip:user1@bbb.example.com>;tag=123456d

To: <sip:user1@bbb.example.com>;tag=56789g

Call-ID: b84c4d76f6@10.35.195.2

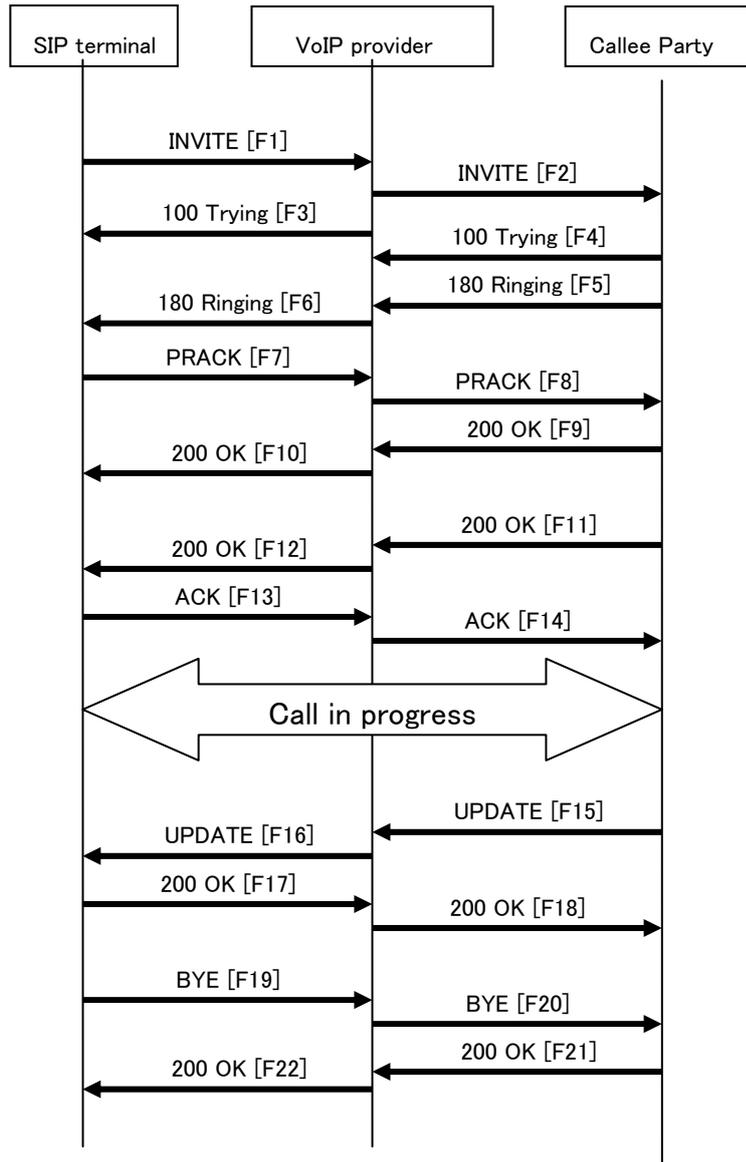
CSeq: 4 REGISTER

Content-Length: 0

i.4 Connection Sequence 1: (call/receive 100rel used, call/receive timer used, call/receive UPDATE used)

Appendix Figure 1-4 shows the call establishment sequence when using call/receive 100rel, call/receive timer, and call/receive UPDATE (authentication sequence is omitted). Message examples follow the sequence.

<p>SIP-URL: user1@bbb.example.com TEL: 2221111 IP: 10.35.195.2</p>	<p>host: aaa.example.com IP: 10.35.196.2</p>	<p>SIP-URL: user2@ccc.example.com TEL: 2223333 IP: 10.35.197.4</p>
--	---	--



Appendix Figure 1-4/JJ-90.24: Connection Sequence 1

Message Details:

F1 INVITE 2221111 → aaa.example.com

INVITE sip:2223333@aaa.example.com SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Max-Forwards: 70
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:g1k7j6n@10.35.195.2>
P-Preferrd-Identity: <sip:user1@bbb.example.com>
Privacy: none
Supported: 100rel,timer
Session-Expires: 180
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F2 INVITE aaa.example.com → 2223333

INVITE sip:z3b6tm@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 69
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:g1k7j6n@10.35.195.2>
Privacy: none
Supported: 100rel,timer
Session-Expires: 180
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F3 100 Trying aaa.example.com → 2221111

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Content-Length: 0

F4 100 Trying 2223333 → aaa.example.com

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Content-Length: 0

F5 180 Ringing 2223333 → aaa.example.com

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>

Supported: 100rel,timer
Require: 100rel
RSeq: 5
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content Length:0

F6 180 Ringing aaa.example.com → 2221111

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Require: 100rel
RSeq: 5
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content Length: 0

F7 PRACK 2221111 → aaa.example.com

PRACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
Max-Forwards: 70
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
RAck: 5 1 INVITE
Content-Length: 0

F8 PRACK aaa.example.com → 2223333

PRACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na80e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
Max-Forwards: 69
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf

To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
RAck: 5 1 INVITE
Content-Length: 0

F9 200 OK 2223333 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na80e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
Content-Length: 0

F10 200 OK aaa.example.com → 2221111

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
Content-Length: 0

F11 200 OK 2223333 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Require: timer
Session-Expires: 180;refresher=uas
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp

Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F12 200 OK aaa.example.com → 2221111

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Require: timer
Session-Expires: 180;refresher=uas
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F13 ACK 2221111 → aaa.example.com

ACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g2
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>

Max-Forwards: 70
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F14 ACK aaa.example.com → 2223333

ACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK8374921
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g2
Max-Forwards: 69
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F15 UPDATE 2223333 → aaa.example.com

UPDATE sip:g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:2223333@aaa.example.com>;tag=314159
To: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 UPDATE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported:timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Length: 0

F16 UPDATE aaa.example.com → 2221111

UPDATE sip:g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497bs
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
Max-Forwards: 69
From: <sip:2223333@aaa.example.com>;tag=314159

To: <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 UPDATE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported:timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Length: 0

F17 200 OK 2221111 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497bs
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 UPDATE
Supported:timer
Require:timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Contact: < sip:g1k7j6n@10.35.195.2 >
Content-Length: 0

F18 200 OK aaa.example.com → 2223333

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
From: <sip:2223333@aaa.example.com>;tag=314159
To: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 UPDATE
Supported:timer
Require:timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Contact: < sip:g1k7j6n@10.35.195.2 >
Content-Length: 0

F19 BYE 2221111 → aaa.example.com

BYE sip:7in9s58t1@10.35.197.4:5060 SIP/2.0

Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g7
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 3 BYE
Content-Length: 0

F20 BYE aaa.example.com → 2223333

BYE sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK83749715
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g7
Max-Forwards: 69
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 3 BYE
Content-Length: 0

F21 200 OK 2223333 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK83749715
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g7
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 3 BYE
Content-Length: 0

F22 200 OK aaa.example.com → 2221111

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g7
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 3 BYE
Content-Length: 0

INVITE sip:2223333@aaa.example.com SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Max-Forwards: 70
From: < sip:user1@bbb.example.com>;tag=a6c85cf
To: < sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: < sip:g1k7j6n@10.35.195.2>
Supported: timer
Session-Expires: 180
Allow: ACK,BYE,CANCEL,INVITE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F2 INVITE aaa.example.com → 2223333

INVITE sip:z3b6tm@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: < sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 69
From: < sip:user1@bbb.example.com >;tag=a6c85cf
To: < sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: < sip:g1k7j6n@10.35.195.2>

Supported: timer
Session-Expires: 180
Allow: ACK,BYE,CANCEL,INVITE
Content-Type: application/sdp
Content-Length: 124

v=0

o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F3 100 Trying aaa.example.com → 2221111

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Content-Length: 0

F4 100 Trying 2223333 → aaa.example.com

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Content-Length: 0

F5 180 Ringing 2223333 → aaa.example.com

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Allow: ACK,BYE,CANCEL,INVITE
Content Length:0

F6 180 Ringing aaa.example.com → 2221111

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Allow: ACK,BYE,CANCEL,INVITE
Content Length: 0

F7 200 OK 2223333 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: timer
Require: timer
Session-Expires: 180;refresher=uas
Allow: ACK,BYE,CANCEL,INVITE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F8 200 OK aaa.example.com → 2221111

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: timer
Require: timer
Session-Expires: 180;refresher=uas
Allow: ACK,BYE,CANCEL,INVITE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F9 ACK 2221111 → aaa.example.com

ACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g2
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F10 ACK aaa.example.com → 2223333

ACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK8374921
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g2
Max-Forwards: 69

From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F11 re-INVITE 22223333 → aaa.example.com

INVITE sip: g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported:timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F12 re-INVITE aaa.example.com → 22221111

INVITE sip: g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497bs
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
Max-Forwards: 69
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>

Supported:timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F13 200 OK 2221111 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497bs
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: < sip:g1k7j6n@10.35.195.2>
Supported: timer
Require: timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F14 200 OK aaa.example.com → 2223333

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: < sip:g1k7j6n@10.35.195.2>
Supported: timer
Require: timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F15 ACK 22223333 → aaa.example.com

ACK sip: g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggw
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F16 ACK aaa.example.com → 22221111

ACK sip: g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497bt
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggw
Max-Forwards: 69
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com>;tag=a6c85cf

Call-ID: a84b4c76e6@10.35.195.2

CSeq: 1 ACK

Content-Length: 0

F17 BYE 2223333 → aaa.example.com

BYE sip: g1k7j6n@10.35.195.2 SIP/2.0

Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77gg

Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>

Max-Forwards: 70

From: <sip:2223333@aaa.example.com>;tag=314159

To: <sip:user1@bbb.example.com>;tag=a6c85cf

Call-ID: a84b4c76e6@10.35.195.2

CSeq: 2 BYE

Content-Length: 0

F18 BYE aaa.example.com → 2221111

BYE sip:g1k7j6n@10.35.195.2 SIP/2.0

Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497b

Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77gg

Max-Forwards: 69

From: <sip:2223333@aaa.example.com>;tag=314159

To: <sip:user1@bbb.example.com>;tag=a6c85cf

Call-ID: a84b4c76e6@10.35.195.2

CSeq: 2 BYE

Content-Length: 0

F19 200 OK 2221111 → aaa.example.com

SIP/2.0 200 OK

Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497b

Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77gg

From: <sip:2223333@aaa.example.com>;tag=314159

To: <sip:user1@bbb.example.com>;tag=a6c85cf

Call-ID: a84b4c76e6@10.35.195.2

CSeq: 2 BYE

Content-Length: 0

F20 200 OK aaa.example.com → 2223333

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77gg
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 BYE
Content-Length: 0

Message Details:

F1 INVITE 2221111 → aaa.example.com

INVITE sip:1862223333@aaa.example.com SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Max-Forwards: 70
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
P-Preferrd-identity: <sip: user1@bbb.example.com>
Contact: <sip:g1k7j6n@10.35.195.2>
Supported: 100rel,timer
Session-Expires: 180
Allow: ACK,BYE,CANCEL,INVITE,PRACK
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F2 INVITE aaa.example.com → 2223333

INVITE sip:z3b6tm@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 69
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:g1k7j6n@10.35.195.2>
Privacy:none
P-Asserted-Identity: "2221111"<tel:+812221111>, <sip:user1@bbb.example.com>
Supported: 100rel,timer

Session-Expires: 180
Allow: ACK,BYE,CANCEL,INVITE,PRACK
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F3 100 Trying aaa.example.com → 2221111

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Content-Length: 0

F4 100 Trying 2223333 → aaa.example.com

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Content-Length: 0

F5 180 Ringing 2223333 → aaa.example.com

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com>;tag=a6c85cf

To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Require: 100rel
RSeq: 5
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content Length:0

F6 180 Ringing aaa.example.com → 2221111

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Require: 100rel
RSeq: 5
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content Length: 0

F7 PRACK 2221111 → aaa.example.com

PRACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
Max-Forwards: 70
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
RAck: 5 1 INVITE
Content-Length: 0

F8 PRACK aaa.example.com → 2223333

PRACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0

Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na80e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
Max-Forwards: 69
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
RAck: 5 1 INVITE
Content-Length: 0

F9 200 OK 2223333 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na80e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
Content-Length: 0

F10 200 OK aaa.example.com → 2221111

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
Content-Length: 0

F11 200 OK 2223333 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer

Require: timer
Session-Expires: 180;refresher=uas
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F12 200 OK aaa.example.com → 2221111

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Require: timer
Session-Expires: 180;refresher=uas
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F13 ACK 2221111 → aaa.example.com

ACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g2
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F14 ACK aaa.example.com → 2223333

ACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK8374921
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g2
Max-Forwards: 69
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F15 re-INVITE 2223333 → aaa.example.com

INVITE sip: g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:1862223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported:timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0

o=- 0 0 IN IP4 10.35.197.4

s=-

c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F16 re-INVITE aaa.example.com → 22221111

INVITE sip: g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497bs
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
Max-Forwards: 69
From: <sip:1862223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported:timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F17 200 OK 2221111 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497bs
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
From: <sip:1862223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: < sip:g1k7j6n@10.35.195.2>
Supported: timer

Require: timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE,PRACK
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F18 200 OK aaa.example.com → 2223333

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
From: <sip:1862223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: < sip:g1k7j6n@10.35.195.2>
Supported: timer
Require: timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE,PRACK
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F19 ACK 22223333 → aaa.example.com

ACK sip: g1k7j6n@10.35.195.2 SIP/2.0

Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggw
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:1862223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F20 ACK aaa.example.com → 22221111

ACK sip: g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497bt
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggw
Max-Forwards: 69
From: <sip:1862223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F19 BYE 2221111 → aaa.example.com

BYE sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g7
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 3 BYE
Content-Length: 0

F20 BYE aaa.example.com → 2223333

BYE sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK83749715
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g7
Max-Forwards: 69
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2

CSeq: 3 BYE
Content-Length: 0

F21 200 OK 2223333 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK83749715
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g7
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 3 BYE
Content-Length: 0

F22 200 OK aaa.example.com → 2221111

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g7
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 3 BYE
Content-Length: 0

Message Details:

F1 INVITE 2221111 → aaa.example.com

INVITE sip:2223333@aaa.example.com SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Max-Forwards: 70
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:g1k7j6n@10.35.195.2>
P-Preferrd-identity: <sip: user1@bbb.example.com>
Supported: 100rel,timer
Session-Expires: 180
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F2 INVITE aaa.example.com → 2223333

INVITE sip:z3b6tm@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 69
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:g1k7j6n@10.35.195.2>
P-Asserted-Identity: “2221111”<tel:+812221111>, <sip:user1@bbb.example.com >
Supported: 100rel,timer
Session-Expires: 180

Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F3 100 Trying aaa.example.com → 2221111

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Content-Length: 0

F4 100 Trying 2223333 → aaa.example.com

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Content-Length: 0

F5 180 Ringing 2223333 → aaa.example.com

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159

Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Require: 100rel
RSeq: 5
Allow: ACK,BYE,CANCEL,INVITE,PRACK
Content Length:0

F6 180 Ringing aaa.example.com → 2221111

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Require: 100rel
RSeq: 5
Allow: ACK,BYE,CANCEL,INVITE,PRACK
Content Length: 0

F7 PRACK 2221111 → aaa.example.com

PRACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
Max-Forwards: 70
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
RAck: 5 1 INVITE
Content-Length: 0

F8 PRACK aaa.example.com → 2223333

PRACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na80e

Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
Max-Forwards: 69
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
RAck: 5 1 INVITE
Content-Length: 0

F9 200 OK 2223333 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na80e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
Content-Length: 0

F10 200 OK aaa.example.com → 2221111

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
Content-Length: 0

F11 200 OK 2223333 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Require: timer

Session-Expires: 180;refresher=uas
Allow: ACK,BYE,CANCEL,INVITE,PRACK
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F12 200 OK aaa.example.com → 2221111

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Require: timer
Session-Expires: 180;refresher=uas
Allow: ACK,BYE,CANCEL,INVITE,PRACK
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F13 ACK 2221111 → aaa.example.com

ACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g2
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F14 ACK aaa.example.com → 2223333

ACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK8374921
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g2
Max-Forwards: 69
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F15 re-INVITE 2223333 → aaa.example.com

INVITE sip: g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com >;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported:timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE,PRACK
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4

t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F16 re-INVITE aaa.example.com → 22221111

INVITE sip: g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497bs
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
Max-Forwards: 69
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com >;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported:timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE,PRACK
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F17 200 OK 2221111 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497bs
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com >;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: < sip:g1k7j6n@10.35.195.2>
Supported: timer
Require: timer

Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F18 200 OK aaa.example.com → 2223333

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com >;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: < sip:g1k7j6n@10.35.195.2>
Supported: timer
Require: timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F19 ACK 2223333 → aaa.example.com

ACK sip: g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggw

Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com >;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F20 ACK aaa.example.com → 22221111

ACK sip: g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497bt
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggw
Max-Forwards: 69
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com >;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F21 BYE 2223333 → aaa.example.com

BYE sip: g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77gg
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com >;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 BYE
Content-Length: 0

F22 BYE aaa.example.com → 2221111

BYE sip:g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497b
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77gg
Max-Forwards: 69
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com >;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2

CSeq: 2 BYE
Content-Length: 0

F23 200 OK 2221111 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497b
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77gg
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com >;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 BYE
Content-Length: 0

F24 200 OK aaa.example.com → 2223333

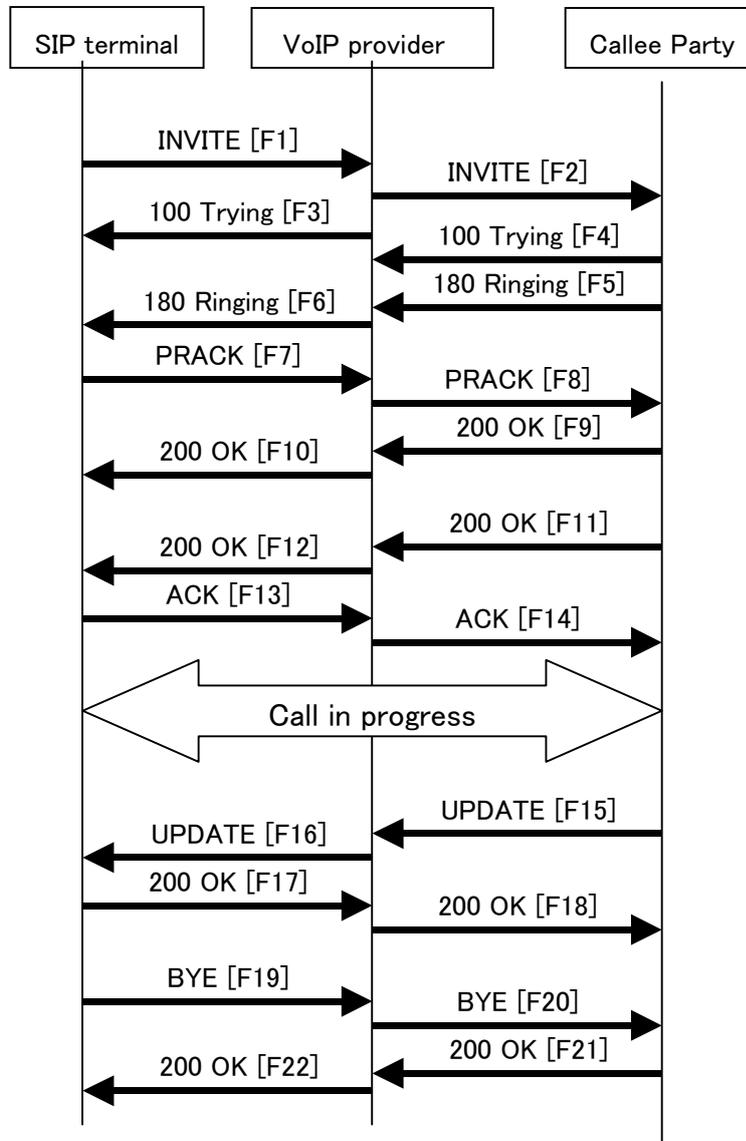
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77gg
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com >;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 BYE
Content-Length: 0

i.8 Connection Sequence 5: (call/receive 100rel used, call timer not used, receive timer used)

Appendix Figure 1-8 shows an example of a session update sequence for the receiving terminal given that the Session-Timer function is equipped only in the receiving terminal and the network. Message examples follow the sequence.

Note that, while the session refresh request is given by the UPDATE method in this example, it could also be given by the INVITE method through an exchange based on the Allow header. See sections 1.5, 1.8, and 1.9.

SIP-URL: user1@bbb.example.com TEL: 2221111 IP: 10.35.195.2	host: aaa.example.com IP: 10.35.196.2	SIP-URL: user2@ccc.example.com TEL: 2223333 IP: 10.35.197.4
--	--	--



Appendix Figure 1-8/JJ-90.24: Connection Sequence 5

Message Details:

F1 INVITE 2221111 → aaa.example.com

INVITE sip:1862223333@aaa.example.com SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Max-Forwards: 70
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:g1k7j6n@10.35.195.2>
Supported: 100rel
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F2 INVITE aaa.example.com → 2223333

* Session-Expires header is generated by the network (but refresher parameter is not specified)

INVITE sip:z3b6tm@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 69
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:g1k7j6n@10.35.195.2>
Supported: 100rel
Session-Expires: 180
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE

Content-Type: application/sdp

Content-Length: 124

v=0

o=- 0 0 IN IP4 10.35.195.2

s=-

c=IN IP4 10.35.195.2

t=0 0

m=audio 5004 RTP/AVP 0

a=rtpmap:0 PCMU/8000

a=ptime:20

F3 100 Trying aaa.example.com → 2221111

SIP/2.0 100 Trying

Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g

From: <sip:user1@bbb.example.com>;tag=a6c85cf

To: <sip:1862223333@aaa.example.com>

Call-ID: a84b4c76e6@10.35.195.2

CSeq: 1 INVITE

Content-Length: 0

F4 100 Trying 2223333 → aaa.example.com

SIP/2.0 100 Trying

Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e

Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g

From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf

To: <sip:1862223333@aaa.example.com>

Call-ID: a84b4c76e6@10.35.195.2

CSeq: 1 INVITE

Content-Length: 0

F5 180 Ringing 2223333 → aaa.example.com

SIP/2.0 180 Ringing

Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e

Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g

Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>

From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf

To: <sip:1862223333@aaa.example.com>;tag=314159

Call-ID: a84b4c76e6@10.35.195.2

CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Require: 100rel
RSeq: 5
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content Length:0

F6 180 Ringing aaa.example.com → 2221111

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Require: 100rel
RSeq: 5
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content Length: 0

F7 PRACK 2221111 → aaa.example.com

PRACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
Max-Forwards: 70
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
RAck: 5 1 INVITE
Content-Length: 0

F8 PRACK aaa.example.com → 2223333

PRACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na80e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g

Max-Forwards: 69
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
RAck: 5 1 INVITE
Content-Length: 0

F9 200 OK 2223333 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na80e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
Content-Length: 0

F10 200 OK aaa.example.com → 2221111

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
Content-Length: 0

F11 200 OK 2223333 → aaa.example.com

* In the event that Initial-INVITE is only specified in the Session-Expires header without the "timer" option-tag being included in the Supported header, the Session Timer function is activated as a refresher on one's own. In this case, "timer" cannot be set in the Require header.

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>

Supported: 100rel,timer
Session-Expires: 180;refresher=uas
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F12 200 OK aaa.example.com → 2221111

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Session-Expires: 180;refresher=uas
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F13 ACK 2221111 → aaa.example.com

ACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g2
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F14 ACK aaa.example.com → 2223333

ACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK8374921
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g2
Max-Forwards: 69
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F15 UPDATE 2223333 → aaa.example.com

UPDATE sip: g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:1862223333@aaa.example.com>;tag=314159
To: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 UPDATE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported:timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Length: 0

F16 UPDATE aaa.example.com → 2221111

UPDATE sip: g1k7j6n@10.35.195.2 SIP/2.0

Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497bs
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
Max-Forwards: 69
From: <sip:1862223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 UPDATE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported:timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Length: 0

F17 200 OK 2221111 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497bs
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
From: <sip:1862223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 UPDATE
Content-Length: 0

F18 200 OK aaa.example.com → 2223333

* Since the caller terminal does not support the Session Timer function, the Session-Expires header would not be attached. If the network, however, implements Session Timer, the network may add “timer” to the Require header and set the Session-Expires header as refresher=uac.

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77ggu
From: <sip:1862223333@aaa.example.com>;tag=314159
To: “2221111” <sip:user1@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 UPDATE
Require:timer
Session-Expires: 180;refresher=uac
Content-Length: 0

F19 BYE 2221111 → aaa.example.com

BYE sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g7

Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 3 BYE
Content-Length: 0

F20 BYE aaa.example.com → 2223333

BYE sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK83749715
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g7
Max-Forwards: 69
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 3 BYE
Content-Length: 0

F21 200 OK 2223333 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK83749715
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g7
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 3 BYE
Content-Length: 0

F22 200 OK aaa.example.com → 2221111

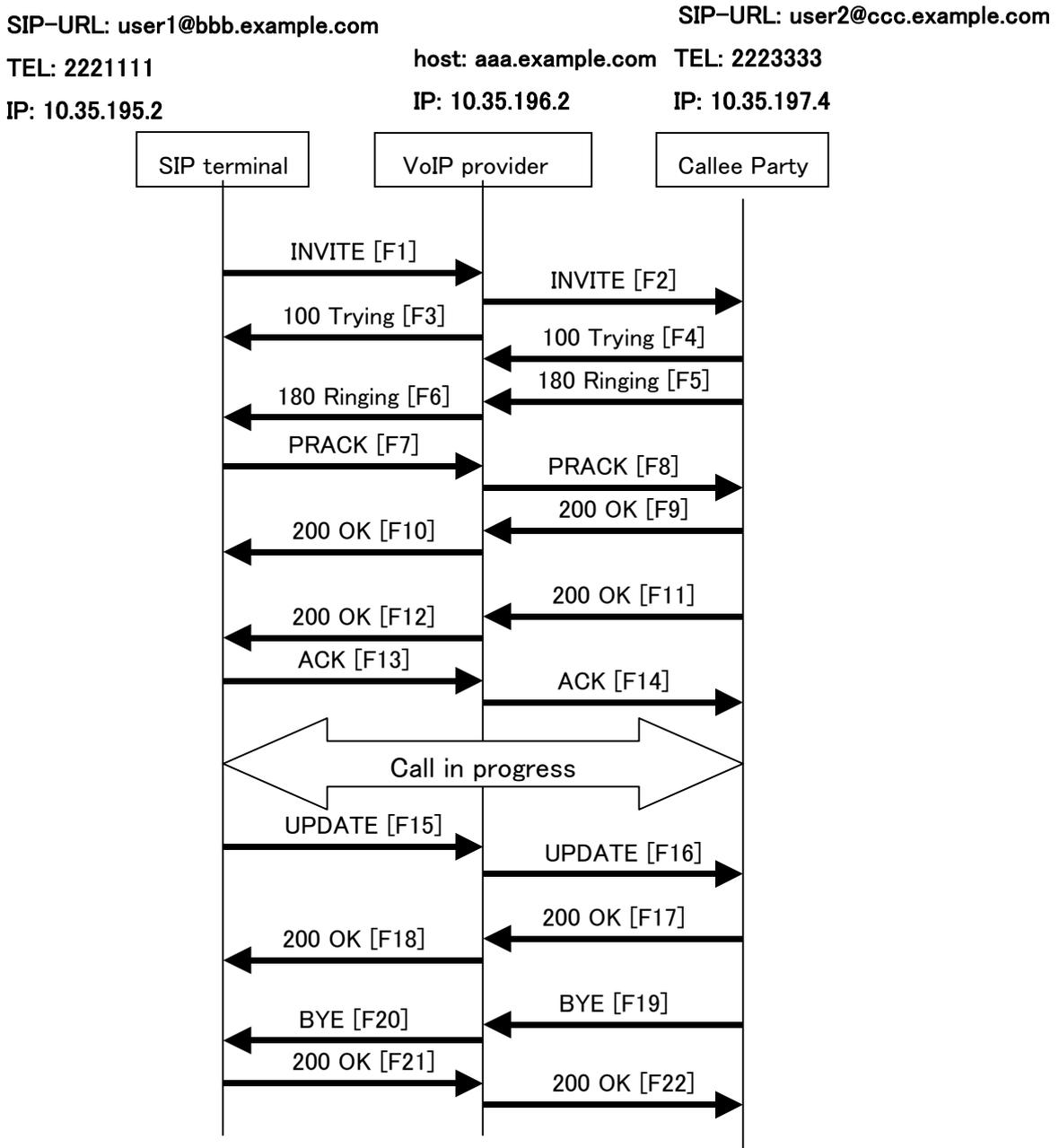
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g7
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 3 BYE

Content-Length: 0

i.9 Connection Sequence 6: (call/receive 100rel used, call timer used, receive timer not used)

Appendix Figure 1-9 shows an example of a session update sequence for the receiving terminal given that the Session-Timer function is equipped only in the caller terminal and the network. Message examples follow the sequence.

Note that, while the session refresh request is given by the UPDATE method in this example, it could also be given by the INVITE method through an exchange based on the Allow header. See sections 1.5, 1.8, and 1.9.



Appendix Figure 1-9/JJ-90.24: Connection Sequence 6

Message Details:

F1 INVITE 2221111 → aaa.example.com

INVITE sip:2223333@aaa.example.com SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Max-Forwards: 70
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:g1k7j6n@10.35.195.2>
Supported: 100rel,timer
Session-Expires: 180
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F2 INVITE aaa.example.com → 2223333

INVITE sip:z3b6tm@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 69
From: "2221111"<sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:g1k7j6n@10.35.195.2>
Supported: 100rel,timer
Session-Expires: 180
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE

Content-Type: application/sdp

Content-Length: 124

v=0

o=- 0 0 IN IP4 10.35.195.2

s=-

c=IN IP4 10.35.195.2

t=0 0

m=audio 5004 RTP/AVP 0

a=rtpmap:0 PCMU/8000

a=ptime:20

F3 100 Trying aaa.example.com → 2221111

SIP/2.0 100 Trying

Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g

From: <sip:user1@bbb.example.com >;tag=a6c85cf

To: <sip:2223333@aaa.example.com>

Call-ID: a84b4c76e6@10.35.195.2

CSeq: 1 INVITE

Content-Length: 0

F4 100 Trying 2223333 → aaa.example.com

SIP/2.0 100 Trying

Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e

Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g

From: "2221111"<sip:user1@bbb.example.com >;tag=a6c85cf

To: <sip:2223333@aaa.example.com>

Call-ID: a84b4c76e6@10.35.195.2

CSeq: 1 INVITE

Content-Length: 0

F5 180 Ringing 2223333 → aaa.example.com

SIP/2.0 180 Ringing

Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e

Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g

Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>

From: "2221111"<sip:user1@bbb.example.com >;tag=a6c85cf

To: <sip:2223333@aaa.example.com>;tag=314159

Call-ID: a84b4c76e6@10.35.195.2

CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel
Require: 100rel
RSeq: 5
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content Length:0

F6 180 Ringing aaa.example.com → 2221111

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel
Require: 100rel
RSeq: 5
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content Length: 0

F7 PRACK 2221111 → aaa.example.com

PRACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
Max-Forwards: 70
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
RAck: 5 1 INVITE
Content-Length: 0

F8 PRACK aaa.example.com → 2223333

PRACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na80e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g

Max-Forwards: 69
From: "2221111"<sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
RAck: 5 1 INVITE
Content-Length: 0

F9 200 OK 2223333 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na80e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
From: "2221111"<sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
Content-Length: 0

F10 200 OK aaa.example.com → 2221111

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
Content-Length: 0

F11 200 OK 2223333 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: "2221111"<sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp

Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F12 200 OK aaa.example.com → 2221111

* Since the receiving terminal does not support the Session Timer function, “timer” would not be specified in the Session-Expires header and Require header. If the network, however, implements Session Timer, the network may add “timer” to the Require header and set the Session-Expires header as refresher=uac.

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F13 ACK 2221111 → aaa.example.com

ACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0

Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g2
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F14 ACK aaa.example.com → 2223333

ACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK8374921
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g2
Max-Forwards: 69
From: "2221111"<sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F15 UPDATE 2221111 → aaa.example.com

UPDATE sip: 7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g3
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 3 UPDATE
Contact: <sip:g1k7j6n@10.35.195.2>
Supported:timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Length: 0

F16 UPDATE aaa.example.com → 2223333

UPDATE sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497bs

Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g3
Max-Forwards: 69
From: "2221111"<sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 3 UPDATE
Contact: <sip:g1k7j6n@10.35.195.2>
Supported:timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Length: 0

F17 200 OK 2221111 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497bs
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g3
From: "2221111"<sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 3 UPDATE
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Contact: < sip:7in9s58t1@10.35.197.4:5060 >
Content-Length: 0

F18 200 OK aaa.example.com → 2223333

* Since the receiving terminal does not support the Session Timer function, "timer" would not be specified in the Session-Expires header and Require header. If the network, however, implements Session Timer, the network may add "timer" to the Require header and set the Session-Expires header as refresher=uac.

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497bs
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g3
From: <sip:user1@bbb.example.com >;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 3 UPDATE
Require:timer
Session-Expires: 180;refresher=uac
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Contact: < sip:7in9s58t1@10.35.197.4:5060 >
Content-Length: 0

F21 BYE 2223333 → aaa.example.com

BYE sip: g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77gg
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:2223333@aaa.example.com>;tag=314159
To: "2221111"<sip:user1@bbb.example.com >;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 BYE
Content-Length: 0

F22 BYE aaa.example.com → 2221111

BYE sip:g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497b
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77gg
Max-Forwards: 69
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com >;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 BYE
Content-Length: 0

F23 200 OK 2221111 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497b
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77gg
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:user1@bbb.example.com >;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 BYE
Content-Length: 0

F24 200 OK aaa.example.com → 2223333

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77gg
From: <sip:2223333@aaa.example.com>;tag=314159
To: "2221111"<sip:user1@bbb.example.com >;tag=a6c85cf

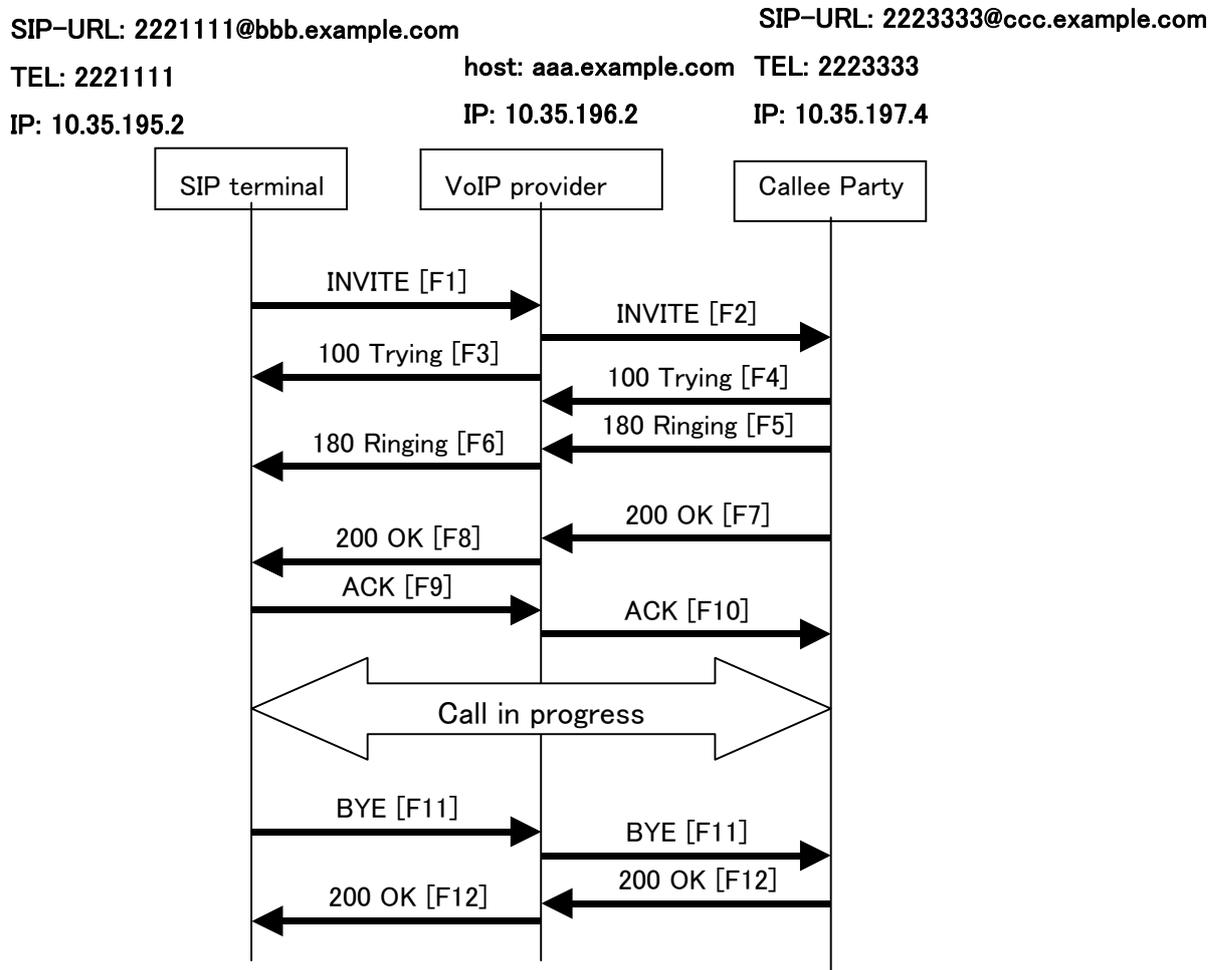
Call-ID: a84b4c76e6@10.35.195.2

CSeq: 1 BYE

Content-Length: 0

i.10 Connection Sequence 7: (call 100rel not used)

Appendix Figure 1-10 shows an example of a call establishment sequence when the SIP terminal on the caller does not support JF-IETF-RFC3262[2] (100rel optional function). The authentication sequence is omitted here. Message examples follow the sequence.



Appendix Figure 1-10/JJ-90.24: Connection Sequence 7

Message Details:

F1 INVITE 2221111 → aaa.example.com

```
INVITE sip:1862223333@aaa.example.com SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Max-Forwards: 70
From: <sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
```

Contact: <sip:g1k7j6n@10.35.195.2>
Supported: timer
Session-Expires: 180
Allow: ACK,BYE,CANCEL,INVITE,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F2 INVITE aaa.example.com → 2223333

INVITE sip:z3b6tm@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 69
From: <sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:1862223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:g1k7j6n@10.35.195.2>
Supported: timer
Session-Expires: 180
Allow: ACK,BYE,CANCEL,INVITE,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F3 100 Trying aaa.example.com → 2221111

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
From: <sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Content-Length: 0

F4 100 Trying 2223333 → aaa.example.com

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
From: <sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Content-Length: 0

F5 180 Ringing 2223333 → aaa.example.com

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported:100rel,timer
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content Length:0

F6 180 Ringing aaa.example.com → 2221111

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:2221111@bbb.example.com>;tag=a6c85cf

To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content Length: 0

F7 200 OK 2223333 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Require: timer
Session-Expires: 180;refresher=uas
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F8 200 OK aaa.example.com → 2221111

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2

CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Require: timer
Session-Expires: 180;refresher=uas
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F9 ACK 2221111 → aaa.example.com

ACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g2
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F10 ACK aaa.example.com → 2223333

ACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK8374921
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g2
Max-Forwards: 69
From: <sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F19 BYE 2221111 → aaa.example.com

BYE sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g7
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 BYE
Content-Length: 0

F20 BYE aaa.example.com → 2223333

BYE sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK83749715
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g7
Max-Forwards: 69
From: <sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 BYE
Content-Length: 0

F21 200 OK 2223333 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK83749715
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g7
From: <sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 BYE
Content-Length: 0

F22 200 OK aaa.example.com → 2221111

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g7
From: <sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2

CSeq: 2 BYE
Content-Length: 0

Contact: <sip:g1k7j6n@10.35.195.2>
Supported: 100rel,timer
Session-Expires: 180
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F2 INVITE aaa.example.com → 2223333

INVITE sip:z3b6tm@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 69
From: "2221111"<sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:g1k7j6n@10.35.195.2>
Supported: 100rel,timer
Session-Expires: 180
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F3 100 Trying aaa.example.com → 2221111

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
From: <sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Content-Length: 0

F4 100 Trying 2223333 → aaa.example.com

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
From: "2221111"<sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Content-Length: 0

F5 180 Ringing 2223333 → aaa.example.com

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: "2221111"<sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: timer
Allow: ACK,BYE,CANCEL,INVITE,UPDATE
Content Length:0

F6 180 Ringing aaa.example.com → 2221111

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:2221111@bbb.example.com>;tag=a6c85cf

To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: timer
Allow: ACK,BYE,CANCEL,INVITE,UPDATE
Content Length: 0

F7 200 OK 2223333 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: "2221111"<sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: timer
Require: timer
Session-Expires: 180;refresher=uas
Allow: ACK,BYE,CANCEL,INVITE,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F8 200 OK aaa.example.com → 2221111

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2

CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: timer
Require: timer
Session-Expires: 180;refresher=uas
Allow: ACK,BYE,CANCEL,INVITE,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F9 ACK 2221111 → aaa.example.com

ACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g2
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F10 ACK aaa.example.com → 2223333

ACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK8374921
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g2
Max-Forwards: 69
From: "2221111"<sip:2221111@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F11 BYE 2223333 → aaa.example.com

BYE sip:g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77gg
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 70
From: <sip:2223333@aaa.example.com>;tag=314159
To: "2221111" <sip:2221111@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 BYE
Content-Length: 0

F12 BYE aaa.example.com → 2221111

BYE sip:g1k7j6n@10.35.195.2 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497b
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77gg
Max-Forwards: 69
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:2221111@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 BYE
Content-Length: 0

F13 200 OK 2221111 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK837497b
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77gg
From: <sip:2223333@aaa.example.com>;tag=314159
To: <sip:2221111@bbb.example.com>;tag=a6c85cf
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 BYE
Content-Length: 0

F14 200 OK aaa.example.com → 2223333

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.197.4:5060;branch=z9hG4bK4na77gg
From: <sip:2223333@aaa.example.com>;tag=314159
To: "2221111" <sip:2221111@bbb.example.com>;tag=a6c85cf

Call-ID: a84b4c76e6@10.35.195.2

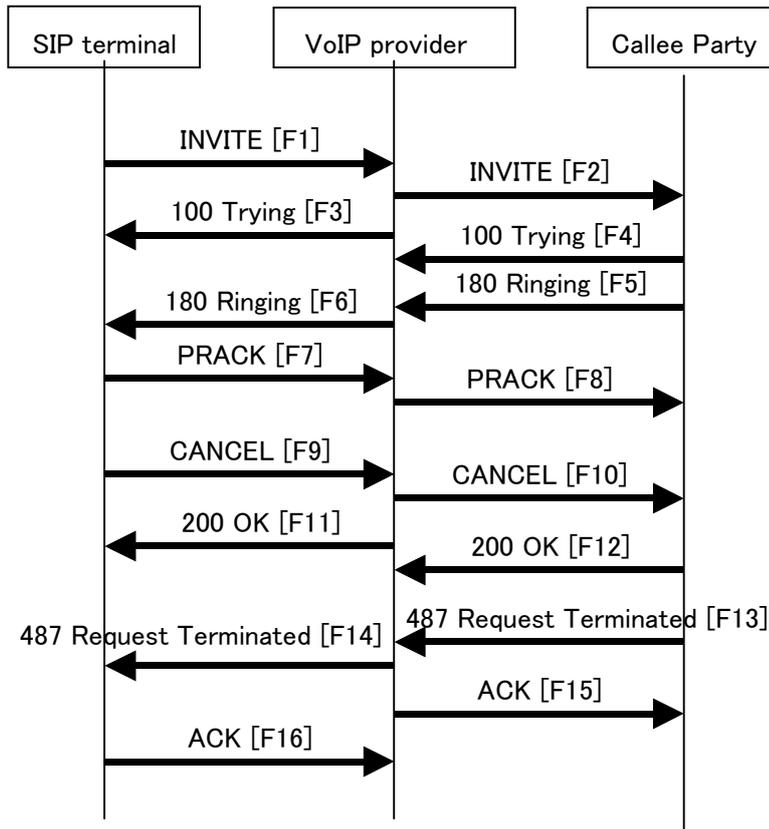
CSeq: 1 BYE

Content-Length: 0

i.12 Call Abortion

Appendix Figure 1-12 shows an example of a disconnect sequence due to an on-hook behavior by the caller terminal during a call. Message examples follow the sequence.

SIP-URL: user1@bbb.example.com	host: aaa.example.com	SIP-URL: user2@ccc.example.com
TEL: 2221111	TEL: 2223333	
IP: 10.35.195.2	IP: 10.35.196.2	IP: 10.35.197.4



Appendix Figure 1-12/JJ-90.24: Call Abandoned

Message Details:

F1 INVITE 2221111 → aaa.example.com

```

INVITE sip:2223333@aaa.example.com SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Max-Forwards: 70
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
    
```

CSeq: 1 INVITE
Contact: <sip:g1k7j6n@10.35.195.2>
Privacy: none
Supported: 100rel,timer
Session-Expires: 180
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20

F2 INVITE aaa.example.com → 2223333

INVITE sip:z3b6tm@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
Max-Forwards: 69
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:g1k7j6n@10.35.195.2>
Privacy: none
Supported: 100rel,timer
Session-Expires: 180
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 124

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 5004 RTP/AVP 0

a=rtpmap:0 PCMU/8000
a=ptime:20

F3 100 Trying aaa.example.com → 2221111

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Content-Length: 0

F4 100 Trying 2223333 → aaa.example.com

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Content-Length: 0

F5 180 Ringing 2223333 → aaa.example.com

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Require: 100rel
RSeq: 5
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content Length:0

F6 180 Ringing aaa.example.com → 2221111

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Record-Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Contact: <sip:7in9s58t1@10.35.197.4:5060>
Supported: 100rel,timer
Require: 100rel
RSeq: 5
Allow: ACK,BYE,CANCEL,INVITE,PRACK,UPDATE
Content Length: 0

F7 PRACK 2221111 → aaa.example.com

PRACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
Max-Forwards: 70
Route: <sip:aaa.example.com:5060;maddr=10.35.196.3;lr>
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
RAck: 5 1 INVITE
Content-Length: 0

F8 PRACK aaa.example.com → 2223333

PRACK sip:7in9s58t1@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na80e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
Max-Forwards: 69
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
RAck: 5 1 INVITE
Content-Length: 0

F9 200 OK 2223333 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na80e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
Content-Length: 0

F10 200 OK aaa.example.com → 2221111

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na80g
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 2 PRACK
Content-Length: 0

F9 CANCEL 2221111 → aaa.example.com

CANCEL sip:2223333@aaa.example.com SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Max-Forwards: 70
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 CANCEL
Content-Length: 0

F10 CANCEL aaa.example.com → 2223333

CANCEL sip:z3b6tm@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Max-Forwards: 69
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 CANCEL
Content-Length: 0

F11 200 OK aaa.example.com → 2221111

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2

CSeq: 1 CANCEL
Content-Length: 0

F12 200 OK 2223333 → aaa.example.com

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 CANCEL
Content-Length: 0

F13 487 Request Terminated 2223333 → aaa.example.com

SIP/2.0 487 Request Terminated
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Content Length:0

F14 487 Request Terminated aaa.example.com → 2221111

SIP/2.0 487 Request Terminated
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 INVITE
Content Length:0

F15 ACK aaa.example.com → 2223333

ACK sip:z3b6tm@10.35.197.4:5060 SIP/2.0
Via: SIP/2.0/UDP 10.35.196.3:5060;branch=z9hG4bK5na77e
Max-Forwards: 70
From: "2221111" <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

F16 ACK 2221111 → aaa.example.com

ACK sip:2223333@aaa.example.com SIP/2.0
Via: SIP/2.0/UDP 10.35.195.2:5060;branch=z9hG4bK4na77g
Max-Forwards: 70
From: <sip:user1@bbb.example.com>;tag=a6c85cf
To: <sip:2223333@aaa.example.com>;tag=314159
Call-ID: a84b4c76e6@10.35.195.2
CSeq: 1 ACK
Content-Length: 0

Appendix ii: Lists of Options

ii.1 Introduction

The following lists of options were compiled for the reader's benefit by extracting the many choices that appear throughout this document. The reader should consult those sections in the specifications portion of the document where these optional items appear (document location) for more detailed information.

Please note that any contention conditions that might exist among these options are not noted in these tables.

Note also that information given in the main document overrides that in these lists of options in the event of any discrepancies.

ii.2 Method of Extracting Options

Options are extracted from the document by identifying choices that must be made in conjunction with the provider's SIP network and then classifying the extracted items into different categories for ease of reference.

ii.3 Format of Option Tables

Appendix Table ii -1 shows and explains the format of the option tables presented here.

Appendix Table ii-1/JJ-90.24: Format example

Item No.	Item (document location)	Options		Remarks
		Network	User	
1	Supported header	Provides timer	Sets timer	
			Does not set timer	
		Provides 100rel	Sets 100rel	
			Does not set 100rel	
		Provides neither	Sets timer, 100rel	
			Does not set timer, 100rel	

Item: An item for which choices must be made for each provider

Network: Conditions provided on the network side

User: Conditions that can be selected by the user.

ii.4 Request Signal

Appendix Table ii-2 lists options in relation to a request signal.

Appendix Table ii-2/JJ-90.24: Request signal

Item No.	Item (document location)	Options		Remarks	
		Network	User		
1	Send re-INVITE at time of media change (10.2.3)	Allows	Uses	Even if network allows it, change request may not be accepted depending on the type of request, connected party, etc.	
			Does not use		
2	Send REGISTER (13.11)	Provides	Uses		
			Does not use		
3	REGISTER update interval (4.1.6)	Does not provide	Does not use		
			Does not specify	Sets any value	
4	Send UPDATE at time of media change (10.2.3)	Allows	Sets specified value	If specified by network, it must be possible to change set value.	
			Does not specify		Sets any value
4	Send UPDATE at time of media change (10.2.3)	Prohibits	Uses		Even if provided by network, use by initial INVITE negotiation may not be possible.
			Does not use		
4	Send UPDATE at time of media change (10.2.3)	Prohibits	Does not use		
			Does not use		

ii.5 Response Signal

Appendix Table ii -3 lists options in relation to a response signal.

Appendix Table ii-3/JJ-90.24: Response signal

Item No.	Item (document location)	Options		Remarks
		Network	User	
1	Forwarding with respect to 3xx response to INVITE (5.4.1)	Sending of 3xx response is possible	Uses Does not use	Forwarding is possible only if confirmation has been made that message is from the network and that there is no security risk.
		No sending of 3xx response is guaranteed	Does not use	

ii.6 SDP

Appendix Table ii -4 lists options in relation to SDP.

Appendix Table ii-4/JJ-90.24: SDP

Item No.	Item (document location)	Options		Remarks
		Network	User	
1	Multiple codec settings in fmt list for "m=" (10.1)	Allows	Makes multiple settings Sets 0 only	Priority order can be changed when making multiple codec settings
		Prohibits	Sets 0 only	
2	ptime setting for "a=" (10.2)	Allows	Sets Does not set	The value may need to be changed at setting time
		Prohibits	Does not set	
3	number of codec which can be set in an offer (10.2)	Specifies	Sets within the specified number	
		Does not specify	Sets any number	
4	Specify priority order for "m=" line in request (10.2)	Specifies	Specifies	
		Does not specify	Specifies Does not specify	
5	Specify priority order for "m=" line in response (10.2)	Specifies	Specifies	
		Does not specify	Specifies Does not specify	
6	dynamic payload type setting for "m=" line in response (10.2)	Allows only value same as offer	Sets value same as offer	
		Allows value different from offer	Sets value same as offer Sets value different from offer	
7	"a=" line settings in response (10.2)	Allows only value same as offer	Sets value same as offer	
		Allows value different from offer	Sets value same as offer Sets value different from offer	

ii.7 Parameters of Various Request Signals

Appendix Tables ii -5 to ii .8 list options in relation to parameters of various request signals.

Appendix Table ii-5/JJ-90.24: REGISTER

Item No.	Item (document location)	Options		Remarks
		Network	User	
1	Request-URI port setting (4.1.2)	Provides default value	Sets Does not set	It must be possible to change value if default value is not provided.
		Provides no default value	Sets Does not set	
2	Setting of Contact q parameter (4.1.3.2)	Allows	Sets Does not set	It must be possible to change value at setting time.
		Prohibits	Does not set	
3	Setting of Contact expires parameter; setting of Expires header (4.1.3.2)	Specifies fixed value	Sets specified value	It must be possible to change value at setting time. Expires must be observed when receiving a 423 error response.
		Specifies no fixed value	Sets any value Does not set	
4	From setting (4.1.3.1)	Allows value different from To	Sets value different from To Sets value same as To	It must be possible to change value if a value different from To is to be set.
		Allows only value same as To	Sets value same as To	

Appendix Table ii-6/JJ-90.24: Initial INVITE

Item No.	Item (document location)	Options		Remarks
		Network	User	
1	Function allowing receive-side terminal to check Request-URI against REGISTER Contact (6.1.2)	Guarantees agreement	Performs check Does not perform check based on agreement	
		Does not guarantee agreement	Does not perform check based on agreement	
2	Request-URI port setting (5.1.2)	Provides default value	Sets Does not set	It must be possible to change value if default value is not provided.
		Provides no default value	Sets	
3	Setting of userinfo in Request-URI in notification or notification restriction of calling number (12.1.1)	Scheme 1	Uses Uses	Note that 184/186 configuration conditions differ depending on network choice.
		Scheme 2	Uses	
		Scheme 3	Uses	
		Scheme 4	Uses	
4	Setting of PRACK in Allow (13.3.5)	Allows setting of PRACK	Sets PRACK Does not set PRACK	
		Prohibits setting of PRACK	Does not set PRACK	
5	Setting of UPDATE in Allow (13.3.5)	Allows setting of UPDATE	Sets UPDATE Does not set UPDATE	
		Prohibits setting of UPDATE	Does not set UPDATE	
6	Setting in Allow of method outside scope of this standard (13.3.5)	Allows setting	Sets method outside scope of this standard Does not set method outside scope of this standard	
		Prohibits setting	Does not set method outside scope of this standard	
7	Setting of userinfo in From (5.1.3)	Provides telephone-subscriber	Sets telephone-subscriber	It must be possible to change value at setting time.
		Provides character string	Sets character string	
8	Setting of userinfo in From in notification or notification restriction of calling number (12.2.1)	Scheme 1	Uses	Note that 184/186 configuration conditions differ depending on network choice.
		Scheme 2	Uses	
		Scheme 3	Uses	
		Scheme 4	Uses	
9	Setting of Min-SE (13.3.5)	Specifies	Sets Does not set	It must be possible to change value at setting time.
		Does not specify	Does not set	
10	Setting of	Scheme 1	Uses	Note that configuration conditions

Item No.	Item (document location)	Options		Remarks
		Network	User	
	P-Preferred-Identity in notification or notification restriction of calling number (12.1.1)			differ depending on network choice.
		Scheme 2	Uses	
		Scheme 3	Uses	
		Scheme 4	Uses	
11	Setting of Privacy in notification or notification restriction of calling number (12.1.1)	Scheme 1	Uses	Note that configuration conditions differ depending on network choice.
		Scheme 2	Uses	
		Scheme 3	Uses	
		Scheme 4	Uses	
12	Setting of Session-Expires (13.3.5)	Allows setting	Sets	It must be possible to change value at setting time.
			Does not set	
		Prohibits setting	Does not set	
13	Setting of timer in Supported (13.3.5)	Allows timer setting	Sets timer	
			Does not set timer	
		Prohibits timer setting	Does not set timer	
14	Setting of 100rel in Supported (13.3.5)	Allows 100rel setting	Sets 100rel	
			Does not set 100rel	
		Prohibits 100rel setting	Does not set 100rel	
15	Setting of Option-tag outside scope of this standard in Supported (13.3.5)	Allows setting	Sets Option-tag outside scope of this standard	
			Does not set Option-tag outside scope of this standard	
		Prohibits setting	Does not set Option-tag outside scope of this standard	
16	Setting of To in notification or notification restriction of calling number (12.1.1)	Scheme 1	Uses	Note that configuration conditions differ depending on network choice.
		Scheme 2	Uses	
		Scheme 3	Uses	
		Scheme 4	Uses	

Appendix Table ii -7/JJ-90.24: re-INVITE

Item No.	Item (document location)	Options		Remarks
		Network	User	
1	Setting of UPDATE in Allow (13.3.7)	Allows setting of UPDATE	Sets UPDATE	
			Does not set UPDATE	
		Prohibits setting of UPDATE	Does not set UPDATE	
2	Setting of Min-SE (13.3.7)	Specifies	Sets	It must be possible to change value at setting time.
			Does not set	
		Does not specify	Does not set	
3	Setting of Session-Expires (13.3.7)	Allows setting	Sets	It must be possible to change value at setting time.
			Does not set	
		Prohibits setting	Does not set	
4	Setting of timer in Supported (13.3.7)	Allows timer setting	Sets timer	
			Does not set timer	
		Prohibits timer setting	Does not set timer	

Appendix Table ii -8/JJ-90.24: UPDATE

Item No.	Item (document location)	Options		Remarks
		Network	User	
1	Setting of Min-SE (13.3.9)	Specifies	Sets	It must be possible to change value at setting time.
			Does not set	
		Does not specify	Does not set	
2	Setting of Session-Expires (13.3.9)	Allows setting	Sets	It must be possible to change value at setting time.
			Does not set	
		Prohibits setting	Does not set	
3	Setting of timer in Supported (13.3.9)	Allows timer setting	Sets timer	
			Does not set timer	
		Prohibits timer setting	Does not set timer	

ii.8 Parameters of Various Response Signals

Appendix Tables ii -9 to ii .11 list options in relation to parameters of various response signals.

Appendix Table ii -9/JJ-90.24: Response settings for an initial INVITE request

Item No.	Item (document location)	Options		Remarks
		Network	User	
1	Setting of PRACK in Allow for 1xx, 2xx responses (13.4.3)	Allows setting of PRACK	Sets PRACK Does not set PRACK	
		Prohibits setting of PRACK	Does not set PRACK	
2	Setting of UPDATE in Allow for 1xx, 2xx responses (13.4.3)	Allows setting of UPDATE	Sets UPDATE Does not set UPDATE	
		Prohibits setting of UPDATE	Does not set UPDATE	
3	Setting in Allow of method outside scope of this standard for 1xx, 2xx responses (13.3.5)	Allows setting	Sets method outside scope of this standard Does not set method outside scope of this standard	
		Prohibits setting	Does not set method outside scope of this standard	
4	Setting of 100rel in Require for 1xx response (13.4.3)	Allows 100rel setting	Sets 100rel Does not set 100rel	Becomes a target of setting when 100rel is set in Supported of received INVITE.
		Prohibits 100rel setting	Does not set 100rel	
5	Setting of timer in Require for 2xx response (13.4.3)	Allows timer setting	Sets timer Does not set timer	Becomes a target of setting when timer is set in Supported of received INVITE.
		Prohibits timer setting	Does not set timer	
6	Setting of RSeq for 1xx response (13.4.3)	Allows setting	Sets Does not set	Becomes a target of setting when 100rel is set in Supported of received INVITE.
		Prohibits setting	Does not set	
7	Setting of Session-Expires for 2xx response (13.4.3)	Allows setting	Sets Does not set	Becomes a target of setting when timer is set in Supported of received INVITE.
		Prohibits setting	Does not set	
8	Setting of timer in Supported for 2xx response (13.4.3)	Allows timer setting	Sets timer Does not set timer	
		Prohibits timer setting	Does not set timer	

Appendix Table ii -10/JJ-90.24: Response settings for a re-INVITE request

Item No.	Item (document location)	Options		Remarks
		Network	User	
1	Setting of UPDATE in Allow for 2xx response (13.4.5)	Allows setting of UPDATE	Sets UPDATE Does not set UPDATE	
		Prohibits setting of UPDATE	Sets UPDATE Does not set UPDATE	
2	Setting of timer in Require for 2xx response (13.4.5)	Allows timer setting	Sets timer Does not set timer	Becomes a target of setting when timer is set in Supported of received INVITE.
		Prohibits timer setting	Does not set timer	
3	Setting of Session-Expires for 2xx response (13.4.5)	Allows setting	Uses Does not use	Becomes a target of setting when timer is set in Supported of received INVITE.
		Prohibits setting	Does not use	
4	Setting of timer in Supported for 2xx response (13.4.5)	Allows timer setting	Sets timer Does not set timer	
		Prohibits timer setting	Does not set timer	

Appendix Table ii -11/JJ-90.24: Response settings for an UPDATE request

Item No.	Item (document location)	Options		Remarks
		Network	User	
1	Setting of timer in	Allows timer setting	Sets timer	Becomes a target of setting when

Item No.	Item (document location)	Options		Remarks
		Network	User	
	Require for 2xx response (13.4.7)		Does not set timer	timer is set in Supported of received INVITE.
		Prohibits timer setting	Does not set timer	
2	Setting of Session-Expires for 2xx response (13.4.7)	Allows setting	Uses	Becomes a target of setting when timer is set in Supported of received INVITE.
			Does not use	
		Prohibits setting	Does not use	
3	Setting of timer in Supported for 2xx response (13.4.7)	Allows timer setting	Sets timer	
			Does not set timer	
		Prohibits timer setting	Does not set timer	

Appendix iii : Background and Role of this Standard

iii.1 Overview

This appendix provides some background to the making of this standard document and describes the criteria used in formulating technical standards.

iii.2 Background to this Standard

At the time of making this standard, there were already several carriers providing SIP-based voice communication services that could be connected to the existing telephone network. Considering this background, the purpose of making this standard was not simply to enable terminals conforming to this standard to interconnect via networks that have the capability of connecting with such terminals. This standard also aimed to formulate a common set of technical standards so that terminals conforming to this standard could make basic voice-call connections with networks and terminals considering not only that connecting networks could already be providing services at the time of standard making but also that callee terminals themselves could already be providing services.

iii.3 Criteria for Formulating Technical Standards

Based on the above background, the following criteria were given particular importance in the decision of technical standards.

iii.3.1 Connected Parties

In principle, this standard has been formulated without violating the stipulations found in RFC documents made by the IETF. At the same time, the technical specifications of this standard have been decided upon so that the terminals targeted by this standard may also connect to parties (networks or terminals) that do not necessarily observe the technical specifications prescribed by IETF, if such parties should currently exist.

iii.3.2 Field Conditions

This standard includes limitations to format in relation to SIP message fields that exceed the scope of RFC specifications. In this regard, it must be kept in mind that some of these limitations result in specifications that call for different sending and receiving conditions. The main reason for this is as follows. As described in Appendix Section iii.2 above, operating specifications for networks and connected parties that connect to terminals that conform to this standard cannot be uniformly specified. This necessitates differences between sending conditions to connected parties and in receiving conditions from connected parties. The idea here is to ensure more reliable connectivity by setting stricter standards for sending conditions so that an actual connected party can accept sent messages and by setting looser standards for receiving conditions so that a message that may be sent from an actual connected party can be accepted.

iii.3.3 Setting Options

The behavior of terminals targeted by this standard is not uniformly determined but is rather based on settings for which a number of options exist. The reason for a standard like this one is that one provider network may require behavior A, for example, to guarantee connections or services, while another provider network may require behavior B, all based on policies dictated by networks or connected terminals. This calls for a provision that enables a terminal to select between settings corresponding to behavior A and behavior B. The actual method of selection is outside the scope of this standard and is not discussed here.

iii.4 Role of Standard

This standard is formulated as a set of technical specifications to enable connection with existing provider networks and a diverse array of audio terminals. It is expected to be used as reference for basic specifications that satisfy the minimum requirements of terminals that aim to use SIP and make voice connections in Japan.

Appendix iv: Setting examples for SDP for wideband codec, etc

iv.1 Overview

This appendix describes setting examples for SDP on typical sending and receiving when SIP terminals with higher audio reproducible codec than G.711 (wideband codec) connects to provider's SIP network via User to Network Interface (interface B). In addition to using wideband codec, setting examples for SDP when using telephone-event to send DTMF is described as examples that a value except G.711 is set in SDP.

iv.2 Setting Examples for SDP

As typical setting examples for SDP when using wideband codec, the following four patterns offer and nine corresponding setting examples for answer are listed as follows.

- 1 Setting examples for SDP when using G.722 [19] as wideband audio codec
- 2 Setting examples for SDP when using G.722.1 [20] as codec using dynamic payload type
- 3 Setting examples for SDP when three kinds of codec (G.722, G.722.1, and G.711) including wideband codec
- 4 Setting examples for SDP when using G.722.1, and also telephone-event to send DTMF

Note that the sequence examples described in this section are considered only as reference material for system implementation and may be changed as needed due to actual service contents and/or terminal functions of provider's SIP network. Note also that the content of these sequence examples do not guarantee connectivity and quality of a communication.

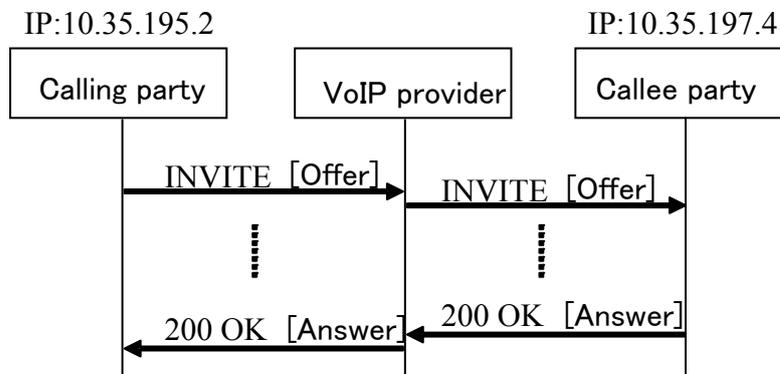
Appendix Table iv-1/JJ-90.24: List of Setting Examples for SDP

No	Offer	Answer	Sequence name
1-①	1 - G.722 - G.711 μ -law	① G.722	Setting example for wideband audio codec (G.722)
1-②		② G.711 μ -law	Setting example for wideband audio codec (G.722) (Select G.711)
2-①	2 - G.722.1 - G.711 μ -law	① G.722.1	Setting example for dynamic payload type
2-②		② G.711 μ -law	Setting example for dynamic payload type (Select G.711)
3-①	3 - G.722.1 - G.722 - G.711 μ -law	① G.722.1	Setting example when offering more than three kinds of codec (Select G.722.1 which is the first priority)
3-②		② G.722	Setting example when offering more than three kinds of codec (Select G.722 which is the second priority)
3-③		③ G.711 μ -law	Setting example when offering more than three kinds of codec (Select G.711)
4-①	4 - G.722.1 - G.711 μ -law - telephone-event	① - G.722.1 - telephone-event	Setting example for SDP when using also telephone-event to send wideband codec and DTMF
4-④		② - G.711 μ -law - telephone-event	Setting example for SDP when using telephone-event to send DTMF

iv.2.1 Sequence Examples for Offer/Answer

1-① Setting example for wideband audio codec (G.722)

A setting example for SDP when using G.722 as wideband voice codec is provided as follows. G.722 is equipped in both originating and terminating terminals. The originating terminal provides G.722 and G.711 as an offer, the terminating terminal selects G.722 as an answer.



[Offer]

A setting example when setting G.722 and G.711 as audio codec and communicating by specifying G.722 as a preferential codec.

```

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 6040 RTP/AVP 9 0
a=rtpmap:9 G722/8000
a=rtpmap:0 PCMU/8000
a=ptime:20
  
```

[Answer]

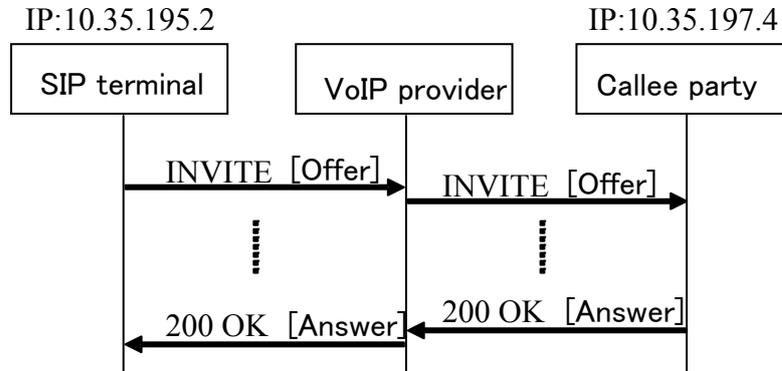
A setting example when selecting G.722 from two codecs set in the offer.

```

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 6008 RTP/AVP 9
a=rtpmap:9 G722/8000
a=ptime:20
  
```

1-② Setting example for wideband audio codec (G.722) (Select G.711)

An originating terminal provides G.722 and G.711 as an offer. G.722 is not equipped in the terminating terminal, or a callee party requires a communication using G.711, and consequently the terminating terminal selects G.711 as an answer.



[Offer]

A setting example when setting G.722 and G.711 as audio codec and communicating by specifying G.722 as a preferential codec.

```

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 6040 RTP/AVP 9 0
a=rtpmap:9 G722/8000
a=rtpmap:0 PCMU/8000
a=ptime:20
  
```

[Answer]

A setting example when selecting G.711 from two codecs set in the offer.

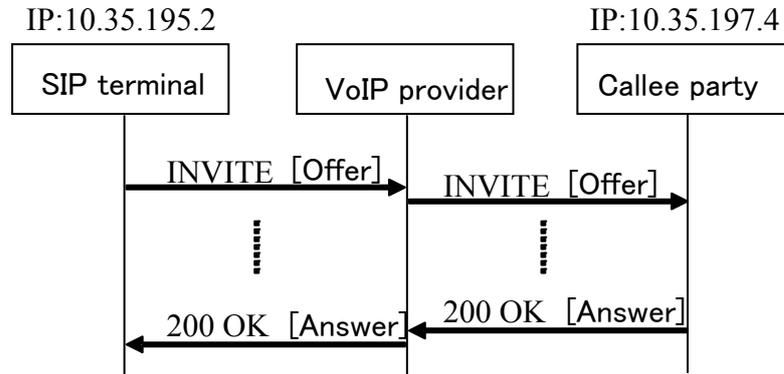
```

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 6008 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20
  
```

2-① Setting example for dynamic payload type

G.722.1 is equipped in both originating and terminating terminals. An originating terminal provides G.722.1 and G.711 as an offer, and consequently the terminating terminal selects G.722.1 as an answer. The originating terminal sets payload type number 102 in the “a=” line because dynamic payload is needed for G.722.1. (Dynamic payload type number 102 is

set in this example. Value from 96 to 127 is available as dynamic payload type.) Regarding the answer, the example describes the case that the “a=” line is not changed from the offer and the same payload type is set in the answer



[Offer]

A setting example when setting G.722.1 and G.711 as audio codec and communicating by specifying G.722.1 as a preferential codec.

```

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 6040 RTP/AVP 102 0
a=rtpmap:102 G7221/16000
a=fmtp:102 bitrate=24000
a=rtpmap:0 PCMU/8000
a=ptime:20
  
```

[Answer]

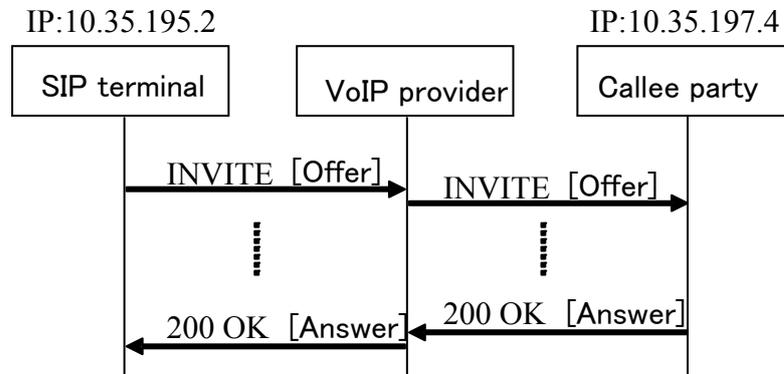
A setting example when selecting G.722.1 from two codecs set in the offer.

```

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 6008 RTP/AVP 102
a=rtpmap:102 G7221/16000
a=fmtp:102 bitrate=24000
a=ptime:20
  
```

2-② Setting example for dynamic payload type (Select G.711)

An originating terminal provides G.722.1 and G.711 as an offer. G.722.1 is not equipped in the terminating terminal, or a callee party requires a communication using G.711, and consequently the terminating terminal selects G.711 as an answer.



[Offer]

A setting example when setting G.722.1 and G.711 as audio codec and communicating by specifying G.722.1 as a preferential codec.

```

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 6040 RTP/AVP 102 0
a=rtpmap:102 G7221/16000
a=fmtp:102 bitrate=24000
a=rtpmap:0 PCMU/8000
a=ptime:20
  
```

[Answer]

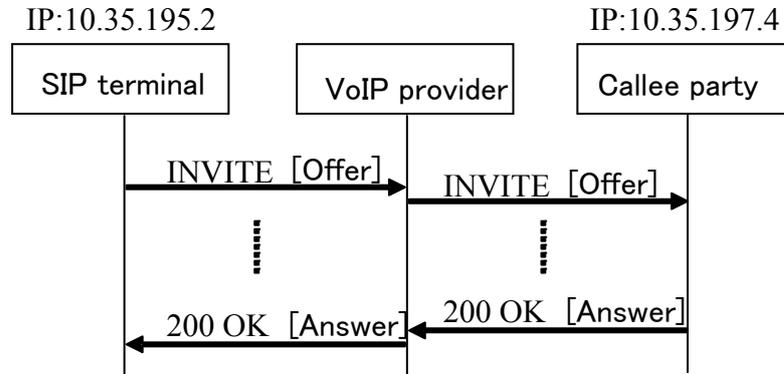
A setting example when selecting G.711 from two codecs set in the offer.

```

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 6008 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20
  
```

3-① Setting example when offering more than three kinds of codec (Select G.722.1 which is the first priority)

G.722.1, G.722, and G.711 are equipped in both originating and terminating terminals. An originating terminal provides an offer in the priority order of G.722.1, G.722, and G.711, and consequently the terminating terminal selects G.722.1 as an answer.



[Offer]

A setting example when setting G.722.1, G.722, and G.711 as audio codec and communicating by specifying the priority in order of G.722.1, G.722, and G.711.

```

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 6040 RTP/AVP 102 9 0
a=rtpmap:102 G7221/16000
a=fmtp:102 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:0 PCMU/8000
a=ptime:20
  
```

[Answer]

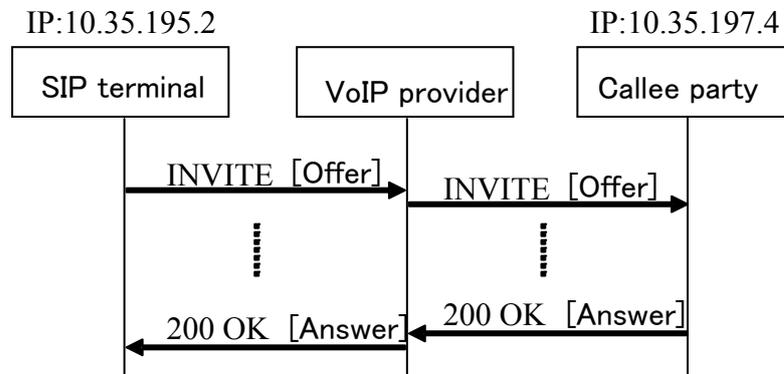
A setting example when selecting G.722.1 from three codecs set in the offer.

```

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 6008 RTP/AVP 102
a=rtpmap:102 G7221/16000
a=fmtp:102 bitrate=24000
a=ptime:20
  
```

3-② Setting example when offering more than three kinds of codec (Select G.722 which is the second priority)

However an originating terminal sets G.722.1 giving the priority, G.722.1 is not equipped in the terminating terminal, or a callee party requires a communication using G.722, and consequently the terminating terminal selects G.722 as an answer.



[Offer]

A setting example when setting G.722.1, G.722, and G.711 as audio codec and communicating by specifying the priority in order of G.722.1, G.722, and G.711.

```

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 6040 RTP/AVP 102 9 0
a=rtpmap:102 G7221/16000
a=fmtp:102 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:0 PCMU/8000
a=ptime:20
  
```

[Answer]

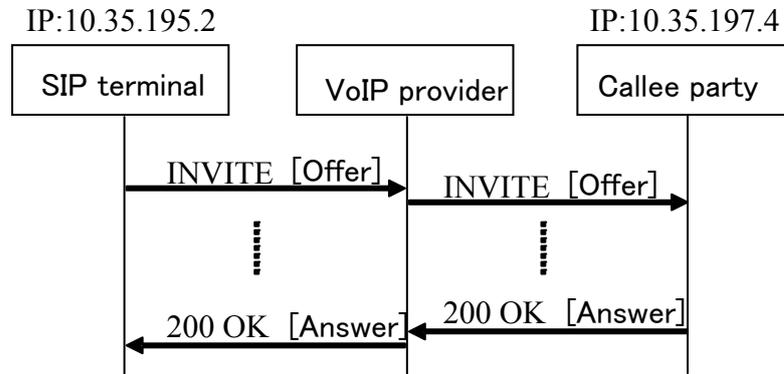
A setting example when selecting G.722 from three codecs set in the offer.

```

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 6008 RTP/AVP 9
a=rtpmap:9 G722/8000
a=ptime:20
  
```

3-③ Setting example when offering more than three kinds of codec (Select G.711)

G.722.1 and G.722 are not equipped in the terminating terminal, or a callee party requires a communication using G.711, and consequently the terminating terminal selects G.711 as an answer.



[Offer]

A setting example when setting G.722.1, G.722, and G.711 as audio codec and communicating by specifying the priority in order of G.722.1, G.722, and G.711.

```

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 6040 RTP/AVP 102 9 0
a=rtpmap:102 G7221/16000
a=fmtp:102 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:0 PCMU/8000
a=ptime:20
  
```

[Answer]

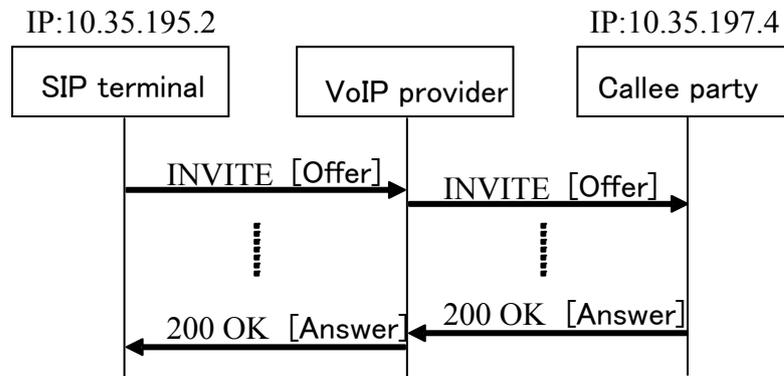
A setting example when selecting G.711 from three codecs set in the offer.

```

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 6008 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20
  
```

4-① Setting example for SDP when using also telephone-event to send wideband codec and DTMF

A setting example for SDP when using wideband codec and also telephone-event is provided as follows. G.722.1 and telephone-event are equipped in both originating and terminating terminals. The terminating terminal selects and sets G.722.1 in an answer as audio codec, and also sets telephone-event. Regarding telephone-event, negotiating in the same “m=” line is recommended because it may cause a confusion on which media stream the telephone-event applies when multiple “m=” lines exist. Regarding parameters set to the “a=” line in an answer, the example describes the case that the same value as in an offer is set.



[Offer]

A setting example communicating by setting G.722.1 and G.711 μ -law, and also telephone-event, as audio codec.

```

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 6040 RTP/AVP 121 0 96
a=rtpmap:121 G7221/16000
a=fmtp:121 bitrate=24000
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
aptime:20
  
```

[Answer]

A setting example when selecting G.722.1 from codecs set in the offer, and also using telephone-event.

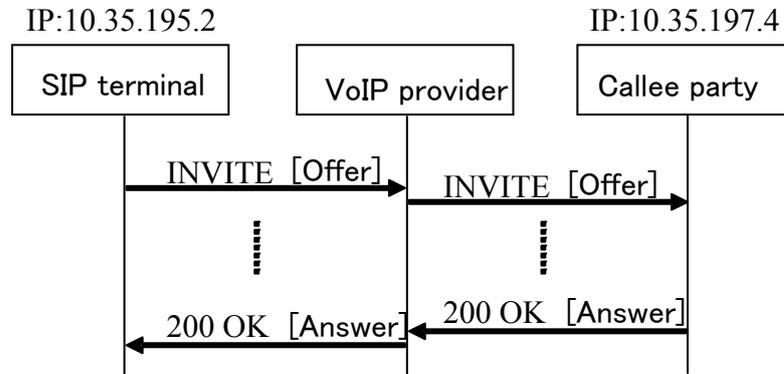
```

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 6008 RTP/AVP 121 96
  
```

a=rtpmap:121 G7221/16000
a=fmtp:121 bitrate=24000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20

4-② Setting example for SDP when using telephone-event to send DTMF

A setting example for SDP when using telephone-event is provided as follows. Telephone-event is equipped in both originating and terminating terminals. The terminating terminal selects and sets G.711 in an answer as audio codec, and sets also telephone-event. Regarding telephone-event, negotiating in the same “m=” line is recommended because it may cause a confusing on which media stream the telephone-event applies when multiple “m=” lines exist. Regarding parameters set to the “a=” line in an answer, the example describes the case that the same value as in an offer is set.



[Offer]

A setting example communicating by setting G.722.1 and G.711 μ -law, and also telephone-event, as audio codec.

```

v=0
o=- 0 0 IN IP4 10.35.195.2
s=-
c=IN IP4 10.35.195.2
t=0 0
m=audio 6040 RTP/AVP 121 0 96
a=rtpmap:121 G7221/16000
a=fmtp:121 bitrate=24000
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
  
```

[Answer]

A setting example when selecting G.711 μ -law from codecs set in the offer, and also using telephone-event.

```

v=0
o=- 0 0 IN IP4 10.35.197.4
s=-
c=IN IP4 10.35.197.4
t=0 0
m=audio 6008 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
  
```

a=rtpmap:96 telephone-event/8000

a=fmtp:96 0-15

a=ptime:20

Technical Specifications on Basic Call Interface for SIP Terminals Connecting with
Provider's SIP Network

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