FFT Doc 10.001 v2.1.1 (~~January~~ June 201~~8~~9)

French Federation of Telecoms
Innovation & Standards Committee
IP Interconnection Working Group
Architecture Sub-group

IP interconnection

Interface specification based on SIP/SDP

FÉDÉRATION FRANÇAISE DES TÉLÉCOMS

French Federation of Telecoms

Internet

http://www.fftelecoms.org

Table of Contents

1 Context

1.1 Purpose

The purpose of this document is to define the SIP/SDP interconnection interface for the interconnection between two French Operators for basic telephony services with deterministic charging for national and international origins and destinations.

This document supports the following basic call capabilities:
- narrowband speech and 3.1 kHz audio services (i.e. including analog fax and data modem calls),
- wideband speech,
- en bloc address signalling,
- early in-band information (forward and backward early media),
- in-band transport of DTMF tones and information (telephone-event for telephony services; G.711 in-band for M2M special usages that are not suitable with telephone-event and only for them),
- Calling party location information,
- User-To-User Information,
- Service access number before translation (for Value Added Services),
- Indication of a call with international origin,
- National short numbers.

and the following supplementary services:
- Calling Line Identification Presentation (CLIP),
- Calling Line Identification Restriction (CLIR),
- Call Forwarding,
- Call Hold.

1.2 Standards

As a rule, the interconnection between two mobile networks shall be governed by the applicable 3GPP standards. The interconnection between two fixed networks shall be governed by the applicable TISPAN/3GPP standards. .

Note: the present document applies to all types of SIP interconnection.

This document also describes optional features of interest to this specification. Other optional features are considered out of the scope of this document but may be considered on a bilateral basis.

2 References

The table below lists the documents that are referenced in current specification. Their use depends on the context as described in dedicated sections of current specification.

[Architecture V2.0~~1.1.2~~_FFT]	“Architecture for IP interconnection”, FFT Doc 09.002, v~~1.1.2~~ 2.0 (~~June~~ May 2018~~4~~)
[RFC3261]	IETF RFC 3261 "Session Initiation Protocol (SIP)"
[RFC3262]	IETF RFC 3262 "Reliability of Provisional Responses in the Session Initiation Protocol (SIP)"
[RFC3264]	IETF RFC 3264 "An Offer/Answer Model with the Session Description Protocol (SDP)"
[RFC3311]	IETF RFC 3311 "The Session Initiation Protocol (SIP) UPDATE method"
[RFC3312]	IETF RFC 3312 "Integration of Resource Management and Session Initiation Protocol (SIP)"
[RFC3323]	IETF RFC 3323 "A Privacy Mechanism for the Session Initiation Protocol (SIP)"
[RFC3325]	IETF RFC 3325 "Private Extensions to the Session Initiation Protocol (SIP) for Network Asserted Identity within Trusted Networks".
[RFC3326]	IETF RFC 3326 "The Reason Header Field for the Session Initiation Protocol (SIP)"
[RFC3407]	IETF RFC 3407 "Session Description Protocol (SDP) Simple Capability Declaration"
[RFC3556]	IETF RFC 3556 “Session Description Protocol (SDP) Bandwidth Modifiers for RTP Control Protocol (RTCP) Bandwidth”

[RFC3966]	IETF RFC 3966 "The tel URI for Telephone Numbers"
[RFC4028]	IETF RFC 4028 "Session Timers in the Session Initiation Protocol (SIP)"
[RFC4458]	IETF RFC 4458 "Session Initiation Protocol (SIP) URIs for Applications such as Voicemail and Interactive Voice Response (IVR)"
[RFC4566]	IETF RFC 4566 "Session Description Protocol (SDP)"
[RFC4733]	IETF RFC 4733 "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals"
[RFC5009]	IETF RFC 5009 "Private Header (P-Header) Extension to the Session Initiation Protocol (SIP) for Authorization of Early Media"
[RFC5806]	IETF RFC 5806 "Diversion Indication in SIP"
[RFC6432]	IETF RFC 6432 “Carrying Q.850 Codes in Reason Header Fields in SIP (Session Initiation Protocol) Responses”
[RFC6567]	IETF RFC 6567 “Problem Statement and Requirements for Transporting User-to-User Call Control Information in SIP”
[RFC7044]	IETF RFC 7044 “An Extension to the Session Initiation Protocol (SIP) for Request History Information” (obsoletes RFC 4244)
[RFC7315]	IETF RFC 7315 “Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3GPP”
[RFC7433]	IETF RFC 7433 “A Mechanism for Transporting User to User Call Control Information in SIP”
[RFC7434]	IETF RFC 7434 ”Interworking ISDN Call Control User Information with SIP”
[RFC7913]	IETF RFC 7913 “P-Access-Network-Info ABNF Update”
[RFC8119]	IETF RFC 8119 “SIP "cause" URI Parameter for Service Number Translation”
[TS 24.229]	3GPP Technical Specification 24.229 "IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3”
[TS 24.628]	3GPP Technical Specification 24.628 "Common basic communication procedures using IP Multimedia (IM)Core Network (CN) subsystem; Protocol specification"
[G.711]	ITU-T Recommendation " Pulse code modulation (PCM) of voice frequencies"
[G.729]	ITU-T Recommendation "Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)"
[G.729 Annex A]	ITU-T Recommendation Annex A "Reduced complexity 8 kbit/s CS-ACELP speech codec"

3 Glossary

CLIP	Calling Line Identity Presentation
CLIR	Calling Line Identity Restriction
DROM	Départements-Régions d'Outre-Mer (French Overseas Departments)
DTMF	Dual-Tone Multi-Frequency
M2M	Machine To Machine
MIME	Multipurpose Internet Mail Extensions
NNI	Network To Network Interface
SIP	Session Initiation Protocol
SDP	Session Description Protocol
TCP	Transport Control Protocol
UDP	User Datagram Protocol
UUI	User-to-User header
URI	Uniform Resource Identifier
VAS	Value Added Services

4 SIP signalling messages

The SIP messages and headers specified in this section must be encoded, filled and handled as specified in the referenced standard in which they are defined.

Request-URI in all SIP requests must be coded and filled according to [RFC3261] and as stated in section 11 for the initial INVITE message.

4.1 Definitions

"Reception" and "Transmission" directions refer to the direction of the messages.

In reception direction, "Supported" means that the header can be present in the message and if received, it must be handled according to the standard. "Mandatory" means that the recipient expects the header to be present. “Not applicable” means that the reception of the header can not occur according to the current specification. By symmetry “Not applicable” is relative only to headers with the status “not sent” in emission.

In transmission direction, "May be sent" means that the header can be present or omitted depending on the transaction or the call context. "Mandatory" means that the header is always present. "Not sent" means that the header shall not be sent.

4.2 Transport protocol

UDP is supported and required for carrying SIP messages. See Maximum message size section 4.5.

Note: According to IETF RFC 3261, TCP must be supported. This requirement arises out of the need to handle large messages. However, the size of messages is limited in the context of this document.

4.3 SIP methods and headers

4.3.1 SIP methods

Table 1 contains the SIP methods required to support the capabilities and services identified in section 1.1.

Mandatory methods
INVITE
RE-INVITE
ACK
BYE
CANCEL
OPTIONS (NOTE)

Table 1: Mandatory SIP methods

NOTE – It is mandatory to support OPTIONS in the reception direction only.

Support for methods not listed in Table 1 is optional, as the Update method which may be used if the optional keep-alive mechanism for active SIP sessions as defined in the [RFC4028] is used on bilateral agreement (see §18.1).

4.3.2 Network behaviour in reception

4.3.2.1 Method inspection

If a SIP method received is recognized but not supported, it shall be rejected as defined in [RFC 3261] by a 405 "Method not allowed" response.

If a SIP method received is not recognized (i.e. not implemented), it shall be rejected as defined in [RFC 3261] by a 501 "Not Implemented" response.

4.3.2.2 Status code inspection

If a non-supported error response is received in a SIP message then the relative call or transaction fails. The list of the supported and of the Not applicable responses with their detailed handling is given in section 4.3.4.3, Table 3.
If a non-recognized final response, i.e. not referenced in the section 4.3.4.3Table 3, is received in a SIP message then it shall be treated as being equivalent to the x00 response code of that class.
If a non-recognized provisional response different than 100 final response, i.e. not referenced in the section 4.3.4.3 Table 3, is received in a SIP message then it shall be treated as being equivalent to a 183 “Session Progress”.

4.3.2.3 ***Header inspection in requests***

If a non-supported SIP header or parameter is received in a SIP request, it shall be ignored unless its corresponding option tag is present in the Require header. The headers or parameters that are not mentioned in the tables from section 4.3.4 to section 4.3.9 are considered as Not applicable headers or parameters.

If a mandatory header is absent or malformed in the request, the request shall be rejected as defined in [RFC 3261].

4.3.2.4 ***Header inspection in responses***

If a non-supported SIP header or parameter is received in a SIP response, it shall be ignored. The headers or parameters that are not present in the tables from section 4.3.4 to section 4.3.9 are considered as non-supported headers or parameters.

If a header necessary for processing the response is absent or malformed in a provisional response, the response shall be discarded.

If a header necessary for processing the response is absent or malformed in a final response except a 2XX response, the response shall be treated as the 500 "Server Internal Failure" response.
If a header necessary for processing the response is absent or malformed in a final 2XX response to an INVITE request, the response shall be acknowledged by sending an ACK and then the dialog shall be terminated.

NOTE – The behaviour in case of receipt of “Not applicable” SIP signalling element is not defined in this specification since this is relative to a context out of the scope of the current document.

4.3.3 Network behaviour in emission

By default only the SIP signalling element (methods, headers, header parameters, response status codes, option tags, …) defined and authorized (mandatory or optional) as described within the current specification can be sent.
Nevertheless according to bilateral agreements, SIP signalling elements not defined or not authorized in the current specification can be exchanged over the interconnection interface.

4.3.4 Initial INVITE method

The initial INVITE request is mandatory as defined in [RFC3261].

4.3.4.1 ***SIP request handling***

The handling of this request is compliant with [RFC3261].

4.3.4.2 ***Supported headers in the request***

Table 2 gives the header status in the initial INVITE for both reception and transmission directions.

Header name	Reference	Reception	Transmission
Accept	[RFC3261]	Supported	May be sent
Allow	[RFC3261]	Supported	May be sent
Call-ID	[RFC3261]	Mandatory	Mandatory
Contact	[RFC3261]	Mandatory	Mandatory
Content-Length	[RFC3261]	Supported	May be sent
Content-Type	[RFC3261]	Mandatory if the body is not empty	Mandatory if the body is not empty
CSeq	[RFC3261]	Mandatory	Mandatory
Diversion	[RFC5806]	Supported with the restrictions described in section 17.2.	May be sent. See section 17.2.
From	[RFC3261]	Mandatory	Mandatory
History-Info	[RFC8119]	Supported for “Service access number before translation” (see section 8).	May be sent for “Service access number before translation” (see section 8).
Max-Forwards	[RFC3261]	Mandatory	Mandatory
Min-SE	[RFC4028]	Supported	May be sent

P-Access-network-info	[3GPP TS.24.229]	Supported. See section 5 and section 6	May be sent. See section 5 and section 6
P-Asserted-Identity	[RFC3325]	Supported. See section 17.1.	May be sent. See section 17.1.
Privacy	[RFC3323]	Supported. See section 17.1.	May be sent. See section 17.1.
Record-Route	[RFC3261]	Not applicable	Not sent
Route	[RFC3261]	Supported	May be sent
Session-Expires	[RFC4028]	Supported	May be sent
Supported	[RFC3261]	Supported	May be sent
Require	[RFC3261]	Not applicable	Not sent
To	[RFC3261]	Mandatory	Mandatory
User-to-User	[RFC7433]	Supported. See section 7	May be sent. See section 7
Via	[RFC3261]	Mandatory	Mandatory

Table 2: Supported SIP headers in the initial INVITE request

4.3.4.3 SIP response handling

SIP responses are handled according to [RFC3261] with the clarifications given in the table below. If a non-supported error response is received, then the relative call or transaction fails.

Multiple SIP provisional responses creating separate early dialogs, as specified in [RFC3261], are supported with the following clarifications:

- Upon receipt of provisional responses containing SDP bodies, the recipient shall use the most recent media session information received for sending media packets during the early dialog phase,
- Confirmed dialogs created by the first 200 OK response for non-existing early dialogs shall override any previously stored dialog information.

SIP response		Reception	Transmission
1xx	100 Trying	Supported	May be sent
	180 Ringing	Supported	Sent when the called user is notified for the incoming call.
	181 Call is being forwarded	Not applicable	Not sent
	182 Queued	Not applicable	Not sent
	183 Session Progress	Supported	May be sent
2xx	200 OK	Supported	Sent when the call is answered.
3xx		Not applicable	Not sent
4xx	400 Bad Request	Supported. The related call or transaction fails.	May be sent
	401 Unauthorized	Not applicable	Not sent
	402 Payment Required	Not applicable	Not sent
	403 Forbidden	Supported. The related call or transaction fails.	May be sent
	404 Not Found	Supported. The related call or transaction fails.	May be sent
	405 Method Not Allowed	Supported	May be sent
	406 Not Acceptable	Supported. The related call or transaction fails.	May be sent
	407 Proxy Authentication Required	Not applicable	Not sent
	408 Request Timeout	Supported	May be sent
	410 Gone	Supported. The related call or transaction fails.	May be sent
	413 Request Entity	Supported The related call or transaction fails.	May be sent

SIP response		Reception	Transmission
	Too Large	The request is not retried.	
	414 Request-URI Too Long	Supported. The related call or transaction fails.	May be sent
	415 Unsupported Media Type	Supported. The related call or transaction fails. The request is not retried.	May be sent
	416 Unsupported URI Scheme	Supported. The related call or transaction fails. The request is not retried.	May be sent
	420 Bad Extension	Supported. The related call or transaction fails. The request is not retried.	May be sent
	421 Extension Required	Not applicable	Not sent
	422 Session Interval Too Small	Supported	May be sent
	423 Interval Too Brief	Not applicable	Not sent
	480 Temporarily Unavailable	Supported. The related call or transaction fails.	May be sent
	481 Call/Transaction Does Not Exist	Supported. The related call or transaction fails.	May be sent
	482 Loop Detected	Supported. The related call or transaction fails.	May be sent
	483 Too Many Hops	Supported. The related call or transaction fails.	May be sent
	484 Address Incomplete	Supported. The related call or transaction fails.	May be sent
	485 Ambiguous	Not applicable	Not sent
	486 Busy here	Supported. The related call or transaction fails.	May be sent
	487 Request Terminated	Supported. The related call or transaction fails.	May be sent
	488 Not acceptable here	Supported. The related call or transaction fails.	Sent if the received request contains an SDP offer proposing non supported media format or IP version.
	491 Request Pending	Supported. For re-INVITE request, the behaviour recommended in [RFC3261]/14.1 on reception of this response is supported.	May be sent. For re-INVITE request, the behaviour recommended in [RFC3261]/14.1 on reception of this response is supported.
	493 Undecipherable	Supported. The related call or transaction fails	May be sent
5xx		Supported. The related call or transaction fails.	May be sent*
6xx	600 Busy Everywhere	Supported. The related call or transaction fails.	May be sent
	603 Decline	Supported. The related call or transaction fails.	May be sent
	604 Does Not Exist Anywhere	Supported. The related call or transaction fails.	May be sent
	606 Not Acceptable	Supported. The related call or transaction fails.	May be sent

Table 3: Handling of SIP responses

*: if the maximum number of simultaneous sessions is exceeded, a 503 response shall be sent with the reason phrase "Exceeded outbound of the service agreement".

4.3.4.4 Supported headers in the responses

Table 4 gives the header status in the SIP responses to the initial INVITE request for both reception and transmission directions.

Header name	Reference	Response code	Reception	Transmission
Accept	[RFC3261]	18X /200	Supported	May be sent
Accept	[RFC3261]	415	Mandatory	Mandatory
Allow	[RFC3261]	All codes	Supported	May be sent
Call-ID	[RFC3261]	All codes	Mandatory	Mandatory
Contact	[RFC3261]	1xx (except 100)	Supported	May be sent
Contact	[RFC3261]	200	Mandatory	Mandatory
Content-Length	[RFC3261]	All codes	Supported	May be sent
Content-Type	[RFC3261]	All codes	Mandatory if the body is not empty.	Mandatory if the body is not empty.
CSeq	[RFC3261]	All codes	Mandatory	Mandatory
From	[RFC3261]	All codes	Mandatory	Mandatory
Min-SE	[RFC4028]	422	Mandatory	Mandatory
P-Asserted-Identity	[RFC3325]	200	Supported. See section 17.1.	May be sent. See section 17.1.
P-Early-Media	[RFC5009]	18x	Supported with the restrictions described in section 12.1.3.	May be sent with the restrictions described in section 12.1.3.
Reason	[RFC3326] and [RFC6432]	All relevant codes	Supported	May be sent
Record-Route	[RFC3261]	18x 200	Not applicable	Not sent
Require	[RFC3261]	18x	Not applicable	Not sent
Require	[RFC3261]	200	Supported	May be sent
Session-Expires	[RFC4028]	200	Supported	May be sent
Supported	[RFC3261]	200	Supported	May be sent
To	[RFC3261]	All codes	Mandatory	Mandatory
Unsupported	[RFC3261]	420	Mandatory	Mandatory
User-to-User	[RFC7433]	All codes (except 100) **if end-to-end responses**	Supported. See section 7	May be sent. See section 7
Via	[RFC3261]	All codes	Mandatory	Mandatory

Table 4: Supported SIP headers in the responses to the initial INVITE request

4.3.5 Re-INVITE method

The re-INVITE request shall be supported as defined in [RFC3261].

4.3.5.1 SIP request handling

The handling of this request shall be compliant with [RFC3261].

4.3.5.2 Supported headers in the request

Table 5 gives the header status in the re-INVITE request for both reception and transmission directions.

Header name	Reference	Reception	Transmission
Accept	[RFC3261]	Supported	May be sent
Allow	[RFC3261]	Supported	May be sent
Call-ID	[RFC3261]	Mandatory	Mandatory
Contact	[RFC3261]	Mandatory	Mandatory
Content-Length	[RFC3261]	Supported	May be sent
Content-Type	[RFC3261]	Mandatory if the body is not empty	Mandatory if the body is not empty
CSeq	[RFC3261]	Mandatory	Mandatory
From	[RFC3261]	Mandatory	Mandatory
Max-Forwards	[RFC3261]	Mandatory	Mandatory
Min-SE	[RFC4028]	Supported	May be sent
Route	[RFC3261]	Supported	May be sent
Session-Expires	[RFC4028]	Supported	May be sent
Supported	[RFC3261]	Supported	May be sent
Require	[RFC3261]	Not applicable	Not sent
To	[RFC3261]	Mandatory	Mandatory
Via	[RFC3261]	Mandatory	Mandatory

Table 5: Supported SIP headers in the re-INVITE request

4.3.5.3 SIP response handling

The handling of the responses shall be compliant with [RFC3261].

1xx responses different from 100 are not expected for the re-INVITE request.

4.3.5.4 Supported headers in the responses

Table 6 gives the header status in the SIP responses to the re-INVITE request for both reception and transmission directions.

Header name	Reference	Response code	Reception	Transmission
Accept	[RFC3261]	200	Supported	May be sent
Accept	[RFC3261]	415	Mandatory	Mandatory
Allow	[RFC3261]	All codes	Supported	May be sent
Call-ID	[RFC3261]	All codes	Mandatory	Mandatory
Contact	[RFC3261]	200	Supported	May be sent
Content-Length	[RFC3261]	All codes	Supported	May be sent
Content-Type	[RFC3261]	200	Mandatory if the body is not empty.	Mandatory if the body is not empty.
CSeq	[RFC3261]	All codes	Mandatory	Mandatory
From	[RFC3261]	All codes	Mandatory	Mandatory
Min-SE	[RFC4028]	422	Mandatory	Mandatory
Require	[RFC3261]	200	Supported	May be sent
Session-Expires	[RFC4028]	200	Supported	May be sent
Supported	[RFC3261]	200	Supported	May be sent
To	[RFC3261]	All codes	Mandatory	Mandatory
Unsupported	[RFC3261]	420	Mandatory	Mandatory
Via	[RFC3261]	All codes	Mandatory	Mandatory

Table 6: Supported SIP headers in the responses to the re-INVITE request

4.3.6 CANCEL method

The CANCEL request shall be supported as defined in [RFC3261].

4.3.6.1 SIP request handling

The handling of this request shall be compliant with [RFC3261].

When the calling party side wishes to terminate the session during the early-dialog phase it is recommended to use the Cancel method instead of the Bye method.

4.3.6.2 Supported headers in the request

Table 7 gives the header status in the SIP CANCEL request for both reception and transmission directions.

Header name	Reference	Reception	Transmission
Call-ID	[RFC3261]	Mandatory	Mandatory
Content-length	[RFC3261]	Supported	May be sent
CSeq	[RFC3261]	Mandatory	Mandatory
From	[RFC3261]	Mandatory	Mandatory
Max-Forwards	[RFC3261]	Mandatory	Mandatory
Reason	[RFC3326]	Supported	May be sent
Route	[RFC3261]	Supported	May be sent
To	[RFC3261]	Mandatory	Mandatory
Via	[RFC3261]	Mandatory	Mandatory

Table 7: Supported SIP headers in the CANCEL request

Both SIP status codes and ITU-T Q.850 cause values in decimal representation are supported in the Reason header, according to [RFC3326].

4.3.6.3 SIP response handling

The handling of the responses shall be compliant with [RFC3261].

4.3.6.4 Supported headers in the responses

Table 8 gives the header status in the responses to the CANCEL request for both reception and transmission directions.

Header name	Reference	Response code	Reception	Transmission
Call-ID	[RFC3261]	All codes	Mandatory	Mandatory
Content-Length	[RFC3261]	All codes	Supported	May be sent
CSeq	[RFC3261]	All codes	Mandatory	Mandatory
From	[RFC3261]	All codes	Mandatory	Mandatory
To	[RFC3261]	All codes	Mandatory	Mandatory
Via	[RFC3261]	All codes	Mandatory	Mandatory

Table 8: Supported SIP headers in the SIP responses to the CANCEL request

4.3.7 ACK method

The ACK request shall be supported as specified in [RFC3261].

4.3.7.1 SIP request handling

The handling of this request shall be compliant with [RFC3261].

4.3.7.2 Supported headers in the request

Table 9 gives the header status in the ACK request for both reception and transmission directions.

Header name	Reference	Reception	Transmission
Call-ID	[RFC3261]	Mandatory	Mandatory
Contact	[RFC3261]	Supported	May be sent
Content-length	[RFC3261]	Supported	May be sent
Content-type	[RFC3261]	Mandatory if the body is not empty	Mandatory if the body is not empty
CSeq	[RFC3261]	Mandatory	Mandatory
From	[RFC3261]	Mandatory	Mandatory
Max-Forwards	[RFC3261]	Mandatory	Mandatory
Route	[RFC3261]	Supported	May be sent
To	[RFC3261]	Mandatory	Mandatory
Via	[RFC3261]	Mandatory	Mandatory

Table 9: Supported SIP headers in the ACK request

4.3.8 BYE method

The BYE request shall be supported as specified in [RFC3261].

4.3.8.1 SIP request handling

The handling of this request shall be compliant with [RFC3261].

4.3.8.2 Supported headers in the request

Table 10 gives the header status in the BYE request for both reception and transmission directions.

Header name	Reference	Reception	Transmission
Accept	[RFC3261]	Supported	May be sent
Allow	[RFC3261]	Supported	May be sent
Call-ID	[RFC3261]	Mandatory	Mandatory
Content-length	[RFC3261]	Supported	May be sent
CSeq	[RFC3261]	Mandatory	Mandatory
From	[RFC3261]	Mandatory	Mandatory
Max-Forwards	[RFC3261]	Mandatory	Mandatory
P-Asserted-Identity	[RFC3325]	Supported	May be sent
Reason	[RFC3326]	Supported	May be sent
Route	[RFC3261]	Supported	May be sent
To	[RFC3261]	Mandatory	Mandatory
User-to-User	[RFC7433]	Supported. See section 7	May be sent. See section 7
Via	[RFC3261]	Mandatory	Mandatory

Table 10: Supported SIP headers in the BYE request

Both SIP status codes and ITU-T Q.850 cause values in decimal representation shall be supported in the reason header, according to [RFC3326].

4.3.8.3 SIP response handling

The handling of the responses shall be compliant with [RFC3261].

4.3.8.4 Supported headers in the responses

Table 11 gives the header status in the SIP responses to the BYE request for both reception and transmission directions.

Header name	Reference	Response code	Reception	Transmission
Accept	[RFC3261]	415	Mandatory	Mandatory
Allow	[RFC3261]	All codes	Supported	May be sent
Call-ID	[RFC3261]	All codes	Mandatory	Mandatory
Content-Length	[RFC3261]	All codes	Supported	May be sent
Cseq	[RFC3261]	All codes	Mandatory	Mandatory
From	[RFC3261]	All codes	Mandatory	Mandatory
To	[RFC3261]	All codes	Mandatory	Mandatory
User-to-User	[RFC7433]	All codes (except 100) **if end-to-end responses**	Supported. See section 7	May be sent. See section 7
Via	[RFC3261]	All codes	Mandatory	Mandatory

Table 11: Supported SIP headers in the responses to the BYE request

4.3.9 OPTIONS method

The OPTIONS method shall be supported as specified in [RFC3261].

4.3.9.1 SIP request handling

The handling of this request shall be compliant with [RFC3261].

4.3.9.2 Supported headers in the request

Table 12 gives the header status in the OPTIONS request for both reception and transmission directions.

Header name	Reference	Reception	Transmission
Accept	[RFC3261]	Supported	May be sent
Allow	[RFC3261]	Supported	May be sent
Call-ID	[RFC3261]	Mandatory	Mandatory
Content-length	[RFC3261]	Supported	May be sent
CSeq	[RFC3261]	Mandatory	Mandatory
From	[RFC3261]	Mandatory	Mandatory
Max-Forwards	[RFC3261]	Mandatory	Mandatory
P-Asserted-Identity	[RFC3325]	Supported	May be sent
Supported	[RFC3261]	Supported	May be sent
To	[RFC3261]	Mandatory	Mandatory
Via	[RFC3261]	Mandatory	Mandatory

Table 12: Supported SIP headers in the OPTIONS request

4.3.9.3 SIP response handling

The handling of the responses shall be compliant with [RFC3261].

4.3.9.4 Supported headers in the responses

Table 13 gives the header status in the SIP responses to the OPTIONS request for both reception and transmission directions.

Header name	Reference	Response code	Reception	Transmission
Accept	[RFC3261]	415	Mandatory	Mandatory
Accept	[RFC3261]	200	Supported	May be sent
Allow	[RFC3261]	All codes	Supported	May be sent
Call-ID	[RFC3261]	All codes	Mandatory	Mandatory
Content-length	[RFC3261]	All codes	Supported	May be sent
CSeq	[RFC3261]	All codes	Mandatory	Mandatory
From	[RFC3261]	All codes	Mandatory	Mandatory
Supported	[RFC3261]	200	Supported	May be sent
To	[RFC3261]	All codes	Mandatory	Mandatory
Unsupported	[RFC3261]	420	Mandatory	Mandatory
Via	[RFC3261]	All codes	Mandatory	Mandatory

Table 13: Supported SIP headers in the responses to the OPTIONS request

4.4 SIP headers compact form

As stated in [RFC3261] it is optional to send SIP headers in compact forms, but implementations must support both the long and short forms of each header name in reception.

4.5 Maximum message size

Each network operator is responsible to check that the maximum size of SIP message and of SDP body it applies end to end does not prevent services to be delivered.

The maximum size of SIP message and of SDP body shall be **exchanged between the two interconnected operators in a bilateral agreement.**

If no agreement is found the following values shall be used by default:

- The size of SIP messages should not exceed 2048 bytes.
- The size of SDP bodies should not exceed 1024 bytes.

5 Calling party's location information for calls towards Value Added Services (VAS)

The calling party's location information can be optionally transmitted over the SIP interconnection interface according to bi-lateral agreement. The current specification takes into account the needs of location information especially for interconnection calls towards VAS. Nevertheless other contexts may exist, which require transmitting same information. In this case the same solution to transport in SIP the location information applies too. The purpose of this chapter is to provide a SIP solution to transport the calling party's location information identical to the one of the SPIROU Location Number parameter.

If transmitted, this location information shall be a network provided one. The location information provided by the network is identical to the one carried by the SPIROU Location Number parameter. It contains the originating network identifier (Operator Code) assigned by ARCEP to originating operator and the location area code associated to calling party's geographic location. This information depends on the originating network nature:

- mobile originating network: BTS/nodeB post code
- fixed originating network: INSEE code of the city, except in case of Paris, Lyon and Marseille where subdivisions are used

The P-Access-Network-Info header field, as defined in [3GPP TS24.229] §7.2A.4, is used to carry this location information. The location information provided by the originating network shall be placed in the "operator-specific-GI" parameter and shall be equal to $R_1R_2C_1C_2C_3C_4C_5$, where R_1R_2 are the 2 digits of the Operator Code and $C_1C_2C_3C_4C_5$ are the 5 digits of the post code or INSEE code. Moreover the np (network provided) parameter shall be present.

If the location information sent at the interconnection interface is invalid (e.g. $C_1C_2C_3C_4C_5$=00000) or improperly formatted (e.g. wrong number of digits), service dysfunctions may appear for non-geographic caller numbers.

Therefore the P-Access-Network-Info header carrying the calling party's location information provided by the network shall be coded according to the following syntax:

P-Access-Network-Info:(access-type / access-class);operator-specific-GI="value"**;network-provided**

The access-type or access-class parameters are by default not significant for the current specification. Nevertheless according to the P-Access-Network-Info header field specification of the [3GPP TS24.229] §7.2A.4 it is mandatory to have one of them and their value always shall be compliant with this specification. Consideration of this field must be done according to a bilateral agreement.

The access-info parameters "operator-specific-GI" and "np" parameters shall always be present. The value of the operator-specific-GI parameter shall be compliant with the 3GPP TS 29.163 standard and the value of the SPIROU Location Number (described in SPIROU1998-005 /edition 1.0 §3.30 Location Number and in the Décision ARCEP n° 05-0521 of December 8th 2005 annex A) populates the operator-specific-GI parameter. The operator-specific-GI is set to the text string between quotes (double quotes) with the sequence of digits found in octet 3 to N (except the filler) starting with the 1st digit:

operator-specific-GI = $R_1R_2C_1C_2C_3C_4C_5XX$

with R_1R_2 are the 2 digits of the Operator Code and $C_1C_2C_3C_4C_5$ are the 5 digits of the Location Area Code, and

with XX 2 digits between 0 to 9 (e.g. 00) for future use.

Hereafter is given an example of P-Access-Network-Info header field for a user located in the Orange mobile access network (61) in Issy les Moulineaux (92130), with XX=00:

P-Access-Network-Info:GSTN;operator-specific-GI="619213000";network-provided

NOTE – This specification requires only "operator-specific-GI" and "np" values as "access-info" parameters. Additional access-info parameters are possible but they are out of scope of the current specification. Therefore they can be exchanged only on bilateral agreement and in this case they always shall be compliant with the P-Access-Network-Info header field specification of the [3GPP TS24.229] §7.2A.4.

6 Indication of a call with international origin

In this section, a call having an “international origin” means that the call is either emitted from the international network or that at least an international interconnection link exist for this call.

During or after call completion, the information that a call received at SIP interconnection interface has an international origin (or not) is required. For example, this information is necessary for some real-time applications (e.g. services triggering, VAS…) or for offline purpose such as charging accounting and billing (e.g. wholesale billing, statistics…).

The information that a call has an international origin should be provided at SIP national interconnection interface for calls towards national destinations, by the national operator that is interconnected to the international operator delivering the call (cf. figure in Annex §24).
When this indication is provided at SIP national interconnection interface:

- End to end transmission of this indication is guaranteed for calls towards national VAS (Z=8) and is guaranteed for all national destinations in full SIP case.
- End to end transmission of this indication is by default not guaranteed for destinations other than national VAS (Z=8) in case of interworking with circuit switched networks, but may be guaranteed according to bilateral agreement.

When the calls that have an international origin are identified at SIP national voice interconnection interface, it shall be done as follows:

The P-Access-Network-Info header field, as defined in [3GPP TS 24.229] §7.2A.4, is used to carry this indication. This indication shall be placed in the "operator-specific-GI" parameter and shall be equal to $R_1R_2C_1C_2C_3C_4C_5X_1X_2$ where:

- R_1R_2 shall be set to “xx” value (*To be completed when ARCEP answer on this question is available*),
- $C_1C_2C_3C_4C_5$ shall be set to “99999” value (other values are reserved for future inter-operators use),
- X_1X_2 shall be set to “00” value.

The P-Access-Network-Info header carrying the indication that a call has an international origin shall be coded according to the following syntax:

P-Access-Network-Info:(access-type / access-class);operator-specific-GI=”value“**;network-provided**

The access-info parameters “operator-specific-GI” and “np” (network provided) shall always be present.
The operator-specific-GI in the access-info field is coded as a text string between double quotes (i.e. quoted-string).
The access-type or access-class parameters are not significant for the current specification. Nevertheless according to [RFC 7315] and [RFC 7913], it is mandatory to have one of them and their value always shall be compliant with these specifications.
For the current specification, the access-type parameter is set to “GSTN” (not significant).

Hereafter is given an example of P-Access-Network-Info header field carrying the indication that a call has an international origin, with access-type set to “GSTN” and within operator-specific-GI, R_1R_2 set to “99” (*example waiting for Arcep answer on this question*) and $C_1C_2C_3C_4C_5$ set to “99999”:

P-Access-Network-Info:GSTN;operator-specific-GI=“999999900”;network-provided

If received, the P-Access-Network-Info header carrying the indication that a call has an international origin shall be transmitted over the SIP interconnection interface.

NOTE: Since there is no standardized solution to carry in SIP the indication that a call has an international origin, a national specific solution is defined here. As the interworking is thereby not described in standards, some precisions are given below:
The indication that a call has an international origin (cf. syntax above) in P-Access-Network-Info header in initial INVITE request is equivalent and should interwork to “1” value of ISUP “National/international call indicator” (bit A) of “Forward Call Indicators” parameter in IAM message (cf. ITU-T Q763, §3.23).
In case of interworking from SIP to ISUP (or SIP-I), upon reception of the SIP P-Access-Network-Info header carrying the indication that a call has an international origin (cf. syntax above), the ISUP “National/international call indicator” (bit A) of “Forward Call Indicators” parameter should be set to “1” value and no ISUP Location Number parameter should be generated in ISUP (or in SIP-I).

7 User To User Information

The SIP User-to-User header (UUI), defined in [RFC7433], has been created to convey transparently in SIP end-to-end user-to-user information, in conformance with the function requirements defined in [RFC6567].
The current FFT specification only considers "ISDN" user-to-user information exchange as specified in [RFC7434] for VAS services framework. This information is analog to and can interwork with the one of the ISDN UUS1 implicit supplementary service. Therefore the UUI header field can be present only in INVITE requests and responses, and in BYE requests and responses. When the UUI header field is used in responses, it can only be utilized in end-to-end responses, for example in 1xx (excluding 100) and 2xx responses.

The syntax of UUI header [RFC7433] is the following:

```
UUI  = "User-to-User" HCOLON uui-value *(COMMA uui-value)
uui-value   = uui-data *(SEMI uui-param)
uui-data    = token / quoted-string
uui-param   = pkg-param / cont-param / enc-param / generic-param
pkg-param   = "purpose" EQUAL pkg-param-value
pkg-param-value = token
cont-param   = "content" EQUAL cont-param-value
cont-param-value = token
enc-param   = "encoding" EQUAL enc-param-value
enc-param-value = token / "hex"
```

The "ISDN" user-to-user information is included in the uui-data element. It is composed of two parts: firstly of a protocol discriminator and secondly of the user information.
The protocol discriminator describes the user information and is specified in table 4-26 of [ITU-T Recommendation Q.931]. It is one octet long.
The length of the user information is assumed to be at most equal to 128 octets.

The procedures for the "ISDN" user-to-user information exchange in SIP shall be compliant with the [RFC7434] with the following clarifications:

- UUI header field shall be present in the initial INVITE request if it is planned to use it in subsequent requests/responses, even when there is no data (except the protocol discriminator octet) to send at that point in time
- Only a single UUI header field can be included in each SIP message
- The "purpose" parameter should be included. Its value shall be equal to "isdn-uui"
- The "content" parameter is optional. If present, it shall be equal to "isdn-uui"
- The "encoding" parameter is optional. If present, it shall be equal to "hex"

An example of a UUI header sent over the SIP interconnection interface is given below:

User-to-User:"04353030303331";purpose=isdn-uui

8 Service access number before translation (for VAS)

Some Value added services (ex: hotline, customer care, Freephone…) are reached dialing a service access number which is not a globally routable number and consequently needs to be translated into a routable SIP or tel URI to process the session establishment.
In order to permit the receiving entity to retrieve the service requested by the calling user, the service access number shall be stored during its translation in the SIP signalling message towards the final destination.

For that purpose, the service access number before translation shall be conveyed in the History-Info header and shall be identified thanks to a History-Info entry containing "cause" SIP URI parameter set to the value "380" as defined in [RFC 8119] which should also contain an "mp" or "rc" header field parameter as defined by [RFC 7044] (i.e. the History-Info entry containing the cause parameter value "380" conveys the service access number after translation and refers to the History-Info entry containing the service access number before translation thanks to "mp" or "rc" parameter if present; otherwise the service access number before translation is contained in the preceding History-Info entry).

The syntax of the History-Info header [RFC 7044] is the following:

```
History-Info          = "History-Info" HCOLON hi-entry *(COMMA hi-entry)
```



```
hi-entry            = hi-targeted-to-uri *(SEMI hi-param)
hi-targeted-to-uri  = name-addr
hi-param            = hi-index/hi-target-param/hi-extension
hi-index            = "index" EQUAL index-val
index-val           =  number *("." number)
number              =  [ %x31-39 *DIGIT ] DIGIT
hi-target-param     = rc-param / mp-param / np-param
rc-param            = "rc" EQUAL index-val
mp-param            = "mp" EQUAL index-val
np-param            = "np" EQUAL index-val
hi-extension        = generic-param
```

The cause-param parameter is a SIP URI parameter and is defined in [RFC 4458].
The cause URI parameter shall be inserted and set to the value "380" in the History-Info entry (URI) of the service access number after translation, as defined in [RFC 8119].

An example of a History-Info header used for service access number before translation and sent over the SIP interconnection interface is given below:

History-Info:
<sip:*ServiceAccessNumber*;user=phone>;index=1,
<sip:*ServiceAccessNumberAfterTranslation*;user=phone;cause=380>;mp=1;index=1.1

9 Message bodies

In the context of this document, the only SIP message body type supported is SDP (application subtype "application/sdp").

10 Supported option tags of SIP extensions

In the context of this document:

- the “timer” option tag is authorized if the optional keep-alive mechanism for active SIP sessions as defined in the [RFC4028] is used on bilateral agreement (see §18.1)
- the “histinfo” option tag is authorized for “Service access number before translation” (see §8).

No other option tag is supported in the context of this document.

11 Identities format, address parameters and signalling mode

The identities formats supported for the Request-URI, and the From, To, P-Asserted-Identity, Diversion and History-Info headers are described in the following table.

The address formats supported for the Route, Via, and Contact headers are also described in the following table.

SIP URI format shall comply with [RFC3261]/19.1 and TEL URI with [RFC3966].

Supported formats in reception direction (NOTE 1)		**Sent formats in transmission direction (NOTE 2)**	
From (for E.164 subscriber numbers)	1. SIP URI like globalnumber@domainname with user=phone 2. SIP URI like globalnumber@IP_address with user=phone 3. Tel URI in global number format	From (for E.164 subscriber numbers)	1. SIP URI like globalnumber@domainname with user=phone 2. SIP URI like globalnumber@IP_address with user=phone 3. Tel URI in global number format

Supported formats in reception direction (NOTE 1)		Sent formats in transmission direction (NOTE 2)	
To (for E.164 subscriber numbers)	1. SIP URI like globalnumber@domainname with user=phone 2. SIP URI like globalnumber@IP_address with user=phone 3. Tel URI in global number format	To (for E.164 subscriber numbers)	1. SIP URI like globalnumber@domainname with user=phone 2. SIP URI like globalnumber@IP_address with user=phone 3. Tel URI in global number format
To (for national short codes)	1.SIP URI in local number format @domainname with user=phone 2. SIP URI in local number format @IP_address with user=phone 3. Tel URI in local number format	To (for national short codes)	1.SIP URI in local number format @domainname with user=phone 2. SIP URI in local number format @IP_address with user=phone 3. Tel URI in local number format
P-Asserted-Identity (for E.164 subscriber numbers)	1. SIP URI like globalnumber@domainname with user=phone 2. SIP URI like globalnumber@IP_address with user=phone 3. Tel URI in global number format	P-Asserted-Identity (for E.164 subscriber numbers)	1. SIP URI like globalnumber@domainname with user=phone 2. SIP URI like globalnumber@IP_address with user=phone 3. Tel URI in global number format
Request-URI (for E.164 subscriber numbers)	1. SIP URI like globalnumber@domainname with user=phone 2. SIP URI like globalnumber@IP_address with user=phone 3. Tel URI in global number format	Request-URI (for E.164 subscriber numbers)	1. SIP URI like globalnumber@domainname with user=phone 2. SIP URI like globalnumber@IP_address with user=phone 3. Tel URI in global number format
Request-URI (for national short codes)	1. SIP URI in local number format @domainname with user=phone 2. SIP URI in local number format @IP_address with user=phone 3. Tel URI in local number format	Request-URI (for national short codes)	1. SIP URI in local number format @domainname with user=phone 2. SIP URI in local number format @IP_address with user=phone 3. Tel URI in local number format
Diversion (for E.164 subscriber numbers)	1. SIP URI like globalnumber@domainname with user=phone 2. SIP URI like globalnumber@IP_address with user=phone 3. Tel URI in global number format	Diversion (for E.164 subscriber numbers)	1. SIP URI like globalnumber@domainname with user=phone 2. SIP URI like globalnumber@IP_address with user=phone 3. Tel URI in global number format
History-Info (for E.164 subscriber numbers)	1. SIP URI like globalnumber@domainname with user=phone 2. SIP URI like globalnumber@IP_address with user=phone (NOTE 4)	History-Info (for E.164 subscriber numbers)	1. SIP URI like globalnumber@domainname with user=phone 2. SIP URI like globalnumber@IP_address with user=phone (NOTE 4)
Via	IP address / port	Via	IP address / port
Route	SIP URI (NOTE 3)	Route	SIP URI (NOTE 3)
Contact	SIP URI (NOTE 3)	Contact	SIP URI (NOTE 3)

NOTE 1 – In the receiving direction, when several formats are listed (e.g. 1. 2. 3…), this means that all formats must be supported.

NOTE 2 – In the sending direction, when several formats are listed, this means that at least one format of the list must be supported.

NOTE 3 – The use of a FQDN instead of an IP address must be agreed between both connecting parties beforehand.

NOTE 4 – Used URI scheme shall be SIP URI. The "cause" URI parameter cannot be added if hi-targeted-to-uri is a tel URI.

Table 14: Supported format identities

Moreover, the following principles shall be taken into consideration:

- Global-number format shall be used for subscriber numbers (for E.164 subscriber numbers and M2M numbers) as described in [RFC3966]. In case of M2M services, the national significant number portion of the global number may belong to the French number range for M2M applications. In the French dialling plan, such numbers begin with 0700 and have an extended length: 13 digit NSNs for Metropolitan France and 12 digit NSNs for a DROM.
- In Global-number format, the "+" is mandatory in front of the number as described in [RFC3966]
- Local-number format, as described in [RFC3966], shall be used for national short codes (French specific non E.164 numbers: 1X, 1XY, 10YT, 118XYZ, 116XYZ, 3BPQ). Whether for Metropolitan or DROM destination, the phone-context parameter is set to +33. For example, 3610 short codes will be conveyed in Request-URI as tel:3610;phone-context=+33 or sip:3610;phone-context=+33@domainname;user=phone.
- The telephone number must contain only digits.
- End-to-end delivery of the display-name received over the interconnection interface is not guaranteed.
- Request-URI identity and To header contain information related to the called party number. From and P-Asserted-Identity headers contain information related to the calling party number, The Diversion header contains information related to the diverting party number. Those identities are always in the form of a E.164 number, except for Request-URI and To header in case of national short codes (see the previous relevant bullet).

  When they belong to the French numbering range, the E.164 numbers shall comply with one of the following formats:
    - +CCZABPQMCDU or,
    - +CC(number portability prefix)ZABPQMCDU (Request-URI identity and To header only), with
        - CC is “33” or the country code allocated to a DROM depending on the value of the ZAB(P) except for VAS services (Z=8) ; In case of Z=8, CC=33 for Metropolitan and DROM destinations, and
        - (number portability prefix) is a number portability prefix as defined by the French regulation authority or,
    - +CC(“ZONE BLANCHE prefix”)$N_1N_2…N_n$ (Request-URI identity and To header only), with
        - CC is “33”, and
        - (“ZONE BLANCHE prefix”) is a prefix in a 600xyz format as defined by the French regulation authority (e.g. 600794 for a national call from Bouygues Telecom network to SFR Network) with x = destination network, y = originating network, z = call type (MOC), and
        - $N_1N_2…N_n$ is a National Significant Number (as defined in ITU-T Recommendation E.164).

  or only for M2M applications:
    - +33700PQMCDUEFGH for Metropolitan France where 700PQMCDUEFGH is a National Significant Number (as defined in ITU-T Recommendation E.164), or
    - +CC700PQMCDUEFG, with CC is the country code allocated to a DROM (e.g. 262) where 700PQMCDUEFG is a National Significant Number (as defined in ITU-T Recommendation E.164), with
        - The value of P determines the value of CC as detailed in the Annex 2 (“Numéros mobiles de longueur étendue (ZAB = 700)”) of the ARCEP décision n° 2012-085 17/07/12 “relative à la réorganisation des tranches de numéros commençant par 06 et 07”.

  When they do not belong to the French numbering range, i.e. correspond to international numbers (“foreign”), the E.164 numbers shall comply with the following format:
    - +CC$N_1N_2…N_n$, with
        - CC is the country code allocated to the relevant country,
        - $N_1N_2…$ Nn is a National Significant Number (as defined in ITU-T Recommendation E.164).

The Unavailable User Identity (« sip:unavailable@unknown.invalid »), as defined in the standard 3GPP TS 23.003 §13.7, shall be used (NOTE 5) in the From header exchanged over the SIP interconnection interface for the following use cases, and only for them:

- Unavailability of a valid telephone number identifying the calling party
- Calls crossing international boundaries without bilateral agreement on CLI information delivery
- 2G or 3G handset mobile access originating calls when the CLIR service is invoked
- Fixed analogic access originating calls when the CLIR service is invoked.

NOTE 5: For backward compatibility reasons, when using an interworking equipment implementing 3GPP release inferior to 12, the Anonymous User Identity (« sip:anonymous@anonymous.invalid »), as defined in the standard 3GPP TS 23.003 §13.6, may be used according to bilateral agreement in the From header exchanged over the SIP interconnection interface for the following use cases, and only for them:

- 2G or 3G handset mobile access originating calls when the CLIR service is invoked
- Fixed analogic access originating calls when the CLIR service is invoked.

Neither call setup nor proper CLIP/CLIR service operation can be guaranteed if the recommended formats in this section are not respected.

The "en bloc" signalling mode shall be used, i.e. the entire called party number shall be included into a single INVITE request. Overlap operations are optional and out of the scope of this document.

NOTE 6: Identities sent over the interconnection interface may exceed 15 digits (max. length authorized by E.164). Nevertheless they remain compliant with the ARCEP's national policies defining the structure of the French numbering. So by default E.164 format shall be applied, nevertheless according to bilateral agreement operators may use numbers exceeding 15 digits.

11.1 ISDN access

In case of a calling user behind an ISDN access where the "Special arrangement" applies (as defined in ETS 300−089) two calling party numbers must be conveyed by the network signalling. One is asserted by the network, originating from the network (NDI, Numéro de Désignation de l'Installation). The other one is provided by the user, originating from the user provided unscreened ISDN "Calling Party Number" information element (NDS, Numéro de Désignation Supplémentaire).
Therefore at the downstream SIP interconnection interface, the value of the SIP "P-Asserted Identity" header field will have to equal to the value of the calling party identity asserted by the network. The value of the SIP "From" header field will have to equal to the calling party identity provided by the user.

12 Media session management

SDP offer/answer exchange shall be performed according to [RFC3261], [RFC3264] and [RFC4566].

SDP information is only supported in the body of INVITE, re-INVITE, ACK, 200 OK (INVITE, re-INVITE) and 18x (INVITE) messages.

At minimum, the SDP parameters used in [RF3264] shall be supported.

Mechanisms and parameters defined for preconditions [RFC3312] and for SDP simple capability declaration [RFC3407] are optional.

12.1 Media session establishment

12.1.1 Initial INVITE message

This section assumes offer/answer rules solely based on [RFC3261] and [RFC3264]. Additional offer/answer rules defined in [RFC3262] and [RFC3311] may be used by bilateral agreement but are out of the scope of this document.

Initial INVITE messages may or may not contain an SDP offer.

NOTE – By default, if an initial INVITE message does not contain an SDP offer, then backward early-media (towards the origin of the call) is not possible (cf. §12.1.3).

Initial INVITE messages with an SDP offer shall not be coded with the address of connection (c= line) set to 0.0.0.0.

When the initial INVITE contains an SDP offer, the SDP answer shall be present in the 200 OK response.

When the initial INVITE does not contain an SDP offer, the SDP offer shall be present in the 200 OK response.

12.1.2 Codec negotiation rules

In a media stream "m=" line, codecs shall be listed in order of preference for SDP negotiation, the first codec format listed being the preferred one.

If an SDP answer is received indicating support of more than one codec different from "telephone-event" among those proposed in the SDP offer, only the first one shall be considered. To switch to another proposed media format of the SDP answer other than "telephone-event", a SDP re-negotiation shall be performed (see section 12.2).

The "a=ptime" is a media attribute which indicates the desired packetization interval that the end point would like to consider in reception for a specific media stream (but not for a specific codec)**.** If the information is available, it is recommended to send the "ptime" parameter over the interconnection interface. The recommended packetisation times for codecs are described in section 13.

If there are no media formats in common in the SDP offer received in:

- an initial INVITE or re-INVITE, it shall be rejected by a 488 "Not acceptable here" response;
- a 200 OK response to the INVITE message, the call shall be released.

If there are no media formats in common in the SDP offer received in:

- an initial INVITE or re-INVITE, it shall be rejected by a 488 "Not acceptable here" response;
- a 200 OK response to the INVITE message, the call shall be released.

12.1.3 Early media

The reception of a SDP answer in a 18x response is not a sufficient indication of an early media coming from a downstream domain. The P-early-media header must be included to guarantee an early media stream sent in the backward direction (towards the origin) or in the backward & forward directions will be taken into account in all cases. The P-Early-Media header present in a 18x response must contain the direction parameter set to “sendrecv” or to “sendonly”. If another value is used, the P-Early-Media header must be ignored. The P-Early-Media header syntax is defined in [RFC5009] and [TS 24.628].

12.2 Media session modification

Once the session is established, the modification of the parameters of the media session shall be supported through the re-INVITE message according to [RFC3261].

12.3 Terminating a session

The procedures used to terminate a session are described in [RFC3261], with the following precision: When the calling party side wishes to terminate the session during the early-dialog phase it is recommended to use the Cancel method instead of the Bye method (cf. §4.3.6).

12.4 RTP/RTCP packet source

In a session, the same IP address and port number shall be used to send and receive RTP packets (symmetric IP address and port number).

> *Note: The port number for sending/receiving RTCP packets MUST be equal to "the port number negotiated for RTP" + 1.*

The [RFC3556] defining SDP Bandwidth Modifiers for RTCP bandwidth can be optionally supported on bilateral agreement.

13 Voice codecs

The list of the supported codec and their usage rules are described in [ArchitectureV~~1.1.2~~2.0_FFT]§4.2.2.1 “Codecs à bande étroite” and §4.2.2.2 “Codecs à large bande”.

14 DTMF transport

For Human to Machine, the “telephone-event” [RFC 4733] must be used for DTMF transport. For this purpose, the support of "telephone-event" must be indicated during the SDP offer/answer exchange and used in accordance with rules described in [ArchitectureV~~1.1.2~~2.0_FFT]§4.2.2.4 “Telephone-event”. If received at interconnection interface, on reception of an SDP offer or answer containing “telephone-event” pseudo-codec, the “telephone-event” pseudo-codec in the m= line shall be transmitted over the interconnection interface. The SDP offer and the SDP answer must contain “telephone-event” over the interconnection interface.

For Machine to Machine (M2M), and only in this case, DTMF transport can be done either using “telephone-event” mode or in G.711 in-band when “telephone-event” mode is not suitable for some special usages (non-voice usages). In this last case, this enables to avoid transcoding from in-band DTMF tones to “telephone-event” and so fulfills the need of DTMF transport transparency for some critical M2M existing specific usages still performed by user equipment or central site servers such as Telealarm or Telemonitoring.
To transport DTMF in G.711 in-band for M2M communications between endpoints, the SDP offer must contain G.711 and may or may not contain “telephone-event”, and the SDP answer must contain G.711 over the interconnection interface and must not contain “telephone-event”.
In order to avoid dysfunctions, G.711 RTP flows shall be transparently transmitted end to end, from caller to called party, meaning without transcoding nor transrating. In particular, on reception of an SDP offer or answer containing G.711 codec at NNI, the G.711 codec in the m= line shall be transmitted over the interconnection interface without modification (it shall not be deleted neither moved in the codecs list). Moreover, the DTMF transport in G.711 in-band shall not be extracted/regenerated and shall not be interworked to “telephone-event” mode. It is also assumed that no interworking from G.711 in-band to “telephone-event” mode nor transcoding nor transrating is applied inside each operator’s network.
Attention should be paid that in-band DTMF is only applicable for sessions using the G.711 codecs. Moreover, when G.711 in-band DTMF is used, some telephony service features are not guaranteed.

General recommendations:
Only one technical solution for DTMF transport shall be used at the same time (either “telephone-event” or G.711 in-band). As a result, the DTMF signals shall not be sent encoded in audio packets using simultaneously “telephone-event” and G.711 in-band (in order to avoid interoperability issues on reception of DTMF signals duplicated in different formats).
Once the session is established, it is not possible to change of DTMF transport mode without re-negotiation.
By default the interconnection equipment shall be transparent to the SDP offer/answer negotiation, in particular for DTMF transport.
The transport of DTMF out-of-band (i.e. using SIP INFO) is forbidden in the context of this document.

Editor’s note: this section may evolve according to the results of the inter-operators tests to be held on DTMF transport for M2M.

15 FAX Modem

Fax modem calls are supported by default by using the G.711 codec (see [ArchitectureV~~1.1.2~~2.0_FFT]§4.2.2.1 “Codecs à bande étroite” for its usage rules) without media session modification.

NOTE – This means that fax modem calls must be established with G.711 as the initial negotiated codec.

In addition T38 mode may be used when bilaterally agreed.
V.152 is optional.

However, there is no guaranty of end to end interoperability because it depends on customer devices, which is beyond the control of the operator.

16 Data Modem

Data modem calls are supported by using the G.711 codec (see [ArchitectureV2.0~~1.1.2~~_FFT]§4.2.2.1“Codecs à bande étroite” for its usage rules) without media session modification.

NOTE – This means that data modem calls must be established with G.711 as the initial negotiated codec.

V.152 is optional.

17 Supplementary services

17.1 CLIP/CLIR

Rule n°1: At the signalling interface, the "P-Asserted-Identity” header must be present in the initial INVITE request (except in some cases, see Note 1) with a telephone number corresponding to the calling party, provided (or verified) by the network operator serving the calling party and expressed in a valid global-number format (see section 11, Table 14).

> *Note 1: in some cases (e.g. calls crossing international boundaries), it is accepted that P-Asserted-Identity is absent due to the lack of bilateral agreement on CLI delivery.*

Rule n°2: The "From" header must be sent with a telephone number identifying the calling party (except in some cases, see section 11) and expressed in valid global-number format (see section 11, Table 14) with a valid content. This rule applies even when the CLIR service is requested (see rule n°3). The upstream operator shall not remove a valid telephone number contained in the “From” header in messages sent over the interconnection interface.

Important:

- If both rules n°1 and n°2 are respected, the content of the “From” header is presented to the CLIP subscriber.
- If one of these two rules detailed above is not respected, the provision of CLIP service to the called party is not guaranteed.
- The presentation of "Display-name" field of the "From" header is not guaranteed.

Rule n°3: The "Privacy" header is used for the CLIR service. The “Privacy” header is defined in [RFC3323] and shall contain at least the values “Id” and “user” for expressing the CLIR service invocation.
NOTE - The "P-Asserted-Identity” header is restricted with the value “id” defined in [RFC3325] in the “Privacy” header and the “From” header is restricted with the value “user” defined in [RFC3323] in the “Privacy” header.

17.2 Call forwarding services

17.2.1 Additional information about parameters and values of the "Diversion" header

[RFC5806] shall be supported in order to represent call fowarding information.

16.1.2.1 Additional information about parameters and values of the "Diversion" header

- *Diversion ="Diversion" ":" # (name-addr *(";" diversion_params))*
- *diversion-params =diversion-reason | diversion-counter | diversion-limit | diversion- privacy | diversion-screen | diversion-extension ;*

- *diversion-reason = "reason" "=" ("unknown" | "user-busy" | "no-answer" | "unavailable" | "unconditional" | "time-of-day" | "do-not-disturb" | "deflection" | "follow-me" | "out-of-service" | "away" | token | quoted-string) ;*

This field is mandatory. Values "unavailable", "time-of-day", "do-not-disturb", "follow-me", "out-of-service" and "away" may be sent over the interconnection interface but need not be taken into account by the recipient.

- *diversion-counter = "counter" "=" 1*2DIGIT ;*

This field is mandatory and its recommended value is '1' for each diversion that occurred as recommended in RFC 5806 (see Note). Otherwise, call delivery and on storage of the diversion data may fail.

> *Note 1 – As a result of interworking with older control protocols (e.g. SSUTR2), the counter may be received with a value of " 5".*

> *Note 2 – The maximum value of the diversion-counter parameter shall be exchanged between the interconnected parties.*

- *diversion-limit = "limit" "=" 1*2DIGIT ;*

This field may be sent over the interconnection interface but needs not be taken into account by the recipient.

- *diversion-privacy = "privacy" "=" ("full" | "name" | "uri" | "off" | token | quoted-string) ;*

This field is recommended. The values "name" and "uri" are not supported. If received, they shall be mapped to "full". If the diverting user has a CLIR service activated, then privacy must be set to "full". If not, privacy must be set to "off".

- *diversion-screen = "screen" "=" ("yes" | "no" | token | quoted-string) ;*

This field may be sent over the interconnection interface but needs not be taken into account by the recipient.

- *diversion-extension = token ["="(token | quoted-string)] ;*

This field may be sent over the interconnection interface but needs not be taken into account by the recipient.

17.2.2 Limitation of the number of diversion and loop issue

Based on bilateral agreement, each network operator shall mention for the interconnection interface the value of its internal limitation on the number of communication diversions allowed, as described in 3GPP TS24.604 §4.5.2.6.1. If no agreement is found, the counter shall be set to 5 by default.

NOTE – The default value "5" is given provided that the resulting Post Dial Delay fulfills the QoS requirements.

An anti-loop mechanism shall be used to avoid loops between the two interconnected networks, eg. by having in each network a limitation procedure when the internal threshold is reached. In this sense the Diversion-counter, enabling to count the number of communication diversions, shall be sent with reliable information."

Reminder, forward of an emergency call is forbidden (Décision ARCEP n° 2010-1233, 14 décembre 2010). Therefore emergency calls delivered to the SIP interconnection interface can not be marked as having already been diverted.

17.3 Call Hold

The Call Hold shall be provided according to the following principles:

- The service is possible only after the dialog is confirmed, i.e after the 200 OK response to the initial INVITE;
- The mechanism described in [RFC 3264], section 8.4, using the direction attribute ("a=") in an updated SDP to request the other party to stop sending media, may be used;
- The mechanism described in [RFC 2543], section B.5, using a connection address ("c=") equal to 0.0.0.0. in an updated SDP to put on hold a call, may be used;
- The mechanism described in [RFC 3264], section 8.4 is preferred.

18 Keep alive

18.1 Keep alive for active SIP sessions

A keep alive mechanism shall be used to check that communications are still active. It can be performed either by sending periodic OPTIONS messages or as defined in [RFC4028]. The support for either of these methods is optional.

When OPTIONS method is used, an OPTIONS message is sent for each confirmed dialog:
- If a response is sent back, the communication is considered still active.
- If no response is sent back, an OPTIONS message is sent again. Then, if again no response is received, the call is released.

The delay between two OPTIONS messages depends on the equipment configuration.

Acknowledgment of OPTIONS messages shall be supported as defined in [RFC 3261].

18.2 Keep alive for interconnection signalling links

A keep alive mechanism shall be used to monitor the general status of the signalling links between connecting equipments.
A similar keep alive mechanism to the sending of periodic OPTIONS messages, as previously described**,** can be used to monitor the general status of the signalling links between connecting equipments. In this case, OPTIONS or INVITE messages are sent as standalone requests.

19 Ringback tone

It is up to the calling side to generate a local ringback tone upon receipt of a 180 “Ringing” answer to an INVITE message. Nevertheless the calling party side need to be prepared to receive ringback tone delivered as early-media (i.e. using the voice codec and as described in §12.1.3) over the interconnection interface by the called party side.

20 Differences with 3GPP/TISPAN standards (informative)

This section outlines difference with standards, for the convenience of the reader. This section is informative only.

- According to 3GPP TS 24.x04, IMS call forwarding services are implemented based on the History-Info header. However, in order to cope with currently available implementations in the market, the services are rendered by other means in the context of this document (Cf. Diversion header). This impacts the headers transiting at the NNI.

21 Codecs and transcoding guidelines (informative)

This section focuses on the narrow band voice codecs that should be used over the IP interconnection interface between two mobile Circuit Switched (CS) R4 networks, two fixed VoIP networks or between a fixed VoIP and a mobile CS R4 network.

From the IP interconnection interface perspective, the media end points in mobile CS networks are the transcoding units located in the mobile MGWs in case of 3G access or in the TRAU in case of 2G access. These transcoding units provide the conversion between G.711 codec and mobile compressed speech codecs (e.g. GSM FR, AMR…). In a direct mobile CS to mobile CS IP interconnection scenario, when TrFO capabilities are supported end-to-end, the media end point can be the mobile device itself.

In fixed VoIP networks there are several possible media end points that need to be considered from the IP interconnection interface perspective: VoIP terminals, IPBXs and MGWs.

In France, the two most common voice codecs used by fixed VoIP media end points are G.711 A law and G.729 (with or without Annex A). G.711 sets the voice quality reference for narrow-band voice codecs from the client perspective and is supported by many VoIP terminals in the consumer market. In certain circumstances however, G.711 is not used because of the lack of access bandwidth (G.711 requires around 106 kbit/s access bandwidth). This is the reason why some fixed VoIP terminals and IPBXs in the business market support exclusively G.729. Fixed MGWs that need to communicate with a wide range of fixed VoIP terminals currently support both G.711 and G.729.

Guideline #1:

It should be noted that mobile MGWs and fixed MGWs are designed to perform transcoding and hence have optimized hardware for this purpose. As a consequence, if transcoding cannot be avoided for particular IP interconnection call configurations and if this configuration involves a MGW (mobile or fixed), it is then recommended that the transcoding takes place in the MGW instead of any other dedicated network equipment.

Guideline #2:

Fixed VoIP terminals or IPBXs that support G.711 should also support G.729 (i.e. include G.729 in SDP offer/answer exchanges) in order to avoid the need for network-based transcoding when communicating with fixed terminals or IPBXs that support or can operate only G.729.

Guideline #3

In the situation whereby a Mobile CS network is interconnected with a fixed VoIP network or a transit network, the edge mobile MGW should be configured to support G.711 but also G.729 in case the distant media endpoint is a fixed VoIP terminal that supports or can operate only G.729.

22 Work plan for the next versions (informative)

This section describes areas considered for further study and that will be addressed in the next version of this specification. This section is informative only.

The following work items have been identified:

- SIP format for SPIROU's "Calling Party Category" parameter information,
- Multiple early-dialogs,
- Other topics have been identified even though the requirement still needs to be confirmed (e.g. ISDN services…).

23 History

History of the present document		
V0.1	11/12/2009	Document creation
V0.2	13/01/2010	Modifications following the 11/01/2010 meeting
V0.3	10/03/2010	Modifications proposed by France Télécom
V0.4	16/03/2010	Modifications following the 15/03/2010 meeting
V0.5	22/03/2010	Modifications following SFR comments and FT updates
V0.6	28/04/2010	Modifications following the 12/04/2010 meeting
V0.7	04/06/2010	Modifications following the 04/06/2010 meeting
V1.0	06/2010	Approved public version
V1.0.1	10/09/2012	Document input to FFT meeting of 10/09/12
V1.0.2	17/09/2012	Modifications following the 10/09/2012 meeting and modifications proposed by France Télécom
V1.0.9	02/10/2012	Modifications following the 01/10/2012 meeting. Version sent to consultation to the FFT members and vendors.
V1.0.9	04/12/12	Modification of §8.1.3 following ALU comments and the 03/12/2012 meeting
V1.1	14/12/12	Approved public version
V1.1.1	15/04/13	Modifications following the 15/04/13 meeting: addition of international calls and short codes.
V1.1.2	10/06/13	Modifications following the 10/06/13 meeting: addition of M2M calls, addition of explanations for ISDN access, addition of a note on the banning of diverting emergency calls.
V1.1.3	15/07/13	Modifications following the 15/07/13 meeting: modification of the Keep-alive mechanism §14 to clarify its mandatory status, modification of the requirement on the SIP header compact form §14, modification of §9, §10, §11 and §12 to take into account the new version of the Architecture document yet covering all codec aspect.
V1.1.4	16/09/13	Modifications following the 3GPP CT3#74 (05-09/08/13) meeting and the FFT 16/09/13 meeting: addition of the Unavailable User Identity in §13.1 (CLIP/CLIR) following the acceptance of the Orange Change Request C3-131201.
V1.1.5	04/10/13	Version sent to consultation to the FFT members and vendors.
V1.1.6	29/11/13	Modification following the phone meeting of 29/11/13: §3 and §18 following ALU comments, §7 following E/// comments.
V1.2	16/12/13	Approved public version.
V1.2.1	20/01/15	Addition of §5 "Calling party's location information for calls towards Value Added Services (VAS)" on the P-ANI header field and §6 "User To User Information" on the UUI header field.
V2.0	02/02/15	Rewording and modification of §9 for specification of the called party numbers in "Zone blanche"
V2.0.1	23/06/16	§5 "Calling party's location information for calls towards Value Added Services (VAS)" correction of a typing error : read GSTN instead of GTSN as access-type in the P-Access-Network-Info header given as example.
V2.0.1.1	23/06/16	- In §9, new rule for VAS number, the code country (Global number with Z=8) and the phone-context (for short number) shall be equal to (+)33 whatever the destination metropolitan or DOM. - In §4.5, no more fixed value for the maximum size of the SIP messages and the SDP bodies but a default value if no agreement can be found by bilateral agreement. - In §5, a precision in case of improperly formatted or invalid location information. - In §15.1, removal of Note 2 from CLIP/CLIR and inclusion into §9.

V2.0.1.2	01/09/16	Remarks from FFT meeting of 28/07/2016 taken into account + clarifications in §3, §4.5, §5 and §9.
V2.0.1.3	04/05/17	Updates to take into account VAS needs in SIP: - Use of History-Info for conveying in SIP the "service access number before translation" in §1.1, §2, §4.3.4.2, §7 (new section), §9, §10. - Forward & backward early-media in §1.1 and §11.1.3 (already allowed in the document). In addition, the following modifications are brought: - In §10, addition of Note 5 allowing the use of "Anonymous" in From header in some specific cases and only for them on bilateral agreement. - In §2 and in §4.3.4.4, addition of a reference to RFC 6432 for conveying Q.850 error codes in Reason header in INVITE responses (already allowed in the document). - Some editorial changes are also made in the document.
V2.0.1.4	29/05/17	In §11.1.3, rewording to indicate that early-media is in backward or in the backward & forward directions, following meeting of 29/05/2017.
V2.0.1.5	24/07/17	In §7, addition of a precision concerning the transport of the service access number before translation (identified by but not in the entry containing the cause "380").
V2.0.1.6	05/09/17	- In order to take into account VAS needs in SIP, addition of a new section §6 to define in SIP the indication that a call has an international origin using specific coding of P-Access-Network-Info header, mention of this use in §4.3.4.2, addition of concerned RFC in §2, update of work plan in §22 and addition of a figure in §24 Annex. - In §7, rewording of a precision concerning the transport of the service access number before translation, following FFT meeting of 24/07/2017. - In §10, addition of ";user=phone" that was missing in an example of a call to 3610 short code.
V2.0.1.7	15/09/17	In §6, modifications on the indication that a call has an international origin, following FFT meetings of 08/09/2017 and of 11/09/2017. Version sent to consultation.
V2.0.1.8	17/10/17	In §14 and in §1.1, modifications to take into account the request of the FFT GT4 "PSTN extinction" on DTMF transport for M2M (possibility to use G.711 in-band for DTMF transport for M2M special usages that are not suitable with telephone-event and only for them).
V2.1	15/01/18	In §6 (indication that a call has an international origin in SIP), replacement of the requirement by a recommendation, following comments received during FFT meeting of 11/12/2017. Approved public version.
V2.1.1	17/06/19	Update of the version of the document "Architecture for IP interconnection", FFT Doc 09.002, from V1.1.2 (June 2014) to version V2.0 (May 2018) (in §2, §13, §14, §15, §16).

24 Annex

This figure gives an end to end macroscopic view of calls that have an international origin towards national SVA: