SIP INTEROP TEST DESCRIPTION

INNOVAPHONE SIP INTEROP TESTS WITH SIP PROVIDER GNTEL

SUMMARY

This article describes the SIP provider Tests done by Com8 when testing the compatibility of a provider. The features are divided in two classes: KO-criteria and optional criteria. When the provider does not pass a test marked as KO-criteria, the provider cannot be accepted in the category recommended SIP Provider.

To differentiate the quality of the offered service, the provider receives an additional rating. The current rating value ranges between 0 and 117 points. The test success rate is displayed as a percent value. It is calculated by using the following formula: (rating points achieved / rating points total) * 100.

Test Name	Importance	Rating Points	Received Points
Basic Call			
Call using G711A	KO	5	5
Call using G711U	KO	5	5
Call using G723	Optional	2	0
Call using G729	Optional	3	3
Overlapped Sending	Optional	2	0
Early Media Channel	KO	5	5
Fax using T.38	Optional	3	*Not tested*
Reverse Media Negotiation	Optional	3	3
CGPN can be suppressed	Optional	2	2
CLIP no screening	Optional	3	0
Long time call possible(>30 min)	KO	5	5
External Transfer	KO	5	5
NAT Detection	Optional	3	*Not tested*
Voice Quality	KO	5	5
Direct Dial In			
Inbound(gnTel->Innovaphone)	KO	5	5
Outbound(Innovaphone->gnTel)	KO	5	5
DTMF			
DTMF tones sent correctly	KO	5	5
DTMF tones received correctly	KO	5	5
Hold/Retrieve			
Call can be put on hold	KO	5	3
Held end hears music on hold / announcement from PBX	Optional	2	2

Transfer with consultation			
Call can be transferred	KO	5	5
Held end hears music on hold	Optional	2	2
Transfer without consultation (alerting only)			
Call can be transferred	KO	5	5
Held end hears music on hold	Optional	2	2
Call returns to transferring device if the third endpoint is not available	KO	5	5
Blind Transfer			
Call can be transferred	Optional	3	3
Held end hears dialing tone	Optional	2	2
Broadcast Group & Waiting Queue			
Caller can make a call to a Broadcast Group	KO	5	5
Caller can make a call to a Waiting Queue	KO	5	5
Announcement if nobody picks up the call	KO	5	5
	FINAL SCORE	**117**	**102**
	(%)	**100**	**87,18**

TEST SCENARIO
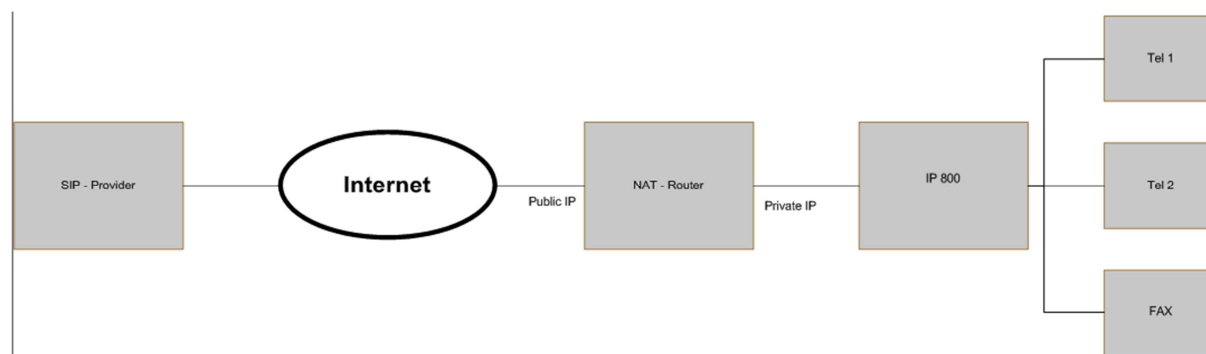
This scenario describes a setup where the PBX and phones are in a private network. The IP800 must use a stun server, in order to send correct SIP - messages. The IP800 works as media relay, all RTP - streams go through the PBX.

BASIC CALL

CALL USING G711A(KO)

Purpose: Test of capability to handle G711A RTP - Streams
Test description:
Tel1 is configured with G711A exclusive coder preference.
Tel1 calls a phone in the PSTN
Remarks: OK

CALL USING G711U(KO)

Purpose: Test of capability to handle G711U RTP - Streams
Test description:
Tel1 is configured with G711U exclusive coder preference.
Tel1 calls a phone in the PSTN
Remarks: OK

CALL USING G723(OPTIONAL)

Purpose: Test of capability to handle G723 RTP - Streams
Test description:
Tel1 is configured with G723 exclusive coder preference.
Tel1 calls a phone in the PSTN
Remarks: NOK. G723 is not supported.

CALL USING G729(OPTIONAL)

Purpose: Test of capability to handle G729 RTP - Streams
Test description:
Tel1 is configured with G729 exclusive coder preference.
Tel1 calls a phone in the PSTN
Remarks: OK

OVERLAPPED SENDING(OPTIONAL)

Purpose: Test if the provider supports Overlap Dialing. If not Enbloc Dialing must be configured on the SIP - trunk.
Test description:
Route from PBX to SIP-provider is not configured with 'Force Enbloc'.
Tel1 calls a phone in the PSTN
Remarks: NOK

EARLY MEDIA CHANNEL(KO)

Purpose: Test if the transmission of RTP - packets is possible before the call has been accepted (200 OK) by both endpoints. This feature is used when the caller receives announcements or dial tones from the SIP - Provider. (i.e. 'The dialed number is incomplete.')
Test description:
Tel1 calls a not existent(incomplete) phone number in the PSTN.
PSTN-Provider will play an announce, i.e. 'Number incomplete'.
The SIP-provider forwards the announcement and terminates the call setup procedure.
Remarks: OK

FAX USING T.38(OPTIONAL)

Purpose: Test of capability to handle T.38 signaling/data. Without this feature Fax over IP is not possible.
Test description:
Fax1 sends a message to a fax machine in the PSTN.
Fax1 receives a message from a fax machine in the PSTN.
Remarks: For sending faxes, separate fax accounts are available (not tested yet). Incoming faxes are only possible with Fax2email.

REVERSE MEDIA NEGATIATION (OPTIONAL)

Purpose: Test if Reverse Media Negotiation is implemented by the provider. If this test fails, media relay and an exclusive coder setting must be configured for this provider, resulting in a much lower amount of concurrent calls.
Test description:
Tel1 calls Tel2. Tel2 picks up the call
Tel2 makes a blind transfer to a phone in the PSTN
This is the quickest way to send a INVITE message without SDP over the SIP-Trunk
Remarks: OK

CGPN CAN BE SUPPRESSED(OPTIONAL)

Purpose: Test if provider accepts anonymous calls
Test description:
Tel1 is configured to not send his CGPN. 'User Setup'->'Number Presentation' = Off
Tel1 calls a phone in the PSTN. The display of the PSTN phone should show as Calling Party: 'anonymous'
Remarks: OK

CLIP NO SCREENING(OPTIONAL)

Purpose: Test if provider accepts calls having a different CGPN as assigned to the SIP trunk
Test description:
Tel1 is configured with CFU to a PSTN phone.
A second PSTN phone calls Tel1, the call is forwarded to the first PSTN phone. The display of the first PSTN phone should show as Calling Party the second PSTN phone.
Remarks: NOK, after 40 seconds busy tone

LONG TIME CALL POSSIBLE(KO)

Purpose: Test if provider interrupts call after session timer expires (RFC 4028). It is possible that the provide awaits Re-Invite after the timer expires. However the session timer is not implemented by innovaphone, so we will not send a Re-Invite.
Test description:
Tel1 calls a phone in the PSTN
call is longer than session expire timeout.(usually 1800 seconds)
Remarks: OK

EXTERNAL TRANSFER(KO)

Purpose: In scenarios where Media Relay is not used, the provider will have to connect both calls by short-circuiting his RTP ports.
Test description:
A phone in the PSTN calls Tel1
Tel1 picks up the call and transfers it to a second PSTN phone.
Manual test of voice quality by speaking/listening on both ends.
Remarks: OK

NAT Detection(optional)

Purpose: In scenario's where Media Relay is not used and the PBX and phones are behind a NAT router, the SIP provider must support a NAT detection mechanism. The provider will receive SIP packets containing private IP addresses, this private IP addresses have to be replaced by the provider with the corresponding public addresses. If this test fails, media relay (with STUN server) and an exclusive coder setting must be configured for this provider, resulting in a much lower amount of concurrent calls.

Test description:
Tel1 is in a private network behind a NAT router.
Tel1 calls a phone in the PSTN.
Check that audio is sent in both directions.
Remarks: NOT Tested

Voice Quality OK?(KO)

Purpose: Simple test of overall voice quality during a call.
Test description:
Tel1 calls Tel2. Tel2 picks up the call.
Manual test of voice quality by speaking/listening on both ends.
Remarks: OK

Direct Dial In

Inbound(gnTel -> Innovaphone)(KO)

Purpose: Test if the provider forwards the call to the PBX using the correct DDI - number(CDPN).
Test description:
Phone in the PSTN calls Tel1.
Check if the CDPN is forwarded correctly to the PBX.
Remarks: OK

Outbound(Innovaphone -> gnTel)(KO)

Purpose: Test if the provider handles the CGPN (trunk number + extension) correctly.
Test description:
Tel1 calls phone in the PSTN.
Check if the PSTN phone display shows the correct CGPN.
Remarks: OK

DTMF

DTMF is also a must have feature for a company. DTMF is crucial for the use of a voicemail system. Currently there are two methods of transferring DTMF signals, by SIP - INFO message or encapsulated in the RTP - packet. Innovaphone supports both types of DTMF signaling. However you must pay attention at the proper configuration of your innovaphone box, since your provider will typical support just one kind of DTMF tone signalization.

DTMF tones sent correctly(KO)

Purpose: Test if DTMF signals going from PBX to Provider are received and interpreted correctly.
Test description:
Tel1 calls phone in the PSTN.
Check if the DTMF - tones are received at the PSTN phone(using SOAP).
Remarks: OK

DTMF TONES RECEIVED CORRECTLY(KO)

Purpose: Test if DTMF signals going from Provider to PBX are received and interpreted correctly.
Test description:
PSTN Phone calls Tel1
Check if the DTMF - tones are received at Tel1(using SOAP).
Remarks: OK

HOLD/RETRIEVE

When a call is put on hold, users normally expect to hear some kind of music/announcement signaling them that they should wait. However there are two possibilities. The PBX generates the announcement or the provider generates it. To test a PBX generated announcement, use the R - key to hold a conversation. This type of holding is tested in the Hold/Retrieve , Transfer with consultation and Transfer with consultation (alerting only) scenario.
To test a provider generated announcement, use the redial key to hold the conversation. This is used when doing a blind transfer.

CALL CAN BE PUT ON HOLD(KO)

Purpose: Test if provider handles hold signalization by Reinvite correctly.
Test description:
Tel1 calls phone in the PSTN.
Tel1 presses the 'R-Key'. Test if call is on hold(display blinking & MoH/announcement).
Tel1 presses again the 'R-Key'. Test if call is retrieved and conversation is continuable.
Remarks: NOK. Call is on hold, but not blinking on display.

HELD END HEARS MUSIC ON HOLD / ANNOUNCEMENT FROM PBX(OPTIONAL)

Purpose: Test if provider handles hold using the send only attribute correctly. The MoH will be transmitted by the PBX to the provider and must be then forwarded to the waiting phone.
Test description:
Tel1 calls phone in the PSTN.
Tel1 presses the 'R-Key'. Test if call signalization is correct and MoH is audible on waiting phone.
Remarks: OK, MOH from innovaphone heard

TRANSFER WITH CONSULTATION

CALL CAN BE TRANSFERRED(KO)

Purpose: Test of call-transfer with consultation
Test description:
Tel1 calls phone in the PSTN.
Tel1 presses the 'R-Key' and dials the number of Tel2.
Test if audio channels between Tel1 and Tel2 are established correctly.
Tel1 hangs up. PSTN phone changes its status from 'hold' to 'active'.

Test if audio channels between PSTN phone and Tel2 are established correctly.
Remarks: OK

HELD END HEARS MUSIC ON HOLD(OPTIONAL)

Purpose: Test for MoH/Announcement on hold phone.
Test description:
Tel1 calls phone in the PSTN.
Tel1 presses the 'R-Key' and dials the number of Tel2.
Test if PSTN phone hears MoH.
Remarks: OK, MOH from innovaphone PBX

TRANSFER WITHOUT CONSULTATION (ALERTING ONLY)

CALL CAN BE TRANSFERRED(KO)

Purpose: Test of call-transfer without consultation
Test description:
Tel1 calls phone in the PSTN.
Tel1 presses the 'R-Key' and dials the number of Tel2.
Tel1 doesn't wait for Tel2 to pick up the call and hangs up. PSTN phone changes its status from 'hold' to 'active'.
Test if audio channels between PSTN phone and Tel2 are established correctly.
Remarks: OK

HELD END HEARS MUSIC ON HOLD(OPTIONAL)

Purpose: Test for MoH/Announcement on hold phone.
Test description:
Tel1 calls phone in the PSTN.
Tel1 presses the 'R-Key' and dials the number of Tel2.
Test if PSTN phone hears MoH.
Remarks: OK, MOH from innovaphone PBX

CALL RETURNS TO TRANSFERRING DEVICE IF THE THIRD ENDPOINT IS NOT AVAILABLE(KO)

Purpose: Test if returning calls are handled correctly by the provider
Test description:
Tel1 calls phone in the PSTN.
Tel1 presses the 'R-Key' and dials the number of Tel2.
Tel1 doesn't wait for Tel2 to pickup the call and hangs up. PSTN phone changes its status from 'hold' to 'active'.
Tel2 doesn't answer the call. Depending on the configured Recall 'Timeout' the call falls back to Tel1.
Tel1 answers calls and checks if audio is working in both directions.
Remarks: OK

BLIND TRANSFER

CALL CAN BE TRANSFERRED(OPTIONAL)

Purpose: Test of call-transfer with consultation
Test description:
Tel1 calls phone in the PSTN.
Tel1 presses the 'Redial-Key' and dials the number of Tel2.
Call is passed to Tel2 and Tel2 picks up the call.
Test if audio channels between PSTN phone and Tel2 are established correctly.
Remarks: OK

HELD END HEARS MUSIC ON HOLD(OPTIONAL)

Purpose: Test for MOH/Announcements on hold phone.
Test description:
Tel1 calls phone in the PSTN.
Tel1 presses the 'R-Key' and dials the number of Tel2.
Test if PSTN phone hears dial-tone(or MOH).
Remarks: OK

BROADCAST GROUP & WAITING QUEUE

CALLER CAN MAKE A CALL TO A BROADCAST GROUP(KO)

Purpose: Test of basic functionality of the Broadcast feature, using Reinvite.
Test description:
PSTN Phone calls a 'Broadcast Group' number.
Tel1 and Tel2 are ringing. Tel1 picks up the call.
Test if audio channels between PSTN phone and Tel1 are established correctly.
Remarks: OK

CALLER CAN MAKE A CALL TO A WAITING QUEUE(KO)

Purpose: Test of basic functionality of the Waiting Queue feature, using Reinvite.
Test description:
PSTN Phone calls a 'Waiting Queue' number.
Tel1 and Tel2 are ringing. Tel1 picks up the call.
Test if audio channels between PSTN phone and Tel1 are established correctly.
Remarks: OK

ANNOUNCEMENT IF NOBODY PICKS UP THE CALL(KO)

Purpose: Test if announcement feature at Waiting queues is working correctly.
Test description:
PSTN Phone calls a 'Waiting Queue' number.
Tel1 and Tel2 are ringing.
PSTN phone hears an announcement from the PBX, i.e. 'All operators are busy. Please hold the line.'
Tel1 picks up the call.
Test if audio channels between PSTN phone and Tel1 are established correctly.
Remarks: OK. MOH from innovaphone PBX

CONFIGURATION SCREENSHOTS

SIP – GK GATEWAY

Name	gnTel
Disable	☐
ID	0130020020 @ sip.gntel.nl
Proxy	
STUN Server	

Authorization

Username 0130020020

Password ●●●●●●●   Retype ●●●●●●●

Media Properties

General Coder Preference G711A ▾ Framesize [ms] 30   Silence Compression ☐   Exclusive ☐

Local Network Coder G711A ▾ Framesize [ms] 30   Silence Compression ☐

Enable T.38 ☐ | Enable SRTP ☐ | Media-Relay ☑ | No DTMF Detection ☐ | Enable PCM ☐

Record to (URL) [ ]

SIP Interop Tweaks

Proposed Registration Interval [s] [ ]

Accept INVITE's from Anywhere ☐

Enforce Sending Complete ☐ (affects outgoing SIP calls only)

From Header when Sending INVITE   CGPN in user part of URI ▾

Identity Header when Sending INVITE   CGPN in user part of URI ▾

Reliability of Provisional Responses   Supported ▾ (affects outgoing SIP calls only)

Internal Registration

Protocol None ▾

| OK | Cancel | Apply | Delete | Help |

NUMBER MAPPINGS

Interface	CGPN-In	CDPN-In	CGPN-Out	CDPN-Out
SIP1 gnTel	i31→0 i→00 n→0	31→0 →00	!→i31852106232	194.140.246.34
SIP2	+			
SIP3	+			
SIP4	+			

Interface	CGPN-In	CDPN-In	CGPN-Out	CDPN-Out
SIP1 gnTel	i31→0 i→00 n→0	31→0 →00	!→31852106232	194.140.246.34
SIP2	+			
SIP3	+			
SIP4	+			

Interface	CGPN-In	CDPN-In	CGPN-Out	CDPN-Out
SIP1 gnTel	i31→0 i→00 n→0	31→0 →00	!→0852106232	194.140.246.34
SIP2	+			
SIP3	+			
SIP4	+			

Interface	CGPN-In	CDPN-In	CGPN-Out	CDPN-Out
SIP1 gnTel	i31→0 i→00 n→0	31→0 →00	!→n852106232	194.140.246.34
SIP2	+			
SIP3	+			
SIP4	+			

Interface	CGPN-In	CDPN-In	CGPN-Out	CDPN-Out
SIP1 gnTel	i31→0 i→00 n→0	31→0 →00	!→852106232	194.140.246.34
SIP2	+			
SIP3	+			
SIP4	+			

Route Settings

From	To		Counter CGPN Maps	
⇄ GW1:PBX ⇄ ⇄ ,	→	TONE:The Netherlands	→	
⇄ 00	→ 00	SIP1:gnTel	b →	From PBX To Sip Provider (International)
⇄ 08	→ 08	SIP1:gnTel	b →	From PBX To Sip Provider (Free Numbers)
⇄ 09	→ 09	SIP1:gnTel	b →	From PBX To Sip Provider (Premium Numbers)
⇄ 0.........	→ 0	SIP1:gnTel	→	From PBX To Sip Provider (National)
⇄	→	SIP1:gnTel	→	From PBX To Sip Provider (Other)
⇄ SIP1:gnTel ⇄ ⇄ !	→ 202	GW1:PBX	→	From Sip Provider To PBX