SIPconnect 1.1

Certification Test Plan

Version 1.1.0

SIP Forum Document Number: TWG-7

MODIFICATION RECORD

Version 1.1.0	August 2017	• Required Tests Table Bug Fixes • Test Procedure and Results Typos • Minor Formatting Fixes
Version 1.0	June 16, 2015	• Certification Program Launch • Added possible problem for 2.1.4b
Version 0.3	May 16, 2014	• All tests written
Version 0.2	April 22, 2014	• Section 1 and 2 Tests Added
Version 0.1	February 12, 2014	• Initial version

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INTRODUCTION

This Conformance Test Plan (CTP) defines a set of test procedures that can be used to verify Conformance between an Enterprise SIP-PBX and a Service Provider network SIP Signaling Entity (SP-SSE). The test cases described in this document cover industry-accepted requirements for the SIP Connect 1.1 interface as well as 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 CTP 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, meaning that the test procedures herein are subject to change based on outcomes from testing, either in the SIP Forum community, CableLabs, UNH-IOL, or other third party test facilities.

Purpose and Scope

The intended use of this CTP is for testing conformanceof a vendor productto the SIPconnect 1.1 Technical Recommendation. The scope of this document includes test cases for the "MUST" and "shall" requirements of the SIPconnect 1.1 Technical Recommendation and other applicable SIP RFCs.

The tests are divided into the following test groups:

- Registration Mode
- Static Mode
- Basic voice calls
- Extended call features
- Media and Session Interactions

DEFINITIONS

Directed Inward Dial (DID)	Calls to Enterprise Public Identities are routed by the SP-SSE to the SIP-PBX and are usually routed by the SIP-PBX directly to a specific user station – bypassing the attendant or operator. This is commonly referred to as "Directed Inward Dial" (DID) service.
Direct Outward Dial (DOD)	Calls from Enterprise Public Identities are routed by the SIP-PBX to the SP-SSE for delivery to the PSTN.
Service Provider SIP-Signaling Entity (SP-SSE)	The Service Provider's point of SIP signaling interconnection with the Enterprise.
SIP-PBX	The Enterprise's point of SIP signaling interconnection with the Service Provider.
SIP Endpoint	A term used in this specification to refer to both SP-SSEs and SIP-PBXes.
Enterprise Public Identity	An Address of Record (AOR) represented as a SIP URI, used to identify a user or group of users served by the SIP-PBX.
Registration AOR	An AOR represented as a SIP URI, used solely to identify the SIP-PBX during registration.
Media Endpoint	Any entity that terminates an RTP/RTCP stream.
Back-to-Back User Agent	A logical entity that receives a request and processes it as a user agent server (UAS). In order to determine how the request should be answered, it acts as a user agent client (UAC) and generates a request to another SIP user agent server (UAS).
DUT	Device Under Test

TEST ORGANIZATION

This document organizes tests by Section based on related test methodology or goals. Each group begins with a brief set of comments pertaining to all tests within that group. This is followed by a series of description blocks; each block describes a single test. The format of the description block is as follows:

Test Label:	The test label and title comprise the first line of the test block. The test label is the test group and test number separated by periods.
Objective:	The Purpose is a short statement describing what the test attempts to achieve. It is usually phrased as a simple assertion of the feature or capability to be tested.
Requirements Tested:	The Requirements Tested section lists cross-references to Appendix A that restate the requirements from the specifications and documentation that might be helpful in understanding and evaluating the test and results.
Discussion:	This section is a general discussion of the test and relevant portion of the specification, including any assumptions made in the design or implementation of the test as well as known limitations.
Test Setup:	The Test Setup section describes the configuration of all devices prior to the start of the test. Different parts of the procedure may involve configuration steps that deviate from what is given in the test setup. If a value is not provided for a protocol parameter, then the protocol's default is used for that parameter.
Procedure:	This section of the test description contains the step-by-step instructions for carrying out the test. These steps include such things as enabling interfaces, unplugging devices from the network, or sending packets from a test station. The test procedure also cues the tester to make observations, which are interpreted in accordance with the expected results given for that test part. Corresponding to some steps are the observable results that can be examined by the tester to verify that the DUT is operating properly. The determination of a pass or fail for each test is usually based on how the DUT's behavior compares to the results described in this section.
Possible Problems:	This section contains a description of known issues with the test procedure, which may affect test results in certain situations.
Modification Record:	This section contains a list of modification to the test case over time due to feedback received from testing and operational communities.

REQUIRED TESTS

Test	Applicable Network Element	Required for Certification
SC-IT.Conf.1.1.1: Registration Setup	SIP-PBX	**REQUIRED[1]**
SC-IT.Conf.1.1.2: Registration Failure	SIP-PBX	**REQUIRED[1]**
SC-IT.Conf.1.1.3: Maintaining Registration	SIP-PBX	**REQUIRED[1]**
SC-IT.Conf.1.1.4: Authentication	SIP-PBX	**REQUIRED[1]**
SC-IT.Conf.1.1.5: TLS Server Mode	SIP-PBX	OPTIONAL
SC-IT.Conf.1.1.6: TLS Certificate Validation	SIP-PBX	OPTIONAL
SC-IT.Conf.1.2.1: SP-SSE Address Acquisition by SIP-PBX	SIP-PBX	**REQUIRED[2]**
SC-IT.Conf.1.2.2: Static Mode Failure Detection	SIP-PBX	OPTIONAL
SC-IT.Conf.1.2.3: TLS Authentication	SIP-PBX	OPTIONAL
SC-IT.Conf.1.3.1: Verification of INVITE Message Parameters When Originating a DOD Call	SIP-PBX	**REQUIRED**
SC-IT.Conf.1.3.2: INVITE Message Processing When Terminating a DID Call	SIP-PBX	**REQUIRED**
SC-IT.Conf.1.3.3: SIP-PBX Support for Privacy when processing INVITE for Terminating DID Call	SIP-PBX	**REQUIRED**
SC-IT.Conf.1.3.4: SIP-PBX Support for Privacy when generating INVITE message for Originating DOD Call	SIP-PBX	**REQUIRED**
SC-IT.Conf.1.4.1: Call Forwarding	SIP-PBX	**REQUIRED**
SC-IT.Conf.1.4.2: Blind Call Transfer - Transferor in SIP-PBX	SIP-PBX	**REQUIRED**
SC-IT.Conf.1.4.3: Blind Call Transfer - Transferee in SIP-PBX	SIP-PBX	**REQUIRED**
SC-IT.Conf.1.4.4: Blind Call Transfer – Transferee and Transfer-To in SIP-PBX	SIP-PBX	**REQUIRED**
SC-IT.Conf.1.4.5: Attended Call Transfer – Transferor in SIP-PBX	SIP-PBX	**REQUIRED**
SC-IT.Conf.1.4.6: Blind Call Transfer – Transferor and Transferee in SIP-PBX	SIP-PBX	**REQUIRED**
SC-IT.Conf.1.4.7: Blind Call Transfer – Transferor and Transfer-to in SIP-PBX	SIP-PBX	**REQUIRED**
SC-IT.Conf.1.4.8: Blind Call Transfer -- Transferee and Transfer-to in SIP-PBX	SIP-PBX	**REQUIRED**
SC-IT.Conf.1.4.9: Blind Call Transfer with Early Media – Transferee and Transfer-to in SIP-PBX	SIP-PBX	**REQUIRED**
SC-IT.Conf.1.4.10: Emergency Call Dial Plan	SIP-PBX	**REQUIRED**
SC-IT.Conf.2.1.1: Registration Setup	SP-SSE	**REQUIRED[1]**
SC-IT.Conf.2.1.2: Registration Failure	SP-SSE	**REQUIRED[1]**
SC-IT.Conf.2.1.3: Inbound Requests	SP-SSE	**REQUIRED[1]**
SC-IT.Conf.2.1.4: Authentication	SP-SSE	**REQUIRED[1]**
SC-IT.Conf.2.1.5: TLS Server Mode	SP-SSE	OPTIONAL

[1] Required Only if Static Mode is Unsupported

[2] Required Only if Registration Mode is Unsupported

SC-IT.Conf.2.2.1: SIP-PBX Address Acquisition by SP-SSE	SP-SSE	**REQUIRED**[2]
SC-IT.Conf.2.2.2: TLS Authentication	SP-SSE	OPTIONAL
SC-IT.Conf.2.3.1: Verification of INVITE Message Parameters When Terminating a DID Call	SP-SSE	**REQUIRED**
SC-IT.Conf.2.3.2: INVITE Processing for Originating a DOD Call	SP-SSE	**REQUIRED**
SC-IT.Conf.2.3.3: Verification of INVITE Parameters for Terminating DID Anonymous Calls	SP-SSE	**REQUIRED**
SC-IT.Conf.2.3.4: Verification of INVITE Parameters for Terminating DID Unknown/Public Calls	SP-SSE	OPTIONAL
SC-IT.Conf.2.4.1: SP-SSE Support for Privacy when generating an INVITE message to Terminate a DID Call	SP-SSE	**REQUIRED**
SC-IT.Conf.2.4.2: SP-SSE Support for Privacy when processing an INVITE message to Originate a DOD Call	SP-SSE	**REQUIRED**
SC-IT.Conf.2.4.3: SP-SSE Call Forwarding	SP-SSE	**REQUIRED**
SC-IT.Conf.2.4.4: Blind Call Transfer – Transferor in SIP_PBX	SP-SSE	**REQUIRED**
SC-IT.Conf.2.4.5: Blind Call Transfer - Transferee in SIP-PBX	SP-SSE	**REQUIRED**
SC-IT.Conf.2.4.6: Blind Call Transfer – Transferee and Transfer-To in SIP-PBX	SP-SSE	**REQUIRED**
SC-IT.Conf.2.4.7: Attended Call Transfer – Transferor in SIP-PBX	SP-SSE	**REQUIRED**
SC-IT.Conf.2.4.8: Blind Call Transfer – Transferor and Transferee in SIP-PBX	SP-SSE	**REQUIRED**
SC-IT.Conf.2.4.9: Blind Call Transfer – Transferor and Transfer-to in SIP-PBX	SP-SSE	**REQUIRED**
SC-IT.Conf.2.4.10: Blind Call Transfer – Transferee and Transfer-to in SIP-PBX	SP-SSE	**REQUIRED**
SC-IT.Conf.2.4.11: Blind Call Transfer with Early Media – Transferee and Transfer-to in SIP-PBX	SP-SSE	**RQUIRED**
SC-IT.Conf.2.4.12: Emergency Services	SP-SSE	**REQUIRED**
SC-IT.Conf.3.1.1: Basic Signaling and SDP Exchange	BOTH	**REQUIRED**
SC-IT.Conf.3.1.2: SDP Version Number	BOTH	**REQUIRED**
SC-IT.Conf.3.2.1: Session Hold	BOTH	**REQUIRED**

REFERENCE ARCHITECTURE

The architecture of the test environment used for SIP Connect 1.1 Conformance testing is shown in Figure 1.

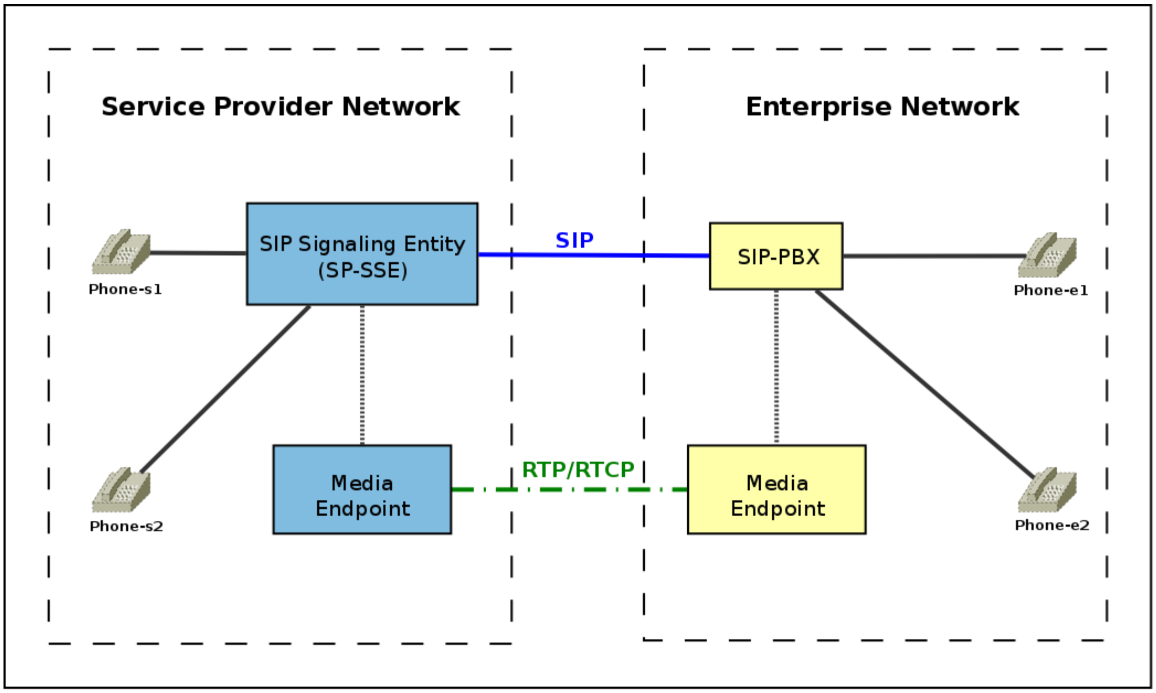

Figure 1: Test Environment

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.

For any given test, the device under test represents either the SP-SSE component, or the SIP-PBX component. The remaining components of the topology are meant to be emulated by testing software.

COMMON TEST SETUP 1

- The SIP-PBX and SP-SSE are configured to use TCP transport.
- The SIP-PBX is configured to support Registration mode.
- The SIP-PBX is configured to register to a sub-domain (sp.lab.com) provided by Service provider (sp.lab.com) from one of its autonomous systems.
- The SIP-PBX is configured with the Service Provider's domain (i.e. "sp.lab.com"), and the range of E.164 numbers.
- The SIP-PBX is SIPconnect 1.1 compliant and configured with a Registration AOR (i.e. "sip:xyz@sp.lab.com").
- The SP-SSE is configured with a table that associates the Registration AOR and Enterprise Public Identities assigned to the SIP-PBX.
- The SP-SSE is capable of handling RFC6140 registration based on Annex A of SIPconnect1.1.

COMMON TEST SETUP 2

- The SIP-PBX and SP-SSE are configured to use TCP transport.
- The 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.lab.com).
- Configure the DNS server to provide the realm (e.g. sp.lab.com), IP address, port and transport protocol (TCP) 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.

COMMON TEST SETUP 3

- The SIP-PBX and SP-SSE are configured to use TCP transport.
- The 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.lab.com).
- Configure the DNS server to provide the realm (e.g. sp.lab.com), IP address, port and transport protocol (TCP) 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.
- The SIP-PBX and SP-SSE are configured to have T.38 fax capabilities enabled.

TEST SECTION 1: SIP-PBX

Scope

Tests in this section cover SIPconnect 1.1 Requirements for SIP-PBX devices.

TEST GROUP 1.1: Registration Mode

Scope

Tests in this group apply to SIP-PBX devices, andcover basic connectivity between SP-SSE and a SIP-PBX in which the SIP-PBX and SP-SSE are configured to use Registration Mode as defined in [SIPconnect].

SC-IT.Conf.1.1.1: Registration Setup

Objective: This test case verifies SIP-PBX Registration during the initial setup phase.

Requirements Tested: REQ24333, REQ24335, REQ24336, REQ24308, REQ24309

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset.

Procedure:

Step	Action	Expected Results	REQ
1.	Reset or restart SIP-PBX to trigger to send REGISTER message to SP-SSE.	Verify the domain name and headers in request URI of the Register message. • REGISTER → Request URI = sip: sp.lab.com • REGISTER → Proxy-Require = gin • REGISTER → Require = gin • REGISTER → Supported = path • REGISTER → To = sip:pbx-1@sp.lab.com • REGISTER → From = sip:pbx-1@sp.lab.com • REGISTER → Contact = sip:192.0.2.4;bnc (or corresponding IP address)	REQ24308

Possible Problems:

- None.

SC-IT.Conf.1.1.2: Registration Failure

Objective: This test case verifies the behavior of a SIP-PBX for various Registration Failures.

Requirements Tested: REQ24339, REQ24340, REQ24341, REQ24342, REQ24345, REQ24346, REQ24348, REQ24349, REQ24351, REQ24352, REQ24356, REQ24360, REQ24361, REQ24422, REQ24423, REQ24424, REQ24424a, REQ24424b, REQ24414, REQ24415, REQ24362, REQ24363

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset.

Procedure:

Part A: No Register Response, Established Connection-based Transport – No Alternative

Step	Action	Expected Results	Requirement
1.	SP-SSE is configured to discard Registration Requests, however is enabled for TCP connections.		
2.	Reset or restart SIP-PBX to trigger to send REGISTER message to SP-SSE.	The SIP-PBX successfully establishes a TCP connection to SP-SSE and transmits a valid Register Request to SP-SSE.	
3.	Wait Timer_F (default: 32 seconds).	The SIP-PBX initiates a new TCP connection to SP-SSE and transmits a valid Register Request to SP-SSE.	REQ24340
4.	Wait registration delay time of 30 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
5.	Wait registration delay time (prev*2) of 60 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
6.	Wait registration delay time (prev*2) of 120 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
7.	Wait registration delay time (prev*2) of 240 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
8.	Wait registration delay time (prev*2) of 480 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
9.	Wait registration delay time (prev*2) of 960 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	

10.	Wait max registration delay time of 960 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	REQ24342

Part B: No Register Response, Established Connection-based Transport – Alternative Available

Step	Action	Expected Results	Requirement
11.	SP-SSE-2 is available as an alternate, and is configured as such on SIP-PBX. SP-SSE and SP-SSE2 are configured to discard Registration Requests, however is enabled for TCP connections.		
12.	Reset or restart SIP-PBX to trigger to send REGISTER message to SP-SSE.	The SIP-PBX successfully establishes a TCP connection to SP-SSE and transmits a valid Register Request to SP-SSE.	
13.	Wait Timer_F (default: 32 seconds).	The SIP-PBX initiates a new TCP connection to SP-SSE and transmits a valid Register Request to SP-SSE.	REQ24340
14.	Wait registration delay time of 30 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE2.	REQ24339, REQ24341

Part C: Transport Error – Alternative Available

Step	Action	Expected Results	Requirement
15.	SP-SSE-2 is configured as an alternate SSE on SIP-PBX. SP-SSE is offline, but configured as primary SSE on SIP-PBX.		
16.	Reset or restart SIP-PBX to trigger to send REGISTER message to SP-SSE.	The SIP-PBX attempts to establish a TCP connection to SP-SSE. The connection fails. The SIP-PBX then transmits a valid Register Request to SP-SSE2.	REQ24339, REQ24341

Part D: Reception of "404 Not Found" Response

Step	Action	Expected Results	Requirement
17.	Reset or restart SIP-PBX to trigger to send REGISTER message to SP-SSE.	SIP-PBX transmits a valid Register Request to SP-SSE.	
18.	SP-SSE transmits a "404 Not Found" Response to SIP-PBX.	The SIP-PBX logs the error to the administrator.	REQ24345

19.	Wait registration delay time of 30 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
20.	Wait registration delay time (prev*2) of 60 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
21.	Wait registration delay time (prev*2) of 120 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
22.	Wait registration delay time (prev*2) of 240 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
23.	Wait registration delay time (prev*2) of 480 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
24.	Wait registration delay time (prev*2) of 960 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
25.	Wait max registration delay time of 960 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	REQ24346

Part E: Multiple Error Responses

Step	Action	Expected Results	Requirement
26.	Reset or restart SIP-PBX to trigger to send REGISTER message to SP-SSE.	SIP-PBX transmits a valid Register Request to SP-SSE.	
27.	SP-SSE transmits a "404 Not Found" Response to SIP-PBX.		
28.	Wait registration delay time of 30 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
29.	SP-SSE transmits a "404 Not Found" Response to SIP-PBX.		
30.	Wait registration delay time (prev*2) of 60 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
31.	SP-SSE transmits a "404 Not Found" Response to SIP-PBX.	The SIP-PBX logs the error to the administrator.	REQ24348, REQ24349

Part F: Reception of "480 Temporarily Unavailable" Response

Step	Action	Expected Results	Requirement
32.	Reset or restart SIP-PBX to trigger to send REGISTER message to SP-SSE.	SIP-PBX transmits a valid Register Request to SP-SSE.	

33.	SP-SSE transmits a "480Temporarily Unavailable" Response to SIP-PBX.		
34.	Wait registration delay time of 30 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
35.	SP-SSE transmits a "480 Temporarily Unavailable" Response to SIP-PBX.		
36.	Wait registration delay time (prev*2) of 60 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
37.	SP-SSE transmits a "480 Temporarily Unavailable" Response to SIP-PBX.	The SIP-PBX logs the error to the administrator.	REQ24351, REQ24352

Part G: SP-SSE Administratively Disabled or Overloaded

Step	Action	Expected Results	Requirement
38.	SP-SSE-2 is offline but is configured as an alternate SP-SSE on SIP-PBX.		
39.	Reset or restart SIP-PBX to trigger to send REGISTER message to SP-SSE.	SIP-PBX transmits a valid Register Request to SP-SSE.	
40.	SP-SSE transmits a "503 Service Unavailable" Response to SIP-PBX. The Response has a "Retry-After" header field with a value of 15 seconds.		
41.	Wait registration delay time of 15 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	REQ24356, REQ24361, REQ24422, REQ24423
42.	SP-SSE transmits a "503 Service Unavailable" Response to SIP-PBX. The Response has a "Retry-After" header field with a value of 45 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE2.	REQ24424
43.	The TCP connection to SP-SSE2 fails.		

44.	Wait the remainder of the previous "Retry-After" time (45 seconds).	The SIP-PBX transmits a valid Register Request to SP-SSE.	REQ24356, REQ24424, REQ24424b

Part H: Other 4xx/5xx/6xx Responses

Step	Action	Expected Results	Requirement
45.	Reset or restart SIP-PBX to trigger to send REGISTER message to SP-SSE.	The SIP-PBX successfully establishes a TCP connection to SP-SSE and transmits a valid Register Request to SP-SSE.	
46.	SP-SSE transmits a "410 Gone" Response to SIP-PBX.		
47.	Wait Timer_F (default: 32 seconds).	The SIP-PBX initiates a new TCP connection to SP-SSE and transmits a valid Register Request to SP-SSE.	REQ24414
48.	SP-SSE transmits a "410 Gone" Response to SIP-PBX.		
49.	Wait registration delay time of 30 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
50.	SP-SSE transmits a "410 Gone" Response to SIP-PBX.		
51.	Wait registration delay time (prev*2) of 60 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
52.	SP-SSE transmits a "410 Gone" Response to SIP-PBX.		
53.	Wait registration delay time (prev*2) of 120 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
54.	SP-SSE transmits a "410 Gone" Response to SIP-PBX.		
55.	Wait registration delay time (prev*2) of 240 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
56.	SP-SSE transmits a "410 Gone" Response to SIP-PBX.		
57.	Wait registration delay time (prev*2) of 480 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	
58.	SP-SSE transmits a "410 Gone" Response to SIP-PBX.		
59.	Wait registration delay time (prev*2) of 960 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	

60.	SP-SSE transmits a "410 Gone" Response to SIP-PBX.		
61.	Wait max registration delay time of 960 seconds.	The SIP-PBX transmits a valid Register Request to SP-SSE.	REQ24415

Part I: Registration-related failures for other Requests

Step	Action	Expected Results	Requirement
62.	Reset or restart SIP-PBX to trigger to send REGISTER message to SP-SSE.	The SIP-PBX successfully establishes a TCP connection to SP-SSE and transmits a valid Register Request to SP-SSE.	
63.	SP-SSE transmits a "200 OK" Response to SIP-PBX.	SIP-PBX successfully registers to SP-SSE.	
64.	Phone e1 places a call to phone s1.	SIP-PBX transmits a SIP INVITE request to SP-SSE.	
65.	Wait Timer_B (default: 32 seconds).	The SIP-PBX initiates a new TCP connection to SP-SSE and transmits a valid Register Request to SP-SSE. SIP-PBX also notifies the Administrator of the error.	REQ24362, REQ24363, REQ24366, REQ24419

Possible Problems:

- None.

SC-IT.Conf.1.1.3: Maintaining Registration

Objective: This test verifies that a SIP-PBX honors the REGISTER expiry timer provided by the SP-SSE.

Requirements Tested: REQ24364

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset.

Procedure:

Step	Action	Expected Results	Requirement
1.	Reset or restart SIP-PBX to trigger to send REGISTER message to SP-SSE.	The SIP-PBX successfully establishes a TCP connection to SP-SSE and transmits a valid Register Request to SP-SSE.	
2.	SP-SSE transmits a "200 OK" Response to SIP-PBX with an Expiry time of 60 seconds.	SIP-PBX successfully registers to SP-SSE.	
3.	Wait 60 seconds.	SIP-PBX transmits a valid Register Request to SP-SSE.	REQ24364
4.	SP-SSE transmits a "200 OK" Response to SIP-PBX with an Expiry time of 120 seconds.	SIP-PBX successfully registers to SP-SSE.	
5.	Wait 120 seconds.	SIP-PBX transmits a valid Register Request to SP-SSE.	REQ24364
6.	SP-SSE transmits a "200 OK" Response to SIP-PBX with an Expiry time of 30 seconds.	SIP-PBX successfully registers to SP-SSE.	
7.	Wait 30 seconds.	SIP-PBX transmits a valid Register Request to SP-SSE.	REQ24364

Possible Problems:

- None.

SC-IT.Conf.1.1.4: Authentication

Objective: This test verifies that a SIP-PBX correctly utilizes Digest Authentication when communicating with a SP-SSE.

Requirements Tested: REQ24337, REQ24368, REQ24370, REQ24371, REQ24416

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset.

Procedure:

Step	Action	Expected Results	Requirement
1.	Reset or restart SIP-PBX to trigger to send REGISTER message to SP-SSE.	The REGISTER Request includes the following headers: • REGISTER → Request URI = sip: sp.lab.com • REGISTER → Proxy-Require = gin • REGISTER → Require = gin • REGISTER → Supported = path • REGISTER → To = sip:pbx-1@sp.lab.com • REGISTER → From = sip:pbx-1@sp.lab.com • REGISTER → Contact = sip:192.0.2.4;bnc (or corresponding IP address)	
2.	SP-SSE transmits a 401 UNAUTHORIZED Response. The Response includes the WWW-Authenticate header.	SIP-PBX transmits a REGISTER Request to SP-SSE with all of the same headers found in Step 1, including a valid Authorization Header.	REQ24337, REQ24368, REQ24370, REQ24371
3.	SP-SSE successfully registers SIP-PBX and transmits a 200 OK Response to SIP-PBX.		
4.	The REGISTER Expiry timer expires.	SIP-PBX transmits a REGISTER Request to SP-SSE with all of the same headers found in Step 1, including a valid Authorization Header.	REQ24416

Possible Problems:

- None.

SC-IT.Conf.1.1.5: TLS Server Mode

Objective: Verify that a SIP-PBX is capable of initializing a TLS connection, acting as a TLS client, and correctly utilizes a SP-SSE acting as a TLS server.

Requirements Tested: REQ24211, REQ24313, REQ24314, REQ24315, REQ24316, REQ24318, REQ24319, REQ24320, REQ24321, REQ24325, REQ24326, REQ24372, REQ24214, REQ24329, REQ24379, REQ24217, REQ24223

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset.

Procedure:

Part A: TLS Registration

Step	Action	Expected Results	Requirement
1.	SIP-PBX and SP-SSE are configured to use TLS with NULL Encryption. The Certificate Authority that signed the certificate used by SP-SSE is configured on SIP-PBX. The identity configured on SP-SSE is a SIP URI equal to the uniformResourceIdentifier in the subjectAltName field of its certificate.		REQ24318
2.	Reset or restart SIP-PBX to trigger to send REGISTER message to SP-SSE.	The SIP-PBX successfully establishes a TLS connection to SP-SSE and transmits a valid Register Request to SP-SSE. There are no SIP messages transmitted without SSL.	REQ24217
3.	SP-SSE transmits a "200 OK" Response to SIP-PBX with an Expiry time of 60 seconds.	SIP-PBX successfully registers to SP-SSE.	REQ24211 REQ24313 REQ24314 REQ24315 REQ24319 REQ24320 REQ24321 REQ24325 REQ24372

4.	Phone-e1 places a call to Phone-s1.	SIP-PBX transmits an INVITE Request to SP-SSE. The TLS connection used in Step 2 is reused. There are no SIP messages transmitted without SSL.	REQ24326 REQ24379
5.	Phone-s1 answers the call. SP-SSE transmits a “200 OK” message to SIP-PBX.	The call is setup successfully. The TLS connection used in Step 2 is reused. There are no SIP messages transmitted without SSL.	REQ24329
6.	Repeat steps 1-5 with a non-NULL Algorithm used as the Encryption cipher.		

Part B: Certificate Hostname

Step	Action	Expected Results	Requirement
7.	SIP-PBX and SP-SSE are configured to use TLS with NULL Encryption. The Certificate Authority that signed the certificate used by SP-SSE is configured on SIP-PBX. The SP-SSE hostname is located only in the subjectCommonName (DN) of the certificate, and is not present in the subjectAltName.		
8.	Reset or restart SIP-PBX to trigger to send REGISTER message to SP-SSE.	The SIP-PBX successfully establishes a TLS connection to SP-SSE and transmits a valid Register Request to SP-SSE. There are no SIP messages transmitted without SSL.	REQ24214
9.	SP-SSE transmits a “200 OK” Response to SIP-PBX with an Expiry time of 60 seconds.	SIP-PBX successfully registers to SP-SSE.	

Part C: DNS Hostname

Step	Action	Expected Results	Requirement

10.	SIP-PBX and SP-SSE are configured to use TLS with NULL Encryption. The Certificate Authority that signed the certificate used by SP-SSE is configured on SIP-PBX. The SP-SSE Certificate uses a DNS name as an identity instead of a SIP URI.		
11.	Reset or restart SIP-PBX to trigger to send REGISTER message to SP-SSE.	The SIP-PBX successfully establishes a TLS connection to SP-SSE and transmits a valid Register Request to SP-SSE. There are no SIP messages transmitted without SSL.	
12.	SP-SSE transmits a "200 OK" Response to SIP-PBX with an Expiry time of 60 seconds.	SIP-PBX successfully registers to SP-SSE.	
13.	Phone-e1 places a call to Phone-s1.	SIP-PBX transmits an INVITE Request to SP-SSE. The TLS connection used in Step 2 is reused. There are no SIP messages transmitted without SSL.	
14.	Phone-s1 answers the call. SP-SSE transmits a "200 OK" message to SIP-PBX.	The call is setup successfully. The TLS connection used in Step 2 is reused. There are no SIP messages transmitted without SSL.	REQ24223

Possible Problems:

- None.

TEST GROUP 1.2: Static Mode

Scope

Tests in this group apply to SIP-PBX devices, and cover basic connectivity between SP-SSE and a SIP-PBX in which the SIP-PBX and SP-SSE are configured to use Static Mode as defined in [SIPconnect].

SC-IT.Conf.1.2.1: SP-SSE Address Acquisition by SIP-PBX

Objective: This test verifies that a SIP-PBX is able to acquire the SP-SSE Address through either DNS or Configuration.

Requirements Tested: REQ24310, REQ24377, REQ24377b, REQ24378, REQ24377a, REQ24404

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset.

Note: SIP-PBX MUST Support Part A, Part B, or Both.

Procedure:

Part A: Manual Configuration

Step	Action	Expected Results	Requirement
1.	Configure SIP-PBX with the FQDN, IP address, port, and transport protocol of SP-SSE.	SIP-PBX has an interface that supports configuring the SP-SSE address, and it is configured successfully. The address field should support both IP Address and hostname, but hostname SHOULD be preferred.	REQ24377b REQ24378
2.	Phone-e1 places a call to Phone-s1.	SIP-PBX transmits an INVITE Request to SP-SSE. The INVITE message uses the FQDN or IP address of SP-SSE in the Request URI, e.g. INVITE sip:+13036611001@sippbx.lab.com;user=phone A stable two-way call is established.	REQ24310 REQ24377

Part B: DNS Configuration

Step	Action	Expected Results	Requirement
1.	SIP-PBX is configured with the Service Provider’s domain name and is restarted, or triggered to configure through DNS.	SIP-PBX transmits a DNS SRV query to determine the IP address, transport protocol, and port number of the SIP-PBX(s) associated with the Service Providers’ domain name	REQ24377a

2.	Phone-e1 places a call to Phone-s1.	SIP-PBX transmits an INVITE Request to SP-SSE. The INVITE message uses the FQDN or IP address of the SP-SSE in the Request URI, e.g. INVITE sip:+13036611001@sippbx.lab.com;user=phone A stable two-way call is established.	REQ24404

Possible Problems:

- None.

SC-IT.Conf.1.2.2: Static Mode Failure Detection

Objective: This test verifies methods used by a SIP-PBX to determine if a SP-SSE has failed.

Requirements Tested: REQ24395, REQ24395.1, REQ24395.2, REQ24396

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset.

Procedure:

Part A: Periodic OPTIONS Request

Step	Action	Expected Results	Requirement
1.	Configure SIP-PBX to use SP-SSE as it's primary SSE.		
2.	Observe the network for Periodic OPTIONS Requests transmitted by SIP-PBX.	SIP-PBX Transmits an OPTIONS request.	
3.	SP-SSE transmits a "200 OK" Response.		
4.	Observe the network for Periodic OPTIONS Requests transmitted by SIP-PBX.	SIP-PBX Transmits an OPTIONS request.	
5.	SP-SSE does not respond to the OPTIONS request.	SIP-PBX may determine that SP-SSE is unreachable, and take steps to confirm.	REQ24395, REQ24395.1

Part B: Periodic CR/LF

Step	Action	Expected Results	Requirement
6.	Configure SIP-PBX to use SP-SSE as it's primary SSE.		
7.	Observe the network for Periodic Carriage Return/Line Feed character transmissions.	SIP-PBX Transmits a CR/LF character.	
8.	SP-SSE transmits a TCP ACK.		
9.	Observe the network for Periodic Carriage Return/Line Feed character transmissions.	SIP-PBX Transmits a CR/LF character.	

10.	SP-SSE transmits a TCP RST.	SIP-PBX may determine that SP-SSE is unreachable, and take steps to confirm.	REQ24395, REQ24395.2

Part C: Configurable Keep-Alive

Step	Action	Expected Results	Requirement
11.	If SIP-PBX supports either of Part A or Part B, this test must be supported.		
12.	Configure the Keep-Alive mechanism to confirm the connection every 30 seconds.		
13.	Observe the network for no more than 30 seconds.	SIP-PBX transmits either an OPTIONS request, or a CR/LF character to SP-SSE.	
14.	Configure the Keep-Alive mechanism to confirm the connection every 15 seconds.		
15.	Observe the network for no more than 30 seconds.	SIP-PBX transmits either an OPTIONS request, or a CR/LF character to SP-SSE.	REQ24396

Possible Problems:

- None.

SC-IT.Conf.1.2.3: TLS Authentication

Objective: This test verifies the use of the TLS Mutual Authentication Model, as well as other TLS requirements, for SIP-PBX devices in Static Mode.

Requirements Tested: REQ24380, REQ24382, REQ24383, REQ24385, REQ24386, REQ24387, REQ24390, REQ24391, REQ24397, REQ24211, REQ24213, REQ24215, REQ24221

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset.

Procedure:

Part A: Session initiated by SIP-PBX

Step	Action	Expected Results	Requirement
1.	SIP-PBX and SP-SSE are configured to use TLS with NULL Encryption. Each are configured with certificates signed by an CA (i.e. not self-signed). The Certificate Authority that signed the certificate used by SP-SSE is configured on SIP-PBX. The identity configured on SIP-PBX is a SIP URI equal to the uniformResourceIdentifier in the subjectAltName field of its certificate. Likewise, the CA that signed the certificate used by SIP-PBX is configured on SP-SSE.		REQ24380 REQ24390 REQ24391
2.	Phone-e1 places a call to Phone-s1.	SIP-PBX initiates a TLS connection with SP-SSE. SIP-PBX transmits an INVITE Request to SP-SSE. There are no SIP messages transmitted without SSL.	REQ24382 REQ24217 REQ24221

3.	SP-SSE establishes a TLS connection with SIP-PBX. Phone-s1 answers the call. SP-SSE transmits a “200 OK” message to SIP-PBX.	The call is setup successfully. The TLS connection used in Step 2 is reused. There are no SIP messages transmitted without SSL.	
4.	Repeat steps 1-5 with a non-NULL Algorithm used as the Encryption cipher.		

Part B: Session initiated by SP-SSE

Step	Action	Expected Results	Requirement
5.	SIP-PBX and SP-SSE are configured to use TLS with NULL Encryption. Each are configured with certificates signed by an CA (i.e. not self-signed). The Certificate Authority that signed the certificate used by SP-SSE is configured on SIP-PBX. Likewise, the CA that signed the certificate used by SIP-PBX is configured on SP-SSE.		REQ24380
6.	Phone-s1 places a call to Phone-e1. SP-SSE initiates a TLS connection with SIP-PBX. SP-SSE transmits an INVITE Request to SIP-PBX. The INVITE request contains a "transport=TLS" URI parameter.	SIP-PBX establishes a TLS connection with SP-SSE. SIP-PBX responds with a “200” OK message to SP-SSE. SIP-PBX ignores the “transport=TLS” parameter. There are no SIP messages transmitted without SSL.	REQ24382 REQ24213 REQ24215
7.	Phone-e1 answers the call.	The call is setup successfully. The TLS connection used in Step 2 is reused. There are no SIP messages transmitted without SSL.	
8.	Repeat steps 1-5 with a non-NULL Algorithm used as the Encryption cipher.		

Part C: Reject Non-TLS Connection Attempts

Step	Action	Expected Results	Requirement

9.	SIP-PBX is configured to use TLS with NULL Encryption. It is configured with a certificate signed by a CA (i.e. not self-signed). SIP-PBX is specifically configured to require TLS for all session traffic. SP-SSE is not configured to use TLS.		REQ24380
10.	Phone-s1 places a call to Phone-e1. SP-SSE initiates a TCP connection with SIP-PBX.	SIP-PBX must reject the connection attempt, and must not establish a TLS connection with SP-SSE.	REQ24382 REQ24387

Possible Problems:

- None.

TEST GROUP 1.3: Basic Call Origination (DOD) and Termination (DID)

Scope:

Tests in this group cover basic calls originating from the SIP-PBX toward the SP-SSE. This section describes guidelines for populating the Request-URI and the "P-Asserted-Identity", "To", "From", "Privacy", and "Route" header fields for new-dialog INVITE requests sent between an Enterprise and a Service Provider. The test cases in this and following groups are by default to be run in Registration mode over a TCP transport.

SC-IT.Conf.1.3.1: Verification of INVITE Message Parameters When Originating a DOD Call

Objective: This test case verifies the format of the INVITE Request message generated by a SIP-PBX when originating a DOD call.

Requirements Tested: REQ24201, REQ24209, REQ24225, REQ24245, REQ24246, REQ24247, REQ24249, REQ24250

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-e1 places a call to Phone-s1.	SIP-PBX transmits an INVITE Request to SP-SSE. Verify the headers in the INVITE Request according to **Table A** below.	REQ24225, REQ24245, REQ24246, REQ24247, REQ24249, REQ24250

Table A:

Request-URI	Verify the **Request-URI** header in INVITE Request is populated with one of the following forms: • sip:+13036611001@sp.lab.com; user=phone • sip:3036611001@sp.lab.com The SP domain name must be used in the host part.
To	Verify that **To** header field in the INVITE message is populated with one of the following forms: • sip:+13036611001@sp.lab.com; user=phone • sip:3036611001@sp.lab.com
P-Asserted-Identity	Verify that the INVITE message contains a **P-Asserted-Identity**header field with one the following forms. • P-Asserted-Identity: "Joe Smith" <sip: +13035555555@sp.lab.com; user=phone><sip:13036611001@sp.lab.com> • P-Asserted-Identity: tel:+13035555555 (Note: The display name may or may not be present in the P-Asserted-Identity header field.)
From	Verify the **From** header in the INVITE message contains the URI with the following elements: • The Enterprise domain name (i.e. From: <sip:+13036621001@sp.lab.com;user=phone >)

Possible Problems:

- None.

SC-IT.Conf.1.3.2: INVITE Message Processing When Terminating a DID Call

Objective: This test case verifies a SIP-PBX correctly processes an INVITE Request message received from an SP-SSE when terminating a DID call.

Requirements Tested: REQ24201, REQ24209, REQ24232, REQ24233, REQ24234, REQ24241, REQ24242, REQ24243

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode.

Procedure:

Part A: Use of Request-URI field

Step	Action	Expected Results	Requirement
1.	SP-SSE transmits an INVITE request to SIP-PBX. The "Request-URI" field is populated with the valid Destination URI for Phone-e1. The "To" field is populated with an unknown destination (e.g. sip:+12225553000@unknown.com)	SIP-PBX processes the INVITE request without error.	REQ24233 REQ24234"

Part B: Processing P-Asserted-Identity

Step	Action	Expected Results	Requirement
2.	SP-SSE transmits an INVITE request to SIP-PBX. The From and P-Asserted-Identity fields contains a SIP-URI as follows: • The E.164 calling number in the user part • The Service provider domain name in the host name part. (e.g. sip:+13036611001@sp.lab.com;\ user=phone>)	SIP-PBX processes the INVITE request without error.	REQ24241

Part C: Processing 2 P-Asserted-Identity Fields

Step	Action	Expected Results	Requirement

3.	SP-SSE transmits an INVITE request to SIP-PBX. The From and P-Asserted-Identity fields contains a SIP-URI as follows: • The E.164 calling number in the user part • The Service provider domain name in the host name part. (e.g. sip:+13036611001@sp.lab.com;\ user=phone>) In addition, a second P-Asserted-Identity Field contains a Tel URI (e.g. tel:+1-222-555-3000	SIP-PBX processes the INVITE request without error.	REQ24242

Possible Problems:

- None.

SC-IT.Conf.1.3.3: SIP-PBX Support for Privacy when processing INVITE for Terminating DID Call

Objective: This test case verifies a SIP-PBX correctly supports the "Privacy" header field.

Requirements Tested: REQ24244

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode.

Procedure:

Step	Action	Expected Results	Requirement
1.	SP-SSE transmits an INVITE request to SIP-PBX. The "From" header field URI contains: "Anonymous"<sip:anonymous@anonymous.invalid> The Privacy Header = 'id'	SIP-PBX processes the INVITE request without error. The caller ID is displayed as "anonymous"	REQ24244

Possible Problems:

- None.

SC-IT.Conf.1.3.4: SIP-PBX Support for Privacy when generating INVITE message for Originating DOD Call

Objective: This test verifies that a SIP-PBX correctly formats an INVITE message when privacy has been request for originating a DOD call.

Requirements Tested:REQ24251, REQ24252, REQ24253

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode.

Procedure:

Part A: Privacy Requested, Fulfilled

Step	Action	Expected Results	Requirement
1.	Configure SIP-PBX to suppress the caller id when originating a call.		
2.	Phone-e1 places a call to Phone-s1.	SIP-PBX transmits an INVITE Request to SP-SSE. • Verify the INVITE message contains the **Privacy** header with the value, 'id'. • Verify the From header field contains following elements:"Anonymous"<sip:anonymous@anonymous.invalid>	REQ24251, REQ24252

Part B: Privacy Requested, Overridden

Step	Action	Expected Results	Requirement
3.	Configure SIP-PBX to override requests for Privacy when originating a DOD call.		
4.	Phone-e1 places a call to Phone-s1.	SIP-PBX transmits an INVITE Request to SP-SSE. • Verify the INVITE message contains the **Privacy** header with the value, 'None'.	REQ24253

Possible Problems:

- None.

TEST GROUP 1.4: Basic Features

Scope

Tests in this group verify basic features specific to a SIP-PBX device.

SC-IT.Conf.1.4.1: Call Forwarding

Objective: This test verifies the format of the INVITE message transmitted by a SIP-PBX upon receiving an INVITE message for a call flow that should be forwarded.

Requirements Tested: REQ24255

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode.

Procedure:

Step	Action	Expected Results	Requirement
1.	The number associated with Phone-e1 is configured on SIP-PBX to be forwarded to Phone-s2.		
2.	Phone-e1 places a call to Phone-s1.	SIP-PBX transmits an INVITE Request to SP-SSE. Verify fields according to **Table A below.**	REQ24255

Table A:

Request-URI	Correct format. (e.g. sip:+13036611001@sp.lab.com;user=phone or sip:3036611001@sp.lab.com) • The phone number does not contain visual separators such as "-".
Request-URI	Populated with a domain name follows one of the forms below: • Service provider domain name
P-Asserted-Identity	Contains the value for the originating user in the INVITE received from SP-SSE. (e.g. "John Doe" <sip: +13036621001@sp.lab.com ;user=phone>).

Possible Problems:

- None.

SC-IT.Conf.1.4.2: Blind Call Transfer - Transferor in SIP-PBX

Objective: This test verifies the basic call transfer capability of a SIP-PBX.

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

Requirements Tested: REQ24257, REQ24259

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode. Phone-s2 is a phone connected to the service provider network.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-e1 (Transferor) places a call to Phone-s1 (Transferee).	A stable two-way call is established from the SIP-PBX transferor to the SP-SSE transferee.	
2.	Phone-e1 conducts a Blind Call Transfer of Phone-s1 to Phone-s2.	SIP-PBX initiates a second session via an INVITE Request transmitted to Phone-s2.	REQ24257, REQ24259
3.	SP-SSE acknowledges the INVITE Request with a 200 OK Response.	The call for Phone-s1 is successfully transferred from Phone-e1 to Phone-s2.	REQ24257, REQ24259

Possible Problems:

- None.

SC-IT.Conf.1.4.3: Blind Call Transfer - Transferee in SIP-PBX

Objective: This test verifies the basic call transfer capability of a SIP-PBX.

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

Requirements Tested: REQ24257, REQ24259

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode. Phone-s2 is a phone connected to the service provider network.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-s1 (Transferor) originates a call to Phone-e1 (Transferee).	A stable two-way call is established from the SIP-PBX transferor to the SP-SSE transferee.	
2.	Hook Flash on Phone-s1; Phone-s1 conducts a Blind Call Transfer of Phone-e1 to Phone-s2. SP-SSE transmits an INVITE Request to hold the media connection to Phone-e1. (SDP a=inactive or sendonly)	SIP-PBX acknowledges the INVITE Request with a 200 OK Response.	REQ24257, REQ24259

Possible Problems:

- None.

SC-IT.Conf.1.4.4: Blind Call Transfer – Transferee and Transfer-To in SIP-PBX

Objective: This test verifies the basic call transfer capability of a SIP-PBX.

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

Requirements Tested: REQ24257, REQ24259

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode. Phone-e2 is a phone connected to the enterprise network.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-s1 (Transferor) originates a call to Phone-e1 (Transferee).	A stable two-way call is established from the SIP-PBX transferor to the SP-SSE transferee.	
2.	Hook Flash on Phone-s1; Phone-s1 conducts a Blind Call Transfer of Phone-e1 to Phone-e2. SP-SSE transmits an INVITE Request with no SDP to SIP-PBX for Phone-e2.	SIP-PBX acknowledges the INVITE Request with a 200 OK Response.	REQ24257, REQ24259

Possible Problems:

- None.

SC-IT.Conf.1.4.5: Attended Call Transfer – Transferor in SIP-PBX

Objective: This test verifies the basic call transfer capability of a SIP-PBX.

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

Requirements Tested: REQ24257, REQ24259

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode. Phone-s2 is a phone connected to the service provider network.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-e1 (Transferor) originates a call to Phone-s1 (Transferee).	A stable two-way call is established from the SIP-PBX transferor to the SP-SSE transferee.	
2.	Hook Flash on Phone-e1.	SIP-PBX initiates media hold toward phone-s1 (transferee). Verify the SIP-PBX sends a re-INVITE to hold the remote party upon detection of a hookflash.	REQ24257, REQ24259
3.	Phone-e1 places a second call to Phone-s2 and hangs up when completed.	SIP-PBX initiates the second call via an INVITE Request transmitted to Phone-s2. A stable two-way call is established between Phone-s1 and Phone-s2.	

Possible Problems:

- None.

SC-IT.Conf.1.4.6: Blind Call Transfer – Transferor and Transferee in SIP-PBX

Objective: This test verifies the basic call transfer capability of a SIP-PBX.

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

Requirements Tested: REQ24257, REQ24259

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode. Phone-e2 is a phone connected to the enterprise network.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-e1 (Transferor) originates a call to Phone-e2 (Transferee).	A stable two-way call is established.	
2.	Hook Flash on Phone-e1; Phone-e1 originates a call to Phone-s1.	SIP-PBX initiates the second call to phone-s1 (transfer-to) via an INVITE request.	REQ24257, REQ24259
3.	SP-SSE transmits a 200 OK Response.	A stable two-way call is established between Phone-e2 and Phone-s1.	

Possible Problems:

- None.

SC-IT.Conf.1.4.7: Blind Call Transfer – Transferor and Transfer-to in SIP-PBX

Objective: This test verifies the basic call transfer capability of a SIP-PBX.

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

Requirements Tested: REQ24257, REQ24259, REQ24275

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode. Phone-e2 is a phone connected to the enterprise network.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-e1 (Transferor) originates a call to Phone-s2 (Transferee).	A stable two-way call is established.	
2.	Hook Flash on Phone-e1; Phone-e1 originates a call to Phone-s1.	SIP-PBX sends an INVITE to hold the transferee party (SDP: a=inactive, sendonly)	REQ24257, REQ24259, REQ24275

Possible Problems:

- None.

SC-IT.Conf.1.4.8: Blind Call Transfer – Transferee and Transfer-to in SIP-PBX

Objective: This test verifies the basic call transfer capability of a SIP-PBX.

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

Requirements Tested: REQ24257, REQ24259

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode. Phone-e2 is a phone connected to the enterprise network.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-s1 (Transferor) originates a call to Phone-e1 (Transferee).	A stable two-way call is established.	
2.	Hook Flash on Phone-s1. SP-SSE transmits a re-INVITE to SIP-PBX to initiate a media hold.	SIP-PBX transmits a 200 OK Response to SP-SSE.	REQ24257, REQ24259
3.	Phone-s1 originates a call to Phone-e2. SP-SSE transmits an INVITE Request to SIP-PBX.	SIP-PBX successfully completes the transaction.	

Possible Problems:

- None.

SC-IT.Conf.1.4.9: Blind Call Transfer with Early Media – Transferee and Transfer-to in SIP-PBX

Objective: This test verifies the basic call transfer capability of a SIP-PBX.

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

Requirements Tested: REQ24257, REQ24259

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode. Phone-s2 is a phone connected to the service provider network.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-s1 (Transferor) originates a call to Phone-s2 (Transferee).	A stable two-way call is established.	
2.	Hook Flash on Phone-s1; Phone-s1 originates a call to Phone-e1. SP-SSE transmits a INVITE request to SIP-PBX to initiate a media hold.	SIP-PBX transmits a 200 OK Response to SP-SSE.	REQ24257, REQ24259
3.	Phone-s1 transfers Phone-s2 to Phone-e1.	The call between Phone-s2 and Phone-e1 is successfully established.	

Possible Problems:

- None.

SC-IT.Conf.1.4.10: Emergency Call Dial Plan

Objective: This test verifies the emergency call dial plan of a SIP-PBX.

Requirements Tested: REQ24261, REQ24262, REQ24263, REQ24264, REQ24265

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-e1 originates an emergency call by entering 911 or some other emergency number that has been configured.	Verify that the SIP-PBX sends INVITE message. Verify the INVITE's Request-URI is sip:911@sp.lab.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. (Note: The number in the request URI may be different than 911 if both the SP-SSE and SIP-PBX are configured to recognize emergency numbers from other locales.)	REQ24261, REQ24262, REQ24263, REQ24264, REQ24265

Possible Problems:

- None.

TEST SECTION 2: SP-SSE

Scope

Tests in this section cover SIPconnect 1.1 Requirements for SP-SSE devices.

TEST GROUP 2.1: Registration Mode

Scope

Tests in this group cover basic connectivity between SP-SSE and a SIP-PBX in which the SIP-PBX and SP-SSE are configured to use Registration Mode as defined in [SIPconnect].

SC-IT.Conf.2.1.1: Registration Setup

Objective: This test case verifies SP-SSE Registration processing functionality during the initial setup phase.

Requirements Tested: REQ24312, REQ24333

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset.

Procedure:

Step	Action	Expected Results	REQ
1.	SIP-PBX transmits a REGISTER Request to SP-SSE. The REGISTER Request includes the following headers: • REGISTER → Request URI = sip: sp.lab.com • REGISTER → Proxy-Require = gin • REGISTER → Require = gin • REGISTER → Supported = path • REGISTER → To = sip:pbx-1@sp.lab.com • REGISTER → From = sip:pbx-1@sp.lab.com • REGISTER → Contact = sip:192.0.2.4;bnc (or corresponding IP address)	SP-SSE successfully registers SIP-PBX and transmits a 200 OK Response to SIP-PBX. (SP-SSE may transmit a 401 UNAUTHORIZED Response. In this case SIP-PBX transmits a follow up REGISTER Request with an appropriate Authorization Header.)	REQ24312, REG24333

Possible Problems:

- None.

SC-IT.Conf.2.1.2: Registration Failure

Objective: This test case verifies the behavior of a SP-SSE in various registration failure scenarios.

Requirements Tested: REQ24343, REQ24344, REQ24347

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset.

Procedure:

Part A: No 302 Moved Temporarily Response

Step	Action	Expected Results	REQ
1.	SP-SSE is configured not to register SIP-PBX.		
2.	SIP-PBX transmits a REGISTER Request to SP-SSE. The REGISTER Request includes the following headers: • REGISTER → Request URI = sip: sp.lab.com • REGISTER → Proxy-Require = gin • REGISTER → Require = gin • REGISTER → Supported = path • REGISTER → To = sip:pbx-1@sp.lab.com • REGISTER → From = sip:pbx-1@sp.lab.com • REGISTER → Contact = sip:192.0.2.4;bnc (or corresponding IP address)	SP-SSE MUST NOT transmit a 302 Moved Temporarily Response to SIP-PBX.	REQ24343

Part B: 404 Not Found Response

Step	Action	Expected Results	REQ
3.	SP-SSE is configured to not have a record of SIP-PBX's AOR in its database.		

4.	SIP-PBX transmits a REGISTER Request to SP-SSE. The REGISTER Request includes the following headers: • REGISTER → Request URI = sip: sp.lab.com • REGISTER → Proxy-Require = gin • REGISTER → Require = gin • REGISTER → Supported = path • REGISTER → To = sip:pbx-1@sp.lab.com • REGISTER → From = sip:pbx-1@sp.lab.com • REGISTER → Contact = sip:192.0.2.4;bnc (or corresponding IP address)	SP-SSE transmits a 404 Not Found Response.	REQ24344

Part C: Stale REGISTER Challenge Response

Step	Action	Expected Results	REQ
5.	Configure SP-SSE to require Authorization.		
6.	SIP-PBX transmits a REGISTER Request to SP-SSE. The REGISTER Request includes the following headers: • REGISTER → Request URI = sip: sp.lab.com • REGISTER → Proxy-Require = gin • REGISTER → Require = gin • REGISTER → Supported = path • REGISTER → To = sip:pbx-1@sp.lab.com • REGISTER → From = sip:pbx-1@sp.lab.com • REGISTER → Contact = sip:192.0.2.4;bnc (or corresponding IP address)	SP-SSE transmits a 401 UNAUTHORIZED Response. The Response includes the WWW-Authenticate header.	
7.	SIP-PBX transmits a REGISTER Request to SP-SSE with all of the same headers found in Step 5, including a valid Authorization Header.	SP-SSE successfully registers SIP-PBX and transmits a 200 OK Response to SIP-PBX.	

8.	SIP-PBX transmits a REGISTER Request to SP-SSE with all of the same headers as found in Step 5. This REGISTER Request has an invalid Authorization Header.	SP-SSE transmits exactly one of the following error Responses: • 401 Unauthorized • 407 Proxy Authentication Required • 403 Forbidden	REQ24347

Possible Problems:

- In Part C, SP-SSE may be configured to silently discard the Registration Message in Step 8.

SC-IT.Conf.2.1.3: Inbound Requests

Objective: This test verifies that upon successful registration, a SP-SSE correctly routes incoming, out-of-dialog, requests to the SIP-PBX for which the URI is registered.

Requirements Tested: REQ24373

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset.

Procedure:

Step	Action	Expected Results	REQ
1.	SIP-PBX transmits a REGISTER Request to SP-SSE. The REGISTER Request includes the following headers: • REGISTER → Request URI = sip: sp.lab.com • REGISTER → Proxy-Require = gin • REGISTER → Require = gin • REGISTER → Supported = path • REGISTER → To = sip:pbx-1@sp.lab.com • REGISTER → From = sip:pbx-1@sp.lab.com • REGISTER → Contact = sip:192.0.2.4;bnc (or corresponding IP address)	SP-SSE successfully registers SIP-PBX and transmits a 200 OK Response to SIP-PBX. (SP-SSE may transmit a 401 UNAUTHORIZED Response. In this case SIP-PBX transmits a follow up REGISTER Request with an appropriate Authorization Header.)	
2.	Phone-s1 places a call to Phone-e1.	SP-SSE transmits an INVITE Request to SIP-PBX. SP-SSE uses the IP address, instead of the domain name. (ex. INVITE sip:+13036611001@192.0.2.4)	REQ24373

Possible Problems:

- None.

SC-IT.Conf.2.1.4: Authentication

Objective: This test verifies that a SP-SSE correctly authenticates a SIP-PBX utilizing Digest Authentication.

Requirements Tested:REQ24337, REQ24327, REQ24368, REQ24370a

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset.

Procedure:

Part A: Authorization Valid

Step	Action	Expected Results	REQ
1.	Configure SP-SSE to require Authorization.		
2.	SIP-PBX transmits a REGISTER Request to SP-SSE. The REGISTER Request includes the following headers: • REGISTER → Request URI = sip: sp.lab.com • REGISTER → Proxy-Require = gin • REGISTER → Require = gin • REGISTER → Supported = path • REGISTER → To = sip:pbx-1@sp.lab.com • REGISTER → From = sip:pbx-1@sp.lab.com • REGISTER → Contact = sip:192.0.2.4;bnc (or corresponding IP address)	SP-SSE transmits a 401 UNAUTHORIZED Response. The Response includes the WWW-Authenticate header.	REQ24327
3.	SIP-PBX transmits a REGISTER Request to SP-SSE with all of the same headers found in Step 1, including a valid Authorization Header.	SP-SSE successfully registers SIP-PBX and transmits a 200 OK Response to SIP-PBX.	REQ24337, REQ24368, REQ24370a

Part B: Authorization Invalid

Step	Action	Expected Results	REQ
4.	Configure SP-SSE to require Authorization.		

<table>
<tr><td>5.</td><td>SIP-PBX transmits a REGISTER Request to SP-SSE.

The REGISTER Request includes the following headers:
REGISTER → Request URI = sip: sp.lab.com
REGISTER → Proxy-Require = gin
REGISTER → Require = gin
REGISTER → Supported = path
REGISTER → To = sip:pbx-1@sp.lab.com
REGISTER → From = sip:pbx-1@sp.lab.com
REGISTER → Contact = sip:192.0.2.4;bnc (or corresponding IP address)</td><td>SP-SSE transmits a 401 UNAUTHORIZED Response. The Response includes the WWW-Authenticate header.</td><td>REQ24327</td></tr>
<tr><td>6.</td><td>SIP-PBX transmits a REGISTER Request to SP-SSE with all of the same headers found in Step 1, including an invalid Authorization Header.</td><td>SP-SSE transmits a 401 UNAUTHORIZED Response. The Response includes the WWW-Authenticate header.</td><td>REQ24327</td></tr>
</table>

Possible Problems:

- In Part B, SP-SSE may send a 403 AUTHENTICATION FAILURE Response in Step 6.

SC-IT.Conf.2.1.5: TLS Server Mode

Objective: Verify that a SP-SSE is capable of acting as a TLS server and accepting SIP-PBX devices acting as TLS clients.

Requirements Tested: REQ24211, REQ24313, REQ24323, REQ24328, REQ24329, REQ24331, REQ24409, REQ24221

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset.

Procedure:

Step	Action	Expected Results	Requirement
1.	SIP-PBX and SP-SSE are configured to use TLS with NULL Encryption. The Certificate Authority that signed the certificate used by SIP-PBX is configured on SP-SSE. The identity configured on SP-SSE is a SIP URI equal to the uniformResourceIdentifier in the subjectAltName field of its certificate.		REQ24211 REQ24323
2.	SIP-PBX initiates a TLS connection to SP-SSE.	SP-SSE successfully establishes a TLS connection with SIP-PBX.	REQ24313 REQ24217 REQ24221
3.	SIP-PBX transmits a REGISTER message to SP-SSE. The REGISTER message contains a "transport=tls" URI parameter in the Contact Header Field.	SP-SSE transmits a "200 OK" Response to SIP-PBX. SP-SSE ignores the "transport=tls" parameter. There are no SIP messages transmitted without SSL.	REQ24331
4.	SIP-PBX successfully registers to SP-SSE. Phone-s1 places a call to Phone-e1.	SP-SSE transmits an INVITE Request to SIP-PBX using the TLS connection from Step 2.	REQ24328
5.	Phone-e1 answers the call. SIP-PBX transmits a "200 OK" message to SP-SSE.	The call is setup successfully. The TLS connection used in Step 2 is reused. There are no SIP messages transmitted without SSL.	REQ24329 REQ24409

6.	Repeat steps 1-5 with a non-NULL Algorithm used as the Encryption cipher.		

Possible Problems:

- None.

TEST GROUP 2.2: Static Mode

Scope:

Tests in this group apply to SP-SSE devices, and cover basic connectivity between SP-SSE and a SIP-PBX in which the SIP-PBX and SP-SSE are configured to use Static Mode as defined in [SIPconnect].

SC-IT.Conf.2.2.1: SIP-PBX Address Acquisition by SP-SSE

Objective: This test verifies that an SP-SSE is able to acquire the Enterprise Address through either DNS or Configuration.

Requirements Tested: REQ24374, REQ24374a, REQ24374b, REQ24375, REQ24376, REQ24407, REQ24421a, REQ24421b, REQ24408

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset.

Note: SP-SSE MUST Support Part A, Part B, or Both.

Procedure:

Part A: Manual Configuration

Step	Action	Expected Results	Requirement
1.	Configure SP-SSE with FQDN, IP address, port and transport protocol of SIP-PBX.	SP-SSE has an interface that supports manual configuration. The signaling address field should support both IP Address and hostname, but hostname SHOULD be preferred.	REQ24374b REQ24421b REQ24408
2.	Phone-s1 places a call to Phone-e1.	SP-SSE transmits an INVITE Request to SIP-PBX. The INVITE message from the SP-SSE uses the FQDN or IP address of the SIP-PBX in the Request URI, e.g. INVITE sip:+13036611001@sippbx.lab.com;user=phone A stable two-way call is established.	REQ24310 REQ24377

Part B: DNS Configuration

Step	Action	Expected Results	Requirement
3.	SP-SSE is configured with the Enterprise Network's domain name and is restarted, or triggered to configure through DNS.	SP-SSE transmits a DNS SRV query to determine the IP address, transport protocol, and port number of the SIP-PBX(s) associated with the Enterprise Network's domain name	REQ24374a REQ24375 REQ24376

4.	Phone-s1 places a call to Phone-e1.	SP-SSE transmits an INVITE Request to SIP-PBX. The INVITE message from the SP-SSE uses the FQDN or IP address of the SIP-PBX in the Request URI, e.g. INVITE sip:+13036611001@sippbx.lab.com;user=phone A stable two-way call is established.	REQ24421a

Possible Problems:

- None.

SC-IT.Conf.2.2.2: TLS Authentication

Objective: This test verifies the use of the TLS Mutual Authentication Model, as well as other TLS requirements, for SP-SSE devices in Static Mode.

Requirements Tested: REQ24380, REQ24382, REQ24383, REQ24385, REQ24386, REQ24387, REQ24393, REQ24394, REQ24397, REQ24211, REQ24213, REQ24215, REQ24214, REQ24221, REQ24223

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset.

Procedure:

Part A: Session initiated by SIP-PBX

Step	Action	Expected Results	Requirement
1.	SIP-PBX and SP-SSE are configured to use TLS with NULL Encryption. Each are configured with certificates signed by an CA (i.e. not self-signed). The Certificate Authority that signed the certificate used by SP-SSE is configured on SIP-PBX. The identity configured on SP-SSE is a SIP URI equal to the uniformResourceIdentifier in the subjectAltName field of its certificate. Likewise, the CA that signed the certificate used by SIP-PBX is configured on SP-SSE.		REQ24380, REQ24383 REQ24393 REQ24394
2.	Phone-e1 places a call to Phone-s1. SIP-PBX initiates a TLS connection with SP-SSE.	SP-SSE establishes a TLS connection with SIP-PBX.	REQ24382 REQ24385 REQ24217 REQ24221

3.	SIP-PBX transmits an INVITE Request to SP-SSE. The INVITE request contains a "transport=TLS" URI parameter.	Phone-s1 answers the call. SP-SSE transmits a "200 OK" message to SIP-PBX. SP-SSE ignores the "transport=TLS" parameter. The call is setup successfully. There are no SIP messages transmitted without SSL.	REQ24213 REQ24215
4.	Repeat steps 1-5 with a non-NULL Algorithm used as the Encryption cipher.		

Part B: Session initiated by SP-SSE

Step	Action	Expected Results	Requirement
5.	SIP-PBX and SP-SSE are configured to use TLS with NULL Encryption. Each are configured with certificates signed by an CA (i.e. not self-signed). The Certificate Authority that signed the certificate used by SP-SSE is configured on SIP-PBX. Likewise, the CA that signed the certificate used by SIP-PBX is configured on SP-SSE.		REQ24380
6.	Phone-s1 places a call to Phone-e1.	SP-SSE initiates a TLS connection with SIP-PBX. SP-SSE transmits an INVITE Request to SIP-PBX. There are no SIP messages transmitted without SSL.	REQ24382 REQ24386
7.	SIP-PBX establishes a TLS connection with SP-SSE. SIP-PBXresponds with a "200" OK message to SP-SSE. Phone-e1 answers the call.	The call is setup successfully. There are no SIP messages transmitted without SSL.	
8.	Repeat steps 1-5 with a non-NULL Algorithm used as the Encryption cipher.		

Part C: Reject Non-TLS Connection Attempts

Step	Action	Expected Results	Requirement

9.	SP-SSE is configured to use TLS with NULL Encryption. It is configured with a certificate signed by a CA (i.e. not self-signed). SP-SSE is specifically configured to require TLS for all session traffic. SIP-PBX is not configured to use TLS.		REQ24380
10.	Phone-e1 places a call to Phone-s1. SIP-PBX initiates a TCP connection with SP-SSE.	SP-SSE must reject the connection attempt, and must not establish a TLS connection with SIP-PBX.	REQ24382 REQ24387

Part D: Certificate Hostname

Step	Action	Expected Results	Requirement
11.	SIP-PBX and SP-SSE are configured to use TLS with NULL Encryption. Each are configured with certificates signed by an CA (i.e. not self-signed). The Certificate Authority that signed the certificate used by SP-SSE is configured on SIP-PBX. Likewise, the CA that signed the certificate used by SIP-PBX is configured on SP-SSE. The SIP-PBX hostname is located only in the subjectCommonName (DN) of the certificate, and is not present in the subjectAltName.		
12.	Phone-e1 places a call to Phone-s1. SIP-PBX initiates a TLS connection with SP-SSE.	SP-SSE establishes a TLS connection with SIP-PBX.	REQ24214

13.	SIP-PBX transmits an INVITE Request to SP-SSE. The INVITE request contains a "transport=TLS" URI parameter.	Phone-s1 answers the call. SP-SSE transmits a “200 OK” message to SIP-PBX. SP-SSE ignores the “transport=TLS” parameter. The call is setup successfully. There are no SIP messages transmitted without SSL.	

Part E: DNS Hostname

Step	Action	Expected Results	Requirement
14.	SIP-PBX and SP-SSE are configured to use TLS with NULL Encryption. Each are configured with certificates signed by an CA (i.e. not self-signed). The Certificate Authority that signed the certificate used by SP-SSE is configured on SIP-PBX. The SP-SSE Certificate uses a DNS name as an identity instead of a SIP URI. Likewise, the CA that signed the certificate used by SIP-PBX is configured on SP-SSE.		
15.	Phone-e1 places a call to Phone-s1. SIP-PBX initiates a TLS connection with SP-SSE.	SP-SSE establishes a TLS connection with SIP-PBX.	
16.	SIP-PBX transmits an INVITE Request to SP-SSE. The INVITE request contains a "transport=TLS" URI parameter.	Phone-s1 answers the call. SP-SSE transmits a “200 OK” message to SIP-PBX. SP-SSE ignores the “transport=TLS” parameter. The call is setup successfully. There are no SIP messages transmitted without SSL.	REQ24223

Possible Problems:

- None.

TEST GROUP 2.3: Basic Call Termination (DID) and Origination (DOD)

Scope:

Tests in this group cover basic calls originating from the SP-SSE toward the SIP-PBX. This section describes guidelines for populating the Request-URI and the “P-Asserted-Identity”, “To”, “From”, “Privacy”, and “Route” header fields for new-dialog INVITE requests sent between an Enterprise and a Service Provider. The test cases in this and following groups are by default to be run in Registration mode over a TCP transport.

SC-IT.Conf.2.3.1: Verification of INVITE Message Parameters When Terminating a DID Call

Objective: This test case verifies the format of the INVITE Request message generated by a SP-SSE during a DID call when the caller does not request calling number privacy.

Requirements Tested: REQ24201, REQ24209, REQ24225, REQ24230, REQ24231, REQ24236, REQ24237, REQ24240, REQ24398

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX registers to SP-SSE using either Registration Mode or Static Mode.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-s1 places a call to Phone-e1.	SP-SSE transmits an INVITE Request to SIP-PBX. Verify the headers in the INVITE Request according to **Table A** below.	REQ24225 REQ24230 REQ24231 REQ24236 REQ24237 REQ24398

Table A:

Request-URI	Verify the user part of **Request-URI** in INVITE Requestis the Enterprise Public Identity of the called Enterprise user and has the correct format. (i.e. sip:+13036611001@10.32.1.5. Note that this example contains an IP address, but an FQDN would also be acceptable) • The phone number contains a "+" sign • The phone number does not contain a visual separators such as "-". It may also have a Contact URI provided by the SIP PBX in a previous request or response.
From	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 (e.g. From: <sip:+ 13036611001@sp.lab.com;user=phone>)

Possible Problems:

- None.

SC-IT.Conf.2.3.2: INVITE Processing for Originating a DOD Call

Objective: This test case verifies that a SP-SSE correctly processes various INVITE message formats when originating a DOD call.

Requirements Tested: REQ24201, REQ24209,REQ24248

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX registers to SP-SSE using either Registration Mode or Static Mode.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-e1 places a call to Phone-s1. SIP-PBX transmits a SIP INVITE Request to SP-SSE. The "To" header field of the INVITE Request contains an invalid URI for Phone-s1. The "Request-URI" contains a valid URI for Phone-s1.	SP-SSE correctly processes the INVITE Request and transmits a 200 OK Response to SIP-PBX.	REQ24248

Possible Problems:

- None.

SC-IT.Conf.2.3.3: Verification of INVITE Parameters for Terminating DID Anonymous Calls

Objective: This test case verifies the format of the "From" header field in an INVITE Request message generated by a SP-SSE during a DID call.

Requirements Tested: REQ24235, REQ24238

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX registers to SP-SSE using either Registration Mode or Static Mode.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-s1 places a call to Phone-e1. Phone-s1 is configured to request calls be completed anonymously.	SP-SSE transmits an INVITE Request to SIP-PBX. • The **From** header field URI contains: "Anonymous"<sip:anonymous@anonymous.invalid> • The message **does not** include a **P-Asserted-Identity** header	REQ24235, REQ24238

Possible Problems:

- None.

SC-IT.Conf.2.3.4: Verification of INVITE Parameters for Terminating DID Unknown/Public Calls

Objective: This test case verifies the format of the URI in an INVITE Request message generated by a SP-SSE during a DID call when no caller identity is available, and privacy has not been requested.

Requirements Tested: REQ24411

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX registers to SP-SSE using either Registration Mode or Static Mode.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-s1 places a call to Phone-e1. Phone-s1 does not provide any Caller Identification and does not request any privacy.	SP-SSE transmits an INVITE Request to SIP-PBX. • The **From** header field URI contains: "unavailable@unknown.invalid" • The message **does not** include a **P-Asserted-Identity** header	REQ24411

Possible Problems:

- None.

TEST GROUP 2.4: Basic Features

Scope

Tests in this group verify basic features specific to a SP-SSE device.

SC-IT.Conf.2.4.1: SP-SSE Support for Privacy when generating an INVITE message to Terminate a DID Call

Objective: This test case verifies the contents of an INVITE Request message generated by a SP-SSE during a DID call when a caller has requested privacy and the SP-SSE is able to assert an identity.

Requirements Tested: REQ24239, REQ24412

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX registers to SP-SSE using either Registration Mode or Static Mode.

Procedure:

Step	**Action**	**Expected Results**	**Requirement**
1.	Phone-s1 places a call to Phone-e1. Phone-s1 resides on a "trusted" enterprise network and requests privacy.	SP-SSE transmits an INVITE Request to SIP-PBX. The INVITE Request contains the following header fields and values: • **P-Asserted-Identity** header is present • **Privacy** header: 'id' • **From** header:sip:anonymous@anonymous.invalid	REQ24239
2.	None	The **P-Asserted-Identity** field in the INVITE Request from Step 1 SHOULD include a display name along with the URI if it is available and can be delivered according to policy. This requirement is OPTIONAL.	REQ24412

Possible Problems:

- None.

SC-IT.Conf.2.4.2: SP-SSE Support for Privacy when processing an INVITE message to Originate a DOD Call

Objective: This test case verifies that an SP-SSE correctly processes an INVITE message when originating a DOD call.

Requirements Tested: REQ24254

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX registers to SP-SSE using either Registration Mode or Static Mode.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-e1 places a call to Phone-s1. SIP-PBX transmits an INVITE Request to SP-SSE that contains a **Privacy** header with the value: ‘id’.	SP-SSE correctly processes the INVITE Request and transmits a 200 OK Response to SIP-PBX.	REQ24254

Possible Problems:

- None.

SC-IT.Conf.2.4.3: SP-SSE Call Forwarding

Objective: This test verifies that an SP-SSE device correctly processes and accepts a forwarded call from a SIP-PBX.

Requirements Tested: REQ24256

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX registers to SP-SSE using either Registration Mode or Static Mode.

Procedure:

Step	Action	Expected Results	Requirement
1.	The number associated with Phone-e1 is configured to forward all calls to Phone-s2, a phone on the SP-SSE network.		
2.	Phone-s1 places a call to Phone-e1.	SP-SSE transmits an INVITE Request to SIP-PBX.	
3.	SIP-PBX transmits an INVITE Request to SP-SSE to forward the call for Phone-e1 to Phone-s2.	SP-SSE correctly processes the INVITE Request and transmits a 200 OK Response to SIP-PBX.	REQ24256

Possible Problems:

- None.

SC-IT.Conf.2.4.4: Blind Call Transfer - Transferor in SIP-PBX

Objective: This test verifies the basic call transfer capability of a SP-SSE.

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

Requirements Tested: REQ24257, REQ24259

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode. Phone-s2 is a phone connected to the service provider network.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-e1 (Transferor) places a call to Phone-s1 (Transferee).	A stable two-way call is established from the SIP-PBX transferor to the SP-SSE transferee.	
2.	Phone-e1 conducts a Blind Call Transfer of Phone-s1 to Phone-s2. SIP-PBX initiates a second session via an INVITE Request transmitted to Phone-s2.	SP-SSE acknowledges the INVITE Request with a 200 OK Response. The call for Phone-s1 is successfully transferred from Phone-e1 to Phone-s2.	REQ24257, REQ24259

Possible Problems:

- None.

SC-IT.Conf.2.4.5: Blind Call Transfer - Transferee in SIP-PBX

Objective: This test verifies the basic call transfer capability of a SP-SSE.

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

Requirements Tested: REQ24257, REQ24259, REQ24275

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode. Phone-s2 is a phone connected to the service provider network.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-s1 (Transferor) originates a call to Phone-e1 (Transferee).	A stable two-way call is established from the SIP-PBX transferor to the SP-SSE transferee.	
2.	Hook Flash on Phone-s1; Phone-s1 conducts a Blind Call Transfer of Phone-e1 to Phone-s2.	SP-SSE transmits an INVITE Request to hold the media connection to Phone-e1. (SDP a=inactive or sendonly)	REQ24257, REQ24259, REQ24275
3.	SIP-PBX acknowledges the INVITE Request with a 200 OK Response.	The call for Phone-e1 is successfully transferred to Phone-s2.	

Possible Problems:

- None.

SC-IT.Conf.2.4.6: Blind Call Transfer – Transferee and Transfer-To in SIP-PBX

Objective: This test verifies the basic call transfer capability of a SP-SSE.

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

Requirements Tested: REQ24257, REQ24259

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode. Phone-e2 is a phone connected to the enterprise network.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-s1 (Transferor) originates a call to Phone-e1 (Transferee).	A stable two-way call is established from the SIP-PBX transferor to the SP-SSE transferee.	
2.	Hook Flash on Phone-s1; Phone-s1 conducts a Blind Call Transfer of Phone-e1 to Phone-e2.	SP-SSE transmits an INVITE Request with no SDP to SIP-PBX for Phone-e2.	REQ24257, REQ24259
3.	SIP-PBX acknowledges the INVITE Request with a 200 OK Response.	The call for Phone-e1 is successfully transferred to Phone-e2.	

Possible Problems:

- None.

SC-IT.Conf.2.4.7: Attended Call Transfer – Transferor in SIP-PBX

Objective: This test verifies the basic call transfer capability of a SP-SSE.

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

Requirements Tested: REQ24257, REQ24259

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode. Phone-s2 is a phone connected to the service provider network.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-e1 (Transferor) originates a call to Phone-s1 (Transferee).	A stable two-way call is established from the SIP-PBX transferor to the SP-SSE transferee.	
2.	Hook Flash on Phone-e1. SIP-PBX initiates media hold toward phone-s1 (transferee). SIP-PBX sends a re-INVITE to hold the remote party upon detection of a hookflash.	SP-SSE transmits a 200 OK Response.	REQ24257, REQ24259
3.	Phone-e1 places a second call to Phone-s2 and hangs up when completed. SIP-PBX initiates the second call via an INVITE Request transmitted to Phone-s2.	SP-SSE transmits a 200 OK Response. A stable two-way call is established between Phone-s1 and Phone-s2.	

Possible Problems:

- None.

SC-IT.Conf.2.4.8: Blind Call Transfer – Transferor and Transferee in SIP-PBX

Objective: This test verifies the basic call transfer capability of a SP-SSE.

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

Requirements Tested: REQ24257, REQ24259

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode. Phone-e2 is a phone connected to the enterprise network.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-e1 (Transferor) originates a call to Phone-e2 (Transferee).	A stable two-way call is established.	
2.	Hook Flash on Phone-e1; Phone-e1 originates a call to Phone-s1. SIP-PBX initiates the second call to phone-s1 (transfer-to) via an INVITE request.	SP-SSE transmits a 200 OK Response. A stable two-way call is established between Phone-e2 and Phone-s1.	REQ24257, REQ24259

Possible Problems:

- None.

SC-IT.Conf.2.4.9: Blind Call Transfer – Transferor and Transfer-to in SIP-PBX

Objective: This test verifies the basic call transfer capability of a SP-SSE.

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

Requirements Tested: REQ24257, REQ24259

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode. Phone-e2 is a phone connected to the enterprise network.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-e1 (Transferor) originates a call to Phone-s2 (Transferee).	A stable two-way call is established.	
2.	Hook Flash on Phone-e1; Phone-e1 originates a call to Phone-s1. SIP-PBX sends an INVITE to hold the transferee party (SDP: a=inactive, sendonly)	SP-SSE transmits a 200 OK Response. A stable two-way call is established between Phone-s1 and Phone-s2.	REQ24257, REQ24259

Possible Problems:

- None.

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SC-IT.Conf.2.4.10: Blind Call Transfer – Transferee and Transfer-to in SIP-PBX

Objective: This test verifies the basic call transfer capability of a SP-SSE.

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

Requirements Tested: REQ24257, REQ24259

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode. Phone-e2 is a phone connected to the enterprise network.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-s1 (Transferor) originates a call to Phone-e1 (Transferee).	A stable two-way call is established.	
2.	Hook Flash on Phone-s1; Phone-s1 originates a call to Phone-e2.	SP-SSE transmits a re-INVITE to SIP-PBX to initiate a media hold.	REQ24257, REQ24259
3.	SIP-PBX transmits a 200 OK Response to SP-SSE. Phone-s1 originates a call to Phone-e2.	SP-SSE transmits an INVITE Request to SIP-PBX.	

Possible Problems:

- None.

SC-IT.Conf.2.4.11: Blind Call Transfer with Early Media – Transferee and Transfer-to in SIP-PBX

Objective: This test verifies the basic call transfer capability of a SIP-PBX.

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

Requirements Tested: REQ24257, REQ24259

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode. Phone-s2 is a phone connected to the service provider network.

Procedure:

Step	Action	Expected Results	Requirement
1.	Phone-s1 (Transferor) originates a call to Phone-s2 (Transferee).	A stable two-way call is established.	
2.	Hook Flash on Phone-s1; Phone-s1 originates a call to Phone-e1.	SP-SSE transmits a INVITE request to SIP-PBX to initiate a media hold.	REQ24257, REQ24259
3.	SIP-PBX transmits a 200 OK Response to SP-SSE. Phone-s1 transfers Phone-s2 to Phone-e1.	SP-SSE transmits an INVITE request to SIP-PBX to transfer Phone-s2 to Phone-e1.	

Possible Problems:

- None.

SC-IT.Conf.2.4.12: Emergency Services

Objective: This test verifies the ability of an SP-SSE to identify an Emergency Call and correctly ignore any typical session limits applicable to non-emergency calls.

Requirements Tested: REQ24266, REQ24267

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode.

Procedure:

Part A: Emergency Call

Step	Action	Expected Results	Requirement
1.	Phone-e1 originates an emergency call by entering 911 or some other emergency number that has been configured. SIP-PBX sends an INVITE message. The INVITE's Request-URI is sip:911@sp.lab.com. The INVITE message includes P-Asserted-identity header. The P-Asserted-identity header includes the user name and enterprise user identity of the caller.	SP-SSE acknowledges the INVITE Request and transmits a 200 OK Response to SIP-PBX.	REQ24266

Part B: Emergency Call - Disregard Session Limits

Step	Action	Expected Results	Requirement
2.	Configure SP-SSE to accept only 2 simultaneous SIP sessions.		

3.	Phone-e2 is a phone connected to the enterprise network. Phone-e3 is a phone connected to the enterprise network. Phone-s2 is a phone connected to the service provider network. Phone-s3 is a phone connected to the service provider network.		
4.	Phone-e1 places a call to Phone-s1, and Phone-e2 places a call to Phone-s2.	Both calls are established successfully.	
5.	Phone-e3 places a call to Phone-s3.	The call is not completed. SP-SSE may transmit an error response to SIP-PBX.	
6.	Phone-e3 originates an emergency call by entering 911 or some other emergency number that has been configured. SIP-PBX sends an INVITE message. The INVITE's Request-URI is sip:911@sp.lab.com. The INVITE message includes P-Asserted-identity header. The P-Asserted-identity header includes the user name and enterprise user identity of the caller.	SP-SSE processes the call normally, disregarding the configured 2-session limit. The session is not blocked. SP-SSE acknowledges the INVITE Request and transmits a 200 OK Response to SIP-PBX.	REQ24267

Possible Problems:

- None.

TEST SECTION 3: Generic SIPconnect Requirements

Scope

Tests in this section verify requirements that apply equally to both SIP-PBX and SP-SSE devices.

TEST GROUP 3.1: SDP Exchange

Scope

Tests in this group verify basic signaling and SDP Offer/Answer exchange requirements.

SC-IT.Conf.3.1.1: Basic Signaling and SDP Exchange

Objective: This test verifies the basic signaling process and SDP Offer/Answer exchange requirements.

Requirements Tested: REQ24268, REQ24269, REQ24332, REQ24203

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode.

Procedure:

Part A: Direct Inbound Dial – SIP-PBX Only

Step	Action	Expected Results	Requirement
1.	Phone-s1 originates a call to Phone-e1. SP-SSE transmits an INVITE Request to SIP-PBX. The INVITE request does not include "100rel" or "precondition" in Require or Supported header fields.	SIP-PBX transmits a "180 – Ringing" Response, or a "183 – Session in Progress" Response to SP-SSE. The response does not contain SDP.	REQ24269
2.	Phone-e1 answers the call from Phone-s1.	SIP-PBX transmits a "200 – OK" Response to SP-SSE. The response contains an SDP Offer.	REQ24268
3.	SP-SSE transmits an ACK Request to SIP-PBX containing an SDP Answer.	The call is established successfully. All IP Addresses in each packet transmitted by SIP-PBX are publicly routable.	REQ24332

Part B: Direct Outbound Dial – SP-SSE Only

Step	Action	Expected Results	Requirement
4.	Phone-e1 originates a call to Phone-s1. SIP-PBX transmits an INVITE Request to SP-SSE. The INVITE request does not include "100rel" or "precondition" in Require or Supported header fields.	SP-SSE transmits a "180 – Ringing" Response, or a "183 – Session in Progress" Response to SIP-PBX. The response does not contain SDP.	REQ24269

1.	Phone-s1 answers the call from Phone-e1.	SP-SSE transmits a "200 – OK" Response to SIP-PBX. The response contains an SDP Offer.	REQ24268
2.	SIP-PBX transmits an ACK Request to SP-SSE containing an SDP Answer.	The call is established successfully. All IP Addresses in each packet transmitted by SP-SSE are publicly routable.	REQ24332

Possible Problems:

- None.

SC-IT.Conf.3.1.2: SDP Version Number

Objective: This test verifies SDP processing in cases when the version number has changed, and is unchanged, as well as verifies that that version number is correctly updated.

Requirements Tested: REQ24271, REQ24272

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode.

Procedure:

Part A: Direct Inbound Dial – Unchanged SDP Version – SIP-PBX Only

Step	Action	Expected Results	Requirement
1.	Phone-s1 originates a call to Phone-e1. SP-SSE transmits an INVITE Request to SIP-PBX. The INVITE contains a valid and appropriate SDP Offer.	SIP-PBX transmits a “180 – Ringing” Response, or a “183 – Session in Progress” Response to SP-SSE. The response contains a valid SDP Answer.	
2.	Phone-e1 answers the call from Phone-s1.	SIP-PBX transmits a “200 – OK” Response to SP-SSE.	
3.	SP-SSE transmits an ACK Request to SIP-PBX.	The call is established successfully.	
4.	SP-SSE transmits a reINVITE Request to SIP-PBX. The reINVITE contains the same SDP Offer as was given in Step 1, with an unchanged version.	SIP-PBX transmits a “200 – OK” Response to SP-SSE. The response contains a valid SDP Answer, which may be the same as was given in Step 1.	REQ24271 REQ24272

Part B: Direct Outbound Dial – SP-SSE Only

Step	Action	Expected Results	Requirement
5.	Phone-e1 originates a call to Phone-s1. SIP-PBX transmits an INVITE Request to SP-SSE. The INVITE contains a valid and appropriate SDP Offer.	SP-SSE transmits a “180 – Ringing” Response, or a “183 – Session in Progress” Response to SIP-PBX. The response contains a valid SDP Answer.	
6.	Phone-s1 answers the call from Phone-e1.	SP-SSE transmits a “200 – OK” Response to SIP-PBX.	
7.	SIP-PBX transmits an ACK Request to SP-SSE.	The call is established successfully.	

8.	SIP-PBX transmits a reINVITE Request to SP-SSE. The reINVITE contains the same SDP Offer as was given in Step 1, with an unchanged version.	SP-SSE transmits a "200 – OK" Response to SIP-PBX. The response contains a valid SDP Answer, which may be the same as was given in Step 1.	REQ24271 REQ24272

Possible Problems:

- None.

TEST GROUP 3.2: Session Hold

SC-IT.Conf.3.2.1: Session Hold

Objective: This test verifies the correctness of a SIP-PBX or SP-SSE when placing a session on Hold.

Requirements Tested: REQ24305, REQ24306, REQ24307, REQ24403, REQ24274

Test Setup:

Common Test Setup 1 is performed. Following the test, any configuration added by the test is removed or reset. SIP-PBX is registered to SP-SSE using either Registration Mode or Static Mode.

Procedure:

Part A: Call Hold by SIP-PBX – SIP-PBX Only

Step	Action	Expected Results	Requirement
1.	Phone-e1 places a call to Phone-s1.	A stable two-way call is established.	
2.	Phone-e1 initiates a call hold.	SIP-PBX transmits an INVITE Request to SP-SSE. The INVITE message includes an SDP with a version number in the o-line greater than what was used in Step 1. If the SIP-PBX is not providing MOH, then the SDP contains the attribute "a=inactive" or "a=sendonly". If the SIP-PBX is providing MOH, then the SDP contains an attribute "a=sendonly".	REQ24305, REQ24306, REQ24403, REQ24274
3.	SP-SSE acknowledges the INVITE Request.	The call is placed on hold. The RTP stream is one-way from SIP-PBX to SP-SSE.	
4.	Phone-e1 cancels the call hold.	SIP-PBX transmits an INVITE Request to SP-SSE. The INVITE message includes SDP with either with an attribute of "a=sendrecv", or no "a=" attribute. The "o=" attribute session version has been incremented from the value in the previous INVITE.	REQ24274
5.	Phone-s1 initiates a call hold. SP-SSE transmits an INVITE request with an SDP with the 'c' field equal to "0.0.0.0".	SIP-PBX acknowledges the INVITE request.	REQ24307

Part B: Call Hold by SP-SSE – SP-SSE Only

Step	Action	Expected Results	Requirement
6.	Phone-s1 places a call to Phone-e1.	A stable two-way call is established.	
7.	Phone-s1 initiates a call hold.	SP-SSE transmits an INVITE Request to SIP-PBX. The INVITE message includes an SDP with a version number in the o-line greater than what was used in Step 1. If the SP-SSE is not providing MOH, then the SDP contains the attribute "a=inactive" or "a=sendonly". If the SP-SSE is providing MOH, then the SDP contains an attribute "a=sendonly".	REQ24305, REQ24306, REQ24403, REQ24274
8.	SIP-PBX acknowledges the INVITE Request.	The call is placed on hold. The RTP stream is one-way from SIP-PBX to SP-SSE.	
9.	Phone-s1 cancels the call hold.	SP-SSE transmits an INVITE Request to SIP-PBX. The INVITE message includes SDP with either with an attribute of "a=sendrecv", or no "a=" attribute. The "o=" attribute session version has been incremented from the value in the previous INVITE.	REQ24274
10.	Phone-e1 initiates a call hold. SIP-PBX transmits an INVITE request with an SDP with the 'c' field equal to "0.0.0.0".	SP-SSE acknowledges the INVITE request.	REQ24307

Possible Problems:

- None.

APPENDIX A: Requirements Text

This section shows the section number and text for each requirement referenced in this document. Text of Requirements drawn from SIPconnect 1.1 Technical Recommendation.

Requirement	Section	Text	Requirement Level
REQ24201	6	SIP-PBXs and SP-SSEs MUST support SIP in accordance with [RFC 3261] [RFC 3261] and offer-answer in accordance with [RFC 3264], [RFC 3264], as qualified by statements in later sections of this document.	MUST
REQ24203	6	Instead, a SIP-PBX or SP-SSE 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
REQ24205	7	SIP-PBXs MUST support either Registration mode, as specified in Annex A, or Static mode, as described in Annex B. SIP-PBXs MAY support both modes.	MUST
REQ24207	7	SP-SSEs MUST support either Registration mode, as specified in Annex A, or Static mode, as described in Annex B. SP-SSEs MAY support both modes.	MUST
REQ24209	8	SIP-PBXs and SP-SSEs MUST implement TCP.	MUST
REQ24211	8.1	The SIP-PBX and SP-SSE MUST support Transport Layer Security (TLS) v1.0 as described in [RFC2246] and [RFC 3261].	MUST
REQ24213 REQ24215	8.1	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. [RFC 3261] Section 26.2.2 deprecates the "transport=TLS" URI parameter. SIP-PBXes and SP-SSEs MUST ignore this parameter.	MUST

REQ24214	8.1	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
REQ24217	8.1	When presenting a certificate, a SIP-PBX or SP-SSE SHOULD identify itself by means of a SIP URI using type uniformResourceIdentifier in the subjectAltName field, in accordance with [RFC 5280].	SHOULD
REQ24221	8.1	SIP-PBX and SP-SSE 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
REQ24223	8.1	Furthermore, SIP-PBX and SP-SSE 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
REQ24225	9	SIP-PBXs and SP-SSEs 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. The global E.164 number MUST begin with a leading "+", MUST NOT contain a phone-context parameter and MUST NOT include visual separators.	MUST
REQ24225	9	SIP-PBXs and SP-SSEs 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. The global E.164 number MUST begin with a leading "+", MUST NOT contain a phone-context parameter and MUST NOT include visual separators.	MUST
REQ24230	10.1	The SP-SSE MUST ensure that all other header fields in the INVITE request comply with [RFC 3261].	MUST
REQ24231	10.1.1	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

REQ24232	10.1.1	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
REQ24233 REQ24234	10.1.2	As such, the SIP- PBX MUST NOT rely on the contents of "To" header field for routing decisions, but MUST use the Request-URI instead.	MUST NOT
REQ24235	10.1.3	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. sip:anonymous@anonymous.invalid	MUST
REQ24236 REQ24237	10.1.3	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. If any display name information is available and has not been restricted for delivery, it SHOULD also be provided.	MUST
REQ24238	10.1.4	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
REQ24239	10.1.4	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

REQ24240	10.1.4	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
REQ24241	10.1.4	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
REQ24242	10.1.4	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
REQ24243	10.1.4	MUST prefer the SIP URI when two are present.	MUST
REQ24244	10.1.4	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
REQ24245	10.2	The SIP-PBX MUST ensure that all other header fields in the INVITE request comply with [RFC 3261].	MUST
REQ24246	10.2.1	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
REQ24247	10.2.1	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 <-- No Space! sip:92125551212@sp.example.com	MUST
REQ24248	10.2.2	As such, the SP-SSE MUST NOT rely on the "To" header field URI for routing decisions, but use the Request-URI instead.	MUST NOT

REQ24249	10.2.3	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
REQ24250	10.2.4	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
REQ24251	10.2.4	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
REQ24252	10.2.5	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
REQ24253	10.2.5	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
REQ24254	10.2.5	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
REQ24255	11	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

REQ24256	11	An SP-SSE MUST be able to accept forwarded calls from a SIP-PBX. Note that an SP-SSE may enforce policies that include a variety of restrictions on calls forwarded from an untrusted SIP-PBX (e. g., mandating the inclusion of a "Diversion" header field [RFC 5806] with a "From" header field that does not correspond to an Enterprise Public Identify assigned to the SIP-PBX)	MUST
REQ24257 REQ24259	12.1 12.2	Call transfer can be accomplished by the use of REFER requests (a "proxy model") in accordance with [RFC 5589], or by the use of one or more INVITE/re-INVITE requests (a "third party call control model") The SP-SSE and SIP-PBX MUST support the use of INVITE/re-INVITE for initiating and responding to call transfers. The SIP-PBX and the SP-SSE 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
REQ24257 REQ24259	12.1 12.2	Call transfer can be accomplished by the use of REFER requests (a "proxy model") in accordance with [RFC 5589], or by the use of one or more INVITE/re-INVITE requests (a "third party call control model") The SP-SSE and SIP-PBX MUST support the use of INVITE/re-INVITE for initiating and responding to call transfers. The SIP-PBX and the SP-SSE 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
REQ24261	13	The SIP-PBX MUST have a dial plan that recognizes emergency calls.	MUST

REQ24262	13	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
REQ24263	13	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
REQ24264 REQ24265	13	The SIP PBX MUST NOT withhold the "P-Asserted-Identity" header field for privacy reasons and MUST NOT anonymize the "From" header field.	MUST NOT
REQ24266	13	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
REQ24267	13	The SP-SSE MUST NOT apply SIP session limits to emergency calls originated by the SIP-PBX. Note that this does not preclude the SP-SSE rejecting the emergency call for other reasons including local congestion or exceeding limits explicitly applicable for emergency calls.	MUST NOT
REQ24268	14.1	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
REQ24269	14.1	SIP-PBXs and SP-SSEs 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

REQ24271 REQ24272	14.1	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 generate a valid answer (which could be the same SDP sent previously, or could be different).	MUST
REQ24273	14.1	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
REQ24274	14.1	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
REQ24275	14.1	SIP-PBX and SP-SSE implementations sending changes to negotiated media capabilities via SIP reINVITE MUST support [RFC 3261], Section 14 "Modifying an Existing Session".	MUST
REQ24275	14.1	SIP-PBX and SP-SSE implementations sending changes to negotiated media capabilities via SIP reINVITE MUST support [RFC 3261], Section 14 "Modifying an Existing Session".	MUST
REQ24277	14.1	SIP UPDATE MAY be used for this purpose when both endpoints advertise support for [RFC 3311].	MAY
REQ24278	14.2	A Media Endpoint MUST transport and receive voice samples using the real-time transport protocol (RTP) as described in [RFC 3550].	MUST
REQ24279	14.2	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
REQ24280	14.2	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

REQ24281(a and b)	14.2	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
REQ24282	14.2	Any device intended for low-bandwidth operation SHOULD support [ITU-T G.729] codecs with a packetization rate of 20 ms.	SHOULD
REQ24283	14.2	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 and SP-SSE MUST assume that the far end Media Endpoint does not support receiving G.711 discontinuously.	MUST
REQ24285	14.2	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
REQ24286	14.2	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
REQ24287	14.2	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
REQ24288	14.3	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
REQ24289	14.3	Any Media Endpoint that supports receiving DTMF MUST support [RFC 4733] procedures.	MUST
REQ24290	14.3	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

REQ24291	14.3	For any local Media Endpoint that supports receiving telephone-event packets, the SIP-PBX or SP-SSE MUST include the supported events in an "a=fmtp:" line as is described as mandatory in [RFC 4733].	MUST
REQ24293	14.3	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
REQ24294	14.3	a SIP-PBX or SP-SSE 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
REQ24296	14.4	Any Media Endpoint that can introduce echo MUST provide [ITU-T G.168]-compliant echo cancellation.	MUST
REQ24297	14.5	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
REQ24298	14.5	Media Endpoints that support [ITU-T T.38] MUST support User Datagram Protocol Transport Layer (UDPTL) transport.	MUST
REQ24299	14.7	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
REQ24300	14.7	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
REQ24301	14.7	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 NOT

REQ24302	14.7	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
REQ24303	14.7	On receipt of valid RTP packets, a Media Endpoint MUST disable any local ringback tone and play the received media.	MUST
REQ24304	14.7	A Media Endpoint, on receipt of information concerning received SDP, MAY use the information to determine whether RTP packets received are valid	MAY
REQ24305	14.8	The hold initiator MUST set the SDP directionality attribute to "a=sendonly".	MUST
REQ24306 REQ24403	14.8	If the hold initiator does not provide MOH, it MUST set the SDP directionality attribute to "a=inactive" or "a=sendonly". If the hold initiator does not provide MOH, it MUST set the SDP directionality attribute to "a=inactive" or "a=sendonly". The attribute "a=inactive" is RECOMMENDED because it provides an indication to the held entity that MOH is not being provided by the hold initiator.	MUST
REQ24307	14.8	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.	MUST
REQ24308	15	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
REQ24309	15.1.1	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
REQ24310	15.1.1	The SIP-PBX MUST be capable of obtaining information about the SP-SSE, using the procedure described in Section 16.1.1.2.	MUST

REQ24311	15.1.2	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
REQ24312	15.1.2	The SP-SSE MUST obtain the SIP-PBX signaling address/port using SIP registration, as described in Section 15.4.	MUST
REQ24313	15.2	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
REQ24313	15.2	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
REQ24314	15.2	The SIP-PBX MUST be capable of initiating the establishment of a TLS session.	MUST
REQ24315	15.2	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). However, the SIP-PBX does not need to be provisioned with a certificate.	MUST
REQ24316	15.2	The SIP-PBX MUST validate the certificate received during TLS establishment using the path validation procedure described in [RFC 5280].	MUST
REQ24317	15.2	The SIP-PBX SHOULD verify the status of the certificate received during TLS establishment.	SHOULD
REQ24318	15.2	The SIP-PBX MUST be capable of being configured to require use of TLS to initiate a session.	MUST
REQ24319	15.2	When TLS is configured as required for session initiation, 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 NOT

REQ24320	15.2	The SIP-PBX MUST initiate the establishment of the TLS session,	MUST
REQ24321	15.2	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 NOT
REQ24322	15.2	When configuring DNS NAPTR and/or SRV resource records in accordance with Section 15.1.2, the SP-SSE SHOULD indicate support for TLS.	SHOULD
REQ24323	15.2	The SP-SSE MUST be configured with a verifiable digital certificate to secure a TLS session.	MUST
REQ24324	15.2	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
REQ24325	15.2	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
REQ24326	15.2	MUST use that connection to deliver the REGISTER request and all subsequent SIP messages to the SP-SSE.	MUST
REQ24327	15.2	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
REQ24328	15.2	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
REQ24329	15.2	The SIP-PBX and SP-SEE MUST avoid closing down the TLS connection, other than in exceptional circumstances (e.g., for maintenance).	MUST
REQ24329	15.2	The SIP-PBX and SP-SEE MUST avoid closing down the TLS connection, other than in exceptional circumstances (e.g., for maintenance).	MUST

REQ24331	15.2.1	When a SIP-PBX registers, the SP-SSE MUST ignore the transport=tls parameter in the "Contact" header field URI.	MUST
REQ24332	15.3	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
REQ24333	15.4	The SIP-PBX and SP-SEE 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
REQ24333	15.4	The SIP-PBX and SP-SEE 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
REQ24335	15.4	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
REQ24336	15.4	The SIP-PBX MUST insert the Registration AOR in the "From"and "To" header fields of the REGISTER request.	MUST
REQ24337	15.4	The SIP-PBX and SP SSE 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
REQ24337	15.4	The SIP-PBX and SP SSE 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

REQ24339	15.4.1.1	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
REQ24340	15.4.1.1	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
REQ24341	15.4.1.1	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 NOT
REQ24342	15.4.1.1	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
REQ24343	15.4.1.2	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 NOT
REQ24344	15.4.1.3	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
REQ24345	15.4.1.3	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
REQ24346	15.4.1.3	The SIP-PBX SHOULD follow the backoff procedures defined previously in Section 15.4.1.1.	SHOULD

REQ24347	15.4.1.4	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: a 401 Unauthorized, a 407 Proxy Authentication Required or a 403 Forbidden unless the SP-SSE is configured to silently discard these requests based on policy.	MUST
REQ24348	15.4.1.4	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
REQ24349	15.4.1.4	The SIP-PBX SHOULD follow the backoff procedures defined previously in Section 15.4.1.1.	SHOULD
REQ24350	15.4.1.5	If an SP-SSE is unable to complete registration, it MAY issue a 480 Temporarily Unavailable response code for a REGISTER request.	MAY
REQ24351	15.4.1.5	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
REQ24352	15.4.1.5	and MUST follow the backoff procedures defined previously in Section 15.4.1.1.	MUST
REQ24353	15.4.1.6	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
REQ24354 REQ24580	15.4.1.6	and SHOULD include a "Retry-After" header field value indicating how long the SIP-PBX should wait before re-attempting a REGISTER request to the same SP-SSE	SHOULD
REQ24355	15.4.1.6	This "Retry-After" header field value SHOULD include an element of randomness so that all served SIP- PBXes don't become synchronized and repeatedly attempt to register en mass.	SHOULD
REQ24356	15.4.1.6	A SIP-PBX receiving such a response MUST support the "Retry-After" header field,	MUST

REQ24360	15.4.1.6	If an alternate SP-SSE can be successfully reached and Registration succeeds through the alternate, the SIP-PBX MAY discard the "Retry-After" value of the original.	MAY
REQ24361	15.4.1.6	Otherwise, it MUST wait to reattempt registration to the original SP-SSE for the "Retry-After" interval.	MUST
REQ24362	15.4.2	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 consider the request attempt to have failed, assume that the SIP-PBX's registration is no longer active at the SP-SSE, and notify the SIP-PBX administrator if possible through some means.	MUST
REQ24363	15.4.2	The SIP-PBX SHOULD attempt re-registration using the procedures defined previously in Section 15.4.1.1.	SHOULD
REQ24364	15.5	Therefore the SIP-PBX MUST honor the REGISTER expiry time provided by the SP-SSE,	MUST
REQ24366	15.5	If failure is detected a SIP-PBX MUST attempt reconnection	MUST
REQ24368	15.6.1	The SIP-PBX and SP-SEE MUST support the digest authentication scheme as described in Section 22.4 of [RFC 3261].	MUST
REQ24368	15.6.1	The SIP-PBX and SP-SEE MUST support the digest authentication scheme as described in Section 22.4 of [RFC 3261].	MUST
REQ24370	15.6.1	The SIP-PBX MUST support receiving 401 Unauthorized and 407 Proxy Authentication Required from the SP-SSE.	MUST
REQ24370a	15.6.1	The SIP-PBX MUST support receiving 401 Unauthorized and 407 Proxy Authentication Required from the SP-SSE.	MUST

REQ24371	15.6.1	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
REQ24372	15.6.2	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
REQ24373	15.7	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
REQ24374	16.1.1.1	The SIP-PBX MUST provide its SIP signaling address and port to the SP-SSE using one of the following mechanisms:	MUST
REQ24374a REQ24375 REQ24376	16.1.1.1	DNS: The Enterprise Network ensures the existence of a publicly-accessible DNS server that is authoritative for its domain (or a sub-domain delegated by the Service Provider for use by the Enterprise). This DNS server SHOULD provide a DNS interface that supports NAPTR resource records and MUST provide a DNS interface that supports SRV resource records	MUST
REQ24374b	16.1.1.1	Configuration: The Enterprise Network provides information to allow the Service Provider to configure mapping of the Enterprise Fully Qualified Domain Name (FQDN) to the SIP-PBX signaling address/port and transport at the SP-SSE.	MUST
REQ24377	16.1.1.2	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

REQ24377a	16.1.1.2	[RFC 3263] "Locating SIP Servers": SIP-PBX utilizes DNS NAPTR and SRV queries as described in [RFC 3263] to determine the IP address(es), transport protocol(s), and port number(s) of the SP-SSE(s) associated with the Service ProviderÕs domain name. This option assumes that the SIP-PBX has been pre-configured with the domain name of the Service Provider Network.	
REQ24377b REQ24378	16.1.1.2	Configuration: One or more transport protocols and SIP signaling address(es)/port(s) of the SP- SSE are configured in the SIP-PBX. A configured SP-SSE signaling address SHOULD be in the form of a hostname that can be resolved through DNS A/AAAA resource records, rather than an IP address (see additional guidance in Section 17.1).	
REQ24379	16.1.1.2	When a TLS connection already exists, the SIP-PBX MUST reuse that TLS connection for all SIP messages.	MUST
REQ24380	16.2	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
REQ24380	16.2	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
REQ24382	16.2	Both SIP-PBX and SP-SSE MUST be able to initiate the establishment of a TLS session.	MUST
REQ24382	16.2	Both SIP-PBX and SP-SSE MUST be able to initiate the establishment of a TLS session.	MUST
REQ24383	16.2	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
REQ24383	16.2	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

REQ24384	16.2	Both SIP-PBX and SP-SSE MUST validate the certificate received during TLS establishment using the path validation procedure described in [RFC 5280].	MUST
REQ24385	16.2	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
REQ24385	16.2	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
REQ24386	16.2	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 NOT
REQ24386	16.2	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 NOT
REQ24387	16.2	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
REQ24387	16.2	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
REQ24388	16.2	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 NOT
REQ24389	16.2	When configuring DNS NAPTR and/or SRV resource records in accordance with Section 16.1.1.1, the SIP-PBX SHOULD indicate support for TLS.	SHOULD
REQ24390	16.2	The SIP-PBX MUST be configured with a verifiable digital certificate to secure a TLS session.	MUST

REQ24391	16.2	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
REQ24392	16.2	When configuring DNS NAPTR and/or SRV resource records in accordance with Section 16.1.2.1, the SP-SSE SHOULD indicate support for TLS.	SHOULD
REQ24393	16.2	The SP-SSE MUST be configured with a verifiable digital certificate to secure a TLS session.	MUST
REQ24394	16.2	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
REQ24395	16.4	SIP-PBXes that require timely detection of SIP peer failure MAY use any of these mechanisms as keep-alives:.	MAY
REQ24395.1	16.4	Sending an OPTIONS request periodically,	MAY
REQ24395.2	16.4	Sending a carriage return/line feed periodically (TCP only - Note: this is a unidirectional CR/LF with no application layer acknowledgement. This can generate TCP resets if the SIP peer fails)	MAY
REQ24396	16.4	SIP-PBXes that support one of these mechanisms MUST also support a mechanism that allows the keep- alive interval to be configured.	MUST
REQ24397	16.5	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
REQ24398	16.6	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
REQ24399	6	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 NOT

REQ24404	16.1.2.1	The SP-SSE MUST be reachable through a publicly-accessible DNS server.	MUST
REQ24405	16.1.2.1	The DNS server SHOULD provide a DNS interface that supports NAPTR resource records and MUST provide a DNS interface that supports SRV resource records.	SHOULD
REQ24407	16.1.2.2	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 ,and except when a TLS connection already exists, MUST use one of these mechanisms:	MUST
REQ24408	16.1.2.2	A configured SIP-PBX signaling address SHOULD be in the form of a hostname that can be resolved through DNS A/AAAA resource records, rather than an IP address (see additional guidance in Section 17.1).	SHOULD
REQ24409	16.1.2.2	When a TLS connection already exists, the SP-SSE MUST reuse that TLS connection for all SIP messages.	MUST
REQ24410	16.2	Both SIP-PBX and SP-SSE SHOULD verify the status of the certificate received during TLS establishment. Status verification steps include checking the status of all certificates in the chain using certificate revocation lists (CRLs) [RFC 5280] or Online Certificate Status Protocol (OCSP) [RFC 2560].	SHOULD
REQ24411	10.1.3	If no caller identity is available and privacy has not been requested, the SP-SSE SHOULD send a URI containing a host portion with a top level domain of ".invalid", as shown below. unavailable@unknown.invalid	SHOULD
REQ24412	10.1.4	If the "P-Asserted-Identity" header field is to be included, then the SP-SSE SHOULD also include display name information along with the SIP or Tel URI in the "P-Asserted-Identity" header field, if the display name is available and has not been restricted for delivery.	SHOULD

REQ24413	14.6	Media Endpoints SHOULD locally generate call progress tones or announcements, or other suitable indications, when the response to an INVITE request indicates call failure.	SHOULD
REQ24414	15.4.1.7	Any 4xx, 5xx or 6xx-class response to a REGISTER request not explicitly identified above SHOULD be treated in a similar manner as Section 15.4.1.1 unless it can automatically be resolved by the SIP-PBX internally - i.e., unless it is part of an explicit negotiation mechanism or procedure.	SHOULD
REQ24415	15.4.1.7	It SHOULD be treated as a delivery failure with a maximum retry interval of 960 seconds (16 minutes), unless a longer "Retry-After" header field is specified.	SHOULD
REQ24416	15.6.1	In order to avoid unnecessary challenges, the SIP-PBX SHOULD include its authentication credentials using the current nonce in each subsequent request that allows authentication credentials to be sent to the SP-SSE.	SHOULD
REQ24417	8.1	[RFC 3261] Section 26.3.1 states: Proxy servers, redirect servers, and registrars SHOULD possess a site certificate whose subject corresponds to their canonical hostname.	SHOULD
REQ24418	8.1	It is also RECOMMENDED that SIP-PBX and SP-SSE implementations be able to provide a certificate with either a URI or DNS name for backward compatibility.	RECOMMEND
REQ24419	15.5	, and if that fails MUST try an alternative SP-SSE if available, in accordance with Section 15.4.1.1.	MUST
REQ24421a	16.1.2.2	DNS: SP-SSE utilizes DNS NAPTR and SRV queries for the pre-configured domain name of the Enterprise Network, as described in [RFC 3263], to determine the IP address, transport protocol, and port number of the SIP-PBX(s) associated with the Enterprise Network's domain name.	MUST
REQ24421b	16.1.2.2	Configuration: The mapping of the Enterprise FQDN to the SIP-PBX signaling address/port and transport protocol is statically configured in the SP-SSE.	MUST

REQ24422	15.4.1.6	and MUST honor the value as follows: if the value is 32 seconds or less,	MUST
REQ24423	15.4.1.6	it MUST wait the requested time and retry the request to the same SP-SSE; if the value is larger,	MUST
REQ24424	15.4.1.6	it MUST remember the value for that SP-SSE address instance, and try any alternate SP-SSE addresses it can.	MUST
REQ24424a	15.4.1.6	If an alternate SP-SSE can be successfully reached and Registration succeeds through the alternate, the SIP-PBX MAY discard the "Retry-After" value of the original.	MAY
REQ24424b	15.4.1.6	Otherwise, it MUST wait to reattempt registration to the original SP-SSE for the "Retry-After" interval.	MUST
RFC4607		First, it MUST contain an option tag of "gin" in both a "Require" header field and a "Proxy-Require “header field.	MUST
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
RFC4609		A URI with a "bnc" parameter MUST NOT contain user portion.	MUST
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
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
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

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
RFC4614		A registrar compliant with this document MUST support the path mechanism defined in RFC 3327 [10].	MUST
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
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
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
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
RFC4607		First, it MUST contain an option tag of "gin" in both a "Require" header field and a "Proxy-Require “header field.	MUST
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
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