Detailed End Point IVT Test Plan and Report for ALGO 8201 SIP Intercom and CUCM 12.0

Test Result	PASS
Test Date	19th December 2018
Product Name	Algo 8201 Sip Intercom
Product Version # (must be generally available)	1.6.1_rc1
Unified Communications Manager Version	12.0
Product Type(Billing, Voice Recording, phone apps etc):	Endpoint
API/Protocol(s) Used	SIP
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1 Introduction

This document is the detailed Interoperability Verification Test Plan and Report for CUCM 12.0 and Algo 8301 SIP Intercom.

1.2 Entry Criteria

Before testing can begins 3rd party partner shall run this entire test plan in their lab and verify that results. If there are any test cases not supported, not applicable or are not successful, the partner should consult with tekVizion test team. Once testing has been initiated, the device under test is considered frozen for certification testing purposes. No software/firmware load can be changed during the testing period. However, configuration can be modified to accommodate testing.

1.3 Exit Criteria

To be deemed certified as configured, the devices under test should have zero severity 1 and severity 2 defects and up to two severity 3 defects detected.

If a severity 1 or 2 failure occurs, irrespective of who is responsible for the problem (Cisco or the 3rd party product) the testing is considered unsuccessful.

Table 1 Defect Severity Level Description

Severity	Description
1	Catastrophic - Common circumstance causes the entire system or a major subsystem to stop working affects other areas/devices no workaround
2	Severe- Important functions are unusable does not affect other areas/devices no workaround
3	Moderate - Very unusual circumstances cause failure minor feature doesn't work at all there's a low impact workaround

If any tests fail, the configuration will be verified to resolve the issue. If the issue cannot be resolved, the tester will attempt to continue testing if possible. If the testing cannot proceed without this problem being resolved, the testing is considered complete and the devices under test are deemed not certified.

The following procedures are followed when testing fails:

- Preliminary analysis is made to determine the source of the problem. If the problem is related to a device under test, then the problem is reported to that partner. If the problem is deemed Cisco related, the problem will be reported to Cisco, but the partner is responsible to open a TAC case with Cisco developer services. Partner should provide the TAC case number to the test team so they can document it in the report.

- If testing can continue past this failure, the other test cases will be tested and verified for pass or fail. If the testing cannot progress past this problem, testing will be halted and a final test report submitted to Partner and Cisco.
- All problems and resolutions encountered during testing are documented in the final test report.
- If a severity 1 failure occurs, irrespective of whom is responsible for the problem (Cisco or the 3rd party product), the testing is considered unsuccessful.

Any deviations of the test execution or problem acceptance are documented in the test report.

Note: The Cisco approval process may increase/decrease the severity level of the defect after the test cycle, if considered necessary.

2 Product Overview

The 8201 is a one-piece PoE solution as an alternative to the popular 8028 SIP Door phone. The intercom can be surface or flush mounted, fitting in a standard double gang electrical box. The 8201 can be used as easily indoors as out. The device offers an outdoor-rated plastic enclosure, stainless steel faceplate (brass is available for purchase as a separate item), conformal coated electronics, and tactile backlit call button for complete weather protection and long life in any climate.

3 Executive Summary

The following summarizes tekVizion's findings:

- **Test Case Failures:**
    - None
- **Test Cases Not Supported**
    - None
- **Test Cases Not Applicable:**
    - 5.2.12 Functional Test: Conference Call
- **Test Cases that were Not Executed:**
    - None

- **Observations**
    - Algo 8201 SIP Intercom is manually registered with CUCM 12.0 using the Digest Authentication.
    - G 722 codec is supported in the DUT by enabling the G.722 in the web interface of the DUT.
    - DUT do not support conference so it can't be a part of conference scenario
    - DUT can be a part of call functions like call Hold, Resume, and Transfer etc. But these functions cannot be performed on the DUT.
    - DUT doesn't have a button to End/Reject the call.

3.2 Items Not Tested

Features that are specific to the internals of the 3rd party product or any features not listed will not be tested.

- All test included in this test plan were executed unless otherwise noted.

3.3 Assumptions

- Interoperability of 3rd party products – Testing will cover only features in 3rd party products that result in events to and/or from the CUCM or specified PSTN gateway.
- Call Processing – PSTN interface and Cisco SIP call processing traffic for all testing (excluding manual sampling run during traffic) may be generated using simulators.

4 Test Environment

4.2 Administration, Testing and Debugging tools

Tools used/required – Identify any tools required by 3rd party (partner under test). Also add Trace and Debug settings here.

Table 2 Administration, Testing and Debugging Tools

Product Name	Version	Type	Purpose	Units	Notes
Test Tools					
None					
3rd Party Tools					
None					
Debug Tools					
FileZilla Server	0.9.60	File server	For retrieving CDR from CUCM	1	Lab Provided

4.3 Equipment Requirements

Table below identifies all equipment/versions used for in this IVT.

Table 3 Equipment and Product Information

Product	Version	Type	Purpose	Units	Notes
Cisco Products					
CUCM	12.0	Call Server	Call Processing (For Local and Remote)	2	Lab Provided
CUPS	12.0	Presence Server	Presence and Messaging	1	Lab Provided

Product	Version	Type	Purpose	Units	Notes
CIPC	8.6.1.0	Soft Phone	Endpoint	2	Lab Provided
Cisco Jabber	12.0	Soft Phone	Endpoint	1	Lab Provided
3rd Party Products					
Algo 8201 SIP Intercom	1.6.1_rc1	Intercom	Endpoint	1	Customer provided

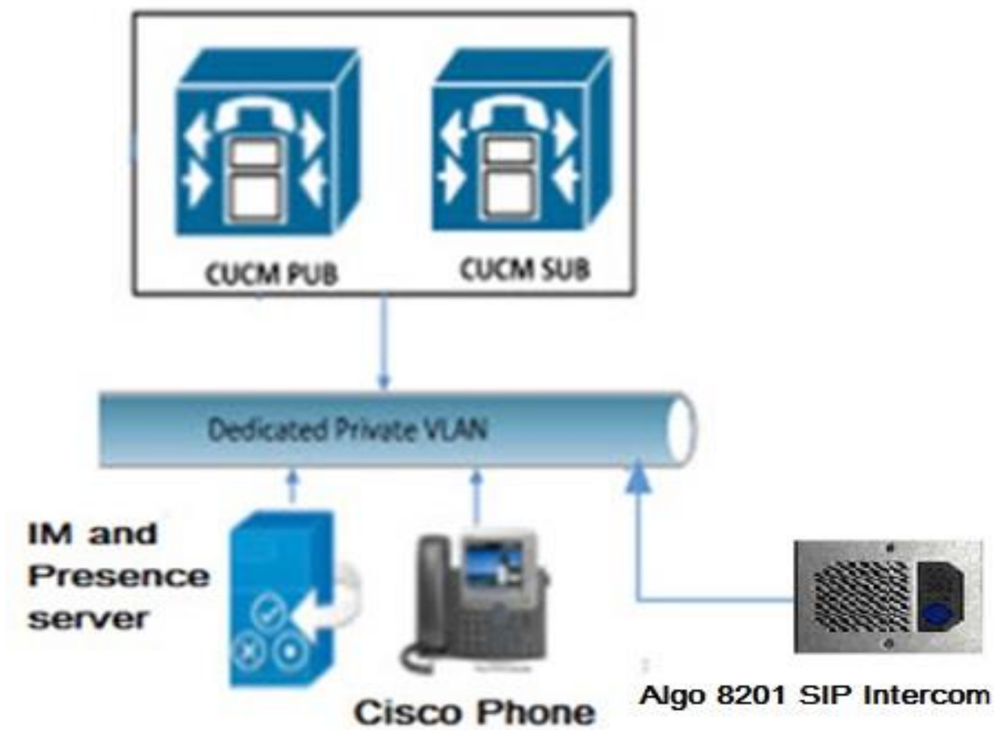
4.4 Cisco Phones

Table 4 Cisco Phones Information

Cisco Phone Model	Phone Firmware Version	Protocol	POE	Units	Notes
9951	sip9951.9-4-2SR3-1	SIP	POE	1	Lab provided
9971	sip9971.9-4-2SR3-1	SIP	POE	1	Lab provided
7942	SIP41.9-4-2SR3-1S	SIP	POE	2	Lab provided
7965	SCCP42.9-4-2SR3-1S	SCCP	POE	1	Lab provided
8845	sip8845_65.12-0-1-11	SIP	POE	1	Lab provided

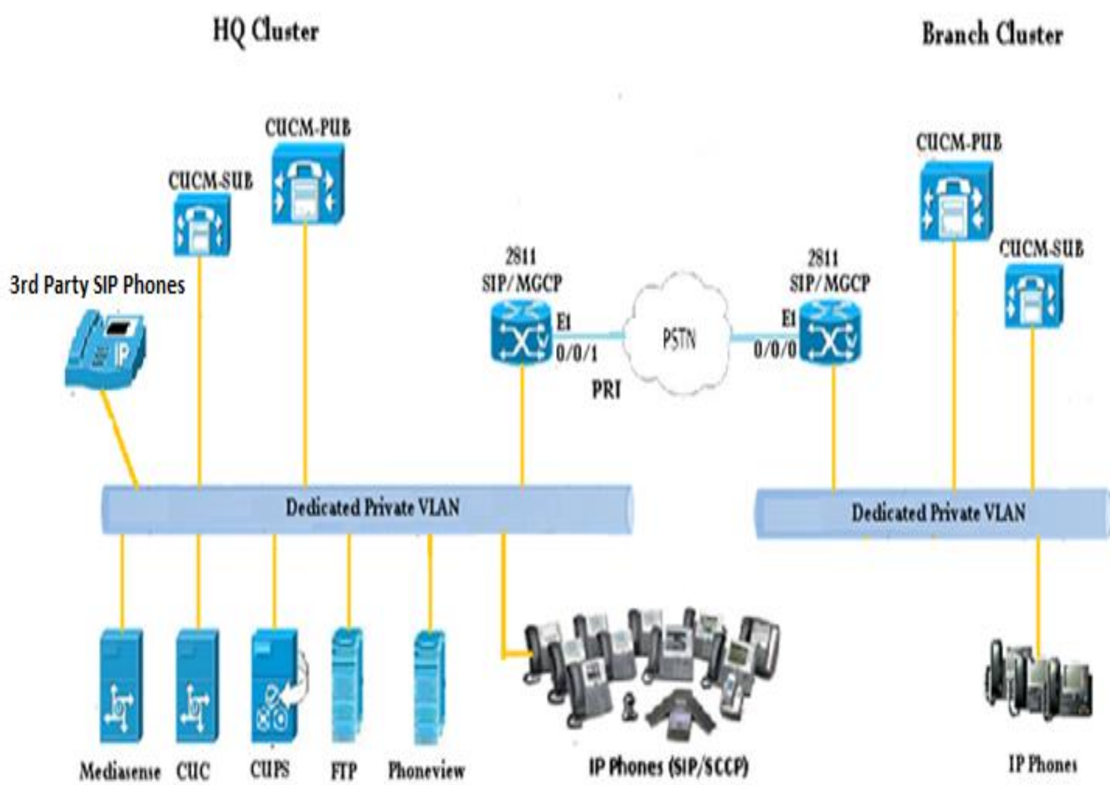
4.5 Deployment Architecture

Figure 1 – Deployment Architecture



4.6 Test Environment Architecture

Figure 2 – Test Environment Architecture



5 Test Cases

This section details the tests that will be performed during the testing period.

Result	Description
PASS	The test case passed with no exceptions
FAIL	The test case failed – details of the failure are noted in the Comments column
N/A	The test case is not applicable to the product under test. Justification must be provided in the Comments column.
N/S	Not supported. While the feature tested by this test case generally would be considered a standard feature for this product category, this specific product (or this specific release) does not support the feature.
N/T	Not tested. The feature is supported by the product under test, but external factors (lab configuration, e.g.) prevented execution of the test. Justification must be provided in the Comments column.

5.1 Phase 1 Installation and Configuration Tests

Test is focused on ensuring that the 3rd party product (DUT) is registered with Call Manager successfully.

5.1.1 Register DUT to Cisco Call Manager

Test Case Details	
Title	Register DUT to Cisco Call Manager
Description	Verify 3rd party endpoints (DUT) are registered in Call Manager successfully
Test Setup	
Procedure	1. Connect the DUT in local CUCM cluster 2. Go to DUT(s) settings to verify network and load information 3. Associate end users to DUT(s) 4. Register DUT in local CUCM cluster with DN's
Expected Results	• CUCM Administration GUI display the DUT(s) • DUT is in "Registered" state • DUT(s) have a DN assigned • DUT(s) network data is correct: (VLAN, DNS, DHCP, TFTP, CUCM) • DUT(s) Phone Load version is correct • Users associated to DUT(s) respectively • DUT(s) registered to Local CUCM with assigned DN's
Observations	PASS

5.2 Phase 2 Functional Test

These tests test the various features of the 3rd party product and its various components. This involves the testing of the product against the Application note and IVT questionnaire requirements to ensure that it functions reliably and consistently in a manner that meets the requirements.

Note: In DUT give Dial to extension as always Local SIP phone1 DN for all the test cases except Hunt group tests. For Hunt group test cases give extension as hunt group pilot number.

5.2.1 Intra-Cluster Calls

Test Case Details	
Title	Intra-Cluster Calls
Description	Verify intra-cluster calls between DUT, SCCP and SIP endpoints
Test Setup	Local CUCM • DUT

	• SCCP: Local SCCP phone1 • SIP: Local SIP phone1
Procedure	1. Local SIP phone1 dials DUT extension and DUT answers the call 2. Check for the audio. 3. Local SIP phone1 on-hook after 30s 4. DUT press INTERCOM button and Local SIP phone1 answers the call 5. Check for the audio. 6. Local SIP phone1 on-hook after 30s 7. Local SCCP phone1 dials DUT extension and DUT answers the call 8. Local SCCP phone1 on-hook after 30s. 9. Check for the audio.
Expected Results	• 3 calls establish with 2-way audio path • 3 calls terminate normally
Observations	PASS

5.2.2 Inter-Cluster Call

Test Case Details	
Title	Inter-Cluster Call
Description	Verify inter-cluster calls between DUT(s), SCCP and SIP endpoints
Test Setup	Local CUCM • DUT(s):DUT Remote CUCM ➢ SCCP: Remote SCCP phone1 ➢ SIP: Remote SIP phone1
Procedure	1. Remote SIP phone1 dials DUT and DUT answers the call 2. Check for the audio. 3. Remote SIP phone1 on-hook after 30s. 4. Remote SCCP phone1 dials DUT and DUT answers the call 5. Check for the audio. 6. Remote SCCP phone1 on-hook after 30s. 7. Calling & Called party release calls alternatively 8. Retrieve CDR from CUCM

	9. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• 2 calls establish with audio path • 2 calls terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	PASS

5.2.3 Off-Net Calls

Test Case Details	
Title	Off-Net Calls
Description	Verify basic calls between DUT(s) and PSTN endpoints
Test Setup	• Local CUCM ➔ DUT(s):DUT • PSTN Phone➔ PSTN (SIP);
Procedure	1. PSTN dials DUT and DUT answers the call 2. PSTN on-hook after 60s 3. Retrieve CDR from CUCM Server 4. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call established with audio path • Calling and Called Parties hear ring-back and ring tone • Call terminate normally • 1 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	PASS

5.2.4 Disable outbound call in DUT

Test Case Details	
Title	Disable outbound call in DUT
Description	Verify inter-cluster calls between DUT, SCCP and SIP endpoints
Test Setup	Local CUCM • DUT

	• Local CUCM : SIP: Local SIP phone1
Procedure	1. DUT press INTERCOM button and Local SIP phone1 answers the call 2. Check for the audio. 3. Local SIP phone1 on-hook after 30s 4. Disable 'answer inbound calls ' in DUT settings 5. Local SIP phone1 dials DUT and hears busy tone 6. Local SIP phone1 on-hook after 5s. 7. Retrieve CDR from CUCM 8. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• 1 call is establish with audio path • Local SIP phone1 hears a busy tone • 2 calls terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	PASS

5.2.5 Multiple calls handling by DUT

Test Case Details	
Title	Multiple calls handling by DUT
Description	Verify Multiple call handling between DUT, SCCP and SIP endpoints
Test Setup	Local CUCM • DUT • Local CUCM : SIP: Local SIP phone1 SCCP: Local SCCP phone1
Procedure	1. DUT press INTERCOM button and Local SIP phone1 answers the call 2. Check for the audio. 3. Local SCCP phone1 dials DUT and hears busy tone 4. Local SCCP phone1 on-hook after 5s. 5. Local SIP phone1 on-hook after 50s. 6. Retrieve CDR from CUCM

	7. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• 1 call is establish with audio path • Local SCCP phone1 hears a busy tone • 2 calls terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	PASS

5.2.6 Functional Test: CFA

Test Case Details	
Title	Functional Test: CFA
Description	Verify "CFA" calls between DUT(s), SCCP, SIP and PSTN endpoints
Test Setup	Local CUCM • DUT • SCCP: Local SCCP phone1 • SIP: Local SIP phone1 Remote CUCM • SIP: Remote SIP phone1 PSTN Phone: PSTN Enable CFA for DN(s): · Device➔Phone➔Local SIP phone1➔CFA➔Local SCCP phone1 (SCCP) · Device➔Phone➔Local SCCP phone1➔CFA➔DUT
Procedure	1. DUT press Intercom Button (Local SIP phone1 DN), Local SCCP phone1 answers and Local SCCP phone1 on-hook after 30s 2. Remote SIP phone1 dials Local SCCP phone1, DUT answers and Remote SIP phone1 1 on-hook after 30s 3. Retrieve CDR from CUCM 4. Check Calling, Called, Duration, Origination & Termination Cause Codes

Expected Results	• Call forward to Local SCCP phone1 and phone rings • Call establish between DUT & Local SIP phone1 with 2-way audio • Call terminated normally • Call forward to DUT and phone rings • Call establish between Remote SIP phone1 & DUT with 2-way audio • Call terminate normally • 4 CDR(s) retrieved • Selected CDR(s) fields match calls
Observations	PASS

5.2.7 Functional Test: CFNA

Test Case Details	
Title	Functional Test: CFNA
Description	Verify "CFNA" calls between DUT(s), SCCP and SIP endpoints

Test Setup	• Local CUCM
	  · DUT(s):DUT
	  · SCCP: Local SCCP phone1; SIP: Local SIP phone1;
	• Remote CUCM
	  · SCCP: Remote SCCP phone1
	• Voicemail and Call Waiting disabled for all DN(s)
	Device➔Phone➔DN➔Line➔
	  · Voicemail➔No Voicemail
	  · Call Waiting➔Max. Calls➔1; Busy Trigger➔1;
	
	Enable CFNA for DN(s):
	· Device➔Phone➔Local SIP phone1➔CFNA➔Local SCCP phone1 (SCCP)
	· Device➔Phone➔Local SIP phone1➔CFNA➔DUT (DUT)

Procedure	1. DUT press Intercom Button (Local SIP phone1 DN), Local SCCP phone1 answers 2. Local SCCP phone1 on-hook after 30s 3. Remote SCCP phone1 dials Local SIP phone1, Local SIP phone1 does not answer and DUT answers 4. Remote SCCP phone1 on-hook after 30s 5. Retrieve CDR from CUCM

	6. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call forward to Local SCCP phone1 after ring timeout • Call establish between DUT & Local SCCP phone1 with 2-way audio • Call terminated normally • Call forward to DUT after ring timeout • Call establish between Remote SCCP phone1 1 & DUT with 2-way audio • Call terminated normally • CDR(s) retrieved • Selected CDR(s) fields match calls
Observations	PASS

5.2.8 Functional Test: CFB

Test Case Details	
Title	Functional Test: CFB
Description	Verify "CFB" calls between DUT(s), SCCP and SIP endpoints
Test Setup	• Local CUCM · DUT(s):DUT; SCCP: Local SCCP phone1,SCCP Phone2 · SIP: Local SIP phone1; • Remote CUCM SCCP: Remote SCCP phone1; SIP: Remote SIP phone1; • Voicemail and Call Waiting disabled for all DNs Enable CFB for DN(s): · Device➔Phone➔DUT➔CFB➔Local SCCP phone1 · Device➔Phone➔Local SIP phone1➔CFB➔Local SCCP phone1 2
Procedure	1. Remote SIP phone1 dials DUT and DUT answers the call 2. Local SIP phone1 dials the DUT, Local SCCP phone1 answer the call. 3. Remote SIP phone1 goes on hook. 4. Local SIP phone1 goes on hook. 5. Remote SCCP phone1 dials Local SIP phone1 and Local SIP phone1 answers

	6. DUT press Intercom Button (Local SIP phone1 DN), Local SCCP phone1 answers and Local SCCP phone12 on-hook after 30s 7. Remote SCCP phone1 goes on hook. 8. Local SCCP phone1 goes on hook. 9. Retrieve CDR from CUCM 10. Check Calling, Called, Duration, origination & termination cause codes matches the calls
Expected Results	• Call establish between Remote SIP phone1 & DUT with 2-way audio • Call forward to Local SCCP phone1 1 and phone rings • Call establish between Local SIP phone1 1 & Local SCCP phone1 with 2-way audio • Remote SIP phone1 end the call. • Local SCCP phone1 end the call. • Call establish between Remote SCCP phone1 & Local SIP phone1 1with 2-way audio • Call forward to DUT and DUT answer the call. • Call establish between Local SCCP phone1 & DUT with 2-way audio • Remote SIP phone1 end the call. • Local SCCP phone1 end the call. • 4 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	PASS

5.2.9 Functional Test: Hold & Resume

Test Case Details	
Title	Functional Test: Hold & Resume
Description	Verify “Hold & Resume” calls between DUT(s), SIP, SCCP and PSTN endpoints
Test Setup	• Local CUCM➔DUT(s):DUT ; SIP: Local SIP phone1; Remove all CFA, CFNA & CFB settings on DN(s) used in previous test cases
Procedure	1. DUT press INTERCOM button and Local SIP phone1 answers the call and Local SIP phone1 hits “Hold” after 20s

	2. Local SIP phone1 hits “Resume” after 20s and Local SIP phone1 on-hook after 30s 3. Local SIP phone1 dials DUT, DUT answers incoming call and Local SIP phone1 on-hold 4. Local SIP phone1 hits “Resume” after 60s and DUT on-hook after 30s 5. Retrieve CDR from CUCM 6. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between Local SIP phone1 & DUT with 2-way audio • DUT is On-Hold (MOH) • Call resume between Local SIP phone1 & DUT • Call terminate normally • Call establish between Local SIP phone1 & DUT with 2-way audio • DUT is On-Hold (MOH) • Call resume between Local SIP phone1 1 & DUT • Call terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	PASS

5.2.10 Functional Test: Blind Transfer

Test Case Details	
Title	Functional Test: Blind Transfer
Description	Verify “Blind Transfer” calls between DUT(s), SIP and SCCP endpoints
Test Setup	• Local CUCM➔DUT(s):DUT; SCCP: Local SCCP phone1; SIP: Local SIP phone1; • PSTN: PSTN
Procedure	1. Local SIP phone1 dials Local SCCP phone1, Local SCCP phone1 answers and hits “Transfer” after 30s 2. Local SCCP phone1 dials DUT, Local SCCP phone1 hits “Transfer” and Local SCCP phone1 is on-hook 3. DUT answer the call 4. Local SIP phone1 goes on-hook after 60s

	5. DUT press INTERCOM button and Local SIP phone1 answers the call and Local SIP phone1 hits "Transfer" after 30s. 6. Local SIP phone1 dials Local SCCP phone1 and hits "Transfer" and DUT is on-hook 7. Local SCCP phone1 answer the call. 8. Local SCCP phone1 goes on-hook after 60s 9. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between Local SIP phone1 & Local SCCP phone1 1 with 2-way audio • Local SIP phone1 is On-Hold (MOH) • Local SCCP phone1 blind transfer to DUT with 2-way audio path • All calls terminate normally • Call establish between DUT & Local SIP phone1 with 2-way audio • DUT is On-Hold (MOH) • Local SIP phone1 blind transfer to Local SCCP phone1 • All calls terminate normally • 4 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	PASS

5.2.11 Functional Test: Consult Transfer

Test Case Details	
Title	Functional Test: Consult Transfer
Description	Verify "Consult Transfer" calls between DUT(s), SIP, SCCP and PSTN endpoints
Test Setup	• Local CUCM➔DUT(s):DUT; SCCP: Local SCCP phone1; SIP: Local SIP phone1; • PSTN: PSTN

Procedure	1. Local SIP phone1 dials Local SCCP phone1 , Local SCCP phone1 answers and hits “Transfer” after 30s 2. Local SCCP phone1 dials DUT, And DUT answers the call 3. Local SCCP phone1 hits “Transfer” and Local SCCP phone1 is on-hook 4. Local SIP phone1 goes on-hook after 60s 5. PSTN dials Local SCCP phone1, Local SCCP phone1 answers and hits “Transfer” after 30s 6. Local SCCP phone1 dials DUT, and DUT answer the call. 7. Local SCCP phone1 hits “Transfer” and Local SCCP phone1 is on-hook 8. PSTN goes on-hook after 60s 9. Check the Calling, Called, Duration, Origination & Termination Cause Codes 10. Retrieve CDR from CUCM 11. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between Local SIP phone1 & Local SCCP phone1 with 2-way audio • Local SIP phone1 is On-Hold (MOH) • Call establish between Local SCCP phone1 & DUT with 2-way audio • Local SCCP phone1 transfer to DUT with 2-way audio path • All calls terminate normally • Call establish between PSTN & Local SCCP phone1 with 2-way audio • PSTN is On-Hold (MOH) • Call establish between Local SCCP phone1 & DUT with 2-way audio • Local SCCP phone1 transfer to DUT • Local SCCP phone1 hears reorder tone • All calls terminate normally • 6 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	PASS

5.2.12 Functional Test: Conference Call

Test Case Details	
Title	Functional Test: Conference Call
Description	Verify Conference call between DUT(s), SIP, SCCP and PSTN endpoints
Test Setup	• Local CUCM➔DUT(s):DUT; SCCP: Local SCCP phone1; SIP: Local SIP phone1; • Service parameter: Drop Ad Hoc Conference ➔ Never (Default) • Media Resource Group (MRG) & Media Resource Group List (MRG_L) • Assign Media Resource: System➔Device Pool➔ep_pool➔Media Resource Group List➔MRG_L
Procedure	1. Local SCCP phone1 dials Local SIP phone1, Local SIP phone1 answers and DUT hits "Conference" after 30s 2. Local SIP phone1dials DUT, DUT answers 3. Local SIP phone1 hits "Conference" after 30s 4. Local SCCP phone1 goes on-hook after 60s 5. Local SIP phone1goes on-hook after 30s 6. PSTN dials Local SIP phone1, Local SIP phone1 answers and hits "Conference" after 30s 7. Local SIP phone1dials DUT, DUT answers 8. Local SIP phone1 hits "Conference" after 30s 9. PSTN goes on-hook after 60s 10. Local SIP phone1goes on-hook after 30s 11. Retrieve CDR from CUCM 12. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between Local SCCP phone1 & Local SIP phone1 with 2-way audio • Local SCCP phone1 is On-hold (MOH) • Local SIP phone1 is conference-in • 3 parties in conference call with 3-way audio • PSTN left conference. Local SIP phone1 & DUT connect directly • All calls terminate normally • Call establish between PSTN & Local SIP phone1 with 2-way audio • PSTN is On-Hold (MOH) • PSTN, Local SIP phone1 & DUT is conference-in

	• All 3 parties in conference call with 3-way audio • PSTN leave conference. Local SIP phone1 & DUT connect directly • All calls terminate normally • CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	N/A This feature is not applicable for the DUT.

5.2.13 Functional Test: Jabber for Windows

Test Case Details	
Title	Functional Test: Jabber for Windows
Description	Verify Jabber calls originating & terminating to DUT(s) endpoints (Jabber for Windows)
Test Setup	• Local CUCM→DUT(s):DUT SCCP: Local SCCP phone1; • Jabber for Windows (Device→Phone→Add New→CSFUSER1:DN:1922; End User:juser01/123456) • Windows PC with Jabber clients installed
Procedure	1. 1922 dials DUT (Duration=30s) 2. 1922 dials Local SCCP phone1 (Duration=30s) 3. Local SCCP phone1 dials DUT (Duration=30s) 4. Calling and Called party goes on-hook alternatively 5. Retrieve CDR from CUCM 6. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• 3 calls establish with 2-way audio • 3 calls terminate normally • 3 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	PASS

5.2.14 Functional Test: IP Communicator

Test Case Details	
Title	Functional Test: IP Communicator
Description	Verify IP Communicator calls originating & terminating to DUT(s) endpoints
Test Setup	• Local CUCM➔DUT(s):DUT, SIP: Local SIP phone1; CIPC:1940; • Launch IP Communicator on a Windows PC
Procedure	1. 1940 dials DUT (Duration=30s) 2. 1940 dials Local SIP phone1 (Duration=30s) 3. Local SIP phone1 dials DUT(Duration=30s) 4. Calling and Called party goes on-hook alternatively 5. Retrieve CDR from CUCM 6. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• 3 calls establish with 2-way audio • 3 calls terminate normally • 3 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	PASS

5.2.15 Functional Test: Hunt Group

Test Case Details	
Title	Functional Test: Hunt Group
Description	Verify "Hunt Group" calls using DUT(s), SCCP, SIP and PSTN endpoints
Test Setup	• Local CUCM➔DUT; SCCP: Local SCCP phone1; SIP: Local SIP phone1; • Remote CUCM ➔ SCCP DN: Remote SCCP phone1; SIP DN: Remote SIP phone1; • PSTN: PSTN1 • Hunt Group Pilot 5000 (1st member-Local SCCP phone1; 2nd member-DUT;), Queuing flag enabled, max. waiting timer=60 secs,

Procedure	1. DUT press Intercom Button(HUNT group pilot number 3000), Local SCCP phone1 answers and Local SCCP phone1 on-hook after 60s 2. Local SIP phone1 dials Local SCCP phone1 and Local SCCP phone1 answers 3. PSTN1 dials 5000 and DUT answers 4. Local SIP phone1 goes on-hook after 60s 5. PSTN goes on hook 6. Retrieve CDR from CUCM 7. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call route to hunt group member DUT • Call establish between DUT & Local SCCP phone1 with 2-way audio • Call terminated normally • Local SIP phone1 and & Local SCCP phone1 members are busy • Call route to hunt group member DUT • Call establish between DUT& Local PSTN with 2-way audio • Call terminated normally • 5 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	PASS

5.2.16 Functional Test: Hunt Group

Test Case Details	
Title	Functional Test: Hunt Group
Description	Verify "Hunt Group" calls on DUT(s) when no members are available
Test Setup	• Local CUCM→DUT; SCCP: Local SCCP phone1; SIP: Local SIP phone1; • Remote CUCM →DUT:DUT3; SCCP DN: Remote SCCP phone1; SIP DN: Remote SIP phone1; • Hunt Group Pilot 5010 (1st member-DUT2), Queuing flag enabled, max. waiting timer=60 secs, • Call Routing→Route/Hunt→Hunt Pilot→5000→Route call to this destination→Local SCCP phone1; • Call Routing→Route/Hunt→Hunt Pilot→5012→Route call to this destination→Local SIP phone1;
Procedure	1. Local SIP phone1 stays off-hook to make it unavailable

	2. DUT press Intercom button (hunt pilot) , Local SCCP phone1 answers and Local SCCP phone1 on-hook after 60s 3. Local SIP phone1 dials 5012, DUT answers and Local SIP phone1 on-hook after 60s 4. Retrieve CDR from CUCM 5. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• HG member-DUT2 is unavailable • Hunt Group has no members available • Call route to hunt group alternate destination Local SCCP phone1 • Call establish between DUT & Local SCCP phone1 with 2-way audio • Call terminate normally • Call route to hunt group alternate destination DUT • Call establish between DUT & Local SIP phone1 with 2-way audio • Call terminated normally • Call terminated normally • 4 CDR(s) retrieved • Selected fields in CDR match calls
Observations	PASS

5.2.17 Functional Test: Hunt Group

Test Case Details	
Title	Functional Test: Hunt Group
Description	Verify “Hunt Group” calls on DUT(s) when maximum queue length exceeded
Test Setup	• Local CUCM➔DUT; SCCP: Local SCCP phone1; SIP: Local SIP phone1; • Hunt Group Pilot 5015 (1st member-Local SIP phone1), Queuing flag enabled, max. waiting timer=60 secs, Route call to Destination disabled; Max. # of callers in queue=2;
Procedure	1. DUT press intercom button (hunt pilot number 5015), Local SIP phone1 answers 2. Local SCCP phone1 dials 5015 3. DUT goes on-hook after 200 secs 4. Retrieve CDR from CUCM

	5. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call route to hunt group member Local SIP phone1 • Call establish between DUT & Local SIP phone1 with 2-way audio • Local SCCP phone1 waiting in queue • Maximum number of callers in queue exceeded • Maximum wait timer exceeded 60s • Local SCCP phoen1 was not terminated to hunt group • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	PASS

5.3 Phase 3 Negative Tests

These tests are executed to determine the ability of the impact on calls, the CUCM and the 3rd party DEVICE. Testing robustness of the application through hardware and software fault insertion i.e. Failover/fallback.

5.3.1 Negative Test: PUB Failure

Test Case Details	
Title	Negative Test: PUB Failure
Description	Verify a PUB failure should not affect stable or transient calls on DUT(s)
Test Setup	• Local CUCM➔DUT SIP: Local SIP phone1;
Procedure	1. Local SIP phone1 dials DUT, DUT answers 2. Local SIP phone1 ends the call 3. Local SIP phone1 dials DUT, DUT answers 4. Access CUCM-PUB server via SSH (Local Cluster) 5. Enter CLI: utils system restart <CR> yes 6. Call terminates normally. 7. After CUCM-PUB restarted, Local SIP phone1 dials DUT, DUT answers 8. Called party goes on-hook 9. Retrieve CDR from CUCM

	10. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between Local SIP phone1 & DUT with 2-way audio • Call establish between Local SCCP phone1 & Local SIP phone1 2 with 2-way audio • CUCM-PUB is restarted • Stable calls not impacted by PUB restart • Call establish between Local SIP phone1 & DUT with 2-way audio • Transient calls not impacted by PUB restart • All calls terminate normally • CUCM-PUB is in-service • All calls successful after PUB failure recovery • 3 CDR(s) retrieved • Selected fields in CDR matches calls
Observations	PASS

5.3.2 Negative Test: SUB Failure

Test Case Details	
Title	Negative Test: SUB Failure
Description	Verify a PUB failure should not affect stable or transient calls on DUT(s)
Test Setup	Local CUCM➔:DUT ; SIP: Local SIP phone1;
Procedure	1. Local SIP phone1dials DUT, DUT answers 2. Local SIP phone1 ends the call 3. Local SIP phone1 dials DUT, DUT answers 4. Access CUCM-SUB server via SSH (Local Cluster) 5. Enter CLI: utils system restart <CR> yes 6. Call terminates normally. 7. After CUCM-SUB restarted, Local SIP phone1 dials DUT, DUT answers 8. Called party goes on-hook 9. Retrieve CDR from CUCM 10. Check Calling, Called, Duration, Origination & Termination Cause Codes

Expected Results	• Call establish between Local SIP phone1 & DUT with 2-way audio • Call establish between Local SCCP phone1 & Local SIP phone1 2 with 2-way audio • CUCM-PUB is restarted • Stable calls not impacted by PUB restart • Call establish between Local SIP phone1 & DUT with 2-way audio • Transient calls not impacted by PUB restart • All calls terminate normally • CUCM-PUB is in-service • All calls successful after PUB failure recovery • 3 CDR(s) retrieved • Selected fields in CDR matches calls
Observations	PASS

5.3.3 Negative Test: Phone Network Failure

Test Case Details	
Title	Negative Test: Phone Network Failure
Description	Verify DUT(s) recovers from a network failure
Test Setup	• Local CUCM→DUT • SCCP: SIP: Local SIP phone1;
Procedure	1. Local SIP phone1 dials DUT, DUT answers 2. Unplug network cable from device DUT 3. Restore the network cable after 60s 4. Local SIP phone1 dials DUT, DUT answers 5. DUT goes on-hook after 60s 6. Retrieve CDR from CUCM 7. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between Local SIP phone1 & DUT with 2-way audio • Network failure reported on device DUT • Stable call drops • Device DUT re-registers after network cable restored

	• Network Data: DNS, DHCP, TFTP, CUCM, VLAN, Load ID are restored on device • Call establish between Local SIP phone1 & DUT with 2-way audio • Call terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	PASS

5.3.4 Negative Test: Phone Power Failure

Test Case Details	
Title	Negative Test: Phone Power Failure
Description	Verify DUT(s) recovers from a power failure
Test Setup	Local CUCM➔DUT; SIP: Local SIP phone1;
Procedure	1. Local SIP phone1 dials DUT, DUT answers 2. Remove power cable from DUT 3. Restore power cable after 60s 4. Local SIP phone1 dials DUT, DUT answers call 5. Local SIP phone1 goes on-hook after 60s 6. Retrieve CDR from CUCM 7. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between Local SIP phone1 & DUT with 2-way audio • DUT lost power • Stable call drops • Device DUT re-registers after power is restored • Network Data: DNS, DHCP, TFTP, CUCM, VLAN, Load ID are restored on device • Call establish between Local SIP phone1 & DUT with 2-way audio • Call terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	PASS

5.4 Phase 4 Miscellaneous Tests

These tests are executed to verify specific information about the third-party products provided by partners

5.4.1 Miscellaneous Test: Codec (G722)

Test Case Details	
Title	Miscellaneous Test: Codec (G722)
Description	Verify URI calls between DUT(s) & SIP endpoints for In-band Codec (G722)
Test Setup	• Local CUCM➔DUT(s): DUT. ; SIP: Local SIP phone1 ; • Go to System➔Region Information➔ Audio Codec Preference List➔ Add New➔ G722➔Select G722 Codec • Go to System➔Region Information➔ Region➔ Add New➔G722-Region➔G722 • Go to System➔Device Pool➔ Add New➔G722-dp➔Region➔G722-Region • Update DUT with device pool=G722
Procedure	1. Local SIP phone1 hits dials DUT 2. DUT answers call 3. Local SIP phone1 goes on-hook after 60s 4. DUT press intercom Button and Local SIP phone1 answer the call 5. Local SIP phone1 goes on-hook after 60s 6. Retrieve CDR from CUCM Server 7. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• 2 calls establish with 2 way audio for G722 codec • 2 calls terminate normally • Voice quality was good for codec type • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	PASS

5.4.2 Long Duration Calls

Test Case Details

Title	Long Duration Calls
Description	Verify long duration calls between DUT(s), SCCP, SIP and PSTN endpoints
Test Setup	Local CUCM➔DUT; SCCP: Local SCCP phone1; SIP: Local SIP phone1;
Procedure	1. Local SIP phone1 dials DUT, DUT answers (Duration: 1 Hr.) 2. Local SCCP phone1 dials DUT, DUT answers (Duration: 1 Hr) 3. DUT press Intercom button and Local SIP phone1 answers (Duration: 1 Hr) 4. Retrieve CDR from CUCM 5. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between Local SIP phone1 & DUT with 2-way audio • Call establish between Local SCCP phone1 & DUT with 2-way audio • All long duration calls were stable with 2-way audio • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	PASS

5.4.3 Miscellaneous Test: Cisco Phone Models

Test Case Details	
Title	Miscellaneous Test: Cisco Phone Models
Description	Verify calls and mid-call features between DUT(s) and various Cisco IP Phone Models
Test Setup	• Local CUCM➔DUT(s):DUT; SCCP: Local SCCP phone1; SIP: Local SIP phone1; • Cisco Phone Models: 8845, 7975, 9971,7842
Procedure	1. Cisco IP Phone dials DUT, DUT answers and Cisco IP Phone on-hook after 120s 2. Cisco IP Phone dials DUT , DUT answers and Cisco IP Phone hits “Hold” after 20s

	3. Cisco IP Phone hits "Resume" after 20s, Cisco IP Phone on-hook after 120s 4. Cisco IP Phone dials DUT, Cisco IP Phone hits "Transfer and Cisco IP Phone on-hook 5. Local SIP phone1 goes on-hook after 120s 6. Repeat steps 1-5 by replacing other Cisco phone models 7. Retrieve CDR from CUCM 8. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Calls establish between DUT & Cisco IP Phone • Call Hold/Resume between DUT & Cisco IP Phone • Blind Transfer between DUT & Cisco IP Phone • CDR(s) retrieved for all the calls • Selected fields in CDR(s) match calls
Observations	PASS