

SIP Trunk Compatibility Report

NEC is pleased to verify that:

OTE S.A. - GREECE

has successfully met the standards for SIP Trunk compatibility with the NEC product listed below.

UNIVERGE SV9300



NEC Enterprise Solutions has performed Interoperability Testing with the Provider / Service listed above on the date specified.

Please always refer to the appropriate SIP Trunk Compatibility Report Index and the latest edition of a specific Compatibility Report on BusinessNet before considering connection.

If a Provider is no longer mentioned in the Index then the Compatibility Report has been withdrawn and connection will no longer be supported by NEC Enterprise Solutions.

Test Completion Date:	4 October 2016
Test Location:	OTE, Greece
Name of Provider:	OTE S.A. http://www.cosmote.gr
System Tested:	UNIVERGE SV9300 Software Version: SC-4351 V3.3.2
SIP Connection Mode:	Carrier Mode (DNS)
Test Plan	Version: 1.0

Please refer to the following page(s) for further information and Configuration Notes.

Executive Summary

IP-PBX model	NEC UNIVERGE SV9300
IP-PBX software version	SC-4351 V3.3.2
SIP Trunk provider	OTE
Test request reference	SIPTR-TS001
Initial test date	31 August 2016
Tested by	Dimitris Kefalas
Version	1.0
Status	Approved

Functionality	Test result Approved/ Not approved	Remarks
Registration	Approved	
Normal Call	Approved	
Call Abandon	Approved	
Call Busy	Approved	
CLIP	Approved	
CLIR	Approved	
Fax / Modem	Approved	Only T.38 FAX
CODEC	Approved	
DTMF (RFC2833)	Approved	
Call diversion	Approved	
Call transfer	Approved	

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Introduction

1 Introduction

NEC Telephone Systems can connect to SIP Internet Telephony Service Providers (ITSPs) to provide Voice over IP services.

Interoperability Testing (IOT) is done to determine if the ITSP is compatible with the Telephone System. If these tests are satisfactorily completed, the ITSP can be “Certified” by NEC Unified Solutions.

The purpose of this document is to detail the test procedure and document the test results.

It should be noted that some ITSPs have their own test and certification procedures. In this case this document should also be filled in for own administrative purposes.

Tests not supported by the IP-PBX under certification can be skipped. Test result = NA

Definitions	Passed :	Test case was successfully completed
	Failed:	Test case was not successfully completed
	Blocked:	Test case could not be executed
	Inconclusive:	Test case could be executed but the results could not be interpreted as either Passed or Failed
	NA	Test case Not Applicable

Abbreviations	IOT	Interoperability Testing
	IP-PBX	Internet Protocol Private Branch Exchange
	ITSP	Internet Telephony Service Provider
	PSTN	Public Switched Telephone Network

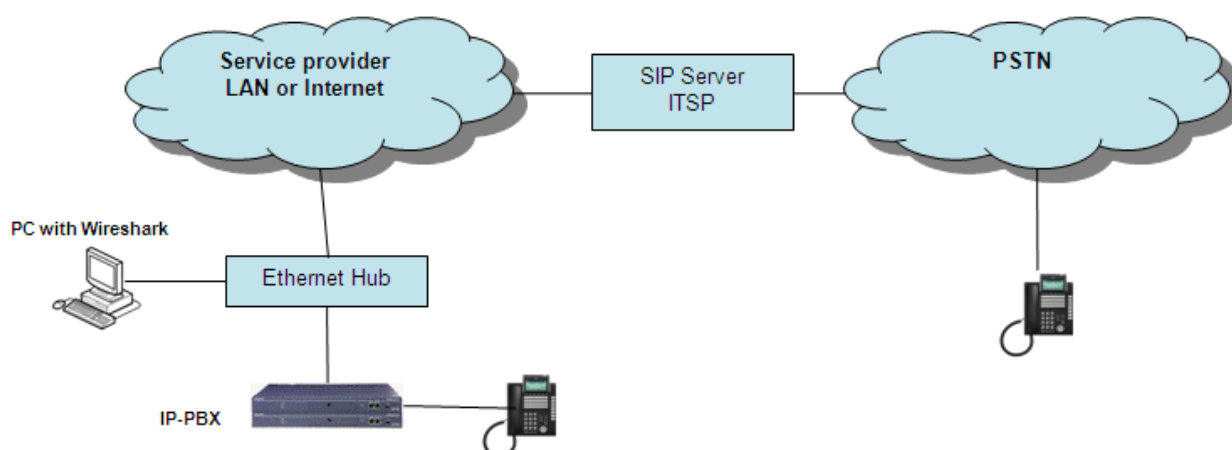
2 Test environment

2.1 Approach

Testing can be carried out remotely (via an Internet connection) or locally (at the ITSP premises). The test suite is the same for both types of test.

For each test, a Wireshark capture file should be obtained and verified. This will help to diagnose and problems that are found, and will be used to ensure compliance with the relevant RFCs.

The diagram below shows the test setup required to perform the IOT.



2.2 Tested equipment

IP-PBX model	SV9300
IP-PBX software version	SC-4351
SIP trunk provider	OTE
SIP server address	ims.otenet.gr
SIP server IP	195.167.16.129
Port	5060
Username	+302144163500@ims.otenet.gr
Password	
Test telephone number(s)	+30214416350x, +30214416351x, +30214416352x
IP device range	80.107.96.32
Subnet device	255.255.255.248
Gateway device	80.107.96.33

Registration

2.3 SIP Header formats

SIP header formats to and from the IPPBX need to be compliant to RFC3261, unless otherwise stated.

3 Registration

The object of these tests is to determine if the IP-PBX is able to register to the ITSP and maintain the registration.

Some ITSPs use a “peering” environment which does not require registration. In this case, this section may be disregarded.

It is preferable to perform testing with multiple SIP accounts. If the ITSP has provided one account only, then skip the multiple account tests.

3.1 Single Account Successful registration

Test Case: 3.1	Title: Single Account Successful registration
Test Procedure: Register the IP-PBX to the SIP Server with a single account	
Expected Results: System should register to the server. Authentication may be required (401 Unauthorized)	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

Registration

3.2 Single Account Registration Refresh

Test Case: 3.2	Title: Single Account Registration Refresh
Test Procedure: Maintain a registration to the server for 24 hours	
Expected Results: Re-registration should occur periodically. Registrations should be maintained	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

3.3 Single Account Registration Failure

Test Case: 3.3	Title: Single Account Registration Failure
Test Procedure: Enter invalid authentication information	
Expected Results: Registration should fail and outgoing calls should be disabled. Confirm that registration is successful after correcting authentication information.	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

Registration

3.4 Single Account De-registration

Test Case: 3.4	Title: Single Account De-registration
Test Procedure: Register to SIP Server. Disable registration	
Expected Results: System should de-register from SIP Server	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

3.5 Multiple Account Successful registration

Test Case: 3.5	Title: Multiple Account Successful registration
Test Procedure: Register the IP-PBX to the SIP server with a multiple accounts	
Expected Results: System should register to the server. Authentication may be required (401 Unauthorized)	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

3.6 Multiple Account Registration Refresh

Test Case: 3.6	Title: Multiple Account Registration Refresh
Test Procedure: Maintain multiple registrations to the server for 24 hours	
Expected Results: Re-registration should occur periodically. Registrations should be maintained	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

Registration

3.7 Multiple Account Registration Failure

Test Case: 3.7	Title: Multiple Account Registration Failure
Test Procedure: Enter invalid authentication information	
Expected Results: Registration should fail and outgoing calls should be disabled. Confirm that registration is successful after correcting authentication information.	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

3.8 Multiple Account De-registration

Test Case: 3.8	Title: Multiple Account De-registration
Test Procedure: Register to SIP Server. Disable registration	
Expected Results: System should de-register from SIP Server	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

3.9 SIP Heartbeat

Test Case: 3.9	Title: SIP Heartbeat
Test Procedure: SIP Network initiates SIP heartbeat every x seconds. Verify what message comes from SIP trunk provider so that SIP trunk provider knows that SIP trunk is alive (heartbeat).	
Expected Results: Verify that heartbeat is answered by IP-PBX by means of OK message Verify that network does not lock session due to absence of IP-PBX response.	
Results and Observations:	
Test Case result	<input type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input checked="" type="checkbox"/> NA
Findings	
OTE SIP Trunk doesn't use Hartbeat	
Special configuration needed on IP-PBX	
SIP trunk provider keep alive message	

Registration

Basic Call

4 Basic Call

This section ensures that:

- Outgoing (outbound) calls can be made from the IP-PBX to other SIP users, or to telephones on the PSTN.
- The IP-PBX can successfully receive calls from other SIP users or via the PSTN.

4.1 Outgoing call, release by A-party

Test Case: 4.1	Title: Outgoing call, release by A-party												
Test Procedure: Make a call from an IP-PBX phone to a PSTN phone. <ul style="list-style-type: none"> - Check that the call is established. - Check the voice channel. A-party, release the call. <ul style="list-style-type: none"> - Check the release signaling sequence. 													
Conditions: <table> <tr> <td>IP-PBX</td> <td>Domain <provider></td> </tr> <tr> <td>PSTN</td> <td>PSTN/ISDN phone</td> </tr> <tr> <td>Number A-party</td> <td>+302144163502</td> </tr> <tr> <td>Number B-party</td> <td>+302106111866</td> </tr> <tr> <td>Date and Time</td> <td>2016/08/16 15:56:15</td> </tr> <tr> <td>Protocol SIP (IP-PBX)</td> <td></td> </tr> </table>		IP-PBX	Domain <provider>	PSTN	PSTN/ISDN phone	Number A-party	+302144163502	Number B-party	+302106111866	Date and Time	2016/08/16 15:56:15	Protocol SIP (IP-PBX)	
IP-PBX	Domain <provider>												
PSTN	PSTN/ISDN phone												
Number A-party	+302144163502												
Number B-party	+302106111866												
Date and Time	2016/08/16 15:56:15												
Protocol SIP (IP-PBX)													
Results and Observations: Test Case result <input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA Findings Special configuration needed on IP-PBX													

4.2 Outgoing call, early media

Test Case: 4.2	Title: Outgoing call, early media										
Test Procedure: Make a call from an IP-PBX phone to a PSTN phone. <ul style="list-style-type: none"> - Check that early media is received from the PSTN before call is accepted - Check that the early media is played out correctly at the originating phone 											
Conditions: <table> <tr> <td>IP-PBX</td> <td>Domain <provider></td> </tr> <tr> <td>PSTN</td> <td>PSTN/ISDN phone</td> </tr> <tr> <td>Number A-party</td> <td>+302144163502</td> </tr> <tr> <td>Number B-party</td> <td>+302106111866</td> </tr> <tr> <td>Date and Time</td> <td>2016/08/31 15:56:15</td> </tr> </table>		IP-PBX	Domain <provider>	PSTN	PSTN/ISDN phone	Number A-party	+302144163502	Number B-party	+302106111866	Date and Time	2016/08/31 15:56:15
IP-PBX	Domain <provider>										
PSTN	PSTN/ISDN phone										
Number A-party	+302144163502										
Number B-party	+302106111866										
Date and Time	2016/08/31 15:56:15										

Basic Call

Protocol SIP (IP-PBX)	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

4.3 Incoming call, release by B-party

Test Case: 4.3	Title: Incoming call, release by B-party
Test Procedure:	
Make a call from a PSTN phone to an IP-PBX phone – Check that the call is established. – Check the voice channel. B-party, release the call. – Check the release signalling sequence.	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302106111910
Number B-party	+302144163501
Date and Time	2016/08/31 16:07:12
Protocol SIP (IP-PBX)	
Results and Observations:	
Test Case result	<input type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

4.4 Outgoing call, long duration call

Test Case: 4.4	Title: Outgoing call, long duration call
Test Procedure:	
Make a call from IP-PBX phone to PSTN phone. – Check that the call is established. – Check the voice channel. – Leave the call open, it should only be disconnected after 3 hours by the network. – Check the release signalling sequence.	
Conditions:	
IP-PBX	Domain <provider>

Basic Call

PSTN	PSTN/ISDN phone
Number A-party	+302144163502
Number B-party	+302106111866
Date and Time	2016/09/01 16:25:17
Protocol SIP (IP-PBX)	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

4.5 Incoming call, long duration call

Test Case: 4.5	Title: Incoming call, long duration call
Test Procedure:	
Make a call from PSTN phone to IP-PBX phone. – Check that the call is established. – Check the voice channel. – Leave the call open, it should only be disconnected after 3 hours by the network. – Check the release signalling sequence.	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302106111910
Number B-party	+302144163501
Date and Time	2016/09/05 08:56:28
Protocol SIP (IP-PBX)	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

5 Call Abandon

5.1 IP-PBX to PSTN, A hangs up during ringing

Test Case: 5.1	Title: IP-PBX to PSTN, A hangs up during ringing
Test Procedure:	
<p>Make a call from a IP-PBX phone to a PSTN phone.</p> <p>When called phone is ringing, hang-up caller phone.</p> <ul style="list-style-type: none"> – Check the signalling sequence. 	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302144163502
Number B-party	+302106111866
Date and Time	2016/08/31 16:10:24
Protocol SIP (IP-PBX)	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

5.2 IP-PBX to PSTN, B does not answer

Test Case: 5.2	Title: IP-PBX to PSTN, B does not answer
Test Procedure:	
<p>Make a call from IP-PBX phone to an idle PSTN phone.</p> <p>Do not answer the call.</p> <ul style="list-style-type: none"> – Check the signalling sequence. – Check that SIP event 480 (Temporarily Unavailable) is obtained. 	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302144163502
Number B-party	+302106111866
Date and Time	2016/08/31 16:12:13
Protocol SIP (IP-PBX)	

Call Abandon

Results and Observations:

Test Case result Passed Failed Blocked
 Inconclusive NA

Findings

Special configuration needed on IP-PBX

5.3 PSTN to IP-PBX, A hangs up during ringing

Test Case: 5.3	Title: PSTN to IP-PBX, A hangs up during ringing
Test Procedure:	
From a PSTN phone dial an IP-PBX number.	
When called phone is ringing, hang-up caller phone.	
– Check the signalling sequence.	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302106111910
Number B-party	+302144163501
Date and Time	2016/08/31 16:15:48
Protocol SIP (IP-PBX)	
Results and Observations:	
Test Case result <input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA	
Findings	
Special configuration needed on IP-PBX	

5.4 PSTN to IP-PBX, B does not answer

Test Case: 5.4	Title: PSTN to IP-PBX, B does not answer
Test Procedure:	
Dial from a PSTN phone an IP-PBX number.	
Do not answer the call.	
– Check the signalling sequence.	
– Check that SIP event 480 (Temporarily Unavailable) is obtained.	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302106111910
Number B-party	+302144163501

Call Abandon

Date and Time	2016/08/31 16:19:49
Protocol SIP (IP-PBX)	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

Call Busy

6 Call Busy

6.1 IP-PBX to PSTN, B is busy

(expect SIP event 486, busy here).

Test Case: 6.1	Title: IP-PBX to PSTN, B is busy
Test Procedure:	
<p>Make a call from IP-PBX phone to a busy PSTN phone.</p> <ul style="list-style-type: none"> – Check the signalling sequence. – Check that SIP event 486 (Busy here) is obtained. 	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302144163502
Number B-party	+302106111866
Date and Time	2016/08/31 16:23:16
Protocol	SIP (IP-PBX)
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

6.2 PSTN to IP-PBX, B is busy

(expect SIP event 486, busy here)

Test Case: 6.2	Title: PSTN to IP-PBX, B is busy
Test Procedure:	
<p>Call from a PSTN phone, to an IP-PBX busy subscriber.</p> <ul style="list-style-type: none"> – Check the signalling sequence. – Check that SIP event 486 (Busy here) is obtained. 	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302106111910
Number B-party	+302144163501
Date and Time	2016/08/31 16:24:33
Protocol	SIP (IP-PBX)
Results and Observations:	

Call Busy

Test Case result	<input checked="" type="checkbox"/> Passed	<input type="checkbox"/> Failed	<input type="checkbox"/> Blocked
Findings	<input type="checkbox"/> Inconclusive	<input type="checkbox"/> NA	
Special configuration needed on IP-PBX			

CLIP

7 CLIP

7.1 IP-PBX to PSTN, Normal call, release by A-party.

Test Case: 7.1	Title: IP-PBX to PSTN, Normal call, release by A-party.												
Test Procedure: Make a call from IP-PBX phone to a PSTN phone. <ul style="list-style-type: none"> – Check that the call is established. – Check the voice channel. – Check the number displayed at PSTN phone (B-subscriber). A-party, release the call. <ul style="list-style-type: none"> – Check the release signalling sequence 													
Conditions: <table> <tr> <td>IP-PBX</td> <td>Domain <provider></td> </tr> <tr> <td>PSTN</td> <td>PSTN/ISDN phone</td> </tr> <tr> <td>Number A-party</td> <td>+302144163502</td> </tr> <tr> <td>Number B-party</td> <td>+302106111866</td> </tr> <tr> <td>Date and Time</td> <td>2016/08/31 16:28:48</td> </tr> <tr> <td>Protocol SIP (IP-PBX)</td> <td></td> </tr> </table>		IP-PBX	Domain <provider>	PSTN	PSTN/ISDN phone	Number A-party	+302144163502	Number B-party	+302106111866	Date and Time	2016/08/31 16:28:48	Protocol SIP (IP-PBX)	
IP-PBX	Domain <provider>												
PSTN	PSTN/ISDN phone												
Number A-party	+302144163502												
Number B-party	+302106111866												
Date and Time	2016/08/31 16:28:48												
Protocol SIP (IP-PBX)													
Results and Observations: Test Case result <input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA Findings Special configuration needed on IP-PBX													

7.2 PSTN to IP-PBX, Normal call, release by A-party.

Test Case: 7.2	Title: PSTN to IP-PBX, Normal call, release by A-party.												
Test Procedure: Make a call from a PSTN phone to an IP-PBX phone <ul style="list-style-type: none"> – Check that the call is established. – Check the voice channel. – Check the number displayed at IP-PBX 2 phone (B-subscriber). A-party, release the call. <ul style="list-style-type: none"> – Check the release signalling sequence. 													
Conditions: <table> <tr> <td>IP-PBX</td> <td>Domain <provider></td> </tr> <tr> <td>PSTN</td> <td>PSTN/ISDN phone</td> </tr> <tr> <td>Number A-party</td> <td>+302106111910</td> </tr> <tr> <td>Number B-party</td> <td>+302144163501</td> </tr> <tr> <td>Date and Time</td> <td>2016/08/31 16:30:00</td> </tr> <tr> <td>Protocol SIP (IP-PBX)</td> <td></td> </tr> </table>		IP-PBX	Domain <provider>	PSTN	PSTN/ISDN phone	Number A-party	+302106111910	Number B-party	+302144163501	Date and Time	2016/08/31 16:30:00	Protocol SIP (IP-PBX)	
IP-PBX	Domain <provider>												
PSTN	PSTN/ISDN phone												
Number A-party	+302106111910												
Number B-party	+302144163501												
Date and Time	2016/08/31 16:30:00												
Protocol SIP (IP-PBX)													

CLIP

Results and Observations:

Test Case result

Passed Failed Blocked
 Inconclusive NA

Findings

Special configuration needed on IP-PBX

CLIR

8 CLIR

8.1 IP-PBX to PSTN, initiate CLIR

CLIR initiated via IP-PBX privacy configuration. (from: anonymous@anonymous.invalid).

Test Case: 8.1	Title: IP-PBX to PSTN, initiate CLIR
Test Procedure:	
<p>Make a call from IP-PBX to PSTN, with CLIR activated in IP-PBX.</p> <ul style="list-style-type: none"> – Check in the incoming INVITE message and verify that “From” header set to “anonymous@anonymous.invalid” and Privacy header set to “id” in initial INVITE message. – Include in test result the displayed number on both phones. <p>B-party, release the call.</p> <ul style="list-style-type: none"> – Check the release signalling sequence. 	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302144163502
Number B-party	+302106111866
Date and Time	2016/08/31 16:49:43
Protocol SIP (IP-PBX)	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	Party A display shows "2106111866", Party B display shows "Unknown"
Special configuration needed on IP-PBX	

CLIR

8.2 PSTN to IP-PBX with header "Privacy" set to "id"

CLIR from PSTN CLIR.

Test Case: 8.2	Title: PSTN to IP-PBX with header "Privacy" set to "id"
Test Procedure:	
<p>Make a call from PSTN to IP-PBX, with number suppression from the PSTN.</p> <ul style="list-style-type: none"> - Include in test result the displayed number on both phones. <p>B-party, release the call.</p> <ul style="list-style-type: none"> - Check the release signalling sequence. 	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302106111910
Number B-party	+302144163501
Date and Time	2016/08/31 16:52:32
Protocol	SIP (IP-PBX)
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	Party A display shows "2144163501", Party B display shows "Secrecy invoke"
Special configuration needed on IP-PBX	

9 Fax / Modem

9.1 IP-PBX to PSTN, Modem

Test Case: 9.1	Title: IP-PBX to PSTN, Modem												
Test Procedure: Make a call from IP-PBX connected modem to a PSTN modem. – Check that the call is established. – Check the data transmission. A-party, release the call. – Check the release signalling sequence.													
Conditions: <table> <tr> <td>IP-PBX</td> <td>Domain <provider></td> </tr> <tr> <td>PSTN</td> <td>PSTN/ISDN modem</td> </tr> <tr> <td>Number A-party</td> <td>+302144163505</td> </tr> <tr> <td>Number B-party</td> <td>+302299020399</td> </tr> <tr> <td>Date and Time</td> <td>2016/09/01 10:57:51</td> </tr> <tr> <td>Protocol SIP (IP-PBX)</td> <td></td> </tr> </table>		IP-PBX	Domain <provider>	PSTN	PSTN/ISDN modem	Number A-party	+302144163505	Number B-party	+302299020399	Date and Time	2016/09/01 10:57:51	Protocol SIP (IP-PBX)	
IP-PBX	Domain <provider>												
PSTN	PSTN/ISDN modem												
Number A-party	+302144163505												
Number B-party	+302299020399												
Date and Time	2016/09/01 10:57:51												
Protocol SIP (IP-PBX)													
Results and Observations: Test Case result <input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA Findings Special configuration needed on IP-PBX													

9.2 IP-PBX to PSTN, Fax Clear Channel (G.711)

Test Case: 9.2	Title: IP-PBX to PSTN, Fax Clear Channel (G.711)												
Test Procedure: Configure the IP-PBX to transport fax using G.711. Make a call from IP-PBX connected fax to a PSTN connected fax. – Check that the call is established. – Check the fax transmission. A-party, release the call. – Check the release signalling sequence.													
Conditions: <table> <tr> <td>IP-PBX</td> <td>Domain <provider></td> </tr> <tr> <td>PSTN</td> <td>PSTN/ISDN fax</td> </tr> <tr> <td>Number A-party</td> <td></td> </tr> <tr> <td>Number B-party</td> <td></td> </tr> <tr> <td>Date and Time</td> <td></td> </tr> <tr> <td>Protocol SIP (IP-PBX)</td> <td></td> </tr> </table>		IP-PBX	Domain <provider>	PSTN	PSTN/ISDN fax	Number A-party		Number B-party		Date and Time		Protocol SIP (IP-PBX)	
IP-PBX	Domain <provider>												
PSTN	PSTN/ISDN fax												
Number A-party													
Number B-party													
Date and Time													
Protocol SIP (IP-PBX)													

Fax / Modem

Results and Observations:	
Test Case result	<input type="checkbox"/> Passed <input checked="" type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	G.711 Passthrough FAX Fails but it is not important since OTE supports both G.711 and T.38 and always use its SBC to transcode
Special configuration needed on IP-PBX	

9.3 PSTN to IP-PBX, Modem

Test Case: 9.3	Title: PSTN to IP-PBX, Modem
Test Procedure:	
Make a call a PSTN modem to an IP-PBX modem – Check that the call is established. – Check the data transmission. A-party, release the call. – Check the release signalling sequence.	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN modem
Number A-party	+302109537199
Number B-party	+302144163505
Date and Time	2016/09/01 12:08:40
Protocol SIP (IP-PBX)	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	Special configuration needed on IP-PBX

9.4 PSTN to IP-PBX, Fax Clear channel (G.711)

Test Case: 9.4	Title: PSTN to IP-PBX, Fax Clear channel (G.711)
Test Procedure:	
Configure the IP-PBX to transport fax inband using G.711. Make a call a PSTN fax to an IP-PBX fax. – Check that the call is established. – Check the fax transmission, verify that the fax transmitted inband using G.711. A-party, release the call. – Check the release signalling sequence.	

Fax / Modem

Conditions:

IP-PBX	Domain <provider>
PSTN	PSTN/ISDN fax
Number A-party	
Number B-party	
Date and Time	
Protocol SIP (IP-PBX)	

Results and Observations:

Test Case result	<input type="checkbox"/> Passed <input checked="" type="checkbox"/> Failed <input type="checkbox"/> Blocked
	<input type="checkbox"/> Inconclusive <input type="checkbox"/> NA

Findings	G.711 Passthrough FAX Fails but it is not important since OTE supports both G.711 and T.38 and always use its SBC to transcode
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Special configuration needed on IP-PBX	
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9.5 IP-PBX to PSTN, Fax (T.38)

Test Case: 9.5	Title: IP-PBX to PSTN, Fax (T.38)														
Test Procedure: Configure the IP-PBX to transport fax preferring T.38. Make a call from IP-PBX connected fax to a PSTN connected fax. <ul style="list-style-type: none"> – Check that the call is established. – Check the fax transmission, verify that the called system renegotiates to T.38. A-party, release the call. <ul style="list-style-type: none"> – Check the release signalling sequence. 															
Conditions: <table> <tr> <td>IP-PBX</td> <td>Domain <provider></td> </tr> <tr> <td>PSTN</td> <td>PSTN/ISDN fax</td> </tr> <tr> <td>Number A-party</td> <td>+302144163505</td> </tr> <tr> <td>Number B-party</td> <td>+302103405397</td> </tr> <tr> <td>Date and Time</td> <td>2016/09/01 13:20:22</td> </tr> <tr> <td>Protocol SIP (IP-PBX)</td> <td></td> </tr> <tr> <td>Transport Protocol for T.38</td> <td></td> </tr> </table>		IP-PBX	Domain <provider>	PSTN	PSTN/ISDN fax	Number A-party	+302144163505	Number B-party	+302103405397	Date and Time	2016/09/01 13:20:22	Protocol SIP (IP-PBX)		Transport Protocol for T.38	
IP-PBX	Domain <provider>														
PSTN	PSTN/ISDN fax														
Number A-party	+302144163505														
Number B-party	+302103405397														
Date and Time	2016/09/01 13:20:22														
Protocol SIP (IP-PBX)															
Transport Protocol for T.38															
Results and Observations: Test Case result <input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA Findings Special configuration needed on IP-PBX															

9.6 PSTN to IP-PBX, Fax (T.38)

Test Case: 9.6	Title: PSTN to IP-PBX, Fax (T.38)
Test Procedure: Configure the IP-PBX to transport fax preferring T.38. Make a call a PSTN fax to an IP-PBX fax <ul style="list-style-type: none"> – Check that the call is established. – Check the fax transmission, verify that the called system renegotiates to T.38. A-party, release the call. <ul style="list-style-type: none"> – Check the release signalling sequence. 	

Fax / Modem

Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN fax
Number A-party	+302103405397 (CLIR)
Number B-party	+302144163505
Date and Time	2016/09/01 13:14:35
Protocol SIP (IP-PBX)	
Transport Protocol for T.38	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

9.7 IP-PBX to PSTN, G.729 voicecall fallback to Fax (T.38)

Test Case: 9.7	Title: IP-PBX to PSTN, G.729 voicecall fallback to Fax (T.38)
Test Procedure:	
Configure the IP-PBX to transport fax preferring T.38. And configure it to setup voice calls preferring the G.729 codec.	
Make a call from IP-PBX connected fax to a PSTN connected fax.	
<ul style="list-style-type: none"> - Check that the call is established as a G.729 voice call. - Check the fax transmission, verify that the called system renegotiates to T.38. 	
A-party, release the call.	
<ul style="list-style-type: none"> - Check the release signalling sequence. 	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN fax
Number A-party	+302144163505
Number B-party	+302103405397
Date and Time	2016/09/01 14:29:08
Protocol SIP (IP-PBX)	
Transport Protocol for T.38	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

9.8 IP-PBX to PSTN, G.729 voicecall fallback to G.711 for modem

Test Case: 9.8	Title: IP-PBX to PSTN, G.729 voicecall fallback to G.711 for modem												
Test Procedure: And configure it to setup voice calls preferring the G.729 codec. Make a call from IP-PBX connected modem to a PSTN connected modem. <ul style="list-style-type: none"> - Check that the call is established as a G.729 voice call. - Check the modem transmission, verify that the RTP switches from G.729 to G.711. A-party, release the call. <ul style="list-style-type: none"> - Check the release signalling sequence. 													
Conditions: <table> <tr> <td>IP-PBX</td> <td>Domain <provider></td> </tr> <tr> <td>PSTN</td> <td>PSTN/ISDN modem</td> </tr> <tr> <td>Number A-party</td> <td>+302144163505</td> </tr> <tr> <td>Number B-party</td> <td>+302299020399</td> </tr> <tr> <td>Date and Time</td> <td>2016/09/01 11:14:54</td> </tr> <tr> <td>Protocol SIP (IP-PBX)</td> <td></td> </tr> </table>		IP-PBX	Domain <provider>	PSTN	PSTN/ISDN modem	Number A-party	+302144163505	Number B-party	+302299020399	Date and Time	2016/09/01 11:14:54	Protocol SIP (IP-PBX)	
IP-PBX	Domain <provider>												
PSTN	PSTN/ISDN modem												
Number A-party	+302144163505												
Number B-party	+302299020399												
Date and Time	2016/09/01 11:14:54												
Protocol SIP (IP-PBX)													
Results and Observations: Test Case result <input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA Findings Special configuration needed on IP-PBX													

Fax / Modem

9.9 PSTN to IP-PBX, G.729 voicecall fallback to Fax (T.38)

Test Case: 9.9	Title: PSTN to IP-PBX, G.729 voicecall fallback to Fax (T.38)														
Test Procedure: Configure the IP-PBX to transport fax preferring T.38. And configure it to setup voice calls preferring the G.729 codec. Make a call from PSTN connected fax to a IP-PBX connected fax. <ul style="list-style-type: none"> – Check that the call is established as a G.729 voice call. – Check the fax transmission, verify that the called system renegotiates to T.38. A-party, release the call. <ul style="list-style-type: none"> – Check the release signalling sequence. 															
Conditions: <table> <tr> <td>IP-PBX</td> <td>Domain <provider></td> </tr> <tr> <td>PSTN</td> <td>PSTN/ISDN fax</td> </tr> <tr> <td>Number A-party</td> <td>+302103405397 (CLIR)</td> </tr> <tr> <td>Number B-party</td> <td>+302144163505</td> </tr> <tr> <td>Date and Time</td> <td>2016/09/01 14:35:25</td> </tr> <tr> <td>Protocol SIP (IP-PBX)</td> <td></td> </tr> <tr> <td>Transport Protocol for T.38</td> <td></td> </tr> </table>		IP-PBX	Domain <provider>	PSTN	PSTN/ISDN fax	Number A-party	+302103405397 (CLIR)	Number B-party	+302144163505	Date and Time	2016/09/01 14:35:25	Protocol SIP (IP-PBX)		Transport Protocol for T.38	
IP-PBX	Domain <provider>														
PSTN	PSTN/ISDN fax														
Number A-party	+302103405397 (CLIR)														
Number B-party	+302144163505														
Date and Time	2016/09/01 14:35:25														
Protocol SIP (IP-PBX)															
Transport Protocol for T.38															
Results and Observations: Test Case result <input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA Findings Special configuration needed on IP-PBX															

9.10 PSTN to IP-PBX, G.729 voicecall fallback to G.711 for modem

Test Case: 9.10	Title: PSTN to IP-PBX, G.729 voicecall fallback to G.711 for modem												
Test Procedure: And configure it to setup voice calls preferring the G.729 codec. Make a call from PSTN connected modem to a IP-PBX connected modem. <ul style="list-style-type: none"> – Check that the call is established as a G.729 voice call. – Check the modem transmission, verify that the RTP switches from G.729 to G.711. A-party, release the call. <ul style="list-style-type: none"> – Check the release signalling sequence. 													
Conditions: <table> <tr> <td>IP-PBX</td> <td>Domain <provider></td> </tr> <tr> <td>PSTN</td> <td>PSTN/ISDN modem</td> </tr> <tr> <td>Number A-party</td> <td></td> </tr> <tr> <td>Number B-party</td> <td></td> </tr> <tr> <td>Date and Time</td> <td></td> </tr> <tr> <td>Protocol SIP (IP-PBX)</td> <td></td> </tr> </table>		IP-PBX	Domain <provider>	PSTN	PSTN/ISDN modem	Number A-party		Number B-party		Date and Time		Protocol SIP (IP-PBX)	
IP-PBX	Domain <provider>												
PSTN	PSTN/ISDN modem												
Number A-party													
Number B-party													
Date and Time													
Protocol SIP (IP-PBX)													

Fax / Modem

Results and Observations:

Test Case result

Passed Failed Blocked
 Inconclusive NA

Findings

Can't configure PSTN to setup calls preferring G.729

Special configuration needed on IP-PBX

Codec

10 Codec

10.1 IP-PBX to PSTN outgoing call codec G.729

Test Case: 10.1	Title: IP-PBX to PSTN outgoing call codec G.729
Test Procedure: Configure IP-PBX to use G.729. Make outgoing call. – Try different VSI sizes	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302144163502
Number B-party	+302106111866
Date and Time	2016/09/01 14:55:12
Protocol SIP (IP-PBX)	
Expected Results: G.729 should have highest priority in SDP message. Make a note of actual CODEC used	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

10.2 IP-PBX to PSTN outgoing call codec G.723

Test Case: 10.2	Title: IP-PBX to PSTN outgoing call codec G.723
Test Procedure: Configure IP-PBX to use G.723. Make outgoing call. – Try different VSI sizes	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	
Number B-party	
Date and Time	
Protocol SIP (IP-PBX)	

Codec

Expected Results: G.723 should have highest priority in SDP message. Make a note of actual CODEC used	
Results and Observations:	
Test Case result	<input type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input checked="" type="checkbox"/> NA
Findings	OTE doesn't support G.723
Special configuration needed on IP-PBX	

10.3 IP-PBX to PSTN outgoing call codec G.711

Test Case: 10.3	Title: IP-PBX to PSTN outgoing call codec G.711
Test Procedure: Configure IP-PBX to use G.711. Make outgoing call. – Try different VSI sizes	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302144163502
Number B-party	+302106111866
Date and Time	2016/09/01 14:58:53
Protocol SIP (IP-PBX)	
Expected Results: G.711 should have highest priority in SDP message. Make a note of actual CODEC used	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

10.4 PSTN to IP-PBX incoming call codec G.729

Test Case: 10.4	Title: PSTN to IP-PBX incoming call codec G.729
Test Procedure: Configure ITSP to use G.729. Make call into IP-PBX. – Try different VSI sizes	

Codec

Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	
Number B-party	
Date and Time	
Protocol SIP (IP-PBX)	
Expected Results:	
G.729 should have highest priority in SDP message. Make a note of actual CODEC used	
Results and Observations:	
Test Case result	<input type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input checked="" type="checkbox"/> NA
Findings	Can't configure PSTN to setup calls preferring G.729
Special configuration needed on IP-PBX	

10.5 PSTN to IP-PBX incoming call codec G.723

Test Case: 10.5	Title: PSTN to IP-PBX incoming call codec G.723
Test Procedure:	
Configure ITSP to use G.723. Make call into IP-PBX. – Try different VSI sizes	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	
Number B-party	
Date and Time	
Protocol SIP (IP-PBX)	
Expected Results:	
G.723 should have highest priority in SDP message. Make a note of actual CODEC used	
Results and Observations:	
Test Case result	<input type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input checked="" type="checkbox"/> NA
Findings	OPE doesn't support G.723
Special configuration needed on IP-PBX	

Codec

10.6 PSTN to IP-PBX incoming call codec G.711

Test Case: 10.6	Title: PSTN to IP-PBX incoming call codec G.711
Test Procedure: Configure ITSP to use G.711. – Make call into IP-PBX Try different VSI sizes	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302106111910
Number B-party	+302144163501
Date and Time	2016/09/01 15:00:29
Protocol SIP (IP-PBX)	
Expected Results: G.711 should have highest priority in SDP message. Make a note of actual CODEC used	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

10.7 PSTN to IP-PBX, codec reordering G.729 as preferred, IP-PBX supports G.711

Test Case: 10.7	Title: PSTN to IP-PBX, codec reordering G.729 as preferred, IP-PBX supports G.711
Test Procedure: Configure the SBC to reorder offers coming from MGCF to put G.729 first and G.711 second. Make a call from PSTN to IP-PBX. – Call setup succeeds. – Verify that voice is transmitted with G.729.	

Codec

Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	
Number B-party	
Date and Time	
Protocol SIP (IP-PBX)	
Results and Observations:	
Test Case result	<input type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input checked="" type="checkbox"/> NA
Findings	Can't configure PSTN to setup calls preferring G.729
Special configuration needed on IP-PBX	

10.8 IP-PBX to PSTN, IP-PBX prefers G.729 but supports G.711

Test Case: 10.8	Title: IP-PBX to PSTN, IP-PBX prefers G.729 but supports G.711
Test Procedure:	
Configure the PBX to prefer the G.729, but also support G.711.	
Make a call from IP-PBX 1 connected phone to PSTN.	
<ul style="list-style-type: none"> - Call setup succeeds. - Verify that voice is transmitted with G.729. 	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302144163502
Number B-party	+302106111866
Date and Time	2016/09/01 15:05:57
Protocol SIP (IP-PBX)	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

DTMF

11 DTMF

11.1 IP-PBX to PSTN, DTMF sending

Named Telephony Events to in-band.

Test Case: 11.1	Title: IP-PBX to PSTN, DTMF sending												
Test Procedure: Make a call (call from IP-PBX phone) to a PSTN phone. <ul style="list-style-type: none"> – Check that the call is established. – Check DTMF-tones (with application recognizing DTMF). A-party, release the call. <ul style="list-style-type: none"> – Check the release signalling sequence. 													
Conditions: <table> <tr> <td>IP-PBX</td> <td>Domain <provider></td> </tr> <tr> <td>PSTN</td> <td>PSTN/ISDN phone</td> </tr> <tr> <td>Number A-party</td> <td>+302144163502</td> </tr> <tr> <td>Number B-party</td> <td>+302106111866</td> </tr> <tr> <td>Date and Time</td> <td>2016/09/01 15:23:46</td> </tr> <tr> <td>Protocol SIP (IP-PBX)</td> <td></td> </tr> </table>		IP-PBX	Domain <provider>	PSTN	PSTN/ISDN phone	Number A-party	+302144163502	Number B-party	+302106111866	Date and Time	2016/09/01 15:23:46	Protocol SIP (IP-PBX)	
IP-PBX	Domain <provider>												
PSTN	PSTN/ISDN phone												
Number A-party	+302144163502												
Number B-party	+302106111866												
Date and Time	2016/09/01 15:23:46												
Protocol SIP (IP-PBX)													
Results and Observations: Test Case result <input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA Findings Special configuration needed on IP-PBX													

DTMF

11.2 PSTN to IP-PBX, DTMF receiving

In-band to Named Telephony Events.

Test Case: 11.2	Title: PSTN to IP-PBX, DTMF receiving
Test Procedure:	
<p>Make a call from PSTN phone to an IP-PBX phone</p> <ul style="list-style-type: none"> - Check that the call is established. - Check DTMF-tones (with application recognizing DTMF). <p>B-party, release the call.</p> <ul style="list-style-type: none"> - Check the release signalling sequence. 	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302106111910
Number B-party	+302144163501
Date and Time	2016/09/01 15:29:41
Protocol SIP (IP-PBX)	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	
Special configuration needed on IP-PBX	

12 Call diversion, CFU

12.1 PSTN to IP-PBX diverted to IP-PBX.

Recommended that calling party is ISDN (to test possible diversion indications).

B-party is permanently diverted to another extension number on the same IP-PBX.

Test Case: 12.1	Title: PSTN to IP-PBX diverted to IP-PBX.
Test Procedure:	
Divert IP-PBX extension 1 to extension 2 of the same IP-PBX.	
Make a call from a PSTN phone to IP-PBX extension 1.	
<ul style="list-style-type: none"> – Check that the call is established. – Check the voice channel. – Include in test result the displayed number on both phones. 	
A-party, release the call.	
<ul style="list-style-type: none"> – Check the release signalling sequence. 	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302106111910
Number B-party	+302144163504
Number C-party	501
Date and Time	2016/09/01 16:18:01
Protocol	SIP (IP-PBX)
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	Part A displays 2144163504, Party C displays 2106111910
Special configuration needed on IP-PBX	

12.2 PSTN to IP-PBX diverted to PSTN.

Recommended that calling party is ISDN (to test possible diversion indications).

B-party is permanently diverted to a number in PSTN.

Test Case: 12.2	Title: PSTN to IP-PBX diverted to PSTN.
Test Procedure:	
Divert IP-PBX extension 1 to a number in PSTN.	
Make a call from a PSTN phone to IP-PBX extension 1.	
<ul style="list-style-type: none"> – Check that the call is established. – Check the voice channel. – Include in test result the displayed number on both phones. 	
A-party, release the call.	
<ul style="list-style-type: none"> – Check the release signalling sequence. 	
Conditions:	

Call diversion, CFU

IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302106111910
Number B-party	+302144163504
Number C-party	+306945493331
Date and Time	2016/09/01 16:14:54
Protocol SIP (IP-PBX)	
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	Part A displays 2144163504, Party C displays +302144163500. OTE doesn't support receiving a CLI that's not part of the trunk's DDI range
Special configuration needed on IP-PBX	

Call transfer

13 Call transfer

13.1 PSTN to IP-PBX transfer with consultation to IP-PBX.

Recommended that calling party is ISDN (to test possible diversion indications).

B-party answers the call, and uses a "second line" function on the phone to manually forward the call to another extension on the same IP-PBX.

Test Case: 13.1	Title: PSTN to IP-PBX transfer with consultation to IP-PBX.
Test Procedure:	
<p>Make a call from a PSTN phone to IP-PBX extension 1.</p> <ul style="list-style-type: none"> – Answer the call on IP-PBX extension 1. – Check that the call (1) is established. – Check the voice channel of call (1) between PSTN phone and IP-PBX extension 1. <p>Set-up a call from IP-PBX extension 1 to IP-PBX extension 2 using the "second line" function.</p> <ul style="list-style-type: none"> – Answer the call on IP-PBX extension 2. – Check that the call (2) is established. – Check the voice channel of call (2) between IP-PBX extension 1 and IP-PBX extension 2 – Transfer the original PSTN call (1) to the second one and hang up the phone (in IP-PBX extension 1). – Check that the call (1) is transferred. – Check the voice channel between PSTN phone and IP-PBX extension 2. – Include in test result the displayed number on both phones after the transfer. <p>A-party, release the call.</p> <ul style="list-style-type: none"> – Check the release signalling sequence. 	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302106111910
Number B-party	+302144163501
Number C-party	502
Date and Time	2016/09/01 16:20:54
Protocol	SIP (IP-PBX)
Results and Observations:	
Test Case result	<input checked="" type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	Part A displays 2144163501, Party C displays 2106111910
Special configuration needed on IP-PBX	

Call transfer

13.2 PSTN to IP-PBX blind transfer to IP-PBX.

Recommended that calling party is ISDN (to test possible diversion indications).
 B-party answers the call, and uses a "second line" function on the phone to manually forward the call to another extension on the same IP-PBX.

Test Case: 13.2	Title: PSTN to IP-PBX blind transfer to IP-PBX.
Test Procedure:	
<p>Make a call from a PSTN phone to IP-PBX extension 1.</p> <p>Set-up a call from IP-PBX extension 1 to IP-PBX extension 2 using the "second line" function.</p> <p>Transfer the call (in IP-PBX extension 1) before answering the incoming call on the second number (blind transfer).</p> <ul style="list-style-type: none"> - Check that the call is established. - Check the voice channel. - Include in test result the displayed number on both phones after the transfer. <p>A-party, release the call.</p> <ul style="list-style-type: none"> - Check the release signalling sequence. 	
Conditions:	
IP-PBX	Domain <provider>
PSTN	PSTN/ISDN phone
Number A-party	+302106111910
Number B-party	+302144163501
Number C-party	502
Date and Time	2016/09/01 16:22:50
Protocol	SIP (IP-PBX)
Results and Observations:	
Test Case result	<input type="checkbox"/> Passed <input type="checkbox"/> Failed <input type="checkbox"/> Blocked <input type="checkbox"/> Inconclusive <input type="checkbox"/> NA
Findings	Part A displays 2144163501, Party C displays 2106111910
Special configuration needed on IP-PBX	

Overview of test results

14 Overview of test results

Case	Description	Result				
		Passed	Failed	Blocked	Inconcl.	NA
	Registration					
3.1	Single Account Successful registration	X				
3.2	Single Account Registration Refresh	X				
3.3	Single Account Registration Failure	X				
3.4	Single Account De-registration	X				
3.5	Multiple Account Successful registration	X				
3.6	Multiple Account Registration Refresh	X				
3.7	Multiple Account Registration Failure	X				
3.8	Multiple Account De-registration	X				
	Normal Call					
4.1	Outgoing call, release by A-party	X				
4.2	Outgoing call, early media	X				
4.3	Incoming call, release by B-party	X				
4.4	Outgoing call, long duration call	X				
4.5	Incoming call, long duration call	X				
	Call Abandon					
5.1	IP-PBX to PSTN, A hangs up during ringing	X				
5.2	IP-PBX to PSTN, B does not answer	X				
5.3	PSTN to IP-PBX, A hangs up during ringing	X				
5.4	PSTN to IP-PBX, B does not answer	X				
	Call Busy					
6.1	IP-PBX to PSTN, B is busy	X				
6.2	PSTN to IP-PBX, B is busy	X				
	CLIP					
7.1	IP-PBX to PSTN, Normal call, release by A-party	X				
7.2	PSTN to IP-PBX, Normal call, release by A-party	X				
	CLIR					
8.1	IP-PBX to PSTN, initiate CLIR	X				
8.2	PSTN to IP-PBX with header "Privacy" set to "id"	X				
	FAX/MODEM					
9.1	IP-PBX to PSTN, Modem	X				
9.2	IP-PBX to PSTN, Fax Clear Channel (G.711)		X			
9.3	PSTN to IP-PBX, Modem	X				
9.4	PSTN to IP-PBX, Fax Clear channel (G.711)		X			
9.5	IP-PBX to PSTN, Fax (T.38)	X				
9.6	PSTN to IP-PBX, Fax (T.38)	X				
9.7	IP-PBX to PSTN, G.729 voicecall fallback to Fax (T.38)	X				
9.8	IP-PBX to PSTN, G.729 voicecall fallback to G.711 for modem	X				
9.9	PSTN to IP-PBX, G.729 voicecall fallback to Fax (T.38)					X

Overview of test results

Case	Description	Result				
		Passed	Failed	Blocked	Inconcl.	NA
9.10	PSTN to IP-PBX, G.729 voicecall fallback to G.711 for modem					X
	CODEC					
10.1	IP-PBX to PSTN outgoing call codec G.729	X				
10.2	IP-PBX to PSTN outgoing call codec G.723					X
10.3	IP-PBX to PSTN outgoing call codec G.711	X				
10.4	PSTN to IP-PBX incoming call codec G.729					X
10.5	PSTN to IP-PBX incoming call codec G.723					X
10.6	PSTN to IP-PBX incoming call codec G.711	X				
10.7	PSTN to IP-PBX, codec reordering G.729 as preferred, IP-PBX supports G.711					X
10.8	IP-PBX to PSTN, IP-PBX prefers G.729 but supports G.711	X				
	DTMF					
11.1	IP-PBX to PSTN, DTMF sending	X				
11.2	PSTN to IP-PBX, DTMF receiving	X				
	CALL DIVERSION, CFU					
12.1	PSTN to IP-PBX diverted to IP-PBX.	X				
12.2	PSTN to IP-PBX diverted to PSTN.	X				
	CALL TRANSFER					
13.1	PSTN to IP-PBX transfer with consultation to IP-PBX.	X				
13.2	PSTN to IP-PBX blind transfer to IP-PBX	X				

Issue ID	Case	Description

Document Management

Revision History

Version Number	Revision date	Summary of Changes
0.1	15 December 2009	Creation
0.2	23 December 2009	<>
0.3	20 January 2010	<>
0.4	15 February 2010	<>
0.5	15 March 2010	<>
0.6	29 March 2010	<>
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1.0	4 October 2016	Document Approved

Approved by

Name	Company	Version
André Blokland	NEC Unified Solutions	1.0
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