Information About SIP Compliance with RFC 3261

This appendix describes how the Cisco SIP IP phone complies with the IETF definition of SIP as described in RFC 3261. It has compliance information on the following:

SIP Functions

Function	Supported?
User agent client (UAC)	Yes
User agent server (UAS)	Yes
Proxy server	Third-party only
Redirect server	Third-party only

SIP Methods

Method	Supported?	Comments
INVITE	Yes	The Cisco SIP IP phone supports mid-call changes such as putting a call on hold as signaled by a new INVITE that contains an existing call ID.
ACK	Yes	None.
OPTIONS	Response only	
BYE	Yes	
CANCEL	Yes	
REGISTER	Yes	The Cisco SIP IP phone supports both user and device registration.
REFER	Yes	None.
NOTIFY	Yes	Used for REFER and remote reboot.

SIP Responses

Release 4.0 of the Cisco SIP IP phone supports the following SIP responses:

- 1xx Response—Information Responses, page A-2
- 2xx Response—Successful Responses, page A-3
- 3xx Response—Redirection Responses, page A-3
- 4xx Response—Request Failure Responses, page A-4
- 5xx Response—Server Failure Responses, page A-6
- 6xx Response—Global Responses, page A-7

1*xx* Response—Information Responses

1*xx* Response	Supported?	Comments
100 Trying	Yes	The Cisco SIP IP phone generates this response for an incoming INVITE. Upon receiving this response, the phone waits for a 180 Ringing, 183 Session progress, or 200 OK response.
180 Ringing	Yes	None

1*xx* Response	Supported?	Comments
181 Call Is Being Forwarded	See comments	The Cisco SIP IP phone does not generate these responses,; however, the phone does receive them. The phone processes these responses the same way that it processes the 100 Trying response.
182 Queued		
183 Session Progress		The SIP IP phone does not generate this message. Upon receiving this response, the phone provides early media cut-through and then waits for a 200 OK response.

2*xx* Response—Successful Responses

2*xx* Response	Supported?	Comments
200 OK	Yes	None
202 Accepted	Yes	None

3*xx* Response—Redirection Responses

3*xx* Response	Supported?	Comments
300 Multiple Choices	Yes	None
301 Moved Permanently	Yes	
302 Moved Temporarily	Yes	The Cisco SIP IP phone does not generate this response at this time. Upon receiving this response, the phone sends an INVITE containing the contact information received in the 302 Moved temporarily response.
305 Use Proxy	Yes	The phone does not generate these responses. The gateway contacts the new address in the Contact header field.
380 Alternate Service	Yes	

4*xx* Response—Request Failure Responses

4*xx* Response	Supported?	Comments
400 Bad Request	Yes	The phone generates a 400 Bad Request response for an erroneous request. For an incoming response, the phone initiates a graceful call disconnect (during which the caller hears a busy or fast busy tone) before clearing the call request.
401 Unauthorized	Yes	This response is received only in this release. If a 401 Unauthorized response is received during registration, the phone accepts the response and sends a new request that contains the user's authentication information in the format of the HTTP digest as modified by RFC 3261.
402 Payment Required	Yes	The phone does not generate the 402 Payment Required response.
403 Forbidden	Yes	This response is received only in this release. If the phone receives a 403 Forbidden response, it notifies the user of the response. This response indicates that the SIP server has the request but will not provide service.
404 Not Found	Yes	The Cisco SIP IP phone generates this response if it is unable to locate the callee. Upon receiving this response, the phone notifies the user.
405 Method Not Allowed	See comments	This response is received only in this release. If the phone receives a 405 Method Not Allowed response, it notifies the user of the response.
406 Not Acceptable	See comments	The SIP phone does not generate a 406 Not Acceptable response. For an incoming response, the gateway initiates a graceful call disconnect (during which the caller hears a busy or fast busy tone) before clearing the call request.
407 Proxy Authentication Required	See comments	This response is received only in this release. The 407 Proxy Authentication Required response indicates that the phone must first authenticate itself with the proxy server. If received by the phone, the phone may repeat the INVITE request with a suitable Proxy-Authorization field. This field should contain the authentication information of the user agent for the next outbound proxy or gateway.

4*xx* Response	Supported?	Comments
408 Request Timeout	See comments	The SIP phone does not generate a 408 Request Timeout response. For an incoming response, the gateway initiates a graceful call disconnect (during which the caller hears a busy or fast busy tone) before clearing the call request.
409 Conflict	See comments	This response is received only by the phone in this release. The 409 Conflict response indicates that the INVITE request could not be processed because of a conflict with the current state of the resource. If this response is received, the user is notified.
410 Gone	See comments	This response is received by the phone only in this release. The 410 Gone response indicates that a resource is no longer available at the server and no forwarding address is known.
411 Length Required	See comments	This response is received by the phone only in this release. This response indicates that the user refuses to accept the request without a defined content length. If received, the phone resends the INVITE request if it can add a valid Content-Length header field.
413 Request Entity Too Large	See comments	This response is received only by the phone in this release. If a retry after header field is contained in this response, then the user can attempt the call once again in the retry time provided.
414 Request—URL Too Long	See comments	This response is received only by the phone in this release. The user is notified if this response is received.
415 Unsupported Media	See comments	This response is received only by the phone in this release. The user is notified if this response is received.
420 Bad Extension	See comments	This response is received only by the phone in this release. The user is notified if this response is received. If the phone does not understand the protocol extension specified in the Require field, the 420 Bad Extension response is generated.
480 Temporarily Unavailable	See comments	The phone sends this response if Do Not Disturb (DND) is active on the phone.

4*xx* Response	Supported?	Comments
481 Call Leg/Transaction Does Not Exist	See comments	This response is received only by the phone in this release. The user is notified if this response is received.
482 Loop Detected		
483 Too Many Hops		
484 Address Incomplete		
485 Ambiguous	See comments	This response is received only by the phone in this release. If a new contact is received, the phone might reinitiate the call.
486 Busy Here	Yes	The Cisco SIP IP phone generates this response if the called party is off-hook and the call cannot be presented as a call-waiting call. Upon receiving this response, the phone notifies the user and generates a busy tone.
487 Request Canceled	Yes	This response indicates that the initial request is terminated with a BYE or CANCEL request.
488 Not Acceptable	Yes	The Cisco SIP IP phone receives and generates this response.

5*xx* Response—Server Failure Responses

5*xx* Response	Comments
500 Internal Server Error	The Cisco SIP IP phone does not generate these 5*xx* responses. For an incoming response, the SIP IP phone initiates a graceful call disconnect.
501 Not Implemented	
502 Bad Gateway	
503 Service Unavailable	
504 Gateway Timeout	
505 Version Not Supported	

6*xx* Response—Global Responses

6*xx* Response	Comments
600 Busy Everywhere	The Cisco SIP IP phone does not generate these 6*xx* responses. For an incoming response, the SIP IP phone initiates a graceful call disconnect.
603 Decline	
604 Does Not Exist Anywhere	
606 Not Acceptable	

SIP Header Fields

Header Field	Supported?
Accept	Yes
Accept-Encoding	Yes
Accept-Language	Yes
Allow	Yes
Also	Yes
Authorization	Yes
Call-ID	Yes
Contact	Yes
Content-Encoding	Yes
Content-Length	Yes
Content-Type	Yes
Cseq	Yes
Date	Yes
Encryption	No
Expires	Yes
From	Yes
Hide	No
Max-Forwards	Yes
Organization	No
Priority	No
Proxy-Authenticate	Yes

Header Field	Supported?
Proxy-Authorization	Yes
Proxy-Require	Yes
Record-Route	Yes
Referred-By	Yes
Referred-To	Yes
Remote-Party-ID	Yes
Replaces	Yes
Requested-By	Yes
Require	Yes
Response-Key	No
Retry-After	Yes
Route	Yes
Server	Yes
Subject	No
Timestamp	Yes
To	Yes
Unsupported	Yes
User-Agent	Yes
Via	Yes
Warning	Yes
WWW-Authenticate	Yes

SIP Session Description Protocol Usage

SDP Headers	Supported?
v—Protocol version	Yes
o—Owner or creator and session identifier	Yes
s—Session name	Yes
t—Time description	Yes
c—Connection information	Yes

SDP Headers	Supported?
m—Media name and transport address	Yes
a—Media attribute lines	Yes

Transport Layer Protocols

Protocol	Supported?
Unicast UDP	Yes
Multicast UDP	No
TCP	No

SIP Security Authentication

Basic Authentication	No
Digest Authentication	Yes
Proxy Authentication	No
PGP	No

SIP DNS Records Usage

DNS Resource Record Type	Supported?
Type A	Yes
Type SRV	Yes

SIP DTMF Digit Transport

Transport Type	Supported?
RFC 2833	Yes
In-band tones	Yes