CableLabs® draft proposal

SIP Forum SIPconnect1.1 Interop Test Plan

Draft

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Document Status

Document Title	SIP Forum SIPconnect1.1 CableLabs® draft proposal
Date	November 28, 2011
Status	Work in Progress

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1 INTRODUCTION

This suggested Interop Test Plan (ITP) defines a set of test procedures that can be used to verify interoperability between a SIP-PBX and an SIP-based Service Provider network. The test cases described in this document cover industry-accepted requirements for the SIP Trunking interface, including:

- SIP Forum SIPconnect 1.1 Technical Recommendation
- Applicable IETF RFCs, such as…
    - RFC3261 “SIP: Session Initiation Protocol”
    - RFC4566 “SDP: Session Description Protocol”
    - RFC3264 “An Offer/Answer Model with the Session Description Protocol (SDP)”
    - RFC6140 “Registration for Multiple Phone Numbers in the Session Initiation Protocol (SIP)”

Note that each test case in this ITP lists one or more requirements associated with the test case. It is important to note that the specific requirements listed are the primary focus of the test case, although additional requirements may also be tested.

This document is a work in progress (WIP), meaning that the test procedures herein are subject to change based on outcomes from testing, either in the SIP Forum community, CableLabs, or other third party test facilities.

1.1 Purpose and Scope

This ITP is intended to be used for testing interoperability between different vendor products and network elements involved in SIP Trunking. The scope of this document includes test cases for the “MUST” and “shall” requirements of the SIPconnect 1.1 Technical Recommendation, applicable, and other related SIP RFCs such as RFC6140.

The tests are divided into the following test groups:

- Configuration (Registration and Static Modes)
- Basic voice calls
- Extended call features
- Media and Session Interactions

1.2 Test Environment

The architecture of the test environment used for SIP Trunking ITP testing is shown in Figure 1.

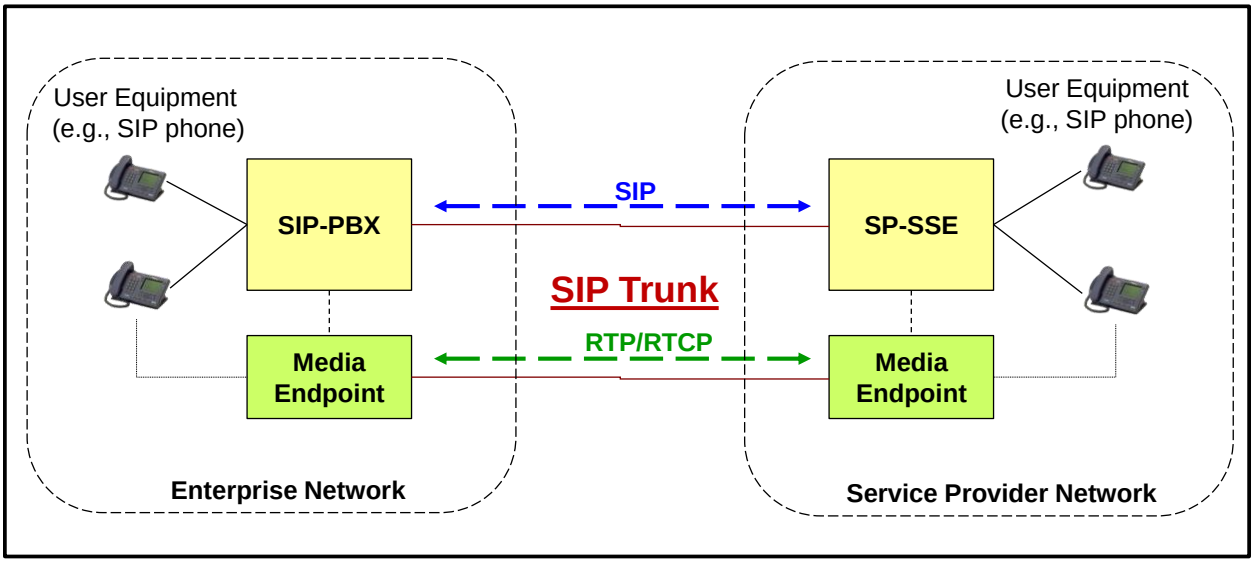

Figure 1. Test Environment

The test environment diagram, and the test plans that follow, use the naming conventions defined in SIPconnect1.1:

- The SIP-PBX and the SP-SSE (Service Provider SIP Signaling Entity) terminate the SIP Trunk SIP signaling interface between the enterprise and Service Provider network
- The Media Endpoint represents the termination point for RTP and RTCP streams in the enterprise and Service Provider network. The Media Endpoint could be the media termination point at the user equipment, or it could be a media server or media-relay device within the enterprise or Service Provider network.

2 TEST GROUP 1: SIP-PBX AND SP-SSE REQUIREMENTS

This section tests the basic interactions between the SIP-PBX in the enterprise network and SP-SSE located in Service Provider Network. Test cases include registration, security, basic call establishment, and call features commonly provided by a SIP-PBX.

2.1 Test Subgroup 1.1: Registration and Static Modes

1.1.1 Registration Mode

The test cases in this section assume that the SIP-PBX is configured for Registration mode instead of Static mode.

1.1.1.1 – SIP-PBX Support for Registration Mode

Objective: This test case verifies the SIP-PBX and SP-SSE support the GIN registration as per RFC6140.

Device Under Test (DUT): SIP-PBX, SP-SSE

Requirements Tested:

SIPConnect1.1: REQ24205, REQ24207, REQ24308, REQ24309, REQ24333, REQ24335, REQ24336, REQ24231, REQ24312, REQ24373

***RFC6140*:** RFC4607, RFC4608, RFC4609, RFC4610, RFC4611, RFC4612, RFC4613, RFC4614, RFC4615, RFC4616, RFC4617, RFC4618

Please note: In the case of the RFC6140 requirements tested above, "RFC" refers to the requirement tag number, not an actual IETF RFC. In other words, for RFCs that are tagged as specification requirements it is CableLabs policy to tag these requirements numbers with "RFC" in the same way as the "REQ" convention is used for CableLabs specs. The tagged requirements that are tested from RFC6140 are listed in the traceability matrix in Appendix A, and these coincide with the tag numbers listed as being tested in this test case.

Test Equipment: SIP-PBX with SIP phones, SIP Core, IP traffic sniffer for WAN-side

Test Case Setup:

- The SIP-PBX is configured to support Registration mode.
- The SIP-PBX is configured to register to a sub-domain (sp.redlab.com) provided by Service provider (sp.redlab.com) from one of its autonomous systems.
- The SIP-PBX is configured with the Service Provider's domain (i.e. "sp.redlab.com"), and the range of E.164 numbers.
- The SIP-PBX is SIP Connect 1.1 compliant and configured with a Registration AOR (i.e. sip:xyz@sp.redlab.com).
- The SIP core is configured with a table that associates the Registration AOR and Enterprise Public Identities assigned to the SIP-PBX.
- The SIP core is capable of handling RFC6140 registration based on Annex A of SIPConnect1.1.

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Start capturing packets on WAN-side.		NA	
2	Reset or restart SIP-PBX to trigger to send REGISTER message to SP-SSE.	SIP-PBX sends new REGISTER message.	NA	
3	Stop the packet captures and verify the REGISTER message from step 2.	Verify the domain name and headers in request URI of the Register message. • REGISTER → Request URI = sip: sp.redlab.com • REGISTER → Proxy-Require = gin • REGISTER → Require = gin • REGISTER → Supported = path • REGISTER → To = sip:pbx-1@sp.redlab.com • REGISTER → From = sip:pbx-1@sp.redlab.com • REGISTER → Contact = sip:192.0.2.4;bnc (or corresponding ip address)	☐ Pass ☐ Fail	REQ24312 REQ24333 REQ24335 REQ24336 RFC4607 RFC4608 RFC4609 RFC4610 RFC4611 RFC4612 RFC4613 RFC4614 RFC4615 RFC4616 RFC4617 RFC4618
4	Start capturing the traffic on WAN-side again.		NA	
5	To verify that SP-SSE has associated SIP-PBX's registration AOR with all the E.164 numbers, from outside of the SIP-PBX network, make a call to a phone number associated with the SIP-PBX (i.e. 303-661-1001)	Verify that the SIP phone rings.	☐ Pass ☐ Fail	
6	Go on hook on the phone from step 5 and end the call.		NA	
7	Stop the capture from step 4.		NA	
8	Find the INVITE message from step 5 from the capture.	Verify the INVITE message from the core uses IP address, instead of the domain name. (ex. INVITE sip:+13036611001@192.0.2.4)	☐ Pass ☐ Fail	REQ24373
9	Clean Up.		NA	

Post Test Procedure:

End of Test Case

1.1.2 Registration Mode – Security and Authentication

1.1.2.1 – Registration Mode When TLS is Enabled

Objective: This test case verifies the SIP-PBX and SP-SSE support for GIN registration as per RFC6140 when TLS is enabled.

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24205, REQ24207, REQ24211, REQ24213, REQ24308, REQ24309, REQ24313, REQ24314, REQ24315, REQ24318, REQ24319, REQ24320, REQ24321, REQ24325, REQ24326, REQ24333, REQ24372, REQ24386, REQ24331

Test Equipment: SIP-PBX with SIP phones, SIP Core, IP traffic sniffer for WAN-side

Test Case Setup:

- SIP-PBX is configured to support Registration mode.
- The SIP-PBX is configured to register to a sub-domain (sp.redlab.com) provided by Service provider (sp.redlab.com) from one of its autonomous systems.
- The SIP-PBX is configured with the Service Provider's domain (i.e. "sp.redlab.com"), and the range of E.164 numbers.
- The SIP-PBX is SIP Connect 1.1 compliant and configured with a Registration AOR (i.e. sip:xyz@sp.redlab.com).
- The SIP core is configured with a table that associates the Registration AOR and Enterprise Public Identities assigned to the SIP-PBX.
- The SIP core is capable of handling RFC6140 registration based on Annex A of SIPConnect1.1.

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Start capturing packets on the LAN and WAN side.		NA	
2	Reset or restart SIP-PBX to trigger to send REGISTER message to SP-SSE.	ESG sends new REGISTER message. Registration with the SP-SSE is successful.	NA	
3	To verify that SIP-PBX has associated SP-SSE signaling address correctly, initiate a call from the SIP-PBX network to a phone number associated with the SP-SSE.	Verify that the SIP phone rings. Go off hook and verify voice path.	☐ Pass ☐ Fail	
4	Go on hook on both phones and end the call.		NA	

Step #	Action	Expected Results	Pass/ Fail	Requirement #
5	To verify that SIP-PBX has associated SP-SSE signaling address correctly, initiate a call from the SIP-PBX network to a phone number associated with the SP-SSE.	Verify that the SIP phone rings. Go off hook and verify voice path.	☐ Pass ☐ Fail	
6	Go on hook on both phones and end the call.		NA	
7	Stop the IP capture and save trace file		NA	
8	Find the REGISTER message from step 2 from the IP capture. Verify that prior to the REGISTER the SP-SSE and the SIP-PBX establish a TLS session and that the mutual authentication is successful.	TLS mutual authentication is successful. TLS session is established before registration. No attempts to register via TCP or UDP are present.	☐ Pass ☐ Fail	REQ24211 REQ24314 REQ24319 REQ24320 REQ24321 REQ24325 REQ24386
9	Find the INVITE message from step 3 from the IP capture. Verify that prior to the INVITE the SP-SSE and the SIP-PBX establish a TLS session and that the mutual authentication is successful.	TLS mutual authentication is successful.	☐ Pass ☐ Fail	REQ24211 REQ24314 REQ24320 REQ24326
10	Find the INVITE message from step 5 from the IP capture. Verify that prior to the INVITE the SP-SSE and the SIP-PBX establish a TLS session and that the mutual authentication is successful.	TLS mutual authentication is successful.	☐ Pass ☐ Fail	REQ24211 REQ24314 REQ24320 REQ24326
11	Clean Up.		NA	

Post Test Procedure:

End of Test Case

1.1.2.2 – Registration with Authentication

Objective: This test case verifies how SIP-PBX behaves if 401 Unauthorized is received from Service Provider after the REGISTER message was sent.

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24368, REQ24370, REQ24371, REQ24337, REQ24327

Test Equipment: IP traffic sniffer

Test Case Setup:

- SIP-PBX is configured for Registration mode
- SIP-PBX and SP-SSE are configured to use digest authentication during registration

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Reset or restart SIP-PBX.	SIP-PBX attempts a new registration.	NA	
2	SIP-PBX begins initial registration by sending REGISTER message.		NA	
3	SP-SSE transmits a 401-REGISTER response. Record the values of the REGISTER→Call-ID in the 401-REGISTER.	Verify the 401-REGISTER message includes the WWW-Authenticate header.	☐ Pass ☐ Fail	REQ24327
4	SIP-PBX Transmits a second REGISTER **Error! Reference source not found.** message.	SIP-PBX responds with second REGISTER containing credentials.	☐ Pass ☐ Fail	REQ24368 REQ24370 REQ24371 REQ24337
6	Verify the REGISTER message from the step 4.	Verify the following message header field - REGISTER →Security-Client header is NOT present. REGISTER →Security-Server header is NOT present. REGISTER →Security-Verify header is NOT present. Verify the following message header field - REGISTER →Call-ID = Recorded 401-REGISTER→Call-ID.	☐ Pass ☐ Fail	REQ24371
7	SP-SSE transmits a 200 OK response.	Registration with the SP-SSE is successful.	NA	
8	Clean up		NA	

Post Test Procedure:

End of Test Case

1.1.2.3 – Registration Authentication Failure

Objective: This test case verifies the SIP-PBX behavior when more than three consecutive 401 responses are received to repeated registration attempts. The SIP-PBX must notify the administrator.

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24368, REQ24370, REQ24348, REQ24337, REQ24327, REQ24347

Test Equipment: IP traffic sniffer

Test Case Setup:

- SIP-PBX is configured for Registration mode.
- SIP-PBX is configured to send wrong password.
- SIP-PBX is configured to alert the administrator if it fails to successfully register.
- Only one SP-SSE address is configured for the SIP-PBX for registration.

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Reset or restart SIP-PBX.	SIP-PBX attempts a new registration.	NA	
2	SIP-PBX begins initial registration by sending REGISTER message.		NA	
3	SP-SSE replies with a 401-REGISTER response.	SIP-PBX transmits a second REGISTER message.	☐ Pass ☐ Fail	REQ24347
4	SP-SSE replies with a 401-REGISTER response.	SIP-PBX transmits a third REGISTER message.	☐ Pass ☐ Fail	REQ24347
5	SP-SSE replies with a 401-REGISTER response.	SIP-PBX transmits a fourth REGISTER message.	☐ Pass ☐ Fail	REQ24347
6	SP-SSE replies with a 401-REGISTER response.	Administrator is notified about the failure of registration. (i.e. via e-mail, pager) Note: SIP-PBX may continue sending REGISTER messages based on Timer F value and if backoff procedures are followed.	☐ Pass ☐ Fail	REQ24327 REQ24348 REQ24337
6	Clean up		NA	

Post Test Procedure:

End of Test Case

1.1.3 Registration Failures

1.1.3.1 – Registration Failure - SP-SSE unreachable

Objective: This test case verifies proper SIP-PBX behavior when multiple IP addresses configured for the SP-SSE are unreachable during registration.

DUT: SIP-PBX

Requirements Tested: SIPConnect1.1: REQ24339, REQ24341, REQ24342, REQ24362

Test Equipment: IP traffic sniffer

Test Case Setup:

- SIP-PBX is configured for Registration mode.
- Two SP-SSE addresses are configured for the SIP-PBX for registration.
- Timer F is configured to a value of 32 seconds on the SIP-PBX.
- Both SP-SSE addresses must be unresponsive to REGISTER requests from the SIP-PBX.

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Start IP capture. Reset or restart SIP-PBX.	SIP-PBX attempts a new registration.	NA	
2	SIP-PBX begins initial registration by sending REGISTER message to the first SP-SSE address.	Note time of initial REGISTER message.	NA	
3	The SP-SSE does not respond to the initial REGISTER message.		NA	
4	The SIP-PBX sends additional REGISTER messages to the first SP-SSE address.	Multiple REGISTER messages are noted to be sent up to 32 seconds.	☐ Pass ☐ Fail	REQ24339
5	After 32 seconds, the SIP-PBX begins to send REGISTER messages to the second IP address configured for the SP-SSE.	A REGISTER message is sent to the second IP address for the SP-SSE after 32 seconds. No additional REGISTER requests are sent to the first IP address for the SP-SSE.	☐ Pass ☐ Fail	REQ24339 REQ24341
6	Allow the REGISTER attempts to continue until expiry time of Timer F.	After 32 seconds, the REGISTER messages are noted to desist.	☐ Pass ☐ Fail	REQ24339
7	Wait for the SIP-PBX to resume sending REGISTER messages to the first IP address for the SP-SSE.	30 seconds must have elapsed between the last REGISTER message recorded in step 6 and the first REGISTER retry to the first SP-SSE IP address.	☐ Pass ☐ Fail	REQ24342
8	Allow the retry process to continue until the SIP-PBX no longer sends any REGISTER messages.	After 16 minutes (960 seconds) the SIP-PBX stops sending REGISTER requests. This behavior is noted for both IP addresses configured for the SP-SSE.	☐ Pass ☐ Fail	REQ24342

Step #	Action	Expected Results	Pass/ Fail	Requirement #
9	Verify that the SIP-PBX uses the correct back off algorithm when the SP-SSE is unreachable.	For each of the IP addresses that the SIP-PBX attempts to contact the time between REGISTER attempts is as follows: 30 seconds – first attempt 60 seconds – second attempt 120 seconds – third attempt 240 seconds – fourth attempt 480 seconds – fifth attempt 960 seconds – REGISTER attempts desist Additionally, each series of REGISTER attempts must be for a duration of 32 seconds.	☐ Pass ☐ Fail	REQ24339 REQ24342 REQ24362
10	Stop IP capture. Save capture file. Clean up.		NA	

Post Test Procedure:

End of Test Case

1.1.3.2 – Miscellaneous Registration Failures

Objective: This test case verifies proper SIP-PBX behavior when SIP error codes are received in response to REGISTER requests. Specifically, 404 Not Found, 480 Temporarily Unavailable, and 503 Service Unavailable are tested.

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24344, REQ24345, REQ24351, REQ24352, REQ24356, REQ24422, REQ24423, REQ24424, REQ24361, REQ24353

Test Equipment: SIP-PBX with SIP phones, SIP Core, IP traffic sniffer

Test Case Setup:

- SIP-PBX is configured for Registration mode.
- Initially, the Registration AOR for the SIP-PBX is not configured in the database of the SP-SSE.
- Only one IP address for the SP-SSE is configured for the SIP-PBX.
- Configure SIP-PBX as required to send notifications upon receipt of SIP error messages.
- Timer F is configured to a value of 32 seconds on the SIP-PBX.
- If available, SP-SSE is configured to send a Retry-After header value of 30 seconds.

Description:

- This test presumes the SP-SSE is configurable for the types of message responses required to execute the test steps. A test platform which emulates the SP-SSE may be used to execute this test.

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Start IP capture. Reset or restart SIP-PBX.	SIP-PBX attempts a new registration.	NA	
2	SIP-PBX begins initial registration by sending REGISTER message.		NA	
3	The SP-SSE sends a 404 Not Found message in response to the REGISTER.		☐ Pass ☐ Fail	REQ24344
4	The SIP-PBX does not register and sends a notification that the registration attempt has failed.	A notification of registration failure is sent to the system administrator. Note: If the SIP-PBX implements back off procedures additional REGISTER requests may be sent.	☐ Pass ☐ Fail	REQ24345
5	Stop IP capture and SIP-PBX registration attempts if required. Reconfigure SP-SSE so that the registration AOR of the SIP-PBX is available in its database. Configure SP-SSE and SIP-PBX for registration mode as required. Ensure that the SP-SSE is initially unresponsive to REGISTER requests from the SIP-PBX.		NA	
6	Start IP capture. Reset or restart SIP-PBX.	SIP-PBX attempts a new registration.	NA	
7	SIP-PBX begins initial registration by sending REGISTER message.	More than one REGISTER message may be seen.	NA	
8	Trigger SP-SSE to send a 480 Temporarily Unavailable message in response to the REGISTER.	Note the time of the 480 response and the time of the initial REGISTER message sent by the SIP-PBX.	NA	

Step #	Action	Expected Results	Pass/ Fail	Requirement #
9	Upon receipt of the 480 from the SP-SSE, the SIP-PBX treats the registration attempt as a failure.	The SIP-PBX follows back off procedures for retrying registration attempts as follows: After 16 minutes (960 seconds) the SIP-PBX stops sending REGISTER requests. The time between REGISTER attempts is as follows: 30 seconds – first attempt 60 seconds – second attempt 120 seconds – third attempt 240 seconds – fourth attempt 480 seconds – fifth attempt 960 seconds – REGISTER attempts desist Additionally, each series of REGISTER attempts must be for a duration of 32 seconds.	☐ Pass ☐ Fail	REQ24351 REQ24352
10	Stop IP capture. Reconfigure the SP-SSE to be administratively disabled or in an overload condition, such that it will send a 503 Service Unavailable message in response to a REGISTER request. If configurable, ensure that the SP-SSE sends a Retry-After header value of 30 seconds. Configure a second IP address for the SP-SSE that is also unresponsive to REGISTER requests from the SIP-PBX.		NA	
11	Start IP capture. Reset or restart SIP-PBX.	SIP-PBX attempts a new registration.	NA	
12	SIP-PBX begins initial registration by sending REGISTER message.	More than one REGISTER message may be seen.	NA	
13	Trigger SP-SSE to send a 503 Service Unavailable message in response to the REGISTER.	Note the time that the 503 response is sent.	☐ Pass ☐ Fail	REQ24353
14	Verify that the SIP-PBX sends the next REGISTER request after 30 seconds.	After the 503 response the next REGISTER message is sent in 30 seconds and that the request is sent to the same IP address that sent the 503 response. (Note: REGISTER requests to the second IP address for the SP-SSE must not be present.)	☐ Pass ☐ Fail	REQ24422 REQ24423

Step #	Action	Expected Results	Pass/ Fail	Requirement #
15	Stop IP capture. Verify in the IP capture that the Retry-After header is present in the REGISTER message from step 14.	The Retry-After header is present in the REGISTER and is set to a value of 30 seconds.	☐ Pass ☐ Fail	REQ24356 REQ24422
16	Reconfigure SP-SSE to send a Retry-After header value of 60 seconds in response to REGISTER requests. Maintain remaining test set up as described in step 10.		NA	
17	Start IP capture. Reset or restart SIP-PBX.	SIP-PBX attempts a new registration.	NA	
18	SIP-PBX begins initial registration by sending REGISTER message.	More than one REGISTER message may be seen.	NA	
19	Trigger SP-SSE to send a 503 Service Unavailable message in response to the REGISTER.	Note the time that the 503 response is sent.	NA	
20	Verify that the SIP-PBX sends the next REGISTER request after 60 seconds.	After the 503 response the next REGISTER message is sent in 60 seconds and that the request is sent to the second IP address that is configured for the SP-SSE. (Note: REGISTER requests to the IP address from which the 503 was sent must not be present.)	☐ Pass ☐ Fail	REQ24422 REQ24424
21	Wait for REGISTER retries to the second SP-SSE IP address to expire.		NA	
22	Verify that REGISTER requests to the original SP-SSE IP address resume after the expiration time in step 21.	The SIP-PBX reattempts to register with the original SP-SSE IP address.	☐ Pass ☐ Fail	REQ24361
23	Wait two minutes then stop IP capture. Stop SIP-PBX registration process if required.		NA	
24	Verify in the IP capture that the Retry-After header is present in the REGISTER message from step 22.	The Retry-After header is present in the REGISTER and is set to a value of 60 seconds.	☐ Pass ☐ Fail	REQ24356 REQ24424

Step #	Action	Expected Results	Pass/ Fail	Requirement #
25	Verify in the IP capture that the SIP-PBX made two additional REGISTER requests to the correct IP address and that the Retry-After header is not discarded.	The SIP-PBX sends two additional REGISTER requests to the original IP address configured for the SP-SSE. (Note: REGISTER requests to the IP address for the second IP address of the SP-SSE must not be present.) The Retry-After header is present in each of these REGISTER requests and it contains a value of 60 seconds.	☐ Pass ☐ Fail	REQ24361 REQ24424
26	Clean up		NA	

Post Test Procedure:

End of Test Case

1.1.4 Static Mode

1.1.4.1 – Static Mode via DNS Routing

Objective: This test case verifies the SIP-PBX and SP-SSE support for static mode when a publically accessible DNS server is used.

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24374, REQ24377, REQ24205, REQ24207, REQ24310, REQ24231, REQ24311, REQ24404, REQ24407, REQ24421

Test Equipment: SIP-PBX with SIP phones, SIP Core, IP traffic sniffer

Test Case Setup:

- SIP-PBX is configured to support static mode. The SIP-PBX is assigned an FQDN that is a sub-domain of the Service Provide domain (e.g., pbx1.redlab.com).
- Configure the DNS server to provide the realm (e.g. sp.redlab.com), IP address, port and transport protocol (UDP) of the SIP PBX.
- Configure SP-SSE as may be required to provide the domain name of the enterprise prior to starting the test.
- Configure NAT/firewalls on the SIP-PBX and SP-SSE as may be required to enable direct communication between these network elements.
- TLS is not configured for this test.

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Start capturing packets on LAN and WAN sides.		NA	
2	Reset SP-SSE or otherwise trigger it to perform a DNS SRV query to resolve the domain name of the SIP-PBX.	A DNS SRV query is noted to be sent	NA	

Step #	Action	Expected Results	Pass/ Fail	Requirement #
3	Verify that the domain name pre-configured on the SP-SSE resolves to the IP address and port of the SIP PBX.	The enterprise domain is correctly resolved via DNS.	☐ Pass ☐ Fail	REQ24374 REQ24310
4	To verify that SP-SSE has associated the SIP-PBX signaling address correctly, initiate a call from outside of the SIP-PBX network to a phone number associated with the SIP-PBX.	Verify that the SIP phone rings. Go off hook and verify voice path.	☐ Pass ☐ Fail	REQ24377
5	Go on hook on both phones and end the call.		NA	
6	Stop the IP capture and save trace file		NA	
7	Find the INVITE message from step 4 from the IP capture.	Verify the INVITE message from the core uses the SIP-PBX FQDN in the Request URI, e.g. INVITE sip:+13036611001@pbx1.redlab.com;user=phone	☐ Pass ☐ Fail	REQ24310
8	Verify in the IP Capture that no attempts to register with the operator's network have been made.	No REGISTER messages are present in the IP capture.	☐ Pass ☐ Fail	
9	Start IP capture on LAN and WAN sides. To verify that SIP-PBX has associated SP-SSE signaling address correctly, initiate a call from the SIP-PBX network to a phone number associated with the SP-SSE.	Verify that the SIP phone rings. Go off hook and verify voice path.	☐ Pass ☐ Fail	
10	Go on hook on both phones and end the call.		NA	
11	Stop the IP capture and save trace file		NA	
12	Find the INVITE message from step 9 from the IP capture.	Verify the INVITE message from the core uses either the IP address or the FQDN in the Request URI, e.g. INVITE sip:+13036611001@sp.redlab.com;user=phone	☐ Pass ☐ Fail	REQ24310
12	Clean Up.		NA	

Post Test Procedure:

End of Test Case

1.1.4.2 – Static Mode via Manually Configured Routing

Objective: This test case verifies the SIP-PBX and SP-SSE support for static mode when the FQDN→IP address routing information is manually configured at the SP-SSE and the SIP-PBX.

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24374, REQ24377, REQ24205, REQ24207, REQ24310, REQ24231, REQ24407, REQ24421

Test Equipment: SIP-PBX with SIP phones, SIP Core, IP traffic sniffer

Test Case Setup:

- SIP-PBX is configured to support static mode. The SIP-PBX is assigned an FQDN that is a sub-domain of the Service Provide domain (e.g., pbx1.redlab.com).
- Configure SP-SSE as may be required to provide the FQDN, IP address, port and transport protocol of the enterprise prior to starting the test.
- Configure SIP-PBX as may be required for the SP-SSE.
- Configure NAT/firewalls on the SIP-PBX and SP-SSE as may be required to enable direct communication between these network elements.
- TLS is not configured for this test.

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Start capturing packets on LAN and WAN sides.		NA	
2	To verify that the SP-SSE and the associated SIP-PBX are configured correctly, initiate a call from outside of the SIP-PBX network to a phone number associated with the SIP-PBX.	Verify that the SIP phone rings. Go off hook and verify voice path.	☐ Pass ☐ Fail	REQ24310 REQ24374
3	Go on hook on both phones and end the call.		NA	
4	Stop the IP capture and save trace file		NA	
5	Find the INVITE message from step 2 from the IP capture.	Verify the INVITE message from the core uses the FQDN in the Request URI, e.g. INVITE sip:+13036611001@sippbx.redlab.com;user=phone	☐ Pass ☐ Fail	
6	Verify in the IP Capture that no attempts to register with the operator's network have been made.	No REGISTER messages are present in the IP capture.	☐ Pass ☐ Fail	

Step #	Action	Expected Results	Pass/ Fail	Requirement #
7	Start IP capture on LAN and WAN sides. To verify that SIP-PBX has associated SP-SSE signaling address correctly, initiate a call from the SIP-PBX network to a phone number associated with the SP-SSE.	Verify that the SIP phone rings. Go off hook and verify voice path.	☐ Pass ☐ Fail	REQ24310 REQ24374
8	Go on hook on both phones and end the call.		NA	
9	Stop the IP capture and save trace file		NA	
10	Find the INVITE message from step 9 from the IP capture.	Verify the INVITE message from the core uses either the IP address or the FQDN in the Request URI, e.g. INVITE sip:+13036611001@sp.redlab.com;user=phone	☐ Pass ☐ Fail	
11	Clean Up.		NA	

Post Test Procedure:

End of Test Case

1.1.4.3 – Static Mode via DNS Routing when TLS is Enabled

Objective: This test case verifies the SIP-PBX and SP-SSE support for static mode configuration when a publically accessible DNS server is used and TLS is enabled.

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24205, REQ24207, REQ24211, REQ24213, REQ24374, REQ24379, REQ24380, REQ24382, REQ24383, REQ24384, REQ24385, REQ24387, REQ24390, REQ24391, REQ24397, REQ24310, REQ24313, REQ24314, REQ24315, REQ24318, REQ24320, REQ24372, REQ24409, REQ24311. REQ24404, REQ24407, REQ24421, REQ24409

Test Equipment: SIP-PBX with SIP phones, SIP Core, IP traffic sniffer

Test Case Setup:

- SIP-PBX is configured to support static mode. The SIP-PBX is assigned an FQDN that is a sub-domain of the Service Provide domain (e.g., pbx1.redlab.com).
- Configure the DNS server to provide the realm (e.g. sp.redlab.com), IP address, port and transport protocol (UDP) of the SIP PBX.
- Configure SP-SSE as may be required to provide the domain name of the enterprise prior to starting the test.
- Configure NAT/firewalls on the SIP-PBX and SP-SSE as may be required to enable direct communication between these network elements.
- TLS is configured on both the SP-SSE and the SIP-PBX for this test.

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Start capturing packets on LAN and WAN sides.		NA	
2	Reset SP-SSE or otherwise trigger it to perform a DNS SRV query to resolve the domain name of the SIP-PBX.	A DNS SRV query is noted to be sent	NA	
3	Verify that the domain name pre-configured on the SP-SSE resolves to the IP address and port of the SIP PBX.	The enterprise domain is correctly resolved via DNS.	☐ Pass ☐ Fail	
4	To verify that SP-SSE has associated SIP-PBX signaling address correctly, initiate a call from outside of the SIP-PBX network to a phone number associated with the SIP-PBX.	Verify that the SIP phone rings. Go off hook and verify voice path.	☐ Pass ☐ Fail	
5	Go on hook on both phones and end the call.		NA	
6	To verify that SIP-PBX has associated SP-SSE signaling address correctly, initiate a call from the SIP-PBX network to a phone number associated with the SP-SSE.	Verify that the SIP phone rings. Go off hook and verify voice path.	☐ Pass ☐ Fail	
7	Go on hook on both phones and end the call.		NA	
8	Stop the IP capture and save trace file		NA	
9	Find the INVITE message from step 4 from the IP capture. Verify that prior to the INVITE the SP-SSE and the SIP-PBX establish a TLS session and that the mutual authentication is successful.	TLS mutual authentication is successful.	☐ Pass ☐ Fail	REQ24380 REQ24382 REQ24409
10	Find the INVITE message from step 6 from the IP capture. Verify that prior to the INVITE the SP-SSE and the SIP-PBX establish a TLS session and that the mutual authentication is successful.	TLS mutual authentication is successful.	☐ Pass ☐ Fail	REQ24380 REQ24382 REQ24409
11	Clean Up.		NA	

Post Test Procedure:

End of Test Case

1.1.4.4 – Static Mode via Manually Configured Routing When TLS is Enabled

Objective: This test case verifies the SIP-PBX and SP-SSE support for static mode configuration when FQDN→IP address routing information is manually configured at the SP-SSE and the SIP-PBX and TLS is enabled.

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24205, REQ24207, REQ24211, REQ24213, REQ24374, REQ24379, REQ24380, REQ24382, REQ24383, REQ24384, REQ24385, REQ24387, REQ24390, REQ24391, REQ24397, REQ24310, REQ24313, REQ24314, REQ24315, REQ24318, REQ24320, REQ24372, REQ24407, REQ24421, REQ24409

Test Equipment: SIP-PBX with SIP phones, SIP Core, IP traffic sniffer

Test Case Setup:

- SIP-PBX is configured to support static mode. The SIP-PBX is assigned an FQDN that is a sub-domain of the Service Provide domain (e.g., pbx1.redlab.com).
- Configure SP-SSE as may be required to provide the FQDN, IP address, port and transport protocol of the enterprise prior to starting the test.
- Configure SIP-PBX as may be required for the SP-SSE.
- Configure NAT/firewalls on the SIP-PBX and SP-SSE as may be required to enable direct communication between these network elements.
- TLS is configured on both the SP-SSE and the SIP-PBX for this test.

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Start capturing packets on LAN and WAN sides.		NA	
2	To verify that the SP-SSE and the associated SIP-PBX are configured correctly, initiate a call from outside of the SIP-PBX network to a phone number associated with the SIP-PBX.	Verify that the SIP phone rings. Go off hook and verify voice path.	☐ Pass ☐ Fail	
3	Go on hook on both phones and end the call.		NA	
4	To verify that SIP-PBX has associated SP-SSE signaling address correctly, initiate a call from the SIP-PBX network to a phone number associated with the SP-SSE.	Verify that the SIP phone rings. Go off hook and verify voice path.	☐ Pass ☐ Fail	
5	Go on hook on both phones and end the call.		NA	
6	Stop the IP capture and save trace file		NA	

Step #	Action	Expected Results	Pass/ Fail	Requirement #
7	Find the INVITE message from step 2 from the IP capture. Verify that prior to the INVITE the SP-SSE and the SIP-PBX establish a TLS session and that the mutual authentication is successful.	TLS mutual authentication is successful.	☐ Pass ☐ Fail	REQ24380 REQ24382 REQ24409
8	Find the INVITE message from step 4 from the IP capture. Verify that prior to the INVITE the SP-SSE and the SIP-PBX establish a TLS session and that the mutual authentication is successful.	TLS mutual authentication is successful.	☐ Pass ☐ Fail	REQ24380 REQ24382 REQ24409
9	Clean Up.		NA	

Post Test Procedure:

End of Test Case

2.2 Test Subgroup 1.2: Basic Call Establishment

1.2.1 SIP-PBX Call Origination – DOD

1.2.1.1 – Verification of INVITE Message Parameters When Originating a DOD Call

Objective: This test case verifies outgoing INVITE message parameters for an outgoing call from the SIP-PBX to the Service Provider network with an E.164 number.

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24225, REQ24227, REQ24228, REQ24229, REQ24246, REQ24247, REQ24248, REQ24249, REQ24250

Test Equipment: packet analyzer

Test Case Setup:

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Using the phone attached to the SIP-PBX network, originate a call to outside of the SIP-PBX network.		NA	

Step #	Action	Expected Results	Pass/ Fail	Requirement #
2	Verify the INVITE message from the SIP-PBX from step 1.	Verify the Request-URI header in INVITE message is populated with one of the following forms: • sip:+13036611001@sp.redlab.com; user=phone • sip:3036611001@sp.redlab.com	☐ Pass ☐ Fail	REQ24225 REQ24227 REQ24228 REQ24229 REQ24246 REQ24247 REQ24248
3	Verify the INVITE message from the SIP-PBX from step 1.	Verify that To header field in the INVITE message is populated with one of the following forms: • sip:+13036611001@sp.redlab.com; user=phone • sip:3036611234@sp.redlab.com	☐ Pass ☐ Fail	REQ24225 REQ24227 REQ24228 REQ24229 REQ24246 REQ24247 REQ24248
4	Verify the INVITE message from the SIP-PBX from step 1.	Verify that the INVITE message contains a P-Asserted-Identity header field with one the following forms. • P-Asserted-Identity: "Joe Smith" <sip: +13036621001@ sp.redlab.com; user=phone> • P-Asserted-Identity: tel:+13036621001	☐ Pass ☐ Fail	REQ24249
5	Verify the INVITE message from the SIP-PBX from step 1.	Verify the From header in the INVITE message contains the URI with the following elements: • The Service provider domain name (i.e. From:<sip:+13036621001@ sp.redlab.com;user=phone >)	☐ Pass ☐ Fail	REQ24250
6	Check the rest of the messages.	Verify that the SP-SSE responds with 100 Trying message.	☐ Pass ☐ Fail	
7	Check the rest of the messages.	Verify that the SP-SSE responds with 200 OK.	☐ Pass ☐ Fail	
8	Clean up.		NA	

Post Test Procedure:

End of Test Case

1.2.1.2 – Presence of Privacy Header In DOD INVITE Message for Caller ID (Trusted Network)

Objective: This test case verifies outgoing INVITE messages from SIP-PBX for DOD calls when caller id delivery is suppressed and the SIP-PBX trusts the Service Provider network.

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24249, REQ24251, REQ24252, REQ24254

Test Equipment: packet analyzer

Test Case Setup:

- The SIP-PBX is configured to suppress the caller id when originating a call.
- The SP-SSE is configured to honor the privacy requests of the SIP-PBX.

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Using the phones from the SIP-PBX network originate a call to outside of the SIP-PBX network.		NA	
2	Verify the INVITE message from the SIP-PBX from step 1.	Verify the INVITE message contains the Privacy header with the value, 'id'.	☐ Pass ☐ Fail	REQ24252 REQ24254
3	Verify the INVITE message from the SIP-PBX from step 1.	Verify the From header field contains following elements: • "Anonymous"<sip:anonymous@anonymous.invalid> •	☐ Pass ☐ Fail	REQ24251 REQ24252
4	Verify the INVITE message from the SIP-PBX from step 1.	Verify that the INVITE message contains a P-Asserted-Identity header field with one the following forms. • P-Asserted-Identity: "Joe Smith" <sip: +13036621001@ sp.redlab.com ;user=phone> • P-Asserted-Identity: tel:+13036621001	☐ Pass ☐ Fail	REQ24249
5	Check the Caller ID display on the receiving side.	Verify the caller ID is displayed as 'anonymous'.	☐ Pass ☐ Fail	
6	Clean up		NA	

Post Test Procedure:

End of Test Case

1.2.2 SIP-PBX Call Termination - DID

1.2.2.1 – Verification of INVITE Message Parameters When Terminating a DID Call

Objective: This test case verifies the incoming INVITE message is formatted as specified in the SIPConnect 1.1 for DID calls from SP-SSE to SIP-PBX.

DUT: SIP-PBX, SP-SSE

Requirements Tested:

SIPConnect1.1: REQ24233, REQ24243, REQ24225, REQ24227, REQ24228, REQ24229, REQ24232, REQ24233, REQ24234, REQ24230, REQ24236, REQ24240, REQ24241, REQ24398

Test Equipment: packet analyzer

Test Case Setup:

- SP-SSE is configured with the set of E.164 numbers assigned to the SIP-PBX.

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	From outside of SIP-PBX network, initiate a call to an enterprise public identity assigned to the SIP-PBX.	SP-SSE send INVITE message to the SIP-PBX.	NA	
2	Verify the INVITE message from the step 1.	Verify the user part of Request-URI in INVITE message has the correct format. (i.e. sip:+13036611001@10.32.1.5) • The phone number contains a "+" sign • The phone number does not contain a visual separators such as "-". • The INVITE message Request-URI host-name contains the IP address registered by the SIP-PBX.	☐ Pass ☐ Fail	REQ24225 REQ24227 REQ24228 REQ24229 REQ24232 REQ24233 REQ24234
3	Verify the INVITE message from the step 1.	Verify the From header and P-Asserted-Identity header fields both contain a SIP URI with the following elements: • The E.164 calling number in the user part • The Service provider domain name in the host name part • For e.g. From:<sip:+13036611001@sp.redlab.com;user=phone>	☐ Pass ☐ Fail	REQ24225 REQ24227 REQ24228 REQ24229 REQ24240 REQ24241
4	Check the rest of SIP messages.	Verify that the SIP-PBX responds with 100 Trying message.	☐ Pass ☐ Fail	
5	Check the rest of SIP messages.	Verify that the SIP-PBX responds with 200 OK message.	☐ Pass ☐ Fail	

Step #	Action	Expected Results	Pass/ Fail	Requirement #
6	Clean up.		NA	

Post Test Procedure:

End of Test Case

1.2.2.2 – Verification of DID INVITE Parameters for Anonymous Calls

Objective: This test case verifies that the incoming INVITE message for an anonymous DID call to the SIP-PBX from the SP-SSE is formatted as specified in SIPconnect 1.1 when the SP-SSE doesn't trust the SIP-PBX.

DUT: SP-SSE

Requirements Tested: SIPConnect1.1: REQ24241, REQ24242, REQ24250, REQ24251, REQ24235, REQ24238

Test Equipment: packet analyzer

Test Case Setup:

- The phone originating the call is configured to make a call anonymously.

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	From outside of SIP-PBX network, initiate a call to an extension registered by the SIP-PBX.	SP-SSE send INVITE message to IP-PBX.	NA	
2	Check the INVITE message from the step 1.	Verify the From header field URI contains: "Anonymous"<sip:anonymous@anonymous.invalid>	☐ Pass ☐ Fail	REQ24235 REQ24241 REQ24242 REQ24250 REQ24251
3	Check the INVITE message from the step 1.	Verify that there is no P-Asserted-Identity included in the INVITE message.	☐ Pass ☐ Fail	REQ24238 REQ24251
4	Check the Caller ID display on the receiving side.	Verify the caller ID displayed is 'anonymous'.	☐ Pass ☐ Fail	
5	Clean up.		NA	

Post Test Procedure:

End of Test Case

2.3 Test Subgroup 1.3: Basic Features

1.3.1 Caller ID

1.3.1.1 – SIP-PBX Support for Calling Name and Number Delivery

Objective: This test case verifies that the SIP-PBX can send correct caller-id for an originating DOD call.

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24246, REQ24249

Test Equipment: packet analyzer, caller ID display

Test Case Setup: The SIP-PBX is configured to deliver the calling name and number of an originating user for a DOD call with privacy disabled. An extension served by the SIP-PBX is configured with a calling name (e.g., John Doe”). A line on the SP-SSE is equipped with a caller-ID display.

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Using an extension from the SIP-PBX network, originate a call to outside of the SIP-PBX network.	Verify that the INVITE has a P-Asserted-Identity header and a From header that both contain the display name and the Public User Identity for the originating line (e.g. “John Doe” <sip:+13036621001@ sp.redlab.com;user=phone>).	☐ Pass ☐ Fail	REQ24249 REQ24246
2	The phone at the terminating line with the caller ID display rings.	Verify the calling name and number are correctly displayed (i.e. “John Doe - 3036621001”).	☐ Pass ☐ Fail	
3	Clean Up.		NA	

Post Test Procedure:

End of Test Case

1.3.2 Call Forwarding

1.3.2.1 – SIP-PBX Support for Call Forwarding Using a New Dialog INVITE(A)

Objective: This test case verifies that the SIP-PBX forwards calls correctly.

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24255, REQ24246, REQ24247, REQ24249, REQ24256, REQ24398

Test Equipment: packet analyzer

Test Case Setup:

- An extension served by the SIP-PBX is configured to forward all calls to a phone served by the SP-SSE network.

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	From outside of SIP-PBX network, initiate a call to an extension served by the SIP-PBX that is configured to forward all calls to an SP-SSE destination. .	SP-SSE sends INVITE request to SIP-PBX. Note the name and number of the originating user contained in the P-Asserted-Identity header field of the INVITE request. For example: "John Doe" <sip: +13036621001@sp.redlab.com ;user=phone>)	NA	
2	SIP-PBX sends an INVITE request to the SP-SSE.		NA	
3	Verify the INVITE request from the step 2.	Verify the following message Request-URI: INVITE→Request Line→ Request URI is present	☐ Pass ☐ Fail	REQ24255
4	Verify the INVITE request from the step 2.	Verify the Request-URI in INVITE message has the correct format. (e.g. sip:+13036611001@sp.redlab.com; user=phone or sip:3036611001@sp.redlab.com) • The phone number does not contain a visual separators such as "-".	☐ Pass ☐ Fail	REQ24255 REQ24246 REQ24247
5	Verify the INVITE message from the step 2.	Verify the INVITE message Request-URI populated with a domain name follows one of the forms below: • Service provider domain name	☐ Pass ☐ Fail	REQ24246 REQ24247
6	Verify the INVITE message from the step 2.	Verify that the INVITE has a P-Asserted-Identity header that contains the display name and the Public User Identity for the originating user noted is step 1 (e.g. "John Doe" <sip: +13036621001@sp.redlab.com ;user=phone>).	☐ Pass ☐ Fail	REQ24249
7	The phone at the terminating end with the caller ID rings.	Verify the caller ID display shows correct id of the caller (i.e. "John Doe").	☐ Pass ☐ Fail	
8	Clean up.		NA	

Post Test Procedure:

End of Test Case

1.3.3 Call Transfer

1.3.3.1 – Blind Call Transfer - Transferor in SIP-PBX

Objective: This test case verifies that the SIP-PBX and SP-SSE support blind call transfer using the reINVITE request for the following case:

- SIP-PBX: transferor party
- SP-SSE: transferee and transfer-to parties

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24257, REQ24259, REQ24275

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Test Equipment: packet analyzer

Test Case Setup:

- SIP-PBX has configured the transferor extension to support call transfer.

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Using the phone (Transferor) from the SIP-PBX network, originate a call to outside of the SIP-PBX network (to the **Transferee**).	A stable two way call is established from the SIP-PBX transferor to the SP-SSE transferee.	NA	
2	Hook Flash on the transferor phone.	The transferor holds the media with the terminating party (Transferee) and plays a Recall dial tone. • Verify the SIP-PBX sends an INVITE to the transferee to hold the media (SDP a=inactive or sendonly) • Verify the SIP-PBX then plays dial-tone at the transferor and collect digits for second call origination.	☐ Pass ☐ Fail	REQ24257 REQ24259 REQ24289
3	On the transferor phone, dial the number for the transfer-to party on the SP-SSE.	Verify that upon detection of a valid digit string, the SIP-PBX initiates the second call via an INVITE request to the transfer-to party. Verify the transfer-to phone rings and that transferor hears ringback tone.	☐ Pass ☐ Fail	REQ24257 REQ24259
4	Initiate blind transfer on the transferor phone (e.g. Hang up or push blind	SIP-PBX sends INVITE to the Transferee. Verify the INVITE includes SDP offer.	☐ Pass ☐ Fail	REQ24257 REQ24259

Step #	Action	Expected Results	Pass/ Fail	Requirement #
	transfer button).	Verify the transfer-to phone displays the Caller ID of transferee. Verify the transferee hears ring back tone. Verify the transfer-to phone displays the Caller ID of transferee.		REQ24275 REQ24276
5	Answer call at transfer-to party.	Verify transferee and transfer-to establish a call and two-way voice is present.	☐ Pass ☐ Fail	REQ24257 REQ24259
6	Hang up on all phones and clean up.		NA	

Post Test Procedure:

End of Test Case

1.3.3.2 – Blind Call Transfer - Transferee in SIP-PBX

Objective: This test case verify that the SIP-PBX and SP-SSE support blind call transfer using the reINVITE request for the following case:

- SIP-PBX: transferee
- SP-SSE: transferor and transfer-to parties

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24257, REQ24259, REQ24299, REQ24300, REQ24302, REQ24303, REQ24275

Test Equipment: packet analyzer

Test Case Setup:

- SP-SSE has configured the transferor extension to support call transfer.

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Using the phone (Transferor) from the SP-SSE, originate a call to outside of the SIP-PBX network (Transferee).	A stable two way call is established between SP-SSE transferor and SIP-PBX transferee.	NA	
2	Hook Flash on the phone from the step 1 (SP-SSE Transferor).	The transferor holds the media with the terminating party (Transferee) and plays a Recall dial tone. • Verify the SP-SSE sends an INVITE to	NA	REQ24257 REQ24259 REQ24299 REQ24300

Step #	Action	Expected Results	Pass/ Fail	Requirement #
		the transferee to hold the media connection (SDP a=inactive or sendonly) • Verify the SP-SSE then plays dial-tone to the transferor and collects digits for second call origination.		REQ24302 REQ24275 REQ24276
3	On the phone from the step 1, Transferor (SP-SSE), dial the number for another phone (Transferred-to, SP-SSE).	Transferor initiates the second call to transfer-to party via an INVITE. Verify that the transfer-to phone rings and the transferor hears ringback tone.	NA	REQ24257 REQ24259
4	Initiate blind transfer on the transferor phone (e.g. Hang up (or push blind transfer button).).	Verify the transferee hears ring back tone.	☐ Pass ☐ Fail	REQ24299 REQ24300 REQ24302
5	Transfer-to at SP-SSE answers the call.	Verify transferee and transfer-to establish a call and two-way voice is present.	☐ Pass ☐ Fail	REQ24303
6	Hang up on all phones and clean up.		NA	

Post Test Procedure:

End of Test Case

1.3.3.3 – Blind Call Transfer – Transferee and Transfer-To in SIP-PBX

Objective: This test case verifies that the SIP-PBX and SP-SSE support blind call transfer using reINVITE for the following case:

- SIP-PBX: transferee and transfer-to parties
- SP-SSE: transferor party

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24257, REQ24259, REQ24299, REQ24300, REQ24302, REQ24303, REQ24269, REQ24259

Test Equipment: packet analyzer

Test Case Setup:

- The SP-SSE transferor extension is configured to support call transfer.

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Using the phone (Transferor) from the SP-	A stable two way call is established.	NA	

Step #	Action	Expected Results	Pass/ Fail	Requirement #
	SSE network, originate a call to outside of the SIP-PBX network (Transferee).			
2	Hook Flash on the transferor phone.	The transferor initiates media-hole toward the transferee, and plays a Recall dial tone at the transferor phone.	NA	
3	On the transferor dial the number of the transfer-to party on the SIP-PBX.	SP-SSE (Transferor) initiates the second call via an INVITE (with no SDP). Verify that the transfer-to phone rings and the transferor hears ringback tone.	☐ Pass ☐ Fail	REQ24257 REQ24259 REQ24300 REQ24269 REQ24259
	Initiate blind call transfer on the transferor phone (hang up or push the blind-call-transfer button).	Verify that the transferee hears ring back tone.	☐ Pass ☐ Fail	REQ24257 REQ24259 REQ24299
5	Answer call at transfer-to phone.	Verify that a 2-way voice connection is established between the transferee and transfer-to parties.	☐ Pass ☐ Fail	REQ24302 REQ24303
6	Hang up on phones and clean up.		NA	

Post Test Procedure:

End of Test Case

1.3.3.4 – Blind Call Transfer with Early Media – Transferee in SIP-PBX

Objective: This test case verifies that the SIP-PBX and SP-SSE support blind call transfer when the transfer-to party provides early media for the following case:

- SIP-PBX: transferee party
- SP-SSE: transferor and transfer-to parties

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPconnect1.1: REQ24257, REQ24259, REQ24268, REQ24269

Test Equipment: packet analyzer

Test Case Setup:

- SP-SSE has configured the transferor extension to support call transfer, and the transfer-to extension to provide early media (say some terminating feature like solicitor call blocking that requires early media).

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Using the phone (Transferor) from the SP-SSE network, originate a call to SIP-PBX (Transferee).	A stable two way call is established.	NA	
2	Hook Flash on the transferor phone	The transferor initiates media hold toward the transferee and plays a Recall dial tone.	NA	
3	Initiate a call to the transfer-to party from the transferor.	Verify that the transfer-to phone rings and the transferor hears early-media.	NA	
4	Initiate blind transfer on the transferor phone (e.g. Hang up or push blind transfer button).	Verify that the transferee hears the early media.	☐ Pass ☐ Fail	REQ24257 REQ24259
5	Answer call at transfer-to phone.	Verify that a 2-way voice connection is established between the transferee and transfer-to parties.	☐ Pass ☐ Fail	REQ24257 REQ24259 REQ24268 REQ24269
6	Hang up on phones and clean up.		NA	

Post Test Procedure:

End of Test Case

1.3.3.5 – Attended Call Transfer – Transferor in SIP-PBX

Objective: This test case verifies that the SIP-PBX and SP-SSE support attended call transfer using the INVITE message for the following case:

- SIP-PBX: transferor
- SP-SSE: transferee and transfer-to parties

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPconnect 1.1: REQ24257, REQ24259, REQ24268, REQ24269, REQ24302, REQ24303

Test Equipment: packet analyzer

Test Case Setup:

- SIP-PBX transferor is configured to support call transfer.

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Using the phone (Transferor) from the SIP-PBX network, originate a call to outside of the SIP-PBX network (SP-SSE, transferee).	A stable two way call is established.	NA	
2	Hook Flash on the transferor phone.	The SP-SE initiates media hold toward transferee and plays Recall dial tone. • Verify the SIP-PBX sends a re-INVITE to hold the remote party upon detection of a hookflash. • Verify the SIP-PBX then plays dial-tone and collect digits for second call origination.	☐ Pass ☐ Fail	REQ24257 REQ24259 REQ24302
3	Initiate a 2nd call from the transferor phone to the transfer-to party.	Verify that upon detection of a valid digit string, the SIP-PBX initiates the second call via an INVITE. Verify that the transfer-to phone rings and the transferor hears ringback tone.	☐ Pass ☐ Fail	REQ24257 REQ24259 REQ24268 REQ24269 REQ24303
4	Answer the call at the transfer-to phone.	Verify that a two way voice connection is established between the transferor and the transfer-to party.	☐ Pass ☐ Fail	
5	Hang up on the transferor phone (Transferor).	Verify that a 2-way voice connection is established between the transferee and transfer-to party. Verify that any media path reconfigurations are done using reINVITE request (and not REFER request).	☐ Pass ☐ Fail	REQ24257 REQ24259
6	Hang up on all phones and clean up.		NA	

Post Test Procedure:

End of Test Case

1.3.3.6 – Blind Call Transfer -- Transferor and Transferee in SIP-PBX

Objective: This test case verifies that the SIP-PBX and SP-SSE support blind call transfer for the following case:

- SIP-PBX: transferor, transferee
- SP-SSE: transfer-to

DUT: SIP-PBX, SP-SSE

Requirements Tested: SipConnect 1.1: REQ24257, REQ24259, REQ24268, REQ24269, REQ24302, REQ24303

Test Equipment: packet analyzer

Test Case Setup:

- SIP-PBX transferor is configured to support call transfer.

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Using the phone (Transferor) from the SIP-PBX network, originate a call to another extension within the SIP-PBX network (**Transferee**).	A stable two way call is established.	NA	
2	Hook Flash on the phone from the step 1 (SIP-PBX, Transferor).	The transferor holds the media with the terminating party (Transferee) and plays a Recall dial tone. • Verify the SIP-PBX then plays dial-tone on the transferor and collect digits for second call origination.	☐ Pass ☐ Fail	REQ24302
3	On the transferor, dial the number of the transfer-to party.	Verify that upon detection of a valid digit string, the SIP-PBX initiates the second call to the transfer-to party via an INVITE request. Verify that the transfer-to phone rings and the transferor hears ringback tone.	☐ Pass ☐ Fail	REQ24257 REQ24259 REQ24268 REQ24269
4	Initiate blind transfer on the transferor phone (e.g. Hang up or push blind transfer button).	Verify that the transfer-to phone continues to ring. Verify that the transfer-to phone displays the caller-ID of the transferee user. Verify that the transferee hears ringback tone.	☐ Pass ☐ Fail	REQ24257 REQ24259 REQ24268 REQ24269 REQ24302
5	Answer the call at the transfer-to phone.	Verify that a 2-way voice connection is established between the transferee and transfer-to phones.	☐ Pass ☐ Fail	REQ24257 REQ24259 REQ24303
6	Hang up on all phones and clean up.		NA	

Post Test Procedure:

End of Test Case

1.3.3.7 – Blind Call Transfer -- Transferor and Transfer-to in SIP-PBX

Objective: This test case verifies that the SIP-PBX and SP-SSE support blind call transfer for the following case:

- SIP-PBX: transferor, transfer-to
- SP-SSE: transferee

DUT: SIP-PBX. SP-SSE

Requirements Tested: REQ24257, REQ24259, REQ24268, REQ24269, REQ24302, REQ24303, REQ24275

-

Test Equipment: packet analyzer

Test Case Setup:

- SIP-PBX transferor line is configured to support call transfer.

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Using the phone (Transferor) from the SIP-PBX network, originate a call to outside of the SIP-PBX network (SP-SSE, Transferee).	A stable two way call is established.	NA	
2	Hook Flash on the transferor phone.	The transferor initiates media hold toward the transferee party, and plays Recall dial tone. • Verify the SIP-PBX sends an INVITE to hold the transferee party (SDP: a=inactive, sendonly) • Verify the SIP-PBX then plays dial-tone and collect digits for second call origination from the transferor extension.	☐ Pass ☐ Fail	REQ24257 REQ24259 REQ24268 REQ24269 REQ24275 REQ24276
3	On the transferor phone , dial the number for the transfer-to party.	Verify that upon detection of a valid digit string, the SIP-PBX initiates the second call to the transfer-to party via an INVITE. Verify that the transfer-to party rings and the transferor hears ringback tone.	☐ Pass ☐ Fail	
4	Initiate blind transfer on the transferor phone (e.g. Hang up or push blind transfer button).	Verify that the transferee hears ringback tone, and the transfer-to party continues to ring. Verify the transfer-to phone displays the Caller ID of the transferee.	☐ Pass ☐ Fail	REQ24257 REQ24259 REQ24268 REQ24269 REQ24302
5	Answer the transfer-to	Verify transferee and transfer-to establish a call	☐ Pass	REQ24303

Step #	Action	Expected Results	Pass/ Fail	Requirement #
	phone.	and two-way voice is present.	☐ Fail	
6	Hang up on all phones and clean up.		NA	

Post Test Procedure:

End of Test Case

1.3.3.8 – Blind Call Transfer -- Transferee and Transfer-to in SIP-PBX

Objective: This test case is to verify the SIP-PBX correctly handling the blind call transfer as a transferee and transfer-to, using the INVITE message.

- SIP-PBX: transferee and transfer-to
- SP-SSE: transferor

DUT: SIP-PBX, SP-SSE

Requirements Tested: REQ24257, REQ24259, REQ24268, REQ24269, REQ24302, REQ24303

Test Equipment: packet analyzer

Test Case Setup:

- SP-SSE is configured to support call transfer on the transferor extension.

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Using the phone (Transferor) from the SP-SSE, originate a call to an extension in SIP-PBX network (Transferee).	A stable two way call is established.	☐ Pass ☐ Fail	REQ24257 REQ24259
2	Hook Flash on the transferor.	Verify that the transferor initiates media hold toward the transferee using reINVITE, and plays a Recall dial tone.	NA	REQ24302
3	On the transferor phone, dial the number for the transfer-to party.	Verify that the transferor initiates the second call to the transfer-to party via an INVITE request. Verify that the transferee rings and the transferor hears ringback tone.	NA	REQ24257 REQ24259
4	Initiate blind transfer on the transferor phone (e.g. Hang up (or push blind transfer button).	Verify the transferee hears ring back tone and the transfer-to continues to ring.	☐ Pass ☐ Fail	REQ24302
5	Transfer-to at SIP-PBX answers the call.	Verify transferee and transfer-to establish a call and two-way voice is present.	☐ Pass ☐ Fail	REQ24257 REQ24259

Step #	Action	Expected Results	Pass/ Fail	Requirement #
				REQ24268 REQ24269 REQ24303
6	Hang up on all phones and clean up.		NA	

Post Test Procedure:

End of Test Case

1.3.3.9 – Blind Call Transfer with Early Media – Transferee and Transfer-to in SIP-PBX

Objective: This test case is to verify the SIP-PBX correctly produce the early media during call transfer.

- SIP-PBX: transfer-to
- SP-SSE: transferor, transferee

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24257, REQ24259, REQ24268, REQ24269

Test Equipment: packet analyzer

Test Case Setup:

- SP-SSE transferor is configured to support call transfer.
- .
- The SIP-PBX transfer-to line is configured to provide early-media

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Using the phone (Transferor) from the SP-SSE network, originate a call to another extension at SP-SSE (Transferee).	A stable two way call is established.	NA	
2	Hook Flash on the transferor phone.	The transferor initiates media-hold toward the transferee and plays local Recall dial tone.	NA	
3	On the transferor phone, dial the number of the transfer-to extension in the SIP-PBX phone.	Verify that the SP-SSE (Transferor) initiates the second call using an INVITE request to the transfer-to party in the SIP-PBX. Verify that the transfer-to phone rings and the transferor hears the early-media provided by the transfer-to party.	NA	
5	Initiate blind transfer on the transferor phone (e.g.	Verify the transferee hears the early media from the transfer-to party while the transfer-to phone	☐ Pass	REQ24257

Step #	Action	Expected Results	Pass/ Fail	Requirement #
	Hang up (or push blind transfer button).	continues to ring.	☐ Fail	REQ24259
5	Answer the call at the transfer-to party.	Verify transferee and transfer-to establish a call and two-way voice is present. Verify that any voice path reconfigurations are done within an INVITE transaction.	☐ Pass ☐ Fail	REQ24257 REQ24259 REQ24268 REQ24269
6	Hang up on phones and clean up.		NA	

Post Test Procedure:

End of Test Case

1.3.4 Call Hold

1.3.4.1 – Call Hold by SIP-PBX

Objective: This test case verifies that the SIP-PBX and SP-SSE support call hold (media hold) when it is initiated by a SIP-PBX user.

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24268, REQ24269, REQ24305, REQ24306

Test Equipment: packet analyzer, SP-SSE

Test Case Setup:

- SIP-PBX has configured an extension to allow call hold.
- SIP-PBX may be configured to provide music on hold (MOH).

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Start a traffic capture at both LAN and WAN-side.		NA	
2	Using the phone from the SIP-PBX (A) network, originate a call to outside of the SIP-PBX network (extension at SP-SSE).	Call session is established.	NA	
3	On SIP-PBX (A), start a call hold by pushing dedicated button for call hold or by entering special code.	The call is on hold and the extension outside of SIP-PBX is put on hold (No voice can be heard with each other, If MOH is configured, music can be heard).	☐ Pass ☐ Fail	
4	Cancel the call hold from SIP-PBX (A) side.	The call is back on normal session.	NA	
5	Hang up the call from SIP-PBX (A) side.	The session is disconnected.	NA	
6	Stop the captures and open the capture file for analysis.		NA	
7	Find the INVITE message where the SIP-PBX (A) initiated call hold.	Verify that the INVITE message includes an SDP. If the SIP-PBX is not providing MOH, then verify that the SDP contains the attribute "a=inactive". If the SIP-PBX is providing MOH, then verify the SDP contains an attribute “a=sendonly” and that the RTP stream is only one way from the SIP-PBX to the SP-SSE.	☐ Pass ☐ Fail	REQ24305 REQ24306
8	Find the INVITE message where the SIP-PBX (A) cancelled call hold.	Verify that the INVITE message includes SDP with either with an attribute of "a=sendrecv", or no “a=” attribute.	☐ Pass ☐ Fail	REQ24268 REQ24269
9	Clean up.		NA	

Post Test Procedure:

End of Test Case

1.3.4.2 – Call Hold by SP-SSE

Objective: This test case verifies that the SIP-PBX and SP-SSE support call hold when it is initiated by an SP-SSE user.

DUT: SP-SSE, SIP-PBX

Requirements Tested: SIPConnect1.1: REQ24268, REQ24269, REQ24305, REQ24306

Test Equipment: packet analyzer

Test Case Setup:

- SP-SSE has configured an extension to support initiation of call hold.
- SP-SSE may be configured to provide music on hold (MOH).

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Start a traffic capture at both LAN and WAN-side.		NA	
2	Using the phone from the SP-SSE network, originate a call to the SIP-PBX network (extension at SIP-PBX (A)).	Call session is established.	NA	
3	On SP-SSE extension, start a call hold by pushing dedicated button for call hold or by entering special code.	The extension on the SIP-PBX is put on hold.	NA	
4	Cancel the call hold from SP-SSE extension.	The call is back on normal session.	NA	
5	Hang up the call from SP-SSE extension.	The session is disconnected.	NA	
6	Stop the captures and open the capture file for analysis.		NA	
7	Find the INVITE message where the SP-SSE extension initiated call hold.	Verify that the INVITE message includes SDP. If MOH is not provided by the SP-SSE, then verify that the SDP contains attribute of "a=inactive". If MOH is provided by the SP-SSE, then verify the SDP contains “a=sendonly”, and the RTP stream is only one way from the SP-SSE to the SIP-PBX.	☐ Pass ☐ Fail	REQ24305 REQ24306
8	Find the INVITE message where the SP-SSE extension cancelled call hold.	Verify that the INVITE message includes SDP with no “a=” attribute, or an attribute of "a=sendrecv".	☐ Pass ☐ Fail	REQ24268 REQ24269
9	Clean up.		NA	

Post Test Procedure:

End of Test Case

1.3.5 Early Media

1.3.5.1 – Early Media

Objective: This test case verifies that the SIP-PBX can render early media received from the SP-SSE to the SIP-PBX user.

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24299, REQ24300, REQ24301, REQ24302, REQ24303

Test Equipment: packet analyzer

Test Case Setup:

- SIP-PBX supports early media.
- The terminating line on the SP-SSE is configured with a feature that will trigger 183 response containing an SDP message body to establish an early-media session.

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Using the phone from the SIP-PBX network, originate a call to outside of the SIP-PBX network (extension at SP-SSE).	Verify that the SIP-PBX sends an INVITE request containing an SDP-offer. Verify that the terminating phone in the SP-SSE rings. Verify that the SP-SSE responds to the INVITE with a 183 Session Progress response containing an SDP-preview to establish an early-media session. Verify that the SP-SSE sends an RTP stream to the SIP-PBX media address advertised in the c= line of the SDP-offer. Verify that the SIP-PBX sends an RTP stream to the SP-SSE media address advertised in the c= line of the SDP-preview. Verify that the originating SIP-PBX line does not play local ringback tone, but that the SIP-PBX renders the received RTP stream to the originating user.	☐ Pass ☐ Fail	REQ24301 REQ24299 REQ24300 REQ24302 REQ24303
4	Answer the call. (The session is established.)	Verify that there is two way talk between the originating and terminating phones.	☐ Pass ☐ Fail	
5	Clean up.		NA	

Post Test Procedure:

End of Test Case

1.3.6 Emergency Service

1.3.6.1 – Emergency Service

Objective: This test case verifies if the SIP-PBX correctly handles the emergency call.

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24261, REQ24262, REQ24263, REQ24264, REQ24265, REQ24266

Test Equipment: packet analyzer

Test Case Setup:

- The SIP-PBX extension is set to use enable 'privacy' for all calls.
- The SIP-PBX is configured to use 911 as the emergency number.

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Using the phone from the SIP-PBX network, originate an emergency call by entering 911.	Verify that the SIP-PBX sends INVITE message. Verify the INVITE's Request-URI is sip:911@sp.redlab.com. Verify that the INVITE message includes P-Asserted-identity header. Verify that the P-Asserted-identity header includes the user name and enterprise user identity of the caller.	☐ Pass ☐ Fail	REQ24261 REQ24262 REQ24263 REQ24264 REQ24265 REQ24266
2	Clean up.		NA	

Post Test Procedure:

End of Test Case

3 TEST GROUP 2: MEDIA AND SESSION INTERACTIONS

3.1 Test Subgroup 2.1: Media and Session Interactions

2.1.1 - Media and Session Interactions

2.1.1.1 – SIP-PBX Media Capability Negotiation Via SDP

Objective: This test case verifies that the SIP-PBX supports the SDP offer/answer procedures to establish a voice session for the case where it receives an offerless DID INVITE request.

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24268, REQ24269, REQ24273

Test Equipment:

- Packet analyzer

Test Case Setup: The SP-SSE must be configured to initiate a DID call to the SIP-PBX using an offerless INVITE.

Description: Send an offerless DID INVITE to the SIP-PBX, and verify that the SIP-PBX sends a subsequent SDP offer on the next reliable response to INVITE (in this case 200OK), and finally accepts an SDP answer received in the ACK to establish the 2-way voice connection.

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	Initiate a DID call from SP-SSE to SIP-PBX via an INVITE request containing no SDP offer. The INVITE request should not include “100rel” or “precondition” in Require or Supported header fields.		NA	
2	The SIP-PBX (A) sends 180 or 183 message.	Verify that the 18x response does not contain SDP.	☐ Pass ☐ Fail	REQ24268 REQ24269 REQ24273
4	Answer the call from the terminating SIP-PBX extension. The SIP-PBX (A) sends 200-INVITE.	Verify that the 200-INVITE message from SIP-PBX includes an SDP-offer. Verify that the SP-SSE responds with an ACK containing the SDP-answer. Verify that the call is established and there is a two way voice path.	☐ Pass ☐ Fail	REQ24269 REQ24273
5	Terminate the session and clean up.		NA	

Post Test Procedure:

End of Test Case

2.1.1.2 – SIP-PBX Codec Support and Negotiation Capabilities

Objective: This test case verifies that the SIP-PBX can correctly negotiate a voice codec.

DUT: SIP-PBX, SP-SSE

Requirements Tested: SIPConnect1.1: REQ24268, REQ24271, REQ24272, REQ24281, REQ24285, REQ24293, REQ24294

Test Equipment: packet analyzer

Test Case Setup:

Description:

Procedure:

Step #	Action	Expected Results	Pass/ Fail	Requirement #
1	From outside of SIP-PBX network, initiate a call to an extension inside the SIP-PBX network.	The INVITE includes SDP Offer.	NA	
2	SIP-PBX replies with 200-INVITE and includes SDP Answer.	Verify that there are supported CODECs included in the "Answer". (There can be multiple media descriptors.) Verify that there is only one session description in the answer.	☐ Pass ☐ Fail	REQ24268 REQ24281 REQ24271 REQ24272
3	Verify the Media descriptors in SDP from step 2.	Answer→m=audio	☐ Pass ☐ Fail	REQ24285 REQ24268
4	Verify the Format/Payload Type (PCMU Profile) in SDP from step 2.	Answer → 13	☐ Pass ☐ Fail	REQ24285 REQ24268
5	Verify the Supported CODEC, Port in SDP from step 2.	Answer →<1 - 65,535> Answer →RTP/AVP	☐ Pass ☐ Fail	REQ24268
6	Verify the Frame Size / Packetization Period in SDP from step 2.	Answer → a=ptime is equal to the ptime value from the Offer.	☐ Pass ☐ Fail	REQ24281 REQ24268
7	Verify the Bandwidth Modifier AS Max in bytes in SDP from step 2.	Answer → b=AS is equal to the value from the Offer.	☐ Pass ☐ Fail	REQ24268
8	Verify the Maximum Packets per second in SDP from step 2.	Answer → a=maxprate is equal to the value from the Offer.	☐ Pass ☐ Fail	REQ24268
9	Verify presence of supported events in the SDP.	Answer → a=fmtp is present along with events in the range of 0 – 15.	☐ Pass ☐ Fail	REQ24293 REQ24294
10	Clean up.		NA	

Post Test Procedure:

End of Test Case

APPENDIX A REQUIREMENTS TRACEABILITY

This section provides the mapping between the test cases described in this ITP and the mandatory requirements defined in SIPconnect1.1 and RFC6140 that are tested by those test cases.

REQ Tag	Requirement Text	Rqmt Category	DUT	SIP PBX TC Traced To
REQ24201	SIP-PBXs MUST support SIP in accordance with [RFC 3261] and offer-answer in accordance with [RFC 3264], as qualified by statements in later sections of this document.	MUST	SIP PBX, SP-SSE	All
REQ24399	This document specifies a profile of SIP, as well as specifying some media aspects. Implementations of this Technical Recommendation MUST NOT simply assume that a particular feature or option listed as mandatory in this document is supported by a peer SIP-PBX or SP-SSE.	MUST	NA	NA
REQ24203	Instead, a SIP-PBX MUST use mechanisms specified for SIP (e.g., Supported, Require and Allow header fields) and SDP (e.g., attributes, payload formats) for ascertaining support of a given SIP or SDP extension at a peer SP- SSE or SIP-PBX. Failure to do this can lead to interoperability problems.	MUST	SIP PBX, SP-SSE	1.3.3.1, 1.3.3.5
REQ24205	SIP-PBXs MUST support either Registration mode, as specified in Annex A, or Static mode, as described in Annex B	MUST	SIP PBX	1.1.1.1, 1.1.2.1, 1.1.4.1, 1.1.4.2, 1.1.4.3, 1.1.4.4
REQ24207	SP-SSEs MUST support either Registration mode, as specified in Annex A, or Static mode, as described in Annex B.	MUST	SP-SSE	1.1.1.1, 1.1.2.1, 1.1.4.1, 1.1.4.2, 1.1.4.3, 1.1.4.4
REQ24209	SIP-PBXs MUST implement TCP	MUST	SIP PBX, SP-SSE	1.2.1.1, 1.2.2.1
REQ24211	The SIP-PBX MUST support Transport Layer Security (TLS) v1.0 as described in [RFC 2246] and [RFC 3261].	MUST	SIP PBX, SP-SSE	1.1.2.1, 1.1.4.3, 1.1.4.4
REQ24213	While SIPconnect 1.1 continues to require TLS support at MUST strength, we should note that using TLS for signaling as described in Sections 15.2 and 16.2 does not require the use of the SIPS URI scheme.	MUST	SP-SSE, SIP PBX	NT
REQ24214	When receiving a certificate, SIP-PBX or SP-SSE implementations MUST support extraction of the canonical hostname from the subjectCommonName (CN) if (and only if) it is not present in the subjectAltName.	MUST	SP-SSE, SIP PBX	NT
REQ24215	SIP-PBXes MUST ignore this parameter.	MUST	SIP PBX, SP-SSE	NT
REQ24221	SIP-PBX implementations MUST comply with guidelines relating to usage of the Subject field, specified in RFC 5280 Section 4.1.2.6, and the SubjectAltName field as specified in [RFC 5280] Section 4.2.1.6.	MUST	SIP PBX, SP-SSE	NT
REQ24223	Furthermore, SIP-PBX implementations MUST be able to accept a DNS name as an identity (e.g. proxy1.example.com), instead of a SIP URI as defined in [RFC 3261] (e.g., sip:proxy.example.com).	MUST	SIP PBX, SP-SSE	1.1.2.1, 1.1.4.3
REQ24225	SIP-PBXs MUST be able to support Enterprise Public Identities in the form of a SIP URI containing a global E.164 [ITU-T E.164] number and the "user=phone" parameter.	MUST	SIP PBX, SP-SSE	1.2.1.1, 1.2.2.1
REQ24227	The global E.164 number MUST begin with a leading "+".	MUST	SP-SSE, SIP PBX	1.2.1.1, 1.2.2.1
REQ24228	MUST NOT contain a phone-context parameter	MUST	SP-SSE, SIP PBX	1.2.1.1, 1.2.2.1

REQ Tag	Requirement Text	Rqmt Category	DUT	SIP PBX TC Traced To
REQ24229	and MUST NOT include visual separators.	MUST	SP-SSE, SIP PBX	1.2.1.1, 1.2.2.1
REQ24230	The SP-SSE MUST ensure that all other header fields in the INVITE request comply with [RFC 3261].	MUST	SP-SSE	1.2.2.1
REQ24231	The SP-SSE MUST populate the Request-URI of the INVITE request in accordance with Section 15.7 for Registration mode and in accordance with Section 16.6 for Static mode.	MUST	SP-SSE	1.1.1.1, 1.1.4.1, 1.1.4.2
REQ24232	On receiving an INVITE request from the SP-SSE, the SIP-PBX MUST identify the called user based on the contents of the Request-URI.	MUST	SIP PBX	1.2.2.1
REQ24233	As such, the SIP-PBX MUST NOT rely on the contents of "To" header field for routing decisions	MUST	SIP PBX	1.2.2.1
REQ24234	but MUST use the Request-URI instead	MUST	SIP PBX	1.2.2.1
REQ24235	In cases where the SP-SSE needs to generate an anonymous URI (e.g., for a call incoming to the Service Provider Network from the PSTN for which calling number privacy is requested), the SP-SSE MUST send a URI as shown here.	MUST	SP-SSE	1.2.2.2
REQ24236	If the originating SIP entity supplied an E.164 calling number, and the caller did not request calling number privacy, then the SP-SSE MUST populate the "From" header field with a SIP URI containing the E.164 calling number, the Service Provider domain name, and the "user=phone" parameter as shown below.	MUST	SP-SSE	1.2.2.1
REQ24238	If the caller requested privacy, and the Service Provider Network does not trust the Enterprise Network, then the SP-SSE MUST remove all "P-Asserted-Identity" header fields in the INVITE request before sending the request to the SIP-PBX.	MUST	SP-SSE	1.2.2.2
REQ24239	If the caller requested privacy, and the SP-SSE is able to assert an identity, and the Service Provider Network trusts the Enterprise Network, then the SP-SSE MUST include a "P-Asserted-Identity" header field and a "Privacy" header field with value 'id' in the INVITE request, in addition to providing an anonymous "From" header field URI as specified in Section 10.1.3, before sending the request to the SIP- PBX.	MUST	SP-SSE	NT
REQ24240	If the caller did not request privacy, and the SP-SSE is able to assert an identity, then the SP-SSE MUST include a "P-Asserted-Identity" header field containing a URI identifying the calling user in the INVITE request before sending the request to the SIP-PBX.	MUST	SP-SSE	1.2.2.1
REQ24241	This means that the SIP-PBX MUST support receiving a "P-Asserted-Identity" header field containing any form of URI permissible according to [RFC 3325] and [RFC 5876].	MUST	SIP PBX	1.2.2.1, 1.2.2.2
REQ24242	As described in [RFC 3325], the SIP-PBX MUST accept up to two "P- Asserted-Identity" header fields, one in the form of a Tel URI, and one in the form of a SIP URI.	MUST	SIP PBX	1.2.2.2
REQ24243	SIP-PBX MUST prefer the SIP URI when two are present.	MUST	SIP PBX	NT

REQ Tag	Requirement Text	Rqmt Category	DUT	SIP PBX TC Traced To
REQ24244	The SIP-PBX MUST support receiving a "Privacy" header field from the SP-SSE that contains a priv- value of either 'id' or 'none', as per [RFC 3325], [RFC 5876] and [RFC 3323].	MUST	SIP PBX	1.2.2.2
REQ24245	The SIP-PBX MUST ensure that all other header fields in the INVITE request comply with [RFC 3261].	MUST	SIP PBX	1.2.1.1
REQ24246	If the SIP-PBX has an E.164 number identifying the called user (e.g., derived from a Tel URI or a dial string), the SIP-PBX MUST populate the Request-URI of the INVITE request with a SIP URI of the following form, using the domain name of the Service Provider in the host part: sip:+12128901234@sp.example.com;user=phone	MUST	SIP PBX	1.2.1.1, 1.3.3.1, 1.3.3.2
REQ24247	If the SIP-PBX has a dial string identifying the called user and is unable to convert it to a SIP URI of the "user=phone" form, the SIP-PBX MUST populate the Request-URI of the INVITE request with a SIP URI in the following form: sip: 92125551212@sp.example.com	MUST	SIP PBX	1.2.1.1, 1.3.2.1
REQ24248	As such, the SP-SSE MUST NOT rely on the "To" header field URI for routing decisions, but use the Request-URI instead.	MUST	SP-SSE	1.2.1.1
REQ24249	The SIP-PBX MUST include a "P-Asserted-Identity" header field in the INVITE request in accordance with the rules of [RFC 3325] and [RFC 5876] unless the SIP-PBX needs to withhold the identity for privacy reasons or the SIP-PBX is performing call forwarding and is unable to assert the identity of the original caller.	MUST	SIP PBX	1.2.1.1, 1.2.1.2, 1.3.1.1, 1.3.2.1
REQ24250	The SIP-PBX MUST populate the "From" header field URI with a URI that the SIP PBX wishes to be used for caller identification.	MUST	SIP PBX	1.2.1.1, 1.2.2.2
REQ24251	In cases where the Enterprise Network needs to generate an anonymous URI on behalf of a caller (as opposed to passing on a received anonymous URI), the SIP-PBX MUST send a URI of the form sip:anonymous@anonymous.invalid	MUST	SIP PBX	1.2.1.1, 1.2.1.2
REQ24252	If the SIP-PBX requires privacy for a call by suppressing delivery of caller identity to downstream entities, it MUST include a "Privacy" header field with value 'id' in the INVITE request, in addition to providing an anonymous "From" header field URI as specified in Section 10.2.4.	MUST	SIP PBX	1.2.1.2
REQ24253	If the SP-SSE provides privacy by default and the SIP-PBX requires privacy to be overridden for a call, the SIP-PBX MUST include a "Privacy" header field with value 'none' in the INVITE request.	MUST	SIP PBX	NT

REQ Tag	Requirement Text	Rqmt Category	DUT	SIP PBX TC Traced To
REQ24254	The SP-SSE MUST support receiving a "Privacy" header, from the SIP-PBX that contains a priv-value of either 'id' or 'none', as per [RFC 3325], [RFC 5876] and [RFC 3323].	MUST	SP-SSE	1.2.1.2
REQ24255	In order to forward a call, the SIP-PBX MUST send an INVITE request to the SP-SSE, populated as specified in Section 10.2, with the Request-URI identifying the forwarded-to target destination.	MUST	SIP PBX	1.3.2.1
REQ24256	An SP-SSE MUST be able to accept forwarded calls from a SIP-PBX.	MUST	SP-SSE	1.3.2.1
REQ24257	The SP-SSE MUST support the use of INVITE/re-INVITE for initiating and responding to call transfers.	MUST	SIP PBX, SP-SSE	1.3.3.1 thru 1.3.3.9
REQ24259	The SIP-PBX MUST support both sending and receiving a re-INVITE request with an SDP offer, and sending and receiving a re-INVITE request without an SDP offer.	MUST	SIP PBX, SP-SSE	1.3.3.1 thru 1.3.3.9
REQ24261	The SIP-PBX MUST have a dial plan that recognizes emergency calls.	MUST	SIP PBX	1.3.6.1
REQ24262	When a SIP-PBX routes a call recognized as an emergency call to the SP-SSE, it MUST populate the Request-URI using a dial string URI, as specified in Section 10.2.1, that contains the national emergency services number.	MUST	SIP PBX	1.3.6.1
REQ24263	The SIP PBX MUST include the identity of the caller in the "P-Asserted-Identity" header field, as described in Section 10.2.3, and in the "From" header field, as described in Section 10.2.4, except in territories where the SIP-PBX is required to include other information (such as a Location Identification Number) in one of these header fields.	MUST	SIP PBX	1.3.6.1
REQ24264	The SIP PBX MUST NOT withhold the "P-Asserted-Identity" header field for privacy reasons	MUST	SIP PBX	1.3.6.1
REQ24265	and MUST NOT anonymize the "From" header field.	MUST	SIP PBX	1.3.6.1
REQ24266	The SP-SSE MUST be able to recognize emergency calls based on the presence of the agreed emergency services number in the Request-URI.	MUST	SP-SSE	1.3.6.1

REQ Tag	Requirement Text	Rqmt Category	DUT	SIP PBX TC Traced To
REQ24267	The SP-SSE MUST NOT apply SIP session limits to emergency calls originated by the SIP-PBX.	MUST	SP-SSE	NT
REQ24268	A SP-SSE/SIP-PBX acting on behalf of a Media Endpoint that originates and/or terminates RTP traffic MUST utilize the Session Description Protocol (SDP) as described in [RFC 4566] in conjunction with the offer/answer model described in [RFC 3264] to exchange media capabilities (IP address, port number, media type, send/receive mode, codec, DTMF mode, etc).	MUST	SP-SSE, SIP PBX	1.3.3.4 thru 1.3.3.9, 1.3.4.1, 1.3.4.2, 2.1.1.1, 2.1.1.2
REQ24269	SIP-PBXs MUST be capable of receiving INVITE requests without an SDP offer and supplying an SDP offer in an appropriate response, in accordance with [RFC 3261].	MUST	SIP PBX, SP-SSE	1.3.3.3 thru 1.3.3.8, 1.3.4.1, 1.3.4.2, 2.1.1.1
REQ24271	A SP-SSE/SIP-PBX that participates in SDP offer/answer negotiation MUST be prepared to accept additional offers containing SDP with a version that has not changed	MUST	SP-SSE, SIP PBX	2.1.1.2
REQ24272	A SP-SSE/SIP-PBX that participates in SDP offer/answer negotiation MUST generate a valid answer (which could be the same SDP sent previously, or could be different).	MUST	SP-SSE, SIP PBX	2.1.1.2
REQ24273	A SP-SSE/SIP-PBX that sends additional SDP offers with the same version MUST be prepared to accept answers with SDP which may be the same as the previously received SDP, or may be different.	MUST	SP-SSE, SIP PBX	2.1.1.1
REQ24274	A SP-SSE/SIP-PBX that sends SDP with a change compared to the previously sent SDP MUST increase the version number in the o-line, in accordance with [RFC 4566].	MUST	SP-SSE, SIP PBX	NT
REQ24275	SIP-PBX implementations sending changes to negotiated media capabilities via SIP reINVITE MUST support [RFC 3261], Section 14 "Modifying an Existing Session".	MUST	SIP PBX	1.3.3.1, 1.3.3.2, 1.3.3.7
REQ24278	A Media Endpoint MUST transport and receive voice samples using the real-time transport protocol (RTP) as described in [RFC 3550].	MUST	SP-SSE, SIP PBX	1.3.5.1, 2.1.1.1, 2.1.1.2
REQ24279	Any Media Endpoint that originates and/or terminates RTP traffic over UDP MUST use the same UDP port for sending and receiving session media (i.e. symmetric RTP).	MUST	SP-SSE, SIP PBX	2.1.1.1, 2.1.1.2
REQ24280	Any Media Endpoint that originates and/or terminates RTP traffic MUST be capable of processing RTP packets with a different packetization rate than the rate used for sending.	MUST	SP-SSE, SIP PBX	2.1.1.1, 2.1.1.2
REQ24281	Any Media Endpoint that originates and/or terminates voice traffic MUST support the [ITU-T G.711] µ- Law and A-Law PCM codecs with a packetization rate of 20 ms.	MUST	SP-SSE, SIP PBX	2.1.1.1, 2.1.1.2

REQ Tag	Requirement Text	Rqmt Category	DUT	SIP PBX TC Traced To
REQ24283	In the absence of a specific indication that receiving G.711 discontinuously using the Comfort Noise (CN) payload type defined in [RFC 3389] is supported, the SIP-PBX MUST assume that the far end Media Endpoint does not support receiving G.711 discontinuously.	MUST	SIP PBX, SP-SSE	2.1.1.1, 2.1.1.2
REQ24285	In order to indicate in SDP that receiving G.711 discontinuously is supported by the local Media Endpoint, the SIP-PBX/SP-SSE MUST include payload type 13 in the "m=audio" line as described in [RFC 3389].	MUST	SP-SSE, SIP PBX	2.1.1.2
REQ24286	In the absence of a specific indication that receiving G.729 discontinuously (i.e., [ITU-T G.729] Annex B) is not supported, the SP-SSE/SIP-PBX MUST assume that the far end Media Endpoint supports receiving G729 discontinuously.	MUST	SP-SSE, SIP PBX	NT
REQ24287	In order to indicate in SDP that receiving G,729 discontinuously is not supported by the local Media Endpoint, the "a=fmtp:18 annexb=no" attribute MUST be included.	MUST	SP-SSE, SIP PBX	NT
REQ24288	A SP-SSE/SIP-PBX MUST advertize support for telephone-events [RFC 4733] in its SDP on behalf of any Media Endpoint that supports receiving DTMF digits using [RFC 4733] procedures.	MUST	SP-SSE, SIP PBX	NT
REQ24289	Any Media Endpoint that supports receiving DTMF MUST support [RFC 4733] procedures.	MUST	SP-SSE, SIP PBX	1.2.1.1, 1.2.2.1, 1.3.3.1
REQ24290	Any Media Endpoint that supports sending DTMF MUST use the [RFC 4733] procedures to transmit DTMF tones using the RTP telephone-event payload format, provided that the other side has advertized support for receiving [RFC 4733] in the offer/answer exchange.	MUST	SP-SSE, SIP PBX	NT
REQ24291	For any local Media Endpoint that supports receiving telephone-event packets, the SIP-PBX MUST include the supported events in an "a=fmtp:" line as is described as mandatory in [RFC 4733].	MUST	SIP PBX	NT
REQ24293	To provide backward compatibility with [RFC 2833] implementations, any Media Endpoint MUST be prepared to receive telephone-event packets for all events in the range 0-15	MUST	SP-SSE, SIP PBX	2.1.1.2
REQ24294	and a SIP-PBX MUST be prepared to accept SDP with a payload type mapped to telephone-event, even if it does not have an associated "a=fmtp" line.	MUST	SIP PBX, SP-SSE	NT
REQ24296	Any Media Endpoint that can introduce echo MUST provide [ITU-T G.168]-compliant echo cancellation.	MUST	SP-SSE, SIP PBX	NT
REQ24297	Media Endpoints that support fax (e.g., a SIP media server that can originate/terminate faxes) and Media Endpoints that can act as a T.30 gateway (e.g., a Media Endpoint that supports an RJ11 analog telephone interface) MUST support the [ITU-T T.38] Recommendation.	MUST	SP-SSE, SIP PBX	NT
REQ24298	Media Endpoints that support [ITU-T T.38] MUST support User Datagram Protocol Transport Layer (UDPTL) transport.	MUST	SP-SSE, SIP PBX	NT

REQ Tag	Requirement Text	Rqmt Category	DUT	SIP PBX TC Traced To
REQ24299	When acting as a call originator, the SIP-PBX, upon receipt of a 180 provisional response message (whether reliable [RFC 3262] or unreliable) MUST instruct the Media Endpoint to play local ringback tone to the user.	MUST	SIP PBX	1.3.3.2, 1.3.3.3, 1.3.5.1
REQ24300	Upon receipt of SDP in any 18x provisional response message (reliable [RFC 3262] or unreliable), the SIP-PBX MUST forward this information to the Media Endpoint.	MUST	SIP PBX	1.3.3.2, 1.3.3.3, 1.3.5.1
REQ24301	When acting as a call terminator and expecting the originating end to provide local ringback tone, the Media Endpoint MUST NOT send RTP packets to the originator if a 180 provisional response message was sent.	MUST	SP-SSE, SIP PBX	1.3.5.1
REQ24302	A Media Endpoint, on receipt of an instruction to play local ringback tone, MUST do so until it receives valid RTP packets or is instructed by the SIP-PBX that the call has been answered.	MUST	SP-SSE, SIP PBX	1.3.3.2, 1.3.3.3, 1.3.3.5, 1.3.3.6, 1.3.3.7, 1.3.3.8, 1.3.5.1
REQ24303	On receipt of valid RTP packets, a Media Endpoint MUST disable any local ringback tone and play the received media.	MUST	SP-SSE, SIP PBX	1.3.3.2, 1.3.3.3, 1.3.3.5, 1.3.3.6, 1.3.3.7, 1.3.3.8, 1.3.5.1
REQ24305	The hold initiator MUST set the SDP directionality attribute to "a=sendonly".	MUST	SP-SSE, SIP PBX	1.3.4.1, 1.3.4.2
REQ24306	If the hold initiator does not provide MOH, it MUST set the SDP directionality attribute to "a=inactive" or "a=sendonly".	MUST	SP-SSE, SIP PBX	1.3.4.1, 1.3.4.2
REQ24307	A SP-SSE/SIP-PBX MUST support the ability to receive SDP session descriptions that have the 'c=' field set to all zeros (0.0.0.0), when the addrtype field is IPV4. Note that this is for support of non- compliant remote SIP signaling entities that use this deprecated syntax from RFC 2543, rather than the "a=sendonly" or "a=inactive" syntax specified in [RFC 3264].	MUST	SP-SSE, SIP PBX	NT
REQ24308	The SIP-PBX MUST be capable of provisioning any format of SIP-URI as the Registration AOR, in order to accommodate SP-SSE requirements (i.e., the Registration AOR is not subject to the same constraints as Enterprise Public Identities and could, for example, be an "email-style" SIP URI).	MUST	SIP PBX	1.1.1.1, 1.1.2.1
REQ24309	The SIP-PBX MUST provide its SIP signaling address(es) and port(s) to the SP-SSE using the SIP registration procedure described in Section 15.4.	MUST	SIP PBX	1.1.1.1, 1.1.2.1
REQ24310	The SIP-PBX MUST be capable of obtaining information about the SP-SSE, using the procedure described in Section 16.1.1.2.	MUST	SIP PBX	1.1.4.1 thru 1.1.4.4
REQ24311	The SP-SSE MUST make its SIP signaling address(es) and port(s) available to the Enterprise Network as specified in Section 16.1.2.1.	MUST	SP-SSE	1.1.4.1, 1.1.4.3
REQ24312	The SP-SSE MUST obtain the SIP-PBX signaling address/port using SIP registration, as described in Section 15.4.	MUST	SP-SSE	1.1.1.1

REQ Tag	Requirement Text	Rqmt Category	DUT	SIP PBX TC Traced To
REQ24313	Both SIP-PBX and SP-SSE MUST support the TLS Server Authentication model, whereby the SP-SSE (acting as TLS server), provides its certificate to the SIP-PBX (acting as TLS client) as part of the TLS establishment phase.	MUST	SP-SSE, SIP PBX	1.1.2.1, 1.1.4.3, 1.1.4.4
REQ24314	The SIP-PBX MUST be capable of initiating the establishment of a TLS session.	MUST	SIP PBX	1.1.2.1, 1.1.4.3, 1.1.4.4
REQ24315	The SIP-PBX MUST be capable of being provisioned with either a certification authority certificate or with a copy of the certificate the SP-SSE plans to use (or a fingerprint thereof).	MUST	SIP PBX	1.1.2.1, 1.1.4.3, 1.1.4.4
REQ24316	The SIP-PBX MUST validate the certificate received during TLS establishment using the path validation procedure described in [RFC 5280].	MUST	SIP PBX	NT
REQ24318	The SIP-PBX MUST be capable of being configured to require use of TLS to initiate a session.	MUST	SIP PBX	1.1.2.1, 1.1.4.3, 1.1.4.4
REQ24319	a SIP-PBX MUST NOT initiate sessions with other transports (UDP or TCP), even if the SP-SSE indicates that these are available via DNS NAPTR and/or SRV resource records.	MUST	SIP PBX	1.1.2.1
REQ24320	The SIP-PBX MUST initiate the establishment of the TLS session	MUST	SIP PBX	1.1.2.1, 1.1.4.3, 1.1.4.4
REQ24321	The SIP-PBX MUST NOT utilize other transports (UDP or TCP), even if the SP-SSE indicates that these are available via configuration of DNS NAPTR and/or SRV resource records.	MUST	SIP PBX	1.1.2.1
REQ24323	The SP-SSE MUST be configured with a verifiable digital certificate to secure a TLS session.	MUST	SP-SSE	NT
REQ24324	The SP-SSE MUST use certificates that are signed by a third party certification authority unless the certificates can be validated through some other means, such as being pre-installed at the SIP- PBX or signed by the SP-SSE itself.	MUST	SP-SSE	NT
REQ24325	When using TLS (as a result of being configured to require use of TLS, or as a result of discovering the availability of TLS from DNS), the SIP-PBX MUST establish a TLS connection (if not already established) prior to registration	MUST	SIP PBX	1.1.2.1
REQ24326	When using TLS (as a result of being configured to require use of TLS, or as a result of discovering the availability of TLS from DNS), the SIP-PBX MUST use that connection to deliver the REGISTER request and all subsequent SIP messages to the SP-SSE.	MUST	SIP PBX	1.1.2.1
REQ24327	The SP-SSE MUST authenticate the SIP-PBX using SIP digest authentication, as specified in Section 15.4, and reject the REGISTER request if authentication fails.	MUST	SP-SSE	1.1.2.2, 1.1.2.3

REQ Tag	Requirement Text	Rqmt Category	DUT	SIP PBX TC Traced To
REQ24328	Following successful registration, the SP-SSE MUST use a TLS connection that is authenticated as a connection to this SIP-PBX to deliver all SIP requests to the SIP-PBX.	MUST	SP-SSE	NT
REQ24329	The SIP-PBX MUST avoid closing down the TLS connection, other than in exceptional circumstances (e.g., for maintenance).	MUST	SIP PBX, SP-SSE	NT
REQ24331	When a SIP-PBX registers, the SP-SSE MUST ignore the transport=tls parameter in the "Contact" header field URI.	MUST	SP-SSE	1.1.2.1
REQ24332	Any IP addresses contained within the header fields and message body parts (e.g. SDP) of SIP messages exchanged between the Service Provider and Enterprise Networks MUST be publicly routable addresses, unless the Service Provider Network is providing an implicit NAT traversal function or the two are using a private VPN-style address space.	MUST	SP-SSE, SIP PBX	1.2.1.1, 1.2.2.1
REQ24333	The SIP-PBX MUST support multiple AOR registration in accordance with [RFC 6140], using the provisioned Registration AOR and the set of provisioned Enterprise Public Identities, even if there is only a single provisioned Enterprise Public Identity.	MUST	SIP PBX, SP-SSE	1.1.1.1, 1.1.2.1
REQ24335	In the REGISTER request, the SIP-PBX MUST include a Contact URI in accordance with [RFC 6140] using a suitable domain part, e.g., the SIP-PBX's IP address.	MUST	SIP PBX	1.1.1.1
REQ24336	The SIP-PBX MUST insert the Registration AOR in the "From"and "To" header fields of the REGISTER request.	MUST	SIP PBX	1.1.1.1
REQ24337	The SIP-PBX MUST support the authentication mechanisms outlined in Section 15.6 for digest authentication for the REGISTER requests, using a user name and password agreed to by both parties.	MUST	SIP PBX, SP-SSE	1.1.2.2, 1.1.2.3
REQ24339	If the SIP-PBX fails to receive any response to a REGISTER request in Timer_F time (typically 32 seconds) or encounters a transport error when sending a REGISTER request, the SIP-PBX MUST consider the SP-SSE unreachable and try to register with an alternate SP-SSE address if it has one.	MUST	SIP PBX	1.1.3.1
REQ24340	If the SIP-PBX has an established connection-based transport (e.g., TCP) to the SP-SSE, and Timer_F expires or a transport error is encountered as above, it MUST try to re-establish a connection to the same SP-SSE before considering it unreachable, by resetting Timer_F and sending a new REGISTER request.	MUST	SIP PBX	NT

REQ Tag	Requirement Text	Rqmt Category	DUT	SIP PBX TC Traced To
REQ24342	If no SP-SSE is reachable, or no alternates are available, the SIP-PBX MUST delay reattempting Registration for 30 seconds, and increasing this delay value by doubling it for each successive delivery failure until delivery succeeds, up to a maximum value of 960 seconds.	MUST	SIP PBX	1.1.3.1
REQ24341	The SIP-PBX MUST NOT attempt to re-establish the connection to the same SP-SSE more than once before considering the SP-SSE unreachable	MUST	SIP PBX	1.1.3.1
REQ24343	The SP-SSE MUST NOT issue a 302 Moved Temporarily redirect response to a REGISTER request, to get the SIP-PBX to Register with an alternate SP-SSE address identified by the Contact URI in the response.	MUST	SP-SSE	NT
REQ24344	The SP-SSE MUST issue a 404 Not Found response to a REGISTER request, if the Registration AOR of the SIP-PBX is not found in its database.	MUST	SP-SSE	1.1.3.2
REQ24345	An SIP-PBX receiving such a response to a REGISTER request MUST consider the Registration attempt to have failed, and notify the SIP-PBX administrator if possible through some means.	MUST	SIP PBX	1.1.3.2
REQ24347	If the digest challenge response of the SIP-PBX in its REGISTER request is stale or invalid, the SP-SSE MUST issue one of the following response codes: oa 401 Unauthorized, oa 407 Proxy Authentication Required or oa 403 Forbidden unless the SP-SSE is configured to silently discard these requests based on policy.	MUST	SP-SSE	1.1.2.3
REQ24348	If a SIP-PBX receives more than three responses of 401, 407 or 403 in aggregate, without a different response other than one of those in between, then the SIP-PBX MUST consider the Registration attempt to have failed, and notify the SIP-PBX administrator if possible through some means.	MUST	SIP PBX	1.1.2.3
REQ24351	An SIP-PBX receiving such a response to a REGISTER request MUST act exactly as if delivery to the SP-SSE had failed per Section 15.4.1.1,	MUST	SIP PBX	1.1.3.2
REQ24352	An SIP-PBX receiving such a response to a REGISTER request MUST follow the backoff procedures defined previously in Section 15.4.1.1.	MUST	SIP PBX	1.1.3.2
REQ24353	An overloaded SP-SSE MUST generate a 503 Service Unavailable or 500 Internal Error response code to a REGISTER request, unless it is silently discarding requests due to overload	MUST	SP-SSE	1.1.3.2
REQ24356	A SIP-PBX receiving such a response MUST support the "Retry-After" header field	MUST	SIP PBX	1.1.3.2
REQ24422	A SIP-PBX receiving such a response MUST honor the value as follows: if the value is 32 seconds or less	MUST	SIP PBX	1.1.3.2
REQ24423	A SIP-PBX receiving such a response MUST wait the requested time and retry the request to the same SP-SSE	MUST	SIP PBX	1.1.3.2

REQ Tag	Requirement Text	Rqmt Category	DUT	SIP PBX TC Traced To
REQ24423	A SIP-PBX receiving such a response MUST wait the requested time and retry the request to the same SP-SSE	MUST	SIP PBX	1.1.3.2
REQ24424	A SIP-PBX receiving such a response if the value is larger, it MUST remember the value for that SP-SSE address instance, and try any alternate SP-SSE addresses it can.	MUST	SIP PBX	1.1.3.2
REQ24361	Otherwise, it MUST wait to reattempt registration to the original SP-SSE for the "Retry-After" interval.	MUST	SIP PBX	1.1.3.2
REQ24362	If a SIP-PBX encounters a transport error when attempting to contact the SP-SSE, encounters Timer F expiry (non-INVITE requests) or Timer B expiry (INVITE requests), or receives a 403 response for any non-REGISTER request, the SIP-PBX MUST oconsider the request attempt to have failed, oassume that the SIP-PBX's registration is no longer active at the SP-SSE, and onotify the SIP-PBX administrator if possible through some means.	MUST	SIP PBX	1.1.3.1
REQ24364	Therefore the SIP-PBX MUST honor the REGISTER expiry time provided by the SP-SSE,	MUST	SIP PBX	NT
REQ24366	If failure is detected a SIP-PBX MUST attempt reconnection,	MUST	SIP PBX	NT
REQ24419	and if that fails MUST try an alternative SP-SSE if available, in accordance with Section 15.4.1.1.	MUST	SIP PBX	NT
REQ24368	The SIP-PBX MUST support the digest authentication scheme as described in Section 22.4 of [RFC 3261].	MUST	SIP PBX, SP-SSE	1.1.2.2, 1.1.2.3
REQ24370	The SIP-PBX MUST support receiving 401 Unauthorized and 407 Proxy Authentication Required from the SP-SSE.	MUST	SIP PBX	1.1.2.2, 1.1.2.3
REQ24371	When so challenged by the SP-SSE, the SIP-PBX MUST respond with authentication credentials that are valid within the Service Provider's realm (i.e. based on the username and password supplied by the Service Provider).	MUST	SIP PBX	1.1.2.2
REQ24372	Authentication of the Service Provider by the Enterprise is supported using TLS server authentication. If TLS is required (based on local configuration data), then the SIP-PBX MUST perform TLS server authentication as described in Section 15.2.	MUST	SIP PBX	1.1.2.1, 1.1.4.3, 1.1.4.4

REQ Tag	Requirement Text	Rqmt Category	DUT	SIP PBX TC Traced To
REQ24373	The SP-SSE MUST route inbound out-of-dialog requests targeted at Enterprise Public Identities to the registered SIP-PBX in accordance with [RFC 6140].	MUST	SP-SSE	1.1.1.1
REQ24374	The SIP-PBX MUST provide its SIP signaling address and port to the SP-SSE using one of the following mechanisms:	MUST	SIP PBX	1.1.4.1 thru 1.1.4.4
REQ24376	This DNS server MUST provide a DNS interface that supports SRV resource records [RFC 2782].	MUST	DNS Server	NA
REQ24377	Except when a TLS connection already exists, the SIP-PBX MUST use one of the following mechanisms to obtain the address and port of the SP-SSE and the transport protocol (UDP, TCP or TLS) to be used	MUST	SIP PBX	1.1.4.1, 1.1.4.2
REQ24379	When a TLS connection already exists, the SIP-PBX MUST reuse that TLS connection for all SIP messages.	MUST	SIP PBX	1.1.4.3, 1.1.4.4
REQ24404	The SP-SSE MUST be reachable through a publicly-accessible DNS server	MUST	SP-SSE	1.1.4.1, 1.1.4.3
REQ24405	The DNS server SHOULD provide a DNS interface that supports NAPTR resource records	MUST	DNS Server	NA
REQ24420	and MUST provide a DNS interface that supports SRV resource records.	MUST	DNS Server	NA
REQ24407	The SP-SSE MUST support both of the following mechanisms to obtain the address and port of the SIP-PBX and the transport protocol (UDP, TCP or TLS) to be used	MUST	SP-SSE	1.1.4.1 thru 1.1.4.4
REQ24421	and, except when a TLS connection already exists, MUST use one of these mechanisms:	MUST	SP-SSE	1.1.4.1 thru 1.1.4.4
REQ24409	When a TLS connection already exists, the SP-SSE MUST reuse that TLS connection for all SIP messages	MUST	SP-SSE	1.1.4.3, 1.1.4.4
REQ24380	Both SIP-PBX and SP-SSE MUST support the TLS Mutual Authentication model, whereby both the SP-SSE and the SIP-PBX provide their respective certificate as part of the TLS establishment phase.	MUST	SP-SSE, SIP PBX	1.1.4.3, 1.1.4.4
REQ24382	Both SIP-PBX and SP-SSE MUST be able to initiate the establishment of a TLS session.	MUST	SP-SSE, SIP PBX	1.1.4.3, 1.1.4.4
REQ24383	Both SIP-PBX and SP-SSE MUST be capable of being provisioned with either a certification authority certificate or with a copy of the certificate the peer SIP endpoint plans to use (or a fingerprint thereof).	MUST	SP-SSE, SIP PBX	1.1.4.3, 1.1.4.4

REQ Tag	Requirement Text	Rqmt Category	DUT	SIP PBX TC Traced To
REQ24384	Both SIP-PBX and SP-SSE MUST validate the certificate received during TLS establishment using the path validation procedure described in [RFC 5280].	MUST	SP-SSE, SIP PBX	1.1.4.3, 1.1.4.4
REQ24385	Both SIP-PBX and SP-SSE MUST be capable of being configured to require use of TLS to initiate a session to a particular peer.	MUST	SP-SSE, SIP PBX	1.1.4.3, 1.1.4.4
REQ24386	When TLS is configured to be required for session initiation to a peer, a SIP-PBX or SP-SSE MUST NOT initiate sessions with other transports (UDP or TCP), even if the peer indicates that these are available via configuration of DNS NAPTR and/or SRV resource records.	MUST	SIP PBX, SP-SSE	1.1.2.1
REQ24387	Both SIP-PBX and SP-SSE MUST be capable of being configured to require use of TLS to accept sessions initiated to it by a peer.	MUST	SP-SSE, SIP PBX	1.1.4.3, 1.1.4.4
REQ24388	When TLS is configured to be required to accept sessions initiated from all peers, a SIP-PBX MUST NOT advertise support for other transports (UDP or TCP), via configuration of DNS NAPTR and/or SRV resource records.	MUST	SIP PBX	NT
REQ24390	The SIP-PBX MUST be configured with a verifiable digital certificate to secure a TLS session.	MUST	SIP PBX	1.1.4.3, 1.1.4.4
REQ24391	The SIP-PBX MUST be configured with a certificate signed by a third party certification authority unless the configured certificate can be validated through some other means, such as being pre-installed on the SP-SSE or signed by the SIP-PBX itself.	MUST	SIP PBX	1.1.4.3, 1.1.4.4
REQ24393	The SP-SSE MUST be configured with a verifiable digital certificate to secure a TLS session.	MUST	SP-SSE	NT
REQ24394	The SP-SSE MUST be configured with a certificate signed by a third party certification authority unless the configured certificate can be validated through some other means, such as being pre- installed on the SIP-PBX or signed by the SP-SSE itself.	MUST	SP-SSE	NT
REQ24396	SIP-PBXes that support one of these mechanisms MUST also support a mechanism that allows the keep- alive interval to be configured.	MUST	SIP PBX	NT
REQ24397	If TLS is required (based on local configuration data), then the SP-SSE and SIP-PBX MUST perform TLS mutual authentication as described in Section 16.2.	MUST	SP-SSE, SIP PBX	1.1.4.3, 1.1.4.4
REQ24398	The SP-SSE MUST populate the Request-URI of the INVITE request with the Enterprise Public Identity of the called Enterprise user in the valid form defined in Section 9, or with a Contact URI provided by the SIP PBX in a previous request or response.	MUST	SP-SSE	1.2.2.1, 1.3.2.1

REQ Tag	Requirement Text	Rqmt Category	DUT	SIP PBX TC Traced To
Note: The following requirements are per RFC 6140. Requirements numbers shown are CableLabs tag numbers, not a reference to an RF				
RFC4607	First, it MUST contain an option tag of "gin" in both a "Require" header field and a "Proxy-Require “header field.	MUST	SIP-PBX	1.1.1.1
RFC4608	Second, in at least one "Contact" header field, it MUST include a Contact URI that contains the URI parameter "banc"(which stands for "bulk number contact") and has no user portion(hence no "@" symbol).	MUST	SIP-PBX	1.1.1.1
RFC4609	A URI with a "bnc" parameter MUST NOT contain user portion.	MUST	SIP-PBX	1.1.1.1
RFC4610	Any SIP-PBX implementing the registration mechanism defined in this document MUST also support the path mechanism defined by RFC 3327[10], and MUST include a 'path' option tag in the "Supported" header field of the REGISTER request (which is a stronger requirement than imposed by the path mechanism itself).	MUST	SIP-PBX	1.1.1.1
RFC4611	Aside from the initial "+" symbol, this E.164-formattednumber MUST consist exclusively of digits from 0 through 9 and explicitly MUST NOT contain any visual separator symbols (e.g., "-",".”	MUST	SP-SSE	1.1.1.1
RFC4612	In particular, this means that REGISTER requests that attempt to de-register a single AOR that has been implicitly registered MUST NOT remove that AOR from the bulk registration.	MUST	SP-SSE	1.1.1.1
RFC4613	A registrar that receives a REGISTER request containing a Contact URI with both a "bnc" parameter and a user portion MUST NOT send a 200-class (Success) response.	MUST	SP-SSE	1.1.1.1
RFC4614	A registrar compliant with this document MUST support the path mechanism defined in RFC 3327 [10].	MUST	SP-SSE	1.1.1.1
RFC4615	Aside from the "bnc" parameter, all URI parameters present on the Contact URI in the REGISTER request MUST be copied to the contact value stored in the location service.	MUST	SP-SSE	1.1.1.1
RFC4616	When a SIP-PBX registers with an SSP using a Contact URI containing a "bnc" parameter, that Contact URI MUST NOT include a "user" parameter.	MUST	SIP-PBX	1.1.1.1
RFC4617	A registrar that receives a REGISTER request containing a Contact URI with both a "bnc" parameter and a "user" parameter MUSTNOT send a 200-class (success) response.	MUST	SP-SSE	1.1.1.1
RFC4618	When a SIP-PBX receives a request from an SSP, and the Request URI contains a user portion corresponding to an AOR registered using a Contact URI containing a "bnc" parameter, then the SIP-PBX MUST NOT reject the request (or otherwise cause the request to fail) due to the absence, presence, or value of a "user" parameter on the Request URI.	MUST	SIP-PBX	1.1.1.1

APPENDIX C ACKNOWLEDGEMENTS

CableLabs wishes to thank the Test Development team participants for various contributions and efforts that led to the development of this test plan. Specifically:

Name	Company
Seung Hoon Lee	CableLabs
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Special appreciation is extended to David Hancock for his role in the clarification of numerous specification questions.

John Berg and the Test Development team, CableLabs, Inc.