**3GPP TSG CN Plenary**
**Meeting #10, Bangkok, Thailand**
**6th – 8th December 2000**

**Tdoc NP-000684**

**3GPP TSG-CN-WG1, Meeting #14**
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**Agenda Item:** 8.2

**WI / Topic:** SIP call control protocol for the IM subsystem

**Source:** Lucent Technologies

**Title:** Proposed scope and contents for IP multimedia subsystem stage 3

**Effected Specifications / Releases:** 24.229

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This document contains the output of the drafting group CN1 ~~SIP ad-hoc #1~~ # 14 held ~~October~~ November 2000 in ~~Sophia Antipolis~~Cardiff, and previously contained in N1-~~001115~~001190.

The only amendment made to this version are to incorporate the latest style and template changes.

# 3GPP TS 24.229 V0.0.7~~6~~ (2000-11)

*Technical Specification*

## 3rd Generation Partnership Project;
## Technical Specification Group Core Network;
## IP Multimedia Call Control Protocol based on SIP and SDP

## (Release 5)

GLOBAL SYSTEM FOR
MOBILE COMMUNICATIONS

*Remove GSM logo from the cover page for pure 3rd Generation documents.*

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Keywords

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# Contents

# Foreword

This Technical Specification has been produced by the 3rd Generation Partnership Project (3GPP).

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of the present document, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

Version x.y.z

where:

x the first digit:

1 presented to TSG for information;

2 presented to TSG for approval;

3 or greater indicates TSG approved document under change control.

y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

z the third digit is incremented when editorial only changes have been incorporated in the document.

# Introduction

*This clause is optional. If it exists, it is always the third unnumbered clause.*

# 1 Scope

The present document defines a call control protocol for use in the IP multimedia (IM) Core Network (CN) subsystem based on the Session Initiation Protocol (SIP), and the associated Session Description Protocol (SDP).

The present document is applicable to:

- the interface between the User Equipment (UE) and the Call State Control Function (CSCF);
- the interface between the CSCF and any other CSCF;
- the interface between the CSCF and the Media Gateway Control Function (MGCF); and
- the interface between the CSCF and an external Multimedia IP network.

Where possible this document specifies the requirements for this protocol by reference to specifications produced by the IETF within the scope of SIP and SDP. Where this is not possible, extensions to SIP and SDP are defined within this document. The document has therefore been structured in order to allow both forms of specification.

Editor's note: Noted CN1 SIP ad-hoc #1: The aim of this document is to provide a framework for the documentation of 3GPP agreements concerning the use of SIP, in a form where the final documentation can be readily be drafted without further discussion, once the required form of that documentation is known (i.e. integration into IETF publications, separate 3GPP specifications, etc.) The drafting is therefore performed as a technical specification, and in accordance with the 3GPP drafting rules, but the final publication may be one or more specifications which may only contain some of this material. Decisions on this can only be made once the relationship to IETF has been ascertained.

# 2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.

This specification may contain references to pre-Release-4 GSM specifications. These references shall be taken to refer to the Release 5 version where that version exists. Conversion from the pre-Release-4 number to the Release 4 (onwards) number is given in subclause 6.1 of 3GPP TR 41.001.

[1] draft-ietf-sip-rfc2543bis-~~01~~ 02 (~~August~~ September 2000): "SIP: Session Initiation Protocol".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[2] ~~draft-ietf-sip-info-method-05~~RFC 2976 (~~January~~ October ~~2001~~2000): "The SIP INFO method".

~~Editor's note: The above document cannot be formally referenced until it is published as an RFC. Currently it has received IETF approval for this.~~

[3] draft-ietf-sip-100rel-02 (June 2000): "Reliability of provisional responses in SIP".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[4] draft-ietf-sip-callerprefs-02 (July 2000): "SIP caller preferences and callee capabilities".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[5] draft-ietf-sip-cc-transfer-01 (September 2000): "SIP call control transfer".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[6] draft-ietf-sip-serverfeatures-02 (March 2000): "The SIP supported header".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[7] draft-ietf-sip-session-timer-~~02~~ 03 (~~July~~ October 2000): "The SIP session timer".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[8] draft-sip-manyfolks-~~sip~~-resource-~~01~~ 00 (~~June~~ November 2000): "Integration of resource management and SIP".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[9] draft-~~dcsgroup~~-sip-privacy-~~02~~ 00 (~~June~~ November 2000): "SIP extensions for caller identity and privacy".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[10] draft-~~dcsgroup~~-sip-state-~~02~~ 00 (~~July~~ November 2000): "SIP extensions for supporting distributed call state".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[11] draft-~~dcsgroup~~-sip-call-auth-~~02~~ 00 (~~July~~ November 2000): "SIP extensions for media authorization".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[12] RFC 2327 (April 1998): "SDP: Session Description Protocol".

# 3 Definitions, symbols and abbreviations

## 3.1 Definitions

For the purposes of the present document, the [following] terms and definitions [given in ... and the following] apply.

For the purposes of the present document, the following terms and definitions apply.

**Header:**

Editor's note: To be provided.

**Method:**

Editor's note: To be provided.

**Option tag:** Option tags are unique identifiers used to designate new options in SIP. These tags are used in Require, Supported and Unsupported header fields.

Editor's note: Text extracted from RFC2543bis, but not specified as a definition.

**Stateful proxy:** When stateful, a proxy remembers the incoming request which generated outgoing requests, and the out-going requests.

Editor's note: The above definition requires enhancement. No definition in RFC2543bis.

**Stateless proxy:** A stateless proxy forgets all information once an outgoing request is generated.

Editor's note: The above definition requires enhancement. No definition in RFC2543bis.

**Status code:** The Status-Code is a 3-digit integer result code that indicates the outcome of the attempt to understand and satisfy the request.

Editor's note: Text extracted from RFC2543bis, but not specified as a definition.

For the purposes of the present document, the following terms and definitions given in RFC 2543bis [1] *(Editor's note – working title)* apply.

Editor's note:Noted by CN1 SIP ad-hoc #1: The full definitions are included below for information purposes in the preparation of this specification. They will ultimately be deleted, leaving only the list of terms.

**Client**: An application program that sends SIP requests. Clients may or may not interact directly with a human user. User agents and proxies contain clients (and servers).

**Proxy, proxy server**: An intermediary program that acts as both a server and a client for the purpose of making requests on behalf of other clients. Requests are serviced internally or by passing them on, possibly after translation, to other servers. A proxy interprets, and, if necessary, rewrites a request message before forwarding it.

**Redirect server**: A redirect server is a server that accepts a SIP request, maps the address into zero or more new addresses and returns these addresses to the client. Unlike a proxy server , it does not initiate its own SIP request. Unlike a user agent server , it does not accept calls.

**Registrar**: A registrar is a server that accepts REGISTER requests. A registrar is typically co-located with a proxy or redirect server and MAY make its information available through the location server.

**Server**: A server is an application program that accepts requests in order to service requests and sends back responses to those requests. Servers are either proxy, redirect or user agent servers or registrars.

**Session**: From the SDP specification: "A multimedia session is a set of multimedia senders and receivers and the data streams flowing from senders to receivers. A multimedia conference is an example of a multimedia session." (RFC 2327 [6]) (A session as defined for SDP can comprise one or more RTP sessions.) As defined, a callee can be invited several times, by different calls, to the same session. If SDP is used, a session is defined by the concatenation of the user name, session id, network type, address type and address elements in the origin field.

**User agent client (UAC)**: A user agent client is a client application that initiates a SIP request.

**User agent server (UAS)**: A user agent server is a server application that contacts the user when a SIP request is received and that returns a response on behalf of the user. The response accepts, rejects or redirects the request.

**User agent (UA)**: An application which can act both as a user agent client and user agent server.

## 3.2 Symbols

For the purposes of the present document, the following symbols apply:

*Symbol format*

<symbol> <Explanation>

## 3.3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

| | |
|---|---|
| CN | Core Network |
| CSCF | Call State Control Function |
| IM | IP Multimedia |
| IP | Internet Protocol |
| MGCF | Media Gateway Control Function |
| SDP | Session Description Protocol |
| SIP | Session Initiation Protocol |
| UA | User Agent |
| UE | User Equipment |
| UAC | User Agent Client |
| UAS | User Agent Server |

# 4 General

The Session Initiation Protocol (SIP) as specified in RFC 2543bis [1] (Editor's note – working name) shall apply ~~within the context of~~ except as modified by the defined support of the methods specified in clause 5 of this specification and the defined support of the headers specified in clause 6 of this specification. Unless otherwise specified, the procedures (including error handling and unrecognized information handling) are unmodified.

Editor's note: Noted by CN1 SIP ad-hoc #1: There are two methods of documenting the requirements where the main specification is dealt with elsewhere.

- By endorsement with modification. This identifies how the existing text is modified to create the new specification on a paragraph by paragraph basis. This could result in voluminous documentation, particularly if the IETF text is opened up to correct some of the drafting faults. It also does nothing to improve the structure of the documentation of the protocol.

- By creation of profile proforma to an existing PICS proforma for the SIP specification. But no PICS proforma exists for the SIP specification.

The suggestion in this document is possibly a combination of the two methods above, retaining flexibility

Editor's note: Agreed by CN1 SIP ad-hoc #1: It is proposed that the discussion should be based on the RFC2543bis [1] draft currently still under preparation within the IETF. The reason for doing this is that this draft corrects many errors within the existing RFC 2543, and to document the deviations at this point to the approved RFC would lead to 3GPP rediscussing the issues that have already been resolved in the IETF SIP working group. If, due to delays in the approval of the RFC2543bis [1] [1] draft, it becomes necessary to refer to RFC2543, the documentation of the required differences is an editorial exercise, although the differences are substantial.

Editor's note: This clause should identify the correspondence between SIP logical entities and 3GPP physical entities.

# 5 SIP methods

## 5.1 SIP methods defined by the IETF

Editor's note: Need to formulate a title to this clause that does not include an organisation name.

Editor's note: This clause is to be used to either endorse, or endorse with modification, those SIP methods that have already been documented by the IETF SIP working group.

### 5.1.1 Introduction

Support for a method also requires support for the associated procedures associated with that sending and receipt of that method.

### 5.1.2 INVITE method

The INVITE method, as specified in RFC2543bis [1] *(Editor's note – working title)* clause 4.2.1 shall apply. The INVITE method indicates that the user or service is being invited to participate in a session.

Editor's note: RFC2543bis [1] specifies the following: "This method MUST be supported by SIP proxy, redirect and user agent servers as well as clients." and also "Proxy and redirect servers treat all methods other than INVITE and CANCEL in the same way, by forwarding them accordingly. Thus, no method-specific support is required in these servers for methods other than INVITE and CANCEL." Under what circumstances would a 3GPP entity not support the INVITE method? How should this be represented in this specification?

### 5.1.3 ACK method

The ACK method, as specified in RFC2543bis [1] *(Editor's note – working title)* clause 4.2.2 shall apply. The ACK request confirms that the client has received a final response to an INVITE request. (ACK is used only with INVITE requests.)

Editor's note: RFC2543bis [1] specifies the following: "This method MUST be supported by SIP user agents." and also "Proxy and redirect servers treat all methods other than INVITE and CANCEL in the same way, by forwarding them accordingly. Thus, no method-specific support is required in these servers for methods other than INVITE and CANCEL." Under what circumstances would a 3GPP entity not support the ACK method? How should this be represented in this specification?

### 5.1.4 OPTIONS method

The OPTIONS method, as specified in RFC2543bis [1] *(Editor's note – working title)* clause 4.2.1 shall apply. The server is being queried as to its capabilities.

Editor's note: RFC2543bis [1] specifies the following: "This method MUST be supported by SIP user agents and registrars." and also "Proxy and redirect servers treat all methods other than INVITE and CANCEL in the same way, by forwarding them accordingly. Thus, no method-specific support is required in these servers for methods other than INVITE and CANCEL." Under what circumstances would a 3GPP entity not support the OPTIONS method? How should this be represented in this specification?

### 5.1.5 BYE method

The BYE method, as specified in RFC2543bis [1] *(Editor's note – working title)* clause 4.2.1 shall apply. The user agent client uses BYE to indicate to the server that it wishes to release the call.

Editor's note: RFC2543bis [1] specifies the following: "This method SHOULD be supported by user agent servers." and also "Proxy and redirect servers treat all methods other than INVITE and CANCEL in the same way, by forwarding them accordingly. Thus, no method-specific support is required in these servers for methods other than INVITE and CANCEL." Under what circumstances would a 3GPP entity not support the BYE method? How should this be represented in this specification?

### 5.1.6 CANCEL method

The CANCEL method, as specified in RFC2543bis [1] *(Editor's note – working title)* clause 4.2.1 shall apply. The CANCEL request cancels a pending request with the same Call-ID, To, From and CSeq (sequence number only) header field values, but does not affect a completed request or existing calls. (A request is considered completed if the server has returned a final status response.)

Editor's note: RFC2543bis [1] specifies the following: "This method MUST be supported by proxy servers and SHOULD be supported by all other SIP server types." and also "Proxy and redirect servers treat all methods other than INVITE and CANCEL in the same way, by forwarding them accordingly. Thus, no method-specific support is required in these servers for methods other than INVITE and CANCEL." Under what circumstances would a 3GPP entity not support the CANCEL method? How should this be represented in this specification?

### 5.1.7 REGISTER method

The REGISTER method, as specified in RFC2543bis [1] *(Editor's note – working title)* clause 4.2.1 shall apply. A client uses the REGISTER method to register the address listed in the To header field with a SIP server. A user agent SHOULD register with a local server on startup and periodically thereafter by sending a REGISTER request. The period is given by the expiration time indicated in the registration response. It is RECOMMENDED that the UA registers via multicast and send a registration to its "home" address, i.e., the server for the domain that it uses as its From address in outgoing requests.

Editor’s note: RFC2543bis [1] specifies the following: “Support of this method is RECOMMENDED; registrars MUST support it.” and also “Proxy and redirect servers treat all methods other than INVITE and CANCEL in the same way, by forwarding them accordingly. Thus, no method-specific support is required in these servers for methods other than INVITE and CANCEL.” Under what circumstances would a 3GPP entity not support the REGISTER method? How should this be represented in this specification?

### 5.1.8 INFO method

The INFO method, as specified in draft-ietf-sip-info-method-05 [2] *(Editor’s note – working title)* clause 2 shall apply. The purpose of the INFO message is to carry mid-session information between SIP user agents. This information will generally be carried in message bodies, although it can be carried in headers in the INFO message.

Editor’s note: draft-ietf-sip-info-method-05 specifies this method separately to the main specification. Therefore support at both a sending and receiving SIP entity is presumably optional. Under what circumstances would a 3GPP entity not support the INFO method? How should this be represented in this specification?

### 5.1.9 PRACK method

The PRACK method, as specified in draft-ietf-sip-100rel-02 [3] *(Editor’s note – working title)* shall apply. For a provisional response which is to be sent reliably, the UAC creates a new request, with a method of PRACK, used to acknowledge the provisional response. The PRACK request is like any other non-INVITE request sent within a call. The PRACK contains a header, called RAck, which contains the value of the RSeq header for the provisional response being acknowledged. The RAck header also contains the contents of the CSeq field in the response being acknowledged. The combination of Call-ID, CSeq, and RAck allow the PRACK request to be matched to a specific provisonal response within a specific transaction within a specific call. When the UAS receives the PRACK request, it knows that the provisional response has been received.

Editor’s note: draft-ietf-sip-100rel-02 specifies this method separately to the main specification. Therefore support at both a sending and receiving SIP entity is optional. Under what circumstances would a 3GPP entity not support the PRACK method? How should this be represented in this specification?

### 5.1.10 REFER method

The REFER method, as specified in draft-ietf-sip-cc-transfer-01 [5] *(Editor’s note – working title)* clause 3 shall apply. The REFER method indicates that the recipient should contact a third party using the contact information provided in the method. A success response indicates that the recipient was able to contact the third party. The REFER method follows the session's current signaling path. In particular, the Request-URI of the REFER method identifies the recipient.

Editor’s note: draft-ietf-sip-cc-transfer-01 specifies this method separately to the main specification. Therefore support at both a sending and receiving SIP entity is optional. Under what circumstances would a 3GPP entity not support the REFER method? How should this be represented in this specification?

### 5.1.11 COMET method

The COMET method, as specified in draft-manyfolks-sip-resource-01 [8] (Editor’s note – working title) clause 5 shall apply. The COMET method is used for communicating successful completion of preconditions from the calling to called user agents. The signaling path for the COMET method is the signaling path established as a result of the call setup. This can be either direct signaling between the calling and called user agents or a signaling path involving SIP proxy servers that were involved in the call setup and added themselves to the Record-Route header on the initial INVITE message. The precondition information is communicated in the message body, which contains an SDP. For every agreed precondition, the strength-tag indicates "success" or "failure".

Editor’s note: draft-manyfolks-sip-resource-01 specifies this method separately to the main specification. Therefore support at both a sending and receiving SIP entity is optional. Under what circumstances would a 3GPP entity not support the COMET method? How should this be represented in this specification?

Editor’s note: The referenced document does not contain the formal ABNF notation, this implying that a separate document is envisaged to specify this method. This understanding is further obtained by the fact that the current document is a mixture of SIP enhancements and SDP enhancements.

## 5.2 SIP methods defined within this specification

Editor's note: This clause is to be used to document those SIP methods that require complete documentation within this specification, because they are not defined elsewhere in a form that can be endorsed, either with or without modification.

# 6 SIP headers

## 6.1 SIP headers defined by the IETF

Editor's note: Need to formulate a title to this clause that does not include an organisation name.

Editor's note: This clause is to be used to either endorse, or endorse with modification, those SIP headers that have already been documented by the IETF SIP working group.

### 6.1.1 Introduction

Support for a header also requires support for the associated procedures associated with that sending and receipt of that header.

### 6.1.2 General headers

General header fields apply to both request and response messages. The "general-header" field names can be extended reliably only in combination with a change in the protocol version. However, new or experimental header fields MAY be given the semantics of general header fields if all parties in the communication recognize them to be "general-header" fields.

General headers shall apply as defined in table 1.

**Table 1: General headers**

| Header | Reference | Support and notes |
|---|---|---|
| Accept | RFC2543bis [1] clause 6.6 | |
| Accept-Encoding | RFC2543bis [1] clause 6.7 | |
| Accept-Language | RFC2543bis [1] clause 6.8 | |
| Anonymity | draft-dcsgroup-sip-privacy-02 [9] clause 5.2 | |
| Call-ID | RFC2543bis [1] clause 6.12 | |
| Contact | RFC2543bis [1] clause 6.14 | |
| Cseq | RFC2543bis [1] clause 6.20 | |
| Date | RFC2543bis [1] clause 6.21 | |
| Encryption | RFC2543bis [1] clause 6.22 | |
| From | RFC2543bis [1] clause 6.25~~4~~ | |
| Media-Auth-Token | draft-dcsgroup-sip-call-auth-02 [11] clause 5.1 | |
| MIME-Version | RFC2543bis [1] clause 6.28 | |
| Organization | RFC2543bis [1] clause 6.29 | |
| Record-Route | RFC2543bis [1] clause 6.34 | |
| Remote-Party-ID | draft-dcsgroup-sip-privacy-02 [9] clause 5.1 | |
| Require | RFC2543bis [1] clause 6.35 | |
| Session-expires | draft-ietf-sip-session-timer-02 [7] clause 3 | |
| State | draft-dcsgroup-sip-state-02 [10] clause 5.1 | |
| Supported | RFC2543bis [1] clause 6.41 or draft-ietf-sip-serverfeatures-02 [6] clause 2.1 | |
| Timestamp | RFC2543bis [1] clause 6.42 | |
| To | RFC2543bis [1] clause 6.43 | |
| User-Agent | RFC2543bis [1] clause 6.45 | |
| Via | RFC2543bis [1] clause 6.46 | |

## 6.1.3 Entity headers

The "entity-header" fields define meta-information about the message-body or, if no body is present, about the resource identified by the request. The term "entity header" is an HTTP 1.1 term where the response body can contain a transformed version of the message body. The original message body is referred to as the "entity". We retain the same terminology for header fields but usually refer to the "message body" rather then the entity as the two are the same in SIP.

Entity headers shall apply as defined in table 2.

**Table 2: Entity headers**

| Header | Reference | Support and notes |
|---|---|---|
| Allow | RFC2543bis [1] clause 6.10 | |
| Content-Disposition | RFC2543bis [1] clause 6.15 | |
| Content-Encoding | RFC2543bis [1] clause 6.16 | |
| Content-Language | RFC2543bis [1] clause 6.17 | |
| Content-Length | RFC2543bis [1] clause 6.18 | |
| Content-Type | RFC2543bis [1] clause 6.19 | |
| Expires | RFC2543bis [1] clause 6.24~~3~~ | |

## 6.1.4 Request headers

The "request-header" fields allow the client to pass additional information about the request, and about the client itself, to the server. These fields act as request modifiers, with semantics equivalent to the parameters of a programming language method invocation.

The "request-header" field names can be extended reliably only in combination with a change in the protocol version. However, new or experimental header fields MAY be given the semantics of "request-header" fields if all parties in the communication recognize them to be request-header fields. Unrecognized header fields are treated as "entity-header" fields.

Request headers shall apply as defined in table 3.

**Table 3: Request headers**

| Header | Reference | Support and notes |
|---|---|---|
| Accept-Contact | draft-ietf-sip-callerprefs-02 [4] clause 5.2 | |
| Alert-Info | RFC2543bis [1] clause 6.9 | |
| Authorization | RFC2543bis [1] clause 6.11 | |
| Hide | ~~RFC2543bis [1] clause 6.25~~ | Editor's note: Use of this header is now deprecated by the SIP working group and it will be removed from the next version of the draft. |
| In-Reply-To | RFC2543bis [1] clause 6.26 | |
| Max-Forwards | RFC2543bis [1] clause 6.27 | |
| Priority | RFC2543bis [1] clause 6.30 | |
| Proxy-Authorization | RFC2543bis [1] clause 6.32 | |
| Proxy-Require | RFC2543bis [1] clause 6.33 | |
| Rack | draft-ietf-sip-100rel-02 [3] clause 4 | |
| Refer-to | draft-ietf-sip-cc-transfer-01 [5] clause 3.1 | |
| Referred-by | draft-ietf-sip-cc-transfer-01 [5] clause 3.2 | |
| Request-Disposition | draft-ietf-sip-callerprefs-02 [4] clause 5.5 | |
| Reject-Contact | draft-ietf-sip-callerprefs-02 [4] clause 5.3 | |
| Route | RFC2543bis [1] clause 6.38 | |
| Response-Key | RFC2543bis [1] clause 6.36 | |
| Subject | RFC2543bis [1] clause 6.40 | |

## 6.1.5 Response headers

The "response-header" fields allow the server to pass additional information about the response which cannot be placed in the Status-Line. These header fields give information about the server and about further access to the resource identified by the Request-URI.

Response-header field names can be extended reliably only in combination with a change in the protocol version. However, new or experimental header fields MAY be given the semantics of "response-header" fields if all parties in the communication recognize them to be "response-header" fields. Unrecognized header fields are treated as "entity-header" fields.

Response headers shall apply as defined in table 4.

**Table 4: Response headers**

| Header | Reference | Support and notes |
|---|---|---|
| Error-info | RFC2543bis [1] clause 6.23 | |
| Proxy-Authenticate | RFC2543bis [1] clause 6.31 | |
| Retry-After | RFC2543bis [1] clause 6.37 | |
| Rseq | draft-ietf-sip-100rel-02 [3] clause 4 | |
| Server | RFC2543bis [1] clause 6.39 | |
| Unsupported | RFC2543bis [1] clause 6.44 | |
| Warning | RFC2543bis [1] clause 6.47 | |
| WWW-Authenticate | RFC2543bis [1] clause 6.48 | |

## 6.2 SIP headers defined within this specification

Editor's note: This clause is to be used to document those SIP headers that require complete documentation within this specification, because they are not defined elsewhere in a form that can be endorsed, either with or without modification.

# 7 Option tags

## 7.1 Option tags defined by the IETF

Editor's note: Need to formulate a title to this clause that does not include an organisation name.

Editor's note: This clause is to be used to either endorse, or endorse with modification, those option tags that have already been documented by the IETF SIP working group.

Option tags, and the requirements indicated by use of those option tags, shall apply as defined in table 5.

Support for an option tag also requires support for the associated procedures associated with that sending and receipt of that option tag.

**Table 5: Option tags**

| Option tag | Reference | Support and notes |
|---|---|---|
| 100rel | draft-ietf-sip-100rel-02 [3] clause 4 | |
| Pref | draft-ietf-sip-callerprefs-02 [4] clause 6.1 and clause 6.2 | |
| Privacy | draft-dcsgroup-sip-privacy-02 [9] clause 6.1 | |
| State | draft-dcsgroup-sip-state-02 [10] clause 3 and clause 5 | |
| Timer | draft-ietf-sip-session-timer-02 [7] clause 2 | |

## 7.2 Option tags defined within this specification

Editor's note: This clause is to be used to document those option tags that require complete documentation within this specification, because they are not defined elsewhere in a form that can be endorsed, either with or without modification.

# 8 Status codes

## 8.1 Status codes defined by the IETF

Editor's note: Need to formulate a title to this clause that does not include an organisation name.

Editor's note: This clause is to be used to either endorse, or endorse with modification, those status codes that have already been documented by the IETF SIP working group.

### 8.1.1 Introduction

The first line of a Response message is the Status-Line, consisting of the protocol version followed by a numeric Status-Code and its associated textual phrase.

The Status-Code is a 3-digit integer result code that indicates the outcome of the attempt to understand and satisfy the request. The Reason-Phrase is intended to give a short textual description of the Status-Code. The Status-Code is intended for use by automata, whereas the Reason-Phrase is intended for the human user. The client is not required to examine or display the Reason-Phrase.

**Table 6: Classes of status-code**

| Class digit | Class of response | Definition of class | Required supported defined in table |
|---|---|---|---|
| 1xx | Informational | request received, continuing to process the request | Table 7 |
| 2xx | Success | the action was successfully received, understood, and accepted | Table 8 |
| 3xx | Redirection | further action needs to be taken in order to complete the request | Table 9 |
| 4xx | Client Error | the request contains bad syntax or cannot be fulfilled at this server | Table 10 |
| 5xx | Server Error | the server failed to fulfil an apparently valid request | Table 11 |
| 6xx | Global Failure | the request cannot be fulfilled at any server | Table 12 |

Support for a status-code also requires support for the associated procedures associated with that sending and receipt of that status-code.

### 8.1.2 1xx: request received, continuing to process the request

Status-codes of the informational class shall apply as defined in table 7.

**Table 7: Informational status-codes**

| Status code | Reference | Support and notes |
|---|---|---|
| "100" Trying | RFC2543bis [1] clause 7.1.1 | |
| "180" Ringing | RFC2543bis [1] clause 7.1.2 | |
| "181" Call Is Being Forwarded | RFC2543bis [1] clause 7.1.3 | |
| "182" Queued | RFC2543bis [1] clause 7.1.4 | |
| "183" Session Progress | RFC2543bis [1] clause 7.1.5 | |

### 8.1.3 2xx: the action was successfully received, understood, and accepted

Status-codes of the successful class shall apply as defined in table 8.

**Table 8: Success status-codes**

| Status code | Reference | Support and notes |
|---|---|---|
| "200" OK | RFC2543bis [1] clause 7.2.1 | |

### 8.1.4 3xx: further action needs to be taken in order to complete the request

Status-codes of the redirection class shall apply as defined in table 9.

**Table 9: Redirection status-codes**

| Status code | Reference | Support and notes |
|---|---|---|
| "300" Multiple Choices | RFC2543bis [1] clause 7.3.1 | |
| "301" Moved Permanently | RFC2543bis [1] clause 7.3.2 | |
| "302" Moved Temporarily | RFC2543bis [1] clause 7.3.3 | |
| "305" Use Proxy | RFC2543bis [1] clause 7.3.4 | |
| "380" Alternative Service | RFC2543bis [1] clause 7.3.5 | |

### 8.1.5 4xx: the request contains bad syntax or cannot be fulfilled at this server

Status-codes of the client error class shall apply as defined in table 10.

**Table 10: Client error status-codes**

| Status code | Reference | Support and notes |
|---|---|---|
| "400" Bad Request | RFC2543bis [1] clause 7.4.1 | |
| "401" Unauthorized | RFC2543bis [1] clause 7.4.2 | |
| "402" Payment Required | RFC2543bis [1] clause 7.4.3 | *Editor's note: Referenced text marks this as reserved for future use, therefore at the moment it cannot be used.* |
| "403" Forbidden | RFC2543bis [1] clause 7.4.4 | |
| "404" Not Found | RFC2543bis [1] clause 7.4.5 | |
| "405" Method Not Allowed | RFC2543bis [1] clause 7.4.6 | |
| "406" Not Acceptable | RFC2543bis [1] clause 7.4.7 | |
| "407" Proxy Authentication Required | RFC2543bis [1] clause 7.4.8 | |
| "408" Request Timeout | RFC2543bis [1] clause 7.4.9 | |
| "409" Conflict | RFC2543bis [1] clause 7.4.10 | |
| "410" Gone | RFC2543bis [1] clause 7.4.11 | |
| "411" Length Required | RFC2543bis [1] clause 7.4.12 | |
| "413" Request Entity Too Large | RFC2543bis [1] clause 7.4.13 | |
| "414" Request-URI Too Large | RFC2543bis [1] clause 7.4.14 | |
| "415" Unsupported Media Type | RFC2543bis [1] clause 7.4.15 | |
| "420" Bad Extension | RFC2543bis [1] clause 7.4.16 | |
| "421" Extension Required" | draft-ietf-sip-serverfeatures-02 [6] clause 2.3 | |
| "480" Temporarily not available | RFC2543bis [1] clause 7.4.17 | |
| "481" Call Leg/Transaction Does Not Exist | RFC2543bis [1] clause 7.4.18 | |
| "482" Loop Detected | RFC2543bis [1] clause 7.4.19 | |
| "483" Too Many Hops | RFC2543bis [1] clause 7.4.20 | |
| "484" Address Incomplete | RFC2543bis [1] clause 7.4.21 | |
| "485" Ambiguous | RFC2543bis [1] clause 7.4.22 | |
| "486" Busy Here | RFC2543bis [1] clause 7.4.23 | |
| "487" Request Cancelled | RFC2543bis [1] clause 7.4.24 | |
| "488" Not Acceptable Here | RFC2543bis [1] clause 7.4.25 | |

### 8.1.6 5xx: the server failed to fulfil an apparently valid request

Status-codes of the server error class shall apply as defined in table 11.

**Table 11: Server error status-codes**

| Status code | Reference | Support and notes |
|---|---|---|
| "500" Internal Server Error | RFC2543bis [1] clause 7.5.1 | |
| "501" Not Implemented | RFC2543bis [1] clause 7.5.2 | |
| "502" Bad Gateway | RFC2543bis [1] clause 7.5.3 | |
| "503" Service Unavailable | RFC2543bis [1] clause 7.5.4 | |
| "504" Gateway Time-out | RFC2543bis [1] clause 7.5.5 | |
| "505" SIP Version not supported | RFC2543bis [1] clause 7.5.6 | |
| "580" Precondition Failure | draft-manyfolks-sip-resource-01 [8] clause 6.1 | |

### 8.1.7 6xx: the request cannot be fulfilled at any server

Status-codes of the global failure class shall apply as defined in table 12.

**Table 12: Global failure status-codes**

| Status code | Reference | Support and notes |
|---|---|---|
| "600" Busy Everywhere | RFC2543bis [1] clause 7.6.1 | |
| "603" Decline | RFC2543bis [1] clause 7.6.2 | |
| "604" Does not exist anywhere | RFC2543bis [1] clause 7.6.3 | |
| "606" Not Acceptable | RFC2543bis [1] clause 7.6.4 | |

## 8.2 Status codes defined within this specification

Editor's note: This clause is to be used to document those status-codes that require complete documentation within this specification, because they are not defined elsewhere in a form that can be endorsed, either with or without modification.

# 9 Request-URIs

## 9.1 Parameters of Request-URIs defined by the IETF

Editor's note: Need to formulate a title to this clause that does not include an organisation name.

Editor's note: This clause is to be used to either endorse, or endorse with modification, those parameters of Request-URIs that have already been documented by the IETF SIP working group.

Editor's note: The following table is a direct copy of table 2 of RFC2543bis, and requires modification to reflect changes to the status of the various components within the IM CN subsystem.

Table 13 shows the various parameters of request URIs that are supported within the IN CM subsystem. Support for the parameter also requires support for the associated procedures associated with that parameter.

**Table 13: Use and default values of URL components for SIP headers, Request-URI and references**

| | default | Req.-URI | To | From | Contact | Rec.-Route | external |
|---|---|---|---|---|---|---|---|
| user | – | O | o | o | O | o | o |
| password | – | O | o | - | O | o | o |
| host | mandatory | M | m | m | m | m | m |
| port | 5060 | o | o | o | o | o | o |
| user-param | Ip | O | o | o | O | o | o |
| method | INVITE | - | - | - | o | - | o |
| maddr-param | – | O | - | - | O | m | o |
| ttl-param | 1 | O | - | - | O | - | o |
| transp.-param | udp | o | - | - | o | - | o |
| other-param | o | O | o | o | o | o | o |
| headers | – | - | - | - | o | - | o |

# 10 SDP types

## 10.1 SDP types defined by the IETF

Editor's note: Need to formulate a title to this clause that does not include an organisation name.

Editor's note: This clause is to be used to either endorse, or endorse with modification, those SDP types that have already been documented by the IETF SIP working group.

### 10.1.1 Introduction

Support for a SDP type also requires support for the associated procedures associated with that sending and receipt of that SDP type.

### 10.1.1 Session description types

Session description types shall apply as defined in table 14.

**Table 14: Session description types**

| Type | Description | Reference | Support |
|---|---|---|---|
| v= | (protocol version) | RFC 2327 [12] clause 6 | |
| o= | (owner/creator and session identifier). | RFC 2327 [12] clause 6 | |
| s= | (session name) | RFC 2327 [12] clause 6 | |
| i=* | (session information) | RFC 2327 [12] clause 6 | |
| u=* | (URI of description) | RFC 2327 [12] clause 6 | |
| e=* | (email address) | RFC 2327 [12] clause 6 | |
| p=* | (phone number) | RFC 2327 [12] clause 6 | |
| c=* | (connection information - not required if included in all media) | RFC 2327 [12] clause 6 | |
| b=* | (bandwidth information) | RFC 2327 [12] clause 6 | |
| One or more time descriptions (see below) | | | |
| z=* | (time zone adjustments) | RFC 2327 [12] clause 6 | |
| k=* | (encryption key) | RFC 2327 [12] clause 6 | |
| a=* | (zero or more session attribute lines) | RFC 2327 [12] clause 6 | |
| Zero or more media descriptions (see below) | | | |

### 10.1.2 Time description types

Time description types shall apply as defined in table 15.

**Table 15: Time description types**

| Type | Description | Reference | Support |
|---|---|---|---|
| t= | (time the session is active) | RFC 2327 [12] clause 6 | |
| r=* | (zero or more repeat times) | RFC 2327 [12] clause 6 | |

### 10.1.3 Media description types

Media description types shall apply as defined in table 16.

**Table 16: Media description types**

| Type | Description | Reference | Support |
|---|---|---|---|
| m= | (media name and transport address) | RFC 2327 [12] clause 6 | |
| i=* | (media title) | RFC 2327 [12] clause 6 | |
| c=* | (connection information - optional if included at session-level) | RFC 2327 [12] clause 6 | |
| b=* | (bandwidth information) | RFC 2327 [12] clause 6 | |
| k=* | (encryption key) | RFC 2327 [12] clause 6 | |
| a=* | (zero or more media attribute lines) | RFC 2327 [12] clause 6 | |

## 10.2 Session description types defined within this specification

Editor's note: This clause is to be used to document those session description types that require complete documentation within this specification, because they are not defined elsewhere in a form that can be endorsed, either with or without modification.

# Annex A (informative): Documentation of temporary solutions that are to be submitted to the IETF

Editor's note: This annex provides a temporary space for holding the contents of material that is to be submitted to the IETF, or has been submitted to the IETF but is insufficently processed. Material in this annex will either:

- Process to RFC status, where said RFC will be normatively referenced from the main body of the text.
- Not progress to RFC status, where, after review of the requirements in association with the reasons for lack of progress, the material will be placed in the main body of this document.

Material to be submitted to the IETF will be ideally submitted under the title of draft-3gpp-sip-xxxxxxxxx.

Submission to the IETF under this title should be performed with the approval of the CN1 plenary, either by full meeting, or by an equivalent email correspondence procedure.

# Annex B (informative): Change history

*It is usual to include an annex (usually the final annex of the document) for specifications under TSG change control which details the change history of the specification using a table as follows:*

| Change history | | | | | | | |
|---|---|---|---|---|---|---|---|
| **Date** | **TSG #** | **TSG Doc.** | **CR** | **Rev** | **Subject/Comment** | **Old** | **New** |
| | | | | | Version 0.0.0 Editor's internal draft | | |
| | | | | | Version 0.0.1 Editor's internal draft | | |
| | | | | | Version 0.0.2 Editor's internal draft | | |
| | | N1-001060 | | | Version 0.0.3 Submitted to CN1 SIP adhoc #1 | | |
| 19/10/00 | | N1-001109 | | | Version 0.0.4 Reflecting results of initial CN1 discussion | | |
| 19/10/00 | | N1-001115 | | | Version 0.0.5 Reflecting output of CN1 SIP adhoc#1 discussion | | |
| 09/11/00 | | | | | Version 0.0.6 Revision to include latest template and styles | | |
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