

# **INTERNATIONAL INTERCONNECTION FORUM FOR SERVICES OVER IP**

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## **Interconnection & Roaming IMS Signalling Profile (Release 3.0) May 2016**

## **Revision History**

<b>Date</b>	<b>Rel.</b>	<b>Subject/Comment</b>
14 <sup>th</sup> May 2012	1	First release dedicated to interconnect scenarios
12 <sup>th</sup> May 2013	2.0	Second Release encompassing the scope also with the Roaming Scenario
18 <sup>th</sup> April 2016	3.0	Enhancing the scope to ViLTE , SMS and RCS plus updating the content to the latest 3GPP specifications 3GPP TS 29.165 V.13.2.0. (2015-09)

## EXECUTIVE SUMMARY

This i3 Forum Interconnection & Roaming IMS (I-IMS) Signalling Profile specification defines a signalling profile for a Network to Network Interface (NNI) as defined in the ‘i3 Forum Technical Interconnection Model for International Voice Services (Release 6) [199]’, and is documented in the “Style” of 3GPP TS 29.165.

Section 6 illustrates a detailed compliance to TS 29.165, v13.2.0 (2015-09). In this way, the i3 Forum I-IMS signalling profile can be directly compared to the 3GPP TS 29.165 NNI signalling profile document for negotiating agreements between two carriers. In addition, it allows for future extensibility for support of the GSMA IPX requirements.

This new version of the Interconnection & Roaming Signalling Profile supports an NNI for basic voice services (i.e. Supplementary Services are not in the scope), SMS, basic video and RCS. In general, and also for RCS, it is assumed in this document that the IPX Provider delivers either the transit or the hubbing service to the Service Provider. In other words, the IPX Provider switches the messages. This is different from where the IPX Provider only delivers pure transport and is not service aware.

Future versions will add IMS Supplementary Services for voice and other non-voice IMS Services.

It is recommended that this Interconnection Signalling Profile should be supported as the minimal profile on the Inter-IMS Network to Network Interface (II-NNI) for basic voice, including SMS, video and RCS services in both Interconnect and Roaming scenarios. It is recommended that bilateral agreements describe the support/transparency (or not) of optional elements.

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## **1 Scope of the document**

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Future versions will add IMS Supplementary Services for voice and other non-voice IMS Services.

## 2 Acronyms

3GPP	3rd Generation Partnership Project
ALG	Application Level Gateway
ATCF	Access Transfer Control Function
B2BUA	Back to Back user agent
BGCF	Border Gateway Control Function
CSCF	Call Session Control Function
IBCF	Interconnection Border Control Function
I-BGF	Interconnection Border Gateway Function
I-CSCF	Interrogating-Call Session Control Function
ICSS	IMS Centralized Services
I-IMS	Interconnection & Roaming IMS
II-NNI	Inter-IMS Network to Network Interface
IM-CN	IP Multimedia Core Networks
IMS	IP Multimedia Subsystem
IMS-ALG	Multimedia Subsystem Application Level Gateway
IP	Internet Protocol
IPSec	IP Security
IPv4	Internet Protocol Version 4
IPv6	Internet Protocol Version 6
MGCF	Media Gateway Control Function
MGF	Media Gateway Function
MIME	Multipurpose Internet Mail Extensions
MSC	Mobile Switching Center
NAT	Network Address Translation
NAT-PT	Network Address Translation—Protocol Translation
NNI	Network to Network Interface
P-CSCF	Proxy Call Session Control Function
RCS	Rich Communication Suite
RTP	Real-Time Protocol
SBC	Session Border Controller
S-CSCF	Serving-Call Session Control Function
SCTP	Stream Control Transmission Protocol
SDP	Session Description Protocol
SGF	Signalling Gateway Function
SIP	Session Initiation Protocol
SIP URI	SIP protocol Uniform Resource Identifier
SIP-I	SIP with encapsulated ISUP
SIP-T	SIP for Telephones
SLA	Service Level Agreement
SRVCC	Single Radio Voice Call Continuity
TCP	Transmission Control Protocol
tel-URI	Telephone Uniform Resource Identifier
TRF	Transit and Roaming Function
TrGw	Transition Gateway
TLS	Transport Layer Security
UA	User Agent
UDP	User Datagram Protocol
URI	Uniform Resource Identifier
VoIP	Voice over IP

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- [194] draft-ietf-sipcore-refer-clarifications-02 (January 2015): "Clarifications for the use of REFER with RFC6665".

- [195] draft-ietf-sipcore-refer-explicit-subscription-00 (November 2014): "Explicit Subscriptions for the REFER Method".
- [196] draft-roach-sipcore-6665-clarifications-00 (October 2014): "A clarifications on the use of Globally Routable User Agent URIs (GRUUs) in the Session Initiation Protocol (SIP) Event Notification Framework".
- [197] IETF RFC 2646: "The Text/Plain Format Parameter".
- [198] IETF RFC 1866: "Hypertext Markup Language - 2.0".
- [199] i3 Forum "Technical Interconnection Model for International Voice Services" Release6.0, May 2014"

## 4 NNI Signalling Profile

### 4.1 Profile Overview

The i3 Forum I-IMS signalling profile is a subset of the 3GPP TS 29.165 signalling profile.

### 4.2 Analysis

#### 4.2.1 i3 forum approach and rationale

The i3 Forum I-IMS signalling profile:

1. starts with Section 6 of 3GPP TS 29.165 v13.2.0 (2015-09).
2. retains ***in any case*** all 3GPP table entries
3. further determines the applicability of an item according to the i3 Forum “Technical Interconnection Model for International Voice Services” Release 6.0, May 2014. the applicability of that model may result in striking out the text of a specific table entry.  
(Note that the scope of this version is for basic voice, SMS, video and RCS);
4. gives an analysis of the “Notation” given an item (Mandatory, Conditional or Optional).

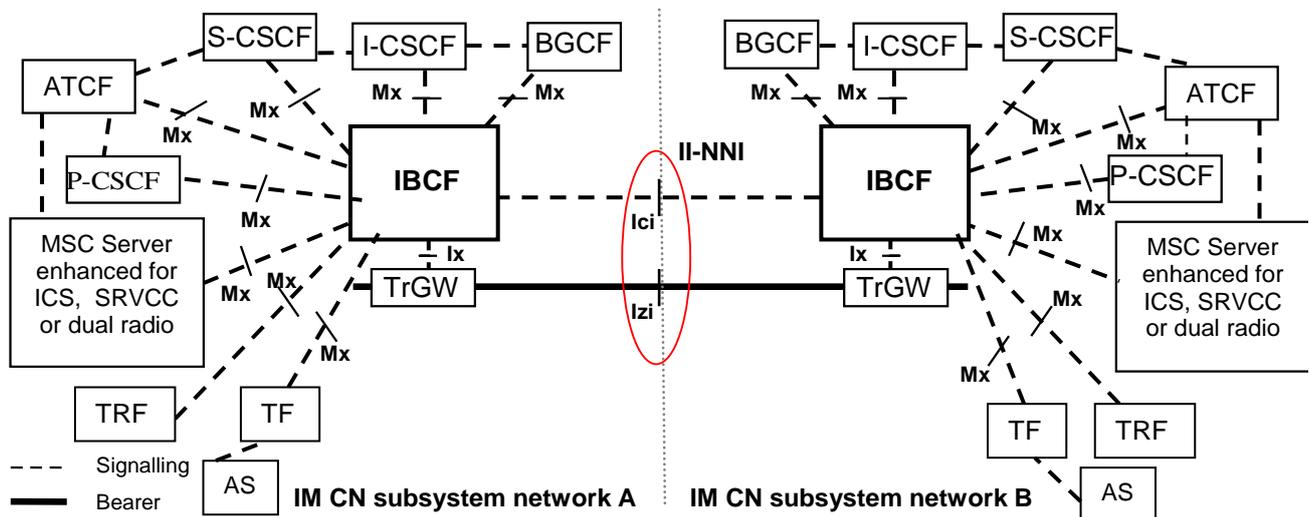
#### 4.2.2 Summary of exceptions to TS 29.165

This I-IMS signaling protocol profile shall be in accordance with 3GPP TS 29.165, with the exceptions noted in section 6. TS 29.165 clause numbers are referenced in this section. The “blank on purpose” are added to maintain section numbering continuity.

## 5 Reference Model for Interconnection

### 5.1 General

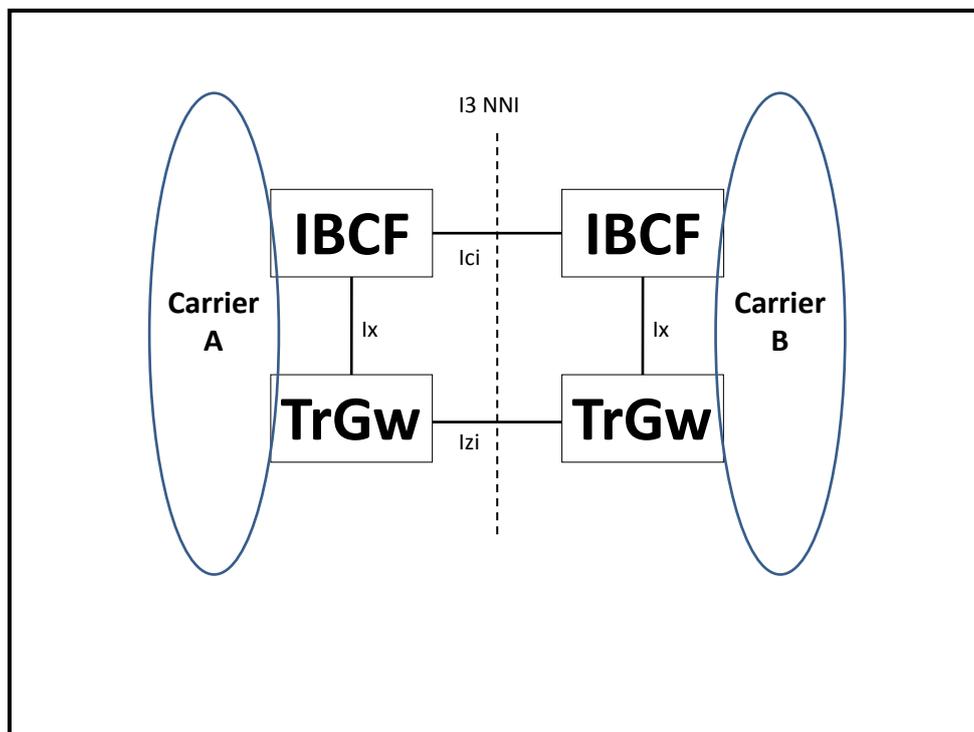
Figure 5.1.1 illustrates the architecture diagram given in 3GPP TS 23.228 [4] showing the Inter-IMS Network to Network Interface (II-NNI) between two IM CN subsystem networks



NOTE: The TRF can reside in a stand-alone entity or can be combined with another functional entity.

**Figure 5.1.1: Inter-IMS Network to Network Interface between two IM CN subsystem networks**

Figure 5.1.2 illustrates the i3 Forum II-NNI, where there are IBCF/TrGW's on either side of the interface. The internal carrier's network environment is out of scope.



**Figure 5.1.2: i3 Forum I-IMS Network to Network Interface**

The protocols over the two reference points Ici and Izi make up the Inter-IMS Network to Network Interface.

The Ici reference point allows IBCFs to communicate with each other in order to provide the communication and forwarding of SIP signalling messaging between IM CN subsystem networks. The Izi reference point allows TrGWs to forward media streams between IM CN subsystem networks.

IMS roaming performed by using II-NNI is considered, when the IBCFs are inserted at the network borders. The applicability of roaming scenario by using II-NNI is based on agreement between the operators.

Whenever the Inter-IMS Network to Network Interface is used to interconnect two IM CN subsystem networks belonging to different security domains, security procedures apply as described in 3GPP TS 33.210 [10].

## 5.2 Functionalities performed by entities at the edge of the network

### 5.2.1 Interconnection Border Control Function (IBCF)

An IBCF provides application specific functions at the SIP/SDP protocol layer in order to perform interconnection between IM CN subsystem networks by using Ici reference point. According to 3GPP TS 23.228 [4], IBCF can act both as an entry point and as an exit point for a network.

The functionalities of IBCF are indicated in the 3GPP TS 23.228 [4] and specified in 3GPP TS 24.229 [5]. They include:

- network topology hiding;
- application level gateway (for instance enabling communication between IPv6 and IPv4 SIP applications, or between a SIP application in a private IP address space and a SIP application outside this address space);
- controlling transport plane functions;
- controlling media plane adaptations;
- screening of SIP signalling information;
- selecting the appropriate signalling interconnect;
- generation of charging data records;
- privacy protection; and
- inclusion of a transit IOI when acting as an entry point for a transit network.

Based on local configuration, the IBCF performs transit routing functions as specified in 3GPP TS 24.229 [5].

The IBCF acts as a B2BUA when it performs IMS-ALG functionality.

### 5.2.2 Transition Gateway (TrGW)

According to 3GPP TS 23.002 [3], the TrGW is located at the network borders within the media path and is controlled by an IBCF. Forwarding of media streams between IM CN subsystem networks is applied over Izi reference point.

The TrGW provides within the media path functions like network address/port translation and IPv4/IPv6 protocol translation. NAT-PT binds addresses in IPv6 network with addresses in IPv4 network and vice versa to provide transparent routing between the two IP domains without requiring any changes to end points. NA(P)T-PT provides additional translation of transport identifier (TCP and UDP port numbers). The approach is similar to that one described also in 3GPP TS 29.162 [8].

Further details are described in 3GPP TS 23.228 [4].

## 6 Control Plane Interconnection

### 6.1 SIP methods and header fields

#### 6.1.1 Blank on purpose

##### 6.1.1.1 Notations of the codes

In table 1, which is equal to table 6.3 of the 3GPP Specification, the status codes "m", "o", "c" and "n/a" have the following meanings:

**Table 1 : Key to notation codes for SIP messages**

Notation code	Notation name	Sending side	Receiving side
m	mandatory	The message shall be supported at II-NNI. Supporting sending a SIP message at the II-NNI means that this message shall be sent over the II-NNI if received from the serving network. It does not imply that network elements inside the serving network or user equipment connected to this network shall support this message.	Supporting receiving a SIP message at the II-NNI means that this message shall be forwarded to the serving network. It does not imply that network elements inside the serving network or user equipment connected to this network are supporting this message.
o	optional	The message may or may not be supported at II-NNI. The support of the method is provided based on bilateral agreement between the operators.	Same as for sending side.
n/a	not applicable	It is impossible to use/support the message.	It is impossible to use/support the message. This message will be discarded by the IBCF.
c <integer>	conditional	The requirement on the message ("m", "o" or "n/a") depends on the support of other optional or conditional items. <integer> is the identifier of the conditional expression.	Same as for sending side.

##### 6.1.1.2 SIP methods

3GPP TS 24.229 [5] defines the methods allowing an IBCF to interconnect to an IBCF placed in another IM CN subsystem.

The following SIP methods are supported on the II-NNI as defined in table 6.1.

The following table is based on table A.5 and table A.163 of 3GPP TS 24.229 [5] and endorsed for this document:

Table 6.1: Supported SIP methods

Item	Method	Ref.	II-NNI		II-NNI (roaming)	
			Sending	Receiving	Sending	Receiving
1	ACK request	IETF RFC 3261 [13]	m	m	m	m
2	BYE request	IETF RFC 3261 [13]	m	m	m	m
3	BYE response	IETF RFC 3261 [13]	m	m	m	m
4	CANCEL request	IETF RFC 3261 [13]	m	m	m	m
5	CANCEL response	IETF RFC 3261 [13]	m	m	m	m
5A	INFO request	IETF RFC 6086 [39]	o	o	o	o
5B	INFO response	IETF RFC 6086 [39]	o	o	o	o
8	INVITE request	IETF RFC 3261 [13]	m	m	m	m
9	INVITE response	IETF RFC 3261 [13]	m	m	m	m
9A	MESSAGE request	IETF RFC 3428 [19]	<u>o x3</u>	<u>o x3</u>	<u>o x3</u>	<u>o x3</u>
9B	MESSAGE response	IETF RFC 3428 [19]	<u>o x3</u>	<u>o x3</u>	<u>o x3</u>	<u>o x3</u>
10	NOTIFY request	IETF RFC 6665 [20]	c1	c1	c1	c1
11	NOTIFY response	IETF RFC 6665 [20]	c1	c1	c1	c1
12	OPTIONS request	IETF RFC 3261 [13]	<u>⊘ x1</u>	<u>⊘ x1</u>	<u>⊘ x1</u>	<u>⊘ x1</u>
13	OPTIONS response	IETF RFC 3261 [13]	<u>⊘ x1</u>	<u>⊘ x1</u>	<u>⊘ x1</u>	<u>⊘ x1</u>
14	PRACK request	IETF RFC 3262 [18]	m	m	m	m
15	PRACK response	IETF RFC 3262 [18]	m	m	m	m
15A	PUBLISH request	IETF RFC 3903 [21]	c1	c1	c1	c1
15B	PUBLISH response	IETF RFC 3903 [21]	c1	c1	c1	c1
16	REFER request	IETF RFC 3515 [22]	o	o	<u>o x2</u>	<u>o x2</u>
17	REFER response	IETF RFC 3515 [22]	o	o	<u>o x2</u>	<u>o x2</u>
18	REGISTER request	IETF RFC 3261 [13]	c2	c2	c2	c2
19	REGISTER response	IETF RFC 3261 [13]	c2	c2	c2	c2
20	SUBSCRIBE request	IETF RFC 6665 [20]	c1	c1	c1	c1
21	SUBSCRIBE response	IETF RFC 6665 [20]	c1	c1	c1	c1
22	UPDATE request	IETF RFC 3311 [23]	m	m	m	m
23	UPDATE response	IETF RFC 3311 [23]	m	m	m	m
c1:	In case of roaming scenario, the support of the method is m, else o.					
c2:	In case of roaming scenario, the support of the method is m, else n/a.					
x1:	<u>Support of OPTIONS in a SIP dialog is mandatory, where support of OPTIONS out of a SIP dialog is optional.</u>					
x2:	<u>Needed to support CONF service as specified within TS 24.147 [106] Section 5.3.1.5.3</u>					
x3:	Mandatory for SMS over IP					

Underlined items in the table above are modifications or additions to table 6.1 in the 3GPP specification.

### 6.1.1.3 SIP header fields

#### 6.1.1.3.1 General

The IBCF shall provide the capabilities to manage and modify SIP header fields according to subclause 5.10 and Annex A of 3GPP TS 24.229 [5] with modifications as described in the following subclauses.

#### 6.1.1.3.2 Trust and no trust relationship

The IBCF acting as exit point applies the procedures described in clause 5.10.2 of 3GPP TS 24.229 [5] before forwarding the SIP signalling to the IBCF acting as entry point. The IBCF acting as entry point applies the procedures described in clause 5.10.3 of 3GPP TS 24.229 [5].

Additionally, in case there is no trust relationship between the two IM CN subsystems connected by II-NNI, the IBCF acting as exit point applies the procedures described in clause 4.4 of 3GPP TS 24.229 [5], before forwarding the SIP signalling.

These procedures may be utilized on a per header field basis to realize overall trust as well as per service level screening of header fields. Trust relationships and trust domains may be defined by inter-operator agreements for individual services and/or individual SIP header fields.

The management of the SIP header fields (if present) over II-NNI in case of a presence or not of a trust relationship between the two interconnected IM CN subsystems is wrapped up in the following table.

**Table 6.2: Management of SIP header fields and parameters over II-NNI in presence or not of a trust relationship**

Item	Header field or parameter	Reference	Trust relationship	Not trust relationship
1	P-Asserted-Identity	IETF RFC 3325 [44]	As specified in TS 24.229 [5], clause 4.4	As specified in TS 24.229 [5], clause 4.4
2	P-Access-Network-Info (NOTE 2)	IETF RFC 7315 [24]	As specified in TS 24.229 [5], clause 4.4	As specified in TS 24.229 [5], clause 4.4
3	Resource-Priority	IETF RFC 4412 [78]	As specified in TS 24.229 [5], clause 4.4	As specified in TS 24.229 [5], clause 4.4
4	History-Info (NOTE 2)	IETF RFC 7044 [25]	As specified in TS 24.229 [5], clause 4.4	As specified in clause 7 of IETF RFC 7044 [25] and in TS 24.229 [5], clause 4.4
5	P-Asserted-Service (NOTE 2)	IETF RFC 6050 [26]	As specified in TS 24.229 [5], clause 4.4 (NOTE 3)	As specified in TS 24.229 [5], clause 4.4 (NOTE 3)
6	P-Charging-Vector	IETF RFC 7315 [24]	As specified in TS 24.229 [5], clause 5.10	As specified in TS 24.229 [5], clause 5.10
7	P-Charging-Function-Addresses (NOTE 4)	IETF RFC 7315 [24]	As specified in TS 24.229 [5], clause 5.10	As specified in TS 24.229 [5], clause 5.10
8	P-Profile-Key (NOTE 2)	IETF RFC 5002 [64]	As specified in TS 24.229 [5], clause 4.4	As specified in TS 24.229 [5], clause 4.4
9	P-Private-Network-Indication (NOTE 1)	IETF RFC 7316 [84]	As specified in TS 24.229 [5], clause 4.4	As specified in TS 24.229 [5], clause 4.4
10	P-Served-User (NOTE 1, NOTE 2)	IETF RFC 5502 [85]	As specified in TS 24.229 [5], clause 4.4	As specified in TS 24.229 [5], clause 4.4
11	Reason (in a response)	IETF RFC 6432 [49]	As specified in TS 24.229 [5], clause 4.4	As specified in TS 24.229 [5], clause 4.4
12	P-Early-Media	IETF RFC 5009 [74]	As specified in TS 24.229 [5], clause 4.4	As specified in TS 24.229 [5], clause 4.4
13	Feature-Caps	IETF RFC 6809 [143]	As specified in TS 24.229 [5], clause 4.4	As specified in TS 24.229 [5], clause 4.4
14	Priority (NOTE 6)	IETF RFC 7090 [184]	As specified in TS 24.229 [5], clause 4.4	As specified in TS 24.229 [5], clause 4.4
15	"iotl" SIP URI parameter (NOTE 7)	IETF RFC 7549 [188]	As specified in TS 24.229 [5], clause 4.4	As specified in TS 24.229 [5], clause 4.4
16	"cpc" tel URI parameter (NOTE 5)	TS 24.229 [5] clause 7.2A.12	As specified in TS 24.229 [5], clause 4.4	As specified in TS 24.229 [5], clause 4.4
17	"oli" tel URI parameter (NOTE 5)	TS 24.229 [5] clause 7.2A.12	As specified in TS 24.229 [5], clause 4.4	As specified in TS 24.229 [5], clause 4.4
18	Restoration-Info (NOTE 2)	TS 24.229 [5] clause 7.2.11	As specified in TS 24.229 [5], clause 4.4	As specified in TS 24.229 [5], clause 4.4
19	Relayed-Charge (NOTE 4)	TS 24.229 [5] clause 7.2.12	As specified in TS 24.229 [5], clause 4.4	As specified in TS 24.229 [5], clause 4.4
20	Service-Interact-Info	TS 24.229 [5] clause 7.2.14	As specified in TS 24.229 [5], clause 4.4	As specified in TS 24.229 [5], clause 4.4

NOTE 1: For a roaming II-NNI, a trust relationship with respect to this header field is required.

NOTE 2: This header field is only applicable on a roaming II-NNI, whereas for the interconnect NNI it is left unspecified

NOTE 3: In addition, value-dependent operator policies may be applied.

NOTE 4: This header field is not applicable at II-NNI.

NOTE 5: The tel URI parameters "cpc" and "oli" can be included in the URI in the P-Asserted-Identity header field.

NOTE 6: Only the "psap-callback" value is part of the trust domain.

NOTE 7: The "iotl" SIP URI parameter can be transported in the Request-URI, Route header field, Path header field, Service-Route header field, "+g.3gpp.trf" header field parameter, "+g.3gpp.atcf-mgmt-uri" header field parameter and in the "ATU-STI" parameter in the "application/vnd.3gpp.srvcc-info+xml" MIME body.

Items stricken out in the table above are not in the scope of this I3 Forum Release, and items underlined are modifications or additions.

#### 6.1.1.3.3 Derivation of applicable SIP header fields from 3GPP TS 24.229 [5]

For any method in table 6.1, the SIP header fields applicable on the II-NNI are detailed in the corresponding method tables for the UA role and proxy role sending behavior in Annex A of 3GPP TS 24.229 [5]. Unless other information is specified in the normative part of the present specification, the applicability of header fields at the II-NNI can be derived for each method from the corresponding tables in annex A of 3GPP TS 24.229 [5] as follows:

- All header fields not present in the corresponding tables in Annex A of 3GPP TS 24.229 or marked as "n/a" in both the "RFC status" and "profile status" columns for the UA role and proxy role sending behaviour of that tables are not applicable at the II-NNI.

NOTE 1: Operators could choose to apply header fields for other SIP extensions on an II-NNI based on bilateral agreements, but this is outside the scope of the present specification.

- All header fields which are marked as "o" in at least one of the "RFC status" or the "profile status" profile columns for the sending behaviour in the corresponding UA role and proxy role tables in annex A of 3GPP TS 24.229 [5] and as "n/a" or "o" in the other such columns are applicable at II-NNI based on bilateral agreement between operators.
- All header fields which are marked as "m" in at least one of the "RFC status" or the "profile status" columns for the sending behaviour in the corresponding UA role or proxy role table in annex A of 3GPP TS 24.229 [5] and as "n/a", "o", or "m" in the other such columns are applicable at the II-NNI.
- If conditions are specified, they are also applicable at the II-NNI and the above rules are applicable to the "n/a", "o" and "m" values within the conditions.

NOTE 2: In the above rules, the RFC profile columns are taken into account in order to enable interworking with non-3GPP networks,

An informative summary of SIP header fields to be used over the II-NNI is proposed in annex A.

#### 6.1.1.3.4 Applicability of SIP header fields on a roaming II-NNI

The following SIP header fields are applicable on a roaming II-NNI but not on a non-roaming II-NNI:

- Authentication-Info
- Authorization
- P-Associated-URI
- P-Called-Party-ID
- P-Preferred-Service
- P-Profile-Key
- P-Served-User
- P-Visited-Network-ID
- Path
- Proxy-Authenticate
- Proxy-Authorization
- Resource-Share
- Restoration-Info
- Service-Route

- WWW-Authenticate

#### 6.1.1.3.5 Applicability of SIP header fields on a non-roaming II-NNI

The following SIP header fields are only applicable on a non-roaming II-NNI:

- P-Refused-URI-List

#### 6.1.1.4 Notations of the codes

Moved to Section 6.1.1.1.

#### 6.1.1.5 Modes of signalling

Overlap signalling may be used if agreement exists between operators to use overlap and which method to be used, otherwise enbloc shall be used at the II-NNI.

### 6.1.2 SDP protocol

#### 6.1.2.1 General

The functional entity closest to the border of an II-NNI (see reference model in Clause 5) shall provide the capabilities specified for that network element in Annex A.3 of 3GPP TS 24.229 [5].

The SDP bodies shall be encoded as described in IETF RFC 3261 [13] and in IETF RFC 4566 [147]. The offer/answer model with the SDP as defined in IETF RFC 3264 [146] shall be applied.

### 6.1.3 Major capabilities

This subclause contains the major capabilities to be supported over the II-NNI.

The table 6.1.3.1 specifies which capabilities are applicable for II-NNI. The profile status codes within table 6.1.3.1 are defined in table 6.1.3.2. For the "Basic SIP" capabilities part of table 6.1.3.1, the last column "Profile status over II-NNI" specifies the general status of applicability of the IETF RFC 3261 [13] main mechanisms described in the 2<sup>nd</sup> column "Capability over the Ici".

For the "Extensions to basic SIP" capabilities part, the last column "Profile status over II-NNI" specifies the general status of applicability of the RFC referenced in the 2<sup>nd</sup> column "Capability over the Ici". If necessary, the applicability of RFCs at the II-NNI level is further detailed in the present Technical Specification.

The columns "Reference item in 3GPP TS 24.229 [5] for the profile status" provide informative references for comparison purposes into the UA and Proxy role major capabilities tables in 3GPP TS 24.229 [5], where the capabilities are defined via additional references.

**Table 6.1.3.1: Major capabilities over II-NNI**

Item	Capability over the Ici	Reference item in TS 24.229 [5] for the profile status		Profile status over II-NNI (non-roaming)	Profile status over II-NNI (roaming)
		UA Role (NOTE 1)	Proxy role (NOTE 2)		
	<b>Basic SIP (IETF RFC 3261 [13])</b>				
1	registrations	1, 2, 2A	-	<del>e2</del> n/a	<del>e2</del> m
2	initiating a session	2B, 3, 4	-	m	m
3	terminating a session	5	3	m	m
4	<del>General proxy behaviour</del>	-	4, 5, 14, 15	n/a	n/a
5	Managing several responses due to forking	9,10	6	m	m
6	<del>support of indication of TLS connections in the Record-Route header</del>	-	7, 8	n/a	n/a
7	Support of authentication	7, 8, 8A	8A	<del>e2</del> n/a	<del>e2</del> m
8	Timestamped requests (Timestamp header field)	6	-	m	m
9	Presence of date in requests and responses (Date header field)	11	9	<del>m</del> o	<del>m</del> o
10	Presence of alerting information data (Alert-info header field)	12	10	o	o
11	Support and handling of the Require header field for REGISTER and other requests or responses for methods other than REGISTER	-	11, 12, 13	m	m
12	Support and reading of the Supported and Unsupported header fields	-	16, 17, 18	m	m
13	Support of the Error-Info header field in 3xx - 6xx responses	-	19	o	o
14	Support and handling of the Organization header field	-	19A, 19B	m	m
15	Support and handling of the Call-Info header field	-	19C, 19D	m	m
16	Support of the Contact header field in 3xx response	-	19E	m	m
16A	<del>Proxy reading the contents of a body or including a body in a request or response</del>	-	19F	n/a	n/a
	<b>Extensions to basic SIP</b>				
16B	<del>TS 24.237 [131]: proxy modifying the content of a body</del>	-	19G	n/a	n/a
17	IETF RFC 6086 [39]: SIP INFO method and package framework	13	20	o	o
17A	IETF RFC 6086 [39]: legacy INFO usage	13A	20A	o	o
18	IETF RFC 3262 [18]: reliability of provisional responses in SIP (PRACK method)	14	21	m	m
19	IETF RFC 3515 [22]: the SIP REFER method	15	22	o	<u>o</u> NOTE i3F-1
19A	<del>draft-ietf-sipcore-refer-clarifications [194]: Clarifications for the use of REFER with RFC6665</del>	15A	22A	n/a	n/a
19B	draft-ietf-sipcore-refer-explicit-subscription [195]: Explicit Subscriptions for the REFER Method	15B	22B	o	o
20	IETF RFC 3312 [40] and IETF RFC 4032 [41]: integration of resource management and SIP (Preconditions framework)	2C, 16	23	o	o
21	IETF RFC 3311 [23]: the SIP UPDATE method	17	24	m	m
22	<del>IETF RFC 3313 [42]: SIP extensions for media authorization (P-Media-Authorization header field)</del>	19	26	n/a	n/a
23	IETF RFC 6665 [20]: SIP specific event notification (SUBSCRIBE/NOTIFY methods)	20, 22, 23	27	c1	c1
23A	<del>draft-roach-sipcore-6665-clarifications [196]: A clarifications on the use of Globally Routable User Agent URIs (GRUUs) in the Session Initiation Protocol (SIP) Event Notification Framework</del>	22A	28	n/a	n/a
24	IETF RFC 3327 [43]: session initiation protocol extension header field for registering non-adjacent contacts (Path header field)	24	29	<del>e2</del> n/a	<del>e2</del> m
25	IETF RFC 3325 [44]: private extensions to the Session Initiation Protocol (SIP) for network asserted identity within trusted networks	25	30	c4	c4
26	<del>IETF RFC 3325 [44]: the P-Preferred-Identity header field extension</del>	-	-	n/a	n/a
27	IETF RFC 3325 [44]: the P-Asserted-Identity header field extension		-	c4	c4
28	IETF RFC 3323 [34], IETF RFC 3325 [44] and IETF RFC 7044 [25]: a privacy mechanism for the Session Initiation Protocol (SIP) (Privacy header field)	26, 26A, 26B, 26C, 26D, 26E, 26F, 26G, 26H	31, 31A, 31B, 31C, 31D, 31E, 31F, 31G, 31H	m	m

29	IETF RFC 3428 [19]: a messaging mechanism for the Session Initiation Protocol (SIP) (MESSAGE method)	27	33	e m NOTE i3F-2	e m NOTE i3F-2
30	IETF RFC 3608 [45]: session initiation protocol extension header field for service route discovery during registration (Service-Route header field)	28	32	e2 n/a	e2 m
31	<del>IETF RFC 3486 [46]: compressing the session initiation protocol</del>	29	34	n/a	n/a
32	IETF RFC 7315 [24]: private header extensions to the session initiation protocol for the 3 <sup>rd</sup> -Generation Partnership Project (3GPP)	30	35	o	e m
32A	<del>IETF RFC 3325 [44]: act as first entity within the trust domain for asserted identity</del>	30A	30A	n/a	n/a
32B	<del>IETF RFC 3325 [44]: act as entity within trust network that can route outside the trust network</del>	30B	30B	n/a	n/a
32C	<del>IETF RFC 3325 [44]: act as entity passing on identity transparently independent of trust domain</del>	30C	30C	n/a	n/a
33	IETF RFC 7315 [24]: the P-Associated-URI header field extension	31	36	e2 n/a	e2 m
34	IETF RFC 7315 [24]: the P-Called-Party-ID header field extension	32	37	e2 n/a	e2 m
35	IETF RFC 7315 [24]: the P-Visited-Network-ID header field extension	33	38, 39	e2 n/a	e2 m
36	IETF RFC 7315 [24]: the P-Access-Network-Info header field extension	34	41, 42, 43	e4 n/a	e4 o
37	<del>IETF RFC 7315 [24]: the P-Charging-Function-Addresses header field extension</del>	35	44, 44A	n/a	n/a
38	IETF RFC 7315 [24]: the P-Charging-Vector header field extension	36	45, 46	c1	c1
39	<del>IETF RFC 3329 [47]: security mechanism agreement for the session initiation protocol</del>	37	47	n/a	n/a
39A	<del>TS 24.229 [5] clause 7.2A.7: Capability Exchange for Media Plane Security</del>	37A	47A	n/a	n/a
40	IETF RFC 3326 [48]: the Reason header field for the session initiation protocol	38	48	o	o
41	IETF RFC 6432 [49]: carrying Q.850 codes in reason header fields in SIP (Session Initiation Protocol) responses	38A	48A	c4	c4
42	IETF RFC 3581 [50]: an extension to the session initiation protocol for symmetric response routing	39	49	o	o
43	IETF RFC 3841 [51]: caller preferences for the session initiation protocol (Accept-Contact, Reject-Contact and Request-Disposition header fields)	40, 40A, 40B, 40C, 40D, 40E, 40F	50, 50A, 50B, 50C, 50D, 50E, 50F	m	m
44	IETF RFC 3903 [21]: an event state publication extension to the session initiation protocol (PUBLISH method)	41	51	c1	c1
45	IETF RFC 4028 [52]: SIP session timer (Session-Expires and Min-SE headers)	42	52	m o	m o
46	IETF RFC 3892 [53]: the SIP Referred-By mechanism	43	53	m o	m o NOTE i3F-1
47	IETF RFC 3891 [54]: the Session Initiation Protocol (SIP) "Replaces" header	44	54	o	o NOTE i3F-1
48	IETF RFC 3911 [55]: the Session Initiation Protocol (SIP) "Join" header	45	55	o	o NOTE i3F-1
49	IETF RFC 3840 [56]: the callee capabilities	46	56	o	o
50	IETF RFC 7044 [25]: an extension to the session initiation protocol for request history information (History-Info header field)	47	57	o	o
50A	IETF RFC 7044 [25]: the "mp" header field parameter	47A	57A	o	o
50B	IETF RFC 7044 [25]: the "rc" header field parameter	47B	57B	o	o
50C	IETF RFC 7044 [25]: the "np" header field parameter	47C	57C	o	o
51	IETF RFC 5079 [57]: Rejecting anonymous requests in the session initiation protocol	48	58	o	o
52	IETF RFC 4458 [58]: session initiation protocol URIs for applications such as voicemail and interactive voice response (NOTE 3)	49	59	o	o
52A	<del>draft-mohali-dispatch-cause-for-service-number [193]: Session Initiation Protocol (SIP) Cause-URI parameter for Service-Number translation</del>	49A	59A	e	e
53	IETF RFC 4320 [59]: Session Initiation Protocol's (SIP) non-	50	61	m	m

	INVITE transactions				
54	<del>IETF RFC 4457 [60]: the P-User-Database private header field extension</del>	51	60	n/a	n/a
55	<del>IETF RFC 5031 [61]: A Uniform Resource Name (URN) for Emergency and Other Well-Known Services</del>	52	62	n/a	n/a
56	IETF RFC 5627 [62]: obtaining and using GRUUs in the Session Initiation Protocol (SIP)	53	63	e4	e4 <u>m</u>
57	Void				
58	IETF RFC 4168 [27]: the Stream Control Transmission Protocol (SCTP) as a Transport for the Session Initiation Protocol (SIP)	55	65	o	o
59	IETF RFC 5002 [64]: the SIP P-Profile-Key private header field extension	56	66, 66A, 66B	e3	e3 <u>o</u>
60	IETF RFC 5626 [65]: managing client initiated connections in SIP	57	67	e4	e4 <u>m</u>
61	<del>IETF RFC 5768 [66]: indicating support for interactive connectivity establishment in SIP</del>	58	68	n/a	n/a
62	IETF RFC 5365 [67]: multiple-recipient MESSAGE requests in the session initiation protocol	59	69	o if 29, else n/a	o if 29, else n/a
63	IETF RFC 6442 [68]: Location conveyance for the Session Initiation Protocol	60	70, 70A, 70B	m o	m o
64	IETF RFC 5368 [69]: referring to multiple resources in the session initiation protocol	61	71	o if 19, else n/a	o if 19, else n/a
65	IETF RFC 5366 [70]: conference establishment using request-contained lists in the session initiation protocol	62	72	o	o
66	IETF RFC 5367 [71]: subscriptions to request-contained resource lists in the session initiation protocol	63	73	o if 23, else n/a	o if 23, else n/a
67	IETF RFC 4967 [72]: dialstring parameter for the session initiation protocol uniform resource identifier	64	74	e2 n/a	e2 <u>m</u>
68	<del>IETF RFC 4964 [73]: the P-Answer-State header extension to the session initiation protocol for the open mobile alliance push to talk over cellular</del>	65	75	e	e
69	IETF RFC 5009 [74]: the SIP P-Early-Media private header field extension for authorization of early media	66	76	c4	c4
70	IETF RFC 4694 [75]: number portability parameters for the 'tel' URI	67, 67A, 67B	77, 77A, 77B	o	o
71	Void				
72	<del>IETF RFC 4411 [77]: extending the session initiation protocol Reason header for preemption events</del>	69	79	e	e
73	<del>IETF RFC 4412 [78]: communications resource priority for the session initiation protocol (Resource Priority header field)</del>	70, 70A, 70B	80, 80A, 80B	e	e
74	IETF RFC 5393 [79]: addressing an amplification vulnerability in session initiation protocol forking proxies	71	81	m	m
75	<del>IETF RFC 5049 [80]: the remote application identification of applying signalling compression to SIP</del>	72	82	n/a	n/a
76	IETF RFC 5688 [81]: a session initiation protocol media feature tag for MIME application sub-types	73	83	e4 <u>o</u>	e4 <u>m</u>
77	IETF RFC 6050 [26]: Identification of communication services in the session initiation protocol	74	84, 84A	e	o
78	<del>IETF RFC 5360 [82]: a framework for consent-based communications in SIP</del>	75, 75A, 75B	85	e	e
79	IETF RFC 7433 [83]: a mechanism for transporting user-to-user call control information in SIP	76	86	e4 <u>o</u>	e4 <u>m</u>
79A	IETF RFC 7434 [83A]: interworking ISDN call control user information with SIP	76A	-	e4 <u>o</u>	e4 <u>m</u>
80	<del>IETF RFC 7316 [84]: The SIP P-Private-Network-Indication private header (P-Header)</del>	77	87	e4	e4
81	IETF RFC 5502 [85]: the SIP P-Served-User private header	78	88	e2 n/a	e2 <u>m</u>
82	Void				
83	<del>draft-dawes-sipping-debug-04 [87]: the P-Debug-ID header extension</del>	80	90	e	e
84	IETF RFC 6228 [88]: the 199 (Early Dialog Terminated) response code	81	91	m	m
85	IETF RFC 5621 [89]: message body handling in SIP	82	92	m	m
86	<del>IETF RFC 6223 [90]: indication of support for keep-alive</del>	83	93	e	e
87	<del>IETF RFC 5552 [91]: SIP Interface to VoiceXML Media Services</del>	84	94	n/a	n/a
88	IETF RFC 3862 [92]: common presence and instant	85	95	o	o

	messaging (CPIM): message format				
89	IETF RFC 5438 [93]: instant message disposition notification	86	96	o	o
90	<del>IETF RFC 5373 [94]: requesting answering modes for SIP (Answer-Mode and Priv-Answer-Mode header fields)</del>	87	97, 97A	e	e
91	Void				
92	IETF RFC 3959 [96]: the early session disposition type for SIP	89	99	e	o
93	Void				
94	draft-ietf-insipid-session-id [124]: End-to-End Session Identification in IP-Based Multimedia Communication Networks	91	101	e	o
95	IETF RFC 6026 [125]: correct transaction handling for 200 responses to Session Initiation Protocol INVITE requests	92	102	m	m
96	IETF RFC 5658 [126]: addressing Record-Route issues in the Session Initiation Protocol (SIP)	93	103	o	o
97	IETF RFC 5954 [127]: essential correction for IPv6 ABNF and URI comparison in IETF RFC 3261 [13]	94	104	m	m
98	IETF RFC 4488 [135]: suppression of session initiation protocol REFER method implicit subscription	95	105	m if 19, else n/a	m if 19, else n/a
99	IETF RFC 7462 [136]: Alert-Info URNs for the Session Initiation Protocol	96	106	e	o
100	TS 24.229 [5] clause 3.1: multiple registrations	97	107	<del>c2</del> n/a	<del>e2</del> m
101	<del>IETF RFC 5318 [141]: the SIP P-Refused-URI-List private-header</del>	98	108	<del>e5</del>	<del>e5</del>
102	IETF RFC 4538 [140]: request authorization through dialog Identification in the session initiation protocol (Target-Dialog header field)	99	109	e	o
103	IETF RFC 6809 [143]: Mechanism to indicate support of features and capabilities in the Session Initiation Protocol (SIP)	100	110	e	o
104	IETF RFC 6140 [160]: registration of bulk number contacts	101	111	<del>e3</del> n/a	<del>e3</del> o
105	IETF RFC 6230 [161]: media control channel framework	102	112	o	o
105A	TS 24.229 [5] clause 4.14: S-CSCF restoration procedures	103	113	<del>e3</del> n/a	<del>e3</del> o
106	IETF RFC 6357 [164]: SIP overload control	104	114	o	o
107	IETF RFC 7339 [165]: feedback control	104A	114A	o	o
108	IETF RFC 7200 [167]: distribution of load filters	104B	114B	o	o
109	<del>TS 24.229 [5] clauses 5.1.2A.1.1, 5.1.3.1, 5.1.6.8, and 5.2.10: Handling of a 380 (Alternative service) response</del>	105	115	n/a	n/a
110	IETF RFC 7090 [184]: Public Safety Answering Point (PSAP) Callback	107	117	o	o
111	<del>draft-holmberg-sipcore-received-realm [185]: Via header field parameter to indicate received realm</del>	106	116	n/a	n/a
112	IETF RFC 7549 [188]: SIP URI parameter to indicate traffic leg	108	118	o (NOTE 4)	o (NOTE 4)
113	TS 24.229 [5] clause 4.14: PCRF based P-CSCF restoration	109	119	<del>e3</del> n/a	<del>e3</del> o
114	TS 24.229 [5] clause 4.14: HSS based P-CSCF restoration	110	120	<del>e3</del> n/a	<del>e3</del> o
115	<del>TS 24.229 [5] clause 7.2.12: the Relayed-Charge header extension</del>	111	121	n/a	n/a
116	TS 24.229 [5]: resource sharing	112	122	<del>e3</del> n/a	<del>e3</del> o
c1: m in case of roaming II-NNI, else o c2: m in case of roaming II-NNI, else n/a c3: o in case of roaming II-NNI, else n/a c4: m in case of trust relationship between the interconnected networks, else n/a c5: o in case of non-roaming II-NNI and loopback traversal scenario, else n/a NOTE 1: The item numbering corresponds to the one provided in table A.4 in TS 24.229 [5]. NOTE 2: The item numbering corresponds to the one provided in table A.162 in TS 24.229 [5]. NOTE 3: A common URI namespace is required to apply this feature on the II-NNI. NOTE 4: For the roaming II-NNI the support of this major capability is recommended. NOTE i3F-1: <a href="#">Needed to support CONF service as specified within TS 24.147 [106] Section 5.3.1.5.3</a> NOTE i3F-2: Mandatory for SMS over IP					

**Table 6.1.3.2: Key to notation codes for major capabilities**

Notation code	Notation name	Explanation
M	mandatory	The capability shall be supported at II-NNI. SIP message relating to this capability shall be sent over the II-NNI if received from the serving network, unless they also make use of other unsupported capabilities. SIP headers or other information elements relating to this capability shall be passed over the II-NNI if received from the sending side. This does not imply that network elements inside the serving network or served network or user equipment connected to these networks shall support this capability.
O	optional	The capability may or may not be supported at II-NNI. The support of the capability is provided based on bilateral agreement between the operators ( <a href="#">i.e. Service Provider and/or carriers according to I3Forum terminology</a> ).
n/a	not applicable	It is impossible to use/support the capability at the II-NNI.
c <integer>	conditional	The support of the capability ("m", "o" or "n/a") depends on the support of other optional or conditional items. <integer> is the identifier of the conditional expression.

## 6.2 Control Plane Transport

### 6.2.1 General

The control plane transport of the II-NNI shall comply with clause 4.2A of 3GPP TS 24.229 [5]. Support of SCTP as specified in IETF RFC 4168 [27] is optional for an IBCF connected by II-NNI. Nevertheless this option is favourable if the operators would like to improve reliability over the ICI.

## 6.3 SIP timers

Table 6.3.1 shows values of SIP timers that should be supported at II-NNI. It contains the following items:

- the first column, titled "SIP Timer", shows the timer names as defined in IETF RFC 3261 [13] or IETF RFC 6026 [125];
- the second column reflects the timer meaning as defined in IETF RFC 3261 [13];
- the third column reflects the reference to the proper section in the IETF RFC 3261 [13] and in 3GPP TS 24.229 [5] and
- the final column lists the values recommended for the functional entities closest to the border of an II-NNI (see reference model in clause 5).

Table 6.3.1 reports information from 3GPP TS 24.229 [5], table 7.7.1. Values between IM CN subsystem elements shown in the second column in 3GPP TS 24.229 [5], table 7.7.1 are applicable for the II-NNI and are reported in the fourth column of table 6.3.1. If there are any differences between table 6.3.1 and 3GPP TS 24.229 [5], table 7.7.1, the information within 3GPP TS 24.229 [5], table 7.7.1 is applicable.

**Table 6.3.1: SIP timers at II-NNI**

SIP Timer	Meaning	Reference	Recommended values
T1	RTT estimate	[13] clause 17.1.1.1 [5] table 7.7.1	500ms default (see NOTE)
T2	The maximum retransmit interval for non-INVITE requests and INVITE responses	[13] clause 17.1.2.2 [5] table 7.7.1	4s (see NOTE)
T4	Maximum duration a message will remain in the network	[13] clause 17.1.2.2 [5] table 7.7.1	5s (see NOTE)
Timer A	INVITE request retransmit interval, for UDP only	[13] clause 17.1.1.2 [5] table 7.7.1	initially T1
Timer B	INVITE transaction timeout timer	[13] clause 17.1.1.2 [5] table 7.7.1	64*T1
Timer C	proxy INVITE transaction timeout	[13] clause 16.6 [5] table 7.7.1	> 3min
Timer D	Wait time for response retransmits	[13] clause 17.1.1.2 [5] table 7.7.1	> 32s for UDP
		[13] clause 17.1.1.2 [5] table 7.7.1	0s for TCP/SCTP
Timer E	non-INVITE request retransmit interval, UDP only	[13] clause 17.1.2.2 [5] table 7.7.1	initially T1
Timer F	non-INVITE transaction timeout timer	[13] clause 17.1.2.2 [5] table 7.7.1	64*T1
Timer G	INVITE response retransmit interval	[13] clause 17.2.1 [5] table 7.7.1	initially T1
Timer H	Wait time for ACK receipt.	[13] clause 17.2.1 [5] table 7.7.1	64*T1
Timer I	Wait time for ACK retransmits	[13] clause 17.2.1 [5] table 7.7.1	T4 for UDP
		[13] clause 17.2.1 [5] table 7.7.1	0s for TCP/SCTP
Timer J	Wait time for non-INVITE request retransmits	[13] clause 17.2.2 [5] table 7.7.1	64*T1 for UDP
		[13] clause 17.2.2 [5] table 7.7.1	0s for TCP/SCTP
Timer K	Wait time for response retransmits	[13] clause 17.1.2.2 [5] table 7.7.1	T4 for UDP
		[13] clause 17.1.2.2 [5] table 7.7.1	0s for TCP/SCTP
Timer L	Wait time for accepted INVITE request retransmits	[125] clause 8.11 [5] table 7.7.1	64*T1
Timer M	Wait time for retransmission of 2xx to INVITE or additional 2xx from other branches of a forked INVITE	[125] clause 8.11 [5] table 7.7.1	64*T1
NOTE:	As a network option, SIP T1 Timer's value can be extended, along with the necessary modifications of SIP T2 and SIP T4 Timer values, to take into account the specificities of the supported services when the MRFC and the controlling AS are under the control of the same operator and the controlling AS knows, based on local configuration, that the MRFC implements a longer value of SIP T1 Timer.		

## 7 RCS

The, here defined, signalling profile is intended to also support RCS over the Interconnect and Roaming NNI. Typically, in RCS 5.1 the following services are supported:

- Stand-alone messaging
- 1-to-1 chat
- Group Chat
- File Transfer
- Content Sharing
- Social Presence Information
- IP Voice Call
- Best Effort Video Call
- Geo-Location Exchange
- Network based Blacklist
- Capability Exchange (based on Presence or SIP OPTIONS)

To support all (or part of) these services over the II-NNI, compared to previous versions of this document, more SIP Methods and capabilities need to be supported over the interface. However it depends on bilateral agreement between the Service Providers whether RCS and specific services defined in RCS will be supported between their end-users and/or for roaming end-users in each others networks. Therefore, in this signalling profile, several SIP Methods and Capabilities are now defined here as 'optional'.

## 8 i3 Forum Recommendations

*It is recommended that this Interconnection Signalling Profile should be supported as the minimal profile on the Inter-IMS Network to Network Interface (II-NNI) for basic voice, SMS, video and RCS services in both Interconnect and Roaming scenarios. It is recommended that bilateral agreements describe the support/transparency (or not) of optional elements.*