



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:

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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:

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1xx Response—Information Responses

| 1xx 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 |

| 1xx 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. |

2xx Response—Successful Responses

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

3xx Response—Redirection Responses

| 3xx 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 | |

4xx Response—Request Failure Responses

| 4xx 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. |

| 4xx 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. |

| 4xx 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. |

5xx Response—Server Failure Responses

| 5xx Response | Comments |
|---------------------------|--|
| 500 Internal Server Error | The Cisco SIP IP phone does not generate these 5xx 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 | |

6xx Response—Global Responses

| 6xx Response | Comments |
|-----------------------------|--|
| 600 Busy Everywhere | The Cisco SIP IP phone does not generate these 6xx 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 |

