INTERNATIONAL INTERCONNECTION FORUM
FOR SERVICES OVER IP

(i3 FORUM)

Workstream "Technical Aspects"
Workstream "Operations"

Interoperability Test Plan for International Voice services

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Document Scope

This pre-service Interoperability Test Plan provides testing guidance for establishing a new bilateral interconnection between international voice carriers, ensuring signaling compatibility, quality and performance levels that meet customer expectations.

This document covers the test approach, specific functionality, assumptions, and test cases for verifying the interconnection between two international VoIP carriers before the delivery of customer traffic with a focus on voice services.

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 for comparison to *i3F Technical Interconnection Model for International Voice Services* [1].
Both carriers will collect and exchange CDRs of 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
- Service quality testing – It is assumed that once this VoIP connection is successfully validated, that the order will go into production testing for destination and quality verification.

Assumptions

a) Prior to testing, the technical and business aspects of the intended interconnection are to be first documented in a formal understanding, e.g., Commercial Agreement.
b) Both carriers will agree to the test scenarios, test plan / process, back-out plan, and the expected test results used to verify the test was successful.
c) Testing shall be performed on a network configuration as documented in *i3F Technical Interconnection Model for International Voice Services* [1].
d) Carriers will exchange *i3F Migration Interconnection Form* [2] to agree on configuration details and to verify system compatibility
e) Each carrier will provide testing numbers that will terminate on phones with Caller ID so that CIN (also abbreviated as CLI) can be verified.
f) Each carrier will provide testing numbers that will terminate on various Fax machines (G.3, superG3).
g) Carries must agree to the CDR format that will be exchanged (either raw or billing/processed CDRs)

Acronyms

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

Reference

[1] i3F Technical Interconnection Model for International Voice Services, rel 6.0, May 2014
[2] i3F Migration Interconnection Form, rel 6.0, May 2014
[3] IETF RFC 3261, "SIP: Session Initiation Protocol", June 2002
[4] 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
[5] ITU-T – P.800.1 "Mean Opinion Score (MOS) terminology" (07/2006)
[6] 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).

Test Strategy

1.1 High Level Configuration

The diagram below describes the high level configuration between the participating carriers to be used during interoperability testing.
The *i3F Migration Interconnection Form* [2] will be exchanged between the carriers covering service and network requirements.

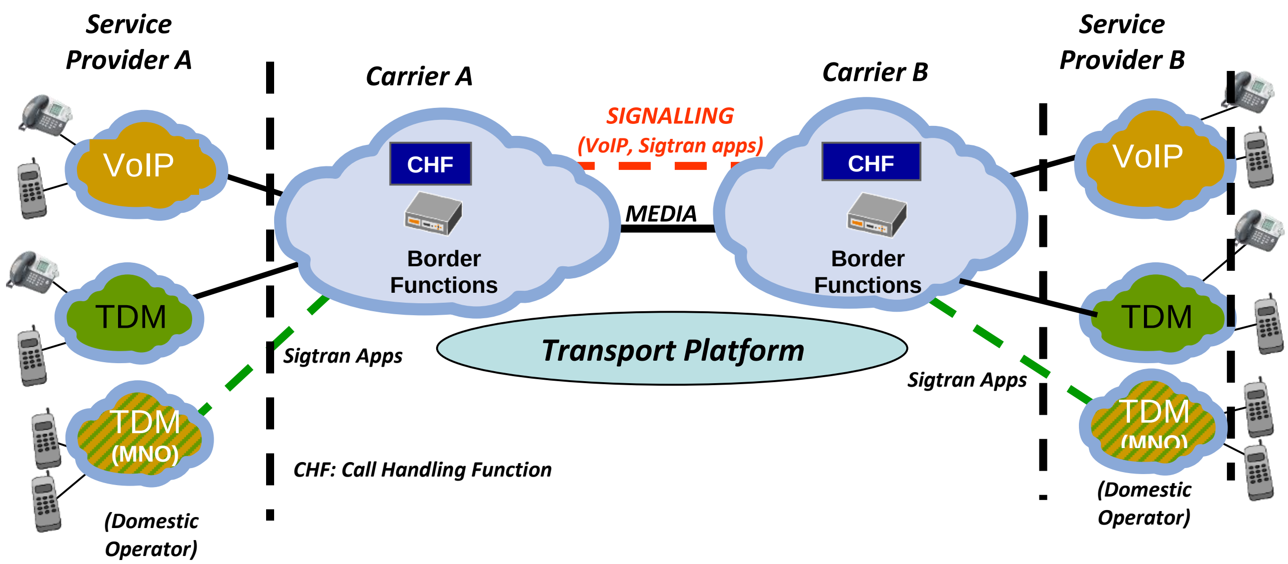

Figure 1 – General Reference Configuration

1.2 Typical Basic VoIP Call Flow between Carriers

The following diagram depicts a typical SIP message flow for a VoIP call between carriers, according to SIP specification *IETF RFC 3261* [3].

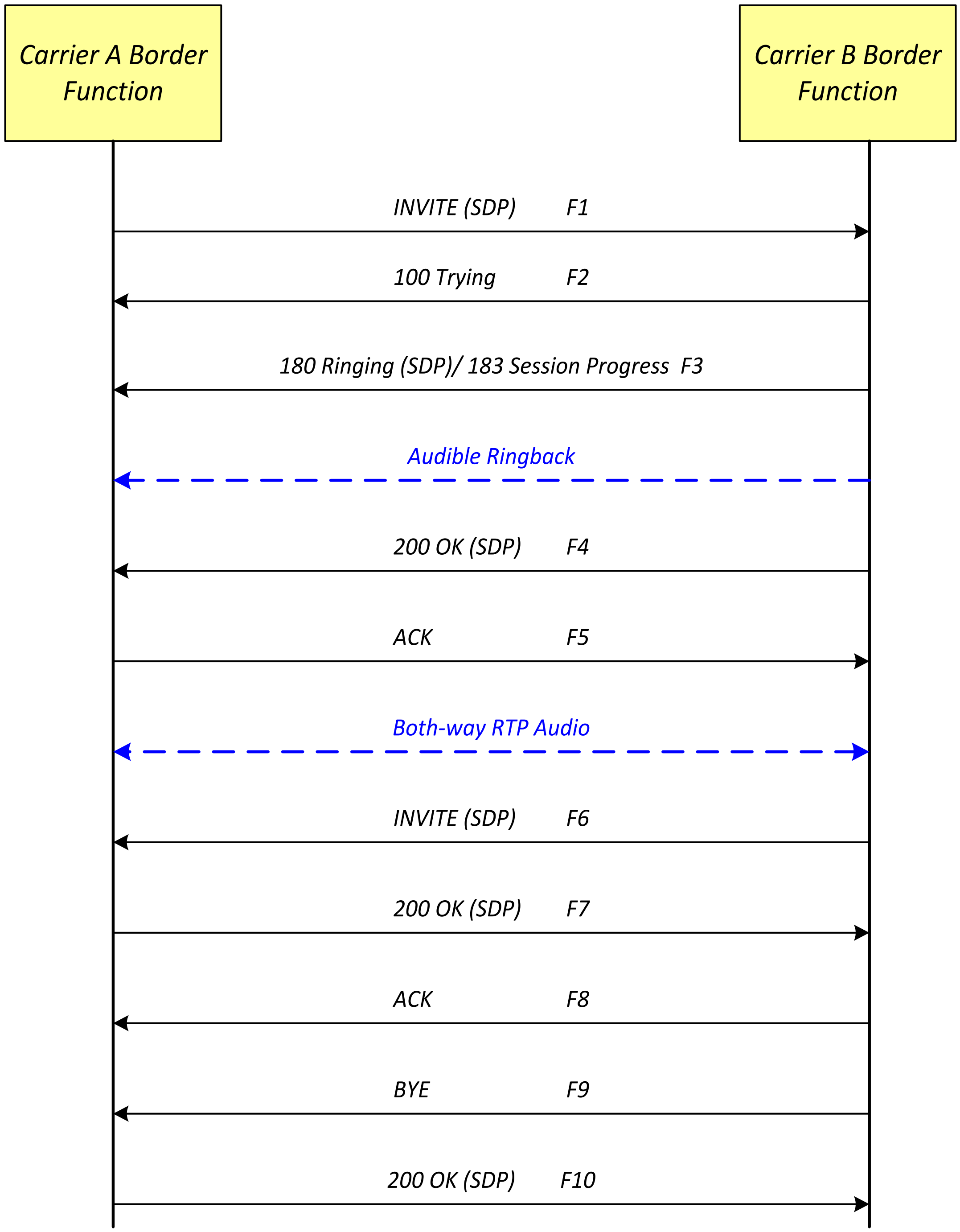

Figure 2 – Basic VoIP Call Flow

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

1.3 Contact Information

Both carriers shall exchange the contact information of testing personnel as part of the *i3Forum Migration Interconnection Form* [2].

1.4 Hardware and Software Configuration

Both carriers shall exchange and document the hardware and software versions of elements to be used during the interoperability testing. In a typical VoIP interconnect each party should provide the vendor and model of the Session Border Controller and Softswitch to be used. The information exchange will aid troubleshooting during the testing phase of the interconnection.

Carrier A	Carrier B
Border Function Vendor X, Release X	Border Function Vendor Y, Release Y
Soft switch vendor, release	Soft switch vendor, release
Call Generator and packet sniffer Vendor X	Call Generator and packet sniffer Vendor Y

Acceptance Test Techniques

Generally, a test plan technique that satisfies each carrier'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 completion.

1.5 Testing entrance criteria

The transmission connectivity is available as per Transport Reference Configuration in the *i3F Migration Interconnection Form* [2].
The test scenarios to be used, the test plan / process, the back-out plan, and the expected test results are established.
Testing phone numbers have been exchanged for voice and fax. If the IP interconnection will carry mobile traffic, then mobile phone numbers, including roaming and voicemail service may also need to be exchanged.

1.6 Test operative practices

Besides direct feedback from partner carrier'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 passed.

1.7 Test exit criteria

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
- CDRs for each of the required test cases are collected by both carriers, exchanged and compared to ensure consistency. It is recommended that billing CDR records be used for validation.

Test case scenarios for new interconnection

The target values of the below parameters are subject to a specific agreement between the two interconnecting carriers, depending on the selected interconnection configuration and the selected technical options. It is recommended that each test should be performed with all relevant codec's that are to be used for production traffic. Verify the following:

- Calls completed with each codec that is to be supported on the interconnect (both narrowband and wideband)
- Review and validate release codes for each test trial

1.8 Voice call flow testing

1.8.1 Normal call release - Calling party clears after answer

- Place test call and verify it completes. The calling party should hang up call. Verify that the call clears normally and the correct release code is received

1.8.2 Normal call release - Called party clears after answer

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

1.8.3 Normal call release - Calling party release while ringing

- 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 that there is a 0 duration CDR and the correct release code is received

1.8.4 Normal call setup to ring No Answer / Timeout

- Place test call. Do not allow the call to complete. Verify that the call clears correctly, there is a 0 duration CDR, and the correct release code is received

1.8.5 Normal call setup to Busy Line / Calling party release

- Place test call. The far side should be busy/off hook. Verify that the call does not complete and that it clears correctly, that there is a 0 duration CDR, and the correct release code is received

1.8.6 Verify proper handling for No Route To Destination

- Place test call. The far termination switch should not have a valid route/capacity available and should reject the call. Verify that the call clears correctly, that there is a 0 duration CDR and the correct release code is received

1.8.7 Verify proper handling for Unallocated Number

- Place test call. The # should not be valid. Verify that the call clears correctly, there is a 0 duration CDR, and the correct release code is received

1.8.8 Verify proper handling for Insufficient Digits (Partial dial)

- Place test call. The # should not be valid. Verify that the call clears correctly, there is a 0 duration CDR and the correct release code is received

1.8.9 Verify "long call" duration (e.g. calls longer than 15 minutes)

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

1.8.10 DTMF – Verify DTMF transmission

- Verify digits passed and received for a DTMF transmission (using agreed upon DTMF method) post answer from the caller for a G.711 call
- Verify digits passed and received for a DTMF transmission (using agreed upon DTMF method) post answer from the caller for another codec call (G.729 for example) call

1.8.11 CLI (calling party number) – Verify that CLI passed and received

1.8.12 Called party number – Verify that the called party number is properly passed and received in the agreed upon format

- Typical format is E.164

1.8.13 CLI Restriction presentation – CLIR (only if agreed upon by both parties)

In case of SIP:
- Check FROM header: "Anonymous@anonymous.invalid"
- Check P-asserted-identity header: "+CC NSN@xxx.xxx.xxx.xxx; 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"

1.8.14 Reachability and keepalive mechanism

- Out of dialog - SIP Options messages (only if agreed upon by both parties). SIP message OPTIONS shall be used to probe signaling reachability during testing phase and as keepalive mechanism in production phase
- It is recommended that for both parties capture network trace and review messages

1.9 Fax Testing

1.9.1 Fax – Carrier A/B originating fax - no codec change

- Verify fax transmission using G.711 pass-through for a G.711 voice originating call

1.9.2 Fax – Carrier A/B originating fax - with codec change

- Verify fax transmission using T.38 for a G.711 voice originating call
- Verify fax transmission using T.38 for another codec (e.g. G.729) voice originating call
- Verify fax transmission using G.711 pass-through for another codec (e.g. G.729) voice originating call

1.9.3 Verify that fax image quality is not deteriorated

- Test *per ETSI EG 202 057-2 "Speech processing transmission and quality aspects (STQ)"* [4]

1.9.4 Verify that 3+ page fax completes

1.10 CDRs comparison

At the end of testing period CDRs shall be exchanged and compared to ensure consistency. Both parties must agree upon the CDR format to be compared. It is recommended that the billing CDR's be used.

An example of the relevant CDR fields to be exchanged is the following:

a) Calling number
b) Called number
c) Start date/time
d) Stop date/time
e) Call duration
f) Release code

1.11 SIP Invite trace

It is recommended that the SIP Invite message be reviewed and validated. External tools may be needed to record and save this trace.

Test case scenarios for adding a new VoIP capacity

For these test cases it is assumed the new capacity will be on same equipment (technology is the same, new IP address, etc), else it will be necessary to perform all tests in previous section.

1.12 Voice call flow testing

- Check at least 10 voice calls to different testing numbers, including mobile terminations (if applicable)
- Calling Party Number - Verify that CLI is properly passed and received in the correct agreed upon format
- Called Party Number – Verify that the called party number is passed and received in the correct agreed upon format
- Verify that all agreed upon codec's are tested

1.13 CDRs comparison

At the end of testing period CDRs shall be exchanged and compared to ensure consistency. It is recommended that the billing CDRs be used for this comparison.

An example of the relevant CDR fields to be exchanged is the following:

a) Calling number
b) Called number
c) Start date/time
d) Stop date/time
e) Call duration
f) Release code

TEST RESULT Record Sheet for New Interconnection

1.14 General information

	Carrier A	Carrier B
Carrier Name		
Date of Test		
Testing personnel contact		
Testing number(s)		
Test fax number(s)		

1.15 Voice call flow testing

Test case number	Description	Pass/ Fail	Codecs tested	SIP Release code	Comments
7.1.1	Normal call release – Calling party clears after answer			BYE	
7.1.2	Normal call release – Called party clears after answer			BYE	
7.1.3	Normal call release – Calling party release while ringing			BYE	
7.1.4	Normal call setup to Ring No Answer / Timeout			480	
7.1.5	Normal call setup to Busy Line / Calling Party Release			486	
7.1.6	Verify Proper handling for No Route To Destination			604	
7.1.7	Verify Proper handling for Unallocated Number			484	
7.1.8	Verify proper handling for Insufficient Digits			484	
7.1.9	Verify “long call” duration – Call should be at least 15 minutes and not disconnect abnormally			BYE	

7.1.10	DTMF – Verify digits received for a DTMF transmission in the agreed upon DTMF methods			BYE	
7.1.11	Calling Party Number - Verify that CLI is properly passed and received in the agreed upon format			BYE	
7.1.12	Called Party Number – Verify that the called party number is received in the agreed upon format			BYE	
7.1.13	CLI Restriction presentation - CLIR (only if agreed upon by both parties)			BYE	
7.1.14	Reachability and keepalive mechanism (SIP Options)			BYE	

1.16 Fax testing

Test case number	Description	Pass/ Fail	Codecs tested	SIP Release code	Comments
7.2.1	Fax transmission test – no fallback			BYE	
7.2.2	Fax transmission test – with fallback			BYE	
7.2.3	Verify fax image quality			BYE	
7.2.4	Verify long fax transmission (3+ page)			BYE	

1.17 CDRs comparison

Test case number	Description	Pass/Fail	Comments
7.3.1	Verify CDR match • Calling number • Called number • Start date/time • Stop date/time • Call duration • Release code		

TEST RESULT Record Sheet for New VoIP Capacity

1.18 General information

	Carrier A	Carrier B
Carrier Name		
Date of Test		
Testing personnel contact		
Testing number(s)		
Test fax number(s)		

1.19 Voice call flow testing

Test case number	Description	Pass/ Fail	Codecs tested	SIP Release code	Comments
8.1.1	Normal call release – Calling party clears after answer- Repeat 10 times to different numbers			BYE	
8.1.2	DTMF – Verify digits received for a DTMF transmission in the agreed upon DTMF methods			BYE	
8.1.3	Calling Party Number - Verify that CLI is properly passed and received in the agreed upon format			BYE	
8.1.4	Called Party Number – Verify that the called party number is received in the agreed upon format			BYE	

1.20 CDRs comparison

Test case number	Description	Pass/Fail	Comments
8.1.5	Verify CDR match • Calling number • Called number • Start date/time • Stop date/time • Call duration • Release code		