INTERNATIONAL INTERCONNECTION FORUM
FOR SERVICES OVER IP

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Workstream "Technical Aspects"

White Paper

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

(Release1.0, May 2009)

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 expected that further work will be required to provide greater clarity in the area of signalling interworking and, as a consequence, as soon as new standards are available i3 Forum will be ready to endorse them and enhance the interconnection model document.

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1 Scope and Objective

This document addresses signalling interworking issues when converting form 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 2, May 2009.

2 Acronyms

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 2.0, May 2009
[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 Signaling 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.

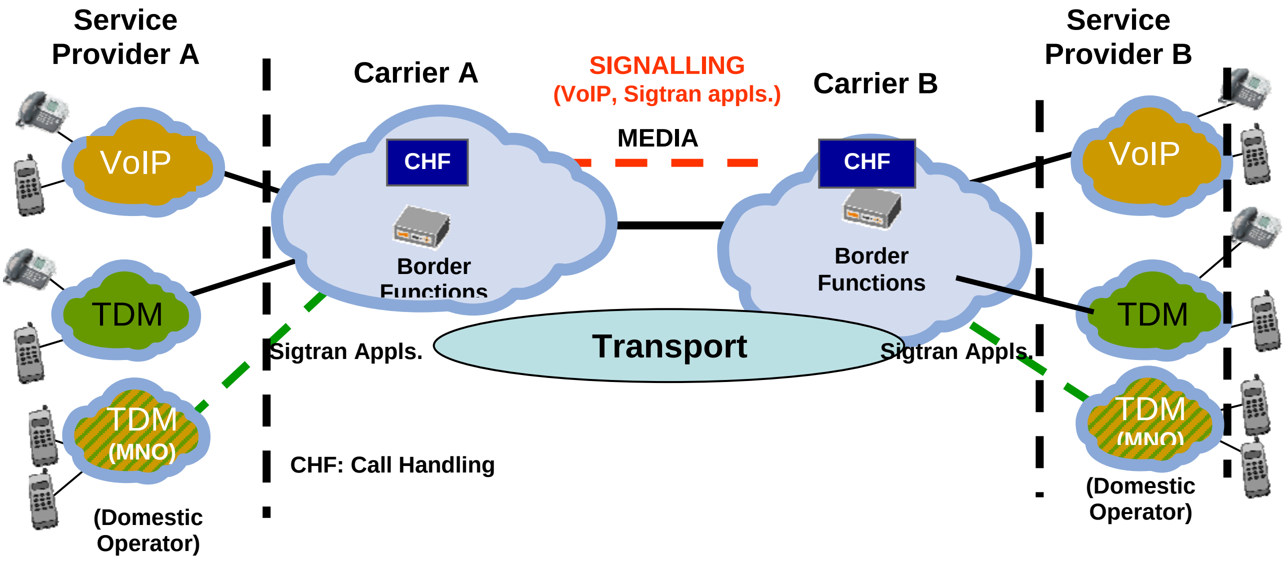


Figure 1 – General Reference Configuration

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 Rec. 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 interworked, 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:

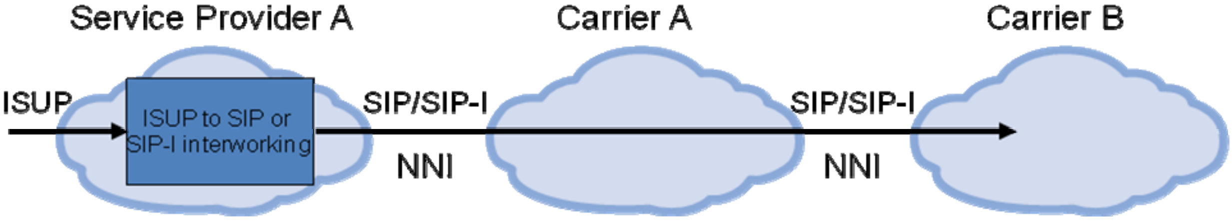
- Loss of end-to-end service information needed to support services
- Automatic re-routing causes used by some carriers, based on one or several specific disconnect causes. Typically, cause value 34 is used to reroute
- Accounting interchanged between clients and carriers based on cause values written 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 2 below), mapping would therefore be the responsibility of the Service Provider.;

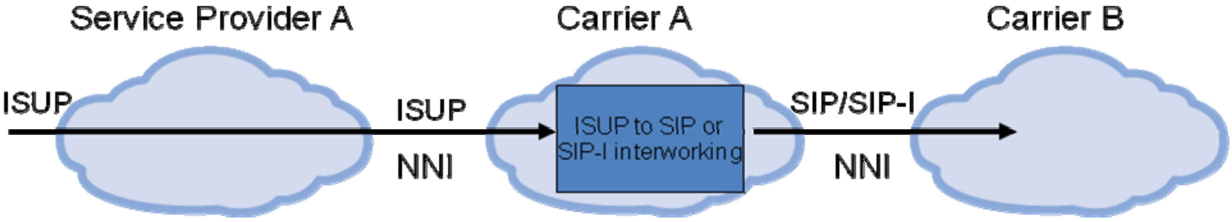
2 – Interworking is performed by one of the carriers (see figure 3 below).

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 2- Interworking function locations, scenario 1



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

Figure 3- Interworking function locations, scenario 2

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 1, 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.

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 equivalent to ISUP disconnect cause values are error response codes 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 2 the mapping from SIP error response codes to ISUP disconnect cause values, according to both [3] and [4] is given. Annex 3 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 possible solution, the use of the Reason header in accordance with IETF RFC 3326 [7] is recommended to enable the inclusion of the ISUP disconnect cause codes.

According to [7], "*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 rerouting 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.

9.2 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, 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.

When there is a difference of significance between the SIP Error 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 Conclusions and Recommendations

As far as message mapping is concerned, the analysis indicates that a complete an 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 but the usage of such mechanism should be required and vendors have to design equipment accordingly.

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.

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 expected that further work will be required to provide greater clarity in the area of signalling interworking and, as a consequence, as soon as new standards are available i3 Forum will be ready to endorse them and enhance the interconnection model document [1].

11 ANNEX 1 "Mapping from ISUP messages to SIP messages

#	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

12 ANNEX 2 "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")*

13 ANNEX 3 "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