Telenor Networks

Test descriptions
for Signalling System No.7
national interconnect
between Telenor Networks
and another telecom operator
Telenor Norwegian national interconnect
ISUP v2:
ISDN-SIP end-to-end

SIP-SS7 Gateway/SIP – ISUP signalling

IP telephony interworking with PSTN/ISDN

TABLE OF CONTENTS

1. INTRODUCTION

1.1 Purpose and scope

The standard Telenor national interconnect agreement specifies interconnect tests for the transit ISUP case, the end-to-end case for ISDN-ISDN, ISDN-analogue, ISDN-GSM. However, testing when one end is terminated on an IP telephony gateway has not previously been described. Until 2002 this has not been a problem, because IP telephony (e.g. H.323 and SIP) has usually been connected to the PSTN/ISDN behind an ISDN access. However, from now on SIP-ISUP and H.323-ISUP gateways are expected to be introduced in the network in a more significant scale, and the need arises for test specifications to be used for end-to-end interconnect test with TELENOR when such gateways are part of the configuration.

This document specifies a set of test cases for national interconnect test with TELENOR when TELECOM OPERATOR terminates the calls on a SIP-ISUP test gateway. A detailed description of each test case is given. This includes the purpose of the test case, the test environment, the test procedures and the expected results. In the test descriptions, the “ISUP simulator” shall be substituted with the TELENOR POI when the test is performed as an interconnect test. TELENOR will actually use an ISUP or DSS1 simulator when testing, but the simulator will be located behind the POI.

Each test case has a corresponding entry in the Test log.

This document is a first version, and will be updated based on experience.

The SIP implementation is expected to be according to IETF RFC 2543.

The protocol interworking requirements on both the SIP side and the ISUP side are decided by the network configuration and the SS7 interface protocol requirements. The network configuration to be tested will be the SIP-ISUP gateway as a gateway between a SIP IP telephony application on the public Internet and the Norwegian national PSTN/ISDN network on the national interconnect interface. This means that the protocol requirements for the interworking shall be according to the Telenor Norwegian national interconnect interface.

2. TEST CASE SPECIFICATIONS

2.1 Call from SS7 (ISUP) to SIP access

2.1.1 Circuit supervision, non-allocated circuits

<table>
<tr><td>Test case ID:</td><td colspan="3">SIP VoIP-ISUP-01</td></tr>
<tr><td>Test object:</td><td colspan="3">Circuit supervision for ISUP signalling, non-allocated circuits.</td></tr>
<tr><td>Test purpose:</td><td colspan="3">The purpose of this test case is to verify that on receipt of a CIC relating to a circuit which does not exist, the SIP Gateway will discard the message and alert the maintenance system.
See Q.784 test no. 1.1.</td></tr>
<tr><td>Test case dependencies:</td><td colspan="3">Infrastructure in place.</td></tr>
<tr><td>Test Set-up:</td><td colspan="3">ISUP simulator towards the SIP/SS7 GW.
Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.

<u>Pre-test conditions:</u>
Arrange the data in the ISUP simulator such that the CIC identifies a circuit that does not exist between the SIP GW and the ISUP simulator.</td></tr>
<tr><td>Post-test actions:</td><td colspan="3">Analyse the signalling procedures. Verify that they are correct according to
• RFC 2543(SIP)
• Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification)
• The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.</td></tr>
<tr><td colspan="2">Test Procedure Steps:</td><td colspan="2">Expected Results:</td></tr>
<tr><td>1.</td><td>ISUP simulator calls SIP access</td><td>ISDN:</td><td>Q.764 IAM</td></tr>
<tr><td>2.</td><td></td><td>SIP GW:</td><td>The SIP GW discards the message and alerts the maintenance system.</td></tr>
<tr><td>3.</td><td></td><td></td><td></td></tr>
<tr><td>4.</td><td></td><td></td><td></td></tr>
<tr><td>5.</td><td></td><td></td><td></td></tr>
<tr><td>6.</td><td></td><td></td><td></td></tr>
</table>

2.1.2 Circuit supervision, reset of circuits, RSC received on an idle circuit

Test case ID:	SIP VoIP-ISUP-02
Test object:	Reset of circuits for ISUP signalling, RSC received on an idle circuit.
Test purpose:	The purpose of this test case is to verify that on receipt of a reset circuit message from SS7 (ISUP), the SIP Gateway will respond by sending a release complete. See Q.784 test no. 1.2.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test condition: The circuit is idle.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
7.	ISUP simulator sends a reset-circuit message to the SIP GW	ISDN:	Q.764 RSC
8.	The SIP GW disconnects	SIP GW:	Q.764 RLC The circuit shall be idle.
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2.1.3 Circuit supervision, reset of circuits, RSC received on a remotely blocked circuit

Test case ID:	SIP VoIP-ISUP-03
Test object:	Reset of circuits for ISUP signalling, RSC received on a remotely blocked circuit.
Test purpose:	The purpose of this test case is to verify that the SIP Gateway is able to react to a reset-circuit message from SS7 (ISUP) for a remotely blocked circuit. See Q.784 test no. 1.2.4.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test condition: The circuit is idle.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
13	ISUP simulator sends a blocking message to the SIP GW	ISDN:	Q.764 BLO
14	SIP GW responds with a blocking acknowledge message	SIP GW:	Q.764 BLA
15	ISUP simulator sends a reset circuit message to the SIP GW	ISDN:	Q.764 RSC
16	SIP GW disconnects	SIP GW:	Q.764 RLC The circuit shall be idle
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2.1.4 Circuit supervision, reset of circuits, circuit group reset received

Test case ID:	SIP VoIP-ISUP-04
Test object:	Reset of circuits for ISUP signalling, circuit group reset received.
Test purpose:	The purpose of this test case is to verify that on receipt of one circuit group message from SS7 (ISUP), the SIP Gateway will respond by sending a circuit group reset acknowledge message. See Q.784 test no. 1.2.5.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test condition:</u> All circuits are idle.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 "SIP-ISUP interworking for Telenor national interconnect ISUP version 2".

Test Procedure Steps:		**Expected Results:**	
19	ISUP simulator sends circuit group reset message to the SIP GW	ISDN:	Q.764 GRS
20	SIP GW responds with a circuit group reset acknowledge message	SIP GW:	Q.764 GRA • The circuits shall be idle. • Are the status bits in GRA set correctly? • If Range=0, GRS is discarded and GRA is not sent. • If Range>31, GRS is discarded and GRA is not sent.
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2.1.5 Circuit supervision, reset of circuits, circuit group reset received on remotely blocked circuits

Test case ID:	SIP VoIP-ISUP-05
Test object:	Reset of circuits for ISUP signalling, circuit group reset received on remotely blocked circuits.
Test purpose:	The purpose of this test case is to verify that the SIP Gateway is able to react correctly to a circuit group reset message from SS7 (ISUP) for remotely blocked circuits. See Q.784 test no. 1.2.7.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test condition:</u> All circuits are idle.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
25	ISUP simulator sends a blocking message including the blocked circuit x to the SIP GW	ISDN:	Q.764 BLO (CIC=x)
26	SIP GW responds with a blocking acknowledge	SIP GW:	Q.764 BLA
27	ISUP simulator sends a blocking message including the blocked circuit y to the SIP GW	ISDN:	Q.764 BLO (CIC=y)
28	SIP GW responds with a blocking acknowledge	SIP GW:	Q.764 BLA
29	ISUP simulator sends a circuit group reset message including the blocked circuits x and y to the SIP GW	ISDN:	Q.764 GRS (including CIC= x, y)
30	SIP GW responds with a circuit group reset acknowledge message	SIP GW:	Q.764 GRA The circuits shall be idle.

2.1.6 Circuit supervision, circuit group blocking/unblocking, CGB and CGU received

Test case ID:	SIP VoIP-ISUP-06
Test object:	Circuit group blocking/unblocking for ISUP signalling, CGB and CGU received.
Test purpose:	The purpose of this test case is to verify that when the circuit group blocking feature is initiated from SS7 (ISUP), the correct response is provided from the SIP GW to the SS7 (ISUP). See Q.784 test no. 1.3.1.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test condition:</u> All circuits are idle.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
31	ISUP simulator sends a circuit group blocking message with the circuit group supervision message type indicator set to “maintenance oriented” to the SIP GW.	ISDN:	Q.764 CGB
32	SIP GW responds with a circuit group blocking acknowledge message	SIP GW:	Q.764 CGBA Verify that a call cannot be originated from the SIP GW on the circuits indicated by the range and status parameter in the CGB message.
33	ISUP simulator sends one circuit group unblocking message with circuit group supervision message type set to “maintenance oriented” to the SIP GW.	ISDN:	Q.764 CGU

34	SIP GW responds with a circuit group unblocking acknowledge message	SIP GW:	Q.764 CGUA • Verify that a call can be originated from ISUP simulator or SIP GW on the circuits indicated by the range field. • If RANGE=0, CGB is discarded and CGBA is not sent • If RANGE>31, CGB is discarded and CGBA is not sent
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2.1.7 Circuit supervision, circuit group blocking/unblocking, BLO received

Test case ID:	SIP VoIP-ISUP-07
Test object:	Circuit group blocking/unblocking for ISUP signalling, BLO received.
Test purpose:	The purpose of this test case is to verify that when the blocking/unblocking procedure is initiated from SS7 (ISUP), the correct response is provided from the SIP GW to the SS7 (ISUP). See Q.784 test no. 1.3.2.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test condition:</u> The circuit is idle.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
36	ISUP simulator sends a blocking message to the SIP GW.	ISDN:	Q.764 BLO
37	SIP GW responds with a blocking acknowledge message	SIP GW:	Q.764 BLA Verify that a call cannot be originated from the SIP GW on this circuit.
38	ISUP simulator sends an unblocking message to the SIP GW.	ISDN:	Q.764 UBL
39	SIP GW responds with an unblocking acknowledge message	SIP GW:	Q.764 UBA Verify that a call can be originated from the ISUP simulator or the SIP GW on this circuit.
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2.1.8 Circuit supervision, circuit group blocking/unblocking, IAM received on a remotely blocked circuit

Test case ID:	SIP VoIP-ISUP-08
Test object:	Circuit group blocking/unblocking for ISUP signalling, IAM received on a remotely blocked circuit.
Test purpose:	The purpose of this test case is to verify that an IAM from SS7 (ISUP) will unblock a remotely blocked circuit, and that the correct response is provided from SIP to the SS7 (ISUP). See Q.784 test no. 1.3.2.4.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test condition: The circuit is idle.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
41	ISUP simulator sends a blocking message to the SIP GW.	ISDN:	Q.764 BLO
42	SIP GW responds with a blocking acknowledge message	SIP GW:	Q.764 BLA Verify that a call cannot be originated from the SIP GW on this circuit.
43	ISUP simulator calls the SIP access.	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 100 (Trying)

44	The SIP terminal is ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK SIP 200 OK (PRACK confirm) Verify that the call is processed normally at the SIP GW and the blocking status for this circuit is removed at the SIP GW.
45	The SIP user answer the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Connection is established
46	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the connection is idle.

2.1.9 Circuit supervision, receipt of unreasonable signalling information messages, receipt of unexpected messages

Test case ID:	SIP VoIP-ISUP-09
Test object:	Receipt of unreasonable signalling information messages for ISUP signalling, receipt of unexpected messages.
Test purpose:	The purpose of this test case is to verify that the action taken by a signalling point upon receipt of unexpected messages from SS7 (ISUP) is as stated in Q.764 Section 2.10.5.1, and that the correct response is provided from the SIP GW to the SS7 (ISUP). See Q.784 test no. 1.5.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: • Arrange the data in the ISUP simulator such that REL, RLC and other unreasonable messages may be initiated. • The circuit shall be idle and unblocked.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
47	a) ISUP simulator sends a release message to the SIP GW	ISDN:	Q.764 REL
48	a) SIP GW disconnects	SIP GW:	Q.764 RLC Verify that the circuit is idle.
49	b) ISUP simulator sends a release complete message to the SIP GW (No actions are taken at the SIP GW)		Q.764 RLC Verify that the circuit is idle.
50	c) ISUP simulator sends an unreasonable message XXX to the SIP GW		Q.764 XXX

51	c) SIP GW responds with a reset circuit message, or discards the message		Q.764 RSC Verify that the circuit is idle.

2.1.10 Circuit supervision, receipt of unreasonable signalling information messages, receipt of unexpected messages during call setup

Test case ID:	SIP VoIP-ISUP-10
Test object:	Receipt of unreasonable signalling information messages for ISUP signalling, receipt of unexpected messages during call setup.
Test purpose:	The purpose of this test case is to verify that the action taken by a signalling point upon receipt of unexpected messages from SS7 (ISUP) is as stated in Q.764 Section 2.10.5.1, and that the correct response is provided from SIP to the SS7 (ISUP). See Q.784 test no. 1.5.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: • Arrange the data in the ISUP simulator such that other unreasonable messages may be initiated. • The circuit shall be idle and unblocked.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
52	ISUP simulator calls the SIP access	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 100 (Trying)
53	ISUP simulator sends an unreasonable message YYY immediately after IAM YYY – Messages other than the call control messages	ISDN:	Q.764 YYY

54	The SIP GW sends a reset circuit message	SIP GW: SIP GW: SIP User: SIP User: SIP GW:	Q.764 RSC CANCEL SIP 200 OK (CANCEL confirm) SIP 487 (Call Cancelled) SIP ACK
55	ISUP simulator sends release complete	ISDN:	Q.764 RLC Verify that the circuit is idle.
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2.1.11 Bearer service speech, called address sending, en block operation

Test case ID:	SIP VoIP-ISUP-11
Test object:	Bearer service speech for ISUP signalling, called address sending, en block operation.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with en bloc operation can be provided for a call from SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.784 test no. 2.2.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test conditions:</u> • Called termination is free • The exchange data is arranged such that all digits are included in the IAM.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
57	ISUP simulator calls SIP user	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 100 (Trying)

58	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM (if the response is not 180 the ACM will carry a ”called party status” value of ”no indication”.) SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
59	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established
60	SIP user disconnects	SIP: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the connection is idle.
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2.1.12 Bearer service speech, called address sending, overlap operation

Test case ID:	SIP VoIP-ISUP-12
Test object:	Bearer service speech for ISUP signalling, called address sending, overlap operation.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with overlap operation can be provided for a call from SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.784 test no. 2.2.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: • Called termination is free • The exchange data is arranged such that digits are generated in an IAM followed by a SAM.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 "SIP-ISUP interworking for Telenor national interconnect ISUP version 2".

Test Procedure Steps:		**Expected Results:**	
63	ISUP simulator calls SIP user	ISDN: SIP GW: SIP User:	Q.764 IAM followed by SAM SIP INVITE Request SIP 100 (Trying) The SIP GW has implemented timers to insure that all digits have been collected before an INVITE is transmitted to the SIP network.

64	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM (if the response is not 180 the ACM will carry a ”called party status” value of ”no indication”.) SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
65	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established
66	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the connection is idle.
67			
68			

2.1.13 Bearer service speech, called address sending, with HLC

Test case ID:	SIP VoIP-ISUP-13
Test object:	Bearer service speech for ISUP signalling, with HLC.
Test purpose:	The purpose of this test case is to verify that the speech bearer service can be provided for a call from SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP).
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
69	ISUP simulator calls SIP user	ISDN: SIP GW: SIP User:	Q.764 IAM (TMR=speech, USI=speech, ATP: HLC=Telephony) SIP INVITE Request SDP=G.711, G.723.1, (G.728, G.729) For more details, see the attached doc 1. SIP 100 (Trying)

70	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM (if the response is not 180 the ACM will carry a "called party status" value of "no indication".) SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
71	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established
72	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the connection is idle.
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2.1.14 Bearer service speech, successful call setup, ordinary call with various indications in alerting (ACM)

Test case ID:	SIP VoIP-ISUP-14
Test object:	Bearer service speech for ISUP signalling, successful call setup, ordinary call with various indications in alerting.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with various indications in alerting can be provided for a call from SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.784 test no. 2.3.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test condition:</u> The called termination is free
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
75	ISUP simulator calls SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 100 (Trying)

76	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	a) SIP 180 (Ringing) b) SIP 181 (Call is being forwarded) c) SIP 182 (Queued) d) SIP 183 (Session progress) Q.764 ACM (a) For coding, ref. attached doc 1. b) For coding, ref. attached doc 1. c) For coding, ref. attached doc 1. d) For coding, ref. attached doc 1.) SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
77	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established
78	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the connection is idle.
79			
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2.1.15 Bearer service speech, successful call setup, ordinary call with call progress message

Test case ID:	SIP VoIP-ISUP-15
Test object:	Bearer service speech for ISUP signalling, successful call setup, ordinary call with call progress message.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with progress message can be provided for a call from SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.784 test no. 2.3.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Called termination is free. Note: There are 5 test cases described below: a) SIP 180 (mapped to ACM) - SIP 181 (mapped to CPG) b) SIP 180 (mapped to ACM) – SIP 183 (mapped to CPG) c) SIP 181 (mapped to ACM) – SIP 180 (mapped to CPG) d) SIP 181(mapped to ACM) – SIP 182(mapped to CPG) – SIP 180 (mapped to CPG) e) SIP 182 (mapped to ACM) – SIP 180 (mapped to CPG) f) SIP 183 (mapped to ACM) – SIP 180 (mapped to CPG)
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
81	ISUP simulator calls SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 100 (Trying)

82	Address complete	SIP User:	a) SIP 180 (Ringing) b) SIP 180 (Ringing) c) SIP 181 (Call is being forwarded) d) SIP 181 (Call is being forwarded) e) SIP 182 (Queued) f) SIP 183 (Session Progress)
		SIP GW:	Q.764 ACM a) For coding, ref. attached doc 1 b) For coding, ref. attached doc 1 c) For coding, ref. attached doc 1 d) For coding, ref. attached doc 1 e) For coding, ref. attached doc 1 f) For coding, ref. attached doc 1
		SIP GW:	SIP PRACK
		SIP User:	SIP 200 OK (PRACK confirm)
83	SIP terminal ringing	SIP User:	a) SIP 181(Call is being forwarded) b) SIP 183 (Session progress) c) SIP 180 (Ringing) d) SIP 182 (Queued) e) SIP 180 (Ringing) f) SIP 180 (Ringing)
		SIP GW:	Q.764 CPG a) For coding, ref. attached doc 1 b) For coding, ref. attached doc 1 c) For coding, ref. attached doc 1 d) For coding, ref. attached doc 1 e) For coding, ref. attached doc 1 f) For coding, ref. attached doc 1
		SIP GW:	SIP PRACK
		SIP User:	SIP 200 OK (PRACK confirm)

84	SIP terminal ringing (d)	SIP User: SIP GW: SIP GW: SIP User:	d) SIP 180 (Ringing) Q.764 CPG d) For coding, ref. attached doc 1 SIP PRACK SIP 200 OK (PRACK confirm)
85	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established
86	SIP User disconnects	SIP: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the connection is idle.
87			

2.1.16 Bearer service speech, successful call setup, ordinary call with various indications in the connect message (CON)

Test case ID:	SIP VoIP-ISUP-16
Test object:	Bearer service speech for ISUP signalling, successful call setup, ordinary call with various indications in the connect message.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with various indications in the connect message can be provided for a call from SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.784 test no. 2.3.3.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test conditions:</u> Called termination is free. A connect message is returned instead of an answer message from the ISUP simulator.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 "SIP-ISUP interworking for Telenor national interconnect ISUP version 2".

Test Procedure Steps:		**Expected Results:**	
88	ISUP simulator calls SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 200 OK (INVITE confirm)

89	SIP User answers the call	SIP GW: SIP GW:	Q.764 CON Comment: The SIP 200 OK can also be mapped to ACM + ANM in the SIP GW. SIP ACK Verify that the connection is established.
90	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the connection is idle.
91			
92			
93			

2.1.17 Bearer service speech, successful call setup, blocking and unblocking during a call (initiated)

Test case ID:	SIP VoIP-ISUP-17
Test object:	Bearer service speech for ISUP signalling, successful call setup, blocking and unblocking during a call (initiated).
Test purpose:	The purpose of this test case is to verify that the speech bearer service with the circuit blocking and unblocking procedure can be provided for a call from SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.784 test no. 2.3.6.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test conditions:</u> Called termination is free.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
94	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 100 (Trying)
95	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)

9(	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established
97	ISUP simulator initiates a circuit blocking message relating to the circuit used for this call	ISDN:	Q.764 BLO
9ε	The SIP GW responds with a circuit blocking acknowledge message	SIP GW:	Q.764 BLA Verify that the connection still is established
9ς	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the connection is idle. Verify that a call cannot be originated on this circuit by the SIP User.
1(	ISUP simulator sends an unblocking signal	ISDN:	Q.764 UBL
1(	The SIP GW responds with an unblocking acknowledge message	SIP GW:	Q.764 UBA Verify that a call can be successfully originated from either signalling points.
1(			

2.1.18 Bearer service speech, successful call setup, blocking and unblocking during a call (received)

Test case ID:	SIP VoIP-ISUP-18
Test object:	Bearer service speech for ISUP signalling, successful call setup, blocking and unblocking during a call (received).
Test purpose:	The purpose of this test case is to verify that the speech bearer service with the circuit blocking and unblocking procedure can be provided for a call from SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.784 test no. 2.3.7.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Called termination is free.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
1(	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 100 (Trying)

1(	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
1(	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established
1(	SIP GW initiates a circuit blocking message relating to the circuit used for this call	SIP GW:	Q.764 BLO
1(	ISUP simulator responds with a circuit blocking acknowledge message	ISDN:	Q.764 BLA Verify that the connection still is established
1(	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the connection is idle. Verify that a call cannot be originated on this circuit by the ISUP simulator.
1(	SIP GW sends an unblocking signal	SIP GW:	Q.764 UBL
11	ISUP simulator responds with an unblocking acknowledge message	ISDN:	Q.764 UBA Verify that a call can be successfully originated from either signalling points.
11			

2.1.19 Bearer service speech, normal call release, calling party clears before any backward messages (ACM)

Test case ID:	SIP VoIP-ISUP-19
Test object:	Bearer service speech for ISUP signalling, normal call release, calling party clears before any backward messages.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with the calling party releases a call prior to receipt of any backward message, can be provided for a call from SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.784 test no.3.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: The circuit is idle.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
11	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 100 (Trying)
11	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User: SIP User:	Q.764 REL Q.764 RLC SIP CANCEL SIP 200 OK (CANCEL confirm) SIP 487 (Call cancelled) Verify that the connection is idle.
11			

2.1.20 Bearer service speech, normal call release, calling party clears before answer

Test case ID:	SIP VoIP-ISUP-20
Test object:	Bearer service speech for ISUP signalling, normal call release, calling party clears before answer.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with the calling party clears before answer, can be provided for a call from SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.784 test no.3.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test conditions:</u> The called termination is free.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
11	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 100 (Trying)
11	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)

11	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User: SIP User:	Q.764 REL Q.764 RLC SIP CANCEL SIP 200 OK (BYE confirm) SIP 487 (Call cancelled) Verify that the circuit is idle
11			

2.1.21 Bearer service speech, normal call release, calling party clears after answer

Test case ID:	SIP VoIP-ISUP-21
Test object:	Bearer service speech for ISUP signalling, normal call release, calling party clears after answer.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with the calling party clears after answer, can be provided for a call from SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.784 test no.3.3.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test conditions:</u> The called termination is free.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
11	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 100 (Trying)
12	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)

12	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established
12	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the connection is idle.
12			
12			
12			
12			
12			

2.1.22 Bearer service speech, normal call release, called party clears after answer

Test case ID:	SIP VoIP-ISUP-22
Test object:	Bearer service speech for ISUP signalling, normal call release, called party clears after answer.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with the called party clears after answer, can be provided for a call from SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.784 test no.3.4.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: The called termination is free.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
12	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 100 (Trying)
12	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)

1:	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established
1:	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the connection is idle.
1:			
1:			
1:			
1:			
1:			

2.1.23 Bearer service speech, normal call release, suspend initiated by the network

Note: A possible procedure to map the ISUP signalling procedure to SIP is described in draft RFC. However this procedure is not considered necessary.

2.1.24 Bearer service speech, unsuccessful call setup, validate a set of known causes for release

Test case ID:	SIP VoIP-ISUP-24
Test object:	Bearer service speech for ISUP signalling, unsuccessful call setup, validate a set of known causes for release.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with a set of known causes for release, can be provided for a call from SS7 (ISUP) to the SIP User, and that correct response is provided to the SS7 (ISUP). See Q.784 test no.4.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Arrange the data in the SIP terminal such that a release message with a given cause is returned to the ISUP request.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
13	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 100 (Trying)

13	SIP User disconnects (cause=xxx)	SIP User: SIP GW: SIP GW:	SIP a) ex. 404 (Not found) b) ex. 480 (Temporarily unavailable) c) ex. 485 (Ambigious) d) ex. 486 (Busy here) e) ex. 484 (Address incomplete) f) ex. 501 (Not implemented) g) ex. 603 (Decline) SIP ACK Q.764 REL a) cause no.1 (unallocated nbr) b) cause no.18 (no user responding) c) cause no.1 (unallocated nbr) d) cause no.17 (User busy) e) cause no.28 (Invalid number format) f) cause no.38 (Network out of order) g) cause no.21 (Call rejected) The total mapping of SIP responses / ISUP cause values is included in the section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.
13	ISUP simulator sends RLC	ISDN:	Q.764 RLC Verify that the circuit is idle.
14			

2.1.25 Bearer service speech, abnormal situation during a call, timer T1 and T5: failure to receive a RLC

Test case ID:	SIP VoIP-ISUP-25
Test object:	Bearer service speech for ISUP signalling, abnormal situation during a call, timer T1 and T5: failure to receive a RLC.
Test purpose:	The purpose of this test case is to verify that a correct response is provided from the SIP GW to the SS7 (ISUP), when an abnormal situation occurs (failure to receive RLC) for a speech call from SS7 (ISUP) to the SIP User. See Q.784 test no.5.2.3.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test conditions:</u> Arrange the data in the ISUP simulator such that a release complete message is not returned in response to a release message.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
14	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 100 (Trying)
14	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)

14	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established
14	SIP User disconnects	SIP User: SIP GW: SIP GW:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL The ISUP simulator is unable to return a RLC message.
14	SIP GW (second time)	SIP GW:	Q.764 REL The ISUP simulator is unable to return a RLC message. Verify that the release message was sent between 4-15 seconds (T1) after sending of the initial release message.
14	SIP GW sends a reset circuit message	SIP GW:	Q.764 RSC Verify that the circuit has been removed from service. Verify that the maintenance system has been alerted. Verify that a reset circuit message was sent at 1 minute (T5) after sending of the initial release message.
14	The ISUP simulator sends RLC	ISDN:	Q.764 RLC Verify that the circuit is idle.
14			

2.1.26 Bearer service speech, abnormal situation during a call, reset of circuits during a call

Test case ID:	SIP VoIP-ISUP-26
Test object:	Bearer service speech for ISUP signalling, abnormal situation during a call, reset of circuits during a call.
Test purpose:	The purpose of this test case is to verify that a correct response is provided from the SIP GW to the SS7 (ISUP) and SIP, when an abnormal situation occurs (reset of circuits during a call) for a speech call from SS7 (ISUP) to the SIP User. See Q.784 test no.5.3.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Called termination is free.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
14	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 100 (trying)
15	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)

1[illegible]	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established
1[illegible]	ISUP simulator sends a reset-circuit message	ISDN:	Q.764 RSC
1[illegible]	SIP GW responds with release complete	SIP GW: SIP GW: SIP User:	Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the circuit is idle.
1[illegible]			

2.1.27 Bearer service speech, special call setup, dual seizure for controlling SP

Test case ID:	SIP VoIP-ISUP-27
Test object:	Bearer service speech for ISUP signalling, special call setup, dual seizure for controlling signalling point.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with dual seizure can be provided for a call from SS7 (ISUP) to the SIP User, and that correct response is provided to the SS7 (ISUP). See Q.784 test no.6.3.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: • Arrange the data in the ISUP simulator such that the ISUP simulator is the controlling signalling point.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
15	SIP User calls the ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request (call 1) SIP 100 (Trying) Q.764 IAM (cic=x) (call 1)
15	ISUP simulator calls the SIP User simultaneously (containing the same value of cic)	ISDN: SIP GW: SIP User:	Q.764 IAM (cic=x) (call 2) SIP INVITE Request (call 2) SIP 100 (Trying)
15	SIP GW calls the ISUP simulator (automatic repeat attempt)	SIP GW:	Q.764 IAM (cic=y) (call 1)

15	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM (cic=x) SIP PRACK Verify that the ringing tone can be heard on the call originated from the ISUP simulator. SIP 200 OK (PRACK confirm)
15	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM (cic=x) SIP ACK Verify that the connection is established. Verify that on detection of dual seizure, the call initiated by the controlling signalling point (ISUP simulator in this case) is completed, and the non-controlling signalling point is backed off.
16	ISUP simulator sends ACM (cic=y)	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM (cic=y) SIP 180 (ringing) SIP PRACK Verify that the ringing tone can be heard on the call originated from the SIP User. SIP 200 OK (PRACK confirm)
16	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM (cic=y) SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established.
16	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL (cic=x) Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the circuit (cic=x) is idle.
16	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL (cic=y) Q.764 RLC Verify that the circuit (cic=y) is idle.

2.1.28 64 kbit/s unrestricted bearer service, successful call setup

Test case ID:	SIP VoIP-ISUP-28
Test object:	64 kbit/s unrestricted bearer service for ISUP signalling, successful call setup.
Test purpose:	The purpose of this test case is to verify that the 64 kbit/s unrestricted bearer service can be provided for a call from SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.784 test no.7.1.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test conditions:</u> Called termination is free. The coding of IAM in the ISUP simulator: TMR= 64 kbit/s unrestricted, USI (if included) have appropriate information. For example USI has two octets for 64 kbit/s, echo control device indicator in the nature of connection indicator parameter is set to “not included”, echo control device is disabled or a non-echo controlled circuit is selected.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
16	ISUP simulator calls the SIP User	ISDN:	Q.764 IAM (TMR, USI)
		SIP GW:	SIP INVITE Request For detailed coding, see the section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.
		SIP User:	SIP 100 (Trying)

16	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK SIP 200 OK (PRACK confirm)
16	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established. Verify that it is possible to pass data between the SIP User and the ISUP simulator.
16	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.
16			
16			

2.1.29 64 kbit/s unrestricted bearer service, unsuccessful call setup

Test case ID:	SIP VoIP-ISUP-29
Test object:	64 kbit/s unrestricted bearer service for ISUP signalling, unsuccessful call setup.
Test purpose:	The purpose of this test case is to verify that the 64 kbit/s unrestricted bearer service can be provided for a call from SS7 (ISUP) to the SIP User, and that correct response is provided to the SS7 (ISUP) in case of call failure. See Q.784 test no.7.1.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Arrange the data in the SIP terminal such that a release message with a given cause is returned to the ISUP request.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
17	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 100 (Trying)

17	SIP User disconnects (with a error response)	SIP User: SIP GW: SIP GW: ISDN:	SIP a) 404 (Not Found), 485 (Ambiguous), 604(Does not exist anywhere) b) 401 (Unauthorized), 402 (Payment required), 403 (Forbidden) SIP ACK Q.764 REL (a) cause no.1 (Unallocated number) b) cause no. 21 (Call rejected) Q.764 RLC Verify that the circuit is idle.

2.1.30 3.1 kHz audio bearer service, successful call setup, from ISDN access

Test case ID:	SIP VoIP-ISUP-30
Test object:	3.1 kHz audio bearer service for ISUP signalling, successful call setup, from ISDN access.
Test purpose:	The purpose of this test case is to verify that the 3.1 kHz audio bearer service can be provided for a call from SS7 (ISUP) to the SIP User, and that correct response is provided to the SS7 (ISUP). See Q.784 test no.7.2.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Called termination is free. The coding of IAM in the ISUP simulator: TMR=3.1 Khz audio, USI (if included) have appropriate information (for example: USI has two or three octets for 3.1 Khz audio)
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
17	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM (TMR, USI) SIP INVITE Request For detailed coding, see the section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”. SIP 100 (Trying)

17	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
17	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established, and that data/speech is possible.
17	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.
17			
17			

2.1.31 3.1 kHz audio bearer service, successful call setup, from analogue access

Test case ID:	SIP VoIP-ISUP-31
Test object:	3.1 kHz audio bearer service for ISUP signalling, successful call setup, from analogue access.
Test purpose:	The purpose of this test case is to verify that the 3.1 kHz audio bearer service can be provided for a call from SS7 (ISUP) to the SIP User, and that correct response is provided to the SS7 (ISUP). See Q.784 test no.7.2.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Called termination is free. The coding of IAM in the ISUP simulator: TMR is set to “3.1 kHz audio”.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
17	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM (TMR) SIP INVITE Request For detailed coding, see the section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”. SIP 100 (Trying)

17	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
18	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established, and that data/speech is possible.
18	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.

2.1.32 Multi-use bearer service, rejected by the SIP/SS7 GW

Test case ID:	SIP VoIP-ISUP-32
Test object:	Multi-use audio bearer service for ISUP signalling, rejected by the SIP/SS7 GW.
Test purpose:	The purpose of this test case is to verify that a for call with the multi-use bearer service, the multi-use bearer service can be properly rejected by the SIP//SS7 Gateway, and the call successfully completed with fallback to speech for a call from SS7 (ISUP) to the SIP User.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. The content of the IAM is based on the following DSS1 coding: Q.931 SETUP (BC1=speech, BC2=UDI-TA (unrestricted digital information with tones and announcements), HLC1=Telephony, HLC2=Audiovisual)
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
18	ISUP simulator calls the SIP User	ISDN:	Q.764 IAM (TMR=64kbit/s unrestricted preferred, TMR prime value= speech, USI=speech, USI prime=UDI-TA, ATP: HLC1=Telephony, HLC2=Audiovisual)
		SIP GW:	SIP INVITE Request For detailed coding, see the section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.
		SIP User:	SIP 100 (Trying)

1ξ	SIP terminal is ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM (TMU=speech, ATP: progress indicator (pi) no.5) SIP PRACK SIP 200 OK (PRACK confirm) Verify that the SIP gateway returns ACM with TMU=speech. Pi #5: interworking has occurred and has resulted in a telecommunication service change (fallback).
1ξ	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established.
1ξ	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.

2.1.33 Multi-use bearer service supported for 7 kHz telephony, successful call setup with fallback to 7 kHz telephony / speech

Test case ID:	SIP VoIP-ISUP-33
Test object:	Multi-use audio bearer service (supported for 7 kHz telephony) for ISUP signalling, successful call setup with fallback to 7 kHz telephony/speech.
Test purpose:	The purpose of this test case is to verify that a call with the multi-use bearer service supported for 7 kHz telephony/ speech by the SIP//SS7 Gateway is successfully completed with fallback to 7 kHz telephony/speech for a call from SS7 (ISUP) to the SIP User.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. The content of the IAM is based on the following DSS1 coding: Q.931 SETUP (BC1=speech, BC2=UDI-TA (unrestricted digital information with tones and announcements), HLC1=Telephony, HLC2=Audiovisual)
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
18	ISUP simulator calls the SIP User	ISDN:	Q.764 IAM (TMR=64kbit/s unrestricted preferred, USI=speech, USI prime=UDI-TA, ATP: HLC1=Telephony, HLC2=Audiovisual)
		SIP GW:	SIP INVITE Request SDP= G.722, G.711, G.723.1, (G.728, G.729)
		SIP User:	SIP 100 (Trying)

18	SIP terminal is ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK SIP 200 OK (PRACK confirm)
18	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) a) SDP=G.722 b) SDP=G.711, G.723.1 (G.728, G.729) Q.764 ANM a) BC=UDI-TA and HLC=telephony is included in ATP. Echo canceller is not included by the gateway. b) TMU=speech is included. BC=speech and HLC=telephony is included in ATP. Echo canceller is included by the gateway. For detailed coding, see the section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”. SIP ACK Verify that the connection is established.
18	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.

2.1.34 Public supplementary service UUS1 implicit from non SIP User, text message

Test case ID:	SIP VoIP-ISUP-34
Test object:	Supplementary service UUS1, text messages for ISUP signalling.
Test purpose:	The purpose of this test case is to verify that UUI with a text message (UUI content= IA5 characters) can be included in ISUP for a call from the SS7 (ISUP) to the SIP User, that the info is discarded by the SIP gateway, and that correct response is provided to the SS7 (ISUP). See Q.785 test no.1.1.1.1.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
19	ISUP simulator calls the SIP User with UUI=IA5 characters	ISDN: SIP GW: SIP User:	Q.764 IAM (UUI contents=IA5 characters) SIP INVITE Request (no UUI) SIP 100 (Trying) Verify that the UUS1 information in the IAM is discarded in the SIP gateway. Note: UUS standards assume explicit rejection of UUS services. Optionally, this rejection can be performed by the SIP/SS7 GW. However, in our case it is assumed that the SIP/SS7 GW only discards the UUS related signalling information, and that the service rejection is handled by the originating PSTN/ISDN local exchange based on missing response.

19	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
19	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established.
19	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.
19			

2.1.35 Public supplementary service UUS2 from non SIP User, text message

Test case ID:	SIP VoIP-ISUP-35
Test object:	Supplementary service UUS2, text message for ISUP signalling.
Test purpose:	The purpose of this test case is to verify that UUI with a text message (UUI content= IA5 characters) can be included in ISUP for a call from the SS7 (ISUP) to the SIP User, that the info is discarded by the SIP gateway, and that correct response is provided to the SS7 (ISUP).
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
19	ISUP simulator calls the SIP User with UUS2 non essential request	ISDN: SIP GW: SIP User:	Q.764 IAM (UUI contents=IA5 characters) SIP INVITE Request (no UUI) SIP 100 (Trying) Verify that the UUS2 non essential request is discarded by the SIP gateway. Note: UUS standards assume explicit rejection of UUS services. Optionally, this rejection can be performed by the SIP/SS7 GW. However, in our case it is assumed that the SIP/SS7 GW only discards the UUS related signalling information, and that the service rejection is handled by the originating PSTN/ISDN local exchange based on missing response.

19	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
19	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established.
19	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.

2.1.36 Public supplementary service UUS3 from non SIP User, text message

Test case ID:	SIP VoIP-ISUP-36
Test object:	Supplementary service UUS3, text message for ISUP signalling.
Test purpose:	The purpose of this test case is to verify that UUI with a text message (UUI content= IA5 characters) can be included in ISUP for a call from the SS7 (ISUP) to the SIP User, that the info is discarded by the SIP gateway, and that correct response is provided to the SS7 (ISUP).
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
1ʢ	ISUP simulator calls the SIP User with UUS3 non essential request	ISDN: SIP GW: SIP User:	Q.764 IAM (UUI contents=IA5 characters) SIP INVITE Request (no UUI) SIP 100 (Trying) Verify that the UUS3 non essential request is discarded by the SIP gateway. Note: UUS standards assume explicit rejection of UUS services. Optionally, this rejection can be performed by the SIP/SS7 GW. However, in our case it is assumed that the SIP/SS7 GW only discards the UUS related signalling information, and that the service rejection is handled by the originating PSTN/ISDN local exchange based on missing response.

2(	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
2(	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established.
2(	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.

2.1.37 Public supplementary service CUG from non SIP User, successful CUG call

Test case ID:	SIP VoIP-ISUP-37
Test object:	Supplementary service CUG (Closed User Group) for ISUP signalling, successful CUG call.
Test purpose:	The purpose of this test case is to verify that a request for a CUG call can be included in ISUP for a call from the SS7 (ISUP) to the SIP User, that the info is discarded by the SIP gateway, and that correct response is provided to the SS7 (ISUP). In this case the call is set up as a non CUG call. See Q.785 test no.2.1.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
2(	ISUP simulator calls the SIP User with • Optional Forward call indicators parameter: CUG call indicator:1 0 (CUG call, outgoing access allowed) • CUG Interlock Code parameter: CUG Interlock code: interlock code included	ISDN: SIP GW: SIP User:	Q.764 IAM (optional forward call indicators, CUG interlock code) SIP INVITE Request SIP 100 (Trying) Verify that the SIP GW responds with a SIP INVITE (depending on basic call procedures) as a non CUG call.
2(	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)

2(	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established.
2(	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm)
2(			
2(			
2(			

2.1.38 Public supplementary service CUG from non SIP User, unsuccessful CUG call

Test case ID:	SIP VoIP-ISUP-38
Test object:	Supplementary service CUG (Closed User Group) for ISUP signalling, unsuccessful CUG call.
Test purpose:	The purpose of this test case is to verify that a request for a CUG call can be included in ISUP for a call from the SS7 (ISUP) to the SIP User, that the info is discarded by the SIP gateway, and that correct response is provided to the SS7 (ISUP). In this case the CUG call is illegal, and the call is disconnected by the SIP/SS7 Gateway. See Q.785 test no.2.1.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
21	ISUP simulator calls the SIP User with Optional forward call indicators parameter indicating “closed user group call, outgoing access not allowed”. CUG Interlock Code parameter: CUG Interlock code: interlock code included	ISDN: SIP GW:	Q.764 IAM Verify that the request for a CUG call is rejected by the SIP GW.
21	SIP GW disconnects	SIP GW: ISDN:	Q.764 REL (cause #87- User not member of CUG) Q.764 RLC
21			

2.1.39 Public supplementary service CLIP from non SIP User, network provided -1

Test case ID:	SIP VoIP-ISUP-39
Test object:	Supplementary service CLIP (Called line identification presentation) for ISUP signalling, network provided.
Test purpose:	The purpose of this test case is to verify that the CLIP supplementary service can be successfully completed for a call from the SS7 (ISUP) to the SIP User, that the SIP signalling procedures for the CLIP supplementary service are correct, and that correct response is provided to the SS7 (ISUP). See Q.785 test no.3.1.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Arrange the data in the ISUP simulator such that the IAM contains a Calling party number parameter (national (significant) number/ network provided/ presentation allowed).
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.
Test Procedure Steps:	**Expected Results:**

21	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM (Calling party number) Calling party number parameter: • Address signal=cgn • NoA= National (significant) number • Screening indicator=network provided • Address presentation restricted indicator=presentation allowed SIP INVITE Calling address in the From-field: Sip:+47<cgn>@host.domain SIP 100 (Trying) For detailed coding, see the section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.
21	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
21	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established.
21	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the circuit is idle.

2.1.40 Public supplementary service CLIP from non SIP User, network provided -2

Test case ID:	SIP VoIP-ISUP-40
Test object:	Supplementary service CLIP (Called line identification presentation) for ISUP signalling, network provided.
Test purpose:	The purpose of this test case is to verify that the CLIP supplementary service can be successfully completed for a call from the SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.785 test no.3.1.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test conditions:</u> Arrange the data in the ISUP simulator such that the IAM contains a Calling party number parameter (international number/ network provided/ presentation allowed).
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.
Test Procedure Steps:	**Expected Results:**

21	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM (Calling party number) Calling party number parameter: • Address signal=cgn • NoA= international number • Screening indicator=network provided • Address presentation restricted indicator=presentation allowed SIP INVITE Calling address in the From-field: Sip:+<cgn>@host.domain SIP 100 (Trying) For detailed coding, see the section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.
21	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
21	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established.
22	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the circuit is idle.

2.1.41 Public supplementary service CLIP from non SIP User, user provided -3

Test case ID:	SIP VoIP-ISUP-41
Test object:	Supplementary service CLIP (Called line identification presentation) for ISUP signalling, user provided.
Test purpose:	The purpose of this test case is to verify that the CLIP supplementary service can be successfully completed for a call from the SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.785 test no.3.1.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Arrange the data in the ISUP simulator such that the IAM contains a Calling party number parameter (national (significant) number/ user provided/ presentation allowed).
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.
Test Procedure Steps:	**Expected Results:**

22	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM (Calling party number) Calling party number parameter: • Address signal=cgn • NoA= National (significant number) • Screening indicator=user provided, verified and passed • Address presentation restricted indicator=presentation allowed SIP INVITE Calling address in the From-field: Sip: +47<cgn>@host.domain SIP 100 (Trying) For detailed coding, see the section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.
22	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
22	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established.
22	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the circuit is idle.

2.1.42 Public supplementary service CLIP from non SIP User, network provided -4

Test case ID:	SIP VoIP-ISUP-42
Test object:	Supplementary service CLIP (Called line identification presentation) for ISUP signalling, network provided.
Test purpose:	The purpose of this test case is to verify that the CLIP supplementary service can be successfully completed for a call from the SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.785 test no.3.1.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test conditions:</u> Arrange the data in the ISUP simulator such that the IAM contains a Calling party number parameter (national(significant)number /network provided/ presentation allowed) and a Generic number parameter (national(significant) number/ user provided/ presentation allowed).
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:	**Expected Results:**

22	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM (Calling party number, Generic number parameter) Calling party number parameter: • Address presentation restricted indicator=<u>presentation allowed</u> • Screening indicator=<u>network provided</u> • NoA= <u>National (significant number)</u> • Address signal=cgn Generic number parameter: • Number qualifier=additional calling party number • Address presentation restricted indicator=<u>presentation allowed</u> • Screening indicator=<u>user provided, not verified</u> • NoA= <u>national(significant) number</u> • Address signal=cgnadd SIP INVITE Calling address in the From-field: Sip: +47<cgnadd>@host.domain Verify that it is the cgnadd in the Generic number parameter which is mapped to the SIP calling address. The calling party number parameter is discarded if present. SIP 100 (Trying)
22	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)

22	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established.
22	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the circuit is idle.

2.1.43 Public supplementary service CLIP from non SIP User, network provided -5

Test case ID:	SIP VoIP-ISUP-43
Test object:	Supplementary service CLIP (Called line identification presentation) for ISUP signalling, network provided.
Test purpose:	The purpose of this test case is to verify that the CLIP supplementary service can be successfully completed for a call from the SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.785 test no.3.1.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test conditions:</u> Arrange the data in the ISUP simulator such that the IAM contains a Calling party number parameter (international number/ network provided/ presentation allowed) and a Generic number parameter (international number/ user provided/ presentation allowed).
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.
Test Procedure Steps:	**Expected Results:**

22	ISUP simulator calls the SIP User	ISDN:	Q.764 IAM (Calling party number, Generic number parameter) Calling party number parameter: • Address presentation restricted indicator=presentation allowed • Screening indicator=network provided • NoA= international number • Address signal=cgn Generic number parameter: • Number qualifier=additional calling party number • Address presentation restricted indicator=presentation allowed • Screening indicator=user provided, nor verified • NoA= international number • Address signal=cgnadd
		SIP GW:	SIP INVITE Calling address in the From-field: Sip: +<cgnadd>@host.domain Verify that it is the cgnadd in the Generic number parameter which is mapped to the SIP calling address. The calling party number parameter is discarded if present.
		SIP User:	SIP 100 (Trying)
23	SIP terminal ringing	SIP User:	SIP 180 (Ringing)
		SIP GW:	Q.764 ACM
		SIP GW:	SIP PRACK Verify that the ringing tone can be heard.
		SIP User:	SIP 200 OK (PRACK confirm)

23	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established.
23	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the circuit is idle.

2.1.44 Public supplementary service CLIR from non SIP User, network provided

Test case ID:	SIP VoIP-ISUP-44
Test object:	Supplementary service CLIR (Called line identification restriction) for ISUP signalling, network provided.
Test purpose:	The purpose of this test case is to verify that the CLIR supplementary service can be successfully completed for a call from the SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.785 test no.3.3.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Arrange the data in the ISUP simulator such that the IAM contains a Calling party number parameter (national(significant) number/ network provided/ presentation restricted).
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.
Test Procedure Steps:	**Expected Results:**

23	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM (calling party number) Calling party number parameter: • Address presentation restricted indicator=<u>presentation restricted</u> • Screening indicator=<u>network provided</u> • NoA= <u>national (significant) number</u> • Address signal=cgn SIP INVITE, calling address in the From-field: Anonymous<sip:restricted@host.domain • The Calling party number parameter is discarded, and the content not mapped to SIP SIP 100 (Trying)
23	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
23	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established.
23	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the circuit is idle.

2.1.45 Public supplementary service CLIR from non SIP User, network provided/ user provided

Test case ID:	SIP VoIP-ISUP-45
Test object:	Supplementary service CLIR (Called line identification restriction) for ISUP signalling, network provided/ user provided.
Test purpose:	The purpose of this test case is to verify that the CLIR supplementary service can be successfully completed for a call from the SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.785 test no.3.3.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Arrange the data in the ISUP simulator such that the IAM contains a Calling party number parameter (national (significant) number/ network provided/ presentation restricted) and a Generic number parameter (national(significant) number/ user provided/ presentation restricted).
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:	**Expected Results:**

23	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM (calling party number) Calling party number parameter: • Address presentation restricted indicator=presentation restricted • Screening indicator=network provided • NoA= national (significant) number • Address signal=cgn Generic number parameter: • Number qualifier=additional calling party number • Address presentation restricted indicator=presentation restricted • Screening indicator=user provided, not verified • NoA= national(significant) • Address signal=cgnadd SIP INVITE, calling address in the From-field: Anonymous<sip:restricted>@host.domain • The Calling party number parameter is discarded, and the content not mapped to SIP • The Generic number parameter is discarded, and the content is not mapped to SIP SIP 100 (Trying)
23	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)

23	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established.
24	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the circuit is idle.

2.1.46 Public supplementary service COLP from non SIP User

Test case ID:	SIP VoIP-ISUP-46
Test object:	Supplementary service COLP (Connected line identification presentation) for ISUP signalling, request.
Test purpose:	The purpose of this test case is to verify that the COLP supplementary service can be rejected for a call from the SS7 (ISUP) to the SIP User, that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP). See Q.785 test no.6.1.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test conditions:</u> Arrange the data in the ISUP simulator such that the IAM contains a request for COL
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
24	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM (optional forward call indicators: connected line identity request indicator= 1(requested)) SIP INVITE The optional forward call indicators parameter in the IAM is ignored. SIP 100 (Trying)

24	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
24	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM No connected number is sent. SIP ACK Verify that the connection is established.
24	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the circuit is idle.
24			
24			

2.1.47 Public supplementary service Diversion, forwarded call from ISDN

Test case ID:	SIP VoIP-ISUP-47
Test object:	Supplementary service Diversion for ISUP signalling.
Test purpose:	The purpose of this test case is to verify that a redirecting number can be included in ISUP for a call from the SS7 (ISUP) to the SIP GW, that the info is correctly mapped or discarded by the SIP gateway, and that correct response is provided to the SS7 (ISUP).
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 "SIP-ISUP interworking for Telenor national interconnect ISUP version 2".
Test Procedure Steps:	**Expected Results:**

24	ISUP simulator calls the SIP User with forwarded call (CFU)	ISDN: SIP GW: SIP User:	Q.764 IAM (Redirection information, Redirecting number, Original called number) SIP INVITE Request : INVITE sip:"user C" from Called party number in ISUP From sip:"user A" from Calling party number/additional calling party number in ISUP according to CLIP/CLIR rules. To sip:"user B" from Original called number when coded "presentation allowed", not included when coded "presentation restricted". Redirection information not mapped to SIP. SIP 100 (Trying)
24	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
24	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established.
25	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.

2.1.48 Public supplementary service Diversion, call to be forwarded by SIP with forwarding capability in the SIP gateway, unsuccessful

Test case ID:	SIP VoIP-ISUP-48
Test object:	Supplementary service Diversion for ISUP signalling.
Test purpose:	The purpose of this test case is to verify that a SIP gateway with forwarding capability can forward an incoming call from PSTN/ISDN to a SIP User (with call diversion towards PSTN/ISDN invoked by SIP) back to ISDN. In this testcase SIP redirects to PSTN/ISDN (before SIP 18x) with a SIP 302 message, but the call is rejected by the SIP/SS7 GW because of the content of the redirection counter.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) The section 4 "SIP-ISUP interworking for Telenor national interconnect ISUP version 2".

Test Procedure Steps:		**Expected Results:**	
25	ISUP simulator calls the SIP User with forwarded call (CFU)	ISDN: SIP GW:	Q.764 IAM (Redirection information (with a redirection counter=5), Redirecting number, Original called number, Called party number) SIP INVITE Request (no Redirection information, no Redirecting number)
25	SIP User indicating that the resource which the user is attempting to contact is at a different location by sending a 3xx message (SIP redirection)	SIP User: SIP GW:	SIP 302 (Moved temporarily) SIP ACK
25	SIP GW rejects the call	SIP GW: ISDN:	Q.764 REL (cause#21) Q.764 RLC

2.1.49 Public supplementary service Diversion, call to be forwarded by SIP with forwarding capability in the SIP gateway (300-1)

Test case ID:	SIP VoIP-ISUP-49
Test object:	Supplementary service Diversion for ISUP signalling.
Test purpose:	The purpose of this test case is to verify that a SIP gateway with forwarding capability can forward an incoming call from PSTN/ISDN to a SIP User (with call diversion towards PSTN/ISDN invoked by SIP) back to ISDN. In this testcase SIP redirects to PSTN/ISDN (before SIP 18x) with a SIP 300 message.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
25	ISUP simulator calls the SIP User with forwarded call (CFU)	ISDN: SIP GW:	Q.764 IAM (ISUP leg 1) SIP INVITE Request
25	SIP User indicating that the resource which the user is attempting to contact is at a different location by sending a 3xx message (SIP redirection)	SIP User: SIP GW: SIP GW:	SIP 300 (Multiple choices) Q.764 ACM (ISUP leg 1) SIP ACK

25	SIP GW redirects the call to ISDN	SIP GW:	Q.764 IAM (with Redirection information, Redirecting number, Original called number), depending upon whether the SIP redirect is invoked from the Internet or from within a secure IP VPN dedicated to a specific company (ISUP leg 2) For detailed coding, see the section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.
25	ISUP simulator sends ACM	ISDN: SIP GW:	Q.764 ACM (ISUP leg 2) Q.764 CPG (ISUP leg 1)
25	ISUP simulator answers the call	ISDN: SIP GW:	Q.764 ANM (ISUP leg 2) Q.764 ANM (ISUP leg 1) Verify that the connection is established
25	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP GW:	Q.764 REL (ISUP leg 2) Q.764 RLC (ISUP leg 2) Q.764 REL (ISUP leg 1) Q.764 RLC (ISUP leg 1) Verify that the circuit is idle.

2.1.50 Public supplementary service Diversion, call to be forwarded by SIP with forwarding capability in the SIP gateway (301)

Test case ID:	SIP VoIP-ISUP-50
Test object:	Supplementary service Diversion for ISUP signalling.
Test purpose:	The purpose of this test case is to verify that a SIP gateway with forwarding capability can forward an incoming call from PSTN/ISDN to a SIP User (with call diversion towards PSTN/ISDN invoked by SIP) back to ISDN. In this testcase SIP redirects to PSTN/ISDN (before SIP 18x) with a SIP 301 message.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
2(	ISUP simulator calls the SIP User with forwarded call (CFU)	ISDN: SIP GW:	Q.764 IAM (ISUP leg 1) (Redirection information (with a redirection counter=1), Redirecting number, Original called number) SIP INVITE Request (no Redirection information, no Redirecting number)
2(	SIP User indicating that the resource which the user is attempting to contact is at a different location by sending a 3xx message (SIP redirection)	SIP User : SIP GW: SIP GW:	SIP 301 (Moved permanently) Q.764 ACM (ISUP leg 1) SIP ACK

26	SIP GW redirects the call to ISDN	SIP GW:	Q.764 IAM (with Redirection information, Redirecting number, Original called number) (ISUP leg 2) For detailed coding, see the section 4 "SIP-ISUP interworking for Telenor national interconnect ISUP version 2".
26	ISUP simulator sends ACM	ISDN: SIP GW:	Q.764 ACM (ISUP leg 2) Q.764 CPG (ISUP leg 1)
26	ISUP simulator answers the call	ISDN: SIP GW:	Q.764 ANM (ISUP leg 2) Q.764 ANM (ISUP leg 1) Verify that the connection is established
26	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP GW:	Q.764 REL (ISUP leg 2) Q.764 RLC (ISUP leg 2) Q.764 REL (ISUP leg 1) Q.764 RLC (ISUP leg 1) Verify that the circuit is idle.

2.1.51 Public supplementary service Diversion, call to be forwarded by SIP with forwarding capability in the SIP gateway (302-1)

Test case ID:	SIP VoIP-ISUP-51
Test object:	Supplementary service Diversion for ISUP signalling.
Test purpose:	The purpose of this test case is to verify that a SIP gateway with forwarding capability can forward an incoming call from PSTN/ISDN to a SIP User (with call diversion towards PSTN/ISDN invoked by SIP) back to ISDN. In this test case SIP redirects to PSTN/ISDN (before SIP 18X) with a SIP 302 message.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
26	ISUP simulator calls the SIP User with forwarded call (CFU)	ISDN: SIP GW:	Q.764 IAM (ISUP leg 1) (Redirection information (with a redirection counter=1), Redirecting number, Original called number) SIP INVITE Request (no Redirection information, no Redirecting number)
26	SIP User indicating that the resource which the user is attempting to contact is at a different location by sending a 3xx message (SIP redirection)	SIP User: SIP GW: SIP GW:	SIP 302 (Moved temporarily) Q.764 ACM (ISUP leg 1) SIP ACK

26	SIP GW redirects the call to ISDN	SIP GW:	Q.764 IAM (with Redirection information, Redirecting number, Original called number) (ISUP leg 2) For detailed coding, see the section 4 "SIP-ISUP interworking for Telenor national interconnect ISUP version 2".
26	ISUP simulator sends ACM	ISDN: SIP GW:	Q.764 ACM (ISUP leg 2) Q.764 CPG (ISUP leg 1)
27	ISUP simulator answers the call	ISDN: SIP GW:	Q.764 ANM (ISUP leg 2) Q.764 ANM (ISUP leg 1) Verify that the connection is established
27	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP GW:	Q.764 REL (ISUP leg 2) Q.764 RLC (ISUP leg 2) Q.764 REL (ISUP leg 1) Q.764 RLC (ISUP leg 1) Verify that the circuit is idle.

2.1.52 Public supplementary service Diversion, call to be forwarded by SIP with forwarding capability in the SIP gateway (302-2)

Test case ID:	SIP VoIP-ISUP-52
Test object:	Supplementary service Diversion for ISUP signalling.
Test purpose:	The purpose of this test case is to verify that a SIP gateway with forwarding capability can forward an incoming call from PSTN/ISDN to a SIP User (with call diversion towards PSTN/ISDN invoked by SIP) back to ISDN. In this testcase SIP redirects to PSTN/ISDN (after SIP 180 (Ringing)) with a SIP 302 message.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
27	ISUP simulator calls the SIP User with call forwarded on no reply (CFNR)	ISDN:	Q.764 IAM (ISUP leg 1) (Redirection information (with a redirection counter=1), Redirecting number, Original called number, Called party number)
		SIP GW:	SIP INVITE Request (no Redirection information, no Redirecting number)

27	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM (ISUP leg 1) SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
27	SIP User indicating that the resource which the user is attempting to contact is at a different location by sending a 3xx message (SIP redirection)	SIP User: SIP GW: SIP GW:	SIP 302 (Moved temporarily) Q.764 CPG (ISUP leg 1) SIP ACK
27	SIP GW redirects the call to ISDN	SIP GW:	Q.764 IAM (with Redirection information, Redirecting number, Original called number) (ISUP leg 2) For detailed coding, see the section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.
27	ISUP simulator sends CPG	ISDN: SIP GW:	Q.764 ACM (ISUP leg 2) Q.764 CPG (ISUP leg 1)
27	ISUP simulator answers the call	ISDN: SIP GW:	Q.764 ANM (ISUP leg 2) Q.764 ANM (ISUP leg 1) Verify that the connection is established
27	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP GW:	Q.764 REL (ISUP leg 2) Q.764 RLC (ISUP leg 2) Q.764 REL (ISUP leg 1) Q.764 RLC (ISUP leg 1) Verify that the circuit is idle.
27			
28			
28			

2.1.53 Public supplementary service Diversion, call to be forwarded by SIP with forwarding capability in the SIP gateway (300 –2)

Test case ID:	SIP VoIP-ISUP-53
Test object:	Supplementary service Diversion for ISUP signalling.
Test purpose:	The purpose of this test case is to verify that a SIP gateway with forwarding capability can forward an incoming call from PSTN/ISDN to a SIP User (with SIP redirect (3xx) to another SIP User).
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
28	ISUP simulator calls the SIP User B with forwarded call (CFU)	ISDN: SIP GW:	Q.764 IAM SIP INVITE Request
28	SIP User B indicating that the resource which the user is attempting to contact is at a different location by sending a 3xx message (SIP redirection)	SIP User B: SIP GW: SIP GW:	SIP 300 (Multiple choices) Q.764 ACM SIP ACK
28	SIP GW redirects the call to another SIP User C	SIP GW: SIP User C: SIP GW: SIP GW: SIP User C:	SIP INVITE request SIP 180 (Ringing) Q.764 CPG SIP PRACK SIP 200 OK (PRACK confirm) For detailed coding, see the section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

28	SIP User C answers the call	SIP User C: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK
28	ISUP simulator sends REL	ISDN: SIP GW:	Q.764 REL Q.764 RLC Verify that the circuit is idle.

2.1.54 Public supplementary services TP from non SIP User

Test case ID:	SIP VoIP-ISUP-54
Test object:	Supplementary services TP (Terminal Portability) for ISUP signalling.
Test purpose:	The purpose of this test case is to verify that notification for public TP can be included in ISUP for a call from the SS7 (ISUP) to the SIP User, that the info is discarded by the SIP gateway, and that correct response is provided to the SS7 (ISUP).
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
28	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 100 (trying)
28	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
28	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established

2[illegible]	Suspend/Resume (ISDN User initiated)	ISDN: SIP GW:	Q.764 SUS (user initiated) Verify that this message is discarded and not mapped to SIP. Q.764 RES (user initiated) Verify that this message is discarded and not mapped to SIP.
2[illegible]	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.

2.1.55 Public supplementary services HOLD from non SIP User

Test case ID:	SIP VoIP-ISUP-55
Test object:	Supplementary services HOLD (Call Hold) for ISUP signalling.
Test purpose:	The purpose of this test case is to verify that notification for public HOLD can be included in ISUP for a call from the SS7 (ISUP) to the SIP User, that the info is discarded by the SIP gateway, and that correct response is provided to the SS7 (ISUP).
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
29	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 100 (trying)
29	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
29	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established

2S	HOLD Notification	ISDN: SIP GW:	Q.764 CPG (Generic notification indicator: HOLD notification) Verify that the Generic notification indicator parameter is discarded by the SIP GW, and not mapped to SIP.
2S	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.

2.1.56 Public supplementary services CONF/3PTY from non SIP User

Test case ID:	SIP VoIP-ISUP-56
Test object:	Supplementary services CONF/3PTY (Conference calling/Three party service) for ISUP signalling.
Test purpose:	The purpose of this test case is to verify that notification for public CONF/3PTY can be included in ISUP for a call from the SS7 (ISUP) to the SIP User, that the info is discarded by the SIP gateway, and that correct response is provided to the SS7 (ISUP).
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
29	ISUP simulator calls the SIP User	ISDN: SIP GW: SIP User:	Q.764 IAM SIP INVITE Request SIP 100 (trying)
29	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) Q.764 ACM SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
29	SIP User answers the call	SIP User: SIP GW: SIP GW:	SIP 200 OK (INVITE confirm) Q.764 ANM SIP ACK Verify that the connection is established

3(	CONF/3PTY Notification	ISDN: SIP GW:	Q.764 CPG (Generic notification indicator: CONF/3PTY notification) Verify that the Generic notification indicator parameter is discarded by the SIP GW, and not mapped to SIP.
3(	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.

2.2 Call from SIP access to SS7 (ISUP)

2.2.1 Circuit supervision, receipt of unreasonable signalling information messages, receipt of unexpected messages during call setup

Test case ID:	SIP VoIP-ISUP-57
Test object:	Receipt of unreasonable signalling information messages for ISUP signalling, receipt of unexpected messages during call setup.
Test purpose:	The purpose of this test case is to verify that the action taken by a signalling point upon receipt of unexpected messages is as stated in Q.764 Section 2.9.5.1, and that the correct response is provided to the ISUP (SS7). See Q.784 test no. 1.5.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: • Arrange the data in the ISUP simulator such that other unreasonable messages may be initiated. • The circuit shall be idle and unblocked.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
3(	The SIP user calls the ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SIP 100 (Trying) Q.764 IAM
3(	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK SIP 200 OK (PRACK confirm)

3(	ISUP simulator sends an unreasonable message XXX after ACM XXX – Messages other than the call control messages	ISDN: SIP GW:	Q.764 XXX (ex. SUS) No action, the message is ignored.
3(	ISUP simulator answers the call	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established
3(	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.
3(			
3(			

2.2.2 Circuit supervision, receipt of unreasonable signalling information messages, receipt of unexpected messages during a call

Test case ID:	SIP VoIP-ISUP-58
Test object:	Receipt of unreasonable signalling information messages for ISUP signalling, receipt of unexpected messages during a call.
Test purpose:	The purpose of this test case is to verify that the action taken by a signalling point upon receipt of unexpected messages is as stated in Q.764 Section 2.9.5.1, and that the correct response is provided to the SS7. See Q.784 test no. 1.5.3.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: • Arrange the data in the ISUP simulator such that an unexpected RLC and other unreasonable messages may be initiated. • The circuit shall be idle and unblocked.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
3(	The SIP user calls the ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SIP 100 (Trying) Q.764 IAM
31	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK SIP 200 OK (PRACK confirm)
31	ISUP simulator answers the call	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established

31	ISUP simulator sends a message other than REL, RLC, RSC and SUS	ISDN: SIP GW:	Q.764 ACM (example) No action, the message is ignored. Verify that the connection still is established
31	ISUP simulator sends a release complete message (RLC)	ISDN: SIP GW: SIP GW: SIP User: ISDN:	Q.764 RLC Q.764 REL SIP BYE SIP 200 OK (BYE confirm) Q.764 RLC Verify that the circuit is idle.
31			

2.2.3 Bearer service speech, both way circuit selection, IAM sent by non-controlling signalling point

Test case ID:	SIP VoIP-ISUP-59
Test object:	Both way circuit selection for ISUP signalling, IAM sent by non-controlling signalling point.
Test purpose:	The purpose of this testcase is to verify that the SIP User can initiate an outgoing call on a circuit capable of both way operation when the controlling signalling point is the SIP GW, and that the correct response is provided to the SS7. See Q.784 test no. 2.1.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: • Called termination is free • Circuit selected is capable of both way operation • The SIP GW is the controlling signalling point
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
31	The SIP user calls the ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request (call 1) SIP 100 (Trying) (call 1) Q.764 IAM (cic=x) (call 1)
31	ISUP simulator calls the SIP User simultaneously with the same cic=x	ISDN:	Q.764 IAM (cic=x) (call 2) Verify that (call 2) is backed off since the controlling signalling point is the SIP GW.
31	ISUP simulator calls the SIP User (automatic repeat attempt)	ISDN: SIP GW: SIP User:	Q.764 IAM (cic=y) (call 3) SIP INVITE Request (call 3) SIP 100 (Trying) (call 3)

31	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM (cic=x) (call 1) SIP 180 (Ringing) (call 1) SIP PRACK (call 1) SIP 200 OK (PRACK confirm) (call 1)
31	ISUP simulator answers the call	ISDN: SIP GW: SIP User:	Q.764 ANM (cic=x) (call 1) SIP 200 OK (INVITE confirm) (call 1) SIP ACK (call 1) Verify that the connection is established
32	SIP terminal ringing	SIP User: SIP GW: SIP GW: SIP User:	SIP 180 (Ringing) (call 3) Q.764 ACM (cic=y) (call 3) SIP PRACK (call 3) Verify that the ringing tone can be heard on the call originated from the ISUP simulator. SIP 200 OK (PRACK confirm) (call 3)
32	SIP User answers the call	SIP User: SIP GW: SIP User:	SIP 200 OK (INVITE confirm) (call 3) Q.764 ANM (cic=y) (call 3) SIP ACK (call 3) Verify that the connection is established.
32	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL (cic=x) (call 1) Q.764 RLC (call 1) SIP BYE (call 1) SIP 200 OK (BYE confirm) (call 1) Verify that the circuit (cic=x) is idle.
32	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL (cic=y) (call 3) Q.764 RLC (call 3) SIP BYE (call 3) SIP 200 OK (BYE confirm) (call 3) Verify that the circuit (cic=y) is idle.

2.2.4 Bearer service speech, called address sending, en block operation

Test case ID:	SIP VoIP-ISUP-60
Test object:	Bearer service speech for ISUP signalling, called address sending, en block operation.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with en bloc operation can be provided for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User. See Q.784 test no. 2.2.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test conditions:</u> • Called termination is free • The SIP data is arranged such that all digits are included in the SIP INVITE request.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
32	SIP User calls the ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SIP 100 (Trying) Q.764 IAM
32	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM (subscriber free) SIP 180 (Ringing) SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)

32	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established
32	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the circuit is idle.

2.2.5 Bearer service speech, successful call setup, ordinary call with various indications in alerting

Test case ID:	SIP VoIP-ISUP-61
Test object:	Bearer service speech for ISUP signalling, successful call setup, ordinary call with various indications in alerting.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with various indications in alerting can be provided for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User. See Q.784 test no. 2.3.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Called termination is free
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
32	SIP User calls the ISUP simulator	SIP User:	SIP INVITE Request Allowed media coding: G.711 or G.723.1 (G.728, G.729)
		SIP GW:	SIP 100 (Trying)
		SIP GW:	Q.764 IAM (TMR=”speech”) For detailed coding, ref. attached doc 1

32	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM (subscriber free) a) subscriber free b) no indication c) no indication, in-band information or an appropriate pattern is now available d) no indication, call diversion may occur a) SIP 180 (Ringing) with SDP=G.711/G.723.1 (G.728, G.729) b) SIP 183 (Session Progress) c) SIP 183 (Session Progress) with SDP=G.711/G.723.1 (G.728, G.729) d) SIP 181 (Call is being forwarded) For detailed coding, ref. attached doc 1 SIP PRACK SIP 200 OK (PRACK confirmed)
33	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established
33	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.
33			

2.2.6 Bearer service speech, successful call setup, ordinary call with call progress message

Test case ID:	SIP VoIP-ISUP-62
Test object:	Bearer service speech for ISUP signalling, successful call setup, ordinary call with call progress message.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with progress message can be provided for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User. See Q.784 test no. 2.3.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Called termination is free
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
3:	The SIP user calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SIP 100 (Trying) Q.764 IAM
3:	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM (no indication) SIP 183 (Call progress) SIP PRACK SIP 200 OK (PRACK confirm)

33	ISUP simulator sends CPG	ISDN: SIP GW: SIP User: SIP GW:	Q.764 CPG (a) Event indicator="ALERTING" or optional backward call indicators="subscriber free" b) Event indicator="PROGRESS" c) Event indicator or optional backward call indicators= "inband information or an appropriate pattern is now available" d) Event indicator="call forwarded on busy/call forwarded on no reply/call forwarded unconditional", or optional backward call indicators="call diversion may occur"). a) SIP 180 (Ringing) with SDP=G.711/G.723.1 (G.728, G.729) b) SIP 183 (Call Progress) c) SIP 183 (Call Progress) with SDP=G.711/G.723.1 (G.728, G.729) d) SIP 181 (call is being forwarded) SIP PRACK SIP 200 OK (PRACK confirm)
33	ISUP simulator answers the call	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established
33	SIP User disconnects	SIP: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.

2.2.7 Bearer service speech, successful call setup, ordinary call with various indications in the connect message (CON)

Test case ID:	SIP VoIP-ISUP-63
Test object:	Bearer service speech for ISUP signalling, successful call setup, ordinary call with various indications in the connect message.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with various indications in the connect message can be provided for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User. See Q.784 test no. 2.3.3.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Use the following indications of backward call indicators in the CON message: • Called party status indicator=”subscriber free” or “no indication” • ISDN access indicator=”ISDN” or “NON ISDN” Pre-test conditions: Called termination is free. A connect message is returned instead of an answer message from the ISUP simulator.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
33	The SIP user calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SIP 100 (Trying) Q.764 IAM

33	ISUP simulator answers the call	ISDN: SIP GW: SIP User:	Q.764 CON SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established.
34	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.

2.2.8 Bearer service speech, normal call release, calling party clears before any backward messages (ACM)

Test case ID:	SIP VoIP-ISUP-64
Test object:	Bearer service speech for ISUP signalling, normal call release, calling party clears before any backward messages.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with the calling party releases a call prior to receipt of any backward message, can be provided for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User. See Q.784 test no.3.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test conditions:</u> The circuit is idle.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
34	The SIP user calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SIP 100 (Trying) Q.764 IAM
34	The SIP user disconnects	SIP User: SIP GW: SIP GW: SIP GW:	SIP CANCEL request SIP 200 OK (CANCEL confirm) Q.764 REL (cause #16 (Normal call clearing)) SIP 487 (Call cancelled)
34	ISUP simulator responds with RLC	ISDN: SIP User:	Q.764 RLC SIP ACK Verify that the circuit is idle.

2.2.9 Bearer service speech, normal call release, calling party clears before answer

Test case ID:	SIP VoIP-ISUP-65
Test object:	Bearer service speech for ISUP signalling, normal call release, calling party clears before answer.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with the calling party clears before answer, can be provided for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User. See Q.784 test no.3.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: The called termination is free.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
34	The SIP user calls ISUP simulator	SIP User:	SIP INVITE Request
		SIP GW:	SIP 100 (Trying)
		SIP GW:	Q.764 IAM
34	ISUP simulator sends ACM	ISDN:	Q.764 ACM
		SIP GW:	SIP 180 (Ringing)
		SIP User:	SIP PRACK
			Verify that the ringing tone can be heard.
		SIP GW:	SIP 200 OK (PRACK confirm)

34	The SIP user disconnects	SIP User: SIP GW: SIP GW: SIP GW:	SIP CANCEL request SIP 200 OK (CANCEL confirm) Q.764 REL (cause #16 (Normal call clearing)) SIP 487 (Call cancelled)
34	ISUP simulator responds with RLC	ISDN: SIP User:	Q.764 RLC SIP ACK Verify that the circuit is idle.
34			
34			

2.2.10 Bearer service speech, normal call release, calling party clears after answer

Test case ID:	SIP VoIP-ISUP-66
Test object:	Bearer service speech for ISUP signalling, normal call release, calling party clears after answer.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with the calling party clears after answer, can be provided for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User. See Q.784 test no.3.3.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: The called termination is free.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
35	The SIP user calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SIP 100 (Trying) Q.764 IAM
35	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)

35	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established
35	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL (Cause #16 (Normal call clearing)) Q.764 RLC Verify that the circuit is idle.

2.2.11 Bearer service speech, normal call release, called party clears after answer

Test case ID:	SIP VoIP-ISUP-67
Test object:	Bearer service speech for ISUP signalling, normal call release, called party clears after answer.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with the called party clears after answer, can be provided for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User. See Q.784 test no.3.4.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: The called termination is free.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
3ξ	The SIP user calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SIP 100 (Trying) Q.764 IAM
3ξ	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
3ξ	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established

3ξ	ISUP simulator sends REL	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the circuit is idle.
3ξ			
3ξ			

2.2.12 Bearer service speech, normal call release, suspend initiated by the network

Note: A possible procedure to map the ISUP signalling procedure to SIP is described in draft RFC. However this procedure is not considered necessary.

Test case ID:	SIP VoIP-ISUP-68
Test object:	Bearer service speech for ISUP signalling, normal call release, suspend initiated by the network.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with suspend initiated by the network, can be provided for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User. See Q.784 test no.3.5.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: The called termination is free
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
3€	The SIP user calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SIP 100 (Trying) Q.764 IAM
3€	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)

3(	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established
3(	ISUP simulator sends a suspend message (SUS)	ISDN: SIP GW:	Q.764 SUS The SUS message is discarded and not mapped to SIP Timer is started in the gateway
3(	ISUP simulator sends a resume message (RES)	ISDN: SIP GW:	Q.764 RES The RES message is discarded and not mapped to SIP Timer is stopped in the gateway. For more details, ref. attached doc 1. Verify that the connection still is established.
3(	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.
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2.2.13 Bearer service speech, normal call release, suspend and resume initiated by a calling party

Note: A possible procedure to map the ISUP signalling procedure to SIP is described in draft RFC. However this procedure is not considered necessary.

2.2.14 Bearer service speech, normal call release, collision of REL messages

Test case ID:	SIP VoIP-ISUP-69
Test object:	Bearer service speech for ISUP signalling, normal call release, collisions of REL messages.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with collisions of REL messages, can be provided for a call from the SIP User to SS7 (ISUP), and that correct response is provided to the SIP User. See Q.784 test no.3.8.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test conditions:</u> The called termination is free.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
36	The SIP user calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SIP 100 (Trying) Q.764 IAM
37	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)

37	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established
37	SIP User disconnects and the ISUP simulator sends REL at the same time	SIP User: SIP GW: SIP GW: ISDN: SIP GW: ISDN: SIP GW: SIP User:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 REL Q.764 RLC Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the circuit is idle.
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2.2.15 Bearer service speech, unsuccessful call setup, validate a set of known causes for release

Test case ID:	SIP VoIP-ISUP-70
Test object:	Bearer service speech for ISUP signalling, unsuccessful call setup, validate a set of known causes for release.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with a set of known causes for release, can be provided for a call from the SIP User to SS7 (ISUP), and that correct response is provided to the SIP User. See Q.784 test no.4.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Arrange the data in the ISUP simulator such that a release message with a given cause is received and the correct indication is given to the calling party.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
37	The SIP user calls ISUP simulator	SIP User:	SIP INVITE Request
		SIP GW:	SIP 100 (Trying)
		SIP GW:	Q.764 IAM

37	ISUP simulator sends REL (cause=xxx)	ISDN: SIP GW: SIP GW: SIP User:	Q.764 REL (a) cause #1 (unallocated number), cause #3 (no route to destination) b) cause #16 (Normal call clearing) c) cause #18 (No user responding) d) cause #34 (No circuit available) e) cause #42 (switching equipment congestion) Q.764 RLC SIP a) 404 (Not Found) b) 603 (Decline) c) 408 (Request timeout) d) 503 (Service Unavailable) e) 503 (Service Unavailable) The total mapping of ISUP cause values/ SIP responses is included in the section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”. SIP 200 (OK) Verify that the circuit is idle.
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2.2.16 Bearer service speech, abnormal situation during a call, inability to release in response to a REL after ANM

Test case ID:	SIP VoIP-ISUP-71
Test object:	Bearer service speech for ISUP signalling, abnormal situation during a call, inability to release in response to a REL after ANM.
Test purpose:	The purpose of this test case is to verify that a correct response is provided from the SIP gateway to the SS7 (ISUP), when an abnormal situation occurs (failure to receive RLC in response to a REL after ANM) for a call from SIP to SS7 (ISUP). See Q.784 test no.5.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Arrange the data in the ISUP simulator such that it is unable to return the circuit to the idle condition in response to a release message.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
37	The SIP user calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SIP 100 (Trying) Q.764 IAM
37	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)
38	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established

3ε	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL The ISUP simulator is unable to send RLC.
3ε	SIP GW disconnects (second time)	SIP GW: ISDN:	Q.764 REL The ISUP simulator is unable to send RLC. Verify that the release message was sent between 4-15 seconds (T1) after sending of IAM.
3ε	SIP GW sends a reset circuit message	SIP GW:	RSC Verify that the reset circuit message was sent at 1 minute after sending of the IAM.
3ε	ISUP simulator sends a blocking message and alert the maintenance system	ISDN:	Q.764 BLO
3ε	SIP GW responds with blocking acknowledge	SIP GW:	Q.764 BLA Verify that the circuit is blocked
3ε	ISUP simulator sends a release complete message	ISDN:	Q.764 RLC Verify that the circuit is idle.
3ε			

2.2.17 Bearer service speech, abnormal situation during a call, timer T7: waiting for ACM or CON

Test case ID:	SIP VoIP-ISUP-72
Test object:	Bearer service speech for ISUP signalling, abnormal situation during a call, timer T7: waiting for ACM or CON.
Test purpose:	The purpose of this test case is to verify that a correct response is provided from the SIP gateway to the SS7 (ISUP), when an abnormal situation occurs (waiting for ACM or CON) for a call from SIP to SS7 (ISUP). See Q.784 test no.5.2.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Arrange the data in the ISUP simulator such that an address complete message is not returned to the call request.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
38	SIP User calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SIP 100 (Trying) Q.764 IAM The ISUP timer T7 is started at this point.

38	SIP Gateway disconnects (because ACM is not returned to the call request)	SIP GW: SIP GW: SIP User:	Q.764 REL SIP 504 (Gateway timeout) SIP ACK The ISUP timer T7 expires before receipt of an ACM or CON message, so a REL message is sent to cancel the call. Verify that the release message was sent after 20-30 seconds. Verify that the circuit is idle.
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2.2.18 Bearer service speech, abnormal situation during a call, timer T9: waiting for an answer message

Test case ID:	SIP VoIP-ISUP-73
Test object:	Bearer service speech for ISUP signalling, abnormal situation during a call, timer T9: waiting for an answer message.
Test purpose:	The purpose of this test case is to verify that a correct response is provided from the SIP gateway to the SS7 (ISUP), when an abnormal situation occurs (waiting for an answer message) for a call from SIP to SS7 (ISUP). See Q.784 test no.5.2.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: The called party should not answer the call.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
39	SIP User calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SIP 100 (Trying) Q.764 IAM
39	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm) The ISUP timer T9 is started at this point. The called party do not answer the call.

39	SIP GW disconnects (because the called party do not answer the call)	SIP GW: SIP GW: ISDN: SIP User:	Q.764 REL SIP 408 (Request timeout) Comment: The SIP response could also be 504, 480. Q.764 RLC SIP ACK Verify that the release message was sent within a period of T9. Verify that the circuit is idle.
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2.2.19 Bearer service speech, abnormal situation during a call, timer T6: waiting for RES (Network) message

Test case ID:	SIP VoIP-ISUP-74
Test object:	Bearer service speech for ISUP signalling, abnormal situation during a call, timer T6: waiting for RES (Network) message.
Test purpose:	The purpose of this test case is to verify that a correct response is provided from SIP gateway to the SS7 (ISUP), when an abnormal situation occurs (waiting for RES message) for a call from SIP to SS7 (ISUP). See Q.784 test no.5.2.4.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test conditions:</u> Arrange the data in the ISUP simulator such that it is unable to return a resume message (called party will not re-answer).
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
4(	The SIP user calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SIP 100 (Trying) Q.764 IAM
4(	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK Verify that the ringing tone can be heard. SIP 200 OK (PRACK confirm)

4(	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established
4(	ISUP simulator sends a SUS (Network) message	ISDN: SIP GW:	Q.764 SUS The SUS message is discarded and not mapped to SIP. Comment: A possible procedure to map the ISUP signalling procedure to SIP is described in draft RFC. However this procedure is not considered necessary. The ISUP timer T6 is started at this point. The ISUP simulator will not return a RES message. Timer T6 expires
4(	SIP GW disconnects (because the ISUP simulator does not return a RES message)	SIP GW: SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) SIP ACK Q.764 REL Q.764 RLC Verify that the release message was sent within a period of T6 timer. Verify that the circuit is idle.
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2.2.20 Bearer service speech, abnormal situation, timer T12 and T13: failure to receive a BLA

Test case ID:	SIP VoIP-ISUP-75
Test object:	Bearer service speech for ISUP signalling, abnormal situation, timer T12 and T13: failure to receive a BLA.
Test purpose:	The purpose of this test case is to verify that a correct response is provided from SIP gateway to the SS7 (ISUP), when an abnormal situation occurs (failure to receive BLA) for a call from SIP to SS7 (ISUP). See Q.784 test no.5.2.6.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: • The circuit is idle • Arrange the data in the ISUP simulator such that a blocking acknowledgement message is not returned in response to a blocking message.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
41	SIP GW sends a blocking message	SIP GW: ISDN:	Q.764 BLO The ISUP simulator fails to respond with BLA.
41	SIP GW sends a blocking message (second time)	SIP GW:	Q.764 BLO Verify that a blocking message was sent between 4-15 seconds (T12) after sending of the initial blocking message.

41	SIP GW sends a blocking message (third time), and alerts the maintenance system	SIP GW:	Q.764 BLO Verify that a blocking message was sent at 1 minute (T13) after sending of the initial blocking message, and that an alarm is set in the SIP GW.

2.2.21 Bearer service speech, abnormal situation, timer T14 and T15: failure to receive a UBA

Test case ID:	SIP VoIP-ISUP-76
Test object:	Bearer service speech for ISUP signalling, abnormal situation, timer T14 and T15: failure to receive a UBA.
Test purpose:	The purpose of this test case is to verify that a correct response is provided from SIP gateway to the SS7 (ISUP), when an abnormal situation occurs (failure to receive UBA) for a call from SIP to SS7 (ISUP). See Q.784 test no.5.2.7.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: • The circuit is idle • Arrange the data in the ISUP simulator such that an unblocking acknowledgement message is not returned in response to an unblocking message.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**
41	SIP GW sends a blocking message	Q.764 BLO
41	ISUP simulator responds with a blocking acknowledge message	Q.764 BLA
41	SIP GW sends an unblocking message	Q.764 UBL The ISUP simulator fails to respond with UBA.
41	SIP GW sends an unblocking message (second time)	Q.764 UBL Verify that an unblocking message was sent between 4-15 seconds (T14) after sending of the initial unblocking message.

41	SIP GW sends an unblocking message (third time), and alerts the maintenance system	Q.764 UBL Verify that an unblocking message was sent between 4-15 seconds (T14) after sending of the initial unblocking message, and that an alarm is set in the SIP GW.

2.2.22 Bearer service speech, abnormal situation during a call, timer T16 and T17: failure to receive a RLC

Test case ID:	SIP VoIP-ISUP-77
Test object:	Bearer service speech for ISUP signalling, abnormal situation during a call, timer T16 and T17: failure to receive a RLC.
Test purpose:	The purpose of this test case is to verify that a correct response is provided from SIP gateway to the SS7 (ISUP), when an abnormal situation occurs (failure to receive RLC) for a call from SIP to SS7 (ISUP). See Q.784 test no.5.2.8.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: • The circuit is idle • Arrange the data in the ISUP simulator such that a release complete message is not returned in response to a reset circuit message.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
41	The SIP user calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SIP 100 (Trying) Q.764 IAM
41	ISUP simulator responds with an unexpected message (ex. SUS)	ISDN:	Q.764 SUS

42	SIP GW sends a reset circuit message (because of the unexpected message received from ISDN)	SIP GW: ISDN: SIP GW: SIP User:	Q.764 RSC The ISUP simulator fails to respond with a RLC message. SIP 408 (Request timeout) Comment: The SIP response could also be 504, 480. SIP ACK
42	SIP GW sends a reset circuit message (second time) (because the ISUP simulator fails to respond with a RLC message)	SIP GW:	Q.764 RSC Verify that a reset circuit message was sent between 4-15 seconds (T16) after sending of the initial reset circuit message.
42	SIP GW sends a reset circuit message (third time), and alerts the maintenance system	SIP GW:	Q.764 RSC Verify that a reset circuit message was sent at 1 minute (T17) after sending of the initial reset circuit message.
42			

2.2.23 Bearer service speech, abnormal situation, timer T18 and T19: failure to receive a CGBA

Test case ID:	SIP VoIP-ISUP-78
Test object:	Bearer service speech for ISUP signalling, abnormal situation, timer T18 and T19: failure to receive a CGBA.
Test purpose:	The purpose of this test case is to verify that a correct response is provided from SIP gateway to the SS7 (ISUP), when an abnormal situation occurs (failure to receive CGBA) for a call from SIP to SS7 (ISUP). See Q.784 test no.5.2.9.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test conditions:</u> • The circuit is idle • Arrange the data in the ISUP simulator such that a circuit group blocking acknowledgement message is not returned in response to a circuit group blocking message.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
42	SIP GW sends a circuit group blocking message	SIP GW: ISDN:	Q.764 CGB ISUP simulator fails to respond with a CGBA.

42	SIP GW sends a circuit group blocking message (second time) (because the ISUP simulator fails to respond with a CGBA)	SIP GW:	Q.764 CGB Verify that a circuit group blocking message was sent between 4-15 seconds (T18) after sending of the initial circuit group blocking message.
42	SIP GW sends a circuit group blocking message (third time), and alerts the maintenance system	SIP GW:	Q.764 CGB Verify that a circuit group blocking message was sent at 1 minute (T19) after sending of the initial circuit group blocking message, and that an alarm is set in the SIP GW.
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2.2.24 Bearer service speech, abnormal situation, timer T20 and T21: failure to receive a CGUA

Test case ID:	SIP VoIP-ISUP-79
Test object:	Bearer service speech for ISUP signalling, abnormal situation, timer T20 and T21: failure to receive a CGUA.
Test purpose:	The purpose of this test case is to verify that a correct response is provided from SIP gateway to the SS7 (ISUP), when an abnormal situation occurs (failure to receive CGUA) for a call from SIP to SS7 (ISUP). See Q.784 test no.5.2.10.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: • The circuit is idle • Arrange the data in the ISUP simulator such that a circuit group unblocking acknowledgement message is not returned in response to a circuit group unblocking message.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
42	SIP GW sends a circuit group blocking message	SIP GW:	Q.764 CGB
42	ISUP simulator responds with a circuit group acknowledge message	ISDN:	Q.764 CGBA
43	SIP GW sends a circuit group unblocking message	SIP GW: ISDN:	Q.764 CGU The ISUP simulator fails to respond with a CGUA message.

43	SIP GW sends a circuit group unblocking message (second time) (because the ISUP simulator fails to respond with a CGUA message)	SIP GW:	Q.764 CGU Verify that a circuit group unblocking message was sent between 4-15 seconds (T20) after sending of the initial circuit group unblocking message.
43	SIP GW sends a circuit group unblocking message (third time), and alerts the maintenance system	SIP GW:	Q.764 CGU Verify that a circuit group unblocking message was sent at 1 minute (T21) after sending of the initial circuit group unblocking message, and that an alarm is set in the SIP GW.
43			

2.2.25 Bearer service speech, abnormal situation during a call, reset of circuits during a call

Test case ID:	SIP VoIP-ISUP-80
Test object:	Bearer service speech for ISUP signalling, abnormal situation during a call, reset of circuits during a call.
Test purpose:	The purpose of this test case is to verify that a correct response is provided from SIP gateway to the SS7 (ISUP), when an abnormal situation occurs (reset of circuits during a call) for a call from SIP to SS7 (ISUP). See Q.784 test no.5.3.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test conditions:</u> Called termination is free.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
43	The SIP user calls ISUP simulator	SIP User:	SIP INVITE Request
		SIP GW:	SIP 100 (Trying)
		SIP GW:	Q.764 IAM
43	ISUP simulator sends ACM	ISDN:	Q.764 ACM
		SIP GW:	SIP 180 (Ringing)
		SIP User:	SIP PRACK
			Verify that the ringing tone can be heard.
		SIP GW:	SIP 200 OK (PRACK confirm)

43	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established
43	ISUP simulator sends a reset-circuit message	ISDN:	Q.764 RSC
43	SIP GW responds with release complete	SIP GW: SIP GW: SIP User:	Q.764 RLC SIP BYE SIP 200 OK (BYE confirm) Verify that the circuit is idle.

2.2.26 Bearer service speech, special call setup, automatic repeat attempt – blocking of a circuit

Test case ID:	SIP VoIP-ISUP-81
Test object:	Bearer service speech for ISUP signalling, special call setup, automatic repeat attempt, blocking of a circuit.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with automatic repeat attempt can be provided for a call from the SIP User to SS7 (ISUP), and that correct response is provided to the SIP User. See Q.784 test no.6.2.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Arrange the data in the ISUP simulator such that a blocking message is returned in response to the initial address message of the first call request.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
43	The SIP user calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SIP 100 (Trying) Q.764 IAM (cic=x)
44	ISUP simulator sends a blocking message and alert the maintenance system	ISDN:	Q.764 BLO (cic=x)
44	SIP GW responds with blocking acknowledge	SIP GW:	Q.764 BLA (cic=x) Verify that the circuit is blocked

4⁄	The SIP GW calls ISUP simulator (automatic repeat attempt)	SIP GW:	Q.764 IAM (cic=y) Verify that an automatic repeat attempt is made on receipt of the blocking message after sending of an IAM and before any backward messages have been received.
4⁄	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM (cic=y) SIP 180 (Ringing) SIP PRACK Verify that the ringing tone can be heard on the call originated from the SIP User. SIP 200 OK (PRACK confirm)
4⁄	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM (cic=y) SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established
4⁄	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL (cic=y) Q.764 RLC Verify that the circuit (cic=y) is idle.
4⁄			

2.2.27 Bearer service speech, special call setup, automatic repeat attempt – circuit reset

Test case ID:	SIP VoIP-ISUP-82
Test object:	Bearer service speech for ISUP signalling, special call setup, automatic repeat attempt, circuit reset.
Test purpose:	The purpose of this test case is to verify that the speech bearer service with automatic repeat attempt can be provided for a call from the SIP User to SS7 (ISUP), and that correct response is provided to the SIP User. See Q.784 test no.6.2.3.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: • Arrange the data in the ISUP simulator such that a circuit reset message is sent in response to the initial address message of the first call request. • The called termination shall be free.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
44	The SIP user calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SIP 100 (Trying) Q.764 IAM (cic=x)
44	ISUP simulator sends a reset circuit message to the SIP access	ISDN:	Q.764 RSC (cic=x)
44	SIP GW disconnects	SIP GW:	Q.764 RLC (cic=x) The circuit shall be idle

4ξ	The SIP GW calls ISUP simulator (automatic repeat attempt)	SIP GW:	Q.764 IAM (cic=y) Verify that an automatic repeat attempt is made on receipt of the circuit reset after sending of an IAM and before a backward message have been received.
4ξ	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM (cic=y) SIP 180 (Ringing) SIP PRACK Verify that the ringing tone can be heard on the call originated from the SIP User. SIP 200 OK (PRACK confirm)
4ξ	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM (cic=y) SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established
4ξ	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL (cic=y) Q.764 RLC Verify that the circuit (cic=y) is idle.
4ξ			

2.2.28 64 kbit/s unrestricted bearer service, successful call setup

Test case ID:	SIP VoIP-ISUP-83
Test object:	64 kbit/s unrestricted bearer service for ISUP signalling, successful call setup.
Test purpose:	The purpose of this test case is to verify that the 64 kbit/s unrestricted bearer service can be provided for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User. See Q.784 test no.7.1.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Called termination is free.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.
Test Procedure Steps:	**Expected Results:**

45	The SIP user calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SDP=H.221 or H.263 SIP 100 (Trying) Q.764 IAM (TMR, USI) Verify • that the TMR is set to ”64 kbit/s unrestricted” • that the USI (if included) have appropritate information. For example: USI has two octets for 64 kbit/s and at least four octets for any subrate • that the “echo control device indicator” in nature of connection indicators parameter is set to “not included” • that the echo control device is disabled or a non-echo controlled circuit is selected
45	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM (subscriber free) SIP 180 (Ringing) SIP PRACK SIP 200 OK (PRACK confirm)
45	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established Verify that it is possible to pass data between the SIP User and the ISUP simulator.
45	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.
45			
46			

2.2.29 64 kbit/s unrestricted bearer service, unsuccessful call setup

Test case ID:	SIP VoIP-ISUP-84
Test object:	64 kbit/s unrestricted bearer service for ISUP signalling, unsuccessful call setup.
Test purpose:	The purpose of this test case is to verify that the 64 kbit/s unrestricted bearer service can be provided for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User. See Q.784 test no.7.1.2.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Arrange the data in the ISUP simulator such that a release message with a given cause is returned to the request. The suggested causes are: - unallocated number - no circuit available - bearer capability not authorized - bearer capability not presently available - bearer capability not implemented
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 "SIP-ISUP interworking for Telenor national interconnect ISUP version 2".

Test Procedure Steps:		**Expected Results:**	
46	The SIP user calls ISUP simulator	SIP User:	SIP INVITE Request
		SIP GW:	SIP 100 (Trying)
		SIP GW:	Q.764 IAM

46	ISUP simulator sends REL (cause=xxx)	ISDN:	Q.764 REL (a) cause #1 b) cause #34 c) cause #57 d) cause #58 e) cause #65)
		SIP GW:	Q.764 RLC
		SIP GW:	SIP a) 404 (Not Found) b) 503 (Service Unavailable) c) 403 (Forbidden) d) 503 (Service Unavailable) e) 501 (Not implemented)
		SIP User:	SIP ACK Verify that the circuit is idle. For circuits equipped with echo control the echo control device is re-enabled.

2.2.30 Multi-use bearer service supported for 7 kHz telephony, successful call setup with and without fallback to speech

Test case ID:	SIP VoIP-ISUP-85
Test object:	Multi-use bearer service supported for 7 kHz telephony, successful call setup with and without fallback to speech.
Test purpose:	The purpose of this test case is to verify that a call with the multi-use bearer service supported for 7 kHz telephony is properly supported by the SIP/SS7 Gateway, and successfully completed with and without fallback to speech for a call from the SIP User to SS7 (ISUP).
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
46	The SIP user calls ISUP simulator	SIP User:	SIP INVITE Request SDP= G.722 included and G.711 or G.723.1 (G.728, G.729) included
		SIP GW:	SIP 100 (Trying)
		SIP GW:	Q.764 IAM (• TMR= “64 kbit/s, unrestricted preferred” • TMR prime= “speech” • USI = “speech” • USI prime= “UDI-TA” • HLC in ATP= “telephony”)

46	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK SIP 200 OK (PRACK confirm)
46	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM (a) TMU= ”speech” b) BC in ATP= ”UDI-TA”) SIP 200 OK (INVITE confirm) a) SDP = G.711 or G.723.1 (G.728, G.729) b) SDP=G.722 For more details, ref. the attached doc.1 SIP ACK Verify that the connection is established
46	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC Verify that the circuit is idle.
46			
46			
46			

2.2.31 Public supplementary service CLIP from SIP User, calling address is national

Test case ID:	SIP VoIP-ISUP-86
Test object:	Supplementary service CLIP (Called line identification presentation) for ISUP signalling, calling address national.
Test purpose:	The purpose of this test case is to verify that the CLIP supplementary service can be provided for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User. See Q.785 test no.3.1.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Arrange the data in the SIP terminal such that the calling address in the INVITE contains a calling address which will be authenticated and can be mapped to ISUP by the SIP GW as ✓ special arrangement (i.e. mapped to generic number parameter in ISUP) ✓ user provided, not verified ✓ national (significant) number ✓ E.164 number
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.
Test Procedure Steps:	**Expected Results:**

47	The SIP user calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Calling address in the From-field: Sip: +47<cgnadd>@host.domain. The calling address comes from a reliable source (has been authenticated). SIP 100 (Trying) Q.764 IAM (Generic number parameter) Generic number parameter: • Number qualifier=additional calling party number • Address presentation restricted indicator=presentation allowed • Screening indicator=user provided, nor verified • NoA=national (significant) number • Address signal=<cgnadd> Calling party number parameter is not included in the IAM.
47	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK SIP 200 OK (PRACK confirm)
47	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established.
47	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC

2.2.32 Public supplementary service CLIP from SIP User, calling address is international

Test case ID:	SIP VoIP-ISUP-87
Test object:	Supplementary service CLIP (Called line identification presentation) for ISUP signalling, calling address international.
Test purpose:	The purpose of this test case is to verify that the CLIP supplementary service can be provided for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User. See Q.785 test no.3.1.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Arrange the data in the SIP terminal such that the calling address in the INVITE contains a calling address which will be authenticated and can be mapped to ISUP by the SIP GW as ✓ special arrangement (i.e. mapped to generic number parameter in ISUP) ✓ user provided, not verified ✓ international number ✓ E.164 number
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:	**Expected Results:**

47	The SIP user calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Calling address in the From-field: Sip: +CC<cgnadd>@host.domain. The calling address comes from a reliable source (has been authenticated). CC is different from 47. SIP 100 (Trying) Q.764 IAM (Generic number parameter) Generic number parameter: • Number qualifier=additional calling party number • Address presentation restricted indicator=presentation allowed • Screening indicator=user provided, not verified • NoA=international number • Address signal=CC<cgnadd> Calling party number parameter is not included in the IAM.
47	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK SIP 200 OK (PRACK confirm)
47	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established.
47	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC

2.2.33 Public supplementary service CLIP from SIP User, calling address is not authenticated

Test case ID:	SIP VoIP-ISUP-88
Test object:	Supplementary service CLIP (Called line identification presentation) for ISUP signalling, the calling address from the SIP User is not authenticated.
Test purpose:	The purpose of this test case is to verify that the CLIP supplementary service can be provided for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User. See Q.785 test no.3.1.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Arrange the data in the SIP terminal such that the INVITE contains a calling address which has not been authenticated.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
47	The SIP user calls ISUP simulator	SIP User:	SIP INVITE Calling address in the From-field: Sip:+47<cgnadd>@host.domain. The calling address has not been authenticated).
		SIP GW:	SIP 100 (Trying)
		SIP GW:	Q.764 IAM (no information) Neither the Generic number parameter nor the Calling party number parameter is included in the IAM.

47	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK SIP 200 OK (PRACK confirm)
48	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established.
48	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC

2.2.34 Public supplementary service CLIP from SIP User, from IP VPN

<table>
<tr><td>Test case ID:</td><td colspan="2">SIP VoIP-ISUP-86</td></tr>
<tr><td>Test object:</td><td colspan="2">Supplementary service CLIP (Called line identification presentation) for ISUP signalling, calling address national.</td></tr>
<tr><td>Test purpose:</td><td colspan="2">The purpose of this test case is to verify that the CLIP supplementary service can be provided for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User.
See Q.785 test no.3.1.1.</td></tr>
<tr><td>Test case dependencies:</td><td colspan="2">Infrastructure in place.</td></tr>
<tr><td>Test Set-up:</td><td colspan="2">ISUP simulator towards the SIP/SS7 GW.
Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
<u>Pre-test conditions:</u>
Arrange the data in the SIP-ISUP GW terminal such that the INVITE comes from an IP telephony application in an IP VPN directly connected to the gateway and dedicated to a specific customer (company).</td></tr>
<tr><td>Post-test actions:</td><td colspan="2">Analyse the signalling procedures. Verify that they are correct according to
• RFC 2543(SIP)
• Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification)
• The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.</td></tr>
<tr><td colspan="2">Test Procedure Steps:</td><td>Expected Results:</td></tr>
</table>

48	The SIP user calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Calling address in the From-field: Sip: +47<cgn>@host.domain. SIP 100 (Trying) Q.764 IAM (Calling party number parameter) The SIP GW screens the received calling address, and the mapping depends upon the result of the screening. a) The calling party number belongs to the range allocated to the subscriber (company): • Address presentation restricted indicator=presentation allowed • Screening indicator=user provided, verified and passed • NoA=national (significant) number • Address signal=<cgn> b) The calling party number does not belong to the range allocated to the subscriber (company): • Address presentation restricted indicator=presentation allowed • Screening indicator=network provided • NoA=national (significant) number • Address signal=<default number allocated to the subscriber access> Generic number parameter is not included in the IAM.
48	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK SIP 200 OK (PRACK confirm)

4ℇ	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established.
4ℇ	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC

2.2.35 Public supplementary service CLIR from SIP User

Test case ID:	SIP VoIP-ISUP-89
Test object:	Supplementary service CLIR (Called line identification restriction) for ISUP signalling.
Test purpose:	The purpose of this test case is to verify that the CLIR supplementary service can be provided for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User. See Q.785 test no.3.3.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. <u>Pre-test conditions:</u> Arrange the data at the SIP terminal such that the INVITE contains an anonymous calling address.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
48	The SIP user calls ISUP simulator	SIP User:	SIP INVITE Calling address in the From field: Anonymous<sip:restricted>@host.domain
		SIP GW:	SIP 100 (Trying)
		SIP GW:	Q.764 IAM Neither the Calling party number parameter nor the Generic number parameter (additional calling party number) is included.

4ξ	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK SIP 200 OK (PRACK confirm)
4ξ	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established.
4ξ	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC

2.2.36 Public supplementary service CLIR from SIP User, from IP VPN

Test case ID:	SIP VoIP-ISUP-86
Test object:	Supplementary service CLIR (Called line identification restriction) for ISUP signalling, calling address national.
Test purpose:	The purpose of this test case is to verify that the CLIR supplementary service can be provided for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User. See Q.785 test no.3.1.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Arrange the data in the SIP-ISUP GW terminal such that the INVITE comes from an IP telephony application in an IP VPN directly connected to the gateway and dedicated to a specific customer (company).
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:	**Expected Results:**

49	The SIP user calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Calling address in the From-field: Anonymous <sip:restricted>@host.domain SIP 100 (Trying) Q.764 IAM Calling party number parameter: • Address presentation restricted indicator=presentation restricted • Screening indicator=network provided • NoA=national (significant) number • Address signal=<default number allocated to the subscriber access> Generic number parameter is not included in the IAM.
49	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK SIP 200 OK (PRACK confirm)
49	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established.
49	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC

2.2.37 Public supplementary service COLP from SIP User, request

Test case ID:	SIP VoIP-ISUP-90
Test object:	Supplementary service COLP (Connected line identification presentation) for ISUP signalling, request.
Test purpose:	The purpose of this test case is to verify that the COLP supplementary service can be rejected for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User. See Q.785 test no.6.1.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Arrange the data in the ISUP simulator such that the ANM contains the connected number parameter coded as follows: ✓ NoA= National(significant) number ✓ Screening indicator= Network provided ✓ Address presentation restricted indicator = Presentation allowed
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
49	SIP User calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE Request SIP 100 (Trying) Q.764 IAM The connected number is not requested in the IAM (i.e. the optional forward call indicators is not included)

49	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK SIP 200 OK (PRACK confirm)
49	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM (with connected number parameter included coded "presentation allowed") SIP 200 OK (INVITE confirm) The Connected number is discarded and not mapped to SIP if received in ANM. SIP ACK Verify that the connection is established.
49	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC

2.2.38 Public supplementary service COLR from SIP User, network provided

Test case ID:	SIP VoIP-ISUP-91
Test object:	Supplementary service COLR(Connected line identification restriction) for ISUP signalling, network provided.
Test purpose:	The purpose of this test case is to verify that the COLR supplementary service can be provided for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SIP User. See Q.785 test no.6.4.1.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals. Pre-test conditions: Arrange the data in the ISUP simulator such that the ANM contains the connected number parameter coded as follows: ✓ NoA= National(significant) number ✓ Screening indicator= Network provided ✓ Address presentation restricted indicator = Presentation restricted
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
4ς	SIP User calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE SIP 100 (Trying) Q.764 IAM The connected number is not requested in the IAM (i.e. the optional forward call indicators is not included)

4(	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK SIP 200 OK (PRACK confirm)
5(	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM (with connected number parameter included coded "presentation restricted") SIP 200 OK (INVITE confirm) The Connected number is discarded and not mapped to SIP if received in ANM. SIP ACK Verify that the connection is established.
5(	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC

2.2.39 Public supplementary service Diversion, forwarded call from SIP

Test case ID:	SIP VoIP-ISUP-92
Test object:	Supplementary service Diversion for ISUP signalling, forwarded call from SIP.
Test purpose:	The purpose of this test case is to verify the following case: for a forwarded call from SIP to ISDN, there will be no indication to ISUP that the call has been forwarded.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
5(	SIP User calls ISUP simulator with forwarded call	SIP User:	SIP INVITE sip:”user C” -> Called party number in ISUP From : sip:”user A” -> additional calling party number in ISUP To : sip:”user B” -> not included in ISUP
		SIP GW:	SIP 100 (Trying)
		SIP GW:	Q.764 IAM There will be no indication to ISUP that the call has been redirected.

5(	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK SIP 200 OK (PRACK confirm)
5(	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established.
5(	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC
5(			
5(			

2.2.40 Public supplementary service Diversion, call to be forwarded by ISDN

Test case ID:	SIP VoIP-ISUP-93
Test object:	Supplementary service Diversion for ISUP signalling.
Test purpose:	The purpose of this test case is to verify the following case: for a call from SIP to a forwarded ISDN number, there will be no indication to SIP that the called number has been forwarded.
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
5(	SIP User calls ISUP simulator (User B)	SIP User: SIP GW: SIP GW:	SIP INVITE SIP 100 (Trying) Q.764 IAM (User B) The called party has been diverted within ISDN (from User B to User C).

5(	User B (ISUP simulator) sends ACM and indicates CFU	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM (no indication) • optional backward call indicator=”call diversion may occur” • redirection number • call diversion information parameter • redirection number restriction parameter SIP 181 (Call is being forwarded) SIP PRACK SIP 200 OK (PRACK confirm)
51	User C (ISUP simulator) sends CPG (alerting)	ISDN: SIP GW: SIP User: SIP GW:	Q.764 CPG (alerting) SIP 180 (Ringing) SIP PRACK SIP 200 OK (PRACK confirm)
51	User C (ISUP simulator) answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established.
51	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC
51			
51			

2.2.41 Public supplementary services TP from non SIP User

Test case ID:	SIP VoIP-ISUP-94
Test object:	Supplementary services TP (Terminal Portability), notification for ISUP signalling.
Test purpose:	The purpose of this test case is to verify that notification for public TP can be included in ISUP for a call from the SIP User to SS7 (ISUP), that the SIP signalling procedures are correct, and that correct response is provided to the SS7 (ISUP).
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
51	SIP User calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE SIP 100 (Trying) Q.764 IAM
51	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK SIP 200 OK (PRACK confirm)
51	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established.

51	Suspend/Resume (ISDN User initiated)	ISDN: SIP GW: ISDN: SIP GW:	Q.764 SUS (user initiated) SUS (user initiated) is discarded and not mapped to SIP. Q.764 RES (user initiated) RES (user initiated) is discarded and not mapped to SIP. Comment: A possible procedure to map the ISUP signalling procedure to SIP is described in draft RFC. However this procedure is not considered necessary.
51	SIP User disconnects	SIP: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC

2.2.42 Public supplementary services HOLD from non SIP User

Test case ID:	SIP VoIP-ISUP-95
Test object:	Supplementary services HOLD (Call Hold), notification for ISUP signalling.
Test purpose:	The purpose of this test case is to verify that notification for public HOLD can be included in ISUP for a call from the SIP User to SS7 (ISUP), that the info is discarded by the SIP gateway, and that correct response is provided to the SS7 (ISUP).
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
52	SIP User calls the ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE SIP 100 (Trying) Q.764 IAM
52	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK Verify that the ringing tone can be heard on the call originated from the SIP User. SIP 200 OK (PRACK confirm)

52	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established.
52	HOLD Notification	ISDN: SIP GW:	Q.764 CPG (Generic notification indicator: HOLD notification) The Generic notification parameter is discarded and not mapped to SIP.
52	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC

2.2.43 Public supplementary services CONF/3PTY from non SIP User

Test case ID:	SIP VoIP-ISUP-96
Test object:	Supplementary services CONF/3PTY (Conference calling/Three-party service), notification for ISUP signalling.
Test purpose:	The purpose of this test case is to verify that notification for public CONF/3PTY can be included in ISUP for a call from the SIP User to SS7 (ISUP), that the info is discarded by the SIP gateway, and that correct response is provided to the SS7 (ISUP).
Test case dependencies:	Infrastructure in place.
Test Set-up:	ISUP simulator towards the SIP/SS7 GW. Instrument with SIP software monitoring the signalling is connected to the local Ethernet near the terminals.
Post-test actions:	Analyse the signalling procedures. Verify that they are correct according to • RFC 2543(SIP) • Telenor national interconnect ISUP version 2 (see standard Telenor Norwegian national interconnect interface specification) • The section 4 “SIP-ISUP interworking for Telenor national interconnect ISUP version 2”.

Test Procedure Steps:		**Expected Results:**	
52	SIP User calls ISUP simulator	SIP User: SIP GW: SIP GW:	SIP INVITE SIP 100 (Trying) Q.764 IAM
52	ISUP simulator sends ACM	ISDN: SIP GW: SIP User: SIP GW:	Q.764 ACM SIP 180 (Ringing) SIP PRACK Verify that the ringing tone can be heard on the call originated from the SIP User. SIP 200 OK (PRACK confirm)

5‹	ISUP simulator answers	ISDN: SIP GW: SIP User:	Q.764 ANM SIP 200 OK (INVITE confirm) SIP ACK Verify that the connection is established.
5‹	CONF/3PTY Notification	ISDN: SIP GW:	Q.764 CPG (Generic notification indicator: CONF/3PTY notification) The generic notification parameter is discarded and not mapped to SIP.
5‹	SIP User disconnects	SIP User: SIP GW: SIP GW: ISDN:	SIP BYE SIP 200 OK (BYE confirm) Q.764 REL Q.764 RLC

3. TEST RESULTS

System which has been tested:	
Network configuration:	
Tester:	
Date:	
Comments:	

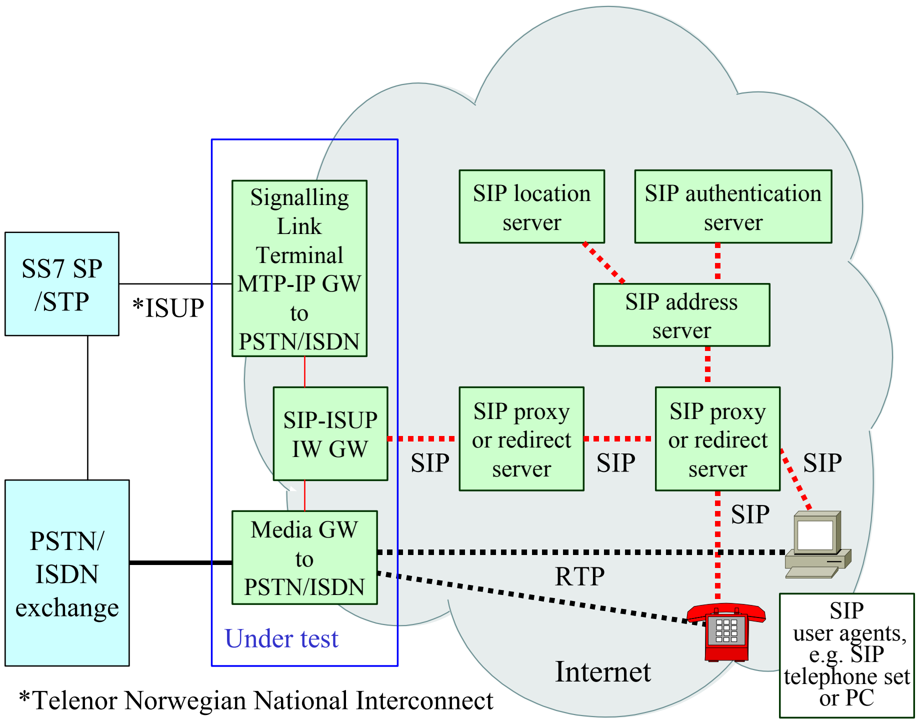
Test case results:

Test case ID	**Test case result (OK/NOK/NA/Not Performed)**	**Comments**

4. SIP-ISUP INTERWORKING FOR TELENOR NATIONAL INTERCONNECT ISUP VERSION 2

The SIP-SS7 gateway functionality will depend upon the network configuration. The requirements will be significantly different between the case when SIP is used as an access network, and when SIP is used on a network internal (NNI) interface within the PSTN/ISDN (the “bridging case”).

The SIP network specified here is considered to be an access network (SIP-T is not considered relevant). The SIP network is also considered as an international network. The SIP/SS7 gateway tested by Telenor FSN will be connected with Telenor Norwegian national interconnect ISUP version 2 on the SS7 side. As a consequence the SIP/SS7 gateway needs to have national interconnect gateway functionality, international gateway functionality, and “ISDN local exchange” type functionality.



*Telenor Norwegian National Interconnect ISUP, based on ETS 300 646-1 (1997)

Figure: Network Configuration

Note that the Internet is considered as an international access network for SIP calls, regardless of whether or not there are one or more SIP proxies between the SIP/SS7 gateway and the SIP user terminal.

The protocol requirements of the SIP/SS7 gateway are implicitly specified in detail by the Telenor national interconnect specification for the ISUP side, see attached specification. Note that the Telenor national interconnect ISUP version 2 is almost identical to the ETSI standard ETS 300 646-1 (1997) for national interconnect ISUP.

Note in particular that correct handling of the Calling party number, Generic number, Redirecting number and Redirection information parameters in ISUP are critically important because of Norwegian regulatory requirements.

Because the Norwegian interconnect specification is written as an application document to a number of ITU-T and ETSI standards, and interworking to SIP is not explicitly treated, it could be difficult to interpret the documentation and deduce the exact requirements in each case. The mapping tables below are provided to make such an interpretation easier.

For some cases, the available specifications do not indicate a specific mapping, and then some reasonable solutions are proposed. For some procedures and services with low priority, a simplified response is proposed.

In order to fulfil the protocol requirements on the SS7 side, the following coding and procedures will be desirable.

4.1 Call from ISUP to SIP

4.1.1 Basic call

Nature of connection indicators

Input	Response
IAM from PSTN/ISDN with a legal value of Nature of connection indicators	SIP INVITE, no specific procedure for basic call depending upon the value of the Nature of connection indicators

Forward call indicators

Input	Response
IAM from PSTN/ISDN with a legal value of the Forward call indicators	SIP INVITE, no specific procedure for basic call depending upon the value of the Forward call indicators

Calling party's category

Input	Response
IAM from PSTN/ISDN with a legal value of Calling party's category	SIP INVITE, no specific procedure for basic call depending upon the value of the Calling party's category

Transmission medium requirement

Input	Response
IAM from PSTN/ISDN with TMR value: speech or 3.1 kHz audio	SIP INVITE with media codings allowed: G.711, G.723.1, (G.728, G.729). G.722 not allowed. Echo canceller included in the gateway (cancelling echo from the PSTN/ISDN side, with two-way end delay of 60 ms).
IAM from PSTN/ISDN with TMR value: 64 kbit/s unrestricted	It is assumed that 64 kbit/s unrestricted interconnect with SIP will only be used for video telephony: SIP INVITE with media coding H.221/H.263 allowed (depending upon the contents of the received USI parameter). Echo canceller is not included by the gateway (i.e. "G.711 clear coding"). Gateway or MCU with bonding capability is outside the scope of this specification, only mapping of single 64 kbit/s channels is considered.
Support of multi-use bearer service (IAM from PSTN/ISDN with TMR value= 64 kbit/s unrestricted preferred). 3 possible options are identified: a) Fallback generated by the PSTN/ISDN gateway, TMR value= 64 kbit/s unrestricted preferred is not passed to the SIP/SS7 GW, but changed to speech. b) Multi-use bearer service properly rejected by the SIP/SS7 GW. IAM from	a) See response for TMR= speech. b) The call is continued on the SIP side as a speech call, see response for TMR= speech on the SIP side. TMU = speech is sent by the gateway to the PSTN/ISDN. c) SIP INVITE with media coding allowed: G.722, G.711, G.723.1, (G.728, G.729). - If the destination terminal indicates G.722 as the desired coding, this is used. BC= UDI-TA and HLC= telephony is included by the gateway in ATP in ANM to PSTN/ISDN. Echo canceller is not included by the gateway. - If the destination terminal indicates G.711, G.723.1, (G.728, G.729) as the desired coding, this is used. BC= speech and HLC= telephony in ATP and TMU= speech is included by the gateway in ANM to PSTN/ISDN. Echo

PSTN/ISDN with TMR value: 64 kbit/s unrestricted preferred, TMR prime value= speech. c) Multi-use bearer service supported for 7 kHz telephony (recommended solution). IAM from PSTN/ISDN with TMR value: 64 kbit/s unrestricted preferred, TMR prime value= speech.	canceller is included by the gateway.

Called party number

Input	Response
IAM from PSTN/ISDN with nature of address= national(significant)number	SIP INVITE, called address: sip:+47<cpn>@host.domain;
IAM from PSTN/ISDN with nature of address= international number	SIP INVITE, called address: sip:+<cpn>@host.domain;
IAM from PSTN/ISDN with nature of address= ported number, address signal= pfx,cpn	SIP INVITE, called address: sip:+47<cpn>@host.domain;

Optional forward call indicators

Input	Response
IAM from PSTN/ISDN with Simple segmentation indicator	See Q.764 (1993), section 2.1.12. However, support of the simple segmentation procedure is not considered to be a priority for this release of the SIP/SS7 GW. No other specific procedure for basic call.
IAM from PSTN/ISDN with legal values of Optional forward call indicators	SIP INVITE, except for closed user group call, outgoing access not allowed (see CUG supplementary service). No specific procedure for basic call.

Propagation delay counter

Input	Response
IAM from PSTN/ISDN with X ms value of the propagation delay counter	SIP INVITE. Call history value X+150 ms is included by the gateway in ANM to the PSTN/ISDN. However, support of the propagation delay determination procedure is not considered to be a priority for this release of the SIP/SS7 GW.

Other basic call related parameters

Input	
IAM from PSTN/ISDN with Access Transport	See procedures for multi-use bearer service. No other specific procedure for basic call is required.
IAM from PSTN/ISDN with Connection request	Not supported, request ignored.
IAM from PSTN/ISDN with User service information	See procedures for multi-use bearer service. No other specific procedure for basic call is required.
IAM from PSTN/ISDN with User service information prime	See procedures for multi-use bearer service. No other specific procedure for basic call is required.
IAM from PSTN/ISDN with parameter compatibility information	Support of the compatibility procedure is a general requirement for ITU-T rec. Q.764 implementations. However, in this case the complete set of legal messages, parameters and parameter values is specified in the national interconnect ISUP specification. No other signalling information is permitted, and gateway policing functions apply. No specific procedure for basic call is required in the receive direction based on the received compatibility information.
IAM from PSTN/ISDN with Transmission medium requirement prime	See procedures for multi-use bearer service. Note that if TMR= 64 kbit/s unrestricted preferred, without TMR prime included in IAM, the call shall be released with cause 111 (protocol error).
IAM from PSTN/ISDN with Location number	SIP INVITE. The Location number parameter is discarded.
IAM from PSTN/ISDN with User teleservice information	SIP INVITE. The User teleservice information parameter is discarded.

Release

Input	Response
REL from PSTN/ISDN before ACM	SIP response CANCEL according to RFC2543. RLC sent to PSTN/ISDN by SIP/SS7 GW

Input	Response
Call rejection from SIP before ACM	REL sent to PSTN/ISDN by SIP/SS7 GW with cause value dependent upon the received SIP response: - 401 unauthorised → 21 Call rejected - 402 payment required → 21 Call rejected - 403 forbidden → 21 Call rejected - 404 not found → 1 Unallocated number - 405 method not allowed → 63 Service or option unavailable - 406 not acceptable → 79 Service or option not implemented - 407 proxy authentication required → 21 Call rejected - 408 request timeout → 102 Recovery on timer expiry - 409 conflict → 41 Temporary failure - 410 gone → 22 Number changed - 411 length required → 127 Interworking - 413 request entity too long → 127 Interworking - 414 request URI too long → 127 Interworking - 415 unsupported media type → 79 Service or option not implemented - 420 bad extension → 127 Interworking - 480 temporarily unavailable → 18 No user responding - 483 too many hops → 34 No circuit available - 484 address incomplete → 28 Invalid number format

	- 485 ambiguous → 1 Unallocated number - 486 busy here → 17 User busy - 488 not acceptable here → 31 Normal, unspecified - 500 server internal error → 41 Temporary failure - 501 not implemented → 38 Network out of order - 502 bad gateway → 38 Network out of order - 503 service unavailable → 41 Temporary failure - 504 server time-out → 102 Recovery on timer expiry - 600 busy everywhere → 17 User busy - 603 decline → 21 Call rejected - 604 does not exist anywhere → 1 Unallocated number - 606 not acceptable → 31 Normal, unspecified When the cause parameter is mapped from a received SIP message from the public Internet, the location value shall be set to User. When the cause parameter is mapped from a received SIP message from a secure IP VPN dedicated to a specific company, the location value shall be set to RPN. Otherwise (i.e. when the release is initiated by the gateway) the location value shall be set to RLN.

Early ACM

Input	Response
Timer T11 expiry in the SIP/SS7 gateway	ACM to PSTN/ISDN with Backward call indicators: - charge indicator = “charge” - called party’s status indicator = “no indication” - called party’s category indicator = “ordinary subscriber” - end-to-end method indicator = “no end-to-end method available” - interworking indicator = “no interworking encountered”

	- end-to-end information indicator = “ no end-to-end information available” - ISDN user part indicator = “ISDN user part used all the way” - holding indicator = “holding not requested” - ISDN access indicator = “terminating access ISDN” - echo control indicator = “incoming half echo control device not included” - SCCP method indicator = “no indication”

SIP 180

Input	Response
SIP 180 as the first backward 18x message	ACM to PSTN/ISDN with Backward call indicators: - charge indicator = “charge” - called party’s status indicator = “subscriber free” - called party’s category indicator = “ordinary subscriber” - end-to-end method indicator = “no end-to-end method available” - interworking indicator = “no interworking encountered” - end-to-end information indicator = “ no end-to-end information available” - ISDN user part indicator = “ISDN user part used all the way” - holding indicator = “holding not requested” - ISDN access indicator = “terminating access ISDN” - echo control indicator = “incoming half echo control device not included” - SCCP method indicator = “no indication”

	Access delivery information: - Access delivery indicator = “set-up message generated” In-band alerting tone in the backwards direction is applied by the media gateway in case the TMR value was “speech” or “3.1 kHz audio”.

SIP 181

Input	Response
SIP 181 as the first 18x message	ACM to PSTN/ISDN with Backward call indicators: - charge indicator = “charge” - called party’s status indicator = “no indication” - called party’s category indicator = “ordinary subscriber” - end-to-end method indicator = “no end-to-end method available” - interworking indicator = “no interworking encountered” - end-to-end information indicator = “ no end-to-end information available” - ISDN user part indicator = “ISDN user part used all the way” - holding indicator = “holding not requested” - ISDN access indicator = “terminating access ISDN” - echo control indicator = “incoming half echo control device not included” - SCCP method indicator = “no indication” Optional backward call indicators (use of the optional backward call indicators is service dependent):

	- in-band information indicator = “no indication” - call diversion may occur indicator = “call diversion may occur” - simple segmentation indicator = “no additional information will be sent” (assuming segmentation is not used) - MLPP user indicator = “no indication” - Bits E-H = 0000

SIP 182

Input	Response
SIP 182 as the first 18x message	ACM to PSTN/ISDN with Backward call indicators: - charge indicator = “charge” - called party’s status indicator = “no indication” - called party’s category indicator = “ordinary subscriber” - end-to-end method indicator = “no end-to-end method available” - interworking indicator = “no interworking encountered” - end-to-end information indicator = “ no end-to-end information available” - ISDN user part indicator = “ISDN user part used all the way” - holding indicator = “holding not requested” - ISDN access indicator = “terminating access ISDN” - echo control indicator = “incoming half echo control device not included” - SCCP method indicator = “no indication”

SIP 183

Input	Response
SIP 183 as the first backward 18x message	ACM to PSTN/ISDN with Backward call indicators: - charge indicator = “charge” - called party’s status indicator = “no indication” - called party’s category indicator = “ordinary subscriber” - end-to-end method indicator = “no end-to-end method available” - interworking indicator = “no interworking encountered” - end-to-end information indicator = “ no end-to-end information available” - ISDN user part indicator = “ISDN user part used all the way” - holding indicator = “holding not requested” - ISDN access indicator = “terminating access ISDN” - echo control indicator = “incoming half echo control device not included” - SCCP method indicator = “no indication”

SIP 180

Input	Response
SIP 180 as a later 18x backward message	CPG to PSTN/ISDN with Event information: - Event indicator = “ALERTING” - Event presentation restricted indicator = “no indication” Access delivery information: - Access delivery indicator = “set-up message generated”

	In-band alerting tone in the backwards direction is applied by the media gateway in case the TMR value was "speech" or "3.1 kHz audio".

SIP 181

Input	Response
SIP 181 as a later 18x backward message	CPG to PSTN/ISDN with Event information: - Event indicator = "PROGRESS" - Event presentation restricted indicator = "no indication" Optional backward call indicators (use of the optional backward call indicators is service dependent): - in-band information indicator = "no indication" - call diversion may occur indicator = "call diversion may occur" - simple segmentation indicator = "no additional information will be sent" (assuming segmentation is not used) - MLPP user indicator = "no indication" - Bits E-H = 0000

SIP 182

Input	Response
SIP 182 as a later 18x backward message	CPG to PSTN/ISDN with Event information: - Event indicator = "PROGRESS"

	- Event presentation restricted indicator = "no indication"

SIP 183

Input	Response
SIP 183 as a later 18x backward message	CPG to PSTN/ISDN with Event information: - Event indicator = "PROGRESS" - Event presentation restricted indicator = "no indication"

Release from PSTN/ISDN

Input	Response
REL from PSTN/ISDN before ANM	SIP response CANCEL according to RFC2543. RLC sent to PSTN/ISDN by SIP/SS7 GW

Release from SIP

Input	Response
SIP rejection message before 200 OK (INVITE response)	SIP response according to RFC2543. REL sent to PSTN/ISDN by SIP/SS7 GW. Cause parameter values as for REL before ACM.

SIP 200 OK

Input	Response

SIP 200 OK	ANM to PSTN/ISDN. The media path is through connected in both directions in the media gateway.

SIP 200 OK (no 18x received)

Input	Response
SIP 200 OK (no 18x received)	If no ACM has been sent to PSTN/ISDN, CON is sent to PSTN/ISDN. The media path is through connected in both directions in the media gateway.

Release from PSTN/ISDN

Input	Response
REL from PSTN/ISDN after CON/ANM	SIP response BYE according to RFC2543. RLC sent to PSTN/ISDN by SIP/SS7 GW

Release from SIP

Input	Response
SIP BYE after CON/ANM to PSTN/ISDN	SIP response according to RFC2543. REL(16) sent to PSTN/ISDN by SIP/SS7 GW

4.1.2 Supplementary services

Subaddressing (SUB)

Input	Response
IAM from PSTN/ISDN with Calling party subaddress information element included in ATP	SIP INVITE. The Calling party subaddress information element is discarded.

Calling line identification presentation (CLIP)

Input	Response
IAM from PSTN/ISDN with Generic number parameter, number qualifier= additional calling party number, address presentation restricted indicator= presentation allowed, screening indicator= user provided, not verified, nature of address= national(significant)number, address signal= cgnadd	SIP INVITE. Calling address: sip:+47<cgnadd>@host.domain; Calling party number parameter is discarded if present.
IAM from PSTN/ISDN with Generic number parameter, number qualifier= additional calling party number, address presentation restricted indicator= presentation allowed, screening indicator= user provided, not verified, nature of address= international number, address signal= cgnadd	SIP INVITE. Calling address: sip:+<cgnadd>@host.domain; Calling party number parameter is discarded if present.
IAM from PSTN/ISDN without Generic number parameter (number qualifier= additional calling party number), with Calling party number	SIP INVITE. Calling address: sip:+47<cgn>@host.domain;

parameter, address presentation restricted indicator= presentation allowed, screening indicator= user provided, not verified/network provided, nature of address= national(significant)number, address signal= cgn	
IAM from PSTN/ISDN without Generic number parameter (number qualifier= additional calling party number), with Calling party number parameter, address presentation restricted indicator= presentation allowed, screening indicator= user provided, not verified/network provided, nature of address= international number, address signal= cgn	SIP INVITE. Calling address: sip:+<cgn>@host.domain;
IAM from PSTN/ISDN without Generic number parameter (number qualifier= additional calling party number), without Calling party number parameter	SIP INVITE. Calling address anonymous: sip:"restricted"@host.domain;

Calling line identification restriction (CLIR)

Input	Response
IAM from PSTN/ISDN with - Calling party number parameter coded "presentation restricted" - Generic number	The Calling party number parameter is discarded, and the content is not mapped to SIP. The SIP calling address URL is coded anonymous sip:"restricted"@host.domain;. The Generic number parameter is discarded, and the content is not mapped to SIP.

parameter (additional calling party number) coded "presentation restricted"	

Connected line identification presentation (COLP)

Input	Response
IAM from PSTN/ISDN with request for COL	The request is ignored, no Connected number parameter is sent from the gateway.

Connected line identification restriction (COLR)

Input	Response
IAM from PSTN/ISDN with request for COL	The request is ignored, no Connected number parameter is sent from the gateway.

Call diversion (CFB, CFNR, CFU, CD)

Input	Response
IAM from PSTN/ISDN with Redirection information, Redirecting number, Original called number	INVITE sip:"user C" from Called party number in ISUP From sip:"user A" from Calling party number/additional calling party number in ISUP as applicable for CLIP/CLIR To sip:"user B" from Original called number, included if present and coded "presentation allowed", otherwise not included in ISUP. Redirection information: No procedures required (i.e. the information is not mapped to SIP)
Incoming call from PSTN/ISDN, Call diversion towards PSTN/ISDN invoked by SIP (e.g. SIP redirect before answer= SIP 3xx)	See Q.732.2 (1993). - If the Redirection information parameter was not received in the incoming IAM from PSTN/ISDN, it is generated by the SIP/SS7 gateway and included in the outgoing IAM to the PSTN/ISDN. - If the Redirection information parameter was received in the incoming IAM from PSTN/ISDN, with a value of the redirection counter less than 5, the value of the redirection counter is incremented and the Redirection information

	parameter included in the outgoing IAM to the PSTN/ISDN by the SIP/SS7 gateway. - If the Redirection information parameter was received in the incoming IAM from PSTN/ISDN, with the value of the redirection counter higher than 4, the call is rejected by the SIP/SS7 gateway with REL(21) to the PSTN/ISDN. - If the SIP redirect comes from a secure IP VPN dedicated to a specific company, the Called party number in the incoming IAM from the PSTN/ISDN is mapped to the Redirecting number parameter in the outgoing IAM to the PSTN/ISDN. - If the SIP redirect comes from a secure IP VPN dedicated to a specific company, and the Original called number was received in the incoming IAM from PSTN/ISDN, it is passed in the outgoing IAM to the PSTN/ISDN. - If the SIP redirect comes from a secure IP VPN dedicated to a specific company, and the Original called number was not received in the incoming IAM from the PSTN/ISDN, it is generated by the SIP/SS7 GW (set equal to the Redirecting number) and passed in the outgoing IAM to PSTN/ISDN. - The Redirection number is not generated, and not passed on if received. - The Call diversion information parameter is not generated. - The Calling party number and the additional calling party number received in the Generic number from the incoming PSTN/ISDN are passed on in the outgoing IAM for the redirected call to the PSTN/ISDN. - If the SIP redirect did not come from a secure IP VPN dedicated to a specific company, the Redirecting number and the Original called number are not sent.
Incoming call from PSTN/ISDN, SIP redirect (3xx) to another SIP user.	a) when ACM has not been sent to the PSTN/ISDN: ACM to PSTN/ISDN with Backward call indicators: - charge indicator = “charge” - called party’s status indicator = “no indication”

	- called party's category indicator = "ordinary subscriber" - end-to-end method indicator = "no end-to-end method available" - interworking indicator = "no interworking encountered" - end-to-end information indicator = " no end-to-end information available" - ISDN user part indicator = "ISDN user part used all the way" - holding indicator = "holding not requested" - ISDN access indicator = "terminating access ISDN" - echo control indicator = "incoming half echo control device not included" - SCCP method indicator = "no indication" Optional backward call indicators (use of the optional backward call indicators is service dependent): - in-band information indicator = "no indication" - call diversion may occur indicator = "call diversion may occur" - simple segmentation indicator = "no additional information will be sent" (assuming segmentation is not used) - MLPP user indicator = "no indication" Bits E-H = 0000 b) when ACM has been sent to the PSTN/ISDN CPG to PSTN/ISDN with Event information: - Event indicator = "PROGRESS" - Event presentation restricted indicator = "no indication"

	Optional backward call indicators (use of the optional backward call indicators is service dependent): - in-band information indicator = “no indication” - call diversion may occur indicator = “call diversion may occur” - simple segmentation indicator = “no additional information will be sent” (assuming segmentation is not used) - MLPP user indicator = “no indication” Bits E-H = 0000 Note: Redirection number parameter, Call diversion information parameter or Redirection number restriction parameter shall not be generated by the SIP/SS7 gateway.

Call waiting (CW)

Input	Response
Generic notification received in ISUP	No procedures required, the Generic notification parameter is discarded, and not mapped to SIP.

Call hold (HOLD)

Input	Response
Generic notification received in ISUP	No procedures required, the Generic notification parameter is discarded, and not mapped to SIP.

Terminal portability (TP)

Input	Response
SUS(user initiated), RES(user initiated) received in ISUP	No procedures required, SUS(user initiated) and RES(user initiated) are discarded and not mapped to SIP.

Conference calling (CONF)

Input	Response
Generic notification received in ISUP	No procedures required, the Generic notification parameter is discarded, and not mapped to SIP.

Three-party service (3PTY)

Input	Response
Generic notification received in ISUP	No procedures required, the Generic notification parameter is discarded, and not mapped to SIP.

Closed user group (CUG)

Input	Response
Optional forward call indicators in IAM from PSTN/ISDN indicating “closed user group call, outgoing access not allowed”	SIP/SS7 GW rejects the call with REL(87) towards PSTN/ISDN.
IAM from PSTN/ISDN without Optional forward call indicators, or with another value of the optional forward call indicators	SIP INVITE (depending on basic call procedures) as a non CUG call.

User-to-user signalling (UUS1 implicit, UUS1 explicit, UUS2, UUS3)

Input	Response
User-to-user information, user-to-user indicators	Support of UUS is not required. No signalling response is required from the SIP/SS7 GW. User-to-user indicators and User-to-user information are discarded.

Explicit call transfer (ECT)

Input	Response

Signalling information received from PSTN/ISDN related to a call transfer invoked in the PSTN/ISDN	Optionally the ETSI ISUP signalling procedures for ECT can be supported. However, support of the ECT signalling procedure is not considered to be a priority for this release of the SIP/SS7 GW. In this case the information is discarded.
ECT invoked from the SIP side (e.g. as SIP redirect after answer or as BYE with transfer information) The interpretation of the SIP procedures for call transfer need to be clarified.	A protection timer (1-8h, default 4h) shall be started by the SIP/SS7 GW. Optionally in addition the ETSI ISUP signalling procedures for ECT can be supported. However, support of the ECT signalling procedure is not considered to be a priority for this release of the SIP/SS7 GW.

Malicious call identification (MCID)

Input	Response
-	No ISUP signalling procedures are required towards the PSTN/ISDN

Call completion on busy subscriber (CCBS)

Input	Response
-	Support of CCBS is not required. “CCBS not possible” shall be included in cause diagnostics in REL(17,34) to PSTN/ISDN

4.2 Call from SIP to ISUP

4.2.1 Basic call

Nature of connection indicators

Input	Response

SIP INVITE, the called address can be mapped to a valid number series.	IAM to PSTN/ISDN with Nature of connection indicators: - Satellite indicator = “one satellite circuit in the connection” - Continuity check indicator = “continuity check not required” - Echo control device indicator = “outgoing half echo control device not included” ISUP timer T7 is started according to ISUP procedures.

Forward call indicators

Input	Response
SIP INVITE, the called address can be mapped to a valid number series.	IAM to PSTN/ISDN with Forward call indicators: - National/international call indicator = “call to be treated as an international call” - End-to-end method indicator = “no end-to-end method available” - Interworking indicator = “no interworking encountered” - End-to-end information available = “no end-to-end information available” - ISDN user part indicator = “ISDN user part used all the way” - ISDN user part preference indicator = “ISDN user part preferred all the way” - ISDN access indicator = “originating access ISDN” - SCCP method indicator = “no indication” - Bits L-P = 00000

Calling party's category

Input	Response
SIP INVITE, the called address can be mapped to a valid number series.	IAM to PSTN/ISDN with Calling party's category: - Calling party's category parameter field = "ordinary calling subscriber"

Transmission medium requirement

Input	Response
SIP INVITE, the called address can be mapped to a valid number series. a) Allowed media codings: G.711 or G.723.1 (G.728, G.729), (G.722 not included) b) Allowed media coding: H.221/H.263 c) Allowed media codings: G.722 included and G.711 or G.723.1 (G.728, G.729) included	IAM to PSTN/ISDN with Transmission medium requirement (TMR) parameter with: a) Transmission medium requirement parameter field = "speech", USI = "speech". The SIP/SS7 GW includes a half echo control device, cancelling echo received from the PSTN/ISDN side. b) Transmission medium requirement parameter field = "64 kbit/s unrestricted", USI = "UDI". No echo control device is included. c) Transmission medium requirement parameter field = "64 kbit/s unrestricted preferred", TMR prime = "speech", USI = "speech", USI prime = "UDI-TA", HLC in ATP = "telephony". If TMU = "speech" is received in ACM/CPG/CON/ANM, G.711 or G.723.1 (G.728, G.729) is indicated as media coding in 200 to the SIP calling user, and the SIP/SS7 GW includes a half echo control device cancelling echo received from the PSTN/ISDN side. If BC = "UDI-TA" is received in ATP in CON/ANM, G.722 is indicated as media coding in 200 to the SIP calling user, and no echo control device is included. If neither response is

	received, the call is released with cause #111 (protocol error). Gateway or MCU with bonding capability is outside the scope of this specification, only mapping of single 64 kbit/s channels is considered.

Called party number

Input	Response
SIP INVITE, the called address can be mapped to a valid number series. a) the SIP called address has the form sip:<+47cpn>@host.domain; b) the SIP called address has the form sip:<+CCcpn>@host.domain; (CC different from 47)	IAM to PSTN/ISDN with Called party number: a) nature of address = “national(significant)number”, address signal = <cpn> b) nature of address = “international number”, address signal = <CCcpn> In case the address has more than 16 digits, the first 16 digits are included in the Called party number parameter in IAM, and the remaining in a SAM. In case of overlap signalling (with a subsequent INVITE from the SIP side), the subsequent INVITE is mapped to a SAM. In case of number portability implemented in the gateway, ported numbers are indicated by: nature of address= “ported number”, address signal= <pfx,cpn>. However, in a first phase it is assumed that onward routing is used for number portability.

Other basic call related parameters

Input	Response
SIP INVITE, the called address can be mapped to a	IAM to PSTN/ISDN:

valid number series.	- Assuming the simple segmentation procedure is not used, the optional forward call indicators parameter can be omitted. - The propagation delay counter shall be included with the value 150 ms. - The parameter compatibility information parameter shall be included with appropriate coding according to ETSI recommendations. - The Location number parameter is not generated.

Release from SIP before ACM

Input	Response
SIP CANCEL before ACM from PSTN/ISDN	SIP response according to RFC2543. REL(16) sent to PSTN/ISDN by SIP/SS7 GW

Release from PSTN/ISDN before ACM

Input	Response
REL from PSTN/ISDN before ACM	RLC sent to PSTN/ISDN by SIP/SS7 GW SIP response dependent on the received cause value from PSTN/ISDN: - 1 unallocated number → 404 Not found - 3 no route to destination → 404 Not found - 4 send special information tone → 503 Service unavailable - 12 number ported → 502 Bad gateway - 16 normal call clearing → 603 Decline - 17 user busy → 486 Busy here - 18 no user responding → 408 Request timeout - 19 no answer from user → 480 Temporarily unavailable - 20 subscriber absent → 480 Temporarily unavailable

	- 21 call rejected → 603 Decline - 22 number changed (w/o diagnostic) → 410 Gone - 22 number changed (w/ diagnostic) → 301 Moved permanently - 26 non selected user clearing → 404 Not found - 27 destination out of order → 502 Bad gateway - 28 address incomplete → 484 Address incomplete - 29 facility rejected → 501 Not implemented - 31 normal unspecified → 480 Temporarily unavailable - 34 no circuit available → 503 Service unavailable - 38 network out of order → 503 Service unavailable - 41 temporary failure → 503 Service unavailable - 42 switching equipment congestion → 503 Service unavailable - 47 resource unavailable → Service unavailable - 55 incoming calls barred within CUG → 403 Forbidden - 57 bearer capability not authorised → 403 Forbidden - 58 bearer capability not presently available → 503 Service unavailable - 63 service or option not available, unspecified → 503 Service unavailable - 65 bearer capability not implemented → 501 Not implemented - 69 requested facility not implemented → 501 Not implemented - 79 service or option not implemented, unspecified → 501 Not implemented - 87 user not member of CUG → 503 Service unavailable - 88 incompatible destination → 503 Service unavailable - 95 invalid message, unspecified → 503 Service

	unavailable - 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 recovery on timer expiry → 408 Request timeout - 111 protocol error → 500 Server internal error - 127 interworking, unspecified → 500 Server internal error Other received cause values are mapped as the default value in class, see ITU-T rec. Q.850 (1993), section 2.2.7.

ACM

Input	Response
SIP INVITE, the called address can be mapped to a valid number series. ACM (subscriber free) is received from PSTN/ISDN.	IAM to PSTN/ISDN. - If the TMR value was “speech” or “3.1 kHz audio”: SIP 180 with SDP = G.711/G.723.1 (G.728,G.729) is sent to the SIP user The media path is through connected in the backwards direction by the media gateway. - If the TMR value was “64 kbit/s unrestricted”: SIP 180 is sent to the SIP user - If the TMR value was “64 kbit/s unrestricted preferred”, and TMU has not been received: SIP 180 is sent to the SIP user - If the TMR value was “64 kbit/s unrestricted preferred”, and TMU has been received:

	SIP 183 with SDP = G.711/G.723.1 (G.728,G.729) is sent to the SIP user The media path is through connected in the backwards direction by the media gateway. ISUP timer T7 is stopped and ISUP timer T9 is started according to ISUP procedures.

ACM

Input	Response
SIP INVITE, the called address can be mapped to a valid number series. ACM (no indication) is received from PSTN/ISDN.	IAM to PSTN/ISDN. SIP 183 to the SIP user. ISUP timer T7 is stopped and ISUP timer T9 is started according to ISUP procedures.

ACM

Input	Response
SIP INVITE , the called address can be mapped to a valid number series. ACM (no indication), with Optional backward call indicators = “in-band information or an appropriate pattern is now available”, is received from PSTN/ISDN.	IAM to PSTN/ISDN. SIP 183 with SDP = G.711/G.723.1(G.728,G.729) to the SIP user. The media path is through connected in the backwards direction by the media gateway. ISUP timer T7 is stopped and ISUP timer T9 is started according to ISUP procedures. Note that a cause parameter in ACM or CPG implies that the call is unsuccessful. However, there is no mapping of a cause parameter in ACM or CPG to SIP.

ACM

Input	Response
SIP INVITE, the called address can be mapped to a valid number series. ACM (no indication), with Optional backward call indicators = “call diversion may occur”, is received from PSTN/ISDN.	IAM to PSTN/ISDN. SIP 181 to the SIP user. ISUP timer T7 is stopped and ISUP timer T9 is started according to ISUP procedures.

CPG

Input	Response
SIP INVITE, the called address can be mapped to a valid number series. CPG (after ACM) is received from PSTN/ISDN a) Event indicator = “ALERTING” or backward call indicators = “subscriber free” b) Event indicator = “PROGRESS” c) Event indicator or optional backward call indicators = “in-band information or an appropriate pattern is now available” d) Event indicator = “call	IAM to PSTN/ISDN. SIP 18x to the SIP user. a) See response to ACM(subscriber free) b) SIP 183 c) SIP 183 with SDP = G.711/G.723.1(G.728,G.729) d) SIP 181

forwarded on busy"/ "call forwarded on no reply"/ "call forwarded unconditional", or optional backward call indicators = "call diversion may occur"	

Release from SIP before ANM

Input	Response
SIP CANCEL before ANM from PSTN/ISDN	SIP response according to RFC2543. REL(16) sent to PSTN/ISDN by SIP/SS7 GW

Release from PSTN/ISDN

Input	Response
REL from PSTN/ISDN before ANM	RLC sent to PSTN/ISDN by SIP/SS7 GW SIP response according to mapping of cause values as for REL before ACM.

CON

Input	Response
SIP INVITE, the called address can be mapped to a valid number series. CON from PSTN/ISDN	IAM to PSTN/ISDN. SIP 200 OK to SIP user. The media path is through connected in both directions in the media gateway. ISUP timer T7 is stopped.

ANM

Input	Response
SIP INVITE, the called address can be mapped to a valid number series. ANM from PSTN/ISDN	IAM to PSTN/ISDN. SIP 200 OK to SIP user. The media path is through connected in both directions in the media gateway. ISUP timer T9 is stopped.

BYE from SIP

Input	Response
BYE from SIP after 200 OK (INVITE confirmation)	REL(16) to PSTN/ISDN

REL from PSTN/ISDN

Input	Response
REL from PSTN/ISDN after CON/ANM	SIP BYE

SUS(network initiated)

Input	Response
SIP INVITE, the called address can be mapped to a	IAM to PSTN/ISDN.

valid number series. SUS(network initiated) received from PSTN/ISDN after CON/ANM. RES(network initiated) received from PSTN/ISDN after SUS(network initiated).	The SIP/SS7 GW shall act as a PSTN/ISDN local exchange, and start timer T6 (Q.764) upon receipt of SUS(network initiated). On receipt of RES(network initiated) timer T6 (Q.764) shall be terminated. On timer expiry the call is released by the SIP/SS7 gateway with REL(102) and SIP BYE.

4.2.2 Supplementary services

Subaddressing (SUB)

Input	Response
SIP INVITE	IAM to PSTN/ISDN. The Calling party subaddress information element is not included in IAM to PSTN/ISDN.

Calling line identification presentation (CLIP)

Input	Response
SIP INVITE Calling address: sip:+47<cgnadd>@host.domain;	IAM to PSTN/ISDN with Generic number parameter, number qualifier= additional calling party number, address presentation restricted indicator= presentation allowed, screening indicator= user provided, not verified, nature of

The calling address comes from a reliable source on the Internet (has been authenticated).	address= national(significant)number, address signal= <cgnadd>. Calling party number parameter is not included.
SIP INVITE. Calling address: sip:+CC<cgnadd>@host.domain; CC is different from 47. The calling address comes from a reliable source on the Internet (has been authenticated).	IAM to PSTN/ISDN with Generic number parameter, number qualifier= additional calling party number, address presentation restricted indicator= presentation allowed, screening indicator= user provided, not verified, nature of address= international number, address signal= CC<cgnadd> Calling party number parameter is not included.
SIP INVITE. The calling address comes from a source on the Internet and has not been authenticated.	IAM to PSTNISDN. Neither the Generic number parameter (additional calling party number) nor the Calling party number parameter are included in the IAM.
SIP INVITE. Calling address: sip:+CC<cgn>@host.domain; The SIP INVITE with the calling address comes from the IP telephony application directly on a company specific secure IP VPN.	IAM to PSTN/ISDN with the Calling party number parameter. The contents of the received calling address shall be screened by the gateway (compared to the defined E.164 address range for the company access). • When the received address is within the defined range and CC is 47, the Calling party number parameter shall be coded: address presentation restricted indicator= “presentation allowed”, screening indicator= “user provided, verified and passed”, nature of address= “national(significant) number”, address signal= <cgn> • When the received address is within the defined range and CC is different from 47, the Calling party number shall be coded: address presentation restricted indicator= “presentation allowed”, screening indicator= “user provided, verified and passed”, nature of address= “international number”, address signal= CC<cgn> • When the received address is not within the defined range, the Calling party number parameter shall be coded: address presentation restricted indicator= “presentation allowed”, screening indicator= “network provided”, with the nature of address and address signal according to the default access number defined for the company subscription.

Calling line identification restriction (CLIR)

Input	Response
SIP INVITE. Calling address:"anonymous" sip:"restricted"@host.domain ; The calling address comes from a source on the Internet	IAM to PSTN/ISDN. Neither the Generic number parameter (additional calling party number) nor the Calling party number parameters are included in the IAM.
SIP INVITE. Calling address:"anonymous" sip:"restricted"@host.domain ; The SIP INVITE with the calling address comes from the IP telephony application directly on a company specific secure IP VPN.	IAM to PSTN/ISDN. The Calling party number parameter shall be coded: address presentation restricted indicator= "presentation restricted", screening indicator= "network provided", with the nature of address and address signal according to the default access number defined for the company subscription

Connected line identification presentation (COLP)

Input	Response
SIP INVITE. ANM/CON from PSTN/ISDN with COL included, coded "presentation allowed".	IAM to PSTN/ISDN. The connected number is not requested in the IAM to PSTN/ISDN. The Connected number parameter is discarded and not mapped to SIP if received in ANM/CON.

Connected line identification restriction (COLR)

Input	Response
SIP INVITE. ANM/CON from PSTN/ISDN with COL included, coded "presentation restricted".	IAM to PSTN/ISDN. The connected number is not requested in the IAM to PSTN/ISDN. The Connected number parameter is discarded and not mapped to SIP if received in ANM/CON.

Call diversion (CFB, CFNR, CFU, CD)

Input	Response
SIP INVITE. a) The call has been redirected within the SIP domain b) The Redirection number, the Call diversion information parameter, the redirection number restriction parameter is received in ACM/CPG/CON/ANM from PSTN/ISDN, indicating that the call has been redirected within the ISDN domain.	IAM to PSTN/ISDN. a) No indication to ISUP that the call has been redirected: INVITE sip:"user C" -> Called party number in ISUP From sip:"user A" -> Calling party number/additional calling party number in ISUP as applicable (see rules for CLIP/CLIR) To sip:"user B" -> not included in ISUP b) No procedures required for the Redirection number parameter, the Call diversion information parameter, the Redirection number restriction parameter (i.e. the information is not mapped to SIP).

Call waiting (CW)

Input	Response
SIP INVITE. Generic notification received in ISUP	IAM to PSTN/ISDN. No procedures required, the Generic notification parameter is discarded, and not mapped to SIP.

Call hold (HOLD)

Input	Response
SIP INVITE. Generic notification received in ISUP	IAM to PSTN/ISDN. No procedures required, the Generic notification parameter is discarded, and not mapped to SIP.

Terminal portability (TP)

Input	Response
SIP INVITE. SUS(user initiated), RES(user initiated) received in ISUP	IAM to PSTN/ISDN. No procedures required, SUS(user initiated) and RES(user initiated) are discarded and not mapped to SIP.

Conference calling (CONF)

Input	Response
SIP INVITE. Generic notification received in ISUP	IAM to PSTN/ISDN. No procedures required, the Generic notification parameter is discarded, and not mapped to SIP.

Three-party service (3PTY)

Input	Response
SIP INVITE. Generic notification received in ISUP	IAM to PSTN/ISDN. No procedures required, the Generic notification parameter is discarded, and not mapped to SIP.

Closed user group (CUG)

Input	Response
SIP INVITE.	IAM to PSTN/ISDN without Optional forward call indicators, or with the Optional forward call indicators indicating “non CUG call”.

User-to-user signalling (UUS1 implicit, UUS1 explicit, UUS2, UUS3)

Input	Response
SIP INVITE. User-to-user information, user-to-user indicators	IAM to PSTN/ISDN. UUS not requested from the SIP/SS7 GW. Support of UUS is not required. No signalling response is

received in ISUP.	required from the SIP/SS7 GW. User-to-user indicators and User-to-user information are discarded.

Explicit call transfer (ECT)

Input	Response
Signalling information received from PSTN/ISDN related to a call transfer invoked in the PSTN/ISDN	Optionally the ETSI ISUP signalling procedures for ECT can be supported. However, support of the ECT signalling procedure is not considered to be a priority for this release of the SIP/SS7 GW.
ECT invoked from the SIP side (e.g. SIP redirect after answer or BYE with transfer information) The interpretation of the SIP procedures for call transfer need to be clarified.	A protection timer (1-8h, default 4h) shall be started by the SIP/SS7 GW. Optionally in addition the ETSI ISUP signalling procedures for ECT can be supported. However, support of the ECT signalling procedure is not considered to be a priority for this release of the SIP/SS7 GW.

Malicious call identification (MCID)

Input	Response
SIP INVITE	IAM to PSTN/ISDN. No additional ISUP signalling procedures are required towards the PSTN/ISDN

Call completion on busy subscriber (CCBS)

Input	Response
SIP INVITE. “CCBS possible” received in cause diagnostics in REL(17,34) from PSTN/ISDN.	IAM to PSTN/ISDN. Support of CCBS is not required. No additional signalling procedures are required by the SIP/SS7 GW.

Notes:

The descriptions in this specification are not exhaustive.

For not described issues on the ISUP side, the procedures as specified in the Telenor national interconnect specification apply (e.g. timers).

SIP-T messages shall not be sent by the gateway. ISUP contents received in SIP-T messages from the SIP side shall not be mapped to the ISUP side.

Only ISUP messages and parameters specified in the Telenor national interconnect ISUP version 2, shall be sent by the gateway to the PSTN/ISDN.

The SIP protocol description used is based on RFC2543bis3, which can be found at: http://www.cs.columbia.edu/~hgs/sip/drafts/draft-ietf-sip-rfc2543bis-03.pdf .