INTERNATIONAL INTERCONNECTION FORUM FOR SERVICES OVER IP

(i3 FORUM)

Workstream "Technical Aspects"

Workstream "Operations"

Interoperability Test Plan for International Voice services

(Release 3.0) May 2010

Table of Contents

1 Document Scope

This Pre-Service Inter-Operability Test Plan aims to provide testing guidance when establishing a new transmission bilateral path between international VoIP carriers, ensuring signaling compatibly and providing quality and performance levels that meet customer quality expectations.

This document covers the test approach, specific functionality, assumptions, and test cases that should 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].

Prior to initiating a test plan, each carrier will provide testing numbers that will terminate on phones with Caller ID so that CIN, (also abbreviated as CLI,) can be verified.
In addition, each carrier will provide testing numbers that will terminate on various Fax machines (G.3, superG3).

Lastly, both carriers will agree to the expected successful test parameters for each of the test scenarios.

3 Acronyms

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

4 Reference

[1] i3 Forum "Technical Interconnection Model for International Voice Services", rel...2, May 2009

[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. The Interconnection Form for International Voice Services will be exchanged between the carriers covering service and network requirements; the definition of this form/template is not in the scope of this document.

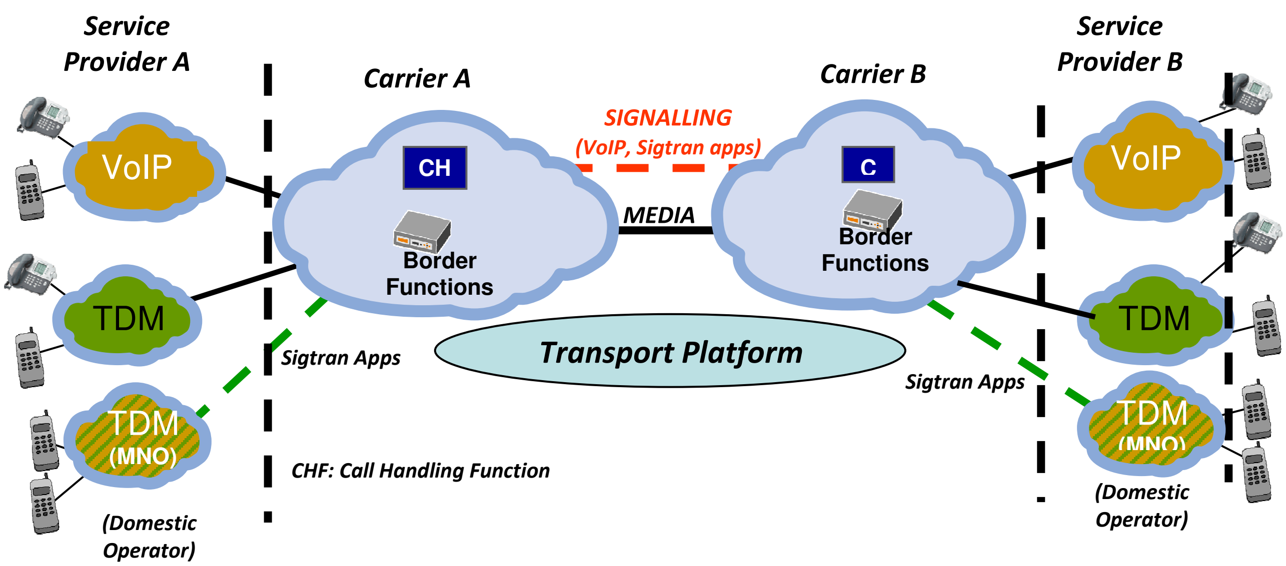

Figure 1 – General Reference Configuration

5.2 Typical Basic VoIP Call Flow between Carriers

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

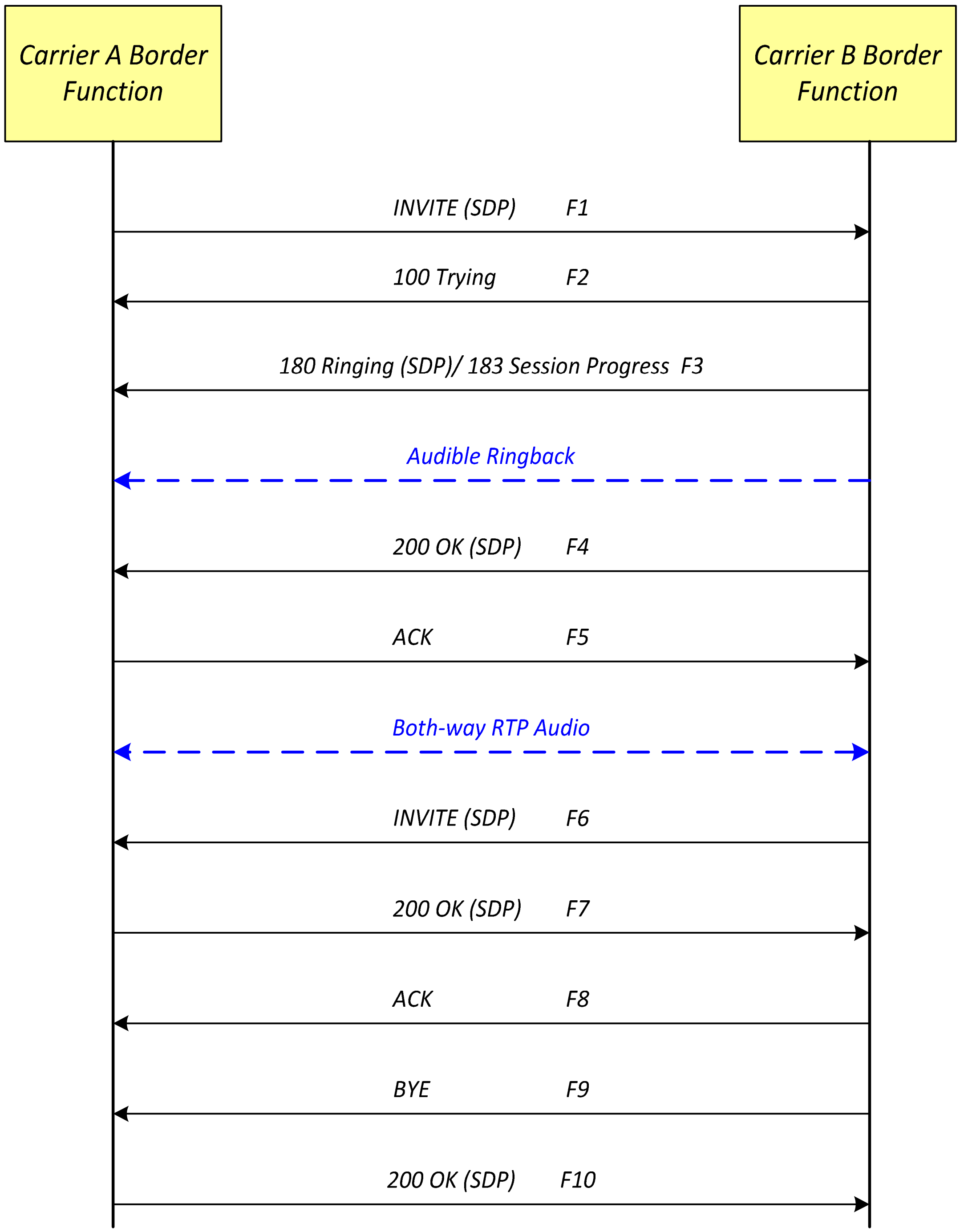

Figure 2 – Basic VoIP Call Flow

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 as part of The Interconnection Form for International Voice Services.

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 aid troubleshooting during the testing phase of the interconnection.

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 (CIN) and fax. If the IP interconnection will carry mobile traffic, then mobile phone numbers, including roaming and voicemail service 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 jointly identified and agreed to test cases are successfully completed*
- *CDRs for each of the required test cases are collected by both carriers, exchanged and compared to ensure consistency.*

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 selected codec*
- *Repeat with all agreed upon codecs*
- *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 Carrier A/B originating verify proper handling for unallocated number

7.2.8 Carrier A/B originating verify Insufficient digits (Partial dial)

7.2.9 Carrier A/B originating modem (If Applicable)

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

7.2.11 DTMF – Verify digits received for a RFC2833 transmission post answer from the caller for another codec (G.729 for example) call

7.2.12 Calling Party Number - Need to verify that CLI is properly passed and received in the correct agreed upon format

7.2.13 Called Party Number – Need to verify that the called party number is passed and received in the correct agreed upon format

7.2.14 CLI Restriction presentation (CLIR) – (Only if agreed upon by both parties)

In case of SIP:
- Check FROM header should be "Anonymous@anonymous.invalid"
- Check P-asserted-identity header "+CC NSN@X.X.X.X;user=phone"
- Check Privacy header "id, header or user"

In case of SIP-I: - check Calling Party Number: "Presentation restricted"
- Check Generic Number presentation: "Presentation restricted"

7.2.15 SIP Options messages to the Carrier

If both carriers agree to use the SIP Options message as a mean of determining the reachability. Refers to SBC for ping at SIP level (testing of layer 5)

7.3 Voice Quality Tests for Carrier A and B

As a general requirement, the VoIP interconnection testing results should achieve the same quality levels as supported 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 measurements/quality levels for each parameter have to be agreed to by the interconnecting carriers.

7.3.1 Mean Opinion Score MOS/PESQ measurement as specified in [2], [3](If applicable)

7.3.2 Post Gateway Ringing Delay

7.4 Fax quality test

7.4.1 Verify that fax image quality is not deteriorated [10]

7.4.2 Verify that 3+ page fax completes

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

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.

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* • *Check selected codec* • *Repeat with all agreed upon codecs* • *Check that at least 10 calls to several testing dialed numbers including mobile complete properly* • *Check the "long call" duration (e.g. calls shall last Langer then 15 minutes)*		
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 unallocated number*		
7.2.8	*Verify proper handling for Insufficient Digits*		
7.2.9	*Carrier A/B originating modem (If Applicable)*		
7.2.10	*DTMF – Verify digits received for a DTMF transmission (using agreed upon DTMF method) post answer from the caller for a G.711 call*		
7.2.11	*DTMF – Verify digits received for a RFC2833 transmission post answer from the caller for another codec (G.729 for example) call*		
7.2.12	*Calling Party Number - Need to verify that CLI is properly passed and received in the agreed upon format*		
7.2.13	*Called Party Number – Need to verify that the called party number is received in the agreed upon format*		
7.2.14	*CLI Restriction presentation (CLIR) – (Only if agreed upon by both parties)* *In case of SIP:*		

	- Check FROM header should be "Anonymous@anonymous.invalid" *- Check P-asserted-identity header "+CC NSN@X.X.X.X;user=phone"* *- Check Privacy header "id, header or user"* *In case of SIP-I: - check Calling Party Number: "Presentation restricted"* *- Check Generic Number presentation: "Presentation restricted"*		
7.2.15	*SIP Options messages to the Carrier* *If both carriers agree to use the SIP Options message as a mean of determining the reachability. Refers to SBC for ping at SIP level (testing of layer 5)*		

8.4 Voice Quality Tests for Carrier A and B

Test case number	***Objective Tests***	***Pass/Fail***	***Comments***
7.3.1	*Mean Opinion Score /PESQ Measurement*		
7.3.2	*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*		
0	*Verify that 3+ page fax completes*		
7.4.3	*Transmission Throughput verify baud rate acceptable*		