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April 21, 2015

Martin Dolly
PTSC Chair

Re: ATIS Letter Ballot PTSC-LB-199 Closing Letter

Dear Martin:

ATIS Letter Ballot PTSC-LB-199 entitled "dpTR – IPNNI Profile" closed on April 20, 2015. The results are as follows:

12	**Approvals**
	Comment: Alcatel-Lucent, Applied Communication Sciences, Ericsson, Sprint, Verizon
0	**Disapprovals**
	No Comment
9	**Abstentions**
	No Comment
5	**Ballots not returned**
26	**Voting members**

Sincerely,

[Original signed by K.Conn]

Kerrianne Conn
ATIS Manager, Knowledge Management

Cc: V. Shaikh
S. Barclay
J. Voss
D. Greco

1 ACS Comments

The following are ACS comments on LB-199:

#	Page	Section	Original Text	Comment
1)		**Global**		Spell out first use of Abbreviations
2)	**1**	**1.2 (Purpose)**	The purpose of this effort is to identify a baseline set of features that should be common to all IP-NNI implementations for voice service.	The purpose of this ~~effort~~ document is to identify a baseline set of features that should be common to all IP-NNI implementations for voice service.
3)	**1**	**1.2**	This document uses key words like MUST, MAY and SHALL in accordance with RFC-2119..	This document uses key words like MUST, MAY and SHALL in accordance with RFC-2119~~.~~.
4)		**3.1**	3.1 Definitions AAA: xxxx. Bbbb: xxxx.	3.1 Definitions ~~AAA: xxxx.~~ ~~Bbbb: xxxx.~~None
5)	**3**	**3.1**	3.1 Acronyms & Abbreviations	Include ETS – Emergency Telecommunications Service
6)	**7**	**4.3**	In general, Carrier providers should view other providers as un-trusted. Figure 3 shows an example when a connected Carrier is judged un-trusted.	In general, a Carrier ~~providers~~ should view other carriers and service providers as un-trusted. Figure 3 shows an example when a connected Carrier is judged un-trusted.
7)	**10**	**5.2.1**	5.2.1 Identifying the Called User When sending a dialog-initiating or standalone request to a peer Carrier network, SIP entities involved in session peering MUST: • Identify the called user or service in the Request-URI of the request, and • Identify the called user or service using a SIP URI as described above in Section 6.2.	Fix reference. 6.2 is not above.
8)	**11**	**5.2.2**	. A non-anonymized identity MUST be populated using the telephone-subscriber syntax form of the SIP URI as described above in Section 5.2.3.	Fix reference. Delete “above”?
9)	**14**	**5.4.2**	Carriers MUST support SIP Overload	Carriers MUST ~~-~~support SIP

			Control including support of the default algorithm [RFC 7339]. Carrier's MAY optional support the Rate Based algorithm based on bilateral agreement between two carriers.	Overload Control including support of the default algorithm [RFC 7339]. Carrier's MAY optionally support the Rate Based algorithm based on bilateral agreement between two carriers.
10)	**14**	**Footnote 5**	Support of SIP Overload Control was defined for in the IETF for 3GPP Release 11, and may not be available for deployment at the time of this documents initial publication.	Support of SIP- Overload Control was defined ~~for~~ in the IETF for 3GPP Release 11, and may not be available for deployment at the time ~~of this documents initial publication~~when this document was published.
11)	**17**	**5.5.6**	The "named telephone events," or "telephone-events" RTP payload [RFC 4733] is the preferred mechanism for transport of DTMF digit events between VoIP endpoints and network elements. by bilateral agreement, in-band DTMF tones might be used across the NNI to avoid transcoding from in-band DTMF tones to named telephone events (DTMF relay), for instance if the media stream is expected to originate and terminate on circuit-switched voice channels in both carrier networks.	The "named telephone events," or "telephone-events" RTP payload [RFC 4733] is the preferred mechanism for transport of DTMF digit events between VoIP endpoints and network elements. ~~by~~ By bilateral agreement, in-band DTMF tones might be used across the NNI to avoid transcoding from in-band DTMF tones to named telephone events (DTMF relay), for instance, if the media stream is expected to originate and terminate on circuit-switched voice channels in both carrier networks.
12)	**18**	**5.6**	**Traffic treatment** Voice and media traffic leaving the sending Border Function towards the receiving Border Function should be treated according to the Expedited Forwarding Per-Hop Behavior **Error! Reference source not found.**, **Error! Reference source not found.**. ETS voice signalling and media traffic leaving the sending Border Function towards the receiving Border Function should be treated according to the VOICE-ADMIT Forwarding Per-Hop Behaviour [Reference to 5865]. Voice signaling traffic leaving the sending Border Function towards the receiving Border Function should be treated according to the Expedited Forwarding Per-Hop Behavior [RFC 3246]**Error! Reference source not found.**, [RFC3247], or alternatively according to the Default Forwarding Per-Hop Behavior [RFC 2597].	What is the difference between "Voice Signaling" and "Signaling?" Is signaling here intended to be "IP Network Signaling"?

			Signalling traffic leaving the sending Border Function towards the sending PE router MUST be treated according to one of the following schemes: • the Expedited Forwarding Per-Hop Behavior, as specified in RFC 3246 and RFC 3247; • the Assured Forwarding Per-Hop Behavior as specified in RFC 2597; • the Default forwarding PHB , as specified in IETF RFC 2597.	
13)	**18**	**5.6**	IP Marking Table	For the "IP Precedence" and "802.1 Q VLAN" values for different traffic classes, provide a reference to where these values are specified.
14)	**19**	**6.2**	2. NOTE: If the terminating network receives an INVITE that does not contain an SDP offer, and wishes to provide early media.......	Is this intended to be a note?
15)	**20**	**6.3**	6.3 Early-Media Carriers SHOULD support P-Early-Media as defined in [RFC 5009].	6.3 Early-Media Carriers SHOULD support P-Early-Media as defined in [RFC 5009]. NOTE: P-Early-Media is required for ETS (NS/EP) support.
16)	**22**	**6.9**	A Carrier network that is required to remain in the signaling path of a forwarded call based on local policy MUST do so using one of the following procedures: 1. forward the INVITE to the forward-to-user while remaining in the signaling path as a SIP Proxy or B2BUA, or	Fix dangling item 1. 1. forward the INVITE to the forward-to-user while remaining in the signaling path as a SIP Proxy or B2BUA.~~, or~~
17)	**22**	**6.10**	***6.10 National Security/Emergency Prepardness (NS/EP)*** Resource Priority Header (RPH) MUST be supported by NS/EP compliant networks,	***6.10 ~~National Security/Emergency Prepardness (NS/EP)~~Emergency Telecommunications Service ETS*** Resource Priority Header (RPH)

			and MUST be transparently passed by non-NS/EP compliant networks.	MUST be supported by ~~NS/EP~~ETS (NS/EP) compliant networks, and MUST be transparently passed by non-~~NS/EP~~ETS compliant networks.
18)	**25**	**7.1.2.2**	*NOTE 1: This header field is only applicable on a roaming NNI whereas for the interconnect NNI it is left unspecified.*	*NOTE 1: This header field is only applicable on a roaming NNI whereas for the interconnect NNI it is left unspecified.*

INSTRUCTIONS: Use this form to enter your comments on PTSC-LB-199 When complete, save the file on your hard drive and attach this file as instructed with your vote in ATIS Workspace (http://www.atis.org/faq/aws/#Ballot).

Section Number - Required.

Current/Proposed Text- Comment - These fields are required. Enter your comment and proposed change in these fields, respectively. Use plain text characters only.

Author	**Section Number**	**Current Text**	**Proposed Text**	**Comment**
ALU-1	4.1	Figure 4.1	Correct figure size to properly fit page	Figure doesn't fit page, correct figure size.
ALU-2	General	RFC-4967	[RFC 4967]	Document references should use consistent format throughout the document
ALU-3	5.2.1	North American supported formats are shown in Table 5.2.	North American supported formats are shown in Table 5.1.	Correct table reference
ALU-4	5.2.2	A non-anonymized identity MUST be populated using the telephone-subscriber syntax form of the SIP URI as described above in Section 5.2.3.	A non-anonymized identity MUST be populated using the telephone-subscriber syntax form of the SIP URI as described in Section 5.2.3.	Section 5.2.2 is referncing section 5.2.3 as "above". Propose to remove the word "above".

ALU-5	5.4.1	…and SHOULD have max-forwards set to ‘1’.	...and SHOULD have max-forwards set to ‘1’ (max-forwards is decremented to 0 when transitmitted).	Desire is that next node does not attempt to propagate the message. This is best accomplished by having the message that the UAS receives with max-forwards set to '0'.
ALU-6	5.4.1	If a requesting SIP entity receives a 486 or 503 response it can send subsequent OPTIONS messages…	If a requesting SIP entity receives a 408 or 503 response it can send subsequent OPTIONS messages	Based on RFC 3261, response to OPTIONS should be the same as INVITE would receive. 486 Busy would not be expected when request is directed to a host instead of a user. 408 should be included as OPTIONS failure to handle case of message being lost.
ALU-7	5.4.1	The sending element SHOULD, after receipt of a 486 response, attempt to avoid establishing new sessions with the heavily loaded peer element until receiving a 200 OK to a subsequent OPTIONS request.	remove statement	Use of 486 in this manner is not common. This profile should not be documenting network specific implementations. The statement should be removed.

ALU-8	5.4.1	If the sending Carrier network fails to receive a response to N consecutive OPTIONS requests, it MUST behave as if a 503 response had been returned.	If the sending Carrier network fails to receive a response SIP Timer F expiry (after multiple retransmission attempts of the OPTIONS requests), it MUST behave as if a 408 response had been returned.	OPTIONS will be retransmitted multiple times by the SIP stack when no response is received. After timer F expires, RFC 3261 indicates that handling is to be per 408 Request Timeout being received.
ALU-9	5.5.4		If the offering network adds additional codecs to the original offer, they should be placed after the offered codecs.	Networks will often add additional codecs to the original offer to improve the possibility of the far node finding a supported codec. These codecs should appear after the natively offered codecs to minimize the possibility of transcoding. It is proposed that an additional statement be added.
ALU-10	5.5.5	SIP entities involved in session peering MUST support fax or modem voice-band data (VBD) pass-through in a G.711 µ-law or A-law audio stream.	SIP entities involved in session peering MUST support fax or modem voice-band data (VBD) pass-through in a G.711 µ-law audio stream.	The requirement should only be applied to µ-law, remove reference to A-law.

ALU-11	6.1		If the repeated SDP answer is not identical to the previous answer, it MUST be ignored.	The last sentence of the first paragraph mentions that repeated SDP answer in 200 OK MUST be the same as previousl answer. But it has been seen that some implementations do not currently following this rule. To be clear about handling of such cases, it is proposed to add additional statement at the end of first paragraph.
ALU-12	6.1.1		Per [RFC-3264], the use of different payload type numbers for the selected codec(s) for sending and receiving RTP MUST be supported.	Since codecs that use dynamic payload type number often have different numbers allocated per implementation, it is common that payload type numbers will not be consistent between networks. A statement should be added that asymmetric payload type number usage MUST be supported.

ALU-12	6.2, item 2	2. NOTE: If the terminating network receives an INVITE that does not contain an SDP offer, and wishes to provide early media to the calling user, it must establish the session over which to do so via an UPDATE request sent after the initial offer / answer exchange has concluded.The originating Carrier network performs the following action on receipt of a provisional response to a call-initiating INVITE.	NOTE: If the terminating network receives an INVITE that does not contain an SDP offer, and wishes to provide early media to the calling user, it must establish the session over which to do so via an UPDATE request sent after the initial offer / answer exchange has concluded. 2. The originating Carrier network performs the following action on receipt of a provisional response to a call-initiating INVITE.	The first sentence of item 2 should be a NOTE associated with the previous major bullet. The second sentence should be the start of item 2.

ALU-13	6.2 NOTE	NOTE: If the terminating network receives an INVITE that does not contain an SDP offer, and wishes to provide early media to the calling user, it must establish the session over which to do so via an UPDATE request sent after the initial offer / answer exchange has concluded.	remove statement	The indicated procedure is to accommodate a means to inform the originating side that early media will be provided lacking use of the P-Early-Media header field. However, the establishment of the media session would have occurred on the completion of the previous offer/answer exchange. Also, it is likely the SDP contained in the UPDATE will simply be a repeat of the previously returned SDP, i.e., the SDP is unchanged. There is no agreed use of UPDATE in this network specific manner. Additionally, notes cannot contain MUST statements. It is proposed that the NOTE be removed.

ALU-14	6.6	Carriers may support features such as click-to-call, where the call is initiated by a 3rd party such as an Application Server on behalf of the originating user. To support such features, SIP entities involved in session peering MUST support the 3PCC procedures described in [RFC 3725].	remove section	This section contains a statement that RFC 3725 MUST be supported. There are many procedures documented in this informative document that are not required to be supported. The procedures that are needed are already accommodated for elsewhere within the NNI Profile. It is proposed that this section be removed.
ALU-15	6.9	1. forward the INVITE to the forward-to-user while remaining in the signaling path as a SIP Proxy or B2BUA, or	1. forward the INVITE to the forward-to-user while remaining in the signaling path as a SIP Proxy or B2BUA.	hanging "or" with unfinished sentence - remove "or" and terminate the sentence.
ALU-16	7.1 and 7.2 subsections	For the purpose of the present document clause 6.1.1.4 TS 29.165 v11.5.0 (2012-12) applies as follows:	remove statement	The text is carry over from the source document. The statement should be removed.
ALU-17	Tables with section 7.1 and 7.2	numerous instances	correct document refernces	incorrect document references resulting from carryover from source document.

ERICSSON Comments on PTSC-LB-199

EDITORIAL

Author	Section Number	Current Text	Proposed Test	Comments
HM	1.2	This often requires Session Border Controllers or IBCF proxies reconcile the signaling between service providers and resolving those ambiguities. Time and effort is also required to document the differences and configure the SBC or IBCF proxy to implement the necessary changes to the on the wire protocol.	This often requires Session Border Controllers or IBCF proxies to reconcile the signaling between service providers and resolve those ambiguities. Time and effort is also required to document the differences and configure the SBC or IBCF proxy to implement the necessary changes to the 'on the wire' protocol.	
	2.		[RFC 5865] IETF RFC - *A Differentiated Services Code Point (DSCP) for Capacity-Admitted Traffic*	Add RFC 5865 after RFC 5009
	3.	3GPP: 3rd Generation Partnership Project ALG: Application Level Gateway	3GPP: 3rd Generation Partnership Project 3PCC: 3rd Party Call Control ALG: Application Level Gateway	Add acronym 3PCC
	3.	MPLS: Multiprotocol Label Switching MSC: Mobile Switching Center	MPLS: Multiprotocol Label Switching MSA: Metropolitan Statistical Area MSC: Mobile Switching Center	Add acronym MSA
	3.	QoS Quality of Service RTP Real-Time Protocol	QoS: Quality of Service RPH: Resource Priority Header RTP: Real-Time Protocol	Add acronym RPH
	Figure 4.1			Remove empty white boxes
	4.3, first paragraph on p.8	The common characteristics of Carrier network elements in this zone are that they are under the full control of the Carrier provider are located in	The common characteristics of Carrier network elements in this zone are that they are under the full control of the Carrier provider, are located in..	
	4.3, second paragraph on p.8	The "trusted zone" will be protected by a combination of various methods. Some examples are physical security	The "trusted zone" will be protected by a combination of various methods. Some examples are physical security	

Author	Section Number	Current Text	Proposed Test	Comments
		of the Carrier network elements, general hardening of the systems, , use of secure signaling, security for OAMP messages separate VPN within the (MPLS/)IP network	of the Carrier network elements, general hardening of the systems, use of secure signaling, security for OAMP messages, separate VPN within the (MPLS/IP) network..	
	4.3, third paragraph on p.8	The equipment may be under the control by either the customer/subscriber or the Carrier provider.	The equipment may be controlled by either the customer/subscriber or the Carrier provider.	
	4.3, p.9 First paragraph	Examples of devices and systems that are operated by an Carrier provider	Examples of devices and systems that are operated by a Carrier provider…	
	4.3, p.9 Second paragraph	The “trusted-but-vulnerable” zone will be protected by a combination of methods. Some examples are physical security of the Carrier network elements, general hardening of the systems, , use of secure	The “trusted-but-vulnerable” zone will be protected by a combination of methods. Some examples are physical security of the Carrier network elements, general hardening of the systems, use of secure	
	5.1, 3rd paragraph	.. could be configured to remove ‘preconditions’ from the Supported header in order to disable the use the SIP preconditions procedures [RFC 3312].	.. could be configured to remove ‘preconditions’ from the Supported header in order to disable the use of the SIP preconditions procedures [RFC 3312].	
	Table 5.2			Remove blank rows, except ones separating the headers
	5.4.2	Carrier's MAY optional support the Rate Based algorithm	Carriers MAY optionally support the Rate Based algorithm…	
	Page 14, footnote 5	Support of SIP Overload Control was defined for in the IETF for 3GPP Release	Support of SIP Overload Control was defined in the IETF for 3GPP Release….	
	Table 5.4			Are there any mandatory entries?

Author	Section Number	Current Text	Proposed Test	Comments
	5.5.3			Last entry in the table needs a correct reference.
	5.5.4	1. Transcoding SHOULD generally avoided;	1. Transcoding SHOULD generally be avoided;	
	5.6			Remove blank row in the table
	5.6	Distinguishing traffic classes		Check fonts and size as they seem to vary from one paragraph to another.
	5.6, p. 18	ETS voice signalling and media traffic leaving the sending Border Function towards the receiving Border Function should be treated according to the VOICE-ADMIT Forwarding Per-Hop Behaviour [Reference to 5865].	ETS voice signalling and media traffic leaving the sending Border Function towards the receiving Border Function should be treated according to the VOICE-ADMIT Forwarding Per-Hop Behaviour [RFC 5865].	
	5.6, p.18, third bullet	the Default forwarding PHB , as specified in IETF RFC 2597.	the Default forwarding PHB , as specified in RFC 2597.	
	6.3.1, last paragraph	..authorization SHOULD be signaled in the P-Early-Media header field of either a subsequent	..authorization SHOULD be signaled in the P-Early-Media header field of a subsequent	Remove 'either' or add more to the sentence
	6.6			Need a correct reference
	6.9	Carrier's MUST support the History-Info Header and SHOULD support of the SIP Diversion header	Carrier's MUST support the History-Info Header and SHOULD support the SIP Diversion header	
	6.9	Item #1		Is there a continuation to item #1?
	Table 7.2	3. BYE response		Delete the duplicate row 3.
	Table 7.4, pages 28-29	*IETF RFC 3455 [24]: the P-Called-Party-ID header field extension*		Does the reference in the [ ] define anything in this document, or was it meaningful in the 3GPP

Author	Section Number	Current Text	Proposed Test	Comments
				standard?
	7.3	The support of IETF RFC 4028, which addresses SIP Timers specification, is optional. The carrier receiving the INVITE message shall comply with IETF RFC 3261 section 16.8 if IETF RFC 4028 is not supported	The support of IETF RFC 4028, which addresses SIP Timers specification, is optional. The carrier receiving the INVITE message shall comply with IETF RFC 3261. Section 16.8 if IETF RFC 4028 is not supported	

INSTRUCTIONS: Use this form to enter your comments on PTSC-LB-199. When complete, save the file on your hard drive and attach this file as instructed with your vote in ATIS Workspace (http://www.atis.org/faq/aws/#Ballot).

Section Number - Required.

Current/Proposed Text- Comment - These fields are required. Enter your comment and proposed change in these fields, respectively. Use plain text characters only.

Author	**Section Number**	**Current Text**	**Proposed Text**	**Comment**
Sprint	3.2			PT as they are not used
Sprint	5.3.1 last paragrap	figure below		Remove last sentence. There is no figure below.
Sprint	2			has reference numbers RFC references in the body do not follow any reference format such as appearing in square brackets. Also several RFCs are referenced in the body but not listed in section 2

Verizon Comments on PTSC-LB-199

EDITORIAL

In-line edits shown in PTSC-2015-00048R000

Author	Section Number	Current Text	Proposed Test	Comments
Mark Desterdick	1.1	This document was developed under a joint ATIS and SIP Forum collaboration. The document defines an IP NNI profile with an emphasis on VoIP. Other Multimedia services will be addressed in subsequent releases.	ADD: This document specifies an NNI profile applicable to the interface between the home network of the originating party and the home network of the terminating party; or between the home network of either party, and a transit network. The interface between the home and visited network of a roaming mobile user, is out of scope.	It should be stated that roaming is out of scope for this NNI.
	1.1		ADD: The scope of this documented is limited to the information exchanged at the reference points illustrated in Figure 4.1. The behavior of network elements upon receipt of such information is governed by other specifications.	It should also clarify that the scope of this document is limited to the information exchanged at the reference points illustrated in section 4.1

Author	Section Number	Current Text	Proposed Test	Comments
	1.2	The call control protocol SIP [RFC 3261] is defined in the IETF and is further refined by 3GPP or ATIS that reflect regional/and/or national differences in implementation.	The call control protocol SIP [RFC 3261] is defined in the IETF and is further refined by 3GPP or ATIS specifications.	Editiorial – added "specifications"
	4.2	More detail relating to interconnect models is provided in section C.2 of this document.	Remove text	Remove the reference or provide correct reference w/in this or other document.
	5.1	When sending a dialog-initiating request to a peer Carrier network	When sending a dialog-initiating or standalone request to a peer Carrier network	Generally speaking any reference to "dialog initiating" requests should be "dialog initiating or standalone".
	5.2.1	Identify the called user or service using a SIP URI as described above in Section **Error! Reference source not found.**	None – check section references throughout document	General comment to editor, check that the references match the sections. Many are "off by one".
	5.2.3	In table 5.2 – North American Numbering Plan formats		Since this is a profile for North America and

<table>
<tr><th>Author</th><th>Section Number</th><th>Current Text</th><th>Proposed Test</th><th>Comments</th></tr>
<tr><td></td><td></td><td><table>
<tr><td>URI</td><td>sip:+CCNSN@host;user=phone</td></tr>
<tr><td>Description</td><td>International number, CC=Country Code, NSN=National Significant Number</td></tr>
<tr><td>Reference</td><td>IETF RFC3966</td></tr>
<tr><td>Headers</td><td>R-URI, To, Request Contact, 3XX Contact,Diversion</td></tr>
</table></td><td></td><td>doesn’t include roaming, is an International number possible</td></tr>
<tr><td></td><td>5.4.2</td><td>Carrier's MAY optional support the Rate Based algorithm based on bilateral agreement between two carriers.</td><td>Carriers MAY…</td><td>Many times throughout this document, the plural case of a noun (e.g., “carriers”) is incorrectly written with an apostrophe before the ‘s’.</td></tr>
<tr><td></td><td>5.4.2</td><td>On receiving a dialog-initiating request that exceeds</td><td>On receiving a dialog-initiating or standalone request that exceeds</td><td></td></tr>
<tr><td></td><td>6.2</td><td>2. NOTE: If the terminating network receives an INVITE that does not contain an SDP offer, and wishes to provide early media to the calling user, it must establish the session over which to do so via an UPDATE request sent after the initial offer / answer exchange has concluded. The originating Carrier network performs the following action on receipt of a provisional response to a call-initiating INVITE.</td><td>NOTE: If the terminating network receives an INVITE that does not contain an SDP offer, and wishes to provide early media to the calling user, it must establish the session over which to do so via an UPDATE request sent after the initial offer / answer exchange has concluded.</td><td>NOTE text should be separated from the numbered list.</td></tr>
</table>

Author	Section Number	Current Text	Proposed Test	Comments
			2. The originating Carrier network performs the following action on receipt of a provisional response to a call-initiating INVITE.	
	6.2	NOTE: If the terminating network receives an INVITE that does not contain an SDP offer, and wishes to provide early media to the calling user, it must establish the session over which to do so via an UPDATE request sent after the initial offer / answer exchange has concluded.	NOTE: If the terminating network receives an INVITE that does not contain an SDP offer, and wishes to provide early media to the calling user, it SHOULD include a P-Early-Media header in the provisional response authorizing backward early media.	Change note to indicate need for P-Early-Media header in response.
	6.3.1	In the event that the nature of early media changes after initially signaled in an 18x response, the new authorization SHOULD be signaled in the P-Early-Media header field of either a subsequent message. Alternatively, the procedures described in **Error! Reference source not found.** may be used.		Is this the correct section reference?
	6.8	The originating Carrier network MAY provide the calling number of the originating user in the P-Asserted-Identity header field of dialog-initiating requests	The terminating Carrier network MAY obtain the calling name and number for caller-ID display from the contents of the P-Asserted-Identity header field contained in dialog-initiating or standalone requests.	
	6.8	The terminating Carrier network MAY obtain the calling name and number for caller-ID display from the contents of the P-Asserted-	The terminating Carrier network MAY obtain the	

Author	Section Number	Current Text	Proposed Test	Comments
		Identity header field contained in dialog-initiating requests.	calling name and number for caller-ID display from the contents of the P-Asserted-Identity header field contained in dialog-initiating or standalone requests.	
	6.9	Carrier's MUST support the History-Info Header and SHOULD support ~~of~~ the SIP Diversion header	Carriers MUST support the History-Info Header and SHOULD support the SIP Diversion header	
	6.9	1. forward the INVITE to the forward-to-user while remaining in the signaling path as a SIP Proxy or B2BUA~~, or~~	1. forward the INVITE to the forward-to-user while remaining in the signaling path as a SIP Proxy or B2BUA.	
	7.1.4	**Table 7. 1 - Major capabilities over NNI** *44 IETF RFC 3903 [21]: an event state publication extension to the session initiation protocol (PUBLISH method) o*	Change "o" to "n/a"	If not supporting roaming should this be n/a?
	7.1.4	*107 draft-ietf-soc-overload-control [165] feedback control m*	Change "m" to "o"	Needs to be 'o' here to align with previous text. Is also 'o' in TS 29.165. Items 106 and 107 should be the same; making one mandatory and

Author	Section Number	Current Text	Proposed Test	Comments
				the other optional doesn't make sense.