

SIP Quick Handbook

Session Initiation Protocol

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Session Initiation Protocol (SIP)

SIP is a signalling protocol used for creating, modifying, and terminating sessions with one or more participants in an IP network. SIP has been adopted by the telecommunications industry as its protocol of choice for signalling. SIP is an RFC standard (RFC 3261) from the Internet Engineering Task Force (IETF), the body responsible for administering and developing the mechanisms that comprise the Internet.

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SIP Methods (Requests)

Method	Description	RFC
ACK	Acknowledgement sent in response to a response to an INVITE request. Final responses are never ACK i.e. 2xx, 3xx, 4xx or 6xx class responses.	3261
BYE	Terminates a session.	3261
CANCEL	Cancels a pending transaction or call attempts.	3261
INFO	Carries session-related signalling information to a user agent within an established media session. It does not change media characteristics of the call.	6086
INVITE	An INVITE method is used to establish media sessions between user agents. In telephony, it is similar to an IAM in ISUP or SETUP in ISDN.	3261 6026
MESSAGE	Transfers instant messages IM using SIP.	3428
NOTIFY	Informs a subscriber about the state of a resource or occurrence of a particular event.	6665
OPTIONS	Queries a server or user agent about its capabilities.	3261
PRACK	Similar to ACK but for responding to reliable provisional responses with the exception of a 100 Trying	3262
PUBLISH	Publishes event state to a SIP events server.	3903
REFER	Indicates that the recipient should contact a third party URI or URL using contact information provided in the refer-to header field.	3515
REGISTER	The REGISTER method is used by a user agent to notify a SIP network of its current CONTACT URI that any request should be routed to.	3261
SUBSCRIBE	Requests current state and state updates from a remote node via the NOTIFY method.	6665
UPDATE	Updates parameters of a non-established session. A re-INVITE would be used to update session parameters in an established session.	3311

Response Codes

Response Code		RFC
Provisional	100 Trying	3261
	180 Ringing	3261
	181 Call is being forwarded	3261
	183 Session Progress	3261
	199 Early Dialog Terminated	6228
Successful	200 OK	3261
	202 Accepted	6665
	204 No Notification	5839
Redirection	300 Multiple Choices	3261
	301 Moved Permanently	3261
	302 Moved Temporarily	3261
	305 Use Proxy	3261
	380 Alternative Service	3261
Request Failure	400 Bad Request	3261
	401 Unauthorized	3261
	402 Payment Required	3261
	403 Forbidden	3261
	404 Not Found	3261
	405 Method not allowed	3261
	406 Not acceptable	3261
	407 Proxy authentication required	3261
	408 Request timeout	3261
	410 Gone	3261
	412 Conditional request failed	3903
	413 Request entity too large	3261
	414 Request-URI too long	3261
	415 Unsupported media type	3261
	416 Unsupported URI scheme	3261
417 Unknown resource-priority	4412	
420 Bad extension	3261	
421 Extension required	3261	
422 Session interval too small	4028	
423 Interval too brief	3261	
424 Bad Location Information	6442	
428 Use identity header	4474	
429 Provide referrer identity	3892	
430 Flow Failed	5626	
433 Anonymity disallowed	5079	
436 Bad identity-info	4474	

Response Code		RFC
437	Unsupported certificate	4474
438	Invalid identity header	4474
439	First Hop Lacks Outbound Support	5626
440	Max-Breadth Exceeded	5393
469	Bad Info Package	6086
470	Consent needed	5360
480	Temporarily unavailable	3261
481	Call Transaction does not exist	3261
482	Loop detected	3261
483	Too many hops	3261
484	Address incomplete	3261
485	Ambiguous	3261
486	Busy Here	3261
487	Request terminated	3261
488	Not acceptable here	3261
489	Bad event	6665
491	Request pending	3261
493	Undecipherable	3261
494	Security agreement required	3329
Server Failure	500 Server internal error	3261
	501 Not implemented	3261
	502 Bad gateway	3261
	503 Service unavailable	3261
	504 Server time-out	3261
	505 Version not supported	3261
	513 Message too large	3261
	580 Precondition failure	3312
Global Failures	600 Busy everywhere	3261
	603 Decline	3261
	604 Does not exist anywhere	3261
	606 Not acceptable	3261

Header Fields

Header Field	Used in	Description / Parameter	RFC
accept	Requests, 2xx, 415	Specifies media types that are acceptable in a response. In SIP a missing accept header field indicates a default value of application / sdp	3261
accept-encoding	Requests, 2xx, 415	This field is similar to accept, but restricts the content-codings that are acceptable in the response.	3841
accept-language	Requests, 2xx, 415	Used in requests to indicate the preferred languages for reason phrases, session descriptions or status responses carried as message bodies in the response.	3261
accept-resource-priority	Request, 2xx, 415		4412
alert-info	200, 417	When present in an INVITE request, the alert-info header field specifies an alternative ring tone to the UAS. When present in a 180 response, it specifies an alternative ringback tone to the UAC	4412
allow	Requests, 180	The allow header field lists the set of methods supported by the UA generating the message	3261
allow-events	Requests, 2xx, 489	The presence of subscribe in the allow header field of any request or response indicates support for SIP events; further in the absence of an allow header field, the simple presence of an allow-events header field is sufficient to indicate that the node that sent the message is capable of acting as a notifier.	6665
answer-mode		Expresses a preference as to whether the target node's user interface waits for user input before accepting the request or, instead accepts the request without waiting on user input.	5373
authentication-info	2xx	Provides for mutual authentication with HTTP digest.	3261
authorization	Requests	Contains authentication credentials of a UA.	3261
authorization		<i>algorithm</i>	
authorization		<i>auts</i>	
authorization		<i>cnonce</i>	
authorization		<i>nc</i>	
authorization		<i>nonce</i>	
authorization		<i>opaque</i>	
authorization		<i>qop</i>	

Header Field <i>continued</i>	Used in	Description / Parameter	RFC
authorization		<i>response</i>	
authorization		<i>realm</i>	
authorization		<i>uri</i>	
authorization		<i>username</i>	
call-id	Copied	The call-id header field uniquely identifies a particular invitation or all registrations of a particular client.	3261
call-info		The call-info header field provides additional information about the caller or callee, depending on whether it is found in a request or response.	3261
call-info		<i>m</i>	
call-info		<i>purpose</i>	
call-info		<i>purpose=call completion m=nr (no reply)</i>	6910
call-info		<i>purpose=call completion m=nl (not logged in)</i>	6910
call-info		<i>purpose=call completion m=bs (busy subscriber)</i>	6910
call-info		<i>purpose=impp (instant messaging presence purpose)</i>	6993
call-info		<i>purpose=ccmp (centralized conference manipulation protocol)</i>	7082
contact	Requests, 1xx, 2xx, 3xx, 485	A contact header field value provides a URI whose meaning depends on the type of request or response it is in.	3261
contact		<i>expires</i>	3261
contact		<i>mp</i>	7044
contact		<i>np</i>	7044
contact		<i>pub-gruu</i>	5627
contact		<i>q</i>	3261
contact		<i>rc</i>	7044
contact		<i>reg-id</i>	5626
contact		<i>temp-gruu</i>	5627
contact		<i>temp-gruu-cookie</i>	6140
content-disposition		The content-disposition header field describes how the message body or, for multipart messages, a message body part is to be interpreted by the UAC or UAS. This SIP header field extends the MIME content-type RFC2183 /// The content-disposition header may be included to describe how the encapsulated ISUP is to be processed.	3261
content-disposition	<i>handling</i>		3204
content-encoding		The content-encoding header field is used as a modifier to the "media-type". When present, its value indicates what additional content codings have been applied to the entity-body. Content-encoding is primarily used to allow a body to be compressed without losing the identity of its underlying media type.	3261
content-language		Example: Content-Language: fr	3261
content-length		The content-length header field indicates the size of the message-body, in decimal number of octets, sent to the recipient.	3261
content-type		The content-type header field indicates the media type of the message-body sent to the recipient.	3261
CSeq	Copied	In a request contains a single decimal sequence number and the request method.	3261
date		Contains the date and time.	3261
diversion		This is a historic header and is being migrated to history-info. Although the SIP history-info header is the solution adopted, the non-standard diversion is still widely used. Please refer to rfc7544 for diversion / history-info interworking.	5806
encryption	-	No longer used	
error-info	300-699	Provides a hint to additional information about the error status response.	3261
event	Requests		6665
event		<i>adaptive-min-rate</i>	6446
event		<i>body</i>	5989
event		<i>call-id</i>	4235
event		<i>effective-by</i>	6080
event		<i>from-tag</i>	4235
event		<i>id</i>	6665
event		<i>include-session-description</i>	4235
event		<i>max-rate</i>	6446
event		<i>min-rate</i>	6446
event		<i>model</i>	6080
event		<i>profile-type</i>	6080
event		<i>shared</i>	7463
event		<i>to-tag</i>	4235
event		<i>vendor</i>	6080
event		<i>version</i>	6080
expires	2xx	Gives the relative time after which the message (or content) expires.	3261

Header Field <i>continued</i>	Used in	Description / Parameter	RFC
feature-caps		This field conveys feature-capability indicators that are used to indicate support of features and capabilities for SIP entities that are not represented by the Uniform Resource Identifier (URI) of the contact header field.	6809
feature-caps		<i>fcap-name</i>	
from	Copied	Indicates the initiator of the request. This may be different from the initiator of the dialog. Request sent by the callee to the caller use the callee's address in the From header field.	3261
from		<i>tag</i>	
geolocation		In order to convey location information, this document specifies three new SIP header fields, Geolocation, Geolocation-Routing and Geolocation-error, which carry a reference to a location object (LO) grant permission to route a SIP request based on the location-value and provide error notifications specific to location errors respectively.	6442
geolocation-error		<i>codes</i> (see rfc for list of codes)	6442
geolocation-routing			6442
hide	-	No longer used	
history-info		For capturing information as to how and why a SIP request arrives at a specific application or user.	7044
history-info		<i>Mp</i>	7044
history-info		<i>Np</i>	7044
history-info		<i>Rc</i>	7044
identity	Requests	Existing security mechanisms in the SIP are inadequate for cryptographically assuring the identity of the end users that originate SIP requests, especially in an interdomian context. New header fields created to address this issue. identity for conveying a signature used for validating the identity, and identity-info for conveying a reference to the certificate of the signer.	4474
identity-info	Requests		4474
in-reply-to	Requests	Enumerates the Call-IDs that this call references or returns. These call-ids may have been cached by the client then included in this header field in a return call.	3261
join	Requests	Used to logically join an existing SIP dialog with a new SIP dialog.	3911
max-breadth		Mechanism for limiting the number of concurrent branches pursued for any given request.	5393
max-forwards	Requests	This field must be used with any SIP method to limit the number of proxies or gateways that can forward the request to the next downstream server.	3261
MIME-version		Example: MIME-Version: 1.0	3261
min-expires	423	Conveys the minimum refresh interval supported for soft-state elements managed by the server.	3261
min-se	Requests, 422	Conveys the minimum allowed value for the session timer.	4028
organization		This field conveys the name of the organization to which the SIP element issuing the request or response belongs.	3261
p-access-network-info		This header is field is useful in SIP-based networks that also provide layer 2 / layer 3 connectivity through different access technologies.	7315
p-access-network-info		<i>cgi-3gpp</i>	7315
p-access-network-info		<i>ci-3gpp2</i>	7315
p-access-network-info		<i>ci-3gpp2-femto</i>	7315
p-access-network-info		<i>dsl-location</i>	7315
p-access-network-info		<i>dvb-rcs2-node-id</i>	7315
p-access-network-info		<i>eth-location</i>	7315
p-access-network-info		<i>fiber-location</i>	7315
p-access-network-info		<i>gstn-location</i>	7315
p-access-network-info		<i>i-wlan-node-id</i>	7315
p-access-network-info		<i>local-time-zone</i>	7315
p-access-network-info		<i>operator-specific-GI</i>	7315
p-access-network-info		<i>utran-cell-id-3gpp</i>	7315
p-access-network-info		<i>utran-sai-3gpp</i>	7315
p-answer-state	Request, 18x, 2xx	Used by the Open Mobile Alliance for Push to Talk over Cellular (PoC)	4964
p-asserted-identity		Enables a network of trusted SIP servers to assert the identity of authenticated users. (sometimes referred to as PAID P-Asserted-IDentity)	5876
p-asserted-service	Requests	This field is used among trusted SIP entities (typically intermediaries) to carry the service information of the user sending a SIP message.	6050
p-associated-uri	2xx	Allows a registrar to return a set of associated URIs for a registered SIP address-of-record. We define the P-Associated-URI header field, used in the 200 (ok) response to a REGISTER request. The header contains the set of associated URIs that are associated with the registered address-of-record.	7315

Header Field <i>continued</i>	Used in	Description / <i>Parameter</i>	RFC
p-called-party-id	Requests	A proxy server inserts this header, typically in an INVITE request, en route to its destination. The header is populated with the request-URI received by the proxy in the request. The user agent server (UAS) identifies to which address of record out of several, the invitation was sent. The UAS can use the information to render different distinctive audio-visual alerting tones.	7315
p-charging-function-address		Charging mechanism for access or services.	7315
p-charging-vector		A globally unique charging identifier	7315
p-dcs-trace-party-id	Requests	In the telephone network, calling identity information is used to support regulatory requirements such as the customer originated trace service, which provide the called party with the ability to report obscene or harassing phone calls to law enforcement.	5503
p-dcs-osps	Requests	Operator Services Position System – enables special call processing, in particular, the busy line verification (BLV) and Emergency Interrupt (EI)	5503
p-dcs-billing-info		Used for billing of network access services	5503
p-dcs-laes		Information needed to support Lawfully Authorized Electronic Surveillance. This header contains the address and port of an electronic surveillance delivery function for delivery of a duplicate stream of event messages related to this call.	5503
p-dcs-redirect		Contains call identifying information needed to support the requirements of Lawfully Authorized Electronic Surveillance of redirected calls.	5503
p-early-media	Request, 18x, 2xx	For use within SIP message in certain SIP networks to authorize the cut-through of backward and or forward early media when permitted by the early media policies of the networks involved.	5009
p-media-authorization	Requests, 101-199, 2xx	This field contains one or more media authorization tokens which are to be included in subsequent resource reservations for the media flows associated with the session.	3313
p-preferred-identity		Used from a user agent to a trusted proxy to carry the identity of the user sending the SIP message wishes to be used for the P-Asserted-Header field.	5876
p-preferred-service	Requests	Used by a user agent sending the SIP request to provide a hint to a trusted proxy of the preferred service that the user wishes to be used for the P-Asserted-Service field value that the trusted element will insert.	6050
p-profile-key		Carries the key of a service profile, that is stored in a user database referred to as HSS, between two proxies, which are referred to as I-CSCF and S-CSCF.	5002
p-user-database		This header field can be added to 'requests' routed from an I-CSCF to an S-CSCF. The P-User-Database P-header contains the address of the HSS handling the user that generated the request.	4457
p-visited-network-id	Requests	Used to convey to the registrar or home proxy in the home network the identifier of a visited network.	7315
path	Requests, 2xx	The REGISTER method does not capture all proxies, the path header works in the same way as route header and records all proxies for future use.	3327 5626
permission-missing		These header fields carry URIs for which a relay did not have permission.	5360
priority	Requests	Indicates the urgency of the request as perceived by the client.	3261
priv-answer-mode		Used in INVITE requests to convey the requester's preference for user-interface handling related to answering of that request. See first header: "Answer-Mode"	5373
privacy		When Privacy = id p-asserted-identity is removed.	3323
proxy-authenticate		Field value contains an authentication challenge.	3323
proxy-authorization	407, 401	Allows the client to identify itself (or its user) to a proxy that requires authentication. A proxy-authorization field value consists of credentials containing the authentication information of the user agent for the proxy and or realm of the resource being requested.	3261
proxy-require	Requests	Used to indicate proxy-sensitive features that must be supported by the proxy.	3261
Rack	PRACK Requests	The Rack header is sent in a PRACK request to support reliability of provisional responses. It contains two numbers and a method tag. The first number is the value from the Rseq header in the provisional response that is being acknowledged. The next number and the method are copied from the Cseq in the response that is being acknowledged.	3262
Reason		Can be used to encapsulate a final error response in a PRACK message due to request no acceptable as opposed to initiation being declined. i.e. forked calls.	3326
reason-phrase		reserved to avoid conflict with rfc6873	
record-route	Requests, 2xx, 18x	This is inserted by proxies in a request to force future requests in the dialog to be routed through the proxy.	3261
recv-info	Requests, 1xx, 2xx & 469	For matching info package types indicated in Recv-Info with those in the Info-Package header field.	6086
refer-events-at		This header is only meaningful in a 2xx class response to a REFER request. If it appears in the heade3r of any other SIP message, its meaning is undefined and it must be ignored.	7641

Header Field <i>continued</i>	Used in	Description / <i>Parameter</i>	RFC
refer-sub	Requests, 2xx	This header when set to 'false' specifies that a REFER-Issuer requests that the REFER-Recipient doesn't establish an implicit subscription and the resultant dialog.	4488
refer-to	Requests	Provides a URL to reference and only appears in a REFER request.	3515 7647
referred-by	Requests	Is a request header field, it can appear in any request. It carries a SIP URI representing the identity of the referrer and optionally the content-id of a body part (the referred-by token) that provides a more secure statement of that identity.	3892
reject-contact	Requests	This header allows the UAC to specify that a UA should not be contacted if it matches any of the values of the header field.	3841
replaces	Requests	The replaces header is used to logically replace an existing SIP dialog with a new SIP dialog. This primitive can be used to enable a call variety of features, for example: Attended Transfer and Call Pickup.	3891
reply-to		Contains a logical return URI that may be different from the FROM header field. If the user wished to remain anonymous, the header field should either be omitted from the request or populated in such a way that does not reveal any private information.	3261
request-disposition	Requests	Carries caller preferences that describe desired request treatment at a server.	3841
require		Used by UACs to tell UASs about options that the UAC expects the UAS to support in order to process the request. This is an optional header, but must be actioned if present.	3261
resource-priority	Requests	The header field marks a SIP request as desiring prioritized access to resources.	4412 7134
retry-after	404, 413, 480, 486, 500, 600, 603	This can be used with a 500 (server internal error) or 503 (service unavailable) response to indicate how long the service is expected to be unavailable to the requesting client and with a 404 (not found), 413 (request entity too large), 480 (temp unavailable), 486 (busy here), 600 (busy), or 603 (decline) response to indicate when the called party anticipates being available again.	3261
route	Request	The route header field is used to force routing for a request through the listed set of proxies.	3261
Rseq	1xx	The Rseq header is used in provisional responses in order to transmit them reliably. It contains a single numeric value from 1 to (2 ³²) - 1	3262
security-client	Requests	This header field contains a list of all the security mechanisms that the client supports.	3329
security-server	4211, 494	List of security mechanisms available and offered by the server.	3329
security-verify	Requests	Subsequent requests from the client MUST contain a security-verify header field that mirrors the server's list received previously in the security-server header.	3329
server	Responses	Information about the software used by the UAS to handle the request.	3261
service-route	2xx		3608 5630
session-expires	Requests, 2xx	The header field conveys the session interval for a SIP session. It is placed only in INVITE or UPDATE requests, as well as in any 2xx response to an INVITE or UPDATE. Like the SIP expires header field, it contains a delta-time.	4028
session-id		The session-id is similar to that of a Call-ID, it is not used for message dialog-matching rules nor does it change or replace call-id usage. It is to provide an identifier for troubleshooting use only. /// = 32(DIGIT / %x61-66) ; 32 chars of 0-9a-f	7329
SIP-Etag	2xx	For each successful PUBLISH request, the ESC (event state compositor) will generate and assign an entity-tag and return it in the SIP-Etag header field of the 2xx response.	3903
SIP-if-match	Requests	When updating previously published event state, PUBLISH requests must contain a single SIP-If-Match header field identifying the specific event state that the request is refreshing, modifying or removing. The header field must contain a single entity-tag that was returned by the ESC (event state compositor) in the SIP-Etag header field of the response to a previous publication.	3903
subject	Requests	Provides a summary or indicates the nature of the call, allowing call filtering without having to parse the session description.	3261
subscription-state	Requests	Contains the state and expiration time of the subscription	6665
supported	Requests, 2xx	The supported header field enumerates all the extensions supported by the UAC or UAS.	3261
suppress-if-match		Contains the last entity-tag seen by the subscriber.	5839
target-dialog	Requests	This header field contains the dialog identifiers for the INVITE dialog between user agents A and B, composed of the Call-ID, local tag, and remote tag.	4538
timestamp		The timestamp header field describes when the UAC sent the request to the UAS.	3261

Header Field <i>continued</i>	Used in	Description / <i>Parameter</i>	RFC
to	Copied, tag added	The to header field specifies the logical recipient of the request.	3261
trigger-consent		Will have a target-uri header field parameter identifying the target URI of the translation.	5360
unsupported	420	Lists the features not supported by the UAS.	3261
user-agent		Contains information about the UAC originating the request.	3261
user-to-user		Contains information pertaining to the transport of UII data relating to call control Q.931 user-user information & Q.763 user-to-user information parameter.	7433
via	Requests, Responses, Copied	The via header field indicates the path taken by the request so far and indicates the path that should be followed in routing responses. The branch ID parameter in the Via header field values serves as a transaction identifier, and is used by proxies to detect loops.	3261 7118
warning	Requests, Responses	Used to carry additional information about the status of a response. Warning header field values are sent with responses and contain a three-digit warning code, host name and warning text. (see warning codes)	3261
www-authenticate	410, 407	Contains an authentication challenge.	3261
X-Nortel-Profile		Ribbon propriety similar to TGRP. All Manufacturer propriety fields start with X	

- **Copied:** header field is copied from the request to the response.
- An empty entry in the "Used in" column indicates that the header field may be present in all requests and responses.

Header Fields – Compact Forms

Compact	Header Field
a	Accept-Contact
b	Referred-By
c	Content-Type
d	Request-Disposition
e	Content-Encoding
f	From
i	Call-ID
j	Reject-Contact
k	Supported
l	Content-Length
m	Contact
n	Identity-Info
o	Event
r	Refer-To
s	Subject
t	To
u	Allow-Events
v	Via
x	Session-Expires
y	Identity

URI Purposes

Value	Description of URI
participation	used to join the conference
streaming	can be used to access the streamed conference data.
event	Can be used to subscribe to the conference event package
recording	Can be used to access the recorded conference data
web-page	Can be used to access a web page that contains additional information of the conference

SIP / SIPS URI Parameters

Parameter Name	Predefined Values	Description	RFC
aai	No	Used to specify a JSON (JavaScript Object Notation) value that is mapped to thesession.connection.aai VoiceXML session	5552
bnc	No		6140
cause	Yes	Voicemail	4458
ccxml	No	Used to specify a 'JSON value' that is mapped to the session.connection.ccxml VoiceXML session variable	5552
comp	Yes	Ability to handle compressed SIP messages	3486
content-type	No	Service announcements & conferencing	4240
delay	No	Specifies a delay interval between announcement repetitions. The delay is measured in milliseconds.	4240
duration	No	Specifies the maximum duration of the announcement.	4240
extension	No		4240
gr	no	Globally Routable User Agent (GRUU) – In SIP the basic unit of reference is the Address of Record (AOR) however in SIP systems a single user can have a number of user agents (handsets, softphones, voicemail etc) that are all referenced by the same AOR. In certain circumstances it is desirable to have an identifier that addresses a single user agent rather than the group of user agents indicated by the AOR.	5627
iotl	Yes	Service announcements & conferencing	7549
locale	No	Specifies the language and optional country variant of the announcement sequence named in the "play=" parameter.	4240
lr	No	loose routing – supports 3261 compliant routing	3261
m	Yes	Server address to be contacted for the user	6910
maddr	No		3261
maxage	No	used to set the max-age value of the cache-control header in conjunction with VoiceXML documents fetched using http	5552
maxstale	No	used to set the max-stale value of the cache-control header in conjunction with VoiceXML documents fetched using http	5552
method	get / post	Used to set the HTTP method applied in the fetch of the initial VoiceXML document.	5552
method	Yes	Specifies the method of the SIP request constructed from the URI	3261
ob	No	Service announcements & conferencing	5626
param[n]	No	Service announcements & conferencing	4240
play	No		4240
postbody	No	Service announcements & conferencing	5552
repeat	No		4240
sg	No	URN of a SIP/SigComp application	6140
sigcomp-id	No	Indicates the address of the retargeting entity	5049
target	No	Transport mechanism to be used for sending SIP messages	4458
transport	Yes	Time to live value of the UDP multicast packet	3261 7118
tgrp		Representing trunk groups in tel/sip	4904
ttl	No	Distinguishes telephone numbers from the user names that look like telephone numbers	3261
user	Yes	Indicates the RUI of the VoiceXML script to execute	3261 4967
voicexml	No		4240

tel URI Parameters

Parameter Name	Predefined Values	Description	RFC
isub	Constrained	isdn-subaddress – A phone number may also contain an isdn subaddress parameter that indicates an ISDN subaddress. (phone extension)	3966
isub-encoding	<i>nsap-ia5</i> <i>nsap-bcd</i> <i>nsap</i>	Indicates the encoding of the isdn-subaddress if this is not specified the default ia5 is used.	4715
ext	Constrained	extension – phone extensions identify stations behind a non-ISDN PBX and are functionally roughly equivalent to ISDN subaddresses.	3966
phone-context	Constrained	A telephone subscriber number can be expressed as a global number (e164) or a local number. All phone numbers MUST use the global form unless they cannot be represented as such. Local numbers MUST be tagged with a phone-context E164 format tel:+442075496044 or Local format with phone-context tel:75496044;phone-context=+4420	3966
enumdi	No value	enum dip – indicates that this e164 number has already had an ENUM query performed by a previous VoIP network element.	4759
npdi	No value	Number Portability dip – This indicates to the downstream servers or switches during call setup that an NP database dip has been performed already.	4694
rn	Constrained	routing-number – a routing number is associated with a geographical or mobile telephone number that has been ported out from a donor carrier to another carrier.	4694
rn-context	Constrained	rn-context parameter describes how the routing number value should be interpreted when the value is not a global-rn. Similar rules as phone-context.	4694
cic	Constrained	Carrier identification code – used to identify a particular service provider maps to TNS is SS7	4694
cic-context	Constrained	cic-context describes how the CIC value is to be interpreted.	4694
tgrp	Constrained	trunk group – this is a way of extending tel and sip uri when interworking with SS7 network to specify which trunk group to use for termination of the call.	4904
trunk-context	Constrained	This parameter works in the similar way to phone-context , we are able to impose specific details about the tgrp : <ol style="list-style-type: none"> 1. Specify a host 2. Specify a domain to use 3. Specify a subset of a gateway domain 4. Global number or any number of leading digits. Unlike phone-context – we can use trunk-context on local and global tel URIs	4904

NB. All tel URI parameters may be used with a SIP URI when USER=PHONE is specified.

SIP Events

Package Name	Subscription	RFC
conference	URI for a conference to learn about other members and conference components	4575
dialog	Users and their changes of state of INVITE-initiated dialog usages in which they are involved	4235
kpml	Dual Tone Multi-Frequency (DTMF) signals for supplemental or mid-call key presses entered at the UA. The Key Press Markup Language (KPML) documents (xml) sent in the SUBSCRIBE define and describe filter specifications for capturing key presses. The KPML documents sent in the NOTIFYs report the captured key presses that match the filter criteria to an application server.	4730
message-summary	Message waiting status and message summaries from a messaging system	3842
poc-settings	Capabilities required by the Push-to-Talk over Cellular (PoC) service.	4354
presence	Users' availability and willingness to communicate with other users on the network	3856
reg	UA's registration state	3680
refer	Status of REFER request	3515
winfo	Set of watchers subscribed to the UA's presence information	3857

Option Tags for SIP extensions

Option Tag	Supported Extension	Description	RFC
100rel	Reliability of provisional responses	This option tag is for reliability of provisional responses. When present in a Supported header, it indicates that the UA can send or receive reliable provisional responses. When present in a Require header in a request it indicates that the UAS MUST send all provisional responses reliably. When present in a Require header in a reliable provisional response, it indicates that the response is to be sent reliably.	3262
199		This option-tag is for indicating support of the 199 Early Dialog Terminated provisional response code. When present in a Supported header of a request, it indicates that the UAC supports the 199 response code. When present in a Require or Proxy-Require header field of a request, it indicates that the UAS, or proxies, MUST support the 199 response code. It does not require the UAS, or proxies, to actually send 199 responses.	6228
answermode		This option tag is for support of the Answer-Mode and Priv-Answer-Mode extensions used to negotiate automatic or manual answering of a request.	5373
early-session	Early-session content disposition	A UA adding the early-session option tag to a message indicates that it understands the early-session content disposition.	3959
eventlist	Subscriptions to lists of resources	Extension to allow subscriptions to lists of resources	4662
explicitsub		This option tag identifies an extension to REFER to suppress the implicit subscription and provide a URI for an explicit subscription.	7614
from-change	Connected identity	This option tag identifies an extension to REFER to suppress the implicit subscription and provide a URI for an explicit subscription.	4916
geolocation-http		The "geolocation-http" option tag signals support for acquiring location information via an HTTP [RFC2616]. A location recipient who supports this option can request location with an HTTP GET and parse a resulting 200 response containing a PIDF-LO object. The URI schemes supported by this option include "http" and "https".	6442
geolocation-sip		The "geolocation-sip" option tag signals support for acquiring location information via the presence event package of SIP [RFC3856]. A location recipient who supports this option can send a SUBSCRIBE request and parse a resulting NOTIFY containing a PIDF-LO object. The URI schemes supported by this option include "sip", "sips", and "pres".	6442
gin		This option tag is used to identify the extension that provides Registration for Multiple Phone Numbers in SIP. When present in a Require or Proxy-Require header field of a REGISTER request, it indicates that support for this extension is required of registrars and proxies, respectively, that are a party to the registration transaction.	6140
gruu	Globally routable user agent URI	This option tag is used to identify the Globally Routable User Agent URI (GRUU) extension. When used in a Supported header, it indicates that a User Agent understands the extension. When used in a Require header field of a REGISTER request, it indicates that the registrar is not expected to process the registration unless it supports the GRUU extension.	5627
histinfo	History-Info header field	When used with the Supported header field, this option tag indicates the UAC supports the History Information to be captured for requests and returned in subsequent responses. This tag is not used in a Proxy-Require or Require header field, since support of History-Info is optional.	7044
ice		This option tag is used to identify the Interactive Connectivity Establishment (ICE) extension. When present in a Require header field, it indicates that ICE is required by an agent.	5768
join	Join header field	Support for the SIP Join Header	3911
multiple-refer		This option tag indicates support for REFER requests that contain a resource list document describing multiple REFER targets.	
norefersub	Suppression of implicit subscriptions	This option tag specifies a User Agent ability of accepting a REFER request without establishing an implicit subscription (compared to the default case defined in [RFC3515]).	4488
nosub		This option tag identifies an extension to REFER to suppress the implicit subscription and indicate that no explicit subscription is forthcoming.	7614
outbound		This option-tag is used to identify UAs and Registrars which support extensions for Client Initiated Connections. A UA places this option in a Supported header to communicate its support for this extension. A Registrar places this option-tag in a Require header to indicate to the registering User Agent that the Registrar used registrations using the binding rules defined in this extension.	5626

Option Tag	Supported Extension	Description	RFC
path	Path header field	A SIP UA that supports the Path extension header field includes this option tag as a header field value in a Supported header field in all requests generated by that UA. Intermediate proxies may use the presence of this option tag in a REGISTER request to determine whether to offer Path service for that request. If an intermediate proxy requires that the registrar support Path for a request, then it includes this option tag as a header field value in a Requires header field in that request.	3327
policy		This option tag is used to indicate that a UA can process policy server URIs for and subscribe to session-specific policies.	6794
precondition	Preconditions for session establishment	Supported – indicates that the offer contains only 'optional' or 'none' strength-tags. Require – indicates that the offer contains one or more 'mandatory' strength-tags, or only 'optional' or 'none' strength-tags.	3312
pref	Caller preferences	Require of a REGISTER – ensures that the registrar supports caller preferences extensions.	3840
Privacy	Privacy mechanism	Proxy-Require – indicates that proxy servers do not forward the request unless they can provide the requested privacy service. Proxies remove this option tag before forwarding the request if the desired privacy function has been performed.	3323
recipient-list- invite		The body contains a list of URIs that indicates the recipients of the SIP INVITE request	5366
recipient-list- message		The body contains a list of URIs that indicates the recipients of the SIP MESSAGE request	5365
recipient-list- subscribe		This option tag is used to ensure that a server can process the recipient-list body used in a SUBSCRIBE request.	5367
record-aware		This option tag is to indicate the ability for the user agent to receive recording indicators in media-level or session-level SDP. When present in a Supported header, it indicates that the UA can receive recording indicators in media- level or session-level SDP.	ietf- siprec- proto- l-18
replaces	Replaces header field	This option tag indicates support for the SIP Replaces header.	3891
resource- priority	Resource-Priority and Accept-Resource-Priority header fields	Indicates or requests support for the resource priority mechanism.	4412
sdp-anat	Alternative network address types of the SDP Grouping framework	The option-tag sdp-anat is defined for use in the Require and Supported SIP [RFC3261] header fields. SIP user agents that place this option-tag in a Supported header field understand the ANAT semantics as defined in [RFC4091].	4092
sec-agree	Security agreement mechanism	This option tag indicates support for the Security Agreement mechanism. When used in the Require, or Proxy-Require headers, it indicates that proxy servers are required to use the Security Agreement mechanism. When used in the Supported header, it indicates that the User Agent Client supports the Security Agreement mechanism. When used in the Require header in the 494 (Security Agreement Required) or 421 (Extension Required) responses, it indicates that the User Agent Client must use the Security Agreement Mechanism.	3329
siprec		This option tag is for identifying that the SIP session is for the purpose of a recording session. This is typically not used in a Supported header. When present in a Require header in a request, it indicates that the UA is either an SRC or SRS capable of handling a recording session.	ietf- siprec- proto- l-18
tdialog	Target-Dialog header field	This option tag is used to identify the target dialog header field extension. When used in a Require header field, it implies that the recipient needs to support the Target-Dialog header field. When used in a Supported header field, it implies that the sender of the message supports it.	4538
timer	Session timers	This option tag is for support of the session timer extension. Inclusion in a Supported header field in a request or response indicates that the UA is capable of performing refreshes according to that specification. Inclusion in a Require header in a request means that the UAS must understand the session timer extension to process the request. Inclusion in a Require header field in a response indicates that the UAC must look for the Session-Expires header field in the response, and process accordingly.	4028
uui		This option tag is used to indicate that a UA supports and understands the User-to-User header field.	7433

SIP Timers (RFC3261)

Timer	Value	RFC 3261 Section	Meaning
T1	500ms	17.1.1.1	RTT estimate
T2	4s	17.1.2.2	The maximum retransmit interval for non-INVITE request and INVITE responses
T4	5s	17.1.2.2	Maximum duration a message will remain in the network
Ta	T1	17.1.1.2	INVITE request retransmit interval, for UDP only
Tb	64*T1	17.1.1.2	Invite transaction
Tc	>3min	16.6	Proxy INVITE transaction timeout
Td	>32ms for UDP	17.1.1.2	Wait time for response retransmits
Te	T1	17.1.2.2	non-INVITE request retransmit interval, UDP only
Tf	64*T1	17.1.2.2	Non-INVITE transaction timeout timer
Tg	T1	17.2.1	INVITE response retransmit interval
Th	64*T1	17.2.1	Wait time for ACK receipt
Ti	T4 for UDP	17.2.1	Wait time for ACK retransmits
Tj	64*T1 for UDP	17.2.2	Wait time for ACK retransmits
Tk	T4 for UDP 0s for TCP/SCTP	17.1.2.2.	Wait time for response retransmits
Tl	64*T1	draft	Wait time for INVITE retransmissions

Session Description Protocol (SDP)

RFC4566

The session description protocol is what describes the session being requested. The SDP is carried in the message body of a SIP request/response. Each attribute line consists of an attribute identified by a single letter, followed by a value.

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Session Level Description

Parameter	Description
v = protocol version	This is the version being used to create the SDP, currently only 0
o = owner / creator and session identifier	The identity of the session initiator
s = session name	Optional name that can be given to the session
i = session information	Additional information that the creator of the session wishes to share with participants
u = URI of description	This contains the URI of a Web site that may contain additional information about the session
e = email address	This usually contains the email address of the creator, where participants can acquire more information
p = phone number	The contact phone number where more information can be provided about the session
c = connection information	Additional information about the connection for the session
b = bandwidth information	The amount of bandwidth to be provided for the session
z = time zone adjustments	Any time zone adjustments to be considered
k = encryption keys	The encryption keys for the session
a = zero or more session attribute lines	The number of attribute lines in the SDP

Time Description

Parameter	Description
t = time the session is active	What time does the session start, for example
r = zero or more repeat times	How many times the session repeats

SDP Capability Negotiation Configuration Parameters

Parameter	Description	RFC
m = media name and transport address	Name of the media, if applicable	6871
i = media title	Title or information field	7006
c = connection information	Additional connection information	7006
b = bandwidth information	Bandwidth required to support the media	7006
k = encryption key	Encryption keys required	4566
a = zero or more attribute lines	Number of attribute lines provided – overrides the session attribute lines	5939
mt = media type		6871
pt = payload type number mapping		6871
t = transport protocol configuration		6871

SDP Media Attribute Lines (a=)

Attribute	Description	RFC
cat	category	4566
keywds	keywords	4566
tool	name and version of tool	4566
ptime	packet sample time	4566
maxptime	maximum packet sample time	4566
recvonly	receive-only mode	4566
sendonly	send only mode	4566
sendrecv	send and receive mode	4566
inactive	no media in either direction	4566
orient	whiteboard orientation	4566
type	conference type	4566
charset	character set	4566
sdplang	language tag	4566
lang	language tag	4566
framerate	frame rate	4566
quality	quality	4566
fntp	format specific parameters	4566
rtpmap	rtpmap attribute	4566
curr	current status attribute (qos)	3312
des	desired status attribute (qos)	3312
conf	confirm status attribute (qos)	3312
mid	media stream identification attribute	5888
group	group attribute	5888
rtcp	rtcp port (used to specify a port number other than the default)	3605
crypto	Encryption keys	4568

SDP Media payload / name

Payload	Name	Audio / Video	Clock rate (Hz)	channels
0	G711 mu-law	a	8000	1
1	reserved			
2	reserved			
3	GSM	a	8000	1
4	G723	a	8000	1
5	DVI4	a	8000	1
6	DVI4	a	16000	1
7	LPC	a	8000	1
8	G711 a-law	a	8000	1
9	G722	a	8000	1
10	L16	a	44100	2
11	L16	a	44100	1
12	QCELP	a	8000	1
13	CN	a	90000	1
14	MPA	a		
15	G728	a	8000	1
16	DVI14	a		
17	DVI14	a		
18	G729	a	8000	1
19	reserved	a		
20	unassigned	a		
21	unassigned	a		
22	unassigned	a		
23	unassigned	a		
24	unassigned	v		
25	CelB	v	90000	
26	JPEG	v	90000	
27	unassigned	v		
28	nv	v	90000	
29	unassigned	v		
30	unassigned	v		
31	H261	v	90000	1
32	MPV	v	90000	
33	MP2T	av	90000	
34	H263	v	90000	1
35-71	unassigned			
72-76	reserved for RTCP conflict avoidance			
77-95	unassigned			
96-127	Dynamic			

RTP payload format media types

In addition to the previously listed RTP payload formats, there are payload formats that do not have static payload numbers assigned but use dynamic payload type number assignment (96-127)

Media Type	Subtype	Clock Rate	Channels	RFC
application	1d-interleaved-parityfec			6015
application	h224	4800		4573
application	parityfec			3009
application	raptorfec			6682
application	rtx			4588
application	smppte336m			6597
application	ulpfec			5109
audio	1d-interleaved-parityfec			6015
audio	32kadpcm	8000		3802 2421
audio	ac3			4184
audio	AMR	8000		4867 3267
audio	AMR-WB	16000		4867 3267
audio	amr-wb+	72000		4352
audio	atrac3	44100		5584
audio	ATRAC-ADVANCED-LOSSLESS			5584
audio	atrac-x			5584
audio	BV16	8000		4298
audio	BV32	16000		4298
audio	clearmode	8000	1	4040
audio	CN			3389
audio	DAT12			3190
audio	dsr-es201108			3557
audio	dsr-es202050	8000		4060
audio	dsr-es202211	8000		4060
audio	dsr-es202212	8000		4060
audio	DV			6469
audio	eac3			4598
audio	EVRC	8000	1	4788
audio	EVRC0	8000	1	4788
audio	EVRC1	8000	1	4788
audio	EVRCB	8000	1	4788
audio	EVRCB0	8000	1	4788
audio	EVRCB1	8000	1	4788
audio	EVRCWB			5188
audio	EVRCWB0			5188
audio	EVRCWB1			5188
audio	fwired			6354
audio	G719	48000		5404
audio	G7221	16000	1	5577
audio	G726-16	8000	1	3551 4856
audio	G726-24	8000	1	3551 4856
audio	G726-32	8000	1	3551 4856
audio	G726-40	8000	1	3551 4856
audio	G729D	8000	1	3551 4856
audio	G729E	8000	1	3551 4856
audio	GSM-EFR	8000	1	3551 4856
audio	L8			3551 4856
audio	raptorfec			6682
audio	RED			2198 3555
audio	rtx			4588
audio	VDVI		1	3551 4856
audio	L20			3190
audio	L24			3190
audio	MP4A-LATM			3016
audio	mpa-robust	90000		5219
audio	parityfec			5109
audio	SMV	8000	1	3558
audio	SMV0	8000	1	3558
audio	t140c			4351
audio	t38			4612
audio	telephone-event			4733

Media Type	Subtype	Clock Rate	Channels	RFC
audio	tone			4733
audio	DVI4			4856
audio	G722			4856
audio	G723			4856
audio	G728			4856
audio	G729			4856
audio	GSM			4856
audio	L16			4856
audio	LPC			4856
audio	PCMA			4856
audio	PCMU			4856
audio	G7291	16000		4749 5459
audio	GSM-HR-08	8000		5993
audio	iLBC	8000		3952
audio	ip-mr_v2.5	16000		6262
audio	MPA	90000		3555
audio	mpeg4-generic			3640 5691 6295
audio	PCMA-WB	16000		5391
audio	PCMU-WB	16000		5391
audio	QCELP			3555
audio	rtp-midi			6295
audio	speex			5574
audio	uemclip			5686
audio	ulpfec			5109
audio	VMR-WB	16000		4348 4424
audio	vorbis			5215
audio	vorbis-config			5215
text	1d-interleaved-parityfec			6015
text	fwdrd			6354
text	parityfec			3009
text	raptorfec			6682
text	red	1000		4102
text	rtx			4588
text	t140	1000		4103
text	ulpfec			5109
video	BMPEG	90000		2343 3555
video	1d-interleaved-parityfec			6015
video	3gpp-tt			4396
video	BT656	90000		2431 3555
video	celB			3555
video	DV	90000		6469
video	H261			4587
video	H263	90000		4628
video	H263-1998	90000		4629
video	H263-2000	90000		4629
video	H264			6184
video	H264-RCDO	90000		6185
video	H264-SVC			6190
video	JPEG			3555
video	JPEG2000			5371
video	MP1S	90000		2250 3555
video	MP2P	90000		2250 3555
video	MP2T			3555
video	MP4V-ES	90000		3016
video	mpeg4-generic			3640
video	MPV			3555
video	nv			4856
video	parityfec			5109
video	pointer	90000		2862
video	raptorfec			6682
video	raw	90000		4175
video	rtx			4588
video	SMPTE292M			3497
video	ulpfec			5109
video	vc1	90000		4425

Example SDP

```
v=0
o=SIPUAC_5.548 126749828 1 IN IP4 66.33.131.9
s=call
c=IN IP4 66.33.131.9
t=0 0
m=audio 21778 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000/1
a=rtpmap:8 PCMA/8000/1
a=rtpmap:18 G729/8000/1
a=rtpmap:101 telephone-event/8000
a=fmtp:18 annexb=no
a=fmtp:101 0-16
a=ptime:20
a=sendrecv
```

```
v=0 (version of SDP presently only 0 (zero) available)
o=SIPUAC_5.548 126749828 1 IN IP4 66.33.131.9 (session identifier and network address)
s=call (the type of session, this is a basic call)
c=IN IP4 66.33.131.9 (ip address unless specified in media)
t=0 0 (time in NTP format, no timings have been specified)
m=audio 21778 RTP/AVP 0 8 18 101 (media, this is an audio session only, with RTP to be sent to port 21778 offering the following codecs: 0=G.711 mu-law, 8=G.711 a-law, 18=G.729 and 101=telephone events)
a=rtpmap:0 PCMU/8000/1 (G.711 mu-law)/8Khz sample rate/1 audio channel
a=rtpmap:8 PCMA/8000/1 (G.711 a-law)/8Khz sample rate/1 audio channel
a=rtpmap:18 G729/8000/1 (G.729)/8Khz sample rate/1 audio channel
a=rtpmap:101 telephone-event/8000 (Support for dtmf tones as per RFC2833/4733)
a=fmtp:18 annexb=no (G.729 no silence suppression)
a=fmtp:101 0-16 (dtmf tones default is 0-15 but if 0-16 is specified then flash break is supported which is a depreciated feature)
a=ptime:20 (this is the packet size for all codecs 20ms)
a=sendrecv (able to send and receive audio)
```

Warning Codes

RFC3261

These provide supplemental information to the status code in SIP response messages when the failure of the transaction results from an SDP problem.

Code	Description
300	Incompatible network protocol: One or more network protocols contained in the session description are not available.
301	Incompatible network address formats: One or more network address formats contained in the session description are not available
302	Incompatible transport protocol: One or more transport protocols described in the session description are not available
303	Incompatible bandwidth units: One or more media types contained in the session description are not available
304	Media type not available: One or more media types contained in the session description are not available
305	Incompatible media format: One or more media formats contained in the session description are not supported
306	Attribute not understood: One or more of the media attributes in the session description are not supported
307	Session description parameter not understood: A parameter other than those listed above was not understood
330	Multicast not available: The site where the user is located does not support multicast
331	Unicast is not available: The site where the user is located does not support unicast communication (usually due to the presence of a firewall)
370	Insufficient bandwidth: The bandwidth specified in the session description or defined by the media exceeds that known to be available
399	Miscellaneous warning: The warning text can include arbitrary information to be presented to a human user or logged.

SIP & ISUP Interworking/Mapping

The mapping of 'release and response messages' & 'SDP and User service information' between the two signalling protocols.

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mapping between ISUP USI/HLC and SIP SDP	24
mapping SIP responses to ISUP	25
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mapping ISUP cause/release to SIP-I	27
mapping SIP responses to UK ISUP	28
mapping UK ISUP cause/release to SIP	29

For more detailed information please refer to:

1. IETF RFC3398 - (ISUP to SIP Mapping)

<http://www.rfc-editor.org/info/rfc3398>

2. ITU-T Q.1912.5 - (Interworking between SIP and BICC or ISUP)

<http://www.itu.int/ITU-T/recommendations/rec.aspx?rec=6999&lang=en>

3. ETSI EN 383 001 - (ETSI TISPAN – Interworking for SIP/SIP-T (BICC & ISUP) [Q.1912.5 modified])

http://www.etsi.org/deliver/etsi_en/383000_383099/383001/01.01.01_60/en_383001v010101p.pdf

4. NICC ND1017:2006/07 - (Interworking between SIP and UK ISUP)

http://www.niccstandards.org.uk/files/current/nd1017_2006_07.pdf

mapping between SIP SDP and ISUP USI/HLC

SIP SDP			ISUP – User Service Information			
Media line (m=)	Bandwidth line (b=)	Attribute line (a=)	Transfer rate	Info Transport rate	User Info Layer 1	High-Layer Characteristics
audio rtp/avp 8	nothing or 64kbps		64kbps	3.1khz audio	G.711 a-law	
audio rtp/avp 0	64kbps	rtpmap pcmu/8000	64kbps	3.1khz audio	G.711 mu-law	
audio rtp/avp 18	as: 64 kbps	rtpmap: 18 g.729/8000	64kbps	unrestricted digital info		
audio rtp/avp dynamic PT	up to 64kbps	rtpmap clearmode/8000	64kbps	3.1khz audio		fax group 2/3
image udptl t38	up to 64kbps		64kbps	3.1khz audio		fax group 2/3
image tcptl t38	up to 64kbps		64kbps	unrestricted digital info		
audio rtp/avp dynamic PT	384kbps	rtpmap clearmode/8000	384kbps	unrestricted digital info		
audio rtp/avp dynamic PT	1472kbps	rtpmap clearmode/8000	1472kbps	unrestricted digital info		
audio rtp/avp dynamic PT	1536kbps	rtpmap clearmode/8000	1536kbps	unrestricted digital info		

mapping between ISUP USI/HLC and SIP SDP

ISUP – User Service Information				SIP SDP		
Transfer rate	Info Transport rate	User Info Layer 1	High-Layer Characteristics	Media line (m=)	Bandwidth line (b=)	Attribute line (a=)
Speech	Speech	G.711		audio rtp/avp 8	as: 64kbps	rtpmap: 8 pcma/8000
Speech	Speech	G.711		audio rtp/avp 8	as: 64kbps	rtpmap: 8 pcma/8000
3.1 Khz audio	3.1 Khz audio	G.711	Telephony	audio rtp/avp 8	as: 64kbps	rtpmap: 8 pcma/8000
3.1 Khz audio	3.1 Khz audio		fax group 2/3	image udptl t38	as: 64kbps	
3.1 Khz audio	3.1 Khz audio		fax group 2/3	image tcptl t38	as: 64kbps	
64Kbps unrestricted	Unrestricted digital info			audio rtp/avp 9	as: 64kbps	rtpmap: 9 G.722/8000
64Kbps unrestricted	Unrestricted digital info			audio rtp/avp dynamic PT	as: 64kbps	rtpmap: clearmode/8000

mapping SIP responses to ISUP

SIP response		ISUP release cause	
400	Bad request	41	Temp failure
401	Unauthorized	21	Call rejected
402	Payment required	21	Call rejected
403	Forbidden	21	Call rejected
404	Not found	1	Unallocated number
405	Method not allowed	63	Service or option unavailable
406	Not acceptable	79	Service option not implemented
407	Proxy authentication required	21	Call rejected
408	Request timeout	102	Recovery on timer expiry
410	Gone	22	Number changed
413	Request entity too long	127	Interworking
414	Request URI too long	127	Interworking
415	Unsupported media type	79	Service option not implemented
416	Unsupported URI scheme	127	Interworking
420	Bad extension	127	Interworking
421	Extension required	127	Interworking
423	Interval too brief	127	Interworking
433	Anonymity disallowed (rfc5079)	24	Call Rejected due to ACR
480	Temp unavailable	18	No user responding
481	Call / transaction does not exist	41	Temp failure
482	Loop detected	25	Exchange routing error
483	Too many hops	25	Exchange routing error
484	Address incomplete	28	Invalid number format
485	Ambiguous	1	Unallocated number
486	Busy here	17	User busy
487	Request terminated		no mapping
488	Not acceptable here		by warning header
500	Server internal error	41	Temp failure
501	Not implemented	79	Not implemented
502	Bad gateway	38	Network out of order
503	Service unavailable	41	Temp failure
504	Server time-out	102	Recovery on timer
504	Version not supported	127	Interworking
513	Message too large	127	Interworking
600	Busy everywhere	17	User busy
603	Decline	21	Call rejected
604	Does not exist anywhere	1	Unallocated number
606	Not acceptable		no mapping

mapping ISUP cause/release to SIP (SIP-T)

ISUP cause value		SIP response	
1	Unallocated number	404	Not found
2	No route to network	404	Not found
3	No route to destination	404	Not found
16	Normal call clearing	Bye or cancel	
17	User busy	486	Busy here
18	No user responding	408	Request timeout
19	No answer from the user	480	Temp unavailable
20	Subscriber absent	480	Temp unavailable
21	Call rejected	403 / 603 Forbidden	
22	Number changed	410	Gone
22	Number changed	301	Moved permanently
23	Redirection to new destination	410	Gone
24	Call rejected due to ACR	433	Anonymity disallowed (rfc5079)
26	Non-selected user clearing	404	Not found
27	Destination out of order	502	Bad gateway
28	Address incomplete	484	Address incomplete
29	Facility rejected	501	Not implemented
31	Normal unspecified	480	Temp unavailable

Resource unavailable

This kind of cause value indicates a temporary failure. Normally the gateway switch would try the next route in the list to try and terminate the call.

ISUP cause value		SIP response	
34	No circuit available	503	Service unavailable
38	Network out of order	503	Service unavailable
41	Temp failure	503	Service unavailable
42	Switch equipment congestion	503	Service unavailable
47	Resource unavailable	503	Service unavailable

Service or option not available

This kind of cause value indicates that there is a problem with the request.

ISUP cause value		SIP response	
55	Incoming calls barred with CUG	403	forbidden
57	Bearer capability not authorised	403	forbidden
58	Bearer capability not presently available	503	Service unavailable
65	Bearer capability no implemented	488	Not acceptable here
70	Only restricted digital available	488	Not acceptable here
79	Service or option not implemented	501	Not implemented
87	User not member of CUG	403	forbidden
88	Incompatible destination	503	Service unavailable
55	Incoming calls barred with CUG	403	forbidden
57	Bearer capability not authorised	403	forbidden
58	Bearer capability not presently available	503	Service unavailable
65	Bearer capability not implemented	488	Not Acceptable here
70	Only restricted digital available	488	Not Acceptable here
79	Service or option not implemented	501	Not implemented

ISUP cause value		SIP response	
87	User not member of CUG	403	forbidden
88	Incompatible destination	503	Service unavailable
90	Non-existent CUG		
91	Invalid transit network selection		
95	Invalid message, unspecified		
97	Message type non-existent or not implemented		
99	Information element non-existent		
102	Recovery of timer expiry	504	Gateway timeout
111	Protocol error	500	Server internal error
127	Interworking unspecified	500	Server internal error

mapping ISUP cause/release to SIP-I

ISUP cause value		SIP-I response	
1	Unallocated number	404	Not found
2	No route to network	500	Server internal error
3	No route to destination	500	Server internal error
4	Send special info tone	500	Server internal error
5	Misdialled trunk prefix	404	Not found
8	Pre-emption	500	Server internal error
9	Pre-emption circuit res.	500	Server internal error
16	Normal call clearing	480	Temporarily unavailable
17	User busy	486	Busy here
18	No user responding	480	Temporarily unavailable
19	No answer from the user	480	Temporarily unavailable
20	Subscriber absent	480	Temporarily unavailable
21	Call rejected	480	Temporarily unavailable
22	Number changed	410	Gone
23	Redirect to new dest.		Not mapped
25	Exchange routing error	480	Temporarily unavailable
27	Destination out of order	502	Bad gateway
28	Address incomplete	484	Address incomplete
29	Facility rejected	500	Server internal error
31	Normal unspecified	480	Temporarily unavailable
34	No circuit available	480	Temporarily unavailable
38	Network out of order	500	Server internal error
41	Temp failure	500	Server internal error
42	Switch equip congestion	500	Server internal error
47	Resource unavailable	500	Server internal error
55	Incoming calls barred with CUG	500	Server internal error
57	Bearer capability not authorised	500	Server internal error
58	Bearer capability not presently available	500	Server internal error
65	Bearer capability not implemented	500	Server internal error
70	Only restricted digital available	500	Server internal error
79	Service or option not implemented	500	Server internal error
87	User not member of CUG	500	Server internal error
88	Incompatible destination	500	Server internal error
90	Non-existent CUG		
91	Invalid transit network selection		
95	Invalid message, unspecified		
97	Message type non-existent or not implemented		
99	Information element non-existent		
102	Recovery of timer expiry	480	Temporarily unavailable
111	Protocol error	500	Server internal error
127	Interworking unspecified	480	Temporarily unavailable

mapping SIP responses to UK ISUP

SIP response	ISUP release cause
400 Bad request	95 Invalid message, unspecified
401 Unauthorized	63 Service or option not available, unspecified
402 Payment required	63 Service or option not available, unspecified
403 Forbidden	63 Service or option not available, unspecified
404 Not found	1 Unallocated (unassigned) number
405 Method not allowed	63 Service or option not available, unspecified
406 Not acceptable	79 Service or option not implemented, unspecified
407 Proxy authentication required	63 Service or option not available, unspecified
408 Request timeout	18 No user responding
410 Gone	22 Number changed
413 Request entity too long	111 Protocol error, unspecified
414 Request URI too long	111 Protocol error, unspecified
415 Unsupported media type	79 Service or option not implemented, unspecified
416 Unsupported URI scheme	127 Interworking, unspecified
420 Bad extension	79 Service or option not implemented, unspecified
421 Extension required	79 Service or option not implemented, unspecified
423 Interval too brief	63 Service or option not available, unspecified
433 Anonymity disallowed (rfc5079)	24 Call rejected due to ACR supplementary service
480 Temp unavailable	31 Normal, unspecified
481 Call / transaction does not exist	95 Invalid message, unspecified
482 Loop detected	25 Exchange routing error
483 Too many hops	25 Exchange routing error
484 Address incomplete	28 Invalid number format (address incomplete)
485 Ambiguous	1 Unallocated (unassigned) number
486 Busy here	34 No circuit/channel available
487 Request terminated	31 Normal, unspecified
488 Not Acceptable Here	79 Service or option not implemented, unspecified
491 Request Pending	No mapping
493 Undecipherable	127 Interworking, unspecified
500 Server internal error	47 Resource unavailable, unspecified
501 Not implemented	79 Service or option not implemented, unspecified
502 Bad gateway	111 Protocol error, unspecified
503 Service unavailable	42 Switching equipment congestion
504 Server time-out	102 Recovery on timer expiry
505 Version not supported	127 Interworking, unspecified
513 Message too large	111 Protocol error, unspecified
580 Precondition Failure	34 No circuit/channel available
600 Busy everywhere	17 User busy
603 Decline	21 Call rejected
604 Does not exist anywhere	4 Send special information tone
606 Not acceptable	79 Service or option not implemented, unspecified

mapping UK ISUP cause/release to SIP

ISUP cause value		SIP response	
1	Unallocated number	404	Not found
2	No route to network	404	Not found
3	No route to destination	404	Not found
4	Send special information tone	604	Does not exist anywhere
5	Misdialled trunk prefix (national use)	404	Not found
8	Pre-emption	480	Temporarily unavailable
9	Pre-emption – circuit reserved for reuse	480	Temporarily unavailable
16	Normal call clearing	480	Temporarily unavailable
17	User busy	600	Busy Everywhere
18	No user responding	408	Request timeout
19	No answer from the user	480	Temp unavailable
20	Subscriber absent	480	Temp unavailable
21	Call rejected	603	Decline
22	Number changed	410	Gone
23	Redirection to new destination	302	Moved temporarily
24	Call rejected due to ACR	433	Anonymity disallowed (rfc5079)
25	Exchange Routing Error	483	Too many hops
27	Destination out of order	480	Temp unavailable
28	Address incomplete	484	Address incomplete
29	Facility rejected	403	Forbidden
31	Normal unspecified	480	Temp unavailable
34	No circuit available	600	Busy Everywhere if "Location = User"
34	No circuit available	486	Busy Here if "Location is other"
38	Network out of order	500	Server Internal Error
41	Temp failure	500	Server Internal Error
42	Switch equipment congestion	503	Service unavailable
43	Access information discarded	500	Server Internal Error
44	Requested circuit/channel not available	500	Server Internal Error
46	Precedence call blocked	500	Server Internal Error
47	Resource unavailable	500	Server Internal Error
50	Requested Facility not subscribed	403	Forbidden
53	Outgoing calls barred with CUG	403	Forbidden
55	Incoming calls barred with CUG	403	Forbidden
57	Bearer capability not authorised	488	Not acceptable here
58	Bearer capability not presently available	503	Service unavailable
62	Inconsistency in designated outgoing access information and subscriber class	403	Forbidden
63	Service or option not available, unspecified	403	Forbidden
65	Bearer capability no implemented	501	Not implemented
69	Requested facility not implemented	501	Not implemented
70	Only restricted digital available	488	Not acceptable here
79	Service or option not implemented	501	Not implemented
87	User not member of CUG	403	Forbidden
88	Incompatible destination	488	Not acceptable here
90	Non-existent CUG	404	Not found
91	Invalid transit network selection	404	Not found
95	Invalid message, unspecified	502	Bad gateway
97	Message type non-existent or not implemented	502	Bad gateway
99	Information element non-existent	502	Bad gateway
102	Recovery of timer expiry	504	Server timeout
103	Parameter non-existent or not implemented	502	Bad gateway
110	Message with unrecognised parameter	502	Bad gateway
111	Protocol error, unspecified	502	Server internal error
127	Interworking unspecified	502	Server internal error

Example call flows

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SIP to ISUP normal call flow

SIP	MGC/MG	PSTN
1 -----INVITE----->		
<-----100-----		
	-----IAM----->	
	<-----ACM-----	3
	<=====Ringing=====	
4 <-----18x-----		
<=====Ringing=====		
	<-----CPG-----	5
6 <-----18x-----		
	<-----ANM-----	7
	<=====2way Audio=====>	
8 <-----200-----		
<=====2way Audio=====>		
9 -----ACK----->		

1. When a SIP user wishes to begin a session with a PSTN user, the SIP node issues an INVITE request.
2. Upon receipt of an INVITE request, the gateway maps it to an IAM message and sends it to the ISUP network.
3. The remote ISUP node indicates that the address is sufficient to set up a call by sending back an ACM message.
4. The 'called party status' code in the ACM message is mapped to a provisional response () and returned to the SIP node. This response may contain SDP to establish an early media stream (as shown in the diagram). If no SDP is present, the audio will be established in both directions after step 8.
5. If the ISUP variant permits, the remote ISUP node may issue a variety of Call Progress (CPG) messages to indicate, for example that the call is being forwarded.
6. Upon receipt of a CPG message, the gateway will map the event code to a SIP provisional response and send it to the SIP node.
7. Once the PSTN user answers, an Answer (ANM) message will be sent to the gateway.
8. Upon receipt of the ANM, the gateway will send a 200 message to the SIP node.
9. The SIP node, upon receiving an INVITE final response (200), will send an ACK to acknowledge receipt.

ISUP setup failure

SIP	MGC/MG	PSTN
1 -----INVITE----->		
<-----100-----		
	-----IAM----->	2
	<-----REL-----	3
	-----RLC----->	4
5 <-----4xx+-----		
6 -----ACK----->		

1. When a SIP user wishes to begin a session with a PSTN user, the SIP node issues an INVITE request.
2. Upon receipt of an INVITE request, the gateway maps it to an IAM (initial address message) message and sends it to the ISUP network.
3. Since the remote ISUP node is unable to complete the call, it will send a REL (release)
4. The gateway releases the circuit and confirms that it is available for reuse by sending an RLC (release complete)
5. The gateway translates the cause code in the REL to a SIP error response and sends it to the SIP node.
6. The SIP node sends an ACK to acknowledge receipt of the INVITE final response.

SIP to SIP normal call flow

```

SIPUA                SIP Server                SIPUA
1 |-----INVITE----->|                    |
2 |<-----100-----|                    |
  |                    |-----INVITE----->| 3
  |                    |<-----100-----|
  |                    |<-----18X-----| 4
  |<-----18X-----|                    |
  |                    |<=====Ringing=====|
  |                    |                    |
  |                    |<-----200-----| 5
  |<-----200-----|                    |
6 |-----ACK----->|                    |
  |                    |-----ACK----->|
  |<=====2way Audio=====| 7
  |                    |                    |
8 |-----BYE----->|                    |
  |                    |-----BYE----->|
  |                    |<-----200-----| 9
  |<-----200-----|                    |
  |                    |                    |

```

1. SIP UA (user agent) initiates a call and sends out an invite to the SIP server.
2. Upon receipt of invite sip server issues a 100 trying prior to sending out invite to distant SIP UA
3. invite sent out to setup session to called SIP UA.
4. a 18x sent back usually a 180 ringing with SDP (ringtone is now heard by the caller)
5. The called sip ua answers the call and a 200 ok is sent. (if no SDP was sent in the 18x than it will be sent here in the 200)
6. The caller SIP UA acknowledges the 200 OK with an ACK message.
7. Two way speech is now established.
8. The caller ends the call and the sip ua sends out a BYE message.
9. Called sip ua acknowledges the bye with a 200 ok.

The specifications and material listed herein is for informational purposes only and subject to change without notification. Please refer to the latest RFC documentation at <http://www.rfc-editor.org/>