Testspezifikation für NGN Interconnection Kompatibilitätstests

Version: 3.0.0

Herausgegeben vom Arbeitskreis technische und betriebliche Fragen der Nummerierung und Netzzusammenschaltung.

Erarbeitet vom Unterarbeitskreis Signalisierung

Editor: Thomas Fischer, Deutsche Telekom AG

0 Inhalt

1 History

Version	Datum	Änderung
1.0.0	19.02.2013	Verabschiedet auf der 140. Tagung des AKNN in Köln
1.0.1	23.02.2015	Zu überprüfende Elemente des Notrufs eingefügt
1.0.2	23.02.2015	Änderungen zu CDIV, multiple early dialogues Ansagen in Fall von CS multiple Call Forwarding
1.0.3	24.02.2015	Einfügung des P-Asserted-Identity in 181 in Fall von Call Forwarding und editorielle Änderungen.
1.0.4	16.11.2015	Ergebnis Editorentagung 12.11.2015
1.0.5	07.12.2015	Ergebnis UAKS 142 und Editorentagung vom 30.11.2015: Korrektur Forking, Klarstellung CFB als „user determined user busy“, editorielle Änderungen.
1.0.6	23.02.2016	Ergebnis UAKS 143: Entfernen Note aus Testfall „SS_bcall_029“
1.1.0	10.05.2016	Verabschiedet auf der 144. Tagung des UAKS
1.1.1	26.06.2017	ANNEX A wurde, beschlossen in der 147. Tagung, aus dem Dokument genommen. Editorielle Anpassungen.
1.1.2	14.06.2018	Umstieg auf ETSI TS 101 585 als Referendokument.
1.1.3	16.06.2018	Editorielle Änderung
1.1.4	12.09.2018	Editorielle Änderung
1.1.5	17.09.2018	Editorielle Änderung zum UAKS 153
1.1.6	09.11.2018	Vorlage zur Verabschiedung im UAKS
1.1.7	22.11.2018	Editoriell berarbeitete Vorlage zur Verabschiedung im UAKS
1.2.0	18.12.2018	Vom UAKS verabschiedete Version
2.0.1	19.09.2022	Erweiterung um Testfälle zu Blocking of Calls und International Calls für das Kapitel 15 „Procedures to fulfill TKG §120 Requirements for Number Transmission“
2.0.2	20.09.2022	Ergebnis UAKS 170: Aufnahme der überarbeiteten Testfälle zu TKG §120 Requirements for Number Transmission und Aufnahme von Testfällen ziu unterschiedlichen Privacy Szenarien. Erweitung des Kaptiel 5.2 Definition um einen Hinweis zur Behandlung von Response Code mit abwiechenden reason phrase.
2.1.0	20.12.2022	Verabschiedet auf der 171. Tagung des UAKS
3.0.0	09.05.2023	Verabschiedet auf der 200. Tagung des AKNN

2 Foreword

This technical specification defines End-to-End compatibility test specification between two national network operators. The test cases defined in this document are the basis for a bilateral test campaign. If a test campaign between two national network operators is agreed, exactly the test cases described in this document shall be performed. In addition further test cases may be necessary and will be defined and agreed multilaterally

3 Scope

This technical specification refers to the ETSI test specification TS 101 585 [2]. Further is described in clause "Endorsement notice"
This technical specification provides the test cases for the test interconnetion test phase before starting the live traffic between national network operators to avoid complications due to a compatibility mismatch.
The "Arbeitskreises für technische und betriebliche Fragen für Nummerierung und Netzzusammenschaltung (AKNN)" agreed to perform standardized test cases to ensure untroubled traffic.
Before starting the real traffic, using the test case defined in this document, the compatibility between the End Devices in the particular networks could be verified and in addition problems with network internal configuration issues. Network internal procedures are out of scope regarding this document.
The point of observation is the common Ici and the Izi interface as described in [1]

4 References and <u>Endorsement notice</u>

4.1 References

The following referenced documents are necessary for the application of the present document. The reference in [1] and [2] applies.

[1] Unterarbeitskreis Signalisierung Specification of the NGN-Interconnection Interface, Version 4.0.0 vom 13.09.2022,
herausgegeben vom Arbeitskreis für technische und betriebliche Fragen der Nummerierung und Netzzusammenschaltung (AKNN)

[2] ETSI TS 101 585 V2.1.1 (2018-02) Core Network and Interoperability Testing (INT); IMS interconnection tests at the Ic Interface; Test Suite Structure and Test Purposes (TSS&TP)

[3] IETF RFC 3323 (2002): A Privacy Mechanism for the Session Initiation Protocol (SIP)

[4] IETF RFC 3325 (2002): Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks

[5] IETF RFC 7044 (2014): An Extension to the Session Initiation Protocol (SIP) for Request History Information

4.2 Endorsement notice

The following section describes the endorsement of ETSI TS 101 585 [2] for the german national networks

Markings used within the text:

Text modified due to AKNN requirements that is added or deleted compared to ETSI TS 101 585 is shown as

- underlined and blue (example Unterarbeitskreis Signalisierung added text)
- or cursive and stricken and red (*~~example Unterarbeitskreis Signalisierung stricken text~~*)

In case a clause number referres to document [2] it is presented with a "§", i.e. §7.5 referres to chapter "Emergency call" in [2].

5 Definitions and Abbreviations

5.1 Abbreviations

For the purposes of the present document, the abbreviations given in [2] and the following apply:

AKNN	Arbeitskreis Nummerierung und Netzzusammenschaltung
CDR	Charging Data Record
DSLAM	Digital Subscriber Line Access Multiplexer
IAD	Integrated Access Device
IC	Interconnection
ICP	Interconnection Partner
IP	Internet Protokoll
LQO	Listening Quality Objective
MOS	Mean Opinion Score
NGN	Next Generation Network
PESQ	Perceptual evaluation of speech quality
PoI	Point of Interconnection
QoS	Quality of Service
PSAP	Public Service Answering Point
RTP	Real Time Transport Protocol
RTCP	Real Time Transport Control Protocol
RTCP-XR	Real Time Transport Control Protocol Extended Report
SBC	Session Border Controller

UAK-S	Unterarbeitskreis Signalisierung des AKNN
VNB	Transit network operator (Verbindungsnetzbetreiber)

5.2 Definitions

For the purposes of the present document, the terms and definitions given in ETSI TS 101 585 [2] apply:

5.2.1 Reference configuration

The reference configuration is described in ETSI TS 101 585 [2] clause 5.3

5.2.2 Note on the evalution of tests

Response codes to be checked in the test may also contain reason phrase different from the default value. Such Response codes are treated in th same way as Response codes with default reason phrase.

5.3 Test case selection

5.3.1 Selection of End Devices

Which test cases are applicalbe during the test campagne sdepends on the used End devices in the particular network, could be chosen as described in ETSI TS 101 585 [2] clause 5.3

5.3.2 Selection Expressions

The supported network options in the particular national network could be identified in ETSI TS 101 585 [2] clause 6. Additional requirements are described in this clause.

Table 6-1: Selection expression applicable in the Test Purposes

SELECTION EXPRESSION:		Support Network A	Support Network B
Network capabilities			
SE 1:	The originating network (Network A) sends the P-Charging-Vector header?		
SE 2:	The originating network (Network A) sends a subset of parameters in the P-Charging-Vector header?		
SE 3:	The P-Early-Media header is supported?		
SE 4:	Overlap procedure using multiple INVITE method is supported?		
~~**SE 5:**~~	~~*Overlap sending using in-dialog method is supported?*~~		
SE 6:	The Network supports the PSTN XML schema?		
SE 7:	The resource reservation procedure is supported?		
SE 8:	Does the network perform the "Fall back" procedure (PSTN or MGCF)?		
SE 9:	The network is untrusted?		
SE 10:	Originating network does not have a number portability data base, the number portability look up is done in the interconnected network?		
SE 11:	The network supports the REFER method?		
SE 12:	The Network supports the 3 party call control procedure (REFER interworking)?		
SE 13:	The Number Portability is supported?		
SE 14:	Carrier Selection is performed?		
SE 15:	The Network is a Long distance carrier?		
SE 16:	SIP Support of Charging is supported?		
SE 17:	The interworking ISUP - SIP I is performed in the network?		
SE 17a:	The Network supports the Session Timers in the Session Initiation Protocol (SIP)?		
SE 17b:	The Network supports the forking of INVITE requests?		
SE 17c:	The called network hosts the public savety answering point?		
SE 17d:	The calling network supports the P-Germany-Tariff header?		
SE 17e:	The called network hosts a service in offline charging?		
SE 17f:	The network B supports Call Blocking service for prohibited calling numbers according to TKG §120?		
SE 17g:	The network A has international Interconnection or supports transit of international calls?		
Supplementary services			
SE 18:	The network supports the Originating Identification Presentation (OIP)?		
SE 19:	The network supports the "Special arrangement" procedure for the originating user?		
SE 20:	The network supports the Originating Identification Restriction (OIR)?		
SE 20a:	The network A supports privacy service with different settings of the privacy header which are not covered by the OIR service?		
SE 21:	The Network supports the Terminating Identification Presentation (TIP)?		
SE 22:	The network supports the "Special arrangement" procedure for the terminating user?		
SE 23:	The Network supports the Terminating Identification Restriction (TIR)?		
SE 24:	The Network supports the session HOLD procedure?		
SE 25:	The network supports Communication Forwarding Unconditional (CFU)?		
SE 26:	The network supports Communication Forwarding Busy (CFB)?		
SE 27:	The network supports Communication Forwarding No Reply (CFNR)?		
SE 28:	The Network supports Communication Forwarding Not Logged in (CFNL)?		
SE 29:	The Network supports Communication Deflection?		
SE 30:	The Network supports the CDIV Notification procedure?		
SE 31:	The Network supports conference (CONF)?		

SELECTION EXPRESSION:		Support	Support
		Network A	Network B
SE 32:	The Network supports the Communication Barring procedure (CB) - (Black list for incoming calls)?		
SE 33:	The Network supports the Anonymous Communication Rejection (ACR)?		
SE 34:	The Network supports the Closed User Group (CUG)?		
SE 35:	The Network supports the Communication Waiting (CW) service?		
SE 36:	The Network supports the T_{AS-CW} timer?		
SE 37:	The Network supports Explicit Communication Transfer (ECT)?		
SE 38:	The network supports Malicious Communication Identification (MCID)?		
SE 39:	The Network supports Message Waiting Indication (MWI)?		
SE 40:	The Network supports Completion of Communications to Busy Subscriber (CCBS)?		
SE 41:	The Network supports Completion of Communications by No Reply (CCNR)?		
	Terminal capabilities		
SE 42:	The End device requires the resource reservation?		
SE 43:	The End device supports Fax transmission via G.711 codec?		
SE 44:	The End device supports Fax transmission via V.152 codec?		
SE 45:	The End device supports Fax transmission via m-line T.38 codec?		
SE 46:	A SIP end device is used supporting an ISDN user equipment and the PSTN XML Schema is used?		
SE 47:	End device is located in the PSTN or PLMN?		
SE 48:	The terminating UE supports the from-change tag procedure **and sends a second user identity in an UPDATE request after the dialogue is confirmed?**		
SE 49:	The end device performs ECT using the 'Blind/assured transfer'?		
SE 50:	The end device performs ECT using the 'Consultative transfer'?		
SE 51:	The end device supports the **Resource reservation** procedure?		
	PSTN/PLMN Supplementary services		
SE 52:	CLIP/CLIR is supported in the PSTN/PLMN part of the network?		
SE 52A:	The network supports the "Special arrangement" procedure for the originating user?		
SE 53:	COLP/COLR is supported in the PSTN/PLMN part of the network?		
SE 53A:	The network supports the "Special arrangement" procedure for the terminating user?		
SE 54:	HOLD is supported in the PSTN/PLMN part of the network?		
SE 55:	CDIV unconditional is supported in the PSTN/PLMN part of the network?		
SE 55A:	CDIV busy is supported in the PSTN/PLMN part of the network?		
SE 55B:	CDIV no reply is supported in the PSTN/PLMN part of the network?		
SE 55C:	CDIV Mobile subscriber not reachable is supported in the PSTN/PLMN part of the network?		
SE 55D:	CDIV call deflection is supported in the PSTN/PLMN part of the network?		
SE 56:	CONF/3PTY is supported in the PSTN/PLMN part of the network?		
SE 57:	ACR is supported in the PSTN/PLMN part of the network?		
SE 58:	CUG is supported in the PSTN/PLMN part of the network?		
SE 59:	CW is supported in the PSTN/PLMN part of the network?		
SE 60:	ECT is supported in the PSTN/PLMN part of the network?		
SE 61:	MCID is supported in the PSTN/PLMN part of the network?		
SE 61A:	Call Completion is supported in the PSTN/PLMN part of the network?		
SE 62:	SUB is supported in the PSTN/PLMN part of the network?		
SE 63:	UUS is supported in the PSTN/PLMN part of the network?		
SE 64:	TP is supported in the PSTN/PLMN part of the network?		

6 Test purposes

Modify Section §7.1.1 "Test purposes for Basic call, Successful" as follows

Test case number	SS_bcall_001
Test case group	BCALL/successful
Comments	Establish a communication from network A to Network B **Check**: Ensure the property of speech **Check**: Are the media streams terminated after the 200 OK BYE was sent? **Check**: Optional:are the headers mentioned in table B-1 marked as '**o**' - optional **present** in the Requests and Responses? **Check**: Optional: are the headers mentioned in table B-1 marked as 'n/a' - not applicable **not present** in the Requests and Responses? Repeat this test in reverse direction. Repeat this test with all chosen end devices

Test case number	SS_bcall_002
Test case group	BCALL/successful
Comments	Establish a communication from network A to Network B **Check**: Ensure the property of speech **Check**: Are the media streams terminated after the 200 OK BYE was sent? **Check**: Optional:are the headers mentioned in table B-1 marked as '**o**' - optional **present** in the Requests and Responses? **Check**: Optional: are the headers mentioned in table B-1 marked as 'n/a' - not applicable **not present** in the Requests and Responses? Repeat this test in reverse direction. Repeat this test with all chosen end devices

<table>
<tr><td>Test case number</td><td>SS_bcall_040A</td></tr>
<tr><td>Test case group</td><td>BCALL/successful</td></tr>
<tr><td>Reference</td><td>[1]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td></td></tr>
<tr><td>Test purpose</td><td>Support of ‘isub’ URI parameter.

Ensure that call establishment is performed correctly and the INVITE contains an ‘isub’ parameter in
- To header and optionally Request URI
- P-Asserted-Identity</td></tr>
<tr><td>Configuration</td><td></td></tr>
<tr><td>SIP Parameter</td><td>INVITE
To: <userpart; isub= 0A0204;isub-encoding=user-specified@domain>

P-Asserted-Identity: <userpart; isub= 0A0204;isub-encoding=user-specified@domain>

SDP
Content-Type: application/sdp
:
:
m=audio < Port number> RTP/AVP <Dynamic PT>
a=rtpmap: <Dynamic PT> CLEARMODE/8000

Content-Type: application/vnd.etsi.cug+xml
Content-Disposition: signal;handling= ......

…….
<…cug>
<…networkIndicator>9490</… networkIndicator>
<….cugInterlockBinaryCode>0F03</…cugInterlockBinaryCode>
<…CugCommunicationIndicator>11</
…cugCommunicationIndicator>
</…cug></td></tr>
<tr><td colspan="2">Message flow</td></tr>
<tr><td colspan="2"><table>
<tr><td>SIP (Network A)</td><td colspan="3">Interconnection Interface</td><td>SIP (Network B)</td></tr>
<tr><td></td><td></td><td>INVITE</td><td>→</td><td></td></tr>
<tr><td></td><td>←</td><td>100 Trying</td><td></td><td></td></tr>
<tr><td></td><td>←</td><td>180 Ringing</td><td></td><td></td></tr>
<tr><td></td><td>←</td><td>200 OK INVITE</td><td></td><td></td></tr>
<tr><td></td><td></td><td>ACK</td><td>→</td><td></td></tr>
<tr><td></td><td></td><td>Communication</td><td></td><td></td></tr>
<tr><td></td><td>→</td><td>BYE</td><td></td><td></td></tr>
<tr><td></td><td></td><td>200 OK BYE</td><td>←</td><td></td></tr>
</table></td></tr>
<tr><td>Comments</td><td>Establish a communication from network A to Network B
Check: Is the CLEARMODE codec preset in the INVITE?
Check: Are the URI parameters ‘isub’ and ‘isub-encoding’ present in the To header or in the P-Asserted-Identity in the INVITE request?
Check: Is the CUG XML body present in the INVITE request?
Repeat this test in reverse direction.</td></tr>
</table>

Modify Section §7.1.5.9 “Closed User Group (CUG)” as follows

The Test case “SS_cug_001” is not applicable

Modify Section §7.4 "Carrier Selection" as follows

NOTE: The following test cases in this chapter only apply if Deutsche Telekom is network A.

Test case number	SS_csel_001
Test case group	SIP-SIP/CS
SIP Parameter	INVITE 1: Request line sip: + <CC> <NDC> <SN>>~~[;cic=(carrier ID)]~~@**tariff.**<hostname> user=phone SIP/2.0 INVITE 2: Request line sip: + <CC> <NDC> <SN>[;npdi] [;rn=<CC> D<xyz> <NDC> <SN>]@<hostname>; user=phone SIP/2.0
Comments	**Check**: Is the sub domain pattern '**tariff**' present at the beginning of the hostportion **only** of the initial INVITE sent from network A to network B? ~~**Check:** Is the optional 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier?~~ **Check:** Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A? **Check:** Is optional the 'rn' parameter present in the Request URI of the INVITE request sent from network B to network A? ~~**NOTE 1:** The 'cic' parameter may be absent according national regulations or national agreements.~~ ~~**NOTE 2:** It is possible that further information is available in the Request line regarding the end user charging in case of Carrier selection.~~ Repeat this test in reverse direction.

Test case number	SS_csel_002
Test case group	SIP-SIP/CS
SIP Parameter	INVITE: Request line sip: + <CC> <NDC> <SN>~~[;cic=(carrier ID)]~~@**tariff**. <hostname>; user=phone SIP/2.0 INVITE: Request line sip: + <CC> <NDC> <SN>[;npdi] [;rn=+ <CC> D<xyz> <NDC> <SN>]@<hostname>;user=phone SIP/2.0
Comments	**Check**: Is the sub domain pattern '**tariff**' present at the beginning of the hostportion **only** of the initial INVITE sent from network A to network B? ~~**Check:** Is the optional 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier?~~ **Check:** Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A? **Check:** Is optional the 'rn' parameter present in the Request URI of the INVITE request sent from network B to network A? ~~**NOTE 1:** The 'cic' parameter may be absent according national regulations or national agreements.~~ ~~**NOTE 2:** It is possible that further information is available in the Request line regarding the end user charging in case of Carrier selection.~~ Repeat this test in reverse direction.

Test case number	SS_csel_003
Test case group	SIP-SIP/CS
SIP Parameter	INVITE: Request line sip: + <CC> <NDC> <SN>~~[;cic=(carrier ID)]~~@**tariff.**<hostname>; user=phone SIP/2.0 INVITE: Request line sip: + <CC> <NDC> <SN>[;npdi] [;rn=+ <CC> D<xyz> <NDC> <SN>]@<hostname>;user=phone SIP/2.0
Comments	**Check:** Is the sub domain pattern 'tariff' present at the beginning of the hostportion only of the initial INVITE sent from network A to network B? ~~**Check:** Is the optional 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier?~~ **Check:** Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A? **Check:** Is optional the 'rn' parameter present in the Request URI of the INVITE request sent from network B to network A? ~~**NOTE 1:** The 'cic' parameter may be absent according national regulations or national agreements.~~ ~~**NOTE 2:** It is possible that further information is available in the Request line regarding the end user charging in case of Carrier selection.~~ Repeat this test in reverse direction.

Test case number	SS_csel_004
Test case group	SIP-SIP/CS
SIP Parameter	INVITE: Request line sip: + <CC> <NDC> <SN>~~[;cic=(carrier ID)]~~@ **tariff.**<hostname> user=phone SIP/2.0 INVITE: Request line sip: + <CC> <NDC> <SN>;npdi [;rn=<Number portability routing number>]@<hostname>; user=phone SIP/2.0
Comments	**Check:** Is the sub domain pattern 'tariff' present at the beginning of the hostportion only of the initial INVITE sent from network A to network B? ~~**Check:** Is the optional 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier?~~ **Check:** Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A? **Check:** Is optional the 'rn' parameter present in the Request URI of the INVITE request sent from network B to network A? ~~**NOTE 1:** The 'cic' parameter may be absent according national regulations or national agreements.~~ ~~**NOTE 2:** It is possible that further information is available in the Request line regarding the end user charging in case of Carrier selection.~~ Repeat this test in reverse direction.

Test case number	SS_csel_005
Test case group	SIP-SIP/CS
SIP Parameter	INVITE: Request line sip: + <CC> <NDC> <SN>@**tariff.**<hostname> user=phone SIP/2.0 Content-Type: application/vnd.etsi.cug+xml Content-Disposition: ….;handling= required ……. <…:cug> …… <..: cugCommunicationIndicator>11</…: cugCommunicationIndicator> <…:cug> INVITE: Request line sip: + <CC> <NDC> <SN@<hostname>;user=phone SIP/2.0 Content-Type: application/vnd.etsi.cug+xml Content-Disposition: ….;handling= required ……. <…:cug> …… <..: cugCommunicationIndicator>11</…: cugCommunicationIndicator> <…:cug>
Comments	**Check**: Is the sub domain pattern '**tariff**' present at the beginning of the hostportion only of the initial INVITE sent from network A to network B? **Check**: Is the 'npdi' parameter present in the userinfo of the INVITE request sent from network B to network A? **Check**: Is **optional** the 'rn' parameter present in the userinfo of the INVITE request sent from network B to network A? **Check**: Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '11' as a 'cug' child element? **Check**: Is the session setup not rejected?

Test case number	SS_csel_006
Test case group	SIP-SIP/CS
Reference	7.3/[1]
SELECTION EXPRESSION	SE 7 AND SE 8
Test purpose	*No sending of tariff announcements from transit network (VNB) by using call-by-call services and Call forwarding* User A and user C are located in network B. Ensure that no announcement is provided to user A if user B has activated call forwarding to user C by using carrier seletion via Network B (VNB). Ensure that user B is able to activate call forwarding to user C and user B is able to select network B as a selected carrier 'call-by-call'.
Configuration	
SIP Parameter	INVITE 1: Request line sip: + <CC> <NDC> <SN>[;npdi]@<hostname;user=phone SIP/2.0 INVITE 2: Request line sip: + <CC> <NDC> <SN>@**tariff.**<hostname;user=phone SIP/2.0 INVITE 2: Request line sip: + <CC> <NDC> <SN>[;npdi] [;rn=<CC> D<xyz> <NDC> <SN>]@<hostname; user=phone SIP/2.0
Message flow **SIP (Network A, Deutsche Telekom)** **Interconnection Interface** **SIP (Network B)** ← INVITE 1 INVITE 2 → ← INVITE 3 **Apply post test routine**	
Comments	**Check**: Is the sub domain pattern '**tariff**' present at the beginning of the hostportion **only** of the initial INVITE sent from network A to network B? **Check**: Is **no** announcement playing with tariff information during early dialogue to user A in Network B

Test case number	SS_csel_007
Test case group	SIP-SIP/CS
Reference	7.3/[1]
SELECTION EXPRESSION	SE 7 AND SE 8
Test purpose	*Sending of tariff announcements from transit network (VNB) by using call-by-call services* User A and user B are located in network A. Ensure that the selected transit network (VNB) is able to send announcement with tariff information to user A during call setup. Ensure that user A is able to call user B and user A is able to select network B as a selected carrier 'call-by-call'
Configuration	
SIP Parameter	INVITE 1: Request line sip: + <CC> <NDC> <SN>@**tariff.**<hostname> user=phone SIP/2.0 183 1 [P-Early-Media: sendonly (inactive)] SDP a= sendrecv 180 1 [P-Early-Media: sendonly (inactive)] SDP a= sendrecv INVITE 2: Request line sip: + <CC> <NDC> <SN>[;npdi] [;rn=<CC> D<xyz> <NDC> <SN>]@<hostname>; user=phone SIP/2.0
Message flow **SIP (Network A, Deutsche Telekom)** **Interconnection Interface** **SIP (Network B)** INVITE 1 → ← 183 (SDP) 1 ← 180 (SDP) 1 ← INVITE 2 **Apply post test routine**	
Comments	**Check**: Is the sub domain pattern '**tariff**' present at the beginning of the hostportion **only** of the initial INVITE sent from network A to network B? **Check**: Is a 183 or 180 sent from network B to establish an early dialogue? **Check**: Is a SDP present in the 183 or 180 as a SDP answer? **Check**: Is network B playing an announcement with tariff information during early dialogue? **Note**: The P-Early-Media header is optional.

Test case number	SS_csel_008
Test case group	SIP-SIP/CS
Reference	7.3/[1]
SELECTION EXPRESSION	SE 7 AND SE 8
Test purpose	*No sending of tariff announcements from transit network (VNB) by using preselection services and Call forwarding* User A and user C are located in network B. Ensure that no announcement is provided to user A if user B has activated call forwarding to user C by using carrier seletion via Network B (VNB). Ensure that user B is able to activate call forwarding to user C and user B is “preselected” to network B as a selected carrier.
Configuration	
SIP Parameter	INVITE 1: Request line sip: + <CC> <NDC> <SN>[;npdi] [;rn=<CC> D<xyz> <NDC> <SN>]@<hostname; user=phone SIP/2.0 INVITE 2: Request line sip: + <CC> <NDC> <SN>@**tariff.**<hostname;user=phone SIP/2.0 INVITE 2: Request line sip: + <CC> <NDC> <SN>[;npdi] [;rn=<CC> D<xyz> <NDC> <SN>]@<hostname; user=phone SIP/2.0
Message flow **SIP (Network A, Deutsche Telekom)** **Interconnection Interface** **SIP (Network B)** ← INVITE 1 INVITE 2 → ← INVITE 3 **Apply post test routine**	
Comments	**Check**: Is the sub domain pattern ‘**tariff**’ present at the beginning of the hostportion **only** of the initial INVITE sent from network A to network B? **Check**: Is **no** announcement playing with tariff information during early dialogue to user A in Network B

Modify Section §7.4“Emergency call“ as follows

Test case number	SS_ecall_001
Test case group	SIP-SIP/EmC
Reference	14/ [1]
SELECTION EXPRESSION	
Test purpose	**Request line in the INVITE with Geographic Location (Arc Band).** User A attempts to call a PSAP located in network B. Ensure that the userinfo in the INVITE contains the PSAP routing number in a global number format without hex digits and an ‘rn’ parameter containing the **PSTN PSAP routing number** in a global number format with hex digits.~~Ensure that the Request line in the INVITE contains the emergency number and a 'rn' parameter containing the PSAP routing number. In addition a location information may be present:~~ ~~• Geolocation header and corresponding PIDF-LO Element~~ ~~• User-to-User header~~ ~~• National solution to convey location information~~ ~~to make location information available for the PSAP.~~
Configuration	

SIP Parameter	INVITE: Request line sip+ <CC> 1982 <NDC> <xy~~(emergency number~~)> ~~[~~; rn =+<CC> <NDC> CC <xy>~~(PSAP routing number)]~~@hostname;user = phone SIP/2.0 User to user container: a) Civic location (chapter 14.3.1.2) b) Geographic location (chapter 14.3.1.3) b1) Ellipsoid point chapter B2) Ellipsoid point with uncertainty ellipse B3) Polygon B4) Ellipsoid arc B5) Reference coordinates b6) coverage area and description of radio cell Geolocation header: a) Civic location (chapter 14.3.2.2) b) Geographic location b1) Point (chapter 14.3.2.3.1) B2) Ellipse (chapter 14.3.2.3.2) B3) Polygon (chapter 14.3.2.3.3) B4) Arc band (chapter 14.3.2.3.4) b5) description of radio cell (chapter 14.3.2.3.5) Example: cellular network UUS and the related geolocation header contain Arc Band format (the relevant part is marked with colour): INVITE sip:+491982615121@t-mobile.de; user=phone SIP/2.0 … Call-Info: <cid:1234567890@t-mobile.de>;purpose=emergencyCallData Geolocation: <cid:target123@t-mobile.de> Content-Type: multipart/mixed; boundary=boundary1 User-to-User: 001D427594358480732200038A420640062F210010839A9; encoding=hex; purpose=isdn-uui P-Access-Network-Info: 3GPP-E-UTRAN-FDD;utran-cell-id-3gpp=2620100791d31a00;network-provided --boundary1 Content-Type: application/sdp … --boundary1 Content-Type: application/pidf+xml Content-Disposition: by-reference;handling=optional Content-ID: <target123@t-mobile.de> <?xml version="1.0" encoding="UTF-8"?> <presence xmlns="urn:ietf:params:xml:ns:pidf" xmlns:gp="urn:ietf:params:xml:ns:pidf:geopriv10" xmlns:gml="http://www.opengis.net/gml" xmlns:gs="http://www.opengis.net/pidflo/1.0" xmlns:cl="urn:ietf:params:xml:ns:pidf:geopriv10:civicAddr" entity="pres:123@t-mobile.de"> <tuple id="arcband"> <status> <gp:geopriv> <gp:location-info> <gml:location> <gs:ArcBand srsName="urn:ogc:def:crs:EPSG::4326" xmlns:gs="http://www.opengis.net/pidflo/1.0" xmlns:gml="http://www.opengis.net/gml"> <gml:pos> 49.8967 8.6228 </gml:pos> <gs:innerRadius uom="urn:ogc:def:uom:EPSG::9001"> 0 </gs:innerRadius> <gs:outerRadius uom="urn:ogc:def:uom:EPSG::9001"> 2005 </gs:outerRadius> <gs:startAngle uom="urn:ogc:def:uom:EPSG::9102"> 328 </gs:startAngle> <gs:openingAngle uom="urn:ogc:def:uom:EPSG::9102"> 64 </gs:openingAngle> </gs:ArcBand> </gml:location>

	<con:confidence>100</con:confidence> <cl:civicAddress xml:lang="de"> <cl:LOC>Mobilfunkzelle</cl:LOC> <cl:ADDCODE>26201F1080939A</cl:ADDCODE> </cl:civicAddress> </gp:location-info> <gp:usage-rules> <gp:retransmission-allowed>yes</gp:retransmission-allowed> <gp:retention-expiry>2013-08-31T12:00:00Z</gp:retention-expiry> </gp:usage-rules> </gp:geopriv> </status> <timestamp>2013-08-01T12:00:00Z</timestamp> </tuple> </presence>
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE ➔ **Apply post test routine**
Comments	**Check:** Is the URI in the userinfo of the Request line in a global number format containing the PSAP routing number? **Check:** ~~Optional~~: Is the URI 'rn' parameter containing the PSAP Routing Number Number in a global number format with hex digits “CC”? **Check:** Is the user parameter set to 'phone'? **Check:** Is the location information present in the initial INVITE request. ~~Geolocation header~~ ~~PIDF-LO Element XML 'geopriv' sub element~~ ~~Or~~ ~~User-to-User header~~ ~~Or~~ ~~National solution~~ **NOTE**: The routing prefix (0)198x has to be assigned by the BNetzA Repeat this test in reverse direction.

Test case number	SS_ecall_002
Test case group	SIP-SIP/EmC
Reference	14/ [1]
SELECTION EXPRESSION	
Test purpose	**Request line in the INVITE with Civic Location.** User A attempts to call a PSAP located in network B. Ensure that the userinfo in the INVITE contains the PSAP routing number in a global number format without hex digits and an 'rn' parameter containing the **PSTN PSAP routing number** in a global number format with hex digits.
Configuration	

SIP Parameter	INVITE: Request line sip+ <CC>1982 <NDC><xy> [; rn =+<CC><NDC> CC <xy>@hostname;user = phone SIP/2.0 Example: fixed network UUS and the related geolocation header contain ellipse format (the relevant part is marked with colour): INVITE sip:+491982615121@t-mobile.de; user=phone SIP/2.0 … Call-Info: <cid:1234567890@t-mobile.de>;purpose=emergencyCallData Geolocation: <cid:target123@t-mobile.de> Content-Type: multipart/mixed; boundary=boundary1 User-to-User: 001D4275943584807322000038A420640062F210010839A9; encoding=hex; purpose=isdn-uui --boundary1 Content-Type: application/sdp … --boundary1 Content-Type: application/pidf+xml Content-Disposition: by-reference;handling=optional Content-ID: <target123@t-mobile.de> <?xml version="1.0" encoding="UTF-8"?> <presence xmlns="urn:ietf:params:xml:ns:pidf" xmlns:gp="urn:ietf:params:xml:ns:pidf:geopriv10" xmlns:gml="http://www.opengis.net/gml" xmlns:gs="http://www.opengis.net/pidflo/1.0" xmlns:cl="urn:ietf:params:xml:ns:pidf:geopriv10:civicAddr" entity="pres:123@t-mobile.de"> <tuple id=" abcd123456"> <status> <gp:geopriv> <gp:location-info> <cl:civicAddress xml:lang="de"> <cl:country>DE</cl:country> <cl:A1>HE</cl:A1> <cl:A2>Darmstadt</cl:A2> <cl:A3>Darmstadt</cl:A3> <cl:A5>06411000a</cl:A5> <cl:A6>Heinrich-Hertz-</cl:A6> <cl:STS>Str.</cl:STS> <cl:HNO>3</cl:HNO> <cl:HNS>-7</cl:HNS> <cl:PC>64295</cl:PC> <cl:FLR>4</cl:FLR> <cl:UNIT>E2</cl:UNIT> <cl:ROOM>4.E2.04</cl:ROOM> <cl:LMK>Deutsche Telekom</cl:LMK> <cl:LOC>TZ Rhein-Main</cl:LOC> </cl:civicAddress> </gp:location-info> <gp:usage-rules> <gbp:retransmission-allowed>yes</gbp:retransmission-allowed> <gbp:retention-expiry>2013-08-31T12:00:00Z</gbp:retention-expiry> </gp:usage-rules> </gp:geopriv> </status> <timestamp>2013-08-01T12:00:00Z</timestamp> </tuple> </presence> --boundary1 Content-Type: application/emergencyCall.ProviderInfo+xml Content-ID: 1234567890@telekom.de Content-Disposition: by-reference; handling=optional <?xml version="1.0" encoding="UTF-8"?> <emergencyCall.ProviderInfo xmlns="urn:ietf:params:xml:ns:emergencyCall.ProviderInfo"> <DataProviderString>Telekom</DataProviderString> <ProviderID>D150</ProviderID> <contactURI>sip:+492911234567@telekom.de;user=phone</contactURI> <ProviderIDSeries>BNetzA</ProviderIDSeries> </emergencyCall.ProviderInfo> --boundary1--

Message flow			
SIP (Network A)	Interconnection Interface INVITE → **Apply post test routine**		SIP (Network B)
Comments	**Check:** Is the URI in the userinfo of the Request line in a global number format containing the PSAP routing number? **Check:** ~~Optional~~: Is the URI 'rn' parameter containing the PSAP Routing Number Number in a global number format with hex digits "CC"? **Check:** Is the user parameter set to 'phone'? **Check:** Is the location information present in the initial INVITE request. **NOTE**: The routing prefix (0)198x has to be assigned by the BNetzA Repeat this test in reverse direction.		

The section "§7.7 Quality of Service (optional)" does not apply

6.1 Add following section “Mobile Service Prefix (Mobilfunkservicevorwahl)”

Test case number	SS_msv_001
Test case group	SIP-SIP/msv
Reference	7.1.2.8/[2]
SELECTION EXPRESSION	SE 9a
Test purpose	*Successful call to a service when offline charging applies.* Ensure that the P-Germany-Tariff header is present in the initial INVITE request and the ‘tariff’ tag is set to the relevant tariff value when the mobile user attempts a communication with a service in offline charging e.g. 0900..
Configuration	
SIP Parameter	INVITE: P-Germany-Tariff: tariff=xy
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE ➔ **Apply post test routine**
Comments	**Check**: is P-Germany-Tariff header present in the initial INVITE request? **Check**: Is the ‘tariff’ value set to the relevant tariff applicable for tis service? **Check**: Is the communication successful?

6.2 Add following section “Transit connections”

Test case number	SS_tran_001
Test case group	SIP-SIP/tran
Reference	
SELECTION EXPRESSION	[Network B] SE 8
Test purpose	*Successful conection via a transit operator network. Passing of History-Info header* User A and User B are located in Network A. Network B is a transit network. Ensure that a History-Info header is unchanged transferred via Network B if Network B is a transit network (VNB).
Configuration	
SIP Parameter	INVITE 1: History-Info INVITE 2: History-Info
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE 1 → ← INVITE 2 **Apply post test routine**
Comments	**Check**: Is a History-Info header present in the INVITE request received from Network A? **Check**: Is a History-Info header present in the INVITE request received from Network B and is the History- Info header identical to the History-Info header sent to Network B?

6.3 Add following section “TKG §120 Requirements for Number Transmission”

Test case number	SS Call_Blocking_001
Test case group	SIP-SIP/Service/TKG §120 Requirements for Number Transmission
Reference	15.1/ [1]
SELECTION EXPRESSION	SE 17f
Test purpose	**Blocking of calls with prohibited Calling Numbers** Ensure that the Call Blocking Service in network B rejects all calls from network A with prohibited calling numbers according to TKG §120 in P-Asserted-Identity and/or FROM header with Response 403 “Forbidden”.
Configuration	Network A set up calls with prohibited calling numbers according to TKG §120 Network B has activated Call Blocking Service List of prohibited calling numbers according to TKG §120 • Emergency Call Numbers, • Directory Enquiries, • Mass Calls, • Premium Services or • Short Codes
SIP Parameter	**INVITE** **P-Asserted-Identity:<sip:prohibited number@networkA.de>** **AND/OR** **From<:sip:prohibited number@networkA.de>** **403 Forbidden** **Reason: SIP;cause=403;text="CLI Forbidden" (optional)**
Message flow	
SIP (Network A) **Interconnection Interface** **SIP (Network B)**	INVITE → ← 403 Forbidden (Note 1) (cause=403;text="CLI Forbidden) ACK →
Comments	Check:Contains P-Asserted-Identity and/or From header from network A prohibited calling numbers according to TKG §120? Check:Is a 403 Forbidden sent from Network B to Network A? Check:If a Reason header is present, is the cause value equal to 403 and the text set to "CLI Forbidden"? Repeat this test with all defined prohibited calling numbers.

Test case number	SS_International_Calls_001
Test case group	SIP-SIP/Service/TKG §120 Requirements for Number Transmission
Reference	15.2/ [1]
SELECTION EXPRESSION	SE 17g
Test purpose	**International Calls with German national calling number in FROM header** If a call with German national calling number contains in FROM header is established via an international call leg, ensure that the INVITE from network A still contains a privacy header with value 'user'. Also ensure that the INVITE from network A contains the P-Germany-Origin header with domain of the network with the international interface.
Configuration	
SIP Parameter	**INVITE** **From:<sip:German-number@orig-network.de>** **Privacy: user** **P-Germany-Origin: international=network-international-lc.de**
Message flow	
SIP (Network A) **Interconnection Interface** **SIP (Network B)** INVITE ➔ **Apply post test routine**	
Comments	Check: Contains the From header from network A an German national number? Check: Does the INVITE from network A contains a Privacy header with value "user"? Check: Does the INVITE from network A contains a P-Germany-Origin header with domain of the network with the international interface?

<table>
<tr><td>Test case number</td><td>SS_International_Calls_002</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/TKG §120 Requirements for Number Transmission</td></tr>
<tr><td>Reference</td><td>15.2/ [1]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>SE 17g</td></tr>
<tr><td>Test purpose</td><td>International Calls with German national calling number in P-Asserted-Identity

If a call with German national calling number contains in P-Asserted-Identity is established via an international call leg, ensure that the INVITE from network A still contains a privacy header set to “id”.
Also ensure that the INVITE from network A contains the P-Germany-Origin header with domain of the network with the international interface.</td></tr>
<tr><td>Configuration</td><td></td></tr>
<tr><td>SIP Parameter</td><td>INVITE
P-Asserted-Identity:<sip:German-number@orig-network.de>

Privacy: id

P-Germany-Origin: international=network-international-lc.de</td></tr>
<tr><td colspan="2">Message flow</td></tr>
<tr><td colspan="2">SIP (Network A)   Interconnection Interface   SIP (Network B)
INVITE ➔
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Check: Contains the P-Asserted-Identity from network A a German national number?
Check: Does the INVITE from network A contains a Privacy header with value “id”?
Check: Does INVITE from network A contains a P-Germany-Origin header with domain of the network with the international interface?</td></tr>
</table>

Test case number	SS_International_Calls_003
Test case group	SIP-SIP/Service/TKG §120 Requirements for Number Transmission
Reference	15.2/ [1]
SELECTION EXPRESSION	SE 17g
Test purpose	**International Calls with German national calling number in P-Asserted-Identity and FROM header** If a call with German national calling number contains in P-Asserted-Identity and FROM header is established via an international call leg, ensure that the INVITE from network A still contains a privacy header with value “id” and 'user'. Also ensure that the INVITE from network A contains the P-Germany-Origin header with domain of the network with the international interface.
Configuration	
SIP Parameter	**INVITE** **P-Asserted-Identity:<sip:German-number@orig-network.de>** **AND** **From:<sip:German-number@orig-network.de>** **Privacy: id and user** **P-Germany-Origin: international=network-international-lc.de**
Message flow	
SIP (Network A) **Interconnection Interface** **SIP (Network B)** INVITE ➔ **Apply post test routine**	
Comments	Check:Contains P-Asserted-Identity and From header from network A German national number? Check:Does the INVITE from network A contains a Privacy header with value “id” and “user”? Check:Does INVITE from network A contains a P-Germany-Origin header with domain of the network with the international interface?

6.4 Add following section “Different privacy settings”

Test case number	SS_privacy_001
Test case group	SIP-SIP/Service/Different privacy setting
Reference	[4]
SELECTION EXPRESSION	SE 20a
Test purpose	**Terminating user does not receive the identity of the originating user from P-Asserted-Identity header.** If network A supports privacy service with different settings for privacy header, ensure that the P-Asserted-Identity still contains identity information, and the privacy is set to 'id' only. The terminating user does not receive the identity of the originating user from the P-Asserted-Identity header.
Configuration	Privacy service for Originating user: Privacy is only set to id
SIP Parameter	**INVITE** **P-Asserted-Identity: default public user identity** **Privacy: id**
Message flow	
SIP (Network A) **Interconnection Interface** **SIP (Network B)**	INVITE ➔ **Apply post test routine**
Comments	Check: Is the P-Asserted-Identity is present? Check: Is the Privacy header only set to 'id'? Check: Does terminating user not receive the P-Asserted-Identity header? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_privacy_002
Test case group	SIP-SIP/Service/Different privacy setting
Reference	[3]
SELECTION EXPRESSION	SE 20a
Test purpose	**Terminating user receives anonymized FROM header** If network A supports privacy service with different settings for privacy header, ensure that the FROM header still contains identity information and the privacy is set to 'user'. The terminating user receives anonymized FROM header.
Configuration	Privacy service for Originating user: Privacy is only set to user
SIP Parameter	**INVITE** **From:original value** **Privacy: user**
Message flow	
SIP (Network A) **Interconnection Interface** **SIP (Network B)**	INVITE ➔ **Apply post test routine**
Comments	Check: Is the originating identification information in FROM header present? Check: Is the Privacy header only set to 'user'? Check: Does terminating user receive anonymized FROM header? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_privacy_003
Test case group	SIP-SIP/Service/Different privacy setting
Reference	[3]
SELECTION EXPRESSION	SE 20a
Test purpose	**Terminating user does not receive the identity of the originating user contains in headers according to RFC3323.** If network A supports privacy service with different settings for privacy header, ensure that privacy is set to 'header' only and all according to headers contains identity information. Terminating user does not receive the identity of the originating user contains in headers according to RFC3323.
Configuration	Privacy service for Originating user: Privacy is only set to header
SIP Parameter	**INVITE** **headers which contain identity information** **Privacy: header**
Message flow	
SIP (Network A)	**Interconnection Interface** / INVITE ➔ / **Apply post test routine** / **SIP (Network B)**
Comments	Check: Are one or more headers contains identity information present? Check: Is the Privacy header only set to 'header'? Check: Does terminating user not receive identity information related to this header? Check: Is the FROM header not affected? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_privacy_004
Test case group	SIP-SIP/Service/Different privacy setting
Reference	[5]
SELECTION EXPRESSION	SE 20a
Test purpose	**Terminating user does not receive information from History-Info header** If network A supports privacy service with different settings for privacy header, ensure that network A is sending History-Info header with related information and the privacy is set to ´history´. The terminating user does not receive any information from History-Info header
Configuration	Privacy service for Originating user: Privacy is set to history
SIP Parameter	**INVITE** **History-Info: any information** **Privacy: history**
Message flow	
SIP (Network A)	**Interconnection Interface** / INVITE ➔ / **Apply post test routine** / **SIP (Network B)**
Comments	Check: Is the History-Info header present? Check: Is the Privacy header set to history? Check: Does terminating user not receive any information from History-Info header? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

ANNEX A Add the following section "Header fields used in the basic call" (Informativ)

This table is used to verify optionally the occurance of mandatory, optional and not applicable headers in the basic call. This could be done e.g. in the test cases SS_bcall_001 and SS_bcall_002 and the test cases to Supplementary services.
The meaning of indication of the following tables is shown in table A-1.

Table A-1

Indication	Meaning
m	mandatory; supported by sending and receiving
n/a	not applicable; not supported
o	optional; may be supported based on bilateral agreements

Table A-1: Meaning of Indication in the last Column

Please consider that optional header fields **may be present** in requests or responses.

Table A-2

Item	Header field	UAK-S profile
1	Accept	m
2	Accept-Contact	o
3	Accept-Encoding	n/a
4	Accept-Language	o
4a	Accept-Resource-Priority	o
5	Alert-Info	see Table 8.3.3/[1]
6	Allow	m
7	Allow-Events	n/a
8	Authentication-Info	n/a
9	Authorization	n/a
9a	Answer-Mode	o
13	Content-Disposition	m
14	Content-Encoding	o
15	Content-Language	o
19	Date	o
20	Error-Info	n/a
21	Expires	o
21a	Flow-Timer	n/a
21b	Feature-Caps	o
22	Event	see Table 8.3.3/[1]
24	Geolocation	m
24a	Geolocation-Error	n/a
24b	Geolocation-Routing	n/a
25	History-Info	m
25a	Info-Package	o
26	In-Reply-To	n/a
27	Join	n/a
27a	Max-Breadth	n/a
29	Min-Expires	o
30	MIME-Version	o
31	Min-SE	o
32	Organization	o
33	P-Access-Network-Info	see Table 8.3.3/[1]
33a	P-Answer-state	n/a
35	P-Asserted-Service	n/a
35a	P-Associated-URI	n/a
36	P-Called-Party-ID	n/a
37	P-Charging-Function-Addresses	n/a
38	P-Charging-Vector	o
38a	P-Debug-Id	n/a
39	P-Early-Media	o
39A	P-Germany-Tariff	o
40	P-Media-Authorization	n/a
41	P-Preferred-Identity	n/a
42	P-Preferred-Service	o
43	P-Private-Network-Indication	n/a
44	P-Profile-Key	n/a
44a	P-Refused-URI-List	n/a
45	P-Served-User	n/a
46	P-User-Database	n/a
47	P-Visited-Network-ID	n/a
47a	Path	n/a
47b	Permission-Missing	n/a
47c	Policy-Contact	o
48	Priority	o
48a	Priv-Answer-Mode	n/a
50	Proxy-Authenticate	n/a

51	Proxy-Authorization	n/a
52	Proxy-Require	n/a
52a	RAck	o
53	Reason	m
54	Record-Route	o
54a	Recv-Info	o
55	Referred-By	see Table 8.3.3/[1]
55a	Refer-Sub	n/a
55b	Refer-To	n/a
56	Reject-Contact	n/a
56a	Relayed-Charge	n/a
57	Replaces	see Table 8.3.3/[1]
58	Reply-To	n/a
59	Request-Disposition	n/a
60	Require	o
61	Resource-Priority	o
61a	Resource-Share	o
61b	Restoration-Info	n/a
61a	Retry-After	o
62	Route	o
62a	RSeq	o
63	Security-Client	n/a
63a	Security-Server	n/a
64	Security-Verify	n/a
65	Server	n/a
65c	Service-Interact-Info	o
65a	Service-Route	n/a
65b	Session-ID	o
66	Session-Expires	o
66a	SIP-ETag	n/a
66b	SIP-If-Match	n/a
67	Subject	n/a
67a	Subscription-State	see Table 8.3.3/[1]
67b	Suppress-If-Match	o
68	Supported	m
68a	Target-Dialog	o
69	Timestamp	o
71	Trigger-Consent	m
71a	Unsupported	o
72	User-Agent	o
73	User-to-User	m
75	Warning	o
76	WWW-Authenticate	n/a