VoIP Interkonnektion – Test Specification

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Vertrag	Vertrag betreffend Verbindung von VoIP Fernmeldeanlagen und -diensten

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1   Introduction

1. This pre-service Test Specification is based on recommendations by i3 forum [2] and provides testing guidance for establishing a new bilateral VoIP Interconnection between SIP telephony service providers, ensuring service interoperability, signaling compatibility, quality and performance levels that meet customer expectations.

2. This document covers the test approach, specific functionality, assumptions, and test cases for verifying the VoIP Interconnection between two SIP telephony service providers before the delivery of customer traffic with a focus on voice services.

3. This Test Specification may, however, also be used when changes are made to an existing VoIP Interconnection. This could be for example when further services are added, or one or more network connections ('Netzverbindung') are added. The service providers may agree on a reduced number of test cases to perform. It is assumed the new capacity will be on the same equipment (technology is the same, SBC vendor and model are the same, new IP address only, etc), else it will be necessary to perform the complete set of tests.

4. Test cases in this document shall cover calls in both directions, unless the VoIP Interconnection, concerned service or network connection is unidirectional only.

5. Test cases in this document shall cover calls to all applicable types of networks (i.e. VoIP, TDM and mobile/MNO), as depicted in the high level configuration diagrams below. Furthermore, if the service provider has different types of VoIP networks "behind" his border function, calls to/from the different VoIP networks shall also be tested.

6. Both service providers will capture and record call traces for each of the test call scenarios for comparison to the specification 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface'. All final call traces are to be kept for future reference.

7. Both service providers will collect Call Detail Records of the test calls for billing verification.

8. The following items are not in the scope of this document:
   – How the call traces and CDRs are captured
   – What equipment is used to capture call traces and CDRs
   – Service quality testing – It is assumed that once this VoIP Interconnection is successfully validated, the service will go into production. Any further testing for service quality verification will be agreed bilaterally.

2   Assumptions

1. Both service providers will agree to the test scenarios, test plan / process and the expected test results used to verify that the test was successful.

2. In case of discrepancies between the text in this document and the text in 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface', the specifications in 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion' takes precedence over this document.

3. Testing shall be performed on a network configuration as documented in the 'Handbuch Technik' and 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface'.

4. Service providers will exchange the necessary configuration details (covering service and network

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requirements) to be able to verify system compatibility.

5. Each service provider will provide testing numbers that will terminate on hard phones or soft clients with Caller ID so that CLI can be verified.

6. Each service provider will also provide testing numbers that will terminate on various Fax machines (e.g. G.3).

7. For billing verification purposes accumulated billing data (e.g. total number of calls, total call duration) shall be exchanged between the two parties.

   In case inacceptable discrepancies are found during the billing verification (see 'Handbuch Abrechnung'), it may be beneficial or necessary to exchange individual CDRs for trouble shooting purposes. In this case service providers must agree on the CDR format that will be exchanged (either raw or processed CDRs).

8. In case individual CDRs are to be exchanged between the two parties, all numbers that could be used to identify a particular customer must first be anonymised (the last four digits shall be obscured) before the data is handed over to the other party in order to adhere to applicable Swiss laws and regulations.

3 Test Strategy

3.1 Test Configuration

1. The test configuration, ie. the endpoints that should be used for the tests, will be agreed between the service providers. Both parties should use a reasonable set of endpoints to cover the different call scenarios, e.g. different kind of endpoints (e.g. VoIP phones, fax, router with analog port) on each system (ie. VoIP and TDM).

3.2 Contact Information

1. Both service providers shall exchange the contact information of the testing personnel.

3.3 Hardware and Software Configuration

1. Both service providers shall exchange and document the vendor and model of the Session Border Controller used including hardware and software versions. Information concerning the test equipment used may optionally be exchanged. The information may aid troubleshooting during the testing phase of the VoIP Interconnection.

4 Acceptance Test Techniques

1. Generally, a test plan technique that satisfies each service provider's needs will be agreed upon, that may include the following phases:

   – Preparation – Exchange contact and technical information; agree upon steps / timeline.

   – Testing – Validation tests: call completion, call teardown, audio quality, etc.

   – Completion - Final collaboration / confirmation of test success and documentation of test results. All final call traces are to be kept for future reference.

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4.1 Testing Entrance Criteria

1. The transmission (IP) connectivity is available as per exchanged configuration details and specification in the 'Handbuch Technik'.

2. The test scenarios to be used, the test plan / process and the expected test results are agreed.

3. Testing phone numbers have been exchanged for voice including voice mail service (Combox) and fax, for all applicable types of networks (as described in the previous chapters above).

4.2 Test Operative Practices

1. Besides direct feedback from partner service provider's testing personnel, a validation of the test results may come from call trace analysis. Call traces could be exchanged if needed. CDRs are captured to verify whether a test case should be marked as passed.

4.3 Test Exit Criteria

1. The test cycle will end when all of the following conditions are met:
   – The jointly identified and agreed test cases are successfully completed and both parties agree to the results.
   – The test completion and acceptance is confirmed in written form by both parties
   – CDRs for each of the required test cases are collected by both service providers, billing data are exchanged and compared to ensure consistency.

5 Test Case Scenarios for New VoIP Interconnection

1. Review and validate response messages and codes for each test scenario below.

5.1 IP Layer

5.1.1 Quality of Service Markings on Layer 3

1. Verify Quality of Service (QoS) markings that optionally have been agreed and implemented on IP layer (i.e. layer 3).

5.2 Basic Call and Supplementary Services – Positive Test Scenarios

5.2.1 Normal call - Calling party clears after answer

1. Place test call using G.711 codec and verify it completes. Verify that media is established in both directions. The calling party should hang up call. Verify that the call clears normally and the correct response is received.
   Verify that CLI (incl. From and P-Asserted-Identity headers) is properly passed and received in the agreed upon format.
   Verify that the called party number is properly passed and received in the agreed upon format.

5.2.2 Normal call - Called party clears after answer

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1. Place test call and verify it completes. The called party should hang up call. Verify that the call clears normally and the correct response is received.

5.2.3 Normal call - Calling party release while ringing

1. Place test call. While call is ringing and before it has completed, the calling party should hang up the call. Verify that the call clears correctly and the correct response is received.

5.2.4 Normal call - No Answer / Timeout

1. Place test call. Do not allow the call to complete. Verify that the call clears correctly and the correct response is received. There are two methods; CANCEL from the originating network or 480 from the terminating network.

5.2.5 Normal call to Busy Subscriber / Calling party release

1. Place test call. The far side should be busy. Verify that the call does not complete and that it clears correctly and the correct response is received.

5.2.6 Long duration call (call longer than 60 minutes)

1. Place test call and verify it completes. Keep the call connected for a period of at least 60 minutes. Verify SIP keep-alive timers (typical timer value is 15 minutes). Verify that the call does not disconnect abnormally. The calling party should hang up call. Verify that the call clears correctly.

5.2.7 Call with CLI Restriction - CLIR

1. Verify that all relevant SIP headers are set correctly:
   - the P-asserted-identity header must contain a valid subscriber number of the calling party
   - the From header must be set to "Anonymous"
   - the Privacy header must be set to "id".

   For further information, please refer to 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface'.

5.2.8 CLI presentation when P-Asserted-Identity and From headers contain different numbers

1. Verify that the number contained in the From header is presented to the called party.

   This corresponds to a situation where Special Arrangement applies (i.e. in ISUP a call with Generic Number).

5.2.9 Call Hold

1. Set up a normal call A to B over the NNI.

   The B-party (called party) puts the A-party (calling party) on hold, using Call Hold based on the IETF RFC 3264 (Offer/Answer in SDP) method. The obsolete method using a null address (c=IN IP 0.0.0.0) is not supported at the NNI.

   For further information, please refer to 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface'.

5.2.10 Call Transfer

1. Set up a normal call A (in service provider A network) to B (in service provider B network) over the NNI. To transfer the call, the B-party (called party) puts the A-party (calling party) on hold using the method

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for Call Hold above and establishes a new call to party C (in service provider A network) over the NNI. After the call between the B- and C-Parties is established, the B-party modifies both call legs media information to connect A and C together while the B-party's proxy remains in the signaling path.

Call Transfer must be based on the SIP Re-INVITE method. The SIP REFER method is not supported at the NNI.

For further information, please refer to 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface'.

5.2.11 Call Diversion using History-Info header

1. Set up a normal call A to B over the NNI. The B-party (called party) has Call Forwarding active to a C-party in service provider A's network. I.e. both A- and C-party belong to service provider A's network.

The History-Info header is the preferred method to reflect a call forwarding condition. A reason cause shall be presented if known.

For further information, please refer to 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface'.

5.2.12 Call Diversion using Diversion header

1. Set up a normal call A to B over the NNI. The B-party (called party) has Call Forwarding active to a C-party in service provider A's network. I.e. both A- and C-party belong to service provider A's network.

The Diversion header method must be supported for backward compatibility reasons.

For further information, please refer to 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface'.

5.2.13 Call to voice mail service (Combox)

1. Place a call directly to voice mail service (Combox) across the NNI.

Normally voice mail numbers are prefixed with +41 860.

5.2.14 Call to a ported number

1. Place a call to a number ported out to the other party's network or to a third party provider "behind" the other party's network.

Unless otherwise agreed between the two service providers the NPRN of the terminating network shall be sent across the NNI as a prefix for ported numbers.

For further information, please refer to 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface'.

5.2.15 Parallel Forking (Parallel/Simultaneous Ringing)

1. Place a call to a number with parallel forking (also known as parallel ringing or simultaneous ringing) to multiple endpoints ("targets"), where one endpoint is in service provider A's network and the other endpoint is in service provider B's network.

Verify that the end user equipment/devices at both endpoints are ringing simultaneously before answer. Now pick up the call at one of the endpoints and verify that bothway media is established. The call at the other endpoint shall stop ringing and clear correctly.

Repeat the test call, but this time pick up the call at the other endpoint.

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5.3 Basic Call and Supplementary Services – Negative Test Scenarios

5.3.1 Call to Unallocated Number

1. Place test call. The number should be in the correct format, but not be assigned to any subscriber. Verify that the call clears correctly and the correct response is received.

5.3.2 Call with Insufficient Digits (Partial dial)

1. Place test call. The number should not be valid (i.e. number is too short). Verify that the call clears correctly and the correct response is received.

5.4 Early Media Handling

5.4.1 Local ringbacktone - provisional response without SDP and PRACK

1. The terminating network sends 180 response containing no SDP. The originating network applies local ringback tone.

   For further information, please refer to 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface'.

5.4.2 Remote ringbacktone with early media - provisional response with SDP and PRACK

1. The terminating network sends 180 or 183 response containing SDP. The originating network establishes an early media session with the media endpoint described by the SDP.

   For further information, please refer to 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface'.

5.4.3 Early media with multiple terminating endpoints

1. Several media sessions to be established (serially) between the originating user agent and one or more intermediate media endpoints before the call is connected to the final target called user agent.

   For further information, please refer to 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface'.

5.5 DTMF, Media and Audio Codecs

5.5.1 DTMF for a call with G.711 codec

1. Verify digits sent and received for a DTMF transmission (using agreed upon DTMF method) post answer from the caller for a G.711 call.

   DTMF signals must be sent according to the IETF RFCs 2833/4733. Out-of-band tramsission of DTMF digits with SIP INFO and SIP NOTIFY is currently not supported across the NNI.

   For further information, please refer to 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface'.

5.5.2 DTMF for a call with a non G.711 codec

1. Verify digits sent and received for a DTMF transmission (using agreed upon DTMF method) post answer from the caller for another codec call (G.722 for example) call.

   DTMF signals must be sent according to the IETF RFCs 2833/4733. Out-of-band tramsission of DTMF digits with SIP INFO and SIP NOTIFY is currently not supported across the NNI.

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For further information, please refer to 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface'.

5.5.3 Non G.711 voice call

1. Both end user equipment/devices support a non G.711 codec (e.g. G.722). Place test call using a non G.711 codec and verify it completes. Verify that media is established in both directions. Verify that the call clears normally and the correct response is received.
   It is recommended that this test is performed with all relevant codecs that are to be used for production traffic (i.e. both narrow- and wideband).

5.5.4 Voice call with SDP offer containing multiple media descriptors

1. If an interconnection SBC receives an SDP offer containing multiple media descriptors, it must act on the media descriptors and include all of them in the same order in the response, including non-zero ports and zero ports for the offered media according to its capabilities as specified in 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface'.

5.6 Redundancy, Keepalive and Loop Detection Mechanisms

5.6.1 Call Forwarding Loop Detection

1. If INVITE message is received with Max-Forward header equal to zero (0), SBC must respond with 483 Too Many Hops. Other methods for call forwarding loop detection may be supported.

   For further information, please refer to 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface'.

5.6.2 Reachability and keepalive mechanism

1. Out of dialog SIP message OPTIONS shall be used to probe signaling reachability and as keepalive mechanism. The partner network shall respond with 200 OK or 483 Too Many Hops.

5.6.3 Reselection in originating network on SIP response code from terminating network

1. Place test call. A situation should be sought between the two interconnection partners, which provokes the terminating network to reject the call with a SIP response code (typically 503 "Service Unavailable") which invokes reselection (i.e. a retry) on the 2nd path in the originating network.

   As described in 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface' the interconnection SBC shall crank back to its own core network which must then re-route via the second interconnection SBC and path.

   Verify the call completes successfully via the 2nd path. The calling or called party should hang up the call after answer. Verify that the call clears normally and the correct response is received.

5.7 Fax Calls

1. The Swisscom SIP devices support either "G.711 only" or both T.38 and G.711 for fax transmission. All Swisscom SIP devices that support both modes for fax transmission, support fallback to G.711.

2. Depending on the capabilities of the SIP device(s) on the partner side, certain test cases below may not be applicable.
   For further information, please refer to 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion –

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Public Network-to-Network Interface'.

3. Swisscom normally uses a standard set of test pages for originating fax test calls from Swisscom's side. These test pages contain a mixture of text, tables and drawings in order to verify the fax image quality. It is recommended that the partner uses similar test pages for originating fax test calls from his side.

4. Verify that fax image quality is not deteriorated (test per ETSI EG 202 057-2 "Speech processing transmission and quality aspects (STQ)" [1]) for each test case below.

5.7.1   Originating fax call using G.711

1. Verify fax transmission with 3 pages using G.711 pass-through end-to-end for a G.711 voice originating call.

5.7.2   Originating fax call using T.38

1. Verify fax transmission with 3 pages using T.38 fax relay end-to-end for a G.711 voice originating call.

5.7.3   Originating fax call with fallback to G.711

1. Originating fax begins with T.38. Terminating side does not support T.38. Call falls back to G.711 pass-through end-to-end. Verify that fax with 3 pages is completed successfully.

5.7.4   Originating fax call with codec change from a non G.711 call

1. Verify fax transmission with 3 pages using G.711 pass-through end-to-end for another codec (e.g. G.722) voice originating call.

5.7.5   Long fax transmission using G.711

1. Verify fax transmission with 10 pages using G.711 pass-through end-to-end for a G.711 voice originating call. Verify that all pages are transmitted successfully.

5.7.6   Long fax transmission using T.38

1. Verify fax transmission with 10 pages using T.38 fax relay end-to-end for a G.711 voice originating call. Verify that all pages are transmitted successfully.

5.7.7   Long fax transmission after fallback to G.711

1. Originating fax begins with T.38. Terminating side does not support T.38. Call falls back to G.711 pass-through end-to-end. Verify that fax with 10 pages is completed successfully. Verify that all pages are transmitted successfully.

5.7.8   Long fax transmission after codec change from a non G.711 call

1. Verify fax transmission with 10 pages using G.711 pass-through end-to-end for another codec (e.g. G.722) voice originating call. Verify that all pages are transmitted successfully.

5.7.9   Calls to special destinations

1. The calls to special destinations as listed below should be tested according to the testcases described in chapter 5.2.1 to 5.2.9.

2.
   - International Calls

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International calls should be tested if at least one of the operators has direct international interconnections

3

- Calls to short numbers

  Calls to short numbers should be tested if at least one of the operators supports such numbers

4

- Emergency Call to test number

  Such calls can be done at any time, they will call a voicemail system that simulates the emergency center

5

- Emergency Call to real emergency number

  Such calls shall be done at least once before a new VoIP interconnection is used for non-test traffic. Important: The calls should be agreed with the emergency organisation in advance.

6

- Calls to value-added-services

  If supported by at least one of the operators calls to value added services should be tested including:
    o Check of filtering (positive & negative tests)
    o Check of billing

7

- Calls with Call-by-Call Carrier Selection

  If supported by at least one of the operators calls with Call-by-Call Carrier Selection should be tested including positive tests and negative tests (CbC not allowed for A-party)

5.8 SIP Protocol Trace Verification

1. It is recommended that the SIP INVITE message from test case 5.1.1 is reviewed and validated against the appropriate chapters in 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface'. External tools may be needed to record and save this trace.

5.8.1 Header compression

1. Header compression shall not be used in IP/UDP/TCP/RTP headers.

5.8.2 Require header

1. The Require header shall not be used whenever possible. Instead, the Supported header shall be used identifying supported SIP extensions.

5.8.3 Allow header

1. Check that Allow header contains preferably the following methods only:
    INVITE, ACK, CANCEL, BYE, PRACK, OPTIONS, UPDATE.

5.8.4 Mandatory headers

1. Check that each INVITE message contains the mandatory headers listed in chapter "SIP Header Support" of 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface'.

5.8.5 Optional headers

1. Check optional headers, which may be present in the INVITE message, as specified in chapter "SIP

Header Support" of 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface'.

2. Check that there are no other headers than mentioned above present in the INVITE message.

5.8.6   Session-Expires header

1. The Session-Expires header as per 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface' shall be set to 1800 in all relevant SIP messages (e.g. INVITE, 200 OK).

5.8.7   Request-URI format

1. Verify R-URI Format:
   - Over-decadic numbers are currently not supported.
   - A dial string may include only the following characters: 0-9,*,# ('#' character is represented as "%23").
   - Dial string is expressed in the User-URI portion of the SIP URI.

5.8.8   URI format

1. Verify that the URI format conforms to 'Anhang 1 zum Handbuch Technik VoIP Interkonnektion – Public Network-to-Network Interface' chapter 'SIP URI Addressing Scheme'.

5.8.9   SDP content

1. Verify the SDP content:
   - Verify that G.711 A-law with 20m packetisation is offered in the SDP negotiation.
   - Further codecs (except G.711 A-law) may be present in the SDP offer.
   - Dynamic payload shall not be used for standard codecs (unless explicitly foreseen in the codec standards reference documentation).
   - Voice Activity Detection (VAD) shall not be used unless mandatory by the codec profile.
   - Asymmetric packetisation may be used for all none-mandatory codecs (all except G.711 A-law) but must not be guaranteed by the partner network.
   - Video codecs may be offered by the partner network. (Media type Video is, however, currently not supported by Swisscom.)

5.9   Billing Data Comparison

1. At the end of testing period billing data shall be exchanged and compared to ensure consistency. Both parties must agree upon the billing data to be compared. It is recommended that the billing CDRs are used as a basis for this comparison. An example of the relevant billing data to be exchanged is the following:

a) Number of calls

b) Total call duration

2. The following test calls shall at minimum be performed at the end of the testing period in both directions (as applicable), in order to generate CDRs and to verify billing data:
   - 5.2.1 Normal call - Calling party clears after answer
     - Verify call duration in CDR.

- 5.2.2 Normal call - Called party clears after answer
  - o Verify call duration in CDR.
- 5.2.3 Normal call - Calling party release while ringing
  - o Verify that the call duration is zero (0) in CDR.
- 5.2.4 Normal call - No Answer / Timeout
  - o Verify that the call duration is zero (0) in CDR.
- 5.2.5 Normal call to Busy Subscriber / Calling Party Release
  - o Verify that the call duration is zero (0) in CDR.
- 5.2.6 Long duration call (call longer than 60 minutes)
  - o Verify that the total call duration is calculated correctly. If there are several CDRs, check that billing system processes it correctly.
- 5.2.7 Call with CLI Restriction - CLIR
  - o Verify call duration in CDR.
  - o Verify that A-Number appears in CDR in correct format (taken from PAI header).
- 5.2.8 CLI presentation when P-Asserted-Identity and From headers contain different numbers
  - o Verify call duration in CDR.
  - o Verify that A-Number appears in CDR in correct format (taken from PAI header).
- 5.2.11 Call Diversion using History-Info header
  - o Verify call duration in CDR.
  - o Swisscom additionally verifies that the correct redirecting number appears in CDR.
- 5.2.12 Call Diversion using Diversion header
  - o Verify call duration in CDR.
  - o Swisscom additionally verifies that the correct redirecting number appears in CDR.
- 5.2.13 Call to voice mail service (Combox)
  - o Verify call duration in CDR.
- 5.2.14 Call to a ported number
  - o Verify call duration in CDR.
- 5.4.1 Local ringbacktone - provisional response without SDP and PRACK
  - o Verify call duration in CDR (billing starts with 200 OK).
- 5.4.2 Remote ringbacktone with early media - provisional response with SDP and PRACK
  - o Verify call duration in CDR (billing starts with 200 OK).

6   Test Case Scenarios for Changes to an Existing VoIP Interconnection – Reduced Scope

1. If changes are done on an existing VoIP Interconnection a reasonable set of test cases should be agreed and executed in advance of the change and executed immediate after the change. The test cases may include all kind of tests, e.g. call tests, billing tests and redundancy tests.

7   Test Result Record Sheet for New VoIP Interconnection

1. The test results will be recorded in a Microsoft Excel table. This table will be used as well to define the

test scope of the test runs.

8 List of References

Reference	Title
[1] ETSI EG 202 057-2	"Speech processing transmission and quality aspects (STQ); user related QoS parameter definitions and measurements; Part 2: Voice Telephony, Group 3 Fax, modem data services and SMS"; October 2005
[2] i3 forum Interoperability Test Plan	Interoperability Test Plan for International Voice services (Release 6) May 2014

9 Abbreviations and Acronyms

CDR	Call Detail Record
CLI	Calling Line Identity
CLIR	Calling Line Identity Restriction
DTMF	Dual-Tone Multi-Frequency
ETSI	European Telecommunications Standards Institute
IETF	Internet Engineering Task Force
ISUP	ISDN User Part
MNO	Mobile Network Operator
NNI	Network-to-Network Interface
NPRN	Number Portability Routing Number
RFC	Request for Comments
RTP	Real-Time Transport Protocol
SBC	Session Border Controller
SDP	Session Description Protocol
SIP	Session Initiation Protocol
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
URI	Uniform Resource Identifier

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