Detailed End Point IVT Test Plan and Report for Cisco Communications Manager and Ascom

Test Result	Pass
Test Date	September 9th, 2021
Product Name	Ascom IP-DECT IPBS2-A3/1B1
Product Version # (must be generally available)	11.4.4
Unified Communications Manager Version	14.0.1.10000-20
Product Type(Billing, Voice Recording, phone apps etc):	IP-DECT
API/Protocol(s) Used	SIP, DECT, TLS/SRTP
Partner IVT Contact Name:	Karl-Magnus Olsson
Partner IVT Contact Email:	Karl-Magnus.Olsson@ascom.com
IVT Lab Location (EMEA or US):	US
Partner Main Support Number	+46-31-559450 [1]
Partner Main Support Email	support@ascom.com [1]

1. Technical support for the Ascom Myco3 wireless handsets should initially be obtained through a local Ascom supplier or, only if unavailable, Ascom global technical support

Revision History

Revision	Author	Date	Comment
1.0	Ashwin George	June 10th-2021	Initial draft for Cisco review
1.1	Austin Morrison	Sep 9th-2021	Report with observations

Table of Contents

1 Introduction

This document is the detailed Interoperability Verification Test Plan and Report for Cisco Unified Communications Manager 14 and Ascom IPBS2-A3/1B1 11.4.4

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

Ascom IP-DECT - Proven technology in a new innovative package

The Ascom IP-DECT system combines the VoIP and the traditional wireless DECT solution in an innovative package offering a future proof, reliable and flexible wireless solution for on-site voice and personal alarms in a secure radio environment. The solution is based on standards like SIP that enables interoperability with most of the leading PBX suppliers today.

The scalability and flexibility of the Ascom IP-DECT offering is best in class based on the possibility to start small with just one or two IP-DECT access points and later scale up to 1000s of access points and 100s of IP-DECT gateways offering wireless services to 100 000s of users on different sites all over the globe. Installing and maintaining larger systems must be as simple as possible and by offering easy registration and easy installation, users and IP-DECT hardware components can be set up in a smooth and fast manner.

Ascom takes security seriously and therefore the IP-DECT system follows the DECT security stipulated by the DECT forum and ETSI as well as supporting SRTP in order to offer end to end VoIP security.

The Ascom IP-DECT System enables extended sales opportunities for on-site wireless communication. For example, IP-DECT enables cost efficient smaller systems, with modularity up to very large installations, feature rich distributed system installations (local office/site) and a wide variety of modern connection possibilities.

The Ascom IP-DECT System consists of the following components: IP-DECT Access Points with internal antennas and external antennas, IP-DECT Gateways, TDM-DECT Base Stations and Analogue VoIP Gateway

3 Executive Summary

The following summarizes tekVizion's findings:

- Test Case Failures:
  -N/A

- Features Not Supported:
  - 6.2.4 Intra-Cluster SIP URI Calling
  - 6.2.5 Inter-Cluster SIP URI Calling
  - 6.2.16 Functional Test: Directed Call Park
  - 6.2.19 Functional Test: Barge
  - 6.2.20 Functional Test: cBarge
  - 6.2.21 Functional Test: Shared Line – Hold/Resume
  - 6.2.24 Functional Test: Video Endpoints
  - 6.2.25 Functional Test: Extension Mobility
  - 6.2.30 Functional Test: Join Across Lines
  - 6.2.31 Functional Test: Hotline
  - 6.2.35 Functional Test: iDivert
  - 6.2.36 Functional Test: CFA & iDivert
  - 6.2.37 Functional Test: MCID
  - 6.2.38 Functional Test: Mobile Connect
  - 6.2.39 Functional Test: Mobile Voice Access (MVA)
  - 6.2.40 Functional Test: Enterprise Feature Access (EFA)
  - 6.4.6: Miscellaneous Test: Multiple Lines

- Test Cases that are Not Applicable:
  - 6.2.34 Functional Test: Do Not Disturb (DND)
  - 6.3.3 Negative Test: Phone Network Failure

- Test Cases that were Not Executed:
  - N/A

- Observations:
  - DUT requires upload of COP license file (COP file 14.0v1) to CUCM in order to employ additional CUCM features such as Callback, Group Pickup, Meet-Me, Call Pickup, etc. Filename must end in ".sha512" to comply with CUCM 14 security restrictions.
  - To register to CUCM, user must enable "Add Instance ID To The User Registration With The IP-PBX" on DUT.
  - To allow DND & true busy tone/response, user must enable "No Validation of Request URI" in DECT Station. Otherwise, CUCM responds with 500 Internal

Server Error and DECT handsets show ‘Temporary Failure’.
- DECT handsets must be configured for ‘Encrypted’ on CUCM if Admin is configuring DUT for TLS/SRTP. CUCM uses Null/SHA for Authenticated and DUT rejects this due to low security concerns. Due to CUCM still showing DUT as an untrusted device, CUCM CDR will show calls involving DUT as ‘0’ for unsecure, even though TLS/SRTP is successfully negotiated. User must also enable ‘Use Local Contact Port As Source Port for TCP/TLS Connections” for ‘TSIP’ & ‘SIPS’ on DUT.
- To utilize softkeys, user must use specific star codes configured in DECT Base Station.
- Callback for DUT utilizes ‘Presence’ in the SUBSCRIBE Event Header. This is not supported for PSTN usage. To enable Callback, user must enable “KPML Support”, and disable “Enbloc Dialing”, “Allow DTMF Through RTP”, and “Send Inband DTMF” on Base Station.
- Directed Call Pickup is a supported DUT feature, but does not meet test plan criteria. DUT does not support BLF lines/multiple lines.
- For SUB/SRST Failover, user must manually point DECT Base Station to SRST as a secondary proxy.
- Admin should primarily configure handsets from DUT, but some configuration can also be done from Ascom WinPDM software.

4 Features Tested

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

4.2 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.

4.3 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.

5 Test Environment

5.2 Administration, Testing and Debugging tools

Table 2 Administration, Testing and Debugging Tools

Product Name	Version	Type	Purpose	Units
Test Tools				
Wireshark	1.12.7	Network scanner	Viewing SIP traffic as it crosses the network and interacts with network devices	1
3rd Party Tools				
Ascom WinPDM	3.15.2	Ascom proprietary software	Used to configure & update Ascom equipment	1
Debug Tools				
Cisco RTMT	14.0	Real-Time Monitoring Tool	Used to track all data internal and external from the CUCM in real time	1
Cisco CDR	14.0	CDR	Call Detail Records to verify Call Security Status and other details	1

5.3 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	14.0.1.10000-20	PBX	Configuring & registering phones, call routing	2	VM hosted in-lab
IM&P	14.0.1.10000-16	IM & Presence server	Handles instant messaging and presence	1	VM hosted in-lab
Cisco 3845	12.4(24)T	ISR	PSTN Gateway	1	Physical device
3rd Party Products					
IPBS2-A3/1B1	11.4.4	IP-DECT Base Station	Making calls with DECT handsets over an IP network	2	Customer provided. Requires COP file 14.0v1-SIP.k3.cop.sha512
(D63 Messenger) DH7 – ABAA/1J	1.30	DECT Handsets	DECT handsets for making calls	3	Customer provided

5.4 Cisco Phones

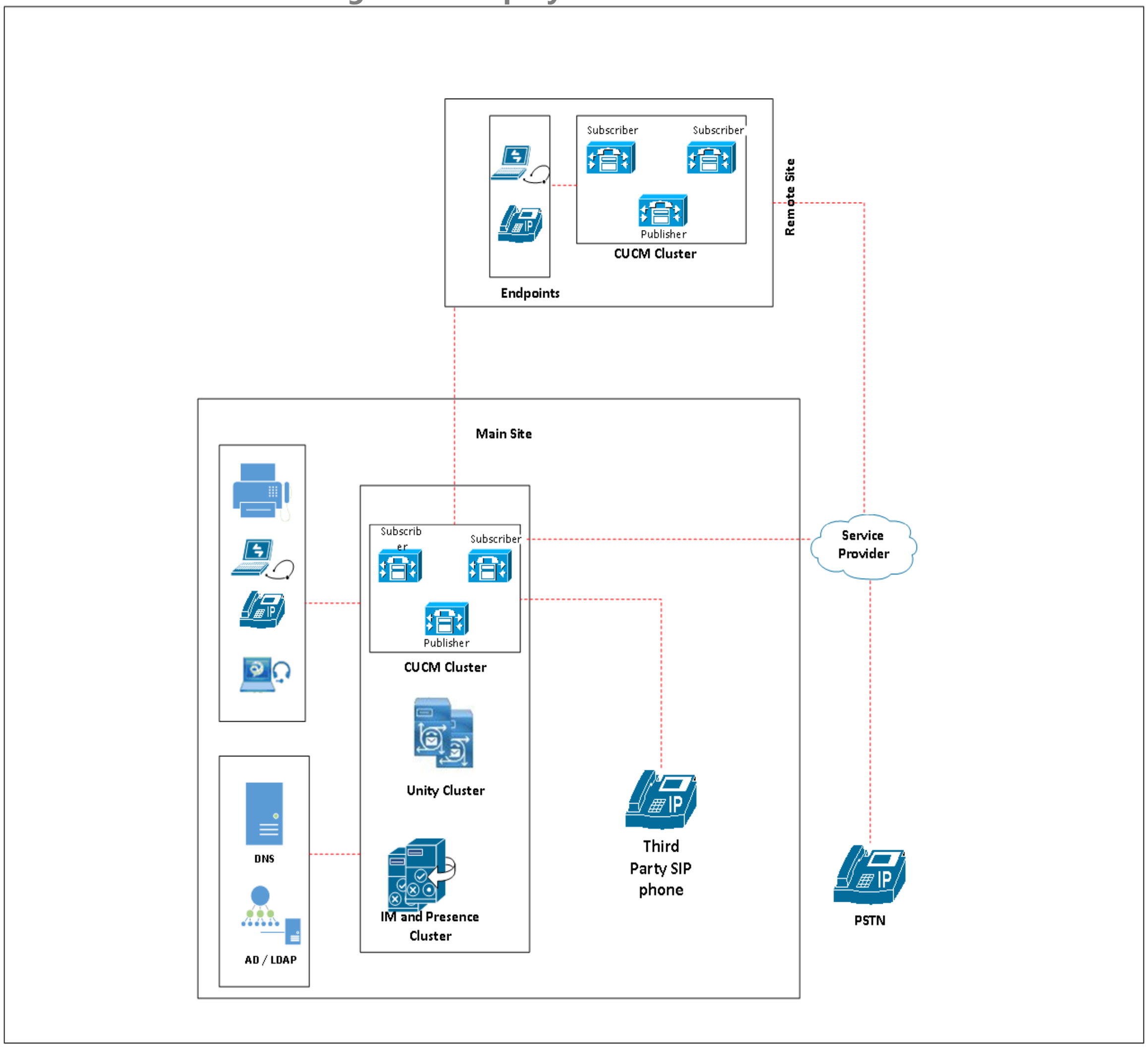
Table 4 Cisco Phones Information

Cisco Phone Model	Phone Firmware Version	Protocol	POE/Power	Units
9971	sip9971.9-4-2SR4-1	SIP	PoE	1

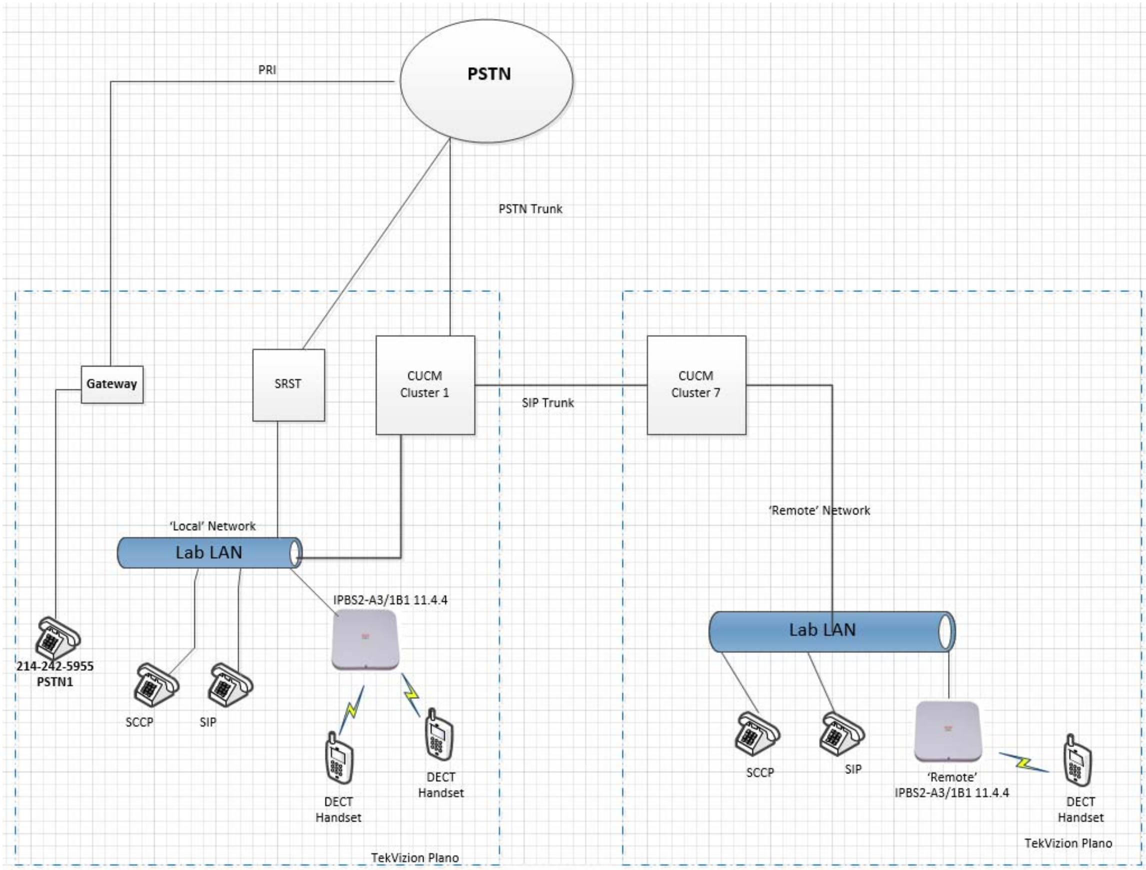
Cisco Phone Model	Phone Firmware Version	Protocol	POE/Power	Units
7965G	SCCP45.9-4-2SR4-3S	SCCP	PoE	1
7945G	SCCP45.9-4-2SR4-3S	SCCP	PoE	1
7821	sip78xx.14-0-1-0001-135	SIP	PoE	1

5.5 Deployment Architecture

Figure 1 – Deployment Architecture



5.6 Test Environment Architecture


PRI
PSTN
PSTN Trunk
Gateway
SRST
CUCM
Cluster 1
SIP Trunk
CUCM
Cluster 7
'Local' Network
Lab LAN
'Remote' Network
Lab LAN
IPBS2-A3/1B1 11.4.4
214-242-5955
PSTN1
SCCP
SIP
DECT
Handset
DECT
Handset
TekVizion Plano
SCCP
SIP
'Remote'
IPBS2-A3/1B1 11.4.4
DECT
Handset
TekVizion Plano

6 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.
Blocked	Other test case failures prevented the execution of this test. Reference to the corresponding failed test case must be provided in the Comments column.

6.1 Phase 1 Installation and Configuration Tests

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

6.1.1 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 two DUT(s) in local CUCM cluster 2. Connect one DUT in remote CUCM cluster 3. Go off-hook on DUT(s) to check for dial tone 4. Go to DUT(s) settings to verify network and load information 5. Associate end users to DUT(s) 6. Register 2 DUT's in local CUCM cluster with DN's 7. Register 1 DUT in Remote CUCM cluster with a DN
Expected Results	• CUCM Administration GUI display the DUT(s) • DUT(s) are in "Registered" state • DUT(s) have a DN assigned • Dial tone played when phone goes off-hook • 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 • DUT registered to Remote CUCM with assigned DN.
Observations	**Pass** To register to CUCM, user must enable "Add Instance ID To The User Registration With The IP-PBX" on DUT. User also requires COP file (14.0v1-SIP.k3.cop.sha512) uploaded to CUCM.

6.2 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.

6.2.1 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):DUT1 & DUT2; Remote CUCM • DUT:DUT3 • SCCP: Remote SCCP Phone1 • SIP: Remote SIP Phone1
Procedure	1. DUT1 dials DUT3 and DUT3 answers the call. 2. DUT3 on-hook after 30s 3. DUT2 dials Remote SCCP Phone1 and Remote SCCP Phone1 answers the call 4. Check for bidirectional audio. 5. Remote SCCP Phone1 on-hook after 60s 6. Repeat steps 3 to 5 with Calling Party:DUT2 & Called Party: Remote SIP Phone1 7. Calling & Called party release calls alternatively 8. Retrieve CDR from CUCM 9. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• 3 calls establish with 2-way audio path • Calling and Called Parties hear ring-back and ring tone • DUT receives Caller ID • 3 calls terminate normally • 3 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Pass**

6.2.2 Intra-Cluster Calls

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Intra-Cluster Calls</td></tr>
<tr><td>Description</td><td>Verify intra-cluster calls between DUT, SCCP and SIP endpoints</td></tr>
<tr><td>Test Setup</td><td>Local CUCM
DUT(s): DUT1 & DUT2
SCCP: Local SCCP Phone1
SIP: Local SIP Phone1</td></tr>
<tr><td>Procedure</td><td>DUT1 dials DUT2 and DUT2 answers the call
DUT1 on-hook after 30s
DUT2 dials Local SCCP Phone1 and Local SCCP Phone1 answers the call.
Check for two- way audio.
Local SCCP Phone1 on-hook after 60s
Repeat steps 3 to 5 with Calling Party: Local SIP Phone1 & Called Party: DUT1
Repeat steps 3 to 5 with Calling Party:DUT2 & Called Party: Local SIP Phone1
Calling & Called party release calls alternatively
Retrieve CDR from CUCM
Check Calling, Called, Duration, Origination & Termination Cause Codes</td></tr>
<tr><td>Expected Results</td><td>4 calls establish with 2-way audio path
Calling and Called Parties hear ring-back and ring tone
DUT receives Caller ID
4 calls terminate normally
4 CDR(s) retrieved
Selected fields in CDR(s) match table
<table>
<tr><th>CDR field</th><th>Call 1</th><th>Call 2</th><th>Call 3</th><th>Call 4</th></tr>
<tr><td>callingPartyNumber</td><td>DUT1</td><td>DUT2</td><td>Local SIP Phone1</td><td>DUT2</td></tr>
<tr><td>OriginalCalledPartyNumber</td><td>DUT2</td><td>Local SCCP Phone1</td><td>DUT1</td><td>Local SIP Phone1</td></tr>
<tr><td>finalCalledPartyNumber</td><td>DUT2</td><td>Local SCCP Phone1</td><td>DUT1</td><td>Local SIP Phone1</td></tr>
<tr><td>origCause_Value</td><td>16</td><td>0</td><td>16</td><td>0</td></tr>
<tr><td>destCause_Value</td><td>0</td><td>16</td><td>0</td><td>16</td></tr>
<tr><td>duration</td><td>30</td><td>60</td><td>60</td><td>60</td></tr>
</table></td></tr>
<tr><td>Observations</td><td>Pass</td></tr>
</table>

6.2.3 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):DUT1 & DUT2; • PSTN Phone➔ PSTN1 (SIP);
Procedure	1. DUT1 dials PSTN1 and PSTN1 answers the call 2. PSTN1 on-hook after 60s 3. Repeat steps1 and 2 with Calling Party:PSTN1 & Called Party:DUT2 4. Retrieve CDR from CUCM Server 5. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• 2 Calls establish with 2-way audio path • Calling and Called Parties hear ring-back and ring tone • DUT receives Caller ID • 2 Calls terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Pass**

6.2.4 Intra Cluster SIP URI Calling

Test Case Details	
Title	Intra Cluster SIP URI Calling
Description	Verify intra-cluster SIP URI calls between DUT and SIP endpoints
Test Setup	Local CUCM • DUT(s):DUT1 (URI:dutuser01@abc.inc); DUT2 (URI:dutuser02@abc.inc); • SIP: Local SIP Phone1 (URI:cuser20@abc.inc); Configure Speed Dial on button 3 for DUT1, DUT2, Local SIP Phone1: · Device➔Phone➔DUT1➔Add new SD➔dutuser02@abc.inc on both fields · Device➔Phone➔DUT2➔Add new SD➔cuser20@abc.inc on both fields · Device➔Phone➔Local SIP Phone1➔Add new

	SD→dutuser01@abc.inc on both fields
Procedure	1. DUT1 hits Speed Dial button 3 and DUT2 answers the call 2. DUT2 goes on-hook after 30s 3. Local SIP Phone1 hits Speed Dial button 3 and DUT1 answers 4. Local SIP Phone1 goes on-hook after 30s 5. DUT2 hits Speed Dial button 3 and Local SIP Phone1 answers 6. DUT2 goes on-hook after 30s 7. Retrieve CDR from CUCM Server 8. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• DUT(s) receives Caller ID • 3 calls establish with 2 way audio • 3 calls terminate normally • 3 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Not Supported** DUT does not support SIP-URI calling.

6.2.5 Inter Cluster SIP URI Calling

Test Case Details	
Title	Inter Cluster SIP URI Calling
Description	Verify inter-cluster SIP URI calls between DUT and SIP endpoints
Test Setup	Local CUCM • DUT(s):DUT1 (URI: dutuser01@abc.inc) & DUT2(URI: dutuser02@abc.inc); • SIP: Local SIP Phone1 (URI: cuser20@abc.inc); Remote CUCM • DUT :DUT3 (URI: rdutuser01@abc.inc); • SIP: Remote SIP Phone1 (URI: rcuser20@abc.inc); Configure Speed Dial on button 3 for DUT1, DUT2 & Remote SIP Phone1: · Device→Phone→DUT1→Add new SD→rdutuser01@abc.inc on both fields · Device→Phone→DUT2→Add new SD→rcuser20@abc.inc on both fields · Device→Phone→Remote SIP Phone1→Add new

	SD➔dutuser01@abc.inc on both fields
Procedure	1. DUT1 hits Speed Dial button 3 and DUT3 answers the call 2. DUT3 goes on-hook after 30s 3. DUT2 hits Speed Dial button 3 and Remote SIP Phone1 answers 4. Remote SIP Phone1 goes on-hook after 30s 5. Remote SIP Phone1 hits Speed Dial button 3 and DUT1 answers 6. DUT1 goes on-hook after 30s 7. Retrieve CDR from CUCM Server 8. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• DUT(s) receive Caller ID • 3 calls establish with 2 way audio • 3 calls terminate normally • 3 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Not Supported** DUT does not support SIP-URI calling.

6.2.6 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(s):DUT1 & DUT2 • SCCP: Local SCCP Phone1 • SIP: Local SIP Phone1 Remote CUCM • DUT:DUT3 • SCCP: Remote SCCP Phone1 • SIP: Remote SIP Phone1 PSTN Phone: PSTN1 Enable CFA for DN(s): · Device➔Phone➔DUT1➔CFA➔Local SCCP Phone1 (SCCP) · Device➔Phone➔DUT2➔CFA➔DUT3 (DUT) · Device➔Phone➔DUT3➔CFA➔Remote SIP Phone1 (SIP) · Device➔Phone➔Remote SCCP Phone1➔CFA➔DUT2 (DUT)

	· Device➔Phone➔Local SIP Phone1➔CFA➔DUT3 (DUT) · Device➔Phone➔PSTN1➔DUT1 (DUT)
Procedure	1. DUT1 dials DUT2, DUT3 answers and DUT1 on-hook after 30s 2. DUT3 dials DUT1, Local SCCP Phone1 answers and Local SCCP Phone1 on-hook after 30s 3. DUT2 dials DUT3, Remote SIP Phone1 answers and DUT2 on-hook after 30s 4. DUT2 dials Local SIP Phone1 and DUT3 answers 5. DUT1 dials Remote SCCP Phone1 and DUT1 on-hook - hears busy tone 6. DUT2 goes on-hook 7. DUT2 dials PSTN1 and DUT1 answers 8. PSTN1 goes on-hook after 30s 9. PSTN1 dials DUT2 and DUT3 answers 10. DUT3 goes on-hook after 30 secs 11. Retrieve CDR from CUCM 12. Check Calling, Called, Duration, Origination & Termination Cause Codes **Note:** Upon test completion, remove "CFA" feature for devices before proceeding to next test case CUCM Administration GUI: Device➔Phone➔DN➔Line➔Call Forward All ➔Destination➔blank
Expected Results	• CFA" phones displays the CFA # on screen • Call forward to DUT3 and phone rings • Call establish between DUT1 & DUT3 with 2-way audio • Call terminate normally • Call forward to Local SCCP Phone1 and phone rings • Call establish between DUT3 & Local SCCP Phone1 with 2-way audio • Call terminate normally • Call forward to Remote SIP Phone1 and phone rings • Call establish between DUT2 & Remote SIP Phone1 with 2-way audio • Call terminate normally • Call forward to DUT3 and phone rings • Call establish between DUT2& DUT3 with 2-way audio • Call forward to DUT2 and phone rings

	• Call forward to DUT3 and phone returns busy tone • DUT1 hears busy tone and release call • Call on DUT2 terminate normally • Call forward to DUT1 and phone rings • Call establish between DUT2& DUT1 with 2-way audio • Call terminate normally • Call forward to DUT3 and phone rings • Call between 2102225400 & DUT3 with 2-way audio • Call terminate normally • 7 CDR(s) retrieved • Selected CDR(s) fields match calls
Observations	**Pass** Must enable “No Validation of Request URI” in DUT via Advanced > SIP. Otherwise, for Steps 4-5: CUCM sends REFER to DUT with XML data showing <statustext>Busy<statustext> and DUT responds with 403 Forbidden. CUCM sends 183 with Reason header showing Q.850; cause=17. CUCM sends another REFER to DUT with <tonetype>DtLineBusy<tonetype> and DUT responds with another 403 Forbidden to seconds REFER and CUCM sends 500 Internal Server Error.

6.2.7 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):DUT1 & DUT2;
	  · SCCP: Local SCCP Phone1; SIP: Local SIP Phone1;
	• Remote CUCM
	  · DUT:DUT3; SCCP: Remote SCCP Phone1 SIP: Remote SIP Phone1;
	• Voicemail and Call Waiting disabled for all DN(s) Device➔Phone➔DN➔Line➔
	  · Voicemail➔NoVoiceMail
	  · Call Waiting➔Max. Calls➔1; Busy Trigger➔1;
	
	Enable CFNA for DN(s):
	· Device➔Phone➔DUT1➔CFNA➔Local SCCP Phone1 (SCCP)
	· Device➔Phone➔DUT2➔CFNA➔DUT3 (DUT)
	· Device➔Phone➔DUT3➔CFNA➔Remote SIP Phone1 (SIP)
	· Device➔Phone➔Remote SCCP Phone1➔CFNA➔DUT2 (DUT)

	· Device→Phone→Local SIP Phone1→CFNA→DUT3 (DUT)
Procedure	1. DUT1 dials DUT2, DUT2 does not answer and DUT3 answers 2. DUT1 goes on-hook after 30s 3. DUT3 dials DUT1, DUT1 does not answer and Local SCCP Phone1 answers 4. Local SCCP Phone1 goes on-hook after 30s 5. DUT2 dials DUT3, DUT3 does not answer and Remote SIP Phone1 answers 6. DUT2 goes on-hook after 30s 7. DUT2 dials Local SIP Phone1, Local SIP Phone1 does not answer and DUT3 does not answer 8. Remote SIP Phone1 answers call 9. DUT3 dials Remote SCCP Phone1, Remote SCCP Phone1 does not answer and DUT2 does not answer 10. DUT3 goes on-hook - hears busy tone 11. Remote SIP Phone1 goes on-hook 12. Retrieve CDR from CUCM 13. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call forward to DUT3 after ring timeout • Call establish between DUT1 & DUT3 with 2-way audio • Call terminate normally • Call forward to Local SCCP Phone1 after ring timeout • Call establish between DUT3 & Local SCCP Phone1 with 2-way audio • Call terminate normally • Call forward to Remote SIP Phone1 after ring timeout • Call establish between DUT2 & Remote SIP Phone1 with 2-way audio • Call terminate normally • Call forward to Remote SIP Phone1 after ring timeout • Call establish between DUT2& Remote SIP Phone1 with 2-way audio • Call terminate normally • DUT3 hears busy tone and terminate call • Call on Remote SIP Phone1 terminate normally • Call forward to DUT1 after ring timeout • Call establish between DUT2& DUT1 with 2-way audio

	• Call terminate normally • Call forward to DUT3 after ring timeout • 5 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass Must enable "No Validation of Request URI" in DUT via Advanced > SIP. Otherwise, for Steps 7-10: CUCM sends REFER to DUT with XML data showing <statustext>Busy<statustext> and DUT responds with 403 Forbidden. CUCM sends 183 with Reason header showing Q.850; cause=17. CUCM sends another REFER to DUT with <tonetype>DtLineBusy<tonetype> and DUT responds with another 403 Forbidden to seconds REFER and CUCM sends 500 Internal Server Error.

6.2.8 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):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1;
	• Remote CUCM
	  · DUT:DUT3; SCCP: Remote SCCP Phone1; SIP: Remote SIP Phone1;
	• Voicemail and Call Waiting disabled for all DNs
	
	Enable CFB for DN(s):
	· Device→Phone→DUT1→CFB→Local SCCP Phone1 (SCCP)
	· Device→Phone→DUT2→CFB→DUT3 (DUT)
	· Device→Phone→DUT3→CFB→Remote SIP Phone1 (SIP)
	· Device→Phone→Remote SIP Phone1→CFB→DUT2(DUT)
	· Device→Phone→Local SCCP Phone1→CFB→DUT3 (DUT)

Procedure	1. Local SCCP Phone1 dials DUT2, DUT2 answers 2. DUT1 dials DUT2, DUT3 answers and DUT1 on-hook after 30s 3. Remote SCCP Phone1 dials DUT3 and DUT3 answers 4. DUT1 dials DUT2, DUT1 on-hook – hears busy tone 5. Remote SCCP Phone1 goes on-hook 6. DUT1 dials Local SCCP Phone1, DUT3 answers; DUT1 on-hook after 30s

	7. DUT2 goes on-hook 8. Local SIP Phone1 dials Local SCCP Phone1, Local SCCP Phone1 answers 9. Remote SCCP Phone1 dials DUT3, DUT3 answers 10. DUT2 dials Local SCCP Phone1, DUT2 on-hook – hears busy tone 11. Local SCCP Phone1 goes on-hook 12. Local SCCP Phone1 dials Remote SIP Phone1, Remote SIP Phone1 answers 13. DUT1 dials Remote SIP Phone1, DUT2 answers and DUT2 on-hook after 30s 14. Local SIP Phone1 dials DUT2 and DUT2 answers 15. DUT1 dials Remote SIP Phone1, DUT1 goes on-hook –hears busy tone 16. Local SIP Phone1 & Remote SIP Phone1 goes on-hook 17. Retrieve CDR from CUCM 18. Check Calling, Called, Duration, origination & termination cause codes matches the calls
Expected Results	• Call establish between Local SCCP Phone1 & DUT2 with 2-way audio • Call forward to DUT3 and phone rings • Call establish between DUT1 & DUT3 with 2-way audio • DUT1 terminate call • Call establish between Remote SCCP Phone1 & DUT3 with 2-way audio • DUT1 hears busy tone and release call • Remote SCCP Phone1 terminate call • Call forward to DUT3 and phone rings • Call establish between DUT1 & DUT3 with 2-way audio • DUT2 terminate call • Call establish between Local SIP Phone1 & Local SCCP Phone1 with 2-way audio • Call establish between Remote SCCP Phone1 & DUT3 with 2-way audio • DUT2 hears busy tone and release call • Local SCCP Phone1 terminate call • Call establish between Local SCCP Phone1 & Remote SIP Phone1 with 2-way audio • Call forward to DUT2 and phone rings • Call establish between DUT1 & DUT2 with 2-way audio

	• DUT2 terminate call • Call establish between Local SIP Phone1 & DUT2 with 2-way audio • DUT1 hears busy tone and release call • Local SIP Phone1 & Remote SIP Phone1 terminate call • Call establish between Local SIP Phone1 & DUT1 with 2-way audio • Local SIP Phone1 & DUT2 terminate call • 12 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Pass** Must enable “No Validation of Request URI” in DUT via Advanced > SIP. Otherwise: CUCM sends REFER to DUT with XML data showing <statustext>Busy<statustext> and DUT responds with 403 Forbidden. CUCM sends 183 with Reason header showing Q.850; cause=17. CUCM sends another REFER to DUT with <tonetype>DtLineBusy<tonetype> and DUT responds with another 403 Forbidden to seconds REFER and CUCM sends 500 Internal Server Error.

6.2.9 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):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT3; SCCP: Remote SCCP Phone1; • PSTN: PSTN1 • Remove all CFA, CFNA & CFB settings on DN(s) used in previous test cases • Call Waiting enabled on all DN(s) Device➔Phone➔DN➔Line➔Call Waiting➔Max Calls➔4; Busy Trigger➔2;
Procedure	1. DUT1 dials DUT2, DUT2 answers and DUT1 hits “Hold” after 20s 2. DUT1 hits “Resume” after 20s and DUT1 on-hook after 30s 3. DUT1 dials DUT3, DUT3 answers 4. Remote SCCP Phone1 dials DUT1, DUT1 answers incoming call and DUT3 on-hold 5. DUT1 hits “Resume” after 60s and DUT1 on-hook after 30s 6. DUT1 dials DUT2, DUT2 answers and DUT1 hits “Hold” after 30s 7. DUT1 dials Remote SCCP Phone1, Remote SCCP Phone1 answers

	and Remote SCCP Phone1 on-hook after 30s 8. DUT2 goes on-hook after 30s 9. DUT2 dials DUT1, DUT1 answers and DUT2 hits "Hold" after 30s 10. DUT1 goes on-hook 10s later while call is on-hold 11. DUT2 goes-hook 12. Repeat steps 1-2 for SCCP. Replace DUT2 with Local SCCP Phone1 13. Repeat steps 1-2 for SIP. Replace DUT1 with Local SIP Phone1 14. Repeat steps 1-2 for PSTN. Replace DUT2 with PSTN1 15. Retrieve CDR from CUCM 16. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT1 & DUT2 with 2-way audio • DUT2 is On-Hold (MOH) • Call resume between DUT1 & DUT2 • Call terminate normally • Call establish between DUT1 & DUT3 with 2-way audio • DUT3 is On-Hold (MOH) • Call establish between DUT1 & Remote SCCP Phone1 with 2-way audio • Call on Remote SCCP Phone1 terminated normally • Call resume between DUT1 & DUT3 with 2-way audio • Call terminate normally • Call establish between DUT1 & DUT2 with 2-way audio • DUT2 is On-Hold (MOH) • Call establish between DUT1 & Remote SCCP Phone1 with 2-way audio • Remote SCCP Phone1 terminate call normally • Call resume between DUT1 & DUT2 with 2-way audio • Call terminate normally • Call establish between DUT2 & DUT1 with 2-way audio • DUT1 is On-Hold (MOH) • DUT1 terminate call during active hold • 9 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass DUT must have call waiting enabled. Not enabled by default. Steps 4-5: DUT must switch back to first call with DUT2. Ending call with Remote SCCP phone transfer DUT2 to Remote SCCP. Remote SCCP must manually end call. Steps 6-8: DUT makes new call. Must switch back to DUT2 after Remote

	SCCP hangs up, otherwise DUT will remain showing call is hung. Hanging up doesn't switch back to DUT2 but instead ends all calls. Steps 9-11: DUT2 shows no indication that DUT1 hung up.

6.2.10 Functional Test: Call Waiting

Test Case Details	
Title	Functional Test: Call Waiting
Description	Verify Call Waiting calls between DUT(s), SIP and SCCP endpoints
Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT3; SCCP: Remote SCCP Phone1; SIP: Remote SIP Phone1; • Call Waiting enabled for all DN(s): Device➔Phone➔DN➔Line➔Call Waiting➔Max. Calls➔4; Busy Trigger➔2;
Procedure	1. DUT1 dials DUT2, DUT2 answers 2. DUT3 dials DUT1, DUT1 answers incoming call 3. DUT3 goes on-hook after 30s 4. Local SCCP Phone1 dials DUT2, DUT2 answers incoming call 5. Local SCCP Phone1 goes on-hook after 30s 6. DUT1 goes on-hook after 60s 7. DUT1 dials Local SCCP Phone1, Local SCCP Phone1 answers 8. Local SIP Phone1 dials DUT1, DUT1 answers incoming call 9. Local SIP Phone1 goes on-hook after 30s 10. DUT1 goes on-hook after 60s 11. Remote SIP Phone1 dials DUT2, DUT2 answers 12. Remote SCCP Phone1 dials DUT2, DUT2 answers incoming call 13. Remote SCCP Phone1 goes on-hook after 60s 14. Remote SIP Phone1 goes on-hook after 30s 15. Retrieve CDR from CUCM 16. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT1 & DUT2 with 2-way audio • DUT1 notified of incoming call (tone /display) • DUT1 answers incoming call • DUT2 is On-Hold (MOH) • Call establish between DUT1 & DUT3 with 2-way audio

	• DUT1 & DUT3 terminate normally • Call resume between DUT1 & DUT2 • DUT2 notified of incoming call (tone /display) • DUT2 answers incoming call • DUT1 is On-Hold (MOH) • Call establish between DUT2 & Local SCCP Phone1 with 2-way audio • DUT2 & Local SCCP Phone1 terminate normally • Call resume between DUT1 & DUT2 • DUT1 & DUT2 terminate normally • Call establish between DUT1 & Local SCCP Phone1 with 2-way audio • DUT1 notified of incoming call (tone /display) • DUT1 answers incoming call • Local SCCP Phone1 is On-Hold (MOH) • Call establish between DUT1 & Local SIP Phone1 with 2-way audio • DUT1 & Local SIP Phone1 terminate normally • Call resume between DUT1 & Local SCCP Phone1 • DUT1 & Local SCCP Phone1 terminate normally • Call establish between Remote SIP Phone1 & DUT2 with 2-way audio • DUT2 notified of incoming call (tone /display) • DUT2 answers incoming call • Remote SIP Phone1 is On-Hold (MOH) • Call establish between DUT2 & Remote SCCP Phone1 with 2-way audio • DUT2 & Remote SCCP Phone1 terminate normally • Call resume between Remote SIP Phone1 & DUT2 • Remote SIP Phone1 & DUT2 terminate normally • 7 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Pass**

6.2.11 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):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT3; • PSTN: PSTN1 • Invalid DN:7777;
Procedure	1. DUT1 dials DUT2, DUT2 answers and DUT2 hits “Transfer” after 30s 2. DUT2 dials DUT3, DUT2 hits “Transfer” and DUT2 is on-hook 3. DUT3 goes on-hook after 60s 4. DUT1 dials DUT2, DUT2 answers and DUT1 hits “Transfer” after 30s 5. DUT1 dials 7777, DUT1 hits “Transfer” and DUT1 is on-hook 6. DUT2 goes on-hook – hears reorder tone 7. DUT1 dials Local SCCP Phone1, Local SCCP Phone1 answers and DUT1 hits “Transfer” after 30s 8. DUT1 dials DUT2, DUT1 hits “Transfer” and DUT1 is on-hook 9. Local SCCP Phone1 goes on-hook after 60s 10. DUT1 dials Local SIP Phone1, Local SIP Phone1 answers and DUT1 hits “Transfer” after 30s 11. DUT1 dials Local SCCP Phone1, DUT1 hits “Transfer” and DUT1 is on-hook 12. Local SCCP Phone1 goes on-hook after 20s 13. DUT2 dials 234-DUT3, DUT3 answers 14. DUT1 dials Local SIP Phone1, Local SIP Phone1 answers and Local SIP Phone1 hits “Transfer’” after 30s 15. Local SIP Phone1 dials DUT3, Local SIP Phone1 hits “Transfer” and Local SIP Phone1 is on-hook 16. DUT1 goes on-hook - hears busy tone 17. Retrieve CDR from CUCM 18. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT1 & DUT2 with 2-way audio • DUT1 is On-Hold (MOH) • DUT1 blind transfer to DUT3 with 2-way audio path • All calls terminate normally • Call establish between DUT1 & DUT2 with 2-way audio • DUT2 is On-Hold (MOH) • DUT2 blind transfer to Invalid DN:7777 • DUT2 hears reorder tone • All calls terminate normally • Call establish between DUT1 & Local SCCP Phone1 with 2-way

	audio • Local SCCP Phone1 is On-Hold (MOH) • Local SCCP Phone1 blind transfer to DUT2 with 2-way audio • All calls terminate normally • Call establish between DUT1 & Local SIP Phone1 with 2-way audio • Local SIP Phone1 is On-Hold (MOH) • Local SIP Phone1 blind transfer to Local SCCP Phone1 with 2-way audio path • All calls terminate normally • Call establish between DUT2 & DUT3 with 2-way audio • Call establish between DUT1 & Local SIP Phone1 with 2-way audio • DUT1 is On-Hold (MOH) • DUT1 blind transfer to DUT3 • DUT1 hears a busy tone • All calls terminate normally • 15 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Pass** Ascom WinPDM software used to configure transfer capabilities. "RR+DN" is used for blind transfer. Steps 5-6: When DUT attempted to blind transfer DUT2 to non-existing DN, DUT1 got error tone and returned to call with DUT2. Call was not released. Steps 15-16: Cisco phones do not support blind transfer, so Local SIP Phone heard busy tone and was not able to transfer DUT1 to DUT3.

6.2.12 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):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT3; • PSTN: PSTN1
Procedure	1. DUT1 dials DUT2, DUT2 answers and DUT2 hits "Transfer'" after 30s 2. DUT2 dials DUT3, DUT3 answers 3. DUT2 hits "Transfer" after 30s and DUT2 is on-hook 4. DUT3 goes on-hook after 60s

	5. DUT 1 dials Local SCCP Phone1, Local SCCP Phone1 answers and DUT1 hits "Transfer" after 30s 6. DUT1 dials DUT2, DUT2 answers and DUT1 hits "Transfer" after 30s 7. DUT1 goes on-hook 8. DUT 2 goes on-hook after 60s 9. DUT1 dials Local SIP Phone1, Local SIP Phone1 answers and DUT1 hits "Transfer'" after 30s 10. DUT1 dials Local SCCP Phone1, Local SCCP Phone1 answers and DUT1 hits "Transfer" after 30s 11. DUT1 goes on-hook 12. Local SCCP Phone1 goes on-hook after 60s 13. DUT2 dials PSTN1, PSTN1 answers 14. DUT2 hits "Transfer" after 30s, DUT2 dials DUT1 and DUT1 answers 15. DUT2 hits "Transfer" after 30s, DUT2 is on-hook 16. PSTN1 goes on-hook after 60s 17. Retrieve CDR from CUCM 18. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT1 & DUT2 with 2-way audio • DUT1 is On-Hold (MOH) • DUT1 consult transfer to DUT3 with 2-way audio path • All calls terminate normally • Call establish between DUT1 & Local SCCP Phone1 with 2-way audio • Local SCCP Phone1 is On-Hold (MOH) • Local SCCP Phone1 consult transfer to DUT2 with 2-way audio path • All calls terminate normally • Call establish between DUT1 & Local SIP Phone1 with 2-way audio • Local SIP Phone1 is On-Hold (MOH) • Local SIP Phone1 consult transfer to Local SCCP Phone1 with 2-way audio path • All calls terminate normally • Call establish between DUT2 & PSTN1 with 2-way audio • PSTN1 is On-Hold (MOH) • PSTN1 consult transfer to DUT1 with 2-way audio path • All calls terminate normally • 12 CDR(s) retrieved • Selected fields in CDR(s) match calls

Observations	Pass

6.2.13 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):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM →DUT:DUT3; • PSTN: PSTN1 • 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. DUT1 dials DUT2, DUT2 answers and DUT2 hits “Conference” after 30s 2. DUT2 dials DUT3, DUT3 answers 3. DUT2 hits “Conference” after 30s 4. DUT1 goes on-hook after 60s 5. DUT3 goes on-hook after 30s 6. DUT1 dials DUT2, DUT2 answers and DUT1 hits “Conference” after 30s 7. DUT1 dials Local SCCP Phone1, Local SCCP Phone1 answers and DUT1 hits “Conference” after 30s 8. SCCP phone 1 goes on-hook after 60s 9. DUT 1 goes on-hook after 30s 10. DUT1 dials Local SIP Phone1, Local SIP Phone1 answers and DUT1 hits “Conference” after 30s 11. DUT1 dials PSTN1, PSTN1 answers 12. DUT1 hits “Conference” after 30s 13. Local SIP Phone1 hits “Conference” after 30s and Local SIP Phone1 dials DUT3 14. DUT3 answers, Local SIP Phone1 hits “Conference” after 30s 15. PSTN1 hits “Conference” after 30s, PSTN1 dials DUT2, DUT2 answers and PSTN1 hits “Conference” after 30s 16. PSTN1 goes on-hook after 60s 17. DUT1 goes on-hook after 30s

	18. Local SIP Phone1 goes on-hook after 30s 19. DUT1 dials Local SIP Phone1, Local SIP Phone1 answers and DUT1 hits "Conference" after 30s 20. DUT1 dials Local SCCP Phone1, DUT1 resumes call before Local SCCP Phone1 answers 21. Local SIP Phone1 goes on-hook after 30s 22. Retrieve CDR from CUCM 23. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT1 & DUT2 with 2-way audio • DUT1 is On-hold (MOH) • DUT3 is conference-in • 3 parties in conference call with 3-way audio • DUT1 left conference.DUT2 & DUT3 connect directly • All calls terminate normally • Call establish between DUT1 & Local SIP Phone1 with 2-way audio • Local SIP Phone1 is On-Hold (MOH) • PSTN1, DUT3 & DUT2 is conference-in • All 5 parties in conference call with 5-way audio • PSTN1, DUT1 & Local SIP Phone1 leave conference. DUT3 & DUT2 connect directly • All calls terminate normally • Call establish between DUT1 & Local SIP Phone1 with 2-way audio • Local SIP Phone1 is placed on-hold (MOH) • Conference setup was cancelled • Call between DUT1 and Local SIP Phone1 resumed • All calls terminate normally • CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Pass** In order to initiate conference, DECT handsets must utilize internal soft key features. 'R' is pressed to initiate a new call, then once new call is answered, 'R3' is pressed to conference the parties. Pressing 'R' put first call on hold. Once this is done and party is rung, DECT handset cannot cancel call without cancelling all calls. Ergo, steps 19-20 cannot be performed.

6.2.14 Functional Test: Call Park

Test Case Details	
Title	Functional Test: Call Park

Description	Verify "Call Park" call for a DUT(s), SIP, SCCP and PSTN endpoints
Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT3; • PSTN: PSTN1 • Call Park Code: Routing➔Call Park➔3001
Procedure	1. DUT1 dials DUT3, DUT3 answers 2. DUT1 hits "Park" softkey after 10s 3. DUT2 dials park code:3001 after 20s 4. DUT3 goes on-hook after 30s 5. DUT1 dials Local SCCP Phone1, Local SCCP Phone1 answers 6. Local SCCP Phone1 hits "Park" softkey after 10s 7. DUT2 dials park code:3001 after 20s 8. DUT1 goes on-hook after 30s 9. Local SIP Phone1 dials DUT2, DUT2 answers 10. DUT2 hits "Park" softkey after 10s 11. DUT1 dials park code:3001 after 20s 12. Local SIP Phone1 goes on-hook after 30s 13. PSTN1 dials DUT1, DUT1 answers 14. DUT1 hits the "Park" softkey after 10s 15. DUT1 dials park code:3001 after 20s 16. PSTN1 goes on-hook 17. Retrieve CDR from CUCM 18. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT1 & DUT3 with 2-way audio • DUT3 is parked • DUT2 picks up parked call • Call establish between DUT2 & DUT3 with 2-way audio • Call terminate normally • Call establish between DUT1 & Local SCCP Phone1 with 2-way audio • DUT1 is parked • DUT2 picks up parked call • Call establish between DUT2 & DUT1 with 2-way audio • Call terminate normally • Call establish between Local SIP Phone1 & DUT2 with 2-way audio • Local SIP Phone1 is parked

	• DUT1 picks up parked call • Call establish between DUT1 & Local SIP Phone1 with audio path • Call terminate normally • Call establish between PSTN1 & DUT1 with 2-way audio • PSTN1 is parked • DUT1 picks up parked call • Call establish between DUT1 & PSTN1 with 2-way audio • Call terminate normally • 8 CDR(s) retrieved • Selected fields in the CDR matched the calls
Observations	**Pass** DUT uses specific star code to initiate parking.

6.2.15 Functional Test: Call Park Reversion

Test Case Details	
Title	Functional Test: Call Park Reversion
Description	Verify "Call Park Reversion" call for DUT(s), SIP and SCCP endpoints
Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT3; • Call Park Code: Routing➔Call Park➔3001 • Service Parameter: Call Park Reversion Timer ➔60s
Procedure	1. DUT1 dials DUT3, DUT3 answers 2. DUT1 hits "Park" softkey after 10s 3. Do not pickup parked call for 60s 4. DUT1 is ringing and DUT1 answers 5. DUT3 goes on-hook after 30s 6. DUT1 dials Local SCCP Phone1, Local SCCP Phone1 answers 7. Local SCCP Phone1 hits "Park" softkey after 10s 8. Do not pickup parked call for 60s 9. Local SCCP Phone1 is ringing and Local SCCP Phone1 answers 10. DUT1 goes on-hook after 30s 11. Local SIP Phone1 dials DUT2, DUT2 answers 12. DUT2 hits "Park" softkey after 10s 13. Do not pickup parked call for 60s 14. DUT2 is ringing, DUT2 answers 15. Local SIP Phone1 goes on-hook after 30s 16. Retrieve CDR from CUCM

	17. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT1 & DUT3 with 2-way audio • DUT3 is parked • DUT1 picks up parked call • Call establish between DUT1 & DUT3 with 2-way audio • Call terminate normally • Call establish between DUT1 & Local SCCP Phone1 with 2-way audio • DUT1 is parked • Local SCCP Phone1 picks up parked call • Call establish between Local SCCP Phone1 & DUT1 with 2-way audio • Call terminate normally • Call establish between Local SIP Phone1 & DUT2 with 2-way audio • Local SIP Phone1 is parked • DUT2 picks up parked call • Call establish between DUT2 & Local SIP Phone1 with 2-way audio • Call terminate normally • 6 CDR(s) retrieved • Selected fields in CDR match calls
Observations	**Pass** DUT uses specific star code to initiate parking.

6.2.16 Functional Test: Directed Call Park

Test Case Details	
Title	Functional Test: Directed Call Park
Description	Verify "Assisted Directed Call Park" call between DUT(s) and SIP endpoints
Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2; SIP: Local SIP Phone1 and Local SIP Phone2; SCCP: Local SCCP Phone1; • Remote CUCM ➔DUT:DUT3; • Enterprise Parameter: BLF For Call Lists ➔Enable • Directed Call Park DN-3011:Routing➔Directed Call Park➔3011 & Retrieval Prefix * • Add BLF Call Park: Device➔Device Settings➔Phone Button Template➔Copy template➔BLF➔Line 4➔Call Park BLF • Update Phone Button Template for all DN(s)::Device➔Phone➔DN➔Phone Button Template➔BLF

	• Directed Call Park DN provisioned for all (DN(s):Device➔Phone➔DN➔Line 4 BLF➔DN:3011
Procedure	1. DUT3 dials Local SIP Phone1, Local SIP Phone1 answers 2. Local SIP Phone1 hits "BLF" button for Assisted Directed Call Park after 20s 3. Local SIP Phone1 goes on-hook 4. DUT2 dials *3011 to retrieve call when the BLF is flashing 5. DUT3 goes on-hook after 30s 6. Local SIP Phone2 dials DUT3, DUT3 answers 7. Local SIP Phone2 hits "BLF" button for Assisted Directed Call Park after 20s 8. Local SIP Phone2 goes on-hook 9. Local SCCP Phone1 hits dials *3011 to retrieve call when BLF is flashing 10. Local SCCP Phone1 goes on-hook after 30s 11. Retrieve CDR from CUCM 12. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT3 & Local SIP Phone1 with 2-way audio • DUT3 is parked • DUT2 retrieves directed parked call • Call establish between DUT3 & DUT2 with 2-way audio • Calls terminate normally • Call establish between Local SIP Phone2 & DUT3 with 2-way audio • DUT3 is parked • Local SCCP Phone1 retrieves directed park call • Call establish between Local SCCP Phone1 & DUT3 with 2-way audio • Calls terminated normally • 4 CDR(s) retrieved • Selected fields in CDR match calls
Observations	**Not Supported** DUT does not support multiple lines or BLF keys. Supports Directed Call Pickup but not within test plan criteria.

6.2.17 Functional Test: Meet-Me

Test Case Details	
Title	Functional Test: Meet-Me

Description	Verify "Meet-Me" Conference call using DUT(s), SCCP and SIP endpoints
Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT3; • Meet-Me #: Call Routing➔Meet-Me➔Add New➔5555 [meet-me #] & create CTI_RP with DN:5555) • Meet-Me Conference initiator (DUT2): Device➔Phone➔DN:DUT2➔Calling Search Space➔css-meetme
Procedure	1. DUT2 goes off hook & hits "Meet-Me" softkey and dials 5555 2. Local SCCP Phone1 dials 5555 3. Local SIP Phone1 dials 5555 4. DUT3 dials 444-5555 5. All 4 members go on-hook after 120 secs 6. Retrieve CDR from CUCM 7. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• DUT2, Local SCCP Phone1, Local SIP Phone1, & DUT3 forward to conference bridge port • All 4 parties in conference with 4-way audio • Conference call terminate normally • 4 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Pass** DUT must use unique Call Service URI star code configured in DECT Station.

6.2.18 Functional Test: Callback

Test Case Details	
Title	Functional Test: Callback
Description	Verify "Callback" calls between DUT(s), SCCP, SIP and PSTN endpoints
Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2; SCCP :Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT3; • PSTN: PSTN1 • VM and CW disabled for all phones ;

Procedure	1. DUT1 dials DUT2, DUT2 answers 2. DUT3 dials DUT2, DUT3 hits "Callback" softkey and exits 3. DUT2 goes on-hook after 60s 4. DUT3 redials DUT2 after callback alert 5. DUT2 answers, DUT2 goes on-hook after 60s 6. DUT1 dials Local SCCP Phone1, Local SCCP Phone1 answers 7. DUT2 dials Local SCCP Phone1, DUT2 hits "Callback" softkey and exits 8. DUT1 goes on-hook after 60s 9. DUT2 redials Local SCCP Phone1 after callback alert 10. Local SCCP Phone1 answers and Local SCCP Phone1 goes on-hook after 60s 11. Local SIP Phone1 dials DUT1, DUT1 answers 12. DUT2 dials Local SIP Phone1, DUT2 hits "Callback" softkey and exits 13. DUT1 goes on-hook after 60s 14. DUT2 redials Local SIP Phone1 after callback alert 15. Local SIP Phone1 answers and DUT2 goes on-hook after 60s 16. DUT2 dials PSTN1, PSTN1 answers 17. DUT1 dials PSTN1, DUT1 hits "Callback" softkey & exits 18. PSTN1 goes on-hook after 60s 19. DUT1 redials PSTN1 after callback alert 20. Retrieve CDR from CUCM 21. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT1 & DUT2 with 2-way audio • DUT3 hears a busy tone & displays "Callback Active" • DUT1 & DUT2 terminate normally • DUT3 receives callback alert with single button redial • Call establish between DUT3 & DUT2 with 2-way audio • Call terminate normally • Call establish between DUT1 & Local SCCP Phone1 with 2-way audio • DUT2 hears a busy tone & displays "Callback Active" • DUT1 & Local SCCP Phone1 terminate normally • DUT2 receives callback alert with single button redial • Call establish between Local SCCP Phone1 & DUT2 with 2-way audio • Call terminated normally

	• Call establish between Local SIP Phone1 & DUT1 with 2-way audio • DUT2 hears a busy tone & displays "Callback Active" • Local SIP Phone1 & DUT1 terminate normally • DUT2 receives callback alert with single button redial • Call establish between Local SIP Phone1 & DUT2 with 2-way audio • Call terminate normally • Call establish between DUT2 & PSTN1 with 2-way audio • DUT1 hears a busy tone & displays "Callback Active" • DUT1 & PSTN1 terminate normally • DUT1 receives callback alert with single button redial • Call establish between DUT2 & PSTN1 with 2-way audio • Call terminate normally • 14 CDR(s)s retrieved • Selected fields in CDR(s) match calls
Observations	**Pass** Must enable presence subscription on Trunk Security Profile for intra-cluster callback. PSTN portion of test case (Steps 16-19) are not possible in current lab setup and not realistic to IRL application and ergo have been skipped.

6.2.19 Functional Test: Barge

Test Case Details	
Title	Functional Test: Barge
Description	Verify "Barge" call using DUT(s), SCCP, SIP and PSTN endpoints

Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1;
	• Remote CUCM ➔DUT:DUT3;
	• PSTN: PSTN1
	• Cluster-wide Service Parameters:
	  · Built In Bridge Enable➔On
	  · Party Entrance Tone➔True
	• Device➔Phone➔DN:
	  · Shared line DN:1901 added to devices with DN:DUT1, DUT2,Local SCCP Phone1, & Local SIP Phone1;
	  · Privacy on Phones with shared lines➔Off
	  · Single Button Barge➔Barge

Procedure	1. DUT3 dials 1901 2. 1901 answers (Shared line on DUT1)

	3. DUT2 hits line 1901 and selects barge after 20s 4. DUT3 goes on-hook after 60s 5. DUT1 dials 1901 6. 1901 answers (Shared line on Local SCCP Phone1) 7. DUT2 hits line 1901 and selects barge after 20s 8. Local SCCP Phone1 goes on-hook after 60s 9. DUT3 dials 1901 10. 1901 answers (Shared line on Local SIP Phone1) 11. DUT1 hits line 1901 and selects barge after 20s 12. DUT1 goes on-hook after 60s 13. DUT3 goes on-hook after 70s 14. PSTN1 dials 1901 15. 1901 answers (Shared line on DUT1) 16. DUT2 hits line 1901 and selects barge after 20s 17. PSTN1 goes on-hook after 60s 18. Retrieve CDR from CUCM 19. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT3 & 1901 with 2-way audio • All 3 parties conference-in with 3-way audio • Barged conference terminate normally • Call establish between DUT1 & 1901 with 2-way audio • All 3 parties conference-in with 3-way audio • Barged conference terminate normally • Call establish between DUT3 & 1901 with 2-way audio • All 3 parties conference with 3-way audio • Barged conference terminate normally • Final call terminate normally • Call establish between PSTN1 & 1901 with 2-way audio • All 3 parties conference with 3-way audio • Barged conference terminate normally • 8 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Not Supported** DUT does not support Barge.

6.2.20 Functional Test: cBarge

Test Case Details	
Title	Functional Test: cBarge

Description	Verify "cBarge" call using DUT(s), SCCP, SIP and PSTN endpoints

Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1;
	• Remote CUCM ➔DUT:DUT3;
	• PSTN: PSTN1
	• Enable Barge feature by setting Cluster-wide Service Parameters:
	  · Built In Bridge Enable➔On
	  · Party Entrance Tone➔True
	• Device➔Phone➔DN:
	  · Shared line DN:1901 added to devices with DN:DUT1, DUT2,Local SCCP Phone1, & Local SIP Phone1;
	  · Privacy on Phones with shared lines➔Off
	  · Single Button Barge➔cBarge

Procedure	1. DUT3 dials 1901, 1901 answers (Shared line on DUT1) 2. DUT2 selects line 1901 after 20s 3. 3.1901 goes on-hook after 60s (Shared line on DUT1) 4. DUT3 goes on-hook after 80s 5. PSTN1 dials 1901, 1901 answers (Shared line on Local SCCP Phone1) 6. Local SIP Phone1 selects line 1901 after 20s 7. DUT1 selects line 1901 Select 1901 after 30s 8. 1901 goes on-hook after 60s (Shared line on DUT1) 9. Remaining 3 parties go-hook after 120s 10. Retrieve CDR from CUCM 11. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT3 & 1901 with 2-way audio • All 3 parties conference-in with 3-way audio • cBarge conference terminate normally • Call between PSTN1 & 1901 with 2-way audio • All 4 parties conference-in with 4-way audio • DUT1 terminated conference 1st • 3 parties terminate cBarge conference after 120s • 11 CDR(s) retrieved • Selected fields in CDR match calls
Observations	**Not Supported** DUT does not support cBarge.

6.2.21 Functional Test: Shared Line – Hold/Resume

Test Case Details	
Title	Functional Test: Shared Line – Hold/Resume
Description	Verify " Hold/Resume" call on a shared line using DUT(s), SCCP and SIP endpoints
Test Setup	• Local CUCM→DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM →DUT:DUT3; • Shared line DN:1901 added to devices with DN:DUT2,Local SCCP Phone1, & Local SIP Phone1; • Privacy on Phones with shared lines→Off
Procedure	1. DUT1 dials 1901, 1901 answers (Shared Line on DUT2) 2. 1901 hits "Hold" softkey after 30s 3. 1901 hits "Resume" softkey after 30s 4. 1901 goes on-hook after 80s 5. DUT1 dials 1901, 1901 answers (Shared Line on Local SCCP Phone1) 6. 1901 hits "Hold" softkey after 30s 7. 1901 hits "Resume" softkey after 30s 8. DUT1 goes on-hook after 80s 9. DUT1 dials 1901, 1901 answers (Shared Line on Local SIP Phone1) 10. 1901 hits "Hold" softkey after 30s 11. 1901 hits "Resume" softkey after 30s 12. 1901 goes on-hook after 80s 13. Retrieve CDR from CUCM 14. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT1 & 1901 with 2-way audio • DUT1 is On-Hold (tone or silence) • DUT1 & 1901 resume call • Call terminate normally • Call establish between DUT1 & 1901 with 2-way audio • DUT1 is On-Hold (tone or silence) • DUT1 & 1901 resume call • Call terminate normally • Call establish between DUT1 & 1901 with 2-way audio • DUT1 is On-Hold (tone or silence)

	• DUT1 & 1901 resume call • Call terminate normally • 4 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Not Supported** DUT does not support Shared Lines.

6.2.22 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):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT3; • Jabber for Windows (Device➔Phone➔Add New➔CSFUSER1:DN:1922; End User:juser01/123456) • Windows PC with Jabber clients installed
Procedure	1. DUT1 dials 1922 (Duration=30s) 2. 1922 dials DUT2 (Duration=30s) 3. DUT3 dials 444-1922 (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**

6.2.23 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):DUT1 & DUT2; SCCP :Local SCCP Phone1; SIP: Local SIP Phone1; CIPC:1940; • Remote CUCM ➔DUT:DUT3; • Launch IP Communicator on a Windows PC
Procedure	1. DUT1 dials 1940 (Duration=30s) 2. 1940 dials DUT2 (Duration=30s) 3. DUT3 dials 444-1940 (Duration=30s) 4. 1940 dials 234-DUT3 (Duration=30s) 5. Calling and Called party goes on-hook alternatively 6. Retrieve CDR from CUCM 7. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• 4 calls establish with 2-way audio • 4 calls terminate normally • 4 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass

6.2.24 Functional Test: Video Endpoints

Test Case Details	
Title	Functional Test: Video Endpoints
Description	Verify video calls originating & terminating to DUT(s) endpoints
Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2; • Remote CUCM ➔DUT:DUT3; • Video Capable phone DN: 2003
Procedure	1. DUT1 dials 2003 (Duration=30s) 2. 2003 dials DUT2 (Duration=30s) 3. DUT3 dials 444-2003 (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 • If DUT is video-capable, 2-way video/audio streaming occurs from

	both devices with acceptable quality • 3 calls terminate normally • 3 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Not Supported** DUT does not support Video.

6.2.25 Functional Test: Extension Mobility

Test Case Details	
Title	Functional Test: Extension Mobility
Description	Verify DUT(s) supports "Extension Mobility" call
Test Setup	• Local CUCM→DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM →DUT:DUT3; SCCP DN: Remote SCCP Phone1; SIP DN: Remote SIP Phone1; • PSTN: PSTN1 • Extension Mobility Service activated & started • Extension Mobility Service provisioned : Device→Device Settings→Phone Service→Add New→Extension Mobility • Create Virtual Device Profile: Device→Device Profile→Add New→EM_DUT1 with DN:1934 • Extension Mobility enabled on DUT1: Device→Phone→DUT1→Enable Extension Mobility checked • Extension Mobility Service subscribed on DUT1:Device→Phone→Select "Subscribe/Unsubscribe Services" →EM • Create User/PIN: emuser01/123456; Associate device profile EM_DUT1 to user under Extension Mobility; EMCC checked;
Procedure	1. DUT1 hits "Services" button and selects EM service 2. DUT1 logs in with "emuser01/123456" 3. 1934 dials Local SCCP Phone1, Local SCCP Phone1 answers and 1934 on-hook after 30s 4. Local SIP Phone1 dials 1934, 1934 answers and Local SIP Phone1 on-hook after 30s 5. DUT2 dials 1934, 1934 answer and DUT2 on-hook after 30s 6. 1934 dials 234-Remote SCCP Phone1, Remote SCCP Phone1 answers and 1934 on-hook after 30s

	7. Remote SIP Phone1 dials 1934, 1934 answers and Remote SIP Phone1 on-hook after 30s 8. 1934 dials PSTN1, PSTN1 answers 9. PSTN1 goes on-hook after 30s 10. 1934 hits "Services" button and selects EM service 11. 1934 logs out 12. Retrieve CDR from CUCM 13. Check Calling, Called, Duration, Origination & Termination 14. Cause Codes
Expected Results	• Login successful – phone rebooted with DN:1934 • 6 calls establish with 2-way audio • All calls terminate normally • 1934 logs out and device rebooted to DUT1 device profile • 6 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Not Supported** DUT does not support Extension Mobility.

6.2.26 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(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM →DUT:DUT3; SCCP DN: Remote SCCP Phone1; SIP DN: Remote SIP Phone1; • PSTN: PSTN1 • Hunt Group Pilot 3000 (1st member-DUT2; 2nd member-Local SCCP Phone1; 3rd member-Local SIP Phone1;), Queuing flag enabled, max. waiting timer=60 secs, • Route call to Destination=DUT3;
Procedure	1. DUT1 dials 3000, DUT2 answers and DUT1 on-hook after 60s 2. DUT2 dials Local SIP Phone1, Local SIP Phone1 answers 3. DUT1 dials 3000, Local SCCP Phone1 answers and Local SCCP Phone1 on-hook after 60s 4. Local SIP Phone1 goes on hook after 70s

	5. DUT2 dials Local SCCP Phone1, Local SCCP Phone1 answers 6. DUT1 dials 3000, Local SIP Phone1 answers and DUT1 on-hook after 60s 7. PSTN1 dials 3000➔Local SIP Phone1 answers 8. Local SIP Phone1 goes on-hook after 60s 9. Local SCCP Phone1 goes on-hook 10. Retrieve CDR from CUCM 11. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call route to hunt group member DUT2 • Call establish between DUT1 & DUT2 with 2-way audio • Call terminate normally • DUT2 & Local SIP Phone1 members are busy • Call route to hunt group member Local SCCP Phone1 • Call establish between DUT1 & Local SCCP Phone1 with 2-way audio • Call terminate normally • DUT2 & Local SCCP Phone1 members are busy • Call route to hunt group member Local SIP Phone1 • Call establish between DUT1 & Local SIP Phone1 with 2-way audio • Call terminate normally • Call route to hunt group member Local SIP Phone1 • Call establish between PSTN1 & Local SIP Phone1 with 2-way audio • Call terminate normally • 6 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Pass**

6.2.27 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(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT3; SCCP DN: Remote SCCP Phone1; SIP DN: Remote SIP Phone1; • PSTN: PSTN1

	• Hunt Group Pilot 3010 (1st member-DUT2), Queuing flag enabled, max. waiting timer=60 secs, • Call Routing→Route/Hunt→Hunt Pilot→3010→Route call to this destination→234-DUT3; • Call Routing→Route/Hunt→Hunt Pilot→3012→Route call to this destination→Local SCCP Phone1; • Call Routing→Route/Hunt→Hunt Pilot→3013→Route call to this destination→Local SIP Phone1; • Call Routing→Route/Hunt→Hunt Pilot→3014→Route call to this destination→9PSTN1;
Procedure	1. DUT2 stays off-hook to make it unavailable 2. DUT1 dials 3010, DUT3 answers and DUT3 on-hook after 60s 3. DUT1 dials 3012, Local SCCP Phone1 answers and Local SCCP Phone1 on-hook after 60s 4. DUT1 dials 3013, Local SIP Phone1 answers and DUT1 on-hook after 60s 5. DUT1 dials 3014, PSTN1 answers 6. PSTN1 goes on-hook after 60s 7. DUT2 goes on-hook 8. Retrieve CDR from CUCM 9. 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 DUT3 • Call establish between DUT1 & DUT3 with 2-way audio • Call terminate normally • Call route to hunt group alternate destination Local SCCP Phone1 • Call establish between DUT1 & Local SCCP Phone1 with 2-way audio • Call terminated normally • Call routed to hunt group alternate destination Local SIP Phone1 • Call establish between DUT1 & Local SIP Phone1 with 2-way audio • Call terminated normally • Call routed to hunt group alternate destination PSTN1 • Call establish between DUT1 & PSTN1 with 2-way audio • Call terminated normally • 4 CDR(s) retrieved

	• Selected fields in CDR match calls
Observations	Pass

6.2.28 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(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT3; SCCP DN: Remote SCCP Phone1; SIP DN: Remote SIP Phone1; • PSTN: PSTN1 • Hunt Group Pilot 3015 (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. DUT1 dials 3015, Local SIP Phone1 answers 2. DUT2 dials 3015 3. DUT3 dials 3015 4. DUT1 goes on-hook after 200 secs 5. Retrieve CDR from CUCM 6. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call route to hunt group member Local SIP Phone1 • Call establish between DUT1 & Local SIP Phone1 with 2-way audio • DUT2 & DUT3 waiting in queue • Maximum number of callers in queue exceeded • Maximum wait timer exceeded 60s • Both calls (DUT3 & DUT2) were not terminated to hunt group • 3 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass

6.2.29 Functional Test: Secure Endpoint

Test Case Details	
Title	Functional Test: Secure Endpoint

Description	Verify “Authenticated” call between DUT(s), SCCP, SIP and PSTN endpoints
Test Setup	• Local CUCM→DUT(s):DUT1 & DUT2 (Authenticated); · SCCP: Local SCCP Phone1; SIP: Local SIP Phone1 (Non-Secure); · SCCP: Local SCCP Phone2; SIP: Local SIP Phone2 (Authenticated); • Remote CUCM →DUT:DUT3 (Non-Secure); • PSTN: PSTN1; • Enterprise Parameter: Cluster Security Mode→1 (Mixed Mode) • Assign authenticated Phone Security Profiles to devices: · DUT1 & DUT2:Device→Phone→DN→Device Security Profile→ 3rd_Party_SIP_Advanced_Secure_Authenticated · Local SCCP Phone2: Device Security Profile==>7975_SCCP_Authenticated · Local SIP Phone2: Device Security Profile→8945_SIP_Authenticated
Procedure	1. DUT1 dials DUT2, DUT2 answers and DUT1 on-hook after 30s 2. DUT1 dials DUT3, DUT3 answers and DUT3 on-hook after 30s 3. DUT1 dials Local SCCP Phone2, Local SCCP Phone2 answers and Local SCCP Phone2 on-hook after 30s 4. Local SIP Phone2 dials DUT2, DUT2 answers and DUT2 on-hook after 30s 5. Local SCCP Phone1 dials DUT1, DUT1 answers and DUT1 on-hook after 30s 6. DUT2 dials Local SIP Phone1, Local SIP Phone1 answers and Local SIP Phone1 on-hook after 30s 7. DUT1 dials PSTN1, PSTN1 answers 8. DUT1 goes on-hook after 30s 9. Retrieve CDR from CUCM 10. Check the Calling, Called, Duration, Secured Status, Origination & termination Cause Codes
Expected Results	• Authenticated call between DUT1 & DUT2 with 2-way audio • Call terminate normally • Non-Secure call between DUT1 & DUT3 with 2-way audio • Call terminate normally • Authenticated call between DUT1 & Local SCCP Phone2 with 2-way audio • Call terminate normally

	• Authenticated call between Local SIP Phone2 & DUT2 with 2-way audio • Call terminate normally • Non-Secure call between DUT1 & Local SCCP Phone1 with 2-way audio • Call terminate normally • Non-Secure call between Local SIP Phone1 & DUT2 with 2-way audio • Call terminate normally • Non-Secure call between PSTN1 & DUT2 • Call terminate normally • 7 CDR(s) retrieved • Selected fields in CDR (s) match calls Note: callSecuredstatus in CDR = 1 callSecuredstatus in CDR = 0 for unsecured calls
Observations	**Pass** DUT does not support Authenticated due to CUCM using Null/SHA encryption. DUT requires higher-grade encryption method. DUT and Cisco phones set for Encrypted. While TLS & SRTP was successfully negotiated between DUT devices and Cisco devices, CDR still shows calls involving DUT as '0' due to DUT being listed as 'not trusted'. User must also enable 'Use Local Contact Port As Source Port for TCP/TLS Connections" for 'TSIP' & 'SIPS' on DUT.

6.2.30 Functional Test: Join Across Lines

Test Case Details	
Title	Functional Test: Join Across Lines
Description	Verify "Join Across Lines" calls between DUT(s), SCCP, SIP and PSTN endpoints
Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT3; • PSTN: PSTN1; • Enable JAL for all phones :Device➔Phone➔DN➔Join across Lines➔On • Shared Line 1901 assigned to DUT2, Local SCCP Phone1, & Local SIP Phone1: Device➔Phone➔DN➔2nd line➔DN=1901
Procedure	1. DUT1 dials 1901, 1901 answers (Shared Line on DUT2) 2. DUT3 dials DUT2, DUT2 answers

	3. DUT2 selects line 1901 and hits softkey “Join” 4. DUT3 goes on-hook after 120s 5. DUT1 dials 1901, 1901 answers (Shared Line on Local SCCP Phone1) 6. DUT2 dials Local SCCP Phone1, Local SCCP Phone1 answers 7. selects line 1901 and hits softkey “Join” 8. DUT2 goes on-hook after 120s 9. DUT1 dials 1901, 1901 answers (Shared Line on Local SIP Phone1) 10. DUT3 dials Local SIP Phone1, Local SIP Phone1 answers 11. Local SIP Phone1 selects line 1901 and hits softkey “Join” 12. DUT1 goes on-hook after 120s 13. PSTN1 dials 1901 14. 1901 answers (Shared line on DUT2) 15. DUT3 dials DUT2, DUT2 answers 16. DUT2 selects line 1901 and hits softkey “Join” 17. PSTN1 goes on-hook after 120s 18. Retrieve CDR from CUCM 19. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT1 & 1901 with 2-way audio • 1901 is On-Hold (MOH) • Call establish between DUT2 & DUT3 with 2-way audio • DUT1 & DUT3 joined in a call, DUT2 drops from call • Call terminate normally • Call establish between DUT1 & 1901 with 2-way audio • 1901 is On-Hold (MOH) • Call establish between DUT2 & Local SCCP Phone1 with 2-way audio • DUT1 & DUT2 joined in a call, Local SCCP Phone1 drops from call • Call terminate normally • Call establish between DUT1 & 1901 with 2-way audio • 1901 is placed on-hold (MOH) • Call establish between DUT2 & Local SIP Phone1 with 2-way audio • DUT1 & DUT3 join in a call. Local SIP Phone1 drops from call • Call terminate normally • Call establish between PSTN1 & 1901 with 2-way audio • 1901 is placed on-hold (MOH) • Call establish between DUT3 & DUT2 with 2-way audio • PSTN1 & DUT3 join in a call. DUT2 drops from call

	• Call terminate normally • CDR(s)s retrieved • Selected fields in CDR(s) match call
Observations	**Not Supported** DUT does not support Join Across Lines/JAL.

6.2.31 Functional Test: Hotline

Test Case Details	
Title	Functional Test: Hotline
Description	Verify "Hotline" calls between DUT(s), SCCP, SIP and PSTN endpoints
Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT3; • PSTN: PSTN1; • Hotline Configuration to dial out 234-DUT3: · Call Routing➔ Class of Control ➔Partition➔Add New➔pt_hotline_DUT3 · Call Routing➔ Class of Control ➔Calling Search Space➔Add New➔css_hotline_DUT3 · Call Routing➔Translation Pattern➔Add New: ➢ Translation Pattern➔ blank ➢ Partition➔pt_hotline_DUT3 ➢ CSS➔css_hotline_DUT3 ➢ Called Party Transform Mask➔DUT3 • Assign DUT1, Local SCCP Phone1, Local SIP Phone1 a CSS for Intercluster Hotline: Device➔Phone➔DN➔CSS➔css_hotline_DUT3 • Assign DUT2 a CSS for PSTN Hotline: Device ➔ Phone ➔ DN ➔CSS➔ css_hotline_PSTN1 to DN:DUT2
Procedure	1. DUT1 goes off-hook, DUT3 rings & answers 2. DUT1 goes on-hook after 30s 3. Local SCCP Phone1 goes off-hook, DUT3 rings & answers 4. Local SCCP Phone1 goes on-hook after 30s 5. Local SIP Phone1 goes off-hook, DUT3 rings & answers 6. DUT3 goes on-hook after 30s 7. DUT2 goes off-hook, PSTN1 rings & answers 8. DUT2 goes on-hook after 30

	9. Retrieve CDR from CUCM 10. Check the Calling, Called, Duration, Origination & Termination 11. Cause Codes
Expected Results	• DUT3 ringing • Call establish between DUT1 & DUT3 with 2-way audio • Call terminate normally • DUT3 ringing • Call establish between Local SCCP Phone1 & DUT3 with 2-way audio • Call terminate normally • DUT3 ringing • Call establish between Local SIP Phone1 & DUT3 with 2-way audio • Call terminate normally • PSTN1 ringing • Call establish between DUT2 & PSTN1 with 2-way audio • Call terminated normally • 4 CDR()s retrieved • Selected fields in CDR match calls
Observations	**Not Supported** DUT does not support Hotline. DUT handsets do not go 'off-hook'.

6.2.32 Functional Test: Group Pickup

Test Case Details	
Title	Functional Test: Group Pickup
Description	Verify "Group Pickup" calls between DUT(s), SCCP, SIP and PSTN endpoints
Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT3; • PSTN: PSTN1; • Group Pickup configured on all phones; Group: Sales (DN: DUT1 & Local SCCP Phone1); Group: TAC (DN: DUT2 & Local SIP Phone1);
Procedure	1. DUT3 dials DUT1 2. Local SCCP Phone1 goes off-hook, hits "Group Pickup" softkey 3. Local SCCP Phone1 enters Sales Group_Pickup DN:3005 4. Local SCCP Phone1 goes on-hook after 60s 5. DUT3 dials Local SIP Phone1 6. DUT2 goes off-hook, hits "Group Pickup" softkey

	7. DUT2 enters TAC Group_Pickup DN:3006 8. DUT3 goes on-hook after 60s 9. PSTN1 dials Local SCCP Phone1 10. DUT1 goes off-hook, hits “Group Pickup” softkey 11. DUT1 enters Sales Group_Pickup DN:3005 12. PSTN1 goes on-hook after 60s 13. Retrieve CDR from CUCM 14. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• DUT1 in alerting state • Call establish between DUT3 & Local SCCP Phone1 with 2-way audio • Call terminate normally • Local SIP Phone1 in alerting state • Call establish between DUT3 & DUT2 with 2-way audio • Call terminate normally • Local SCCP Phone1 in alerting state • Call establish between PSTN1 & DUT1 with 2-way audio • Call terminate normally • 6 CDR(s) retrieved • Selected fields in CDR match calls
Observations	**Pass** Groups must be associated to each other on CUCM Group Pickup section. Auto Call Pickup Enabled must be set to “True” in Service Parameters. DUT uses unique internal Call Service URI star code to activate Group Pickup/

6.2.33 Functional Test: Do Not Disturb (DND)

Test Case Details	
Title	Functional Test: Do Not Disturb (DND)
Description	Verify “Do Not Disturb Ringer Off ” feature is supported for DUT(s) endpoints

Test Setup	• Local CUCM→DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1;
	• Remote CUCM →DUT:DUT3;
	• PSTN: PSTN1;
	• Enable DND on DN:DUT2
	  · Service Parameters→BLF Status Depicts DND →True
	  · Device→Device Settings > Softkey Template, add Do Not

	Disturb to a softkey template (Alerting and Connected state) · Device➔Phone➔DN:DUT2: ➢ Do Not Disturb➔checked ➢ DND Option: Ringer Off ➢ DND Incoming Call Alert: Flash Only
Procedure	1. DUT1 dials DUT2, DUT1 goes on-hook after 5 secs 2. DUT3 dials DUT2, DUT3 goes on-hook after 5 secs 3. Local SCCP Phone1 dials DUT2, Local SCCP Phone1 goes on-hook after 5 secs 4. Local SIP Phone1 dials DUT2, Local SIP Phone1 goes on-hook after 5 secs 5. PSTN1 dials DUT2, PSTN1 goes on-hook 6. Retrieve CDR from CUCM 7. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• DUT2 flashes to indicate incoming call • Called party hears a ring back tone • DUT2 is given an option to answer call • Call terminated by Called party • 5 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Pass** DND configured at DUT. DUT does not support CUCM features for DND. DECT handsets send star code commands to DECT station to initiate DND locally.

6.2.34 Functional Test: Do Not Disturb (DND)

Test Case Details	
Title	Functional Test: Do Not Disturb (DND)
Description	Verify "Do Not Disturb Call Reject " feature is supported on DUT(s) endpoints
Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1, Local SCCP Phone2; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT3; • PSTN: PSTN1; • Enable DND on:DUT1

	· Device➔Phone➔DN:DUT1➔update BLF Status Depicts DND➔True · Device➔Device Settings > Softkey Template, add Do Not Disturb to a softkey template *Alerting and Connected state · Device➔Phone➔DN:DUT2: ➢ Do Not Disturb➔checked ➢ DND Option: Call Reject ➢ DND Incoming Call Alert: Beep Only
1. Procedure	1. Local SCCP Phone2 dials DUT1, DUT1 answers 2. DUT2 dials DUT1, DUT1 hits "DND" softkey in connected state 3. DUT2 goes on-hook 4. DUT3 dials DUT1, DUT1 hits "DND" in connected state 5. DUT3 goes on-hook 6. Local SCCP Phone1 dials DUT1, DUT1 hits "DND" softkey in connected state 7. Local SCCP Phone1 goes on-hook 8. Local SIP Phone1 dials DUT1, DUT1 hits "DND" softkey in connected state 9. Local SIP Phone1 goes on-hook 10. PSTN1 dials DUT1, DUT1 hits "DND" in connected state 11. Local SCCP Phone2 goes on-hook after 200s 12. Retrieve CDR from CUCM 13. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between Local SCCP Phone2 & DUT1 with 2-way audio • DUT1 hears ringback for all incoming calls in connected state • CUCM rejects call with Reason: User Busy • DUT1 hears a beep for all the rejected calls • 5 calls terminate with User Busy • 1st call terminate normally • 6 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Not Applicable** DUT does not support CUCM features for DND. DECT handsets send star code commands to DECT station to initiate DND locally. DUT handsets cannot enable DND while receiving active call. Performing DND test would

	be repeating previous test.

6.2.35 Functional Test: iDivert

Test Case Details	
Title	Functional Test: iDivert
Description	Verify "IDivert " call between DUT(s), SCCP, SIP and PSTN endpoints

Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1;
	• Remote CUCM ➔DUT:DUT3;
	• PSTN: PSTN1;
	• VM enabled on all phones; VM Pilot # 7000; Device➔Phone➔DN➔Line➔Voicemail➔Default
	• Enable iDivert on DN:DUT2
	  · Service Parameter➔Legacy Immediate Divert➔True
	  · Device➔Device Settings➔ Softkey Template➔SIP_EP_User (Add iDivert to template - Connected, On Hold & Ring states)
	  · Device➔Phone➔DN:DUT2➔Softkey Template➔SIP_EP_User

Procedure	1. DUT1 dials DUT2, DUT2 hits "iDivert" softkey during ringing state 2. DUT1 leaves a voicemail and goes on-hook 3. DUT3 dials DUT2, DUT2 hits "iDivert" softkey in ringing state 4. DUT3 goes on-hook without leaving a message 5. Local SCCP Phone1 dials DUT2, DUT2 hits "iDivert" softkey during ringing state 6. Local SCCP Phone1 leaves a voicemail and goes on-hook 7. Local SIP Phone1 dials DUT2, DUT2 hits "iDivert" softkey during ringing state 8. Local SIP Phone1 leaves a voicemail and goes on-hook 9. PSTN1 dials DUT2 10. DUT2 hits "iDivert" softkey during ringing state 11. PSTN1 leaves a voicemail and goes on-hook 12. Retrieve CDR from CUCM 13. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• DUT1 directed to DUT2's voicemail box • DUT3 directed to DUT2's voicemail box

	• Local SCCP Phone1 directed to DUT2's voicemail box • Local SIP Phone1 directed to DUT2's voicemail box • PSTN1 directed to DUT2's voicemail box • MWI "On" when voicemail present for DUT2 • DUT2 able to retrieve all 4 voicemails • MWI "Off" only after 4th voicemail was retrieved • All calls terminate normally • 5 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Not Supported** DUT does not support iDivert.

6.2.36 Functional Test: CFA & iDivert

Test Case Details	
Title	Functional Test: CFA & iDivert
Description	Verify "CFA" & "iDivert " call between DUT(s), SCCP, SIP and PSTN endpoints

Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1;
	• Remote CUCM ➔DUT:DUT3;
	• PSTN: PSTN1;
	• VM enabled on all phones; CFA enabled on DUT2➔DUT1; VM Pilot # 7000;
	• Enable iDivert on DN:DUT1
	    · Legacy Immediate Divert Service Parameter Set to True
	    · Device➔Device Settings➔Softkey Template➔SP_EP_User (Add iDivert to template - Connected, On Hold & Ring states)
	    · Device➔Phone➔DN:DUT1➔Softkey Template➔SIP_EP_User

Procedure	1. DUT3 dials DUT2, DUT1 hits "iDivert" softkey in ringing state 2. DUT3 leaves voicemail and goes on-hook 3. Local SCCP Phone1 dials DUT2, DUT1 answers 4. DUT1 hits "iDivert" softkey during connected state (after 10s) 5. Local SCCP Phone1 leaves a voicemail and goes on-hook 6. Local SIP Phone1 dials DUT2, DUT1 answers 7. DUT1 hits "iDivert" softkey during connected state (after 20s) 8. Local SIP Phone1 leaves a voicemail and goes on-hook

	9. PSTN1 dials DUT2, DUT1 answers 10. PSTN1 goes on-hook without leaving voicemail 11. Retrieve CDR from CUCM 12. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• DUT2 call forward to DUT1 (ringing state) • DUT3 directed to DUT1's voicemail box • Call terminate normally • Call establish between Local SCCP Phone1 & DUT1 with 2-way audio • Local SCCP Phone1 directed to DUT1's voicemail box after 10s • Call terminate normally • Call establish between Local SIP Phone1 & DUT1 with 2-way audio • Local SIP Phone1 directed to DUT1's voicemail box after 20s • Call terminate normally • Call establish between PSTN1 & DUT1 with 2-way audio • PSTN1 directed to DUT1's voicemail box after 20s • Call terminate normally • MWI "On" when a voicemail is left on DUT1's mailbox • DUT1 was able to retrieve all 3 voicemails • MWI "Off" only after 3rd message was retrieved • 5 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Not Supported** DUT does not support iDivert.

6.2.37 Functional Test: MCID

Test Case Details	
Title	MCID
Description	Verify DUT(s) is able to mark a call malicious
Test Setup	• Local CUCMDUT(s):DUT1 & DUT 2; SCCP: SCCP phone 1; SIP: SIP phone 1 • **CUCM Global Parameter Settings** • Local CUCM Phones – DUT1 ; SIP : ; • Assign a softkey template that includes MCID button to all phones
Procedure	1. DUT1 dials Local SIP Phone 1 2. Local SIP Phone 1 answers 3. Local DUT 1 hits "MCID" softkey 4. Local SIP Phone 1 goes on-hook after 60s

<table>
<tr><td></td><td>5. Repeat steps 1-4 using SCCP phone1
6. Retrieve CDR from CUCM
7. Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>

Call established between DUT 1 & Local SIP Phone 1 (talking state)
Call terminated normally
2 CDR Records retrieved
Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, origCause_Value, duration fields in the CDR table for each call
MALICIOUS CALL (Callflag) indication is displayed in the application.

<table>
<tr><th>CDR field</th><th>Call 1</th></tr>
<tr><td>Comment</td><td>callflag=MALICIOUS</td></tr>
<tr><td>origCause_Value</td><td>16</td></tr>
</table>
</td></tr>
<tr><td>Observations</td><td>Not Supported
DUT does not support MCID.</td></tr>
</table>

6.2.38 Functional Test: Mobile Connect

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Functional Test: Mobile Connect</td></tr>
<tr><td>Description</td><td>Verify DUT(s) supports Mobile Connect call to a remote SIP endpoint</td></tr>
<tr><td>Test Setup</td><td>

Local CUCM→DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1;
Remote CUCM →DUT:DUT3; SIP: Remote SIP Phone1;
Configure CUCM Service Parameter: Device Mobility Mode→On
Mobile Voice Access service running on CUCM-PUB
Mobile Voice Access enabled on Voice Gateway
Remote Cluster Single Number Reach (SNR) configured for Remote SIP Phone1 – Remote Device:444-DUT2

Add Remote Destination Profile:Device→Device Settings→Remote Destination Profile→Add New→mobile1_rdp

Userid:rcuser21
Line: Remote SIP Phone1
Add New Remote Destination:

Name→Mobility_1
Destination Number→ 444DUT2
Check “Enable Unified Mobility”, “Enable Single Number Reach” & “Enable Move to Mobile”

Device Settings→Softkey Template→SIP_EP_User→Add “Mobility” softkey (On-Hook & Connected)
User Management→End User→rcuser21→Check “Enable

</td></tr>
</table>

	Mobility" &"Enable Mobile Voice Access" · Device➔Phone➔Remote SIP Phone1➔Owner userid➔rcuser21 · Remote Device: Device➔Phone➔DUT2➔Line➔No Answer Ring Duration➔60
Procedure	1. DUT1 dials Remote SIP Phone1, DUT2 answers 2. DUT2 sends DTMF *74" after 30s 3. Remote SIP Phone1 answers 4. DUT2 goes on-hook 5. Remote SIP Phone1 hits "Mobility" softkey after 30s 6. DUT2 answers 7. Remote SIP Phone1 goes on-hook 8. DUT1 goes on-hook after 60s 9. Repeat steps 1-8 and replace the Calling DN: Local SCCP Phone1 10. Repeat steps 1-8 and replace the Calling DN: Local SIP Phone1 11. Retrieve CDR from CUCM 12. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Both Remote SIP Phone1 & DUT2 are ringing • Call establish between DUT1 & DUT2 with 2-way audio • Call is transferred to device Remote SIP Phone1 • Call establish between DUT1 & Remote SIP Phone1 with 2-way audio • Local SIP Phone1 handoff call back to DUT2 • Call restored between DUT1 & DUT2 • Final Call terminated normally • Results for SCCP and SIP call are similar as above • 6 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Not Supported** DUT does not support Mobility/Mobile Connect.

6.2.39 Functional Test: Mobile Voice Access (MVA)

Test Case Details	
Title	Functional Test: Mobile Voice Access (MVA)
Description	Verify Inbound Mobile Voice Access (MVA) calls from DUT(s) endpoints
Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1;

	• Remote CUCM ➔DUT:DUT3;
	• CUCM Service Parameter:
	  · Enable Enterprise Feature Access➔True
	  · Enable Mobile Voice Access➔True
	  · Mobile Voice Access Number➔8005555
	  · Matching Caller ID with Remote Destination➔ Partial Match
	  · Number of Digits for Caller ID Partial Match➔7
	• Mobile Voice Access service running on CUCM-PUB
	• Mobile Voice Access enabled on Voice Gateway
	• MVA # provisioned in CUCM: Media Resources➔Mobile Voice Access➔Add New➔8005555
	• Local Cluster Single Number Reach (SNR) configured for DUT2➔Remote Device:234-DUT3:
	  · Add Remote Destination Profile:Device➔Device Settings➔Remote Destination Profile➔Add New➔mobile1_rdp
	    ➢ Userid:dutuser02
	    ➢ Line:DUT2
	    ➢ Add New Remote Destination:
	      · Name➔Mobility_1
	      · Destination Number➔ 234DUT3
	      · Check “Enable Unified Mobility”, “Enable Single Number Reach” & “Enable Move to Mobile”
	  · Device Settings➔Softkey Template➔SIP_EP_User➔Add “Mobility” softkey (On-Hook & Connected)
	  · User Management➔End User➔dutuser02➔Check ”Enable Mobility” &”Enable Mobile Voice Access”
	  · Device➔Phone➔DUT2➔Owner userid➔dutuser02
	  · Remote Device: Device➔Phone➔DUT3➔Line➔No Answer Ring Duration➔60

Procedure	1. (Mobil device) dials MVA #8005555 2. Mobil User enters DUT3# or 234DUT3# 3. Mobil User enters PIN:123456# & DN:DUT1# 4. DUT1 answers 5. DUT3 sends DTMF *74 to handoff session after 30s 6. DUT1 goes on-hook after 60s 7. Retrieve CDR from CUCM 8. Check the Calling, Called, Duration, Origination & Termination Cause

	Codes
Expected Results	• Mobile user prompted for Caller ID: DUT3 • Mobile user prompted for PIN & Destination DN • DUT1 is ringing • Call establish between DUT3 & DUT1 with 2-way audio • Call hand-off to DUT2 • Call terminate normally • 1 CDR retrieved • Selected fields in CDR matches call
Observations	**Not Supported** DUT does not support MVA.

6.2.40 Functional Test: Enterprise Feature Access (EFA)

Test Case Details	
Title	Functional Test: Enterprise Feature Access (EFA)
Description	Verify Inbound Enterprise Feature Access (EFA) - Hold/Resume call from a DUT endpoint

Test Setup	• Local CUCM→DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1;
	• Remote CUCM →DUT:DUT3;
	• Service Parameter:
	    · Enable Enterprise Feature Access→True
	    · Enable Mobile Voice Access→True
	• Mobile Voice Access service running on CUCM-Publisher
	• Mobile Voice Access enabled on Voice Gateway
	• EFA # provisioned in CUCM: Call Routing→Mobility→Enterprise Feature Access Configuration→Add New→9005555
	• Local Cluster Single Number Reach (SNR) configured for DUT2→Remote Device:234-DUT3:
	    · Add Remote Destination Profile:Device→Device Settings→Remote Destination Profile→Add New→mobile1_rdp
	        ➢ Userid:dutuser02
	        ➢ Line:DUT2
	        ➢ Add New Remote Destination:
	            · Name→Mobility_1
	            · Destination Number→ 234DUT3
	            · Check “Enable Unified Mobility”, “Enable Single

	Number Reach" & "Enable Move to Mobile" · Device Settings→Softkey Template→SIP_EP_User→Add "Mobility" softkey (On-Hook & Connected) · User Management→End User→dutuser02→Check "Enable Mobility" &"Enable Mobile Voice Access" · Device→Phone→DUT2→Owner userid→dutuser02 · Remote Device: Device→Phone→DUT3→Line→No Answer Ring Duration→60
Procedure	1. DUT3 (Mobil device) dials EFA #9005555 2. Mobil user prompted to enter remote device DN DUT3# 3. Mobil User enters PIN:123456#, Option 1 & DN:DUT1# 4. DUT1 answers call 5. DUT3 sends DTMF *81 to place call on-hold after 30s 6. DUT3 sends DTMF *83 to resume call after 30s 7. DUT3 goes on-hook after 120s 8. Retrieve CDR from CUCM 9. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Mobile user prompted for Caller ID: DUT3 • Mobile user prompted for PIN • Selects option 1 and enters Called DN:DUT1 • DUT1 is ringing • Call establish between DUT3 & DUT1 with 2-way audio • DUT1 is On-Hold • Call resumes • Cal terminate normally • 1 CDR retrieved • Selected fields in CDR matches call
Observations	**Not Supported** DUT does not support EFA.

6.3 Phase 3 Negative Tests

These tests are executed to determine the ability of the impact on calls, the CUCM and the 3rd party device when combinations of the aforementioned fail by power failure or network connectivity problems.

6.3.1 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(s):DUT1 & DUT2;SCCP:Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM→DUT:DUT3;
Procedure	1. DUT1 dials DUT2, DUT2 answers 2. Local SIP Phone1 dials 234-DUT3, DUT3 answers 3. Access CUCM-PUB server via SSH (Local Cluster) 4. Enter CLI: utils system restart <CR> yes 5. Local SCCP Phone1 dials DUT2, DUT2 answers 2nd incoming call 6. Called party goes on-hook for all 3 calls 7. Repeat steps 1-2,5-6 after CUCM-PUB recovery 8. Retrieve CDR from CUCM 9. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT1 & DUT2 with 2-way audio • Call establish between Local SIP Phone1 & DUT3 with 2-way audio • CUCM-PUB is restarted • Stable calls not impacted by PUB restart • Call establish between Local SCCP Phone1 & DUT2 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 • 5 CDR(s) retrieved • Selected fields in CDR matches calls
Observations	**Pass** Tested two ways: CUCM CallManager service shut down while DUTs in active call; CUCM CallManager service shut down and DUTs register to SUB while PUB is down, then CallManager re-enabled and DUTs re-register to PUB and verify call functionality. Both were successful. DUTs active calls remained up, however DUTs had to register to SUB for DUT2 to get call from Local SCCP.

6.3.2 Negative Test: SUB Failure

Test Case Details	
Title	Negative Test: SUB Failure
Description	Verify a SUB failure should not affect stable calls on DUT(s)
Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2;SCCP:Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM➔DUT:DUT3;
Procedure	1. DUT1 dials DUT2, DUT2 answers 2. Local SIP Phone1 dials DUT3, DUT3 answers call 3. Access CUCM-SUB server via SSH (Local Cluster) 4. Enter CLI: utils system restart <CR> yes 5. Local SCCP Phone1 dials DUT2 6. Called party goes on-hook for all 3 calls 7. Repeat steps 1-2, after CUCM-SUB recovery 8. Retrieve CDR from CUCM 9. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT1 & DUT2 with 2-way audio • Call establish between Local SIP Phone1 & DUT3 with 2-way audio • CUCM-SUB is restarted • Stable calls not impacted by SUB restart • Transient calls impacted by SUB restart

	• Call between Local SCCP Phone1 & DUT2 unsuccessful • All stable calls terminate normally • CUCM-SUB is in-service • All calls successful after SUB failure recovery • 4 CDR(s) retrieved • Selected fields in CDR match calls
Observations	**Pass** DUTs registered to PUB. SUB CallManager services stopped and started. Calls were not impacted.

6.3.3 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(s):DUT1 & DUT2;SCCP:Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM➔DUT:DUT3;
Procedure	1. DUT1 dials DUT2, DUT2 answers 2. Unplug network cable from device DN:DUT1 3. Restore the network cable after 60s 4. Local SIP Phone1 dials DUT1, DUT1 answers 5. DUT1 goes on-hook after 60s 6. Retrieve CDR from CUCM 7. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT1 & DUT2 with 2-way audio • Network failure reported on device DN:DUT1 • Stable call drops • Device DUT1 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 & DUT1 with 2-way audio • Call terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Not Applicable**

	DECT handsets are registered and connected to same IP-DECT station. Cannot disconnect network from handsets without disconnecting it from both.

6.3.4 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(s):DUT1 & DUT2;SCCP:Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM➔DUT:DUT3;
Procedure	1. DUT1 dials DUT2, DUT2 answers 2. Remove power cable from DUT2 3. Restore power cable after 60s 4. Local SIP Phone1 dials DUT2, DUT2 answers call 5. DUT2 goes on-hook after 60s 6. Retrieve CDR from CUCM 7. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT1 & DUT2 with 2-way audio • DUT2 lost power • Stable call drops • Device DUT2 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 & DUT2 with 2-way audio • Call terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Pass** Tested by removing power from DECT handset connected to IP-DECT station.

6.3.5 Negative Test: Abnormal Call Scenarios

Test Case Details	
Title	Negative Test: Abnormal Call Scenarios

Description	Verify calls on DUT for negative call scenarios (Invalid DN, Busy DN, Abandoned, RNA)
Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2;SCCP:Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM➔DUT:DUT3; • Call Waiting disabled for DUT1 & DUT1 • Device Profile: Busy Trigger set to 1 for DN: DUT1 & DUT2 • Voicemail disabled for Local SCCP Phone1 & Local SIP Phone1;
Procedure	1. DUT1 dials Local SCCP Phone1, Local SCCP Phone1 answers 2. DUT2 dials DUT1 3. DUT2 dials Local SCCP Phone1 4. DUT2 dials Local SIP Phone1 (RNA) 5. DUT2 dials DUT3 6. DUT2 goes on-hook before DUT3 answers (Abandoned) 7. DUT1 goes on-hook after 120 secs 8. Retrieve CDR from CUCM 9. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT1 & Local SCCP Phone1 with 2-way audio • DUT2 hears busy tone for calls to DUT1 & Local SCCP Phone1 • DUT2 hears ring back timeout for call to Local SIP Phone1 • 1 Call terminated normally • 5 unsuccessful call attempts • 6 CDR(s) retrieved • Selected fields in CDR match calls
Observations	**Pass**

6.4 Phase 4 Miscellaneous Tests

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

6.4.1 Miscellaneous Test: Codec (G722 & G729)

Test Case Details	
Title	Miscellaneous Test: Codec (G722 & G729)
Description	Verify URI calls between DUT(s) & SIP endpoints for In-band Codec (G722, G729)
Test Setup	• Local CUCM➔DUT(s):DUT1 (dutuser01@abc.inc) ; DUT2 (dutuser02@abc.inc); SCCP: Local SCCP Phone1 (cuser01@abc.inc); SIP: Local SIP Phone1 (cuser20@abc.inc); • Remote CUCM ➔DUT:DUT3; • Go to System➔Region Information➔ Audio Codec Preference List➔ Add New➔ G722➔Select G722 Codec • Go to System➔Region Information➔ Audio Codec Preference List➔ Add New➔ G729➔Select G729ab Codec • Go to System➔Region Information➔ Region➔ Add New➔G722-Region➔G722 • Go to System➔Region Information➔ Region➔ Add New➔G729-Region➔G729 • Go to System➔Device Pool➔ Add New➔G722-dp➔Region➔G722-Region • Go to System➔Device Pool➔ Add New➔G729-dp➔Region➔G729-Region • Update DUT1, DUT2 with device pool=G722-dp • Configure Speed Dial for DUT1, DUT2, Local SIP Phone1: · Device➔Phone➔DUT1➔Add new SD➔dutuser02@abc.inc · Device➔Phone➔DUT2➔Add new SD➔cuser20@abc.inc · Device➔Phone➔Local SIP Phone1➔Add new SD➔dutuser02@abc.inc · Device➔Phone➔Local SCCP Phone1➔Add new SD➔dutuser01@abc.inc
Procedure	1. DUT1 hits Speed dial button 2. DUT2 answers call 3. DUT2 goes on-hook after 60s 4. Local SIP Phone1 hits Speed dial button

	5. DUT2 answers call 6. Local SIP Phone1 goes on-hook after 60s 7. DUT2 hits Speed dial button 8. Local SIP Phone1 answers call 9. DUT2 goes on-hook after 60s 10. Local SCCP Phone1 hits Speed dial button 11. DUT1 answers call 12. DUT1 goes on-hook after 60s 13. Repeat steps 1-9 with device pool of G720-dp for DUT1 & DUT2 14. Retrieve CDR from CUCM Server 15. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• DUT receives both the Caller ID and URI • 4 calls establish with 2 way audio for G722 codec • 4 calls terminate normally • 4 calls establish with 2 way audio for G729 codec • 4 calls terminate normally • Voice quality was good for both codec types • 8 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass DUT only supports G722.2 and CUCM does not support G722.2

6.4.2 Miscellaneous Test: DUT display features

Test Case Details	
Title	Miscellaneous Test: DUT display features
Description	Verify different packetization period support on DUT(s) endpoints
Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT3; • Configure Service Parameter: Preferred G.711 Millisecond Packet Size :10
Procedure	1. DUT1 dials DUT2, DUT2 answers 2. DUT1 goes on-hook after 60s 3. DUT1 dials DUT3, DUT3 answers 4. DUT3 goes on-hook after 60s

	5. DUT1 dials Local SCCP Phone1, Local SCCP Phone1 answers 6. DUT1 goes on-hook after 60s 7. Local SIP Phone1 dials DUT1, DUT1 answers 8. DUT1 goes on-hook after 60s 9. Repeat steps 1-8 with Packet Size=20 10. Repeat steps 1-8 with Packet Size=30 11. Retrieve CDR from CUCM Server 12. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT1 & DUT2 with 2-way audio • Call establish between DUT1 & DUT3 with 2-way audio • Call establish between DUT1 & Local SCCP Phone1 with 2-way audio • Call establish between DUT1 & Local SIP Phone1 with 2-way audio • All calls with good audio quality • All calls terminate normally • 6 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	**Pass** Configurable packet size range in DUT is 20-200ms but DUT will also adapt to 10ms packet size if negotiated by other end. 10ms packet size was not tested.

6.4.3 Miscellaneous Test: DUT Screen Features

Test Case Details	
Title	Miscellaneous Test: DUT Screen Features
Description	Verify the features displayed on the screen of DUT
Test Setup	• Local CUCM→ DUT (s):DUT1 & DUT2;
Procedure	1. Check DUT:DUT1 phone display for: 2. Missed Calls 3. Placed Calls 4. Received Calls 5. Date & Time 6. Clear Call History (Missed, Placed, Received) 7. Redial or Dial from Call History List 8. Soft keys for call features 9. Multiple lines

	10. Edit Called #
Expected Results	• Able to access all these features from the DUT's phone display
Observations	Pass DUT does not support multiple lines.

6.4.4 Miscellaneous Test: Long Duration Calls

Test Case Details	
Title	Miscellaneous Test: Long Duration Calls
Description	Verify long duration calls between DUT(s), SCCP, SIP and PSTN endpoints
Test Setup	• Local CUCM➔DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT3; • PSTN DN: PSTN1
Procedure	1. DUT1 dials DUT2, DUT2 answers (Duration: 1 Hr.) 2. Local SIP Phone1 dials DUT3, DUT3 answers (Duration: 1 Hr) 3. Repeat step 1 by replacing Called DN: PSTN1 4. Repeat step 2 by replacing the Calling DN: Local SCCP Phone1 5. Retrieve CDR from CUCM 6. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between DUT1 & DUT2 with 2-way audio • Call establish between Local SIP Phone1 & DUT3 with 2-way audio • Call establish between DUT1 & PSTN1 with 2-way audio • Call establish between Local SCCP Phone1 & DUT3 with 2-way audio • All long duration calls were stable with 2-way audio • 4 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass

6.4.5 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):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT3; • Cisco Phone Models: 6961,8861, 8945, 7925, 9971, DX650
Procedure	1. DUT1 dials Cisco IP Phone, Cisco IP Phone answers and DUT1 on-hook after 120s 2. Cisco IP Phone dials DUT1, DUT1 answers and Cisco IP Phone on-hook after 120s 3. DUT2 dials Cisco IP Phone , Cisco IP Phone answers and DUT2 hits "Hold" after 20s 4. DUT2 hits "Resume" after 20s, Cisco IP Phone on-hook after 120s 5. Cisco IP Phone dials 234-DUT3, DUT3 answers and DUT3 on-hook after 90s 6. DUT1 dials Cisco IP Phone, Cisco IP Phone answers and Cisco IP Phone hits "Transfer" after 20s 7. Cisco IP Phone dials DUT2, Cisco IP Phone hits "Transfer and Cisco IP Phone on-hook 8. DUT1 goes on-hook after 120s 9. Cisco IP Phone dials DUT1, DUT1 answers and DUT1 hits "Transfer" after 20s 10. DUT1 dials 234-DUT3, DUT3 answers 11. DUT1 hits "Transfer" after 30s, DUT1 on-hook 12. DUT3 goes on-hook after 120s 13. Cisco IP Phone dials DUT2, DUT2 answers 14. DUT2 hits "Conference" after 30s, DUT2 dials 234-DUT3 15. DUT3 answers, DUT2 hits "Conference" after 30s 16. DUT2 goes on-hook after 120s 17. Cisco IP Phone and DUT3 goes on-hook after 200s 18. Repeat steps 1-17 by replacing DN: Cisco IP Phone with DN(s) of other Cisco phone models 19. Retrieve CDR from CUCM 20. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Intra-cluster calls establish between DUT & Cisco IP Phone • Inter-cluster calls establish between DUT & Cisco IP Phone • Call Hold/Resume between DUT & Cisco IP Phone

	• Blind Transfer between DUT & Cisco IP Phone • Consult Transfer between DUT & Cisco IP Phone • Conference Call between DUT & Cisco IP Phone • CDR(s) retrieved for all the calls • Selected fields in CDR(s) match calls
Observations	Pass Tested with a 6945, 7941, 8811, 8945, 9971, & DX650.

6.4.6 Miscellaneous Test: Multiple Lines

Test Case Details	
Title	Miscellaneous Test: Multiple Lines
Description	Verify DUT is able to handle calls and mid-call features on multiple lines
Test Setup	• Local CUCM→DUT(s):DUT1 & DUT2; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM →DUT:DUT3; SIP: Remote SIP Phone1; SCCP: Remote SCCP Phone1; • Assumption: DUT is Advanced 3rd Party SIP Endpoint with multiple lines
Procedure	1. Provision all the lines for both DUT1 & DUT2 2. Device→Phone→DN→Line (Example DN Range: 7120 -7150) 3. Initiate calls on all the lines between DUT1 & DUT2 4. Calling & Called parties goes on-hook alternatively at random duration 5. Initiate intra-cluster & inter-cluster calls on all lines to SIP, SCCP & PSTN endpoints 6. Calling & Called parties goes on-hook alternatively at random Duration. 7. Initiate calls and perform mid-call features between these lines (Hold/Resume, Transfer, Conference, CFNA, CFB) 8. Retrieve CDR from CUCM 9. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• All calls establish successfully with good audio quality • Caller ID presented for all calls • Mid-call features works as designed • All calls release normally • CDR (s) retrieved

	• Selected fields in CDR match calls
Observations	**Not Supported** DUT does not support multiple lines.

6.5 Phase 5 SRST Failover Tests

These tests have been added at the request of the partner to verify SRST failover functionality between the third-party products and the Cisco phone devices.

6.5.1 SRST Failover Test: DUT Registers with SRST

Test Case Details	
Title	SRST Failover Test: DUT Registers with SRST
Description	Verify DUT is able to register to SRST when CUCM call services fail
Test Setup	• Local CUCM➔ DUT(s):DUT; SIP: Local SIP Phone; • CUCM SRST configuration set. SRST live and available for use i. Go to System➔ SRST➔ Add New

	- Name: SRST; IP Address: x.x.x.x; SIP Network/IP Address: x.x.x.x; Save ii. Go to System➔ Device Pool➔ Default➔ SRST Reference➔ SRST; Save & Apply • SRST set with proper dial-peers and configurations
Procedure	1. Go to Cisco Unified Serviceability➔ Tools➔ Control Center – Feature Services➔ cucmpub 2. Stop CallManager service
Expected Results	• Local Cisco Phone/client and DUT lose call services • Local Cisco Phone/client and DUT register with SRST, display that in Fallover Mode or some other indication that failover has happened/services have been diminished/lost. • Call services restored • DUT displays that it is registered with secondary proxy server
Observations	**Pass** We had to explicitly specify SRST IP as the secondary IP on the DECT Station.

6.5.2 SRST Failover Test: Basic Call using SRST

Test Case Details	
Title	SRST Failover Test: Basic Call using SRST
Description	Verify DUT is able to make basic local calls via SRST when CUCM call services are unavailable
Test Setup	• SRST➔ DUT(s):DUT; SIP: Local SIP Phone; • DUT & Local SIP Phone are registered with SRST • SRST set with proper dial-peers and configurations
Procedure	1. Local SIP Phone dials DUT➔ DUT answers 2. DUT on-hook after 30s 3. DUT dials Local SIP Phone➔ Local SIP Phone answers 4. Local SIP Phone on-hook after 30s
Expected Results	• DUT notified of incoming call (tone/display) • Call established between Local SIP Phone & DUT with 2-way audio • Local SIP Phone notified of incoming call (tone/display) • Call established between DUT & Local SIP Phone with 2-way audio • All calls routed through SRST

	• Calls are cleared successfully
Observations	Pass

6.5.3 SRST Failover Test: Outbound Call to PSTN

Test Case Details	
Title	SRST Failover Test: Outbound Call to PSTN
Description	Verify DUT is able to make outbound calls to PSTN via SRST when CUCM call services are unavailable
Test Setup	• SRST➔ DUT(s):DUT; • DUT is registered with SRST • SRST set with proper dial-peers and configurations
Procedure	1. Initiate call from DUT to PSTN➔ PSTN answers 2. DUT on-hook after 30s
Expected Results	• PSTN shows Caller ID of DUT • Call established between DUT & PSTN with 2-way audio • Call routed through SRST • Call is cleared successfully
Observations	Pass

6.5.4 SRST Failover Test: Inbound Call from PSTN

Test Case Details	
Title	SRST Failover Test: Inbound Call from PSTN
Description	Verify DUT is able to receive inbound calls from PSTN via SRST when CUCM call services are unavailable
Test Setup	• SRST➔ DUT(s):DUT; • DUT is registered with SRST • SRST set with proper dial-peers and configurations
Procedure	1. Initiate call from PSTN to DUT➔ DUT answers 2. PSTN on-hook after 30s
Expected Results	• DUT shows Caller ID of PSTN • Call established between PSTN & DUT with 2-way audio • Call routed through SRST • Call is cleared successfully

Observations	Pass

6.5.5 SRST Failover Test: DUT Re-registers with CUCM

Test Case Details	
Title	SRST Failover Test: Re-registers with CUCM
Description	Verify DUT is able to re-register with CUCM when call services are restored
Test Setup	• SRST➔ DUT(s):DUT; SIP: Local SIP Phone • DUT & Local SIP Phone are registered with SRST • SRST set with proper dial-peers and configurations
Procedure	1. Go to Cisco Unified Serviceability➔ Tools➔ Control Center – Feature Services➔ cucmpub 2. Start CallManager service
Expected Results	• Local Cisco Phone/client & DUT re-register with CUCM • Call services are restored • DUT displays that it is re-registered with primary proxy server
Observations	Pass