This document has been created on May 23rd, 2008 by the i3 Forum.
Its content can be used and disclosed to any third party provided that the i3 Forum is clearly mentionned as the source

INTERNATIONAL INTERCONNECT FORUM FOR SERVICES OVER IP

i3 FORUM

Interoperability Test Plan for bilateral Voice services (V.1.0) May 2008

This document has been created on May 23rd, 2008 by the i3 Forum.
Its content can be used and disclosed to any third party provided that the i3 Forum is clearly mentionned as the source

Table of Contents

1 Document Scope **4**
2 Assumptions **4**
3 Acronyms **4**
4 Reference **5**
5 Test Strategy **5**
5.1 High Level Configuration 5
5.2 Typical Basic VoIP Call Flow between Carriers 6
5.3 Contact Information 7
5.4 Lab Hardware and Software Configuration 7
6 Acceptance Test Techniques **7**
6.1 Pre-Service Inter-Op Testing Entrance Criteria 7
6.2 Test Exit Criteria 7
7 Test case scenarios **8**
7.1 Initial IP Testing between Carrier A and Carrier B 8
7.1.1 Ping Tests towards agreed IP addresses 8
7.1.2 Round trip delay measurement 8
7.1.3 Packet loss measurement 8
7.1.4 Packet jitter measurement 8
7.2 Basic Call Flow and Basic Fax Tests for Carrier A and B 8
7.2.1 Carrier A/B originating normal call release - Calling party clears after answer 8
7.2.2 Carrier A/B originating normal call release - Calling party abandon 8
7.2.3 Carrier A/B originating normal call release - Called party clears after answer 8
7.2.4 Carrier A/B originating normal call setup to ring No Answer / Timeout .. 8
7.2.5 Carrier A/B originating normal call setup to Busy Line / Calling party release 8
7.2.6 Carrier A/B originating verify proper handling for no route to destination 8
7.2.7 Verify proper handling for vacant code in foreign network 8
7.2.8 Carrier A/B Originating Fax 8
7.2.9 Carrier A/B originating modem 8
7.2.10 DTMF – Verify digits for a RFC2833 [4] transmission to a network based call prompter (prior to answer) for a G.729 ([5], [6], [7], [8]) call 8
7.2.11 DTMF – Verify digits received for a RFC2833 transmission post answer from the caller for a G.729 call 8
7.2.12 DTMF – Verify digits received for a RFC2833 transmission post answer from the callee for a G.729 call 9
7.2.13 DTMF – Verify digits for a RFC2833 transmission to a network based call prompter (prior to answer) for a G.711 [9] call 9
7.2.14 DTMF – Verify digits received for a RFC2833 transmission post answer from the caller for a G.711 call 9
7.2.15 DTMF – Verify digits received for a RFC2833 transmission post answer from the callee for a G.711 call 9
7.2.16 SIP Options messages to the Carrier 9
7.3 Voice Quality Tests for Carrier A and B 9
7.3.1 Idle Channel Noise as specified in [11], [12] 9
7.3.2 Echo Performance as specified in [13] 9
7.3.3 Average Speech Level 9

This document has been created on May 23rd, 2008 by the i3 Forum.
Its content can be used and disclosed to any third party provided that the i3 Forum is clearly mentionned as the source

7.3.4 Speech Loss as specified in [14] 9
7.3.5 Mean Opinion Score MOS/PESQ measurement as specified in [2], [3] 9
7.3.6 Post Gateway Ringing Delay 9
7.4 Fax quality test 9
7.4.1 Verify the fax image quality is not deteriorate [10] 9
7.4.2 Verify the fax transmission throughput is acceptable for G3 fax [10] 9
7.5 Scenarios for Mobile Services Over IP Interconnections 10
7.5.1 Mobile Phone Roaming in Country A Receives Calls from Carrier B and Answers the Call 10
7.5.2 Mobile Phone Roaming in Country B Receives Calls from Carrier A and Answers the Call 10
7.5.3 Call Mobile Phone Roaming in Country A is redirected to Voice Mail Box when Called Party is Busy or not Answering 10
7.5.4 Call Mobile Phone Roaming in Country B is redirected to Voice Mail Box when Called Party is Busy or not Answering 10
8 TEST RESULT Record Sheet 11
8.1 Testing Information 11
8.2 Initial IP Testing between Carrier A and Carrier B 11
8.3 Basic Call Flow and Basic Fax Tests for Carrier A and B 12
8.4 Voice Quality Tests for Carrier A and B 13
8.5 Fax quality test 13
8.6 Tests for Mobile Services Over IP Interconnections 13

This document has been created on May 23rd, 2008 by the i3 Forum.
Its content can be used and disclosed to any third party provided that the i3 Forum is clearly mentionned as the source

1 Document Scope

This Pre-Service Inter-Operability Test Plan aims to provide tools in order ~~is~~ to establish a new transmission bilateral path between international VoIP carriers, ensuring signaling compatibly and providing quality and performance levels that meets customer expectations.

This document covers the test approach, specific functionality, assumptions, and test cases that will be performed for the pre-service inter-operability between two international VoIP carriers before the delivery of customer traffic.

Test cases in this document will cover calls in both directions.

Both carriers will capture and record call traces for each of the test call scenarios and verify that the SIP Messages are in compliance with [1]. Both carriers will collect and exchange CDRs for the test calls for billing verification.

The following items are not in the scope of this document:

- How the call traces and Call Detail Records are captured
- What equipment is used to capture call traces and CDR.

2 Assumptions

This testing shall be performed on a network configuration as documented in [1].

Before starting a test plan, each carrier will provide testing phone numbers that will terminate on phones with Caller ID so that CPN, (also abbreviated as CLI,) can be verified.

Before starting a test plan, each carrier will provide testing phone numbers that will terminate on various Fax machines (G.3, superG3).

3 Acronyms

CDR	Call Detail Record
CLI	Calling Line Identity
CPN	Calling Party Number
MOS	Mean Opinion Score
PESQ	Perceptual evaluation of speech quality

This document has been created on May 23rd, 2008 by the i3 Forum.
Its content can be used and disclosed to any third party provided that the i3 Forum is clearly mentionned as the source

4 Reference

[1] IP-IP Forum "Interconnection Model for bilateral Voice Services", V.1, May 2008
[2] ITU-T – P.800.1 "Mean Opinion Score (MOS) terminology" (07/2006)
[3] ITU-T – P.862 "Perceptual evaluation of speech quality (PESQ): An Objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs" (02/200)
[4] IETF RFC 2833, "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals", May 2000
[5] ITU-T Recommendation G.729 "Coding of speech at 8 kbit/s using conjugate-structure algebraic code excited linear-prediction (CS-ALEP (03/96)
[6] ITU-T Recommendation G.729 Annex A "Reduced complexity 8kbit/s CS-ALEP codec" (11/96)
[7] ITU-T Recommendation G.729 Annex B Silence compression scheme for G.729 optimized for terminals conforming to Recommendation V.70" (11/96)
[8] ITU-T Recommendation G.729 Annex A and B
[9] ITU-T Recommendation G.711 "Pulse Code Modulation of Voice Frequencies", 1988
[10] 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.
[11] ITU-T Recommendation G.177 "Transmission Planning for Voiceband Services over Hybrid Internet/PSTN Connections", September 1999
[12] ITU-T Recommendation G.223 "Assumptions for the Calculation of Noise on Hypothetical Reference Circuits for Telephony-International Analogue Carrier Systems 9 pp", November 1988
[13] ITU-T Recommendation G.168 "Digital Network Echo Cancellers", April 2000
[14] ITU-T Recommendation E.855 "Connection integrity objective for the international telephone service", November 1988

5 Test Strategy

5.1 High Level Configuration

The diagram below describes the high level configuration between the participating carriers to be used during the interoperability testing. An interconnection form/template has to be exchanged between the carriers covering service and networks issues; the definition of this form/template is not in the scope of this document.

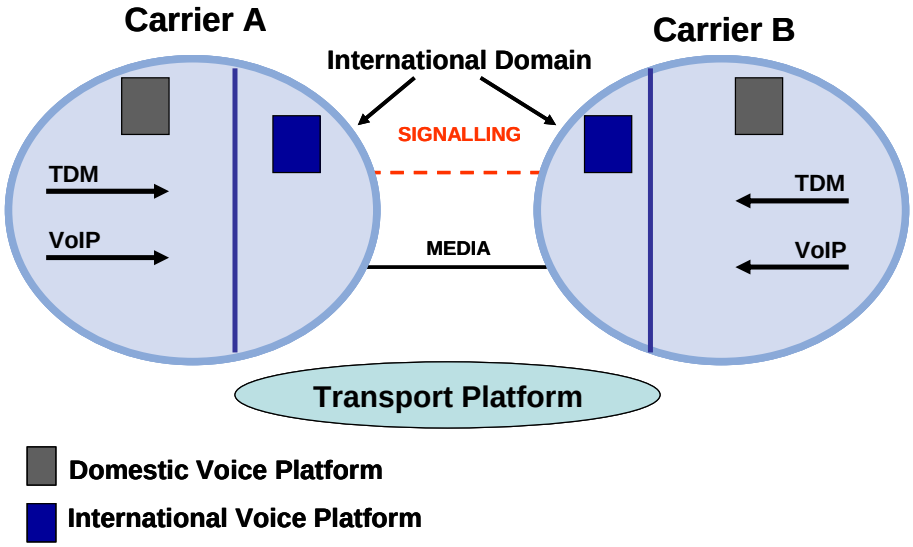

Figure 1 – General Reference Configuration

This document has been created on May 23rd, 2008 by the i3 Forum.
Its content can be used and disclosed to any third party provided that the i3 Forum is clearly mentionned as the source

5.2 Typical Basic VoIP Call Flow between Carriers

The following diagram depicts a typical SIP message flow for a VoIP call between carriers.

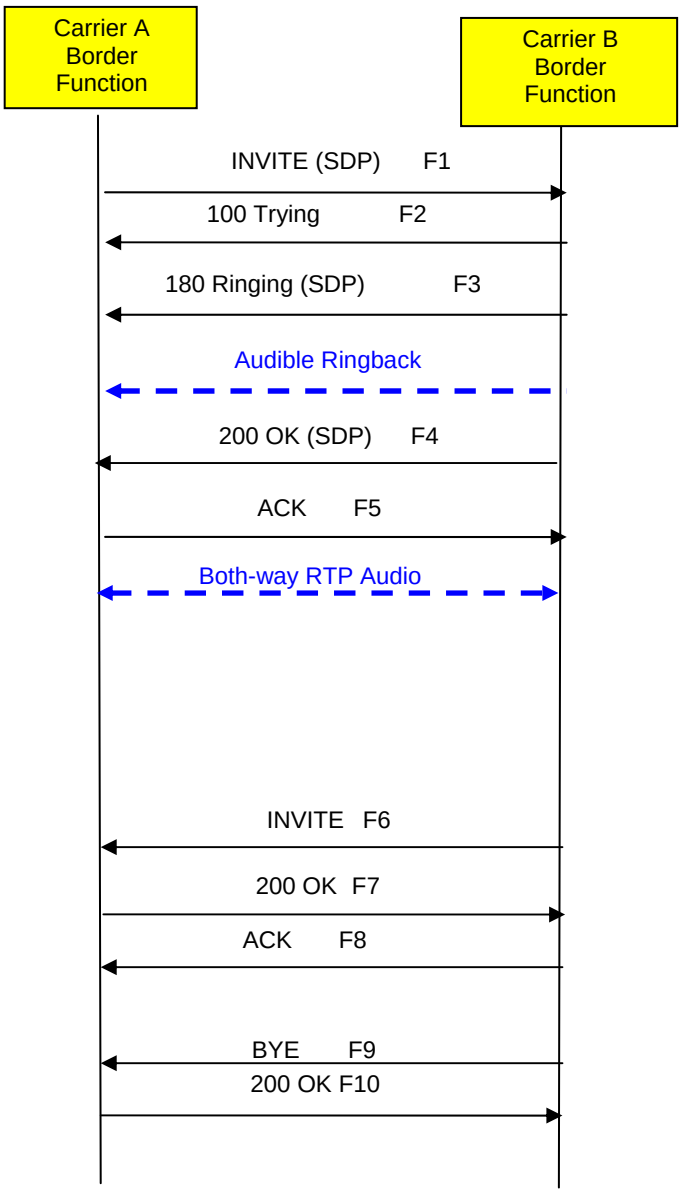

Figure 2 – Basic VoIP Call Flow

This document has been created on May 23rd, 2008 by the i3 Forum.
Its content can be used and disclosed to any third party provided that the i3 Forum is clearly mentionned as the source

SIP uses the following methods:

- INVITE—Indicates that a user or service is being invited to participate in a call session. INVITE F6 is sent to refresh the session prior to expiration of the Session interval for long duration calls.
- OK—Indicates the request has succeeded.
- BYE—Terminates a call and can be sent by either the caller or the called party.
- CANCEL—Cancels any pending searches but does not terminate a call that has already been accepted.
- OPTIONS—Queries the capabilities of servers.
- REGISTER—Registers the address listed in the To header field with a SIP server.
- REFER—Indicates that the user (recipient) should contact a third party for use in transferring parties.
- NOTIFY—Notifies the user of the status of a transfer using REFER. Also used for remote reset.

The following types of responses are used by SIP:

- SIP 1xx—Informational responses
- SIP 2xx—Successful responses
- SIP 3xx—Redirection responses
- SIP 4xx—Client failure responses
- SIP 5xx—Server failure responses
- SIP 6xx—Global failure responses

5.3 Contact Information

Two carriers shall exchange the contact information of testing personnel.

5.4 Lab Hardware and Software Configuration

Two carriers shall exchange and document the hardware and software versions of elements to be used during the interoperability testing. The information exchange will help trouble shooting during testing.

Carrier A	Carrier B
Border Function Vendor X, Releases X	Border Function Vendor Y, Releases Y
Call Generator and packet sniffer Vendor X	Call Generator Vendor and packet sniffer Y

6 Acceptance Test Techniques

6.1 Pre-Service Inter-Op Testing Entrance Criteria

- The transmission connectivity is available as per Service Reference Configuration.
- Testing phone numbers have been exchanged for phone and fax. If the IP interconnection will carry mobile traffic, then mobile phone numbers, voice mail number may need to be exchanged.

6.2 Test Exit Criteria

The test cycle will end when all of the following conditions are met:

- 100% of the identified test cases are successfully completed or mutually agreed that they won't be executed.
- The CDRs for each test call are collected by both carriers, exchanged and compared to ensure consistency.

This document has been created on May 23rd, 2008 by the i3 Forum.
Its content can be used and disclosed to any third party provided that the i3 Forum is clearly mentionned as the source

7 Test case scenarios

CDRs are captured to verify whether a test case should be passed. Call traces could be exchanged if needed; for example, call traces can be used to verify which codec was negotiated.

7.1 Initial IP Testing between Carrier A and Carrier B

7.1.1 Ping Tests towards agreed IP addresses

7.1.2 Round trip delay measurement

7.1.3 Packet loss measurement

7.1.4 Packet jitter measurement

7.2 Basic Call Flow and Basic Fax Tests for Carrier A and B

7.2.1 Carrier A/B originating normal call release - Calling party clears after answer

- Check IP packet TOS marking
- Check selected codec
- Check that at least 10 calls to several testing dialed numbers, including mobile terminations.
- Check the “long call” duration (e.g. calls shall last longer than 15 minutes)

7.2.2 Carrier A/B originating normal call release - Calling party abandon

7.2.3 Carrier A/B originating normal call release - Called party clears after answer

7.2.4 Carrier A/B originating normal call setup to ring No Answer / Timeout

7.2.5 Carrier A/B originating normal call setup to Busy Line / Calling party release

7.2.6 Carrier A/B originating verify proper handling for no route to destination

7.2.7 Verify proper handling for vacant code in foreign network

7.2.8 Carrier A/B Originating Fax

- G3 to G3 Fax call
- SuperG3 to SuperG3 Fax call

7.2.9 Carrier A/B originating modem

7.2.10 DTMF – Verify digits for a RFC2833 [4] transmission to a network based call prompter (prior to answer) for a G.729 ([5], [6], [7], [8]) call

7.2.11 DTMF – Verify digits received for a RFC2833 transmission post answer from the caller for a G.729 call

This document has been created on May 23rd, 2008 by the i3 Forum.
Its content can be used and disclosed to any third party provided that the i3 Forum is clearly mentionned as the source

7.2.12 DTMF – Verify digits received for a RFC2833 transmission post answer from the callee for a G.729 call

7.2.13 DTMF – Verify digits for a RFC2833 transmission to a network based call prompter (prior to answer) for a G.711 [9] call

7.2.14 DTMF – Verify digits received for a RFC2833 transmission post answer from the caller for a G.711 call

7.2.15 DTMF – Verify digits received for a RFC2833 transmission post answer from the callee for a G.711 call

7.2.16 SIP Options messages to the Carrier
If both carriers agree to use the SIP Options message as a mean of determining the reachability.

7.3 Voice Quality Tests for Carrier A and B

As a general requirement, the VoIP interconnection should achieve the same quality levels as provided by TDM interconnections.

Many different international interconnection configurations are possible [1] and each international configuration can generate different end-to-end scenarios (e.g. from TDM to TDM, from IP to TDM, from IP to IP).

As far as the testing model is concerned, no definitive and complete reference model has been yet standardized. As a result, in the following the most relevant parameters are suggested for testing. The actual parameters have to be agreed by the interconnecting carriers.

7.3.1 Idle Channel Noise as specified in [11], [12]

7.3.2 Echo Performance as specified in [13]

7.3.3 Average Speech Level

7.3.4 Speech Loss as specified in [14]

7.3.5 Mean Opinion Score MOS/PESQ measurement as specified in [2], [3]

7.3.6 Post Gateway Ringing Delay

7.4 Fax quality test

7.4.1 Verify the fax image quality is not deteriorate [10]

7.4.2 Verify the fax transmission throughput is acceptable for G3 fax [10]

This document has been created on May 23rd, 2008 by the i3 Forum.
Its content can be used and disclosed to any third party provided that the i3 Forum is clearly mentionned as the source

7.5 Scenarios for Mobile Services Over IP Interconnections

If the IP interconnections between carriers shall carry mobile calls, the following test scenarios may be performed to verify mobile calls can be supported.

7.5.1 Mobile Phone Roaming in Country A Receives Calls from Carrier B and Answers the Call

- Verify the caller ID is presented to the roaming phone.

7.5.2 Mobile Phone Roaming in Country B Receives Calls from Carrier A and Answers the Call

- Verify the caller ID is presented to the roaming phone.

7.5.3 Call Mobile Phone Roaming in Country A is redirected to Voice Mail Box when Called Party is Busy or not Answering

- Verify the caller does not receive open-tree treatment and the subscriber's voice mail box is reached.

7.5.4 Call Mobile Phone Roaming in Country B is redirected to Voice Mail Box when Called Party is Busy or not Answering

This document has been created on May 23rd, 2008 by the i3 Forum.
Its content can be used and disclosed to any third party provided that the i3 Forum is clearly mentionned as the source

8 TEST RESULT Record Sheet

8.1 Testing Information

	Carrier A	Carrier B
Name		
Date of Test		
IP address for Ping		
Testing personnel contact		
Test contact number		
Testing number(s)		
Test fax number		

8.2 Initial IP Testing between Carrier A and Carrier B

	Pass/Fail	Comments
Initial Test		
Ping Test		
Round Trip Delay		
Packet Loss		
Packet Jitter		

The target values of the above parameters are subject to a specific agreement between the two interconnecting carriers depending on the selected interconnection configuration and the selected technical options.

This document has been created on May 23rd, 2008 by the i3 Forum.
Its content can be used and disclosed to any third party provided that the i3 Forum is clearly mentionned as the source

8.3 Basic Call Flow and Basic Fax Tests for Carrier A and B

Test case number	Test case	Pass/Fail	Comments
7.2.1	Originating Normal Call Release - Calling Party Clears After Answer		
7.2.2	Originating Normal Call Release - Calling Party Abandon		
7.2.3	Originating Normal Call Release - Called Party Clears After Answer		
7.2.4	Normal Call Setup to Ring No Answer / Timeout		
7.2.5	Originating Normal Call Setup to Busy Line / Calling Party Release		
7.2.6	Originating Verify Proper handling for no route to destination		
7.2.7	Verify Proper handling for vacant code in foreign network		
7.2.8	Originating Fax		
7.2.9	Originating Modem		
7.2.10	DTMF – Verify digits for a RFC2833 transmission to a network based call prompter (prior to answer) for a G.729 call		
7.2.11	DTMF – Verify digits received for a RFC2833 transmission post answer from the caller for a G.729 call		
7.2.12	DTMF – Verify digits received for a RFC2833 transmission post answer from the callee for a G.729 call		
7.2.13	DTMF – Verify digits for a RFC2833 transmission to a network based call prompter (prior to answer) for a G.711 call		
7.2.14	DTMF – Verify digits received for a RFC2833 transmission post answer from the caller for a G.711 call		
7.2.15	DTMF – Verify digits received for a RFC2833 transmission post answer from the callee for a G.711 call		
7.2.16	SIP Options messages to the Carrier		

This document has been created on May 23rd, 2008 by the i3 Forum.
Its content can be used and disclosed to any third party provided that the i3 Forum is clearly mentionned as the source

8.4 Voice Quality Tests for Carrier A and B

Test case number	Objective Tests	Pass/Fail	Comments
7.3.1	Idle Channel Noise		
7.3.2	Echo Performance		
7.3.3	Average Speech Level		
7.3.4	Speech Loss		
7.3.5	Mean Opinion Score /PESQ Measurement		
7.3.6	PGRD		

The target values of the above parameters are subject to a specific agreement between the two interconnecting carriers depending on the selected interconnection configuration and the selected technical options.

8.5 Fax quality test

Test case number	Objective Tests	Pass/Fail	Comments
7.4.1	fax image quality		
7.4.2	Transmission Throughput		

8.6 Tests for Mobile Services Over IP Interconnections

Test case number	Test case	Pass/Fail	Comments
7.5.1	Mobile Phone Roaming in Country A Receives Calls from Carrier B and Answers the Call		
7.5.2	Mobile Phone Roaming in Country B Receives Calls from Carrier A and Answers the Call		
7.5.3	Call Mobile Phone Roaming in Country A is Redirected to Voice Mail Box when Called Party is Busy or not Answering		
7.5.4	Call Mobile Phone Roaming in Country B is Redirected to Voice Mail Box when Called Party is Busy or not Answering		