INTERNATIONAL INTERCONNECTION FORUM
FOR SERVICES OVER IP

(i3 FORUM)

Workstream "Technical Aspects"

White Paper

Mapping of Signalling Protocols ISUP to/from SIP, SIP-I

(Release 2) May 2010

Executive Summary

Mapping between ISUP and SIP, or ISUP and SIP-I, is a complex area with regard to disconnect cause values and this needs to be considered to ensure optimum behaviour for session control.

The most straightforward case is ISUP to SIP-I in accordance with specification ITU Q1912.5, Annex C Profile C . Since the ISUP message is encapsulated within the SIP message, correct conveyance of the ISUP information is guaranteed. Whereas, when ISUP has to be mapped into SIP there are a number of standards that differ and this has led to different vendor implementations.

A further level of complication exists when an ISUP to SIP conversion takes place in, for example, a Service Provider domain and another ISUP to SIP/SIP-I conversion occurs in the International Carrier domain. The level of end-to-end signalling transparency achieved depends on the compatibility of the two mapping activities. The more divergent these are, the less signalling transparency occurs.

The objective of this document is to be informative, outlining to the carrier industry that inconsistencies do exist under some conditions and may lead to undesired network behaviour. Carriers need to take full account of the complexities and ambiguities described in this paper when entering into bilateral cooperation for new SIP or SIP-I interconnections.

It is the view of the i3 Forum that these problems are sufficiently acute that the industry needs to address the issue as a matter of urgency to agree one common standard for mapping between SIP and ISUP and then implement this on all relevant vendor platforms as quickly as possible. To this end, i3 Forum is currently engaging the relevant standards bodies to initiate this pan-industry activity.

Since the release of issue 1 of this white paper, further detail has been added relating to the extent of the mapping discrepancies and their potential service impacts. In addition, the support of the Reason Header field in SIP has been addressed and examples of where this still lacks all the necessary functionality required, however there is much benefit in having reason Header support to alleviate the majority of mapping issues where ISUP reasons can be retrieved.

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1 Scope and Objective

This document addresses signalling interworking issues when converting from TDM to IP. These issues exist when inter-operating between legacy ISUP networks and next-generation VoIP networks using SIP-based protocols.

Mapping between ISUP and SIP, or ISUP and SIP-I, is a complex area with regard to disconnect cause values and this needs to be considered to ensure optimum behaviour for session control.

The objective of this document is to be informative, outlining to the carrier industry that inconsistencies do exist under some conditions and may lead to undesired network behaviour. Further work is needed in the standardization bodies and this has to be dealt with expeditiously.

The content of this white paper is based on the i3 Forum document "Technical Interconnection model for International Voice Services", Release 3, May 2010.

2 Acronyms

3GPP: 3rd Generation Partnership Program
ABR Answer to Bid Ratio
ASR Answer Seizure Ratio
CDR Call Detail Record
IETF Internet Engineering Task Force
ISDN Integrated Services Digital Network
ISUP ISDN User Part
ITU International Telecommunication Union
KPI Key Performance Indicator
NER Network Efficiency Ratio
NNI Network to Network Interface
RFC Request for Comments
SIP Session Initiation Protocol
SIP-I SIP with encapsulated ISUP
TDM Time Division Multiplexing

3 References

[1] i3 Forum, "Technical Interconnection Model for International Voice Services", Release 3.0, May 2010
[2] IETF RFC 3261 "SIP: Session Initiation Protocol", June 2002
[3] ITU-T Recommendation Q1912.5 "Interworking between Session Initiation Protocol and Bearer Independent Call Control or ISDN User Part", 2004
[4] IETF RFC 3398 – "ISUP to SIP Mapping", December 2002
[5] 3GPP TS 29.163 "Interworking between IP multimedia network and circuit switched networks, version 8.6.0, March 2009
[6] ITU-T Recommendation Q.850 "Usage of codes and location in the digital subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part", May 1998;
[7] IETF RFC 3326 "The Reason Header Field for the Session Initiation Protocol (SIP)", December 2002;
[8] ITU-T Recommendation Q.767. "Application of the ISDN User Part of CCITT Signalling System No. 7 for International ISDN interconnections";

4 Reference Configuration

The general reference configuration for international voice interconnection based on IP protocol is given in [1] and endorsed in this document. Carriers operate switching facilities which are fed with TDM traffic as well as voice over IP traffic from the domestic fixed and mobile networks. The interconnection between two Carriers makes use of signalling protocols and media flows carried onto an IP transport layer.

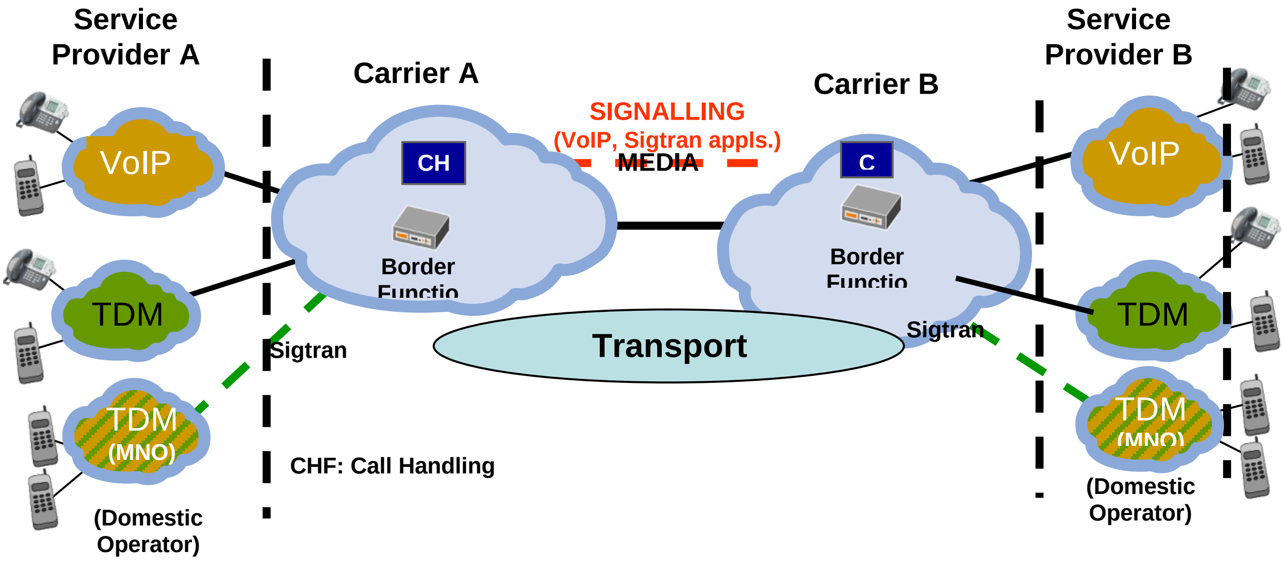


Figure 1 – General Reference Configuration

5 Applicable International standards

This document assumes that international carriers handle interconnections using:

a) IP-protocol interconnections utilising either SIP [2] or SIP-I [3], or
b) TDM-protocol interconnections, based on international ITU-T White Book ISUP v1, v2 and v3**,**

Note: as ITU- T White Book ISUP it is meant the collection of ITU-T recommendations which, in various Study Periods, have specified the ISUP protocol.

It is accepted that other IP interconnection protocols exist but these are outside the scope of this document and will not be addressed.

With regard to the protocol interworking, including disconnect cause to response codes mapping, the following standards apply:

- RFC 3398 – "ISUP to SIP Mapping" [4]
- ITU-T Q.1912.5 [3]
- 3GPP TS 29.163 "Interworking between IP multimedia network and circuit switched networks" [5].

All of these standards detail the mapping between the two protocol stacks applicable to IP and TDM networks. There are, however, significant differences between the mapping schemes as described in this document.

6 Interworking Issues: ISUP-SIP, ISUP- SIP-I

There are a number of issues that need to be addressed when a session encounters protocol interworking as it progresses through multiple carriers. As the protocol used to set-up the session is inter-worked, care must be given to:

1) Messages mapping
2) Parameter mapping
3) Disconnect causes and response codes mapping

A significant potential impact of having poor mapping between protocols would be a degraded service to client operators caused by incorrect behaviours such as:

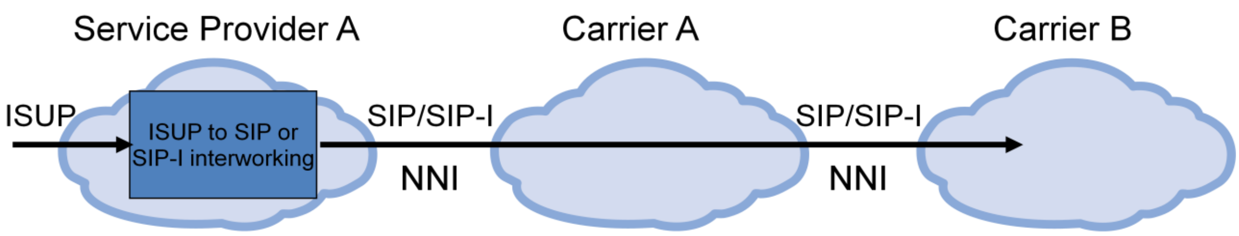
- Loss of end-to-end service information needed to support services
- Automatic re-routing causes used by some carriers, based on one or several specific disconnect causes. Typically, cause value 34 is used to reroute
- Accounting interchanged between clients and carriers based on cause values written to CDRs
- Voice KPI statistics and reporting, dependent on disconnect causes, for example, ASR, ABR, NER.

If interworking is only performed once, two scenarios are possible:

1 – Interworking is performed within the Service Provider domain. In this case, carriers only handle SIP/SIP-I traffic (see figure 2a below), mapping would therefore be the responsibility of the Service Provider;

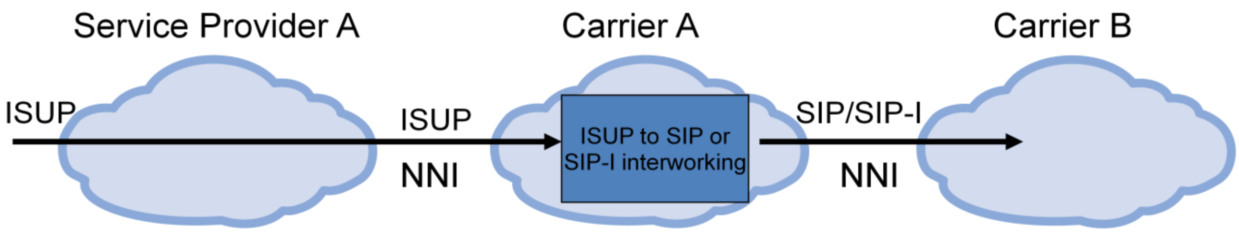
2 – Interworking is performed by one of the carriers (see figure 2b below).

This document focuses on the second scenario and the diagram shows this occurring at Carrier A.



Interworking is performed in the Service Provider A network. Carrier A is unaware of ISUP - SIP mapping

Figure 2a – Interworking function locations



Interworking is performed in the Carrier A network. Carrier A is responsible for ISUP - SIP mapping

Figure 2b – Interworking function locations

7 Message mapping between ISUP – SIP and ISUP –SIP-I

Signalling messages as well as their mandatory and optional parameters used in international interconnect are listed in ITU-T Q.767 [8] "Application of the ISDN User Part in of CCITT Signalling System No. 7 for international ISDN interconnections". The list contains 6 message groups

1) Forward set-up
2) General Supervision
3) Backward Supervision
4) Call Supervision
5) Circuit Supervision
6) Circuit group supervision

These 6 message groupings have, in total, 24 messages. In Annex A, mapping of these messages to SIP, based on 3GPP TS 29.163 [5], can be found.

From this table it can be seen that 11 messages are actually mapped to SIP methods and 13 are not mapped. It is necessary, therefore, to analyse the un-mapped messages to determine if they are indispensable for routing calls between TDM and SIP domains in both directions.

The first three message groups, Forward set-up messages; Backward set-up messages; and General supervision are fully mapped.

The fourth group, Call supervision, is also mapped with the exception of the Forward Transfer (FOT**)** message which is for connecting an operator to assist in call set-up. This message is not essential for call routing and its future use (if, indeed, it will be used), will depend on the operator applications requirements, (for example, foreign language support) for VoIP international interconnects.

In the fifth group, Circuit supervision, and sixth group, Circuit Group supervision, most of the messages are not mapped to the SIP Method. However, in a VoIP domain, there are no actual circuits or circuit groups, so therefore, circuit or circuit group supervision is unnecessary.

8 Parameter Mapping Issues: ISUP-SIP, ISUP- SIP-I

8.1 Considerations on ISUP, SIP interworking

In ISUP/SIP interworking, information carried as parameters in the ISUP message header is mapped to the SIP message. ISUP Information not mapped or inadequately mapped is lost.

As an example, for end-to-end ISDN connections, since some of the ISUP parameters may not get mapped into the SIP messages, it is unclear what level of end-to-end capability can be provided over native SIP interconnections. It is likely this level is variable and depends on the specific IETF RFCs implemented in a given network and thus end-to-end ISDN service cannot be guaranteed. i3 Forum therefore recommends that, where it is intended end-to-end ISDN service traffic is to be delivered across an interconnect, this interconnect shall support the SIP-I protocol. Non-ISDN traffic may be routed over native SIP or SIP-I routes.

8.2 Considerations on ISUP, SIP-I interworking

Mapping ISUP to/from SIP-I is a different case than mapping to/from native SIP since the ISUP messages are encapsulated in the body of the SIP messages. As a result, the carrier conveys the signalling information transparently by using the encapsulation mechanism. The receiving network can extract the full ISUP message from the body of the SIP message.

This encapsulation will ensure the integrity of ISUP parameters and disconnect cause information between service providers.

In this case, for example of end-to-end ISDN connections, since the ISUP parameters are mapped into the SIP messages, end-to-end ISDN service is guaranteed provided that the appropriate bearer capabilities are supported.

9 Disconnect Cause Value Mapping

9.1 ISUP – SIP Interworking

In ISUP, the disconnect cause values contained in the release message are defined in ITU standard Q.850 [6] and are in the range 1 to 127.

In SIP, the error response codes are equivalent to ISUP disconnect cause values and are in the range 4xx, 5xx, and 6xx. There is no one-to-one mapping for each TDM cause into SIP protocol error codes. Consequently, mapping between protocols therefore inevitably leads to loss of cause granularity as previously described.

In a call flow where the origination and termination are both ISUP and the Carrier-Carrier interconnect is based on SIP, the disconnect cause information will get mapped from ISUP to SIP error response code and then back to an ISUP disconnect cause in the next interworking. The end-to-end disconnect cause transparency will be degraded between the two Service Provider networks. Refer to section 7 for more details.

In practice this means that it is possible for a cause value sent from the terminating ISUP node to the originating ISUP node to be changed by the interworking function and the value received will, therefore, be different from that originally returned from the terminating node.

In Annex B the mapping from SIP error response codes to ISUP disconnect cause values, according to both [3] and [4] is given. Annex C gives the mapping of the ISUP disconnect cause values to SIP error response codes according to the same two specifications.

An example of mishandling of disconnect cause is:

Example 1: Call ISUP->SIP->ISUP: Using RFC 3398 mapping

ISUP (REL): 2 – "No route to network" -> SIP (code): 404 - "Not found" -> ISUP (REL) 1- "Unallocated/unassigned number"

Example 2: Call ISUP->SIP->ISUP: Using Q.1912.5 mapping

ISUP (REL): 2 – "No route to network" -> SIP (code): 500 - "Server internal error" -> ISUP (REL) 127- "Interworking unspecified"

In both examples the original cause is REL=2 sent out from the terminating side, but what the backwards carrier received is REL=1, following RFC3398, or REL=127, following Q.1912.5.
Neither of the two standard defined mappings is preserving the original REL value.

As a partial solution, the use of the Reason Header in accordance with IETF RFC 3326 [7] is recommended to enable the inclusion of the ISUP release cause values. This is only a partial solution (though the remaining deficiency is very small) because the Location field information associated with an ISUP Release Cause is not carried. The implications of this are described later in this document.

According to [7], With reference to the Reason Header "*It is normally present in BYE and CANCEL messages but it may be included in any request within a dialog, in any CANCEL request and in any response whose status code explicitly allows the presence of this header field*."

Note, however, that according to RFC 3326, clients and servers are free to ignore this header field. It has no impact on protocol processing or re-routing in most applicable network elements, as it is only accepted as additional information.

It can be concluded from the above extract from RFC3326 that the actual interoperability behaviour between nodes may differ depending on the implementation of such functionality by the respective vendors.

9.1.1 Mapping Stability

During discussion of this issue at i3 Forum, the issue of when mapping between protocols becomes constant regardless of how many further mapping actions take place was considered and is now referred to as mapping stability. A definition of this concept follows:

> *For a given initial input SIP error response code or ISUP release cause value, it is the point at which further conversions between SIP to ISUP and ISUP to SIP result in the same new value of each protocol being returned as that in the previous iteration. From this point, regardless of the number of further iterations, the mapping process results in the same result. The mapping is then said to be **stable.***

9.1.2 Major Mapping Issues with ITU Scheme

The main issue with the ITU mapping scheme is the serious lack of granularity that the mapping generates.

For ISUP to SIP mapping, many ISUP release cause values are mapped to the same SIP error response code, (the most notable example being that of 500 *("Server internal error").* Similarly, for SIP to ISUP mapping, many SIP error response codes are mapped to release cause value 127: *("Interworking unspecified")* Please see Annex B for details.

For network operation, this mapping makes trouble location and resolution very difficult and poses significant issues with the accuracy of any NER statistics generated.

In the case of multiple SIP/ISUP and ISUP/SIP mappings on a single call setup, the ITU scheme achieves mapping stability after usually one iteration and exceptionally, two.

9.1.3 Major Mapping Issues with 3GPP Scheme

The main issues with the 3GPP scheme [5] are the same as those for the ITU mapping scheme. The delta between these two very similar schemes is contained in Annex D.

9.1.4 Major Mapping Issues with RFC 3398 Scheme

With the RFC 3398 scheme, the granularity is much-improved over that provided by the ITU scheme, but a different problem manifests itself where there are multiple SIP/ISUP and ISUP/SIP mappings for a given call set-up. In comparison with the ITU scheme, the RFC3398 scheme in some cases takes several iterations before mapping stability is achieved. This results in a very confused picture of call failure behaviour depending on the point in the chain of network components at which the signalling is analysed.

ISUP Cause Value & Location	SIP error response code	ISUP Cause Value & Location	SIP error response code	ISUP Cause Value & Location	SIP error response code	ISUP Cause Value & Location
19/any	480	18/network	408	102/network	504	102/network

In the example above, the case of multiple SIP/ISUP and ISUP/SIP mappings on a single call setup, the RFC3398 scheme achieves mapping in the worst-case example only after five iterations.

For network operation, this mapping change at re-iteration also makes trouble location and resolution very difficult and again poses significant issues with the accuracy of any NER statistics generated.

9.1.5 Mapping Incompatibility between networks

The scenario needs to be addressed whereby two interconnecting networks use different mapping schemes. There are two cases:

a) ITU before 3398
b) 3398 before ITU.

The effects of the above cases are different but both have the potential to seriously degrade the end to end service, its measurement and trouble location and resolution.

It is therefore recommended that unless unavoidable, native SIP interconnects without Reason Header support, either between two Carriers, or between a Carrier and a Service Provider, should strive to achieve common mapping scheme implementation.

9.2 ISUP – SIP Interworking with Reason Header Support

With the implementation of native SIP that supports Reason Header (RFC 3326), the ISUP cause value is written into this Reason Header field. Providing that both the sending and receiving SIP platforms support this field, then *most* ISUP release cause value information will be preserved and written correctly into the ISUP function at the interworking point on the originating side.

It needs to be understood, however, that this is not a complete preservation of the ISUP information, as the ISUP Location parameter is not written into the Reason Header and therefore the differentiation between locations is lost.

9.2.1 Example 1 SIP error response code 486 – Busy Here

In a SIP-I environment, the next operation of the receiving node will be dependent on the value in the Location parameter.

<u>Location Value = User</u>
The call attempt will be terminated, and a busy indication will be returned towards the caller (for example, either Engaged Tone or the relevant Q850 cause value over a Q931 interface). At the SP-A domain, an attempt may be made to initiate the CCBS capability, depending on the end-user feature-set.

<u>Location Value = Network</u>
The receiving node would take the relevant action based on its routing configuration for the destination involved. This will typically be to either:

a) Make a subsequent call attempt over the next choice of outgoing route for the destination; or
b) If there is no further alternative route for this destination, return Network Congestion as the result and fail the call attempt.

In the terminating ISUP SP function, a SIP response of 486 (Busy Here) should have the location of ‘User’ added if the implemented mapping function is comprehensive in capability.

9.2.2 Example 2 ISUP Release Cause Value 34 - No Circuit/channel Available

In the case of CV=34, again the action to be taken would differ between a transit node receiving this from a succeeding node, and a terminating node receiving this from the end user interface. Because the Location = TN or Location = User information is lost, the effect will be the same as in the previous example related to SIP error response codes.

9.2.3 Predicted Impacts of loss of Location Information

In both of the above examples, even though Reason Header to RFC3326 is implemented in the SIP functions of both Carrier A and Carrier B, the location information from SP-B is not preserved. The significance of this to Carrier A is that it is not possible to differentiate between User Busy and Network Busy.

Carrier A will now inevitably miss-handle some call attempts on receipt of “Busy Here” or ISUP CV=34. It is probable that a Carrier would by default act differently for these, giving one problem in one case and the inverse problem in the other.

In the case where a carrier always regards this as indicating User Busy, and the location of the event was ‘Network’, any alternative routing options in the Carrier A node will not be used, thus failing calls in the case where the User is free but the forward network is busy, the wrong indication would be returned to SP-A and the caller and NER statistics will be impacted, providing a higher value of NER than the network is delivering.

Alternatively, if Carrier A assumes that “Busy Here” indicates network congestion, then in the case of User Busy, further ineffective call set-up attempts would be initiated over all alternative routing choices, the wrong indication would be returned to SP-A and the caller and NER statistics will be impacted, providing a higher value of NER than the network is delivering.

It is possible that in the carrier domain, Busy Here would be treated as User Busy, whilst CV=34 would be treated as network congestion.

9.2.4 IETF Draft regarding Reason in Responses

There is a current initiative within IETF to further qualify SIP responses by use of the Q.850 reason. This is being led by R. Jesske and L. Liess of DT. The Internet draft identity is as follows: draft-jesske-dispatch-reason-in-responses.

i3 Forum recommends that this draft is taken as one input into the wider industry meetings planned for later in 2010 with the objective of agreeing a single mapping standard for the whole industry.

9.3 ISUP – SIP-I Interworking

The issues described in the previous section are at the SIP level also relevant in the SIP-I – ISUP interworking scenario.

In the case of SIP-I, although the SIP messages handled by carriers will still contain the SIP status code values, the actual ISUP disconnect cause values are preserved and encapsulated in the message body.

When there is a difference of significance between the SIP error response code and the ISUP disconnect cause value in SIP-I, the ISUP disconnect cause always takes precedence.

This interworking case is not as complex as interworking via native SIP, since many network elements, even though they do not read the content of ISUP messages, can read and act on the Release field. In case of different criteria, according to ITU-T Q.1912.5 [3], the ISUP value takes precedence over the SIP value.

Note that some vendors may not be compliant to this ITU standard implementation. Consequently, incorrect operation and failure reason reporting could occur.

10 Current Carrier Implementations of Mapping Schemes

The i3 Forum Technical working group conducted a survey of members between November 2009 and January 2010 to ascertain the prevalence of deployment of the two mapping schemes: ITU and RFC 3398. The results do not show a clear preference for either scheme.

It needs to be understood that Carriers and Service Providers will have developed OSS capabilities to act with their current implemented mapping schemes.

International standards bodies, carriers, service providers and vendors will need to work together to agree and implement one common mapping scheme. Once this has been achieved, it will then be necessary for each Carrier and Service Provider to make the required changes to their networks and systems to achieve compatibility with the new mapping scheme.

10.1 Current Carrier Implementations of Mapping Exceptions

A number of Carriers have implemented changes to some mappings within the scheme implemented within their network to better address particular issues with the current schemes. Clearly, this is an undesirable situation and unless the underlying issues are fully addressed, i3 Forum predicts that these exceptions will proliferate in both the Carrier and Service Provider domains, further complicating the transparent conveyance of accurate clearing cause information.

11 Recommendations to Carriers and Interested Industry Bodies

As a consequence of the analysis carried out by the i3 forum technical working group, the following recommendations are given:

a) The industry standards bodies (e.g. ITU, 3GPP, IETF) need to jointly define one common standard for the whole industry for mapping between the ISUP and SIP protocols.

b) Other pan-industry bodies with an interest in this area, (for example: The SIP Forum, The Multi-Service Forum) should work collaboratively, together with i3 Forum, into the combined ITU/IETF initiative to ensure the fullest possible dataset to drive the standard-setting exercise forward is provided.

c) The industry needs to deliver a single, aligned standard that will be described in both the relevant ITU and 3GPP documents and a new IETF RFC.

d) Future changes: no changes or updates to this agreed standard shall be countenanced by a single standards body, but only by full agreement between ITU, 3GPP and IETF.

e) Industry vendors should incorporate the new mapping standard at the earliest opportunity into their existing products to enable deployed networks to be easily enhanced to adopt the new standard.

f) Industry vendors should undertake to ensure that all new products responsible for ISUP/SIP mapping are fully compliant to the pan-industry common standard.

g) The use of the Reason Header field whenever possible when using native SIP

h) Industry introduction of ISUP Location information into native SIP as an addition to reason header support.

12 Conclusions

The objective of this document is to be informative, outlining to the carrier industry that inconsistencies do exist under some conditions and may lead to undesired network behaviour. Carriers need to take full account of the complexities and ambiguities described in this paper when entering into bilateral cooperation for new SIP or SIP-I interconnections.

As far as message mapping is concerned, the analysis indicates that a complete and unambiguous mapping exists between ISUP and SIP messages. The only exception, the *Forward Transfer message*, does not pose a major problem since it is rarely used.

Regarding parameter mapping the use of SIP-I guarantees transparency. Care should be taken, when SIP is used, to ensure implementation supports required capabilities. Specifically, if full ISDN support has to be guaranteed then SIP-I has to be used.

From the perspective of disconnect cause value mapping, there is no one-to-one mapping for each TDM cause into SIP protocol error codes. Consequently, mapping between protocols therefore inevitably leads to loss of cause granularity as previously described. In addition, though three standards are available, these standards are not consistent with each other.

This issue can be solved by using the Reason Header as already recommended by ITU-T. Accordingly, in ISUP to SIP Interworking, the Reason Header field should be populated and preserved whenever this is technically feasible. Usage of this mechanism would require all vendors to design equipment incorporating this capability.

As far as the use of SIP-I is concerned, the transparency of the disconnect cause value is guaranteed by the encapsulation of the ISUP message into the SIP body.

It is the view of the i3 Forum that these problems are sufficiently acute that the industry needs to address the issue as a matter of urgency to agree one common standard for mapping between SIP and ISUP and then implement this on all relevant vendor platforms as quickly as possible. To this end, i3 Forum is currently engaging the relevant standards and other interested bodies to initiate this pan-industry activity.

13 ANNEX A - Mapping from ISUP messages to SIP messages

Mapping based on 3GPP TS 29.163

#	Group	ISUP MESSAGE		SIP MESSAGE
1	*Forward set-up*	IAM	Initial address	INVITE
		SAM	Subsequent address	collecting address - INVITE
3	*General supervision*	COT	Continuity	(success) SDP indicating pre-conditions met. UPDATE
4	*Backward supervision*	ACM	Address complete	180 RINGING or 183 SESSION PROGRESS
5		CON	Connect	200 OK (INVITE)
6		CPG	Call Progress (alerting)	180 RINGING or 183 SESSION PROGRESS
7	*Call supervision*	ANM	Answer	200 OK (INVITE)
8		FOT	Forward transfer	*No Equivalent*
9		REL	Release	BYE, CANCEL
10	*Circuit supervision*	RLC	Release complete	*No Equivalent*
11		CCR	Continuity check request	*No Equivalent*
12		RSC	Reset circuit	200 OK -> BYE 200 OK -> 480 Temporarily Unavailable CANCEL
13		BLO	Blocking	*No Equivalent*
14		UBL	Unblocking	*No Equivalent*
15		BLA	Blocking acknowledgement	*No Equivalent*
16		UBA	Unblocking acknowledgement	*No Equivalent*
17		SUS	Suspend	*No Equivalent*
18		RES	Resume	*No Equivalent*
19	*Circuit Group supervision*	CGB	Circuit group blocking	200 OK -> BYE 200 OK -> 480 Temporarily Unavailable CANCEL
20		CGU	Circuit group unblocking	*No Equivalent*
21		CGBA	Circuit group blocking ack.	*No Equivalent*
22		CGUA	Circuit group unblocking ack.	*No Equivalent*
23		GRS	Circuit group reset	200 OK -> BYE 200 OK -> 480 Temporarily Unavailable CANCEL
24		GRA	Circuit group reset ack.	No Equivalent

14 ANNEX B - Mapping from SIP Response Codes to ISUP Disconnect Cause Values

	The yellow rows indicate a mismatch between the IETF RFC 3398 and ITU-T Rec. Q.1912.5				
4xx, 5xx, 6xx on INVITE		**REL (Cause Value) ISUP (Follows IETF RFC 3398)**		**REL (Cause Value) ISUP (Follows ITU-Q.1912.5)**	
Error Response Code		Cause Value		Cause Value	
400	Bad Request	41	*("Temporary Failure")*	127	*("Interworking unspecified")*
401	Unauthorized	21	*("Call rejected")*	127	*("Interworking unspecified")*
402	Payment Required	21	*("Call rejected")*	127	*("Interworking unspecified")*
403	Forbidden	21	*("Call rejected")*	127	*("Interworking unspecified")*
404	Not Found	1	*("unallocated (unassigned) number)*	1	*("unallocated (unassigned) number)*
405	Method Not Allowed	63	*("Service option not available, unspecified")(Class default)*	127	*("Interworking unspecified")*
406	Not Acceptable	79	*("Service option not implemented, unspecified")*	127	*("Interworking unspecified")*
407	Proxy authentication required	21	*("Call rejected")*	127	*("Interworking unspecified")*
408	Request Timeout	102	*("Recover on Expires timeout")*	127	*("Interworking unspecified")*
410	Gone	22	*("Number changed (without diagnostic)")*	22	*("Number changed (without diagnostic)")*
413	Request Entity too long	127	*("Interworking unspecified")*	127	*("Interworking unspecified")*
414	Request-uri too long	127	*("Interworking unspecified")*	127	*("Interworking unspecified")*
415	Unsupported Media type	79	*("Service option not implemented, unspecified*	127	*("Interworking unspecified")*
416	Unsupported URI scheme	127	*("Interworking unspecified")*	127	*("Interworking unspecified")*
420	Bad Extension	127	*("Interworking unspecified")*	127	*("Interworking unspecified")*
421	Extension required	127	*("Interworking unspecified")*	127	*("Interworking unspecified")*
423	Interval Too Brief	127	*("Interworking unspecified") (Class default)*	127	*("Interworking unspecified")*
480	Temporarily Unavailable	18	*("no user responding")*	20	*("Subscriber absent")*
481	Call/Transaction does not exist	41	*("Temporary Failure")*	127	*("Interworking unspecified")*
482	Loop Detected	25	*("Exchange routing error")*	127	*("Interworking unspecified")*
483	Too many hops	25	*("Exchange routing error")*	127	*("Interworking unspecified")*
484	Address Incomplete	28	*("Invalid number format (address incomplete) ")*	28	*("Invalid Number format(address incomplete)")*
485	Ambiguous	1	*("Unallocated (unassigned) number")*	127	*("Interworking unspecified")*
486	Busy Here	17	*("User busy")*	17	*("User busy")*
487	Request terminated			127	*Interworking or no mapping*
488	Not acceptable here			127	*("Interworking unspecified")*
491	Request Pending				*no mapping*
493	Undecipherable			127	*("Interworking unspecified")*
500	Server Internal error	41	*("Temporary failure")*	127	*("Interworking unspecified")*
501	Not implemented	79	*("Service or option not implemented, unspecified")*	127	*("Interworking unspecified")*

502	Bad Gateway	38	*("Network out of order")*	127	*("Interworking unspecified")*
503	Service Unavailable	41	*("Temporary failure")*	127	*("Interworking unspecified")*
504	Server timeout	102	*("Recovery on timer expiry")*	127	*("Interworking unspecified")*
505	Version not supported	127	*("Interworking, unspecified")*	127	*("Interworking unspecified")*
513	Message too large	127	*("Interworking, unspecified")*	127	*("Interworking unspecified")*
580	Precondition failure			127	*("Interworking unspecified")*
600	Busy Everywhere	17	*("User busy")*	17	*("User busy")*
603	Decline	21	*("Call rejected")*	21	*("Call rejected")*
604	Does not exist anywhere	1	*("Unallocated (unassigned) number")*	1	*("Unallocated number")*
606	Not acceptable			127	*("Interworking unspecified")*

15 ANNEX C - Mapping from ISUP Disconnect Cause Values to SIP Response Codes

	The yellow rows indicate a mismatch between the IETF RFC 3398 and ITU-T Rec. Q.1912.5		

REL ISUP -Cause Disconnect Values -		SIP Message (Follows IETF RFC 3398)		SIP Message (Follows ITU-Q.1912.5)
1	*("Unallocated (unassigned) number")*	404	Not found	404 Not Found
2	*("No route to network")*	404	Not found	500 Server Internal Error
3	*("No route to destination")*	404	Not found	500 Server Internal Error
4	*("Send special information tone")*			500 Server Internal Error
5	*("Misdialled trunk prefix")*			404 Not Found
17	*("User busy")*	486	Busy here	486 Busy Here
18	*("No user response")*	408	Request Timeout	480 Temporarily unavailable
19	*("No answer from the user")*	480	Temporarily unavailable	480 Temporarily unavailable
20	*("Subscriber absent")*	480	Temporarily unavailable	480 Temporarily unavailable
21	*("Call rejected")*	403	Forbidden	480 Temporarily unavailable
22	*("Number changed")*	410 301	Gone or Moved Permanently	410 Gone
23	*("Redirection to new destination")*	410	Gone	No interwork
25	*("Exchange routing error")*			480 Temporarily unavailable
26	*("Non-selected user clearing")*	404	Not found	
27	*("Destination out of order")*	502	Bad Gateway	502 Bad Gateway
28	*("Invalid number format (address incomplete)"*	484	Address incomplete	484 Address Incomplete
29	*("Facility rejected")*	501	Not implemented	500 Server Internal Error
31	*("Normal, unspecified") (Class default)*	480	Temporarily unavailable	480 Temporarily unavailable
Cause Value in the Class 010 (resource unavailable Cause Value No. 34)		503 Service unavailable		486 Busy here if Diagnostics Indicator includes the CCBS indicator
38	*("Network out of order")*	503	Service unavailable	500 Server Internal Error
41	*("Temporary failure")*	503	Service unavailable	500 Server Internal Error
42	*("Switching equipment congestion")*	503	Service unavailable	500 Server Internal Error
44	*("Requested circuit/channel not available")*			500 Server Internal Error
46	*("Precedence call blocked")*			500 Server Internal Error
47	*("Resource unavailable, unspecified")*	503	Service unavailable	500 Server Internal Error
Cause Value in the Class 010 (recource unavailable Cause Value No. 38, 41-44,46,47) (47 is class default)				500 Server Internal Error
50	*("Requested facility not subscribed")*			500 Server Internal Error
55	*("Incoming class barred within Closed User Group (CUG)")*	403	Forbidden	500 Server Internal Error
57	*("Bearer capability not authorized")*	403	Forbidden	500 Server Internal Error
58	*("Bearer capability not presently available")*	503	Service unavailable	500 Server Internal Error
63	*("Service option not available,unspecified") (Class default)*			500 Server Internal Error
65	*("Bearer capability not implemented")*	488	Not acceptable here	500 Server Internal Error
66				500 Server Internal Error
69	*(Requested Facility not implemented)*			500 Server Internal Error
70		488	Not acceptable here	500 Server Internal Error
79	*("Service option not available,unspecified")*	501	Not implemented	500 Server Internal Error

Cause Value in the Class 100 (Service or option not implemented, Cause Value No. 65, 66,69,70,79) (79 is class default)				500 Server Internal Error
87	*("User not member of Closed User Group(CUG)")*	403	Forbidden	500 Server Internal Error
88	*("Incompatible destination")*	503	Service unavailable	500 Server Internal Error
90	*("Non existent CUG")*			500 Server Internal Error
91	*("Invalid transit network")*			404 Not Found
95	*("Invalid message (Class default)")*			500 Server Internal Error
97	*("Message type non-existent or not implemented")*			500 Server Internal Error
99	*("Information element/parameter non-existent or not implemented")*			500 Server Internal Error
102	*("Recover on Expires timeout")*	504	Server Timeout	480 Temporarily unavailable
103	*("Parameter non-existent or not implemented, passed on")*			500 Server Internal Error
110	*("Message with unrecognized parameter, discarded")*			500 Server Internal Error
111	*("Protocol error unspecified") (Class default)*	500	Internal Server Error	500 Server Internal Error
127	*("Interworking, unspecified") (Class default)*	500	Internal Server Error	480 Temporarily unavailable

16 ANNEX D - Mapping delta between ITU-T Q1912.5 and 3GPP TS.29.163 and subsidiary documents.

The 3GPP standard is aligned with ITU Q1912.5 Profile C, however, there are some differences, not least to the formal references to applicable IETF RFCs.

In document TS 29.231, the 3GPP list of RFCs differs from the corresponding list in Q.1912.5, sub-clause C.1.1.2 as follows:

- 3GPP uses RFC3966 “The tel URI for Telephone Numbers” to replace RFC2806 (now obsolete) in Q.1912.5;
- 3GPP uses RFC4566 “SDP: Session Description Protocol” to replace RFC2327 (now obsolete) in Q.1912.5;
- 3GPP adds [optional] RFC4028 “Session Timers in the Session Initiation Protocol (SIP)”;
- 3GPP adds [mandatory] RFC4320 “Actions Addressing Identified Issues with the Session Initiation Protocol's (SIP) Non-INVITE Transaction”;
- 3GPP adds [optional] RFC4715 “The Integrated Services Digital Network (ISDN) Subaddress Encoding Type for tel URI;
- 3GPP adds [optional] RFC5079 “Rejecting Anonymous Requests in the Session Initiation Protocol (SIP)”.

Document TS 29.235, sub-clause 4.2.2.1, states that overlap signalling is not propagated into the 3GPP domain. Where an incomplete number is received at the ITU Q1921.5 interworking unit the 3GPP standard requires this interworking unit to collect all remaining digits prior to the call being forwarded into the 3GPP domain.