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 3) May 2011

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 in most circumstances. However, 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 multiple conversions (i.e. signalling protocol interworking) take place in the International Carrier domain.

The level of end-to-end signalling transparency achieved depends on the compatibility of the two (or more) mapping activities. The more divergent these are the less signalling transparency occurs.

In addition, these mapping inconsistencies can also adversely affect the quality KPIs that are generated in the Carrier and/or Service Provider network, leading to differences in values reported by the parties involved in transporting the call.

This can in turn result in difficulties in identifying the root cause of an issue and, potentially worse, applying incorrect call routing management. This behaviour can affect whether SLAs have been met or not by a particular party.

The objective of this document is twofold. On the one hand, it aims to outline to the carrier industry that inconsistencies do exist under some conditions and may lead to undesired network behaviour. On the other hand, on the basis of the joint activity carried out by 3GPP and i3 Forum in late 2010 / early 2011, it identifies actions Carriers need to take to limit these complexities and ambiguities when implementing new SIP or SIP-I interconnections.

Specifically, it is advised to implement the new mapping specified by 3GPP TS29.163 Release 7.22.0 [5], or any other Release distributed after March 2011 that incorporates this new mapping.

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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 and quality KPI reporting.

The objective of this document is twofold. On the one hand, it aims to outline to the carrier industry that inconsistencies do exist under some conditions and may lead to undesired network behaviour. On the other hand, on the basis of the joint activity carried out by 3GPP and i3 Forum in late 2010 / early 2011, it identifies actions that carriers need to take to limit these complexities and ambiguities when implementing new SIP or SIP-I interconnections.

Specifically, it is advised to implement the new mapping specified by 3GPP TS29.163 Release 7.22.0 [5], or any other Release distributed after March 2011 that incorporates this new mapping.

The content of this white paper complements the i3 Forum document "Technical Interconnection model for International Voice Services", [1].

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 Effictiveness 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 4.0, May 2011

[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" Release 7.22.0, Kansas City, March 2011

[6] 3GPP TS 29.163 "Interworking between IP multimedia network and circuit switched networks" Release 7.21.0, and all versions prior to March 2011.

[7] 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;

[8] IETF RFC 3326 "The Reason Header Field for the Session Initiation Protocol (SIP)", December 2002;

[9] ITU-T Recommendation Q.767. “Application of the ISDN User Part of CCITT Signalling System No. 7 for International ISDN interconnections”;

4 Reference Configuration

Figure 1 below shows the general reference configuration for international voice interconnection, as given in [1]. 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.

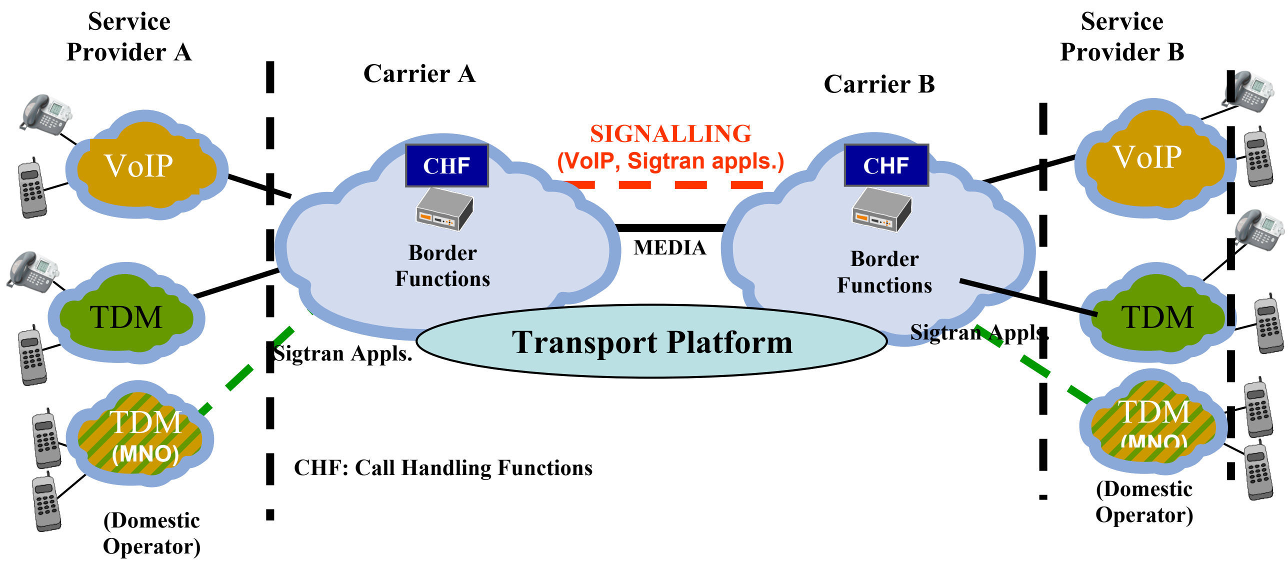


Figure 1 – General Reference Configuration

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:

- ITU-T Q.1912.5 [3]
- 3GPP TS 29.163 “Interworking between IP multimedia network and circuit switched networks” [5], [6]
- IETF RFC 3398 – “ISUP to SIP Mapping” [4]

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 later in this document.

6 Interworking between ISUP-SIP and between ISUP- SIP-I

There are a number of issues that need to be addressed when considering 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) Message mappings
2) Parameter mappings
3) Disconnect cause and response code mappings, which can be influenced by Location Information.

A significant potential impact to Carriers or Service Providers of having poor mapping between protocols is degraded service caused by incorrect behaviours such as:

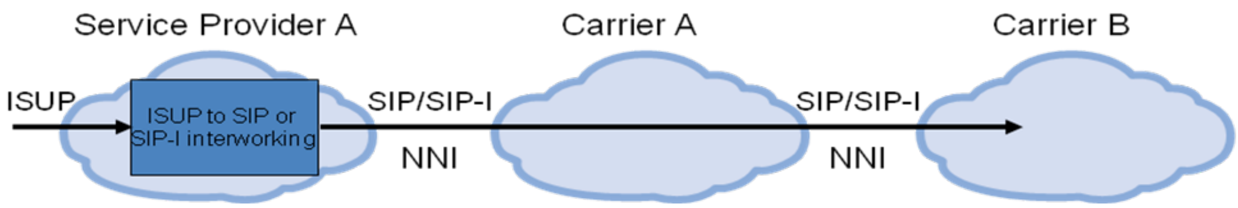
- Loss of end-to-end service information needed to properly support services.
- Incorrect automatic re-routing based on one or several specific disconnect causes. Typically, for instance, disconnect cause value 34 results in rerouting but disconnect cause value 17 does not since it is a user response.
- Accounting discrepancies due to information interchanged between clients and carriers based on disconnect cause values written to CDRs
- Inconsistent voice quality KPI statistics and reporting, dependent on disconnect cause values, for example, ASR, ABR, and 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); the primary mapping would therefore be the responsibility of the Service Provider;

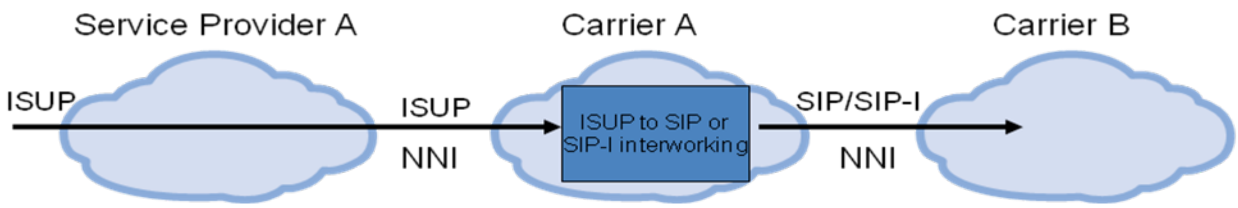
2 – Interworking is performed within the Carrier domain. The primary mapping is therefore performed by one of the carriers (see figure 2b below).

This document focuses on both of these scenarios.



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 mappings between ISUP – SIP and between ISUP –SIP-I

SS7 signalling messages, as well as their mandatory and optional parameters used in international interconnections, are listed in ITU-T Q.767 [9] “Application of the ISDN User Part 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 six message groups comprise, in total, 24 messages. In Annex A, the mapping of these messages to SIP methods, 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, therefore, circuit or circuit group supervision is unnecessary.

8 Parameter Mappings between ISUP-SIP and between 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 that 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 for end-to-end ISDN service traffic 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-I 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-I message.

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

9 Disconnect Cause Value Mappings between ISUP-SIP and between ISUP-SIP-I

9.1 Desired Behaviours in a Disconnect Cause Value to Response Code Mapping

9.1.1 Mapping Granularity

Whenever possible mappings should be specific, i.e., a CV (Cause Value) or Response code should only be used in only one mapping. This will allow for a clearer meaning of the events occurring in the network and, hence, a better chance to identify and debug issues that may occur.

9.1.2 Mapping Stability

The issue of when the mapping between protocols becomes constant regardless of how many further mapping actions take place was considered with i3 Forum 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 disconnect cause value, there is a point at which further conversions between SIP to ISUP and ISUP to SIP result in the same value of each protocol being returned as it was in the previous iteration. From this point on, regardless of the number of further iterations, the mapping process results in the same result. The mapping is then said to be* ***stable.***

The fewer number of mapping cycles needed to achieve this stability, the less likely there will be potential differences in interpretation of cause values between carrier networks and potential differences in KPI's, etc.

9.1.3 Symmetric Mappings

A symmetric mapping is a one to one mapping in both directions that is inherently stable. Such mappings are preferable when possible as they don't introduce any potential ambiguity between carriers. An example of this would be:

CV 24 → SIP Response 433 → CV 24
SIP Response 433-> CV 24 -> SIP Response 433

In this case there are no intermediate transformations possible and the result is unambiguous.

9.1.4 Quality KPI preservation

It is desirable to have quality KPIs calculated in a consistent manner across carrier networks. To do so, KPI calculations have to be agreed upon and cause values/response codes need to be passed/received in a consistent manner between carrier networks. Given that KPIs are typically calculated based on cause values/response codes that distinguish between 'user' and 'network' events (as defined by ITU-T E.425), an ideal mapping will attempt to eliminate the possibility of 'user' class events being mapped back to 'network' class values (and vice versa) so that KPIs can be preserved.

Note: this is implicitly extending the notion of "user" and "network" to SIP, notions which do not exist as such in the SIP framework.

9.2 ISUP – SIP Interworking Issues

9.2.1 Inconsistencies within a given Standard Mapping

In ISUP, the disconnect cause values contained in the release message are defined in ITU standard Q.850 [7] 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 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. Depending on the mappings used at each Carrier and the specific disconnect causes involved, the end-to-end disconnect cause transparency can be degraded between the two Service Provider networks.

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.

Examples of mishandling of disconnect causes follow:

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.

9.2.2 Inconsistencies across different Standard Mappings

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

a) ITU-T Q.1912.5 before IETF RFC 3398
b) IETF RFC 3398 before ITU-T Q.1912.5

The effects of the above cases are different but both give rise to mapping inconsistencies similar to those shown in the previous section. This has the potential to seriously degrade end to end service, quality KPI measurements and trouble location and resolution within and across Carrier and Service Provider networks.

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

9.2.3 Disadvantages of the Current Mapping Schemes

Issues with the IT-T Q.1912.5 scheme

The main issue with the ITU mapping scheme is the 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 a single iteration and, exceptionally, two**.**

Issues with the 3GPP scheme (TS 29.163 V7.21.0 [6])

The main issues with versions of the 3GPP scheme older than March 2011 are the same as those for the ITU mapping scheme described above.

Issues with the IETF 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.

9.3 ISUP – SIP Interworking with Reason Header Support

As a partial solution to some of the issues mentioned above, the use of the Reason Header in accordance with IETF RFC 3326 [8] is recommended to enable the inclusion of the ISUP disconnect cause values. Providing that all the sending and receiving SIP platforms support this field, then *most* ISUP release cause value information will be preserved and may be written correctly into the ISUP function at the interworking point on the originating side.

This is only a partial solution however for the following two reasons:

9.3.1 Optional Inclusion or Interpretation of the Reason Header

According to [8], the Reason Header "*.. 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 [8], clients and servers are free to ignore this header field (for backward compatibility reasons). 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 therefore be concluded that the Reason Header's effect on the actual interoperability behaviour between nodes may differ depending on the implementation of such functionality by the respective vendors.

9.3.2 Loss of Location Information

An additional deficiency, though arguably very small, is due to the Location field information associated with an ISUP Release Cause not being carried in the header. As a result, incomplete ISUP information is passed and therefore the differentiation between locations (User and Network in particular) is lost.

Example: ISUP Release Cause Value 21 – Call Rejected
In the case of CV=21, 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 ability to take different action or even to record the difference will not be possible.

Predicted Impacts of loss of Location Information
In the above example, even though the Reason Header from RFC3326 is implemented in the SIP functions of, for instance, both Carrier A and Service Provider B (assuming the call flows from A –> B),

the location information from Service Provider B is not preserved. The significance of this to Carrier A is that it is not possible to differentiate between User and Network conditions.

Carrier A will now inevitably miss-handle some call attempts on receipt of "ISUP CV=21. It is probable that a Carrier would by default act differently based on the location/class of the event, giving one treatment in one case and the inverse treatment in the other. This could result in re-route attempts being made when they really shouldn't be (or visa versa) and can also negatively affect KPI's such as NER, which may end up being either higher or lower then what they should actually be.

9.3.3 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 DTAG. The Internet draft identity is as follows: draft-jesske-dispatch-reason-in-responses.

9.4 ISUP – SIP-I Interworking

The issues described in the previous section (9.2 and 9.3) are at the SIP level and so might also be relevant in the SIP-I – ISUP interworking scenario. In the case of SIP-I, however, 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.

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, i3 Forum recommends that the ISUP values take precedence over the SIP values.

In Annex E a more detailed discussion on the ISUP-SIP-I interworking issues is provided.

10 Resolving Disconnect Cause Value Inconsistencies

i3 Forum conducted a survey among its members between November 2009 and January 2010 to ascertain the prevalence of deployment of the two mapping schemes: ITU-T Q.1912.5 and RFC 3398. The results did not show a clear preference for either scheme.

In addition, it is noted that a number of Carriers have implemented changes to some of the mappings within a given standard inside 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 Disconnect Cause Values and Response Codes.

In late 2010 / early 2011 i3 Forum and 3GPP worked together to finalize a unique mapping capable of overcoming the previously mentioned issues. The resultant output of this activity is a new version of 3GPP 29.163, dated March 2011 which encompasses releases back to release 7 (i.e. 7.22.0[5]). In Annex B the mapping from SIP error response codes to ISUP disconnect cause values is presented, according to both [3], [4], along with the new joint i3/3GPP recommendation. Annex C details the mappings of the ISUP disconnect cause values to SIP error response codes according to the same set of specifications.

Given that ISUP and SIP are protocols for different network environments, there is no perfect mapping between the two. The new solution is therefore not optimal in every regard. However, because it was derived utilizing the guidelines presented in section 9.1 of this document, which specify the criteria for an ideal mapping, the new proposal is better able to:

1. support the alternative routing requirements of service providers in IP, similar to those in circuit switched networks;
2. reduce the use of many-to-one mappings, greatly reducing the ambiguity that arose because of this in previous proposals;
3. create symmetric mappings of ISUP-SIP and SIP-ISUP, which will allow better understanding and identification of issues within and across Carrier networks;

4. categorize the SIP response codes into User and Network event categories which are useful in the NER calculations;
5. allow for the generation of more uniform quality KPI calculations and reporting.

As a whole, these capabilities will allow the industry to achieve wider compatibility and improved Carrier /Service Provider network behaviour and related reporting. i3 Forum therefore recommends the adoption of this new mapping.

11 Recommendations to involved parties

As a consequence of the analysis carried out by i3 Forum, the following recommendations are given:

a) **It is the recommendation of i3 Forum that all Carriers and Service Providers implement the new mapping found in 3GPP 29.163 (March 2011 Version 7.22 or later [5]) in their networks as soon as is practically feasible**. This is the best approach to insure consistent, correct network behaviours and quality KPI reporting.

   When converting from SIP-I to SIP, if use of the above mapping is not possible, the Carrier must insure that the implementation converts internally to the ISUP Disconnect Cause Values prior to converting to the SIP Response Codes (however this will not necessarily fix the KPI reporting issue if the KPIs are calculated using SIP Response Codes).

b) **Industry vendors should similarly incorporate the new mapping standard at the earliest opportunity into their existing or new products to enable deployed networks to more easily adopt the new standard**.

c) Further, the use of the Reason Header field is recommended whenever possible when using native SIP.

d) Changes or updates to this agreed standard shall not be countenanced by a single standards body, but only by full agreement between ITU, 3GPP and, going forward, other stand bodies such as the IETF. Moreover, other pan-industry bodies with an interest in this area, (for example: The SIP Forum, The Multi-Service Forum) should also work collaboratively together with i3 Forum and 3GPP to ensure future versions of the mappings reflect the fullest possible dataset across the industry.

12 ANNEX A - Mapping from ISUP messages to SIP messages

Mapping specified by 3GPP TS 29.163

#	Group	ISUP MESSAGE		SIP MESSAGE
1	*Forward*	IAM	Initial address	INVITE
	set-up	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*	ANM	Answer	200 OK (INVITE)
8	*supervision*	FOT	Forward transfer	*No Equivalent*
9		REL	Release	BYE, CANCEL
10	*Circuit*	RLC	Release complete	*No Equivalent*
11	*supervision*	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

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

RFC 3398, Q.1912.5 and i3/3GPP TS29.163 V722 Mappings:

The yellow rows indicate a mismatch between the IETF RFC 3398, ITU-T Q.1912.5 and/or i3/3GPP TS29.163 V722, March 20,2011 versions

* Following a Response Code indicates that it should be viewed as a User event; Absence of the symbol indicates that it should be viewed as a Network event. This classification will affect KPI calculations.

4xx, 5xx, 6xx on INVITE		REL (Cause Value) ISUP (Follows IETF RFC 3398)		REL (Cause Value) ISUP (Follows ITU-Q.1912.5)		*REL (Cause Value) ISUP (Follows i3 & 3GPP TS29.163 V722March 20, 2011 Versions)*	
Error Response Code		Cause Value		Cause Value		**Cause Value**	
400	Bad Request	41	("Temporary Failure")	127	("Interworking unspecified")	***111***	***("protocol error, unspecified")***
401	Unauthorized	21	("Call rejected")	127	("Interworking unspecified")	***127***	***("Interworking unspecified")***
402	Payment Required	21	("Call rejected")	127	("Interworking unspecified")	***127***	***("Interworking unspecified")***
403	Forbidden	21	("Call rejected")	127	("Interworking unspecified")	***79***	***("Service option not implemented, unspecified")***
404*	Not Found	1	("unallocated (unassigned) number)	1	("unallocated (unassigned) number)	***1***	***("unallocated (unassigned) number)***
405	Method Not Allowed	63	("Service option not available, unspecified")(Class default)	127	("Interworking unspecified")	***127***	***("Interworking unspecified")***
406	Not Acceptable	79	("Service option not implemented, unspecified")	127	("Interworking unspecified")	***79***	***("Service option not implemented, unspecified")***
407	Proxy authentication required	21	("Call rejected")	127	("Interworking unspecified")	***127***	***("Interworking unspecified")***
408	Request Timeout	102	("Recover on Expires timeout")	127	("Interworking unspecified")	***102***	***("Recover on Expires timeout")***
409	Conflict	--		--		***41***	***("Temporary Failure")***
410*	Gone	22	("Number changed (without diagnostic)")	22	("Number changed (without diagnostic)")	***22***	***("Number changed (without diagnostic)")***
413	Request Entity too long	127	("Interworking unspecified")	127	("Interworking unspecified")	***127***	***("Interworking unspecified")***
414	Request-uri too long	127	("Interworking unspecified")	127	("Interworking unspecified")	***111***	***("protocol error, unspecified")***
415	Unsupported Media type	79	("Service option not implemented, unspecified	127	("Interworking unspecified")	***127***	***("Interworking unspecified")***
416	Unsupported URI scheme	127	("Interworking unspecified")	127	("Interworking unspecified")	***111***	***("protocol error, unspecified")***
420	Bad Extension	127	("Interworking unspecified")	127	("Interworking unspecified")	***111***	***("protocol error, unspecified")***
421	Extension required	127	("Interworking unspecified")	127	("Interworking unspecified")	***111***	***("protocol error, unspecified")***
423	Interval Too Brief	127	("Interworking unspecified") (Class default)	127	("Interworking unspecified")	***127***	***("Interworking unspecified")***
433*	Anonymity Disallowed	--		--		***24***	***("Anonymous Call Reject")***
480*	Temporarily Unavailable	18	("no user responding")	20	("Subscriber absent")	***20***	***("Subscriber absent")***
481	Call/Transaction does not exist	41	("Temporary Failure")	127	("Interworking unspecified")	***127***	***("Interworking unspecified")***
482	Loop Detected	25	("Exchange routing error")	127	("Interworking unspecified")	***127***	***("Interworking unspecified")***
483*	Too many hops	25	("Exchange routing error")	127	("Interworking unspecified")	***25***	***("Exchange routing error")***

484*	Address Incomplete	28	("Invalid number format (address incomplete) ")	28	("Invalid Number format(address incomplete)")	***28***	***("Invalid Number format(address incomplete)")***
485*	Ambiguous	1	("Unallocated (unassigned) number")	127	("Interworking unspecified")	***1***	***("Unallocated (unassigned) number")***
486*	Busy Here	17	("User busy")	17	("User busy")	***17***	***("User busy")***
487	Request terminated			127	Interworking or no mapping	**--**	***Not Mapped***
488*	Not acceptable here			127	("Interworking unspecified")	***50***	***("Requested facility not subscribed")***
491	Request Pending				no mapping		***No Mapping***
493	Undecipherable			127	("Interworking unspecified")		***No Mapping***
500	Server Internal error	41	("Temporary failure")	127	("Interworking unspecified")	**127**	***("Interworking unspecified")***
501	Not implemented	79	("Service or option not implemented, unspecified")	127	("Interworking unspecified")	**79**	***("Service or option not implemented, unspecified")***
502	Bad Gateway	38	("Network out of order")	127	("Interworking unspecified")	***27***	***("Destination out of order")***
503	Service Unavailable	41	("Temporary failure")	127	("Interworking unspecified")	**41**	***("Temporary failure")***
504	Server timeout	102	("Recovery on timer expiry")	127	("Interworking unspecified")	**102**	***("Recovery on timer expiry")***
505	Version not supported	127	("Interworking, unspecified")	127	("Interworking unspecified")	**127**	***("Interworking unspecified")***
513	Message too large	127	("Interworking, unspecified")	127	("Interworking unspecified")	***95***	***("Invalid message, unspecified")***
580	Precondition failure			127	("Interworking unspecified")	***127***	**("Interworking, unspecified")**
600*	Busy Everywhere	17	("User busy")	17	("User busy")	**17**	***("User busy")***
603*	Decline	21	("Call rejected")	21	("Call rejected")	**21**	***("Call rejected")***
604	Does not exist anywhere	1	("Unallocated (unassigned) number")	1	("Unallocated number")	**2**	***("No route to network")***
606*	Not acceptable			127	("Interworking unspecified")	***88***	***("Incompatible destination")***
CANCEL*						***16 or 31***	***("Normal Call Clearing or Normal, unspecified")***

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

RFC 3398, Q.1912.5 and i3/3GPP TS29.163 V722 Mappings:

The yellow rows indicate a mismatch between the IETF RFC 3398, ITU-T Q.1912.5 and/or i3/3GPP TS29.163 V722 March 20, 2011 Versions.

* Following a Disconnect Cause Value indicates that it should be viewed as a User event;
Absence of the symbol indicates that it should be viewed as a Network event.
This classification will affect KPI calculations.

REL ISUP -Cause Disconnect Values -		SIP Message (Follows IETF RFC 3398)		SIP Message (Follows ITU-Q.1912.5)		*SIP Message (Follows i3/3GPP TS29.163 V722March 20, 2011 Versions)*	
1*	*("Unallocated (unassigned) number")*	404	Not found	404	Not Found	***404***	***Not Found***
2	*("No route to network")*	404	Not found	500	Server Internal Error	***604***	***Does Not Exist Anywhere***
3	*("No route to destination")*	404	Not found	500	Server Internal Error	***604***	***Does Not Exist Anywhere***
4	*("Send special information tone")*			500	Server Internal Error	***500***	***Server Internal Error***
5	*("Misdialled trunk prefix")*			404	Not Found	***404***	***Not Found***
8	*("Preemption")*					***500***	***Server Internal Error***
9	*("Preemption – Circuit Reserved")*					***500***	***Server Internal Error***
14	*("QoR: Ported Number")*					***500***	***Server Internal Error***
16*	*("Normal Call Clearning")*						***BYE (cause carried in Reason Header)***
17*	*("User busy")*	486	Busy here	486	Busy Here	***486***	***Busy Here***
18*	*("No user response")*	408	Request Timeout	480	Temporarily unavailable	***480***	***Temporarily unavailable***
19*	*("No answer from the user")*	480	Temporarily unavailable	480	Temporarily unavailable	***480***	***Temporarily unavailable***
20*	*("Subscriber absent")*	480	Temporarily unavailable	480	Temporarily unavailable	***480***	***Temporarily unavailable***
21*	*("Call rejected")*	403	Forbidden	480	Temporarily unavailable	***603 or 403***	***603 Decline for user response. 403 Forbidden for network response.***
22*	*("Number changed")*	410 301	Gone or Moved Permanently	410	Gone	***410***	***Gone***
23*	*("Redirection to new destination")*	410	Gone		No interwork	***410***	***Gone***
24*	*("Anonymous Call Reject")*					***433***	***Anonymity Disallowed***
25*	*("Exchange routing error")*			480	Temporarily unavailable	***483***	***Too Many Hops***
26*	*("Non-selected user clearing")*	404	Not found			***404 or 480***	***404 Not Found for North America 480 Temporarily unavailable for Rest of World***
27	*("Destination out of order")*	502	Bad Gateway	502	Bad Gateway	***502***	***Bad Gateway***
28*	*("Invalid number format (address incomplete)"*	484	Address incomplete	484	Address Incomplete	***484***	***Address Incomplete***
29	*("Facility rejected")*	501	Not implemented	500	Server Internal Error	***501***	***Not implemented***
31*	*("Normal, unspecified") (Class default)*	480	Temporarily unavailable	480	Temporarily unavailable		***BYE (cause carried in Reason Header)***

Cause Value in the Class 010 (resource unavailable Cause Value No. 34)		503	Service unavailable	486 / 480	486 Busy here if Diagnostics Indicator includes the CCBS indicator; otherwise mapping to 480	***503***	***503 Service unavailable***
38	*("Network out of order")*	503	Service unavailable	500	Server Internal Error	***500***	***Server Internal Error***
41	*("Temporary failure")*	503	Service unavailable	500	Server Internal Error	***503***	***Service unavailable***
42	*("Switching equipment congestion")*	503	Service unavailable	500	Server Internal Error	***503***	***Service unavailable***
43	*(“Access information discarded”)*					***500***	***Server Internal Error***
44	*("Requested circuit/channel not available")*			500	Server Internal Error	***503***	***Service unavailable***
46	*("Precedence call blocked")*			500	Server Internal Error	***500***	***Server Internal Error***
47	*("Resource unavailable, unspecified")*	503	Service unavailable	500	Server Internal Error	***503***	***Service unavailable***
50*	*("Requested facility not subscribed")*			500	Server Internal Error	***488***	***Not acceptable here***
53*	*Outgoing cals barred within CUG*					***603***	***Decline***
55*	*("Incoming class barred within Closed User Group (CUG)")*	403	Forbidden	500	Server Internal Error	***603***	***Decline***
57*	*("Bearer capability not authorized")*	403	Forbidden	500	Server Internal Error	***603***	***Decline***
58	*("Bearer capability not presently available")*	503	Service unavailable	500	Server Internal Error	***503***	***Service unavailable***
63	*("Service option not available, unspecified") (Class default)*			500	Server Internal Error	***501***	***Not implemented***
65	*("Bearer capability not implemented")*	488	Not acceptable here	500	Server Internal Error	***500***	***Server Internal Error***
66				500	Server Internal Error		***Not Mapped***
69	*(Requested Facility not implemented)*			500	Server Internal Error	***501***	***Not implemented***
70		488	Not acceptable here	500	Server Internal Error	***501***	***Not implemented***
79	*("Service option not available, unspecified")*	501	Not implemented	500	Server Internal Error	***501***	***Not implemented***
87*	*("User not member of Closed User Group(CUG)")*	403	Forbidden	500	Server Internal Error	***403***	***Forbidden***
88*	*("Incompatible destination")*	503	Service unavailable	500	Server Internal Error	***606***	***Not Acceptable***
90*	*("Non existent CUG")*			500	Server Internal Error	***403***	***Forbidden***
91	*("Invalid transit network")*			404	Not Found	***500***	***Server Internal Error***
95	*("Invalid message (Class default)")*			500	Server Internal Error	***513***	***Message Too Large***
97	*("Message type non-existent or not*			500	Server Internal Error	***501***	***Not implemented***

	implemented")						
98	*Message not compatible with call state or not implemented*					***501***	***Not implemented***
99	*("Information element/parameter non-existent or not implemented")*			500	Server Internal Error	***501***	***Not implemented***
102	*("Recover on Expires timeout")*	504	Server Timeout	480	Temporarily unavailable	***504***	***Server Timeout***
103	*("Parameter non-existent or not implemented, passed on")*			500	Server Internal Error	***501***	***Not implemented***
110	*("Message with unrecognized parameter, discarded")*			500	Server Internal Error	***501***	***Not implemented***
111	*("Protocol error unspecified") (Class default)*	500	Internal Server Error	500	Server Internal Error	***400***	***Bad Request***
127	*("Interworking, unspecified") (Class default)*	500	Internal Server Error	480	Temporarily unavailable	***500***	***Server Internal Error***

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

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

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.

Lastly, from March 2011 onwards, the mapping of ISUP cause values onto SIP Response codes has been enhanced.

16 ANNEX E – Examples of issues with ISUP – SIP-I interworking

In a scenario where an ITU-T complaint [3] carrier is in an end-to-end call path together with multiple carriers and/or service providers, the interworking between SIP-I and other interconnect types may cause an inconsistency in Cause Value/Response Code mappings across these providers. Consequently, incorrect operation and failure reason reporting could occur in the same way described above for native SIP.

In particular, there are issues that arise in scenarios where a Carrier or Service Provider is using Q.1912.5 to convert from ISUP to SIP-I and then subsequent Carrier(s) use 3GPP 29.163 SIP to perform further conversions (independent of the version of 29.163 that is used).

As shown earlier in the document, there are several classes of issue that can arise with mapping conversions. Here we show two classes of issue resulting from two different CV mappings. Other cases exist but are not shown. In the figure that follows, the red box delineates where the problems can arise across Carrier networks and the red highlighted numeric values highlight a few of the Response codes/Disconnect Cause Value combinations that give rise to the issues.

The first issue involves potentially inconsistent reporting of quality KPI's across carriers' networks. Figure 3 depicts two scenarios. The first considers a TDM origination to TDM termination call path and assumes Carrier B utilizes SIP Response Codes for quality KPI calculations

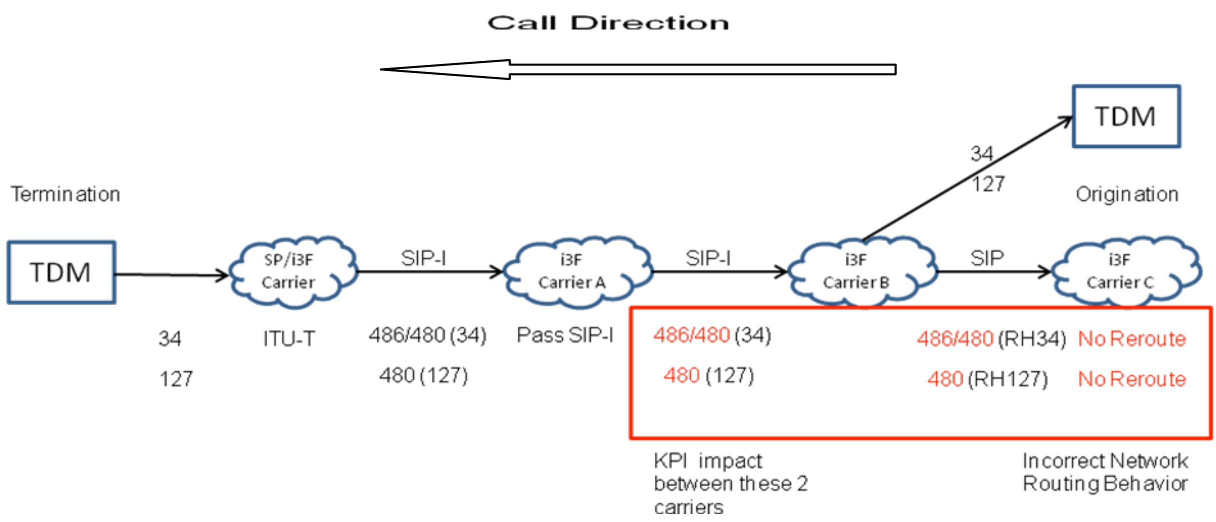


Figure 3 – Examples of Potential SIP-I Interworking Issues

In this scenario the conversion at the terminating Carrier is performed by using ITU-T Q1912.5 to convert from ISUP to SIP-I. The SIP-I Disconnect Cause Values then propagate back through Carrier A and B networks to reach the TDM originator who is connected via ISUP. The correct Disconnect Cause Values are tunnelled intact across the networks of Carriers A and B. However, the SIP Response Codes are inconsistent with the tunnelled values and could cause incorrect KPI reporting, particularly those that classify User vs. Network failures differently. This can be avoided if the KPIs are calculated only based on the Disconnect Cause Values but there are implementations that may only look at the SIP Response Codes (by design or by error).

Now consider the second scenario, depicted by the SIP origination to TDM termination call path in Figure 3. Here assume that both Carrier B and C utilize SIP Response Codes for quality KPI calculations and routing purposes. This scenario is similar to the first example except that the SIP-I Disconnect Cause Values now propagate back through the carriers' networks to reach the originator (i.e. Carrier C) who is connected via SIP, necessitating a SIP-I to SIP conversion. In case the conversion takes place without first internally mapping the ISUP Disconnect Cause Values to the 3GPP 29.163 V7.22.0 defined SIP Response Codes, incorrect routing behaviours may occur when basing these operations on the SIP Response Codes. If, however, the initial mapping at the terminating Carrier had been performed with 3GPP 29.163 V.7.22, then the last conversion to SIP would not have caused any issue even if the mapping back to ISUP did not take place and would allow for the routing and KPI calculations to simply be based on the SIP Response Codes.