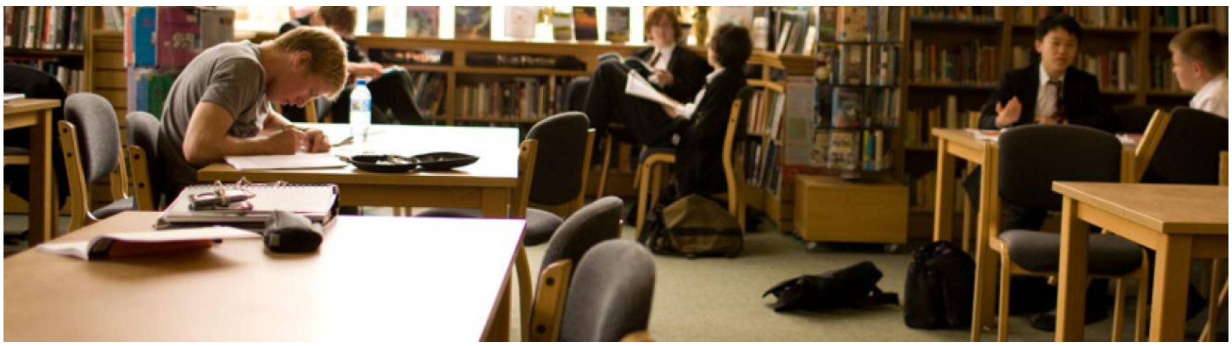
Detailed End Point IVT Test Plan and Report for Cisco Unified Communications Manager 12.5 and Ascom Myco 3



Test Date/ Result (Completed by Cisco or Authorized Test House)	09/10/19 - PASS
Partner Product Name	Myco3
Partner Product Type	SIP Endpoint, Myco 3 WiFi
Partner Product Version #	1.2.1
Cisco Product Name	CUCM
Cisco Product Version	12.5
API/Protocol(s) Used	SIP
Date Testing Completed	September 09, 2019
IVT Contact Email	Karl-Magnus.Olsson@ascom.com

Contents

Pre-Testing Information

The purpose of this section is to gather information about the 3rd party Solution Partner Program (SPP) product being submitted for Interoperability Verification Testing (IVT) in support of receiving a Cisco Compatibility logo. The information collected in this section will be used to complete customization of test plan for the product integration with Cisco product(s).

This section must be completed thoroughly to ensure that products features and requirements are properly understood and reflected appropriately in the test plan. ***The limits stated in this questionnaire will be tested. Anything (limits, functionality, interfaces) not reported in this document will not be supported.***

Complete all sections with tekVizion.

This document will be reviewed for content, completeness and appropriate integration methods by Cisco and will not be submitted for test plan generation or test scheduling until approval. This process generally takes about 10 business days, though can be more or less dependent on complexity and current demand.

IVT Pre-requisites

The following prerequisites must be complete prior to submitting a request for testing:

1) Approved application in SPP for the product pairing being submitted for test.

a) Product Pairing = Cisco Product Major Version + Partner Product Major Version

b) Cisco Product Major Version must be generally available

c) Partner Product Major Version must be generally available

2. Any use of Cisco Intellectual Property (proprietary protocols or interface methods) must have been approved by Cisco and have appropriate agreements in place. This is not appliEPle to standard published integration methods. Questions regarding interface methods should be directed to Developer Services or your Cisco Partner Manager.

Submission Instructions

Provide the requested information on the following pages for the product being submitted for Interoperability Verification Testing (IVT).

Complete Current Test Request Information, Product Category, and Product Description for all product pairings (Cisco Product + Program Member Product) being submitted. Only requests with all required sections completed

1 Interoperability Verification Testing (IVT) Overview

1.1 Interoperability Verification Testing Requirement

Successful completion of Endpoint / USB Accessory IVT is required for Partner Products to be designated as “Cisco Compatible” and for Partner Products to be listed in the Cisco Solution Marketplace.

1.2 IVT Objectives

The IVT program’s objective is to provide verification that 3rd party Partner product(s) meet the following criteria:

- Successfully Integrate and scale as defined by Cisco design guides and 3rd party product specifications
- Install and functionally operate/perform as indicated in collateral and specifications (from integration perspective only)
- Successfully integrate with Cisco products while **not adversely affecting** Cisco product operation or the integrated solution.
- Use only supported integration methods. Supported integration methods (API’s and protocols) can be found on the DevNet web site: https://developer.cisco.com/site/collaboration/overview.gsp

1.3 IVT Focus

Testing is focused on integration points of Partner products and Cisco products, not on the Partner product itself, to ensure quality integrations between 3rd party products and Cisco products.

Test categories include:

- Installation and connectivity of partner product
- Validation of integrated features between Cisco product and partner product
- Negative testing (connectivity failure, redundancy, recovery)
- Performance and load testing of integration points/functionality, using a subset of functional test scenarios

2 Instructions

Provide the requested information on the following pages for the product being submitted for Interoperability Verification Testing (IVT).

1. Complete Current Test Request Information, Product Category, and Product Description for all product pairings (Cisco Product + Program Partner Product) being submitted. Only requests with all required sections completed will be accepted. Failure to provide this information will result in the request being denied.
2. Submission:

   a) Access your Developer Dashboard, go to the Registered Products Tab and select "Actions" and "Add New IVT Request" next to the product to be submitted for IVT

   b) Upload this document to the IVT Request, failure to upload this document will result in an incomplete request

   c) Save using filename: <COMPANY_PRODUCT_VX_X+CISCO_PRODUCT_VX_X>.doc Example Filename: CiscoSystems_FASTAPP_V1_1+CIscoProduct_1_0.doc

Click on link below for detailed instructions:
http://solutionpartner.cisco.com/documents/8974369/0/DeveloperPartnerGuide.pdf

Help or questions related to SPP Portal, listings or application status::solutionpartnerprogram-support@cisco.com

General Questions: Contact your Cisco representative or send email to ivt_questions@cisco.com

3 Product and Testing Information

3.1 IVT Request info here

IVT Request ID: 5614

Cisco Technology: Collaboration - Unified Communications Call Control

Cisco Product and Version: Cisco Unified Communications Manager 12.5

4 Test Set Up and Tools

This section refers to the product test tools that have been used during the development testing of the product being submitted for IVT

Question	Response
What if any commercial test tools are used in the development and test of this product	
Can these tools and test scripts for these products be made available to support IVT	
Are there proprietary test tools that could be made available to support IVT	

5 Product Platform Description

In the table below, provide specific details on the platform/server that your product resides. If your application is an appliance, it will need to be onsite for testing; otherwise, a VM will be provided for your installation of OS and application.

	Minimum Configuration Server Requirements	Maximum Configuration Server Requirements	OS and Version
CPU	N/A	N/A	N/A
Disk	N/A	N/A	N/A
Memory	N/A	N/A	N/A
Max Users supported	N/A	N/A	N/A

5.1 Product Deployment Description

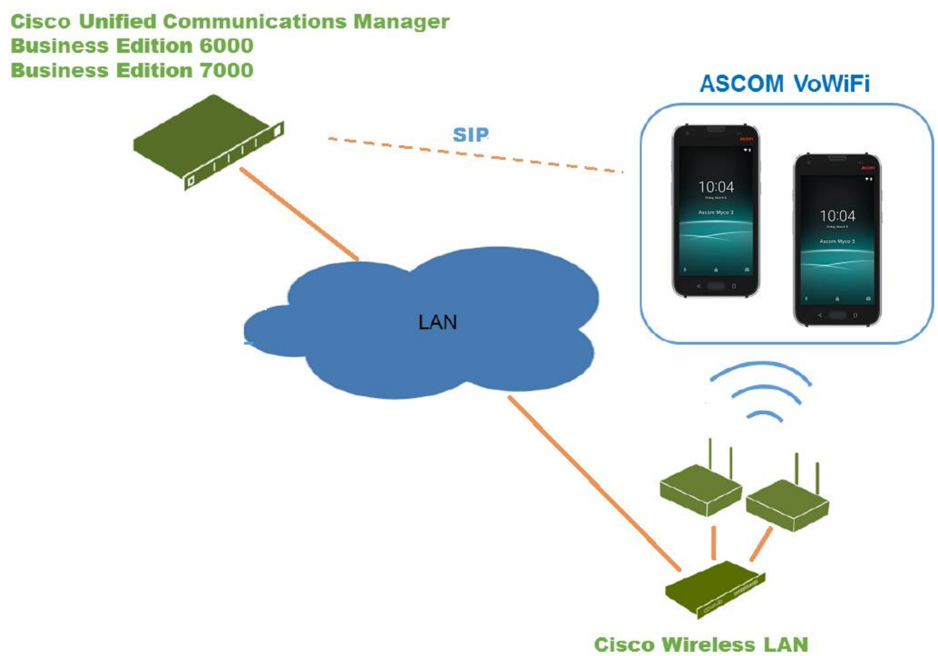
Provide the following information about the product and integration. Each of the items below is **required in order to proceed with test scheduling**.

5.2 Product Description

The Ascom Myco 3 smartphone works with supported apps to bridge digital information gaps—making it easier to communicate, coordinate and execute time-sensitive activities. A professional-grade Android™ mobile device, the Ascom Myco 3 helps streamline workflows, support faster responses and deliver context-rich information to mobile personnel.

With its HD 5” touchscreen, the Ascom Myco 3 offers an easy user interface for managing calls, alerts, photos, messages, scores, waveforms and other critical information. A true hot-swap battery helps ensure constant operation throughout long shifts. And its Android 8.1 OS and Google™ certifications give access to the world’s largest ecosystem of apps.

5.3 Product Integration Diagram



5.4 Product Integrated Use Cases

Barcode scanning: secure verification of patient IDs when administering medications.

Enabling clinicians to access and share lab results, images, requests and alerts while on the go.

Communicate and coordinate effectively. Requests, messages, alerts and tasks go directly to assigned recipients.

6 Test Plan

6.1 Introduction

This document is the detailed Interoperability Verification Test Plan and Report for Cisco Unified Communications Manager and Endpoint/USB Accessory partner product.

6.2 Entry Criteria

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

6.3 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.

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

Table 1. Defect Severity Level

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 is blocked due to this issue, then testing is considered complete and the devices under test will not receive a Compatibility Logo.

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 case with Cisco Developer Services. Partner should provide the Developer Services 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. The Cisco approval process may increase/decrease the severity level of the defect after the test cycle if considered **necessary**.

7 Executive Summary

Short summary of the test effort, summarizing the lab findings during testing.

The following summarizes results:

- Test Case Failures:
  - None
- Features Not Supported:
  - EP-5 – Functional Test: SIP URI
  - EP-6 – Functional Test: SIP URI
  - EP-14 – Functional Test: Conference Call
  - EP-15 – Functional Test: Call Park
  - EP-16 – Functional Test: Call Park Reversion
  - EP:18 – Functional Test: Direct Transfer
  - EP-19 – Functional Test: Automated CDR Creation
  - EP-20 – Functional Test: Meet-Me
  - EP-21 – Functional Test: Callback
  - EP-22 – Functional Test: Barge
  - EP-23 – Functional Test: cBarge
  - EP-24 – Functional Test: Shared Line – Hold/Resume
  - EP-27 – Functional Test: Video Endpoints
  - EP-33 – Functional Test: Join Across Line
  - EP-34 – Functional Test: Hotline
  - EP-35 – Functional Test: Group Pickup
  - EP-36 – Functional Test: Do Not Disturb (DND)
  - EP-37 – Functional Test: Do Not Disturb (DND)
  - EP-38 – Functional Test: iDivert
  - EP-39 – Functional Test: CFA & iDivert
  - EP-44 – Functional Test: Mobile Voice Access (MVA)

- EP-45 – Functional Test: Enterprise Feature Access (EFA)
- EP-64 – Basic call features using Expressway

- Test Cases that are Not Applicable:
    - EP-28 – Functional Test: Extension Mobility
    - EP-40 – Functional Test: Malicious Call
    - EP-41 – Functional Test: Mobile Connect
    - EP-42 – Functional Test: Mobile Connect
    - EP-43 – Functional Test: Mobile Connect
    - EP-52 – Miscellaneous Test: DUT display features
    - EP-56 – Functional Test: Multiple Lines
- Test Cases that were Not Executed:
    - None
- Observations:
    - DUT uses Android OS, ergo Speed Dial is done through configuring contacts as ‘Favorites’ instead of applying a DN to a line/number.
    - DUT is 3rd Party, so MOH doesn’t operate when putting Cisco phones on hold.
    - When DUT is registered with secondary proxy, DUT will occasionally un-authenticate and attempt to re-register to primary proxy. Should primary fail, DUT re-attempts registration with secondary proxy.

8 Testing Details

8.1 Items Tested

Features that are specific in this section are the high level categories the testing will focus on.

- 3rd Party installation, configuration and validation
- Security Requirements
- Functional testing of the various features interfacing through the 3rd party product to the Cisco product
- Negative tests in relation to service outages, restarts, bad files etc.

8.2 Items Not Tested

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

8.3 Assumptions

- Interoperability of 3rd party products – Testing will cover only features in 3rd party products that result in events to and/or from the Cisco product.

8.4 Administration, Testing and Debugging tools

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

Table 2. Administration, Testing and Debugging Tools

Product Name	Version	Type	Purpose	Units	Notes
Test Tools					
Remote Phone Control	4.2	Phone Tool	Controls Physical IP Phones remotely	1	
3rd Party Tools					
Wireshark	1.12.7	Packet Analyzer	Records data traversing the network.	1	
Debug Tools					
N/A	N/A	N/A	N/A	N/A	

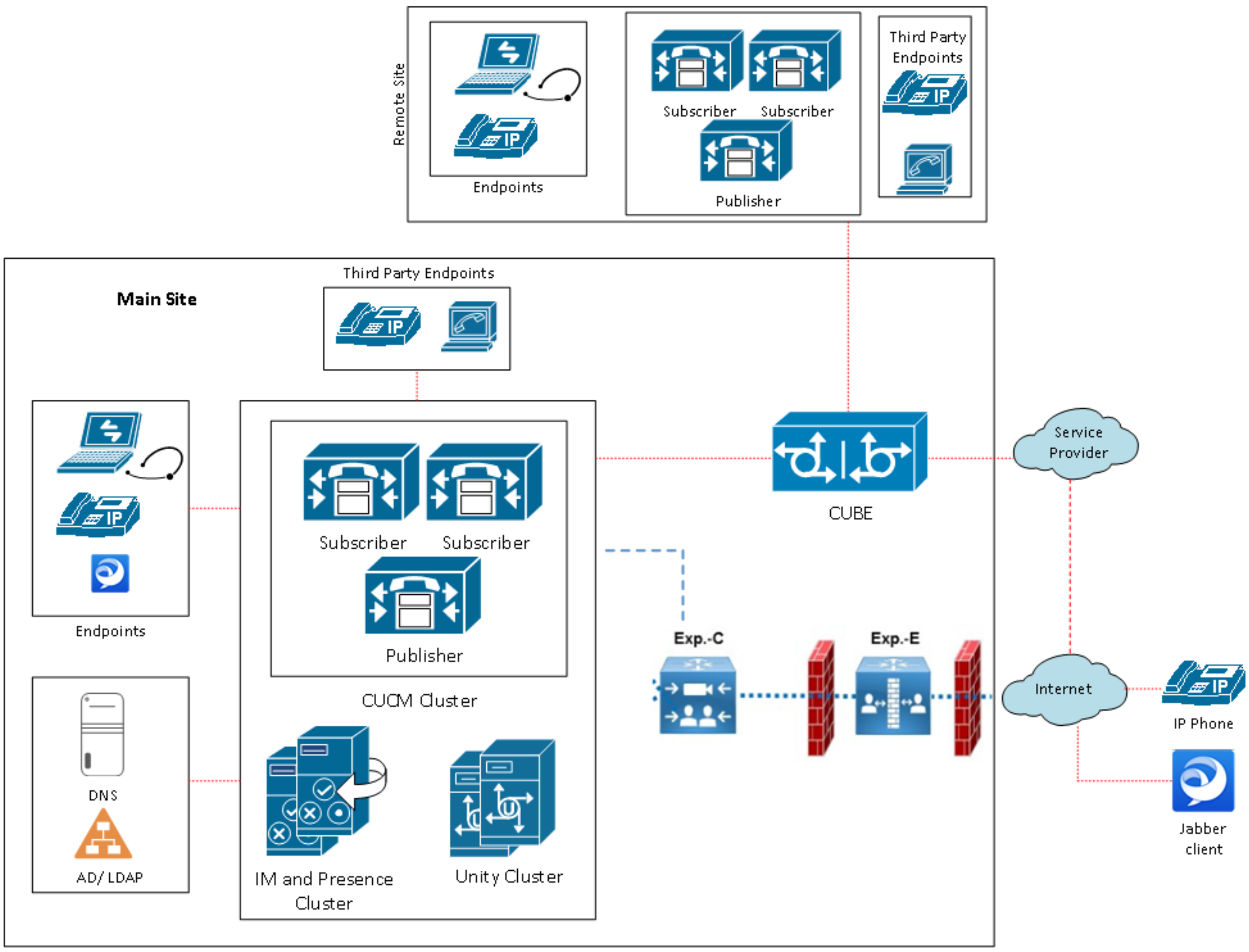
8.5 Equipment Requirements

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

Table 3. Sandbox Topology Components

Product	Version	Units	Description
CUCM	10.5	2 PUB & 2 SUB	HQ & Branch CUCM Clusters
CUPS	10.0	1	Cisco Presence Server
CUC	10.0	1	Cisco Unity Connection
Mediasense	10.0	1	Cisco Mediasense
Cisco 2811	12.4	2	PSTN Gateways
IP Phones	9.4.2	5	6941,79XX, 8851, 8861, 8945, 8961,9951,9971,DX650
Phoneview	N/A	1	Remote Phone Control Server (RPC)
DUT(s)	1.2.1	3 or more	VoIP Cellular Device

8.6 Lab Network Topology



F

8.7 Test Case Result Reporting

Table 4. Test Results Legend

Result	Description
Pass (P)	The test case passed with no exceptions
Fail (F)	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. Provide justification 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 (B)	Other test case failures prevented the execution of this test. Reference the failed test case in the“Comments” column.

9 Test Cases

This section details the tests that will be performed during the testing period. Partner is responsible for identifying any features or functions not supported covered in the test cases prior to start of testing

9.1 Endpoint IVT Workflow & Test Case Mapping

Test Work Flow Sections	Test Case #	Total Tests	A/M
Endpoint Registration & Validation (Step 1 & 2)	EP-1	1	M
Functional Tests (Step 3)	EP-2 → EP-45	44	M
Negative Tests (Step 4)	EP-46 → EP-50	5	M
Miscellaneous Tests (Step 5)	EP-51→ EP-56	6	M
Basic Call Features using Expressway (Step 6)	EP-57→ EP-65	9	M

9.2 Integration Test

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

<table>
<tr><td>Test Case #</td><td>EP-1</td><td>Category</td><td colspan="4">Connect➔Validate</td><td colspan="3">RFC_Standard</td><td>Y</td></tr>
<tr><td>Objective</td><td colspan="10">Verify 3rd party endpoints (DUT) are registered in Call Manager successfully</td></tr>
<tr><td colspan="11">Pre-Test Conditions</td></tr>
<tr><td colspan="11">• Enable auto-registration in Local CUCM with DN range 7100 – 7199
• Local CUCM Cluster➔NPA-NXX➔ 410-444-XXXX
• Remote CUCM Cluster➔NPA-NXX➔322-234-XXXX
• CUCM Administration GUI: https://X.X.X.X:8443 (X.X.X.X= CUCM-PUB IP)
• Hardware VPN Router setup to EP_IVT Lab</td></tr>
<tr><td colspan="4">Test Procedure</td><td colspan="7">Expected Results</td></tr>
<tr><td colspan="4">1. Connect two DUT(s) in local CUCM cluster
2. Connect one DUT in remote CUCM cluster
3. Run “Step1_Endpoint Registration” cmd to register DUT(s)
4. Run “Utility_Device_Status” cmd to check registration status
5. Go off-hook on DUT(s) to check for dial tone
6. Go to DUT(s) settings to verify network and load information
7. In local CUCM cluster, change DN of DUT(s) to 7100 & 7101 with device pool➔ep_pool
8. In remote CUCM cluster, change DN of DUT to 8000
9. Assign all DUT(s) a softkey template of “SIP_EP_User”
10. Associate end users to DUT(s) as follows:
Device➔Phone➔Line➔Associate End Users
➢ DUT:7100➔dutuser01
➢ DUT:7101➔dutuser02
➢ DUT:8000➔rdutuser01

Note:
• Endpoint Registration command is setup to register endpoints to local CUCM Cluster
• To register endpoints to remote CUCM cluster, manually enter the TFTP IP address of remote CUCM to DUT</td><td colspan="7">• DUT(s) goes through CUCM auto-registration process
• 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
• DUT(s) DN changed to proceed with test cases
• DUT(s) softkey template updated to “SIP_EP_User”
• Users associated to DUT(s) respectively</td></tr>
<tr><td colspan="5">Test Results: Comments</td><td>P</td><td>F</td><td>N/A</td><td>N/S</td><td>N/T</td><td>B</td></tr>
<tr><td colspan="5">DUT is 3rd party endpoint and ergo doesn’t support specific softkey template configurations.</td><td>X</td><td></td><td></td><td></td><td></td><td></td></tr>
</table>

9.3 Entrance Tests

Tests will be focused on features and the operational behavior of the 3rd party product (DUT) to ensure it corresponds to its design specifications.

Test Case #	EP-2	**Category**	Entrance Test: Intra-Cluster Calls	**RFC_Standard**	Y
Objective	Verify intra-cluster calls between DUT, SCCP and SIP endpoints				

Pre-Test Conditions

- Local CUCM ➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Configure Audio & Video Playback on an RDP Session for a PC accessing RPC Tool Server (Refer to Lab Guide for instructions)
  - Launch RDP➔Options➔Local Resources ➔Settings➔Play on this computer
  - Access RPC Server via RDP
- RPC is used to remotely control IP Phones :1000 & 2000;

Test Procedure	Expected Results
1. 7100 dials 7101➔7101 answers➔7100 on-hook after 30s 2. 7101 dials 1000➔Select 1000 and answer call using RPC 3. Select “Headphone” icon 4. Enter “Play:AreYouThere.raw” & hit “Send” on the Command Line 5. 7101 speaks “Testing1234”➔1000 on-hook after 60s 6 Repeat steps 2-5 with Calling DN:2000 & Called DN;7100 7. Repeat steps 2-5 with Calling DN:7101 & Called DN;2000 8. Calling & Called party release calls alternatively 9 Retrieve CDR from CUCM 10. Check Calling, Called, Duration, Origination & Termination Cause Codes	• 4 calls establish with 2-way audio path • Calling and Called Parties hear ring-back and ring tone • DUT receives Caller ID • Phone on RPC displays “monitoring active” message with symbol • 7100 & 7101 hear “Are you There” • 1000 & 2000 hear “Testing 1234” • Audio for RPC phones heard on pc running Phoneview • 4 calls terminate normally • 4 CDR(s) retrieved • Selected fields in CDR(s) match table

CDR field	Call 1	Call 2	Call 4	Call 5
callingPartyNumber	7100	7101	7101	2000
OriginalCalledPartyNumber	7101	1000	2000	7100
finalCalledPartyNumber	7101	1000	2000	7100
origCause_Value	16	0	16	0
destCause_Value	0	16	0	16
duration	30	60	60	60

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

Note:

- In RPC Tool, if any of the Phones used in test case is in un-registered state, use any available registered IP Phones. Phone displays without a DN assigned are un-registered.
- Refer to Lab Guide for instructions on:
  - CDR Retrieval from CUCM
  - Phoneview User Guide
  - 2-Way Audio Path Validation

9.4 Features and Services

<table>
<tr><td>Test Case #</td><td>EP-3</td><td>Category</td><td>Functional Test: Inter-Cluster Call</td><td>RFC_Standard</td><td>Y</td></tr>
<tr><td>Objective</td><td colspan="5">Verify inter-cluster calls between DUT(s), SCCP and SIP endpoints</td></tr>
<tr><td colspan="6">Pre-Test Conditions</td></tr>
<tr><td colspan="6">Local CUCM➔DUT(s):7100 & 7101;
Remote CUCM ➔DUT:8000; SCCP:5200; SIP:6200;
RPC is used to remotely control IP Phones: 5200, 6200;</td></tr>
<tr><td colspan="3">Test Procedure</td><td colspan="3">Expected Results</td></tr>
<tr><td colspan="3">1. 7100 dials 2348000➔8000 answers➔8000 on-hook after 30s
2. 7101 dials 2345200➔Select 5200 and answer call using RPC
3. Select “Headphone” icon
4. Enter “Play:AreYouThere.raw” and hit “Send” on the Command Line
5. 7101 speaks “Testing1234” ➔5200 on-hook after 60s
6. Repeat steps 2-5 with Calling DN:7101 & Called DN;6200
7. Calling & Called party release calls alternatively
8. Retrieve CDR from CUCM
9. Check Calling, Called, Duration, Origination & Termination Cause Codes</td><td colspan="3">3 calls establish with 2-way audio path
Calling and Called Parties hear ring-back and ring tone
DUT receives Caller ID
Phone on RPC displays “monitoring active” message with symbol
7100 & 8000 hear “Are you There”
5200 & 6200 hear “Testing 1234”
Audio for RPC phones heard on PC running Phoneview
3 calls terminate normally
3 CDR(s) retrieved
Selected fields in CDR(s) match calls</td></tr>
</table>

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

<table>
<tr><td>Test Case #</td><td>EP-4</td><td>Category</td><td>Functional Test: Off-Net Calls</td><td>RFC_Standard</td><td>Y</td></tr>
<tr><td>Objective</td><td colspan="5">Verify basic calls between DUT(s) and PSTN endpoints</td></tr>
<tr><td colspan="6">Pre-Test Conditions</td></tr>
<tr><td colspan="6">Local CUCM ➔ DUT(s):7100 & 7101;
Remote CUCM ➔DUT:8000;
PSTN: 210-222-5400 (SIP);
RPC is used to remotely control PSTN Phone:2102225401;</td></tr>
<tr><td colspan="3">Test Procedure</td><td colspan="3">Expected Results</td></tr>
<tr><td colspan="3">1. 7100 dials 92102225400➔Select 2102225400 & answer using RPC
2. Select “Headphone” icon
3. Enter “Play:AreYouThere.raw” and hit “Send” on the Command Line
4. 7100 speaks “Testing1234” ➔2102225400 on-hook after 60s
5. Repeat steps1-4 with Calling DN:2102225400 & Called DN:94104447101
6. Retrieve CDR from CUCM Server
7. Check Calling, Called, Duration, Origination & Termination Cause Codes

Note: Inbound & Outbound PSTN calls are dialed with a prefix of “9”</td><td colspan="3">2 Calls establish with 2-way audio path
Calling and Called Parties hear ring-back and ring tone
DUT receives Caller ID
Phone on RPC displays “monitoring active” message with symbol
7100 & 7101 hear “Are you There”
2102225400 hears “Testing 1234”
Audio for RPC phones heard on PC running Phoneview
2 Calls terminate normally
2 CDR(s) retrieved
Selected fields in CDR(s) match calls</td></tr>
</table>

Test Results: Comments	P	F	N/A	N/S	N/T	B

	X					

Test Case #	EP-5	Category	Functional Test: SIP URI	RFC_Standard	Y
Objective	Verify intra-cluster SIP URI calls between DUT and SIP endpoints				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 (URI:dutuser01@abc.inc); 7101 (URI:dutuser02@abc.inc); SIP:2000 (URI:cuser20@abc.inc);
- Configure Speed Dial on button 3 for 7100, 7101, 2000:
  - Device➔Phone➔7100➔Add new SD➔dutuser02@abc.inc on both fields
  - Device➔Phone➔7101➔Add new SD➔cuser20@abc.inc on both fields
  - Device➔Phone➔2000➔Add new SD➔dutuser01@abc.inc on both fields
- RPC is used to remotely control IP Phones: 2000;

Note: Provision URI on device page: Device➔Phone➔DN➔Line➔URI (Known bug if provisioned via End User Page)

Test Procedure	Expected Results
1. 7100 hits Speed Dial button 3➔7101 answers 2. 7101 goes on-hook after 30s 3. 2000 hits Speed Dial button 3 ➔7100 answers 4. 2000 goes on-hook after 30s 5. 7101 hits Speed Dial button 3➔2000 answers 6. 7101 goes on-hook after 30s 7. Retrieve CDR from CUCM Server 8. Check Calling, Called, Duration, Origination & Termination Cause Codes	• 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

Test Results: Comments	P	F	N/A	N/S	N/T	B
DUT doesn't support making calls with SIP URI, only supports digit dialing.				X		

Test Case #	EP-6	Category	Functional Test: SIP URI	RFC_Standard	Y
Objective	Verify inter-cluster SIP URI calls between DUT and SIP endpoints				

Pre-Test Conditions

- Local CUCM➔ DUT(s):7100 (URI: dutuser01@abc.inc) & 7101(URI: dutuser02@abc.inc); SIP:2000 (URI: cuser20@abc.inc);
- Remote CUCM➔ DUT :8000 (URI: rdutuser01@abc.inc); SIP:6200 (URI: rcuser20@abc.inc);
- Configure Speed Dial on button 3 for 7100, 7101 & 6200:
  - Device➔Phone➔7100➔Add new SD➔rdutuser01@abc.inc on both fields
  - Device➔Phone➔7101➔Add new SD➔rcuser20@abc.inc on both fields
  - Device➔Phone➔6200➔Add new SD➔dutuser01@abc.inc on both fields
- RPC is used to remotely control IP Phones: 2000 & 6200;

Note: Provision URI on device page: Device➔Phone➔DN➔Line➔URI (Known bug if provisioned via End User Page)

Test Procedure	Expected Results
1. 7100 hits Speed Dial button 3➔8000 answers 2. 8000 goes on-hook after 30s 3. 7101 hits Speed Dial button 3➔6200 answers 4. 6200 goes on-hook after 30s 5. 6200 hits Speed Dial button 3➔7100 answers 6. 7100 goes on-hook after 30s 7. Retrieve CDR from CUCM Server 8. Check Calling, Called, Duration, Origination & Termination Cause Codes	• 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

Test Results: Comments	P	F	N/A	N/S	N/T	B
DUT doesn't support making calls with SIP URI, only supports digit dialing.				X		

Test Case #	EP-7	Category	Functional Test: CFA	RFC_Standard	Y
Objective	Verify "CFA" calls between DUT(s), SCCP, SIP and PSTN endpoints				
Pre-Test Conditions					

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM➔DUT:8000; SCCP:5200; SIP:6200;
- PSTN: 210-222-5400
- Enable CFA for DN(s):
    - Device➔Phone➔7100➔CFA➔1000 (SCCP)
    - Device➔Phone➔7101➔CFA➔2348000 (DUT)
    - Device➔Phone➔8000➔CFA➔6200 (SIP)
    - Device➔Phone➔5200➔CFA➔4447101 (DUT)
    - Device➔Phone➔2000➔CFA➔2348000 (DUT)
    - Device➔Phone➔2102225400➔94104447100 (DUT)
- RPC is used to remotely control IP Phones: 1000, 2000, 5200, 6200, 2102225400;

Test Procedure	Expected Results
1. 7100 dials 7101➔2638 answers➔7100 on-hook after 30s 2. 2638 dials 4447100➔2635 answers➔2635 on-hook after 30s 3. 7101 dials 2638➔6200 answers➔7101 on-hook after 30s 4.7101 dials 2634➔2638 answers 5. 7100 dials 5200➔7100 on-hook - hears busy tone 6. 7101 goes on-hook 7. 7101 dials 5400➔7100 answers 8. PSTN goes on-hook after 30s 9. PSTN0 dials 7101➔2638 answers 10. 2638 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 highlighted in yellow before proceeding to next test case CUCM Administration GUI: Device➔Phone➔DN➔Line➔Call Forward All ➔Destination➔blank	• "CFA" phones displays the CFA # on scree • Call forward to 8000 and phone rings • Call establish between 7100 & 8000 with 2-way audio • Call terminate normally • Call forward to 1000 and phone rings • Call establish between 8000 & 1000 with 2-way audio • Call terminate normally • Call forward to 6200 and phone rings • Call establish between 7101 & 6200 with 2-way audio • Call terminate normally • Call forward to 8000 and phone rings • Call establish between 7101& 8000 with 2-way audio • Call forward to 7101 and phone rings • Call forward to 8000 and phone returns busy tone • 7100 hears busy tone and release call • Call on 7101 terminate normally • Call forward to 7100 and phone rings • Call establish between 7101& 7100 with 2-way audio • Call terminate normally • Call forward to 8000 and phone rings • Call between 2102225400 & 8000 with 2-way audio • Call terminate normally • 7 CDR(s) retrieved • Selected CDR(s) fields match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

Test Case #	EP-8	Category	Functional Test: CFNA	RFC_Standard	Y
Objective	Verify "CFNA" calls between DUT(s), SCCP and SIP endpoints				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM➔DUT:8000; SCCP:5200 SIP:6200;
- 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➔7100➔CFNA➔1000 (SCCP)
  - Device➔Phone➔7101➔CFNA➔2348000 (DUT)
  - Device➔Phone➔8000➔CFNA➔6200 (SIP)

➢ Device➔Phone➔5200➔CFNA➔4447101 (DUT) ➢ Device➔Phone➔2000➔CFNA➔2348000 (DUT) • RPC is used to remotely control IP Phones: 1000, 2000, 5200, & 6200,	
Test Procedure	**Expected Results**
1. 7100 dials 7101➔7101 does not answer➔8000 answers 2. 7100 goes on-hook after 30s 3. 8000 dials 4447100➔7100 does not answer➔1000 answers 4. 1000 goes on-hook after 30s 5. 7101 dials 2348000➔8000 does not answer➔6200 answers 6. 7101 goes on-hook after 30s 7.7101 dials 2000➔2000 does not answer➔8000 does not answer 8. 6200 answers call 9. 8000 dials 5200➔5200 does not answer➔7101 does not answer 10. 8000 goes on-hook - hears busy tone 11. 6200 goes on-hook 12. Retrieve CDR from CUCM 13. Check Calling, Called, Duration, Origination & Termination Cause Codes **Note:** Upon test completion, remove "CFNA" feature for devices highlighted in yellow before proceeding to next test case CUCM Administration GUI: Device➔Phone➔DN➔Line➔CFNA➔Destination➔blank	• Call forward to 8000 after ring timeout • Call establish between 7100 & 8000 with 2-way audio • Call terminate normally • Call forward to 1000 after ring timeout • Call establish between 8000 & 1000 with 2-way audio • Call terminate normally • Call forward to 6200 after ring timeout • Call establish between 7101 & 6200 with 2-way audio • Call terminate normally • Call forward to 6200 after ring timeout • Call establish between 7101& 6200 with 2-way audio • Call terminate normally • 8000 hears busy tone and terminate call • Call on 6200 terminate normally • Call forward to 7100 after ring timeout • Call establish between 7101& 7100 with 2-way audio • Call terminate normally • Call forward to 8000 after ring timeout • 5 CDR(s) retrieved • Selected fields in CDR(s) match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

Test Case #	EP-9	Category	Functional Test: CFB	RFC_Standard	Y
Objective	Verify "CFB" calls between DUT(s), SCCP and SIP endpoints				
Pre-Test Conditions					
• Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000; • Remote CUCM ➔DUT:8000; SCCP:5200; SIP:6200; • Voicemail and Call Waiting disabled for all DNs • Enable CFB for DN(s):					

- Device➔Phone➔7100➔CFB➔1000 (SCCP)
- Device➔Phone➔7101➔CFB➔2348000 (DUT)
- Device➔Phone➔8000➔CFB➔6200 (SIP)
- Device➔Phone➔6200➔CFB➔4447101(DUT)
- Device➔Phone➔1000➔CFB➔2348000 (DUT)

- RPC is used to remotely control IP Phones: 1000, 2000, 5200 & 6200;

Test Procedure	Expected Results
1. 1000 dials 7101➔7101 answers 2. 7100 dials 7101➔8000 answers➔7100 on-hook after 30s 3. 5200 dials 8000➔8000 answers 4. 7100 dials 7101➔7100 on-hook – hears busy tone 5. 5200 goes on-hook 6. 7100 dials 1000➔8000 answers➔7100 on-hook after 30s 7. 7101 goes on-hook 8. 2000 dials 1000➔1000 answers 9. 5200 dials 8000➔8000 answers 10. 7101 dials 1000➔7101 on-hook – hears busy tone 11. 1000 goes on-hook 12. 1000 dials 6200➔6200 answers 13. 7100 dials 2346200➔7101 answers➔7101 on-hook after 30s 14. 2000 dials 7101➔7101 answers 15. 7100 dials 2346200➔7100 goes on-hook –hears busy tone 16. 2000 & 6200 goes on-hook 17. Retrieve CDR from CUCM 18. Check Calling, Called, Duration, origination & termination cause codes matches the calls **Note:** Upon test completion, remove “CFB” feature for devices highlighted in yellow before proceeding to next test case CUCM Administration GUI: Device➔Phone➔DN➔Line➔CFB➔Destination➔blank	• Call establish between 1000 & 7101 with 2-way audio • Call forward to 8000 and phone rings • Call establish between 7100 & 8000 with 2-way audio • 7100 terminate call • Call establish between 5200 & 8000 with 2-way audio • 7100 hears busy tone and release call • 5200 terminate call • Call forward to 8000 and phone rings • Call establish between 7100 & 8000 with 2-way audio • 7101 terminate call • Call establish between 2000 & 1000 with 2-way audio • Call establish between 5200 & 8000 with 2-way audio • 7101 hears busy tone and release call • 1000 terminate call • Call establish between 1000 & 6200 with 2-way audio • Call forward to 7101 and phone rings • Call establish between 7100 & 7101 with 2-way audio • 7101 terminate call • Call establish between 2000 & 7101 with 2-way audio • 7100 hears busy tone and release call • 2000 & 6200 terminate call • Call establish between 2000 & 7100 with 2-way audio • 2000 & 7101 terminate call • 12 CDR(s) retrieved • Selected fields in CDR(s) match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

Test Case #	EP-10	Category	Functional Test: Hold & Resume	RFC_Standard	Y
Objective	Verify “Hold & Resume” calls between DUT(s), SIP, SCCP and PSTN endpoints				

Pre-Test Conditions						
• Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000; • Remote CUCM ➔DUT:8000; SCCP:5200; • PSTN:210-222-5400; • 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; • RPC is used to remotely control IP Phones: 1000, 2000, 2102225400;						
Test Procedure	**Expected Results**					
1. 7100 dials 7101➔7101 answers➔ 7100 hits "Hold" after 20s 2. 7100 hits "Resume" after 20s➔7100 on-hook after 30s 3. 7100 dials 2638➔2638 answers 4. 5200 dials 444-7100➔7100 answers incoming call➔2638 on-hold 5. 7100 hits "Resume" after 60s➔7100 on-hook after 30s 6. 7100 dials 7101➔7101 answers➔7100 hits "Hold" after 30s 7. 7100 dials 5200➔5200 answers➔5200 on-hook after 30s 8. 7101 goes on-hook after 30s 9. 7101 dials 7100➔7100 answers➔7101 hits "Hold" after 30s 10. 7100 goes on-hook 10s later while call is on-hold 11. 7101 goes-hook 12. Repeat steps 1-2 for SCCP. Replace 7101 with 1000 13. Repeat steps 1-2 for SIP. Replace 7100 with 2000 14. Repeat steps 1-2 for PSTN. Replace 7101 with 92102225400 15. Retrieve CDR from CUCM 16. Check Calling, Called, Duration, Origination & Termination Cause Codes	• Call establish between 7100 & 7101 with 2-way audio • 7101 is On-Hold (MOH) • Call resume between 7100 & 7101 • Call terminate normally • Call establish between 7100 & 8000 with 2-way audio • 8000 is On-Hold (MOH) • Call establish between 7100 & 5200 with 2-way audio • Call on 5200 terminated normally • Call resume between 7100 & 8000 with 2-way audio • Call terminate normally • Call establish between 7100 & 7101 with 2-way audio • 7101 is On-Hold (MOH) • Call establish between 7100 & 5200 with 2-way audio • 5200 terminate call normally • Call resume between 7100 & 7101 with 2-way audio • Call terminate normally • Call establish between 7101 & 7100 with 2-way audio • 7100 is On-Hold (MOH) • 7100 terminate call during active hold • 9 CDR(s) retrieved • Selected fields in CDR(s) match calls					
Test Results: Comments	**P**	**F**	**N/A**	**N/S**	**N/T**	**B**
	X					

Test Case #	EP-11	Category	Functional Test: Call Waiting	RFC_Standard	Y
Objective	Verify Call Waiting calls between DUT(s), SIP and SCCP endpoints				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000; SCCP:5200; SIP:6200;
- Call Waiting enabled for all DN(s): Device➔Phone➔DN➔Line➔Call Waiting➔Max. Calls➔4; Busy Trigger➔2;
- RPC is used to remotely control IP Phones: 1000, 2000, 5200, 6200;

Test Procedure	Expected Results
1. 7100 dials 7101➔7101 answers 2. 8000 dials 444-7100➔7100 answers incoming call 3. 8000 goes on-hook after 30s 4. 1000 dials 7101➔7101 answers incoming call 5. 1000 goes on-hook after 30s 6. 7100 goes on-hook after 60s 7. 7100 dials 1000➔7100 answers 8. 2000 dials 7100➔7100 answers incoming call 9. 2000 goes on-hook after 30s 10. 7100 goes on-hook after 60s 11. 6200 dials 444-7101➔7101 answers 12. 5200 dials 444-7101➔7101 answers incoming call 13. 5200 goes on-hook after 60s 14. 6200 goes on-hook after 30s 15. Retrieve CDR from CUCM 16. Check Calling, Called, Duration, Origination & Termination Cause Codes	• Call establish between 7100 & 7101 with 2-way audio • 7100 notified of incoming call (tone /display) • 7100 answers incoming call • 7101 is On-Hold (MOH) • Call establish between 7100 & 8000 with 2-way audio • 7100 & 8000 terminate normally • Call resume between 7100 & 7101 • 7101 notified of incoming call (tone /display) • 7101 answers incoming call • 7100 is On-Hold (MOH) • Call establish between 7101 & 1000 with 2-way audio • 7101 & 1000 terminate normally • Call resume between 7100 & 7101 • 7100 & 7101 terminate normally • Call establish between 7100 & 1000 with 2-way audio • 7100 notified of incoming call (tone /display) • 7100 answers incoming call • 1000 is On-Hold (MOH) • Call establish between 7100 & 2000 with 2-way audio • 7100 & 2000 terminate normally • Call resume between 7100 & 1000 • 7100 & 1000 terminate normally • Call establish between 6200 & 7101 with 2-way audio • 7101 notified of incoming call (tone /display) • 7101 answers incoming call • 6200 is On-Hold (MOH) • Call establish between 7101 & 5200 with 2-way audio • 7101 & 5200 terminate normally • Call resume between 6200 & 7101 • 6200 & 7101 terminate normally • 7 CDR(s) retrieved • Selected fields in CDR(s) match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

Test Case #	EP-12	**Category**	Functional Test: Blind Transfer	**RFC_Standard**	Y
Objective	Verify “Blind Transfer” calls between DUT(s), SIP and SCCP endpoints				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000;
- PSTN DN: 210-222-5400
- Invalid DN:7777;
- RPC is used to remotely control IP Phones : 1000, 2000, 2102225400;

Test Procedure	**Expected Results**
1. 7100 dials 7101➔7101 answers➔7101 hits “Transfer” after 30s 2. 7101 dials 234-8000➔ 7101 hits “Transfer”➔7101 is on-hook 3. 8000 goes on-hook after 60s 4. 7100 dials 7101➔7101 answers➔7100 hits “Transfer” after 30s 5. 7100 dials 234-7777➔7100 hits “Transfer”➔7100 is on-hook 6. 7101 goes on-hook – hears reorder tone 7. 7100 dials 1000➔1000 answers➔7100 hits “Transfer” after 30s 8. 7100 dials 7101➔7100 hits “Transfer”➔7100 is on-hook 9. 1000 goes on-hook after 60s 10. 7100 dials 2000➔2000 answers➔7100 hits “Transfer” after 30s 11. 7100 dials 1000➔7100 hits “Transfer”➔7100 is on-hook 12. 1000 goes on-hook after 20s 13. 7101 dials 234-8000➔8000 answers 14. 7100 dials 2000➔2000 answers➔2000 hits “Transfer’” after 30s 15. 2000 dials 234-8000➔2000 hits “Transfer”➔2000 is on-hook 16. 7100 goes on-hook - hears busy tone 17 .Retrieve CDR from CUCM 18. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• Call establish between 7100 & 7101 with 2-way audio • 7100 is On-Hold (MOH) • 7100 blind transfer to 8000 with 2-way audio path • All calls terminate normally • Call establish between 7100 & 7101 with 2-way audio • 7101 is On-Hold (MOH) • 7101 blind transfer to Invalid DN:7777 • 7101 hears reorder tone • All calls terminate normally • Call establish between 7100 & 1000 with 2-way audio • 1000 is On-Hold (MOH) • 1000 blind transfer to 7101 with 2-way audio • All calls terminate normally • Call establish between 7100 & 2000 with 2-way audio • 2000 is On-Hold (MOH) • 2000 blind transfer to 1000 with 2-way audio path • All calls terminate normally • Call establish between 7101 & 8000 with 2-way audio • Call establish between 7100 & 2000 with 2-way audio • 7100 is On-Hold (MOH) • 7100 blind transfer to 8000 • 7100 hears a busy tone • All calls terminate normally • 15 CDR(s) retrieved • Selected fields in CDR(s) match calls

Test Results: Comments	**P**	**F**	**N/A**	**N/S**	**N/T**	**B**
When dialing invalid DN, call wasn’t transferred to DUT. Transfer pilot remained on call with called party.	**X**					

<table>
<tr><td>Test Case #</td><td>EP-13</td><td>Category</td><td>Functional Test: Consult Transfer</td><td>RFC_Standard</td><td>Y</td></tr>
<tr><td>Objective</td><td colspan="5">Verify “Consult Transfer” calls between DUT(s), SIP, SCCP and PSTN endpoints</td></tr>
<tr><td colspan="6">Pre-Test Conditions</td></tr>
<tr><td colspan="6">• Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
• Remote CUCM ➔DUT:8000;
• PSTN DN: 210-222-5400
• RPC is used to remotely control IP Phones: 1000, 2000, 2102225400;</td></tr>
<tr><td colspan="3">Test Procedure</td><td colspan="3">Expected Results</td></tr>
<tr><td colspan="3">1.7100 dials 7101➔7101 answers➔7101 hits “Transfer’” after 30s
2.7101 dials 2348000➔8000 answers
3.7101 hits “Transfer” after 30s➔7101 is on-hook
4.8000 goes on-hook after 60s
5.7100 dials 1000➔1000 answers➔7100 hits “Transfer” after 30s
6.7100 dials 7101➔7101 answers➔7100 hits “Transfer” after 30s
7.7100 goes on-hook
8.1000 goes on-hook after 60s
9. 7100 dials 2000➔2000 answers➔7100 hits “Transfer’” after 30s
10. 7100 dials 1000➔1000 answers➔7100 hits “Transfer” after 30s
11. 7100 goes on-hook
12. 1000 goes on-hook after 60s
13. 7101 dials 92102225400➔2103335400 answers
14. 7101 hits “Transfer” after 30s➔7101 dials 7100➔7100 answers
15. 7101 hits “Transfer” after 30s➔7101 is on-hook
16. 2102225400 goes on-hook after 60s
17. Retrieve CDR from CUCM
18. Check the Calling, Called, Duration, Origination & Termination Cause Codes</td><td colspan="3">• Call establish between 7100 & 7101 with 2-way audio
• 7100 is On-Hold (MOH)
• 7100 consult transfer to 8000 with 2-way audio path
• All calls terminate normally
• Call establish between 7100 & 1000 with 2-way audio
• 1000 is On-Hold (MOH)
• 1000 consult transfer to 7101 with 2-way audio path
• All calls terminate normally
• Call establish between 7100 & 2000 with 2-way audio
• 2000 is On-Hold (MOH)
• 2000 consult transfer to 1000 with 2-way audio path
• All calls terminate normally
• Call establish between 7101 & 2102225400 with 2-way audio
• 2102225400 is On-Hold (MOH)
• 2102225400 consult transfer to 7100 with 2-way audio path
• All calls terminate normally
• 12 CDR(s) retrieved
• Selected fields in CDR(s) match calls</td></tr>
</table>

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

<table>
<tr><td>Test Case #</td><td>EP-14</td><td>Category</td><td colspan="3">Functional Test: Conference Call</td><td>RFC_Standard</td><td>Y</td></tr>
<tr><td>Objective</td><td colspan="7">Verify Conference call between DUT(s), SIP, SCCP and PSTN endpoints</td></tr>
<tr><td colspan="8">Pre-Test Conditions</td></tr>
<tr><td colspan="8">• Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
• Remote CUCM ➔DUT:8000;
• PSTN DN: 210-222-5400;
• 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
• RPC remotely control IP Phones: 1000, 2000, 2102225400;</td></tr>
<tr><td colspan="4">Test Procedure</td><td colspan="4">Expected Results</td></tr>
<tr><td colspan="4">1.7100 dials 7101➔7101 answers➔7101 hits “Conference” after 30s
2.7101 dials 2348000➔8000 answers
3. 7101 hits “Conference” after 30s
4. 7100 goes on-hook after 60s
5. 8000 goes on-hook after 30s
6. 7100 dials 7101➔7101 answers➔7100 hits “Conference” after 30s
7. 7100 dials 1000➔1000 answers➔7100 hits “Conference” after 30s
8.1000 goes on-hook after 60s
9.7100 goes on-hook after 30s
10. 7100 dials 2000➔2000 answers➔7100 hits “Conference” after 30s
11. 7100 dials 92102225400➔2102225400 answers
12. 7100 hits “Conference” after 30s
13. 2000 hits “Conference” after 30s➔2000 dials 2348000
14. 8000 answers➔2000 hits “Conference” after 30s
15. 2102225400 hits “Conference” after 30s➔2102225400 dials 7101
16. 7101 answers➔2102225400 hits “Conference” after 30s
17. 2102225400 goes on-hook after 60s
18. 7100 goes on-hook after 30s
19. 2000 goes on-hook after 30s
20. 7100 dials 2000➔2000 answers➔7100 hits “Conference” after 30s
21. 7100 dials 1000➔7100 resumes call before 1000 answers
22. 2000 goes on-hook after 30s
23. Retrieve CDR from CUCM
24. Check the Calling, Called, Duration, Origination & Termination Cause Codes</td><td colspan="4">• Call establish between 7100 & 7101 with 2-way audio
• 7100 is On-hold (MOH)
• 8000 is conference-in
• 3 parties in conference call with 3-way audio
• 7100 left conference.7101 & 8000 connect directly
• All calls terminate normally
• Call establish between 7100 & 2000 with 2-way audio
• 2000 is On-Hold (MOH)
• 2102225400, 8000 & 7101 is conference-in
• All 5 parties in conference call with 5-way audio
• 2102225400, 7100 & 2000 leave conference. 8000 & 7101 connect directly
• All calls terminate normally
• Call establish between 7100 & 2000 with 2-way audio
• 2000 is placed on-hold (MOH)
• Conference setup was cancelled
• Call between 7100 and 2000 resumed
• All calls terminate normally
• CDR(s) retrieved
• Selected fields in CDR(s) match calls</td></tr>
<tr><td colspan="2">Test Results: Comments</td><td>P</td><td>F</td><td>N/A</td><td>N/S</td><td>N/T</td><td>B</td></tr>
<tr><td colspan="2">Conference not implemented on DUT</td><td></td><td></td><td></td><td>X</td><td></td><td></td></tr>
</table>

Test Case #	EP-15	Category	Functional Test: Call Park	RFC_Standard	N
Objective	Verify "Call Park" call for a DUT(s), SIP, SCCP and PSTN endpoints				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000;
- PSTN DN: 210-222-5400;
- Call Park Code: Routing➔Call Park➔3001
- RPC is used to remotely control IP Phones: 1000, 2000, 2102225400;

Test Procedure	Expected Results
1. 7100 dials 2348000➔8000 answers 2. 7100 hits "Park" softkey after 10s 3. 7101 dials park code:3001 after 20s 4. 8000 goes on-hook after 30s 5. 7100 dials 1000➔1000 answers 6. 1000 hits "Park" softkey after 10s 7. 7101 dials park code:3001 after 20s 8. 7100 goes on-hook after 30s 9. 2000 dials 7101➔7101 answers 10. 7101 hits "Park" softkey after 10s 11. 7100 dials park code:3001 after 20s 12. 2000 goes on-hook after 30s 13. 2102225400 dials 7100➔7100 answers 14. 7100 hits the "Park" softkey after 10s 15. 7100 dials park code:3001 after 20s 16. 2102225400 goes on-hook 17. Retrieve CDR from CUCM 18. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• Call establish between 7100 & 8000 with 2-way audio • 8000 is parked • 7101 picks up parked call • Call establish between 7101 & 8000 with 2-way audio • Call terminate normally • Call establish between 7100 & 1000 with 2-way audio • 7100 is parked • 7101 picks up parked call • Call establish between 7101 & 7100 with 2-way audio • Call terminate normally • Call establish between 2000 & 7101 with 2-way audio • 2000 is parked • 7100 picks up parked call • Call establish between 7100 & 2000 with audio path • Call terminate normally • Call establish between 2102225400 & 7100 with 2-way audio • 2102225400 is parked • 7100 picks up parked call • Call establish between 7100 & 2102225400 with 2-way audio • Call terminate normally • 8 CDR(s) retrieved • Selected fields in the CDR matched the calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
Call Park is not implemented on DUT				X		

Test Case #	EP-16	Category	Functional Test: Call Park Reversion	RFC_Standard	N
Objective	Verify “Call Park Reversion” call for DUT(s), SIP and SCCP endpoints				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000;
- Call Park Code: Routing➔Call Park➔3001
- Service Parameter: Call Park Reversion Timer ➔60s
- RPC is used to remotely control IP Phones with DN: 1000, 2000;

Test Procedure	Expected Results
1. 7100 dials 234-8000➔8000 answers 2. 7100 hits “Park” softkey after 10s 3. Do not pickup parked call for 60s 4. 7100 is ringing➔7100 answers 5. 8000 goes on-hook after 30s 6. 7100 dials 1000➔1000 answers 7. 1000 hits “Park” softkey after 10s 8. Do not pickup parked call for 60s 9. 1000 is ringing➔1000 answers 10. 7100 goes on-hook after 30s 11. 2000 dials 7101➔7101 answers 12. 7101 hits “Park” softkey after 10s 13. Do not pickup parked call for 60s 14. 7101 is ringing➔7101 answers 15. 2000 goes on-hook after 30s 16. Retrieve CDR from CUCM 17. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• Call establish between 7100 & 8000 with 2-way audio • 8000 is parked • 7100 picks up parked call • Call establish between 7100 & 8000 with 2-way audio • Call terminate normally • Call establish between 7100 & 1000 with 2-way audio • 7100 is parked • 1000 picks up parked call • Call establish between 1000 & 7100 with 2-way audio • Call terminate normally • Call establish between 2000 & 7101 with 2-way audio • 2000 is parked • 7101 picks up parked call • Call establish between 7101 & 2000 with 2-way audio • Call terminate normally • 6 CDR(s) retrieved • Selected fields in CDR match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
Call Park is not implemented on DUT				X		

Test Case #	EP-17	Category	Functional Test: Directed Call Park	RFC_Standard	N
Objective	Verify "Assisted Directed Call Park" call between DUT(s) and SIP endpoints				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SIP:2000-2003; SCCP:1000;
- Remote CUCM ➔DUT:8000;
- 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
- RPC is used to remotely control IP Phone: 2000;

Test Procedure	Expected Results
1. 8000 dials 4442000➔2000 answers 2. 2000 hits "BLF" button for Assisted Directed Call Park after 20s 3. 2000 goes on-hook 4. 7101 dials *3011 to retrieve call when the BLF is flashing 5. 8000 goes on-hook after 30s 6. 2003 dials 234-8000➔8000 answers 7. 2003 hits "BLF" button for Assisted Directed Call Park after 20s 8. 2003 goes on-hook 9. 1000 hits dials *3011 to retrieve call when BLF is flashing 10. 1000 goes on-hook after 30s 11. Retrieve CDR from CUCM 12. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• Call establish between 8000 & 2000 with 2-way audio • 8000 is parked • 7101 retrieves directed parked call • Call establish between 8000 & 7101 with 2-way audio • Calls terminate normally • Call establish between 2003 & 8000 with 2-way audio • 8000 is parked • 1000 retrieves directed park call • Call establish between 1000 & 8000 with 2-way audio • Calls terminated normally • 4 CDR(s) retrieved • Selected fields in CDR match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
DUT has no BLF field/light.	X					

Test Case #	EP-18	Category	Functional Test: Direct Transfer	RFC_Standard	N
Objective	Verify "Direct Transfer" call from a shared line between DUT(s), SCCP, SIP and PSTN endpoints				
Pre-Test Conditions					
- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000; - Remote CUCM ➔DUT:8000; - PSTN: 210-222-5400; - DN:1901 (shared line) assigned to Line 2 on DUT:7100 - RPC is used to remotely control IP Phones: 1000, 2000, 2102225400;					

Test Procedure	Expected Results
1. 7101 dials 7100➔7100 answers 2. 7100 selects shared line:DN:1901 after 30s 3. 1901 dials 234-8000➔8000 answers 4. Scroll to 1st call and hit select 5. Scroll to 2nd call and hit select and hit "DirTfr" 6. 1901 goes on-hook 7. 7101 goes on-hook after 30s 8. 7100 dials 7101➔7101 answers 9. 7100 selects shared line:DN:1901 after 30s 10. 1901 dials 1000➔1000 answers 11. Scroll to 1st call and hit select 12. Scroll to 2nd call and hit select and hit "DirTfr" 13. 1901 goes on-hook 14. 1000 goes on-hook after 30s 15. 8000 dials 444-7100➔7100 answers 16. 7100 selects shared line:DN:1901 after 30s 17. 1901 dials 2000➔2000 answers 18. Scroll to 1st call and hit select 19. Scroll to 2nd call and hit select and hit "DirTfr" 20.. 1901 goes on-hook 21. 2000 goes on-hook 22. 7100 dials 234-8000 23. 7100 selects shared line:DN:1901 after 30s 24. 1901 dials 92102225400➔2102225400 answers 25. Scroll to 1st call and hit select 26. Scroll to 2nd call and hit select and hit "DirTfr" 27. 1901 goes on-hook 28. 8000 goes on-hook 29. Retrieve CDR from CUCM 30. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• Call establish between 7101 & 7100 with 2-way audio • 7101 is On-Hold (MOH) • Call establish between 1901 & 8000 with 2 way audio • 7101 direct transfer to 8000 with 2 way audio • 1901 dropped off from call • Call terminate normally • Call establish between 7100 & 7101 with 2-way audio • 7101 On-Hold (MOH) • Call establish between 1901 & 1000 with 2 way audio • 7101 direct transfer to 1000 with 2 way audio • 1901 dropped off from call • Call terminate normally • Call establish between 8000 & 7100 with 2-way audio • 8000 is On-Hold (Tone/Silence) • Call establish between 1901 & 2000 with 2 way audio • 7101 direct transfer to 2000 with 2 way audio • 1901 dropped off from call • Call terminate normally • Call establish between 7100 & 8000 with 2-way audio • 8000 is On-Hold (Tone/Silence) • Call between 1901 & 2102225400 with 2 way audio • 7101 direct transfer to 103335500 with 2 way audio • 1901 dropped off from call • Call terminate normally • 8 CDR(s) retrieved • Selected fields in CDR(s) match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
DUT is 3rd party device and doesn't support shared line.				X		

Test Case #	EP-19	Category	Functional Test: Automated CDR Creation	RFC_Standard	N
Objective	Verify joining two "Ad-Hoc Conference" using DUT(s), SIP, SCCP and PSTN endpoints				
Pre-Test Conditions					
• Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000; • Remote CUCM ➔DUT:8000; SCCP:5200; SIP:6200; • PSTN: 210-222-5400;					

• Assign Media Resource: System➔Device Pool➔ep_pool➔Media Resource Group List➔MRG_L • RPC is used to remotely control IP Phones with DN: 1000, 2000, 5200, 6200, 2102225400;

Test Procedure	**Expected Results**
1. 7100 dials 92102225400➔2102225400 answers 2. 7100 hits "Conference" after 30s 3. 7100 dials 7101➔7101 answers➔7100 hits "Conference" after 30s 4. 2000 dials 7101➔7101 answers 2nd incoming call 5. 2000 hits "Conference" after 20s 6. 2000 dials 234-8000➔8000 answers➔2000 hits "Conference" after 20s 7. 2000 dials 1000➔1000 answers➔2000 hits "Conference" after 20s 8. 7101 selects conference 1 and hits the "Join" softkey 9. 7101 goes on-hook after 60s 10. 8000 goes on-hook after 70s 11. All other participants ended call after 120s 12. Retrieve CDR from CUCM 13. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• Call between 7101 & 2102225400 with 2-way audio • 2102225400 is On-Hold (Tone/Silence) • Call establish between 7100 & 7101 with 2-way audio • All 3 participants join in conference-1 • Call establish between 2000 & 7101 with 2-way audio • 7101 is On-Hold (Tone/Silence) • Call establish between 2000 & 8000 with 2-way audio • All 3 participants join in conference-2 • All participants in conference 1 & 2 are joined • 7101 left conference • 8000 left conference • All participants terminate normally • 12 Records retrieved • Selected fields in CDR(s) match calls

Test Results: Comments	**P**	**F**	**N/A**	**N/S**	**N/T**	**B**
DUT doesn't support conference and join softkey.				X		

Test Case #	EP-20	**Category**	Functional Test: Meet-Me	**RFC_Standard**	N
Objective	Verify "Meet-Me" Conference call using DUT(s), SCCP and SIP endpoints				
Pre-Test Conditions					
• Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000; • Remote CUCM ➔DUT:8000; • Meet-Me #: Call Routing➔Meet-Me➔Add New➔5555 [meet-me #] & create CTI_RP with DN:5555) • Meet-Me Conference initiator (7101): Device➔Phone➔DN:7101➔Calling Search Space➔css-meetme • RPC is used to remotely control IP Phones with DN: 1000, 2000;					

Test Procedure	**Expected Results**
1. 7101 goes off hook & hits "Meet-Me" softkey and dials 5555 2. 1000 dials 5555 3. 2000 dials 5555 4. 8000 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	• 7101, 1000, 2000, & 8000 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 **Note:** Change the Calling Search Space for 7101➔ None after test is complete

Test Results: Comments	**P**	**F**	**N/A**	**N/S**	**N/T**	**B**
DUT doesn't support Meet-Me softkey.				X		

Test Case #	EP-21	**Category**	Functional Test: Callback	**RFC_Standard**	N
Objective	Verify "Callback" calls between DUT(s), SCCP, SIP and PSTN endpoints				

Pre-Test Conditions	
• Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000; • Remote CUCM ➔DUT:8000; • PSTN DN: 210-222-5400 • VM and CW disabled for all phones ; • RPC is used to remotely control IP Phones:1000, 2000, 210-222-5400;	
Test Procedure	**Expected Results**
1. 7100 dials 7101➔7101 answers 2. 8000 dials 444-7101➔8000 hits "Callback" softkey and exits 3. 7101 goes on-hook after 60s 4. 8000 redials 444-7101 after callback alert 5. 7101 answers➔7101 goes on-hook after 60s 6. 7100 dials 1000➔1000 answers 7. 7101 dials 1000➔7101 hits "Callback" softkey and exits 8. 7100 goes on-hook after 60s 9. 7101 redials 1000 after callback alert 10. 1000 answers➔1000 goes on-hook after 60s 11. 2000 dials 7100➔7100 answers 12. 7101 dials 2000➔7101 hits "Callback" softkey and exits 13. 7100 goes on-hook after 60s 14. 7101 redials 2000 after callback alert 15. 2000 answers➔7101 goes on-hook after 60s 16. 7101 dials 92102225400➔2102225400 answers 17. 7100 dials 92102225400➔7100 hits "Callback" softkey & exits 18. 2102225400 goes on-hook after 60s 19. 7100 redials 92102225400 after callback alert 20. Retrieve CDR from CUCM 21. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• Call establish between 7100 & 7101 with 2-way audio • 8000 hears a busy tone & displays "Callback Active" • 7100 & 7101 terminate normally • 8000 receives callback alert with single button redial • Call establish between 8000 & 7101 with 2-way audio • Call terminate normally • Call establish between 7100 & 1000 with 2-way audio • 7101 hears a busy tone & displays "Callback Active" • 7100 & 1000 terminate normally • 7101 receives callback alert with single button redial • Call establish between 1000 & 7101 with 2-way audio • Call terminated normally • Call establish between 2000 & 7100 with 2-way audio • 7101 hears a busy tone & displays "Callback Active" • 2000 & 7100 terminate normally • 7101 receives callback alert with single button redial • Call establish between 2000 & 7101 with 2-way audio • Call terminate normally • Call establish between 7101 & 2102225400 with 2-way audio • 7100 hears a busy tone & displays "Callback Active" • 7100 & 2102225400 terminate normally • 7100 receives callback alert with single button redial • Call establish between 7101 & 2102225400 with 2-way audio • Call terminate normally • 14 CDR(s)s retrieved • Selected fields in CDR(s) match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
CallBack is not implemented on DUT				X		

Test Case #	EP-22	Category	Functional Test: Barge	RFC_Standard	N
Objective	Verify "Barge" call using DUT(s), SCCP, SIP and PSTN endpoints				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000;
- PSTN:210-222-5400
- 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:7100, 7101,1000, & 2000;
  - Privacy on Phones with shared lines➔Off
  - Single Button Barge➔Barge
- RPC is used to remotely control IP Phones with DN: 1000, 2000, 2102225400;

Test Procedure	Expected Results
1. 8000 dials 444-1901 2. 1901 answers (Shared line on 7100) 3. 7101 hits line 1901 and selects barge after 20s 4. 8000 goes on-hook after 60s 5. 7100 dials 1901 6. 1901 answers (Shared line on 1000) 7. 7101 hits line 1901 and selects barge after 20s 8. 1000 goes on-hook after 60s 9. 8000 dials 4441901 10. 1901 answers (Shared line on 2000) 11. 7100 hits line 1901 and selects barge after 20s 12. 7100 goes on-hook after 60s 13. 8000 goes on-hook after 70s 14. 2102225400 dials 94104441901 15. 1901 answers (Shared line on 7100) 16. 7101 hits line 1901 and selects barge after 20s 17. 2102225400 goes on-hook after 60s 18. Retrieve CDR from CUCM 19. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• Call establish between 8000 & 1901 with 2-way audio • All 3 parties conference-in with 3-way audio • Barged conference terminate normally • Call establish between 7100 & 1901 with 2-way audio • All 3 parties conference-in with 3-way audio • Barged conference terminate normally • Call establish between 8000 & 1901 with 2-way audio • All 3 parties conference with 3-way audio • Barged conference terminate normally • Final call terminate normally • Call establish between 2102225400 & 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

Test Results: Comments	P	F	N/A	N/S	N/T	B
DUT is 3rd party device and doesn't support shared line.				X		

Test Case #	EP-23	Category	Functional Test: cBarge	RFC_Standard	N
Objective	Verify "cBarge" call using DUT(s), SCCP, SIP and PSTN endpoints				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000;
- PSTN DN: 210-222-5400;
- 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:7100, 7101,1000, & 2000;
  - Privacy on Phones with shared lines➔Off
  - Single Button Barge➔cBarge
- RPC is used to remotely control IP Phones with DN: 1000, 2000, 2102225400;

Test Procedure	Expected Results
1. 8000 dials 444-1901➔1901 answers (Shared line on 7100) 2. 7101 selects line 1901 after 20s 3.1901 goes on-hook after 60s (Shared line on 7100) 4. 8000 goes on-hook after 80s 5. 2102225400 dials 94104441901➔1901 answers (Shared line on 1000) 6. 2000 selects line 1901 after 20s 7. 7100 selects line 1901 Select 1901 after 30s 8. 1901 goes on-hook after 60s (Shared line on 7100) 9. Remaining 3 parties go-hook after 120s 10. Retrieve CDR from CUCM 11. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• Call establish between 8000 & 1901 with 2-way audio • All 3 parties conference-in with 3-way audio • cBarge conference terminate normally • Call between 2102225400 & 1901 with 2-way audio • All 4 parties conference-in with 4-way audio • 7100 terminated conference 1st • 3 parties terminate cBarge conference after 120s • 11 CDR(s) retrieved • Selected fields in CDR match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
DUT is 3rd party device and doesn't support shared line.				X		

Test Case #	EP-24	Category	Functional Test: Shared Line – Hold/Resume	RFC_Standard	N
Objective	Verify " Hold/Resume" call on a shared line using DUT(s), SCCP and SIP endpoints				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000;
- Shared line DN:1901 added to devices with DN:7101,1000, & 2000;
- Privacy on Phones with shared lines➔Off
- RPC is used to remotely control IP Phones with DN: 1000, 2000;

Test Procedure	Expected Results
1. 7100 dials 1901➔1901 answers (Shared Line on 7101)	• Call establish between 7100 & 1901 with 2-way audio
2. 1901 hits "Hold" softkey after 30s	• 7100 is On-Hold (tone or silence)
3. 1901 hits "Resume" softkey after 30s	• 7100 & 1901 resume call
4. 1901 goes on-hook after 80s	• Call terminate normally
5. 7100 dials 1901➔1901 answers (Shared Line on 1000)	• Call establish between 7100 & 1901 with 2-way audio
6. 1901 hits "Hold" softkey after 30s	• 7100 is On-Hold (tone or silence)
7. 1901 hits "Resume" softkey after 30s	• 7100 & 1901 resume call
8. 7100 goes on-hook after 80s	• Call terminate normally
9. 7100 dials 1901➔1901 answers (Shared Line on 2000)	• Call establish between 7100 & 1901 with 2-way audio
10. 1901 hits "Hold" softkey after 30s	• 7100 is On-Hold (tone or silence)
11. 1901 hits "Resume" softkey after 30s	• 7100 & 1901 resume call
12. 1901 goes on-hook after 80s	• Call terminate normally
13. 2001 dials 1901➔1901 answers (Shared Line on 7101)	• Call establish between 2001 & 1901 with 2-way audio
14. 1901 hits "Hold" softkey after 30s	• 2001 is On-Hold (tone or silence)
15. 1901 hits "Resume" softkey after 30s	• 2001 & 1901 resume call
16. 1901 goes on-hook after 80s	• Call terminate normally
17. Retrieve CDR from CUCM	• 4 CDR(s) retrieved
18. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• Selected fields in CDR(s) match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
DUT is 3rd party device and doesn't support shared line.				X		

Test Case #	EP-25	Category	Functional Test: Jabber for Windows	RFC_Standard	Y
Objective	Verify Jabber calls originating & terminating to DUT(s) endpoints (Jabber for Windows)				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000;
- Jabber for Windows (Device➔Phone➔Add New➔CSFUSER1:DN:1922; End User:juser01/123456)
- Windows PC with Jabber clients installed

Test Procedure	Expected Results
1. 7100 dials 1922 (Duration=30s) 2. 1922 dials 7101 (Duration=30s) 3. 8000 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	• 3 calls establish with 2-way audio • 3 calls terminate normally • 3 CDR(s) retrieved • Selected fields in CDR(s) match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

Test Case #	EP-26	Category	Functional Test: IP Communicator	RFC_Standard	Y
Objective	Verify IP Communicator calls originating & terminating to DUT(s) endpoints				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000; CIPC:1940;
- Remote CUCM ➔DUT:8000;
- Launch IP Communicator on a PC: Phone Preferences➔Network:
  - Use this Device Name: CIPC00001940
  - Use this TFTP Servers:10.10.20.211
- CIPC Credentials: ipcuser01/123456
- Windows PC with IP Communicator client installed

Test Procedure	Expected Results
1. 7100 dials 1940 (Duration=30s) 2. 1940 dials 7101 (Duration=30s) 3. 8000 dials 444-1940 (Duration=30s) 4. 1940 dials 234-8000 (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	• 4 calls establish with 2-way audio • 4 calls terminate normally • 4 CDR(s) retrieved • Selected fields in CDR(s) match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

Test Case #	EP-27	Category	Functional Test: Video Endpoints	RFC_Standard	Y
Objective	Verify video calls originating & terminating to DUT(s) endpoints				

Pre-Test Conditions

- Local CUCM➔DUT(s): 7100 & 7101;
- Remote CUCM➔DUT DN:8000;
- Video capable phone DN: 2003

Test Procedure	Expected Results
1. 7100 dials 2003 (Duration=30s) 2. 2003 dials 7101 (Duration=30s) 3. 8000 dials 444-2005 (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	• 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

Test Results: Comments	P	F	N/A	N/S	N/T	B
3rd Party Device configurations don't allow video calls.				X		

Test Case #	EP-28	Category	Functional Test: Extension Mobility	RFC_Standard	N
Objective	Verify DUT(s) supports "Extension Mobility" call				

Pre-Test Conditions

- Local CUCM→DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM →DUT:8000; SCCP DN:5200; SIP DN:6200;
- PSTN: 210-222-5400;
- 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_7100 with DN:1934
- Extension Mobility enabled on 7100: Device→Phone→7100→Enable Extension Mobility checked
- Extension Mobility Service subscribed on 7100:Device→Phone→Select "Subscribe/Unsubscribe Services" →EM
- Create User/PIN: emuser01/123456; Associate device profile EM_7100 to user under Extension Mobility; EMCC checked;
- RPC is used to remotely control IP Phones with DN: 1000, 2000, 5200, 6200, 210222-5400 ;

Test Procedure	Expected Results
1. 7100 hits "Services" button and selects EM service 2. 7100 logs in with "emuser01/123456" 3. 1934 dials 1000→1000 answers →1934 on-hook after 30s 4. 2000 dials 1934→1934 answers→2000 on-hook after 30s 5. 7101 dials 1934→1934 answer→7101 on-hook after 30s 6. 1934 dials 234-5200→5200 answers→1934 on-hook after 30s 7. 6200 dials 444-1934→1934 answers→6200 on-hook after 30s 8. 1934 dials 92102225400→2102225400 answers 9. 2102225400 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 Cause Codes	• 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 7100 device profile • 6 CDR(s) retrieved • Selected fields in CDR(s) match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B

Extension mobility is not support on DUT			**X**			

Test Case #	EP-29	Category	Functional Test: Hunt Group	RFC_Standard	N
Objective	Verify "Hunt Group" calls using DUT(s), SCCP, SIP and PSTN endpoints				
Pre-Test Conditions					

- Local CUCM→DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM →DUT:8000; SCCP DN:5200; SIP DN:6200;
- PSTN: 210-222-5400;
- Hunt Group Pilot 3000 (1^{st} member-7101; 2^{nd} member-1000; 3^{rd} member-2000;), Queuing flag enabled, max. waiting timer=60 secs, Route call to Destination=234-8000;
- RPC is used to remotely control IP Phones with DN: 1000, 2000, 2122225400;

Test Procedure	Expected Results
1. 7100 dials 3000➔7101 answers➔7100 on-hook after 60s 2. 7101 dials 2000➔2000 answers 3. 7100 dials 3000➔1000 answers➔1000 on-hook after 60s 4. 2000 goes on hook after 70s 5. 7101 dials 1000➔1000 answers 6. 7100 dials 3000➔2000 answers➔7100 on-hook after 60s 7. 2122225400 dials 4144443000➔2000 answers 8. 2000 goes on-hook after 60s 9. 1000 goes on-hook 10. Retrieve CDR from CUCM 11. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• Call route to hunt group member 7101 • Call establish between 7100 & 7101 with 2-way audio • Call terminate normally • 7101 & 2000 members are busy • Call route to hunt group member 1000 • Call establish between 7100 & 1000 with 2-way audio • Call terminate normally • 7101 & 1000 members are busy • Call route to hunt group member 2000 • Call establish between 7100 & 2000 with 2-way audio • Call terminate normally • Call route to hunt group member 2000 • Call establish between 21022254000 & 2000 with 2-way audio • Call terminate normally • 6 CDR(s) retrieved • Selected fields in CDR(s) match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

Test Case #	EP-30	Category	Functional Test: Hunt Group	RFC_Standard	N
Objective	Verify “Hunt Group” calls on DUT(s) when no members are available				
Pre-Test Conditions					
• Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000; • Remote CUCM ➔DUT:8000; SCCP DN:5200; SIP DN:6200; • PSTN: 210-222-5400;					

• Hunt Group Pilot 3010 (1st member-7101), Queuing flag enabled, max. waiting timer=60 secs, • Call Routing➔Route/Hunt➔Hunt Pilot➔3010➔Route call to this destination➔234-8000; • Call Routing➔Route/Hunt➔Hunt Pilot➔3012➔Route call to this destination➔1000; • Call Routing➔Route/Hunt➔Hunt Pilot➔3013➔Route call to this destination➔2000; • Call Routing➔Route/Hunt➔Hunt Pilot➔3014➔Route call to this destination➔92102225400; • RPC is used to remotely control IP Phones with DN: 1000, 2000, 2102225400;

Test Procedure	**Expected Results**
1. 7101 stays off-hook to make it unavailable 2. 7100 dials 3010➔8000 answers➔ 8000 on-hook after 60s 3. 7100 dials 3012➔1000 answers➔ 1000 on-hook after 60s 4. 7100 dials 3013➔2000 answers➔ 7100 on-hook after 60s 5. 7100 dials 3014➔2102225400 answers 6. 2102225400 goes on-hook after 60s 7. 7101 goes on-hook 8. Retrieve CDR from CUCM 9. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• HG member-7101 is unavailable • Hunt Group has no members available • Call route to hunt group alternate destination 8000 • Call establish between 7100 & 8000 with 2-way audio • Call terminate normally • Call route to hunt group alternate destination 1000 • Call establish between 7100 & 1000 with 2-way audio • Call terminated normally • Call routed to hunt group alternate destination 2000 • Call establish between 7100 & 2000 with 2-way audio • Call terminated normally • Call routed to hunt group alternate destination 2102225400 • Call establish between 7100 & 2102225400 with 2-way audio • Call terminated normally • 4 CDR(s) retrieved • Selected fields in CDR match calls

Test Results: Comments	**P**	**F**	**N/A**	**N/S**	**N/T**	**B**
	X					

Test Case #	EP-31	**Category**	Functional Test: Hunt Group	**RFC_Standard**	N
Objective	Verify “Hunt Group” calls on DUT(s) when maximum queue length exceeded				

Pre-Test Conditions
• Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000; • Remote CUCM ➔DUT:8000; SCCP DN:5200; SIP DN:6200; • PSTN: 210-222-5400; • Hunt Group Pilot 3015 (1st member-2000), Queuing flag enabled, max. waiting timer=60 secs, Route call to Destination disabled; Max. # of callers in queue=1; • RPC is used to remotely control IP Phones with DN: 1000, 2000, 2102225400;

Test Procedure	**Expected Results**
1. 7100 dials 3015➔2000 answers 2. 7101 dials 3015 3. 8000 dials 444-3015 4. 7100 goes on-hook after 200 secs 5. Retrieve CDR from CUCM 6. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• Call route to hunt group member 2000 • Call establish between 7100 & 2000 with 2-way audio • 7101 & 8000 waiting in queue • Maximum number of callers in queue exceeded • Maximum wait timer exceeded 60s • Both calls (8000 & 7101) were not terminated to hunt group • 3 CDR(s) retrieved • Selected fields in CDR(s) match calls

Test Results: Comments	**P**	**F**	**N/A**	**N/S**	**N/T**	**B**
	X					

Test Case #	EP-32	**Category**	Functional Test: Secure Endpoint	**RFC_Standard**	N
Objective	Verify “Authenticated” call between DUT(s), SCCP, SIP and PSTN endpoints				

Pre-Test Conditions							
• Local CUCM→DUT(s):7100 & 7101 (Authenticated); SCCP:1000; SIP:2000 (Non-Secure); SCCP:1001; SIP:2001 (Authenticated); • Remote CUCM →DUT:8000 (Non-Secure); • PSTN: 2102225400; • Enterprise Parameter: Cluster Security Mode→1 (Mixed Mode) • Assign authenticated Phone Security Profiles to devices: ➢ 7100 & 7101:Device→Phone→DN→Device Security Profile→ 3rd_Party_SIP_Advanced_Secure_Authenticated ➢ 1001: Device Security Profile==>7975_SCCP_Authenticated ➢ 2001: Device Security Profile→8945_SIP_Authenticated • RPC is used to remotely control IP Phones with DN: 1000,1001, 2000, 2001, 2102225400;							
Test Procedure	**Expected Results**						
1. 7100 dials 7101→7101 answers→7100 on-hook after 30s 2. 7100 dials 234-8000→8000 answers→8000 on-hook after 30s 3. 7100 dials 1001→1001 answers→1001 on-hook after 30s 4. 2001 dials 7101→7101 answers→7101 on-hook after 30s 5. 1000 dials 7100→7100 answers→7100 on-hook after 30s 6. 7101 dials 2000→2000 answers→2000 on-hook after 30s 7. 7100 dials 2102225400→2102225400 answers 8. 7100 goes on-hook after 30s 9. Retrieve CDR from CUCM 10. Check the Calling, Called, Duration, Secured Status, Origination & termination Cause Codes	• Authenticated call between 7100 & 7101 with 2-way audio • Call terminate normally • Non-Secure call between 7100 & 8000 with 2-way audio • Call terminate normally • Authenticated call between 7100 & 1001 with 2-way audio • Call terminate normally • Authenticated call between 2001 & 7101 with 2-way audio • Call terminate normally • Non-Secure call between 7100 & 1000 with 2-way audio • Call terminate normally • Non-Secure call between 2000 & 7101 with 2-way audio • Call terminate normally • Non-Secure call between 2102225400 & 7101 • 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						
Test Results: Comments	**P**	**F**	**N/A**	**N/S**	**N/T**	**B**	
	X						

<table>
<tr><td>Test Case #</td><td>EP-33</td><td>Category</td><td colspan="6">Functional Test: Join Across Line</td><td colspan="2">RFC_Standard</td><td>N</td></tr>
<tr><td>Objective</td><td colspan="11">Verify “Join Across Lines” calls between DUT(s), SCCP, SIP and PSTN endpoints</td></tr>
<tr><td colspan="12">Pre-Test Conditions</td></tr>
<tr><td colspan="12">
• Local CUCM→DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
• Remote CUCM →DUT:8000;
• PSTN: 2102225400;
• Enable JAL for all phones :Device→Phone→DN→Join across Lines→On
• Shared Line 1901 assigned to 7101, 1000, & 2000: Device→Phone→DN→2nd line→DN=1901
• RPC is used to remotely control IP Phones with DN: 1000, 2000, 2102225400;
</td></tr>
<tr><td colspan="3">Test Procedure</td><td colspan="9">Expected Results</td></tr>
<tr><td colspan="3">
1. 7100 dials 1901→1901 answers (Shared Line on 7101)
2. 8000 dials 444-7101→ 7101 answers
3. 7101 selects line 1901 and hits softkey “Join”
4. 8000 goes on-hook after 120s
5. 7100 dials 1901→1901 answers (Shared Line on 1000)
6. 7101 dials 1000→1000 answers
7.1000 selects line 1901 and hits softkey “Join”
8. 7101 goes on-hook after 120s
9. 7100 dials 1901→1901 answers (Shared Line on 2000)
10. 8000 dials 444-2000→2000 answers
11. 2000 selects line 1901 and hits softkey “Join”
12. 7100 goes on-hook after 120s
13. 2102225400 dials 94104441901
14. 1901 answers (Shared line on 7101)
15. 8000 dials 444-7101→7101 answers
16. 7101 selects line 1901 and hits softkey “Join”
17. 2102225400 goes on-hook after 120s
18. Retrieve CDR from CUCM
19. Check the Calling, Called, Duration, Origination & Termination Cause Codes
</td><td colspan="9">
• Call establish between 7100 & 1901 with 2-way audio
• 1901 is On-Hold (MOH)
• Call establish between 7101 & 8000 with 2-way audio
• 7100 & 8000 joined in a call, 7101 drops from call
• Call terminate normally
• Call establish between 7100 & 1901 with 2-way audio
• 1901 is On-Hold (MOH)
• Call establish between 7101 & 1000 with 2-way audio
• 7100 & 7101 joined in a call, 1000 drops from call
• Call terminate normally
• Call establish between 7100 & 1901 with 2-way audio
• 1901 is placed on-hold (MOH)
• Call establish between 7101 & 2000 with 2-way audio
• 7100 & 8000 join in a call. 2000 drops from call
• Call terminate normally
• Call establish between 2102225400 & 1901 with 2-way audio
• 1901 is placed on-hold (MOH)
• Call establish between 8000 & 7101 with 2-way audio
• 2102225400 & 8000 join in a call. 7101 drops from call
• Call terminate normally
• CDR(s)s retrieved
• Selected fields in CDR(s) match call
</td></tr>
<tr><td colspan="6">Test Results: Comments</td><td>P</td><td>F</td><td>N/A</td><td>N/S</td><td>N/T</td><td>B</td></tr>
<tr><td colspan="6">Join softkey is not supported</td><td></td><td></td><td></td><td>X</td><td></td><td></td></tr>
</table>

Test Case #	EP-34	Category	Functional Test: Hotline	RFC_Standard	N
Objective	Verify "Hotline" calls between DUT(s), SCCP, SIP and PSTN endpoints				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000;
- PSTN: 2102225400;
- Hotline Configuration to dial out 234-8000:
  - Call Routing➔ Class of Control ➔Partition➔Add New➔pt_hotline_2348000
  - Call Routing➔ Class of Control ➔Calling Search Space➔Add New➔css_hotline_2348000
  - Call Routing➔Translation Pattern➔Add New:
    - Translation Pattern➔ blank
    - Partition➔pt_hotline_2348000
    - CSS➔css_hotline_2348000
    - Called Party Transform Mask➔2348000
- Assign 7100, 1000, 2000 a CSS for Intercluster Hotline: Device➔Phone➔DN➔CSS➔css_hotline_2348000
- Assign 7101 a CSS for PSTN Hotline: Device ➔ Phone ➔ DN ➔CSS➔ css_hotline_2102225400 to DN:7101,
- RPC is used to remotely control IP Phones with DN: 1000,2000, 2102225400;

Procedure	Expected Results
1. 7100 goes off-hook➔8000 rings & answers 2. 7100 goes on-hook after 30s 3. 1000 goes off-hook➔8000 rings & answers 4. 1000 goes on-hook after 30s 5. 2000 goes off-hook➔8000 rings & answers 6. 8000 goes on-hook after 30s 7. 7101 goes off-hook➔2102225400 rings & answers 8. 7101 goes on-hook after 30 9. Retrieve CDR from CUCM 10. Check the Calling, Called, Duration, Origination & Termination Cause Codes **Note:** Upon completion of the test, change CSS for 7100, 7101, 1000 & 2000➔None CUCM Administration: Device➔Phone➔DN➔CSS➔None	• 8000 ringing • Call establish between 7100 & 8000 with 2-way audio • Call terminate normally • 8000 ringing • Call establish between 1000 & 8000 with 2-way audio • Call terminate normally • 8000 ringing • Call establish between 2000 & 8000 with 2-way audio • Call terminate normally • 2102225400 ringing • Call establish between 7101 & 2102225400 with 2-way audio • Call terminated normally • 4 CDR()s retrieved • Selected fields in CDR match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
DUT doesn't support post dial				X		

Test Case #	EP-35	Category	Functional Test: Group Pickup	RFC_Standard	N
Objective	Verify "Group Pickup" calls between DUT(s), SCCP, SIP and PSTN endpoints				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000;
- PSTN: 2102225400;
- Group Pickup configured on all phones; Group: Sales (DN: 7100 & 1000); Group: TAC (DN: 7101 & 2000);
  - Call Routing➔Call Pickup Group➔Add New➔Sales (DN:3005;Visual Alert; Calling & Called party checked)
  - Call Routing➔Call Pickup Group➔Add New➔TAC (DN:3006;Visual Alert; Calling & Called party checked)
  - Device➔Phone➔DN➔update Call Pickup Group to Sales for 7100 & 1000;
  - Device➔Phone➔DN➔update Call Pickup Group to TAC for 7101 & 2000;
- RPC is used to remotely control IP Phones with DN: 1000,2000, 2102225400;

Test Procedure	Expected Results
1. 8000 dials 444-7100 2. 1000 goes off-hook, hits "Group Pickup" softkey 3. 1000 enters Sales Group_Pickup DN:3005 4. 1000 goes on-hook after 60s 5. 8000 dials 444-2000 6. 7101 goes off-hook, hits "Group Pickup" softkey 7. 7101 enters TAC Group_Pickup DN:3006 8. 8000 goes on-hook after 60s 9. 2102225400 dials 4104441000 10. 7100 goes off-hook, hits "Group Pickup" softkey 11. 7100 enters Sales Group_Pickup DN:3005 12. 2102225400 goes on-hook after 60s 13. Retrieve CDR from CUCM 14. Check the Calling, Called, Duration, Origination & Termination Cause Codes **Note:** Upon completion of the test, change Call Pickup Group for 7100, 7101, 1000 & 2000➔None CUCM Administration: Device➔Phone➔DN➔Line➔Call Pickup Group➔None	• 7100 in alerting state • Call establish between 8000 & 1000 with 2-way audio • Call terminate normally • 2000 in alerting state • Call establish between 8000 & 7101 with 2-way audio • Call terminate normally • 1000 in alerting state • Call establish between 2102225400 & 7100 with 2-way audio • Call terminate normally • 6 CDR(s) retrieved • Selected fields in CDR match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
Group Pickup softkey is not implemented in DUT				X		

Test Case #	EP-36	Category	Functional Test: Do Not Disturb (DND)	RFC_Standard	Y
Objective	Verify "Do Not Disturb Ringer Off " feature is supported for DUT(s) endpoints				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000;
- PSTN: 2102225400;
- Enable DND on DN:7101
  - 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:7101:
    - Do Not Disturb➔checked
    - DND Option: Ringer Off
    - DND Incoming Call Alert: Flash Only
- RPC is used to remotely control IP Phones with DN: 1000,2000, 2102225400;

Test Procedure	Expected Results
1. 7100 dials 7101➔7100 goes on-hook after 5 secs 2. 8000 dials 444-7101➔8000 goes on-hook after 5 secs 3. 1000 dials 7101➔1000 goes on-hook after 5 secs 4. 2000 dials 7101➔2000 goes on-hook after 5 secs 5. 2102225400 dials 94104447101➔2102225400 goes on-hook 6. Retrieve CDR from CUCM 7. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• 7101 flashes to indicate incoming call • Called party hears a ring back tone • 7101 is given an option to answer call • Call terminated by Called party • 5 CDR(s) retrieved • Selected fields in CDR(s) match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
DUT is 3rd party device and doesn't support specific softkey template configurations.				X		

Test Case #	EP-37	Category	Functional Test: Do Not Disturb (DND)	RFC_Standard	N
Objective	Verify "Do Not Disturb Call Reject " feature is supported on DUT(s) endpoints				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000;
- PSTN: 2102225400;
- Enable DND on:7100
  - Device➔Phone➔DN:7100➔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:7101:
    - Do Not Disturb➔checked
    - DND Option: Call Reject
    - DND Incoming Call Alert: Beep Only
- RPC is used to remotely control IP Phones with DN: 1000, 1001, 2000, 2102225400;

Test Procedure	Expected Results
1. 1001 dials 7100➔7100 answers 2. 7101 dials 7100➔7100 hits "DND" softkey in connected state 3. 7101 goes on-hook 4. 8000 dials 444-7100➔7100 hits "DND" in connected state 5. 8000 goes on-hook 6. 1000 dials 7100➔7100 hits "DND" softkey in connected state 7. 1000 goes on-hook 8. 2000 dials 7100➔7100 hits "DND" softkey in connected state 9. 2000 goes on-hook 10. 2102225400 dials 7100➔7100 hits "DND" in connected state 11. 1001 goes on-hook after 200s 12. Retrieve CDR from CUCM 13. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• Call establish between 1001 & 7100 with 2-way audio • 7100 hears ringback for all incoming calls in connected state • CUCM rejects call with Reason:User Busy • 7100 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

Test Results: Comments	P	F	N/A	N/S	N/T	B
DUT is 3rd party device and doesn't support specific softkey template configurations.				X		

Test Case #	EP-38	Category	Functional Test: iDivert	RFC_Standard	N
Objective	Verify "IDivert " call between DUT(s), SCCP, SIP and PSTN endpoints				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000;
- PSTN: 2102225400;
- VM enabled on all phones; VM Pilot # 7000; Device➔Phone➔DN➔Line➔Voicemail➔Default
- Enable iDivert on DN:7101
  - 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:7101➔Softkey Template➔SIP_EP_User
- RPC is used to remotely control IP Phones with DN: 1000,2000, 2102225400;

Test Procedure	Expected Results
1. 7100 dials 7101➔7101 hits "iDivert" softkey during ringing state 2..7100 leaves a voicemail and goes on-hook 3. 8000 dials 444-7101➔7101 hits "iDivert" softkey in ringing state 4. 8000 goes on-hook without leaving a message 5. 1000 dials 7101➔7101 hits "iDivert" softkey during ringing state 6. 1000 leaves a voicemail and goes on-hook 7. 2000 dials 7101➔7101 hits "iDivert" softkey during ringing state 8. 2000 leaves a voicemail and goes on-hook 9. 2102225400 dials 94104447101 10. 7101 hits "iDivert" softkey during ringing state 11. 2102225400 leaves a voicemail and goes on-hook 12. Retrieve CDR from CUCM 13. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• 7100 directed to 7101's voicemail box • 8000 directed to 701's voicemail box • 1000 directed to 7101's voicemail box • 2000 directed to 7101's voicemail box • 2102225400 directed to 7101's voicemail box • MWI "On" when voicemail present for 7101 • 7101 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

Test Results: Comments	P	F	N/A	N/S	N/T	B
DUT is configured as a 3rd party Device and ergo doesn't support softkey template configurations.				X		

Test Case #	EP-39	Category	Functional Test: CFA & iDivert	RFC_Standard	N
Objective	Verify "CFA" & "iDivert " call between DUT(s), SCCP, SIP and PSTN endpoints				

Pre-Test Conditions

- Local CUCM→DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM →DUT:8000;
- PSTN: 2102225400;
- VM enabled on all phones; CFA enabled on 7101→7100; VM Pilot # 7000;
- Enable iDivert on DN:7100
  - 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:7100→Softkey Template→SIP_EP_User
- RPC is used to remotely control IP Phones with DN: 1000,2000;

Test Procedure	Expected Results
1. 8000 dials 4447101→7100 hits "iDivert" softkey in ringing state 2. 8000 leaves voicemail and goes on-hook 3. 1000 dials 7101→7100 answers 4. 7100 hits "iDivert" softkey during connected state (after 10s) 5.1000 leaves a voicemail and goes on-hook 6. 2000 dials 7101→7100 answers 7. 7100 hits "iDivert" softkey during connected state (after 20s) 8. 2000 leaves a voicemail and goes on-hook 9. 2102225400 dials 94104447101→7100 answers 10. 2102225400 goes on-hook without leaving voicemail 11. Retrieve CDR from CUCM 12. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• 7101 call forward to 7100 (ringing state) • 8000 directed to 7100's voicemail box • Call terminate normally • Call establish between 1000 & 7100 with 2-way audio • 1000 directed to 7100's voicemail box after 10s • Call terminate normally • Call establish between 2000 & 7100 with 2-way audio • 2000 directed to 7100's voicemail box after 20s • Call terminate normally • Call establish between 2102225400 & 7100 with 2-way audio • 2102225400 directed to 7100's voicemail box after 20s • Call terminate normally • MWI "On" when a voicemail is left on 7100's mailbox • 7100 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

Test Results: Comments	P	F	N/A	N/S	N/T	B
DUT is configured as a 3rd party Device and ergo doesn't support softkey template configurations.				X		

Test Case #	EP-40	Category	Functional Test: Malicious Call	RFC_Standard	N
Objective	Verify DUT(s) is able to mark a call malicious				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000;
- PSTN: 2102225400;
- Update softkey template on device➔SIP_EP_User (Includes MCID softkey)
- RPC is used to remotely control IP Phones with DN: 1000, 2000;

Test Procedure	Expected Results
1. 7100 dials 7101➔7101 answers➔7101 hits “MCID” softkey 2. 7101 goes on-hook after 30s 3. 8000 dials 444-7100➔7100 answers➔7100 hits “MCID” 4. 8000 goes on-hook after 30s 5. 1000 dials 7100➔7100 answers==7100 hits “MCID” softkey 6. 7100 goes on-hook after 30s 7. 2000 dials 7100➔7100 answers➔7100 hits “MCID” softkey 8. 2000 goes on-hook after 30s 9. 2102225400 dials 94104447101➔7101 answers 10. 7101 hits “MCID” softkey 11. 7101 goes on-hook after 30s 12. Retrieve CDR from CUCM 13. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• 3 calls establish with 2-way audio on 7100 • 2 calls establish with 2-way audio on 7101 • Called parties marke as malicious • Call terminate normally • 5 CDR(s) retrieved • Selected fields in CDR match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
MCID softkey not implemented in DUT.			X			

Test Case #	EP-41	Category	Functional Test: Mobile Connect	RFC_Standard	N
Objective	Verify DUT(s) supports Mobile Connect call to a remote DUT endpoint				
Pre-Test Conditions					

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000;
- PSTN: 2102225400;
- Configure CUCM Service Parameter: Device Mobility Mode➔On
- Mobile Voice Access service running on CUCM-PUB
- Mobile Voice Access enabled on Voice Gateway
- **Local Cluster Single Number Reach (SNR)** configured for 7101➔Remote Device:234-8000:
  - Add Remote Destination Profile:Device➔Device Settings➔Remote Destination Profile➔Add New➔mobility_1_rdp
    - Userid:dutuser02
    - Line:7101
    - Add New Remote Destination:
      - Name➔Mobility_1
      - Destination Number➔ 2348000
      - 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➔7101➔Owner userid➔dutuser02
  - Remote Device: Device➔Phone➔8000➔Line➔No Answer Ring Duration➔60
- RPC is used to remotely control IP Phones with DN: 1000,2000, 2102225400;

Test Procedure	Expected Results
1. 7100 dials 7101➔7101 answers➔7101 hits "Mobility" softkey after 30s 2. Select the option to send call to mobile device 3. 8000 answers 4. 7101 goes on-hook after 30s 5. 8000 sends DTMF *74 after 30s for call hanoff 6. 7101 answers 7. 8000 goes on-hook 8. 7101 goes on-hook after 30s 9. Repeat steps 1-7, replace 7100 with SCCP:1000 10. Repeat steps 1-7, replace 7100 with SIP:2000 11. Repeat steps 1-7, replace 7100 with PSTN:2102225400 12. Retrieve CDR from CUCM 13. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• Both 7101(local) & 8000 (remote) are ringing • Call establish between 7100 & 7101 with 2-way audio • Call is transferred to mobile device 8000 • Call establish between 7100 & 8000 with 2-way audio • 8000 handoff call back to 7101 • Call restored between 7100 & 7101 • Final Call terminate normally • Results for SCCP, SIP & PSTN calls are similar as above • 6 CDR(s) retrieved • Selected fields in CDR(s) match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
DUT doesn't support Mobility softkey.			X			

Test Case #	EP-42	Category	Functional Test: Mobile Connect	RFC_Standard	N
Objective	Verify DUT(s) supports Mobile Connect call to a remote PSTN endpoint				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000;
- PSTN: 2102225401;
- Configure CUCM Service Parameter: Device Mobility Mode➔On
- Mobile Voice Access service running on CUCM-PUB
- Mobile Voice Access enabled on Voice Gateway
- **Local Cluster Single Number Reach (SNR)** configured for 7100➔ Remote Device:92102225401
    - Add Remote Destination Profile:Device➔Device Settings➔Remote Destination Profile➔Add New➔mobile2_rdp
        - Userid:dutuser01
        - Line:7100
        - Add New Remote Destination:
            - Name➔Mobility_2
            - Destination Number➔ 92102225401
            - 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➔dutuser01➔Check "Enable Mobility" &"Enable Mobile Voice Access"

◆ Device→Phone→7100→Owner userid→dutuser01 ◆ Remote Device: Device→Phone→2102225401→Line→No Answer Ring Duration→60 • RPC is used to remotely control IP Phones with DN: 1000,2000, 2102225401;

Test Procedure	**Expected Results**
1. 7101 dials 7100→7100 answers 2. 7100 hits "Mobility" softkey after 30s and selects to send call to mobile 3. 2102225401 answers 4. 7100 goes on-hook 5.2102225401 sends DTMF *74 after 30s (call handoff) 6.7100 answers 7.2102225401 goes on-hook 8.7101 goes on-hook after 30s 9. Repeat steps 1-8 and replace the Calling DN:1000 10. Repeat steps 1-8 and replace Calling DN:2000 11. Retrieve CDR from CUCM 12. Check the Calling, Called, Duration, Origination & Termination Cause Codes and Comments	• Both 7101 & 7100 are ringing • Call establish between 7100 & 7101 with 2-way audio • Call transfer to mobile device 2102225401 (PSTN) • Call establish between 7100 & 2102225401 with 2-way audio • 2102225401 handoff call back to 7100 • Call restored between 7100 & 7101 • 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

Test Results: Comments	**P**	**F**	**N/A**	**N/S**	**N/T**	**B**
DUT doesn't support Mobility softkey			**X**			

Test Case #	EP-43	**Category**	Functional Test: Mobile Connect	**RFC_Standard**	N
Objective	Verify DUT(s) supports Mobile Connect call to a remote SIP endpoint				
Pre-Test Conditions					
• Local CUCM→DUT(s):7100 & 7101; SCCP:1000; SIP:2000; • Remote CUCM →DUT:8000; SIP:6201; • 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 6201 – Remote Device:444-7101 ◆ Add Remote Destination Profile:Device→Device Settings→Remote Destination Profile→Add New→mobile1_rdp ○ Userid:rcuser21 ○ Line:6201 ○ Add New Remote Destination: ◆ Name→Mobility_1 ◆ Destination Number→ 4447101					

✚ 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 Mobility" &"Enable Mobile Voice Access" ✚ Device➔Phone➔6201➔Owner userid➔rcuser21 ✚ Remote Device: Device➔Phone➔7101➔Line➔No Answer Ring Duration➔60 • RPC is used to remotely control IP Phones with DN: 1000,2000, 6201;	
Test Procedure	**Expected Results**
1. 7100 dials 2346201➔7101 answers 2. 7101 sends DTMF *74" after 30s 3. 6201 answers 4. 7101 goes on-hook 5. 6201 hits "Mobility" softkey after 30s 6. 7101 answers 7. 6201 goes on-hook 8. 7100 goes on-hook after 60s 9. Repeat steps 1-8 and replace the Calling DN:1000 10. Repeat steps 1-8 and replace the Calling DN:2000 11. Retrieve CDR from CUCM 12. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• Both 6201 & 7101 are ringing • Call establish between 7100 & 7101 with 2-way audio • Call is transferred to device 6201 • Call establish between 7100 & 6201 with 2-way audio • 2000 handoff call back to 7101 • Call restored between 7100 & 7101 • 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

Test Results: Comments	P	F	N/A	N/S	N/T	B
DUT doesn't support Mobility softkey			X			

Test Case #	EP-44	**Category**	Functional Test: Mobile Voice Access (MVA)	**RFC_Standard**	N
Objective	Verify Inbound Mobile Voice Access (MVA) calls from DUT(s) endpoints				
Pre-Test Conditions					
• Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000; • Remote CUCM ➔DUT:8000; • 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 7101→Remote Device:234-8000:
    ◦ Add Remote Destination Profile:Device→Device Settings→Remote Destination Profile→Add New→mobile1_rdp
        ○ Userid:dutuser02
        ○ Line:7101
        ○ Add New Remote Destination:
            ◦ Name→Mobility_1
            ◦ Destination Number→ 2348000
            ◦ 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→7101→Owner userid→dutuser02
    ◦ Remote Device: Device→Phone→8000→Line→No Answer Ring Duration→60
• RPC is used to remotely control IP Phones with DN: 1000,2000;

Test Procedure	**Expected Results**
1.8000 (Mobil device) dials MVA #8005555 2. Mobil User enters 3222348000# or 2348000# 3.Mobil User enters PIN:123456# & DN:7100# 4.7100 answers 5. 8000 sends DTMF *74 to handoff session after 30s 6. 7100 goes on-hook after 60s 7. Retrieve CDR from CUCM 8. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• Mobile user prompted for Caller ID: 3222348000 • Mobile user prompted for PIN & Destination DN • 7100 is ringing • Call establish between 8000 & 7100 with 2-way audio • Call hand-off to 7101 • Call terminate normally • 1 CDR retrieved • Selected fields in CDR matches call

Test Results: Comments	**P**	**F**	**N/A**	**N/S**	**N/T**	**B**
DUT doesn’t support Mobility softkey.				X		

Test Case #	EP-45	**Category**	Functional Test: Enterprise Feature Access (EFA)	**RFC_Standard**	N
Objective	Verify Inbound Enterprise Feature Access (EFA) - Hold/Resume call from a DUT endpoint				
Pre-Test Conditions					
• Local CUCM→DUT(s):7100 & 7101; SCCP:1000; SIP:2000; • Remote CUCM →DUT:8000; • 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 7101➔Remote Device:234-8000: ✚ Add Remote Destination Profile:Device➔Device Settings➔Remote Destination Profile➔Add New➔mobile1_rdp o Userid:dutuser02 o Line:7101 o Add New Remote Destination: ✚ Name➔Mobility_1 ✚ Destination Number➔ 2348000 ✚ 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➔7101➔Owner userid➔dutuser02 ✚ Remote Device: Device➔Phone➔8000➔Line➔No Answer Ring Duration➔60 • RPC is used to remotely control IP Phones with DN: 1000,2000;	
Test Procedure	**Expected Results**
1. 8000 (Mobil device) dials EFA #9005555 2. Mobil user prompted to enter remote device DN 2348000# 3. Mobil User enters PIN:123456#, Option 1 & DN:7100# 4. 7100 answers call 5. 8000 sends DTMF *81 to place call on-hold after 30s 6. 8000 sends DTMF *83 to resume call after 30s 7. 8000 goes on-hook after 120s 8. Retrieve CDR from CUCM 9. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• Mobile user prompted for Caller ID: 2348000 • Mobile user prompted for PIN • Selects option 1 and enters Called DN:7100 • 7100 is ringing • Call establish between 8000 & 7100 with 2-way audio • 7100 is On-Hold • Call resumes • Cal terminate normally • 1 CDR retrieved • Selected fields in CDR matches call

Test Results: Comments	P	F	N/A	N/S	N/T	B
DUT doesn't support Mobility softkey.				X		

9.5 Negative Tests

Test Case #	EP-46	**Category**	Negative Test: PUB Failure	**RFC_Standard**	Y
Objective	Verify a PUB failure should not affect stable or transient calls on DUT(s)				

Pre-Test Conditions							
• Local CUCM➔DUT(s):7100 & 7101;SCCP:1000; SIP:2000; • Remote CUCM➔DUT:8000; • RPC is used to remotely control IP Phones with DN: 1000,2000;							
Test Procedure	**Expected Results**						
1. 7100 dials 7101➔7101 answers 2. 2000 dials 234-8000➔8000 answers 3. Access CUCM-PUB server via SSH (Local Cluster) 4. Enter CLI: **utils system restart <CR>** **yes** 5. 1000 dials 7101➔7101 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	• Call establish between 7100 & 7101 with 2-way audio • Call establish between 2000 & 8000 with 2-way audio • CUCM-PUB is restarted • Stable calls not impacted by PUB restart • Call establish between 1000 & 7101 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						
Test Results: Comments	**P**	**F**	**N/A**	**N/S**	**N/T**	**B**	
	X						

Test Case #	EP-47	**Category**	Negative Test: SUB Failure	**RFC_Standard**	Y
Objective	Verify a SUB failure should not affect stable calls on DUT(s)				

Pre-Test Conditions							
• Local CUCM➔DUT(s):7100 & 7101;SCCP:1000; SIP:2000; • Remote CUCM➔DUT:8000; • RPC is used to remotely control IP Phones with DN: 1000,2000;							
Test Procedure	**Expected Results**						
1. 7100 dials 7101➔7101 answers 2. 2000 dials 234-8000➔8000 answers call 3. Access CUCM-SUB server via SSH (Local Cluster) 4. Enter CLI: **utils system restart <CR>** **yes** 5. 1000 dials 7101 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	• Call establish between 7100 & 7101 with 2-way audio • Call establish between 2000 & 8000 with 2-way audio • CUCM-SUB is restarted • Stable calls not impacted by SUB restart • Transient calls impacted by SUB restart • Call between 1000 & 7101 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						
Test Results: Comments	**P**	**F**	**N/A**	**N/S**	**N/T**	**B**	
	X						

Test Case #	EP-48	**Category**	Negative Test: Phone Network Failure	**RFC_Standard**	Y
Objective	Verify DUT(s) recovers from a network failure				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101;SCCP:1000; SIP:2000;
- Remote CUCM➔DUT:8000;
- RPC is used to remotely control IP Phones with DN: 1000,2000;

Test Procedure	Expected Results
1. 7100 dials 7101➔7101 answers 2. Unplug network EPle from device DN:7100 3. Restore the network EPle after 60s 4. 2000 dials 7100➔7100 answers 5. 7100 goes on-hook after 60s 6. Retrieve CDR from CUCM 7. Check Calling, Called, Duration, Origination & Termination Cause Codes	• Call establish between 7100 & 7101 with 2-way audio • Network failure reported on device DN:7100 • Stable call drops • Device 7100 re-registers after network EPle restored • Network Data: DNS, DHCP, TFTP, CUCM, VLAN, Load ID are restored on device • Call establish between 2000 & 7100 with 2-way audio • Call terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

Test Case #	EP-49	Category	Negative Test: Phone Power Failure	RFC_Standard	Y
Objective	Verify DUT(s) recovers from a power failure				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101;SCCP:1000; SIP:2000;
- Remote CUCM➔DUT:8000;
- RPC is used to remotely control IP Phones with DN: 1000,2000;

Test Procedure	Expected Results
1. 7100 dials 7101➔7101 answers 2. Remove power cable from 7101 3. Restore power cable after 60s 4. 2000 dials 7101➔7101 answers call 5. 7101 goes on-hook after 60s 6. Retrieve CDR from CUCM 7. Check Calling, Called, Duration, Origination & Termination Cause Codes	• Call establish between 7100 & 7101 with 2-way audio • 7101 lost power • Stable call drops • Device 7101 re-registers after power is restored • Network Data: DNS, DHCP, TFTP, CUCM, VLAN, Load ID are restored on device • Call establish between 2000 & 7101 with 2-way audio • Call terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

Test Case #	EP-50	Category	Negative Test: Abnormal Call Scenarios	RFC_Standard	Y
Objective	Verify calls on DUT for negative call scenarios (Invalid DN, Busy DN, Abandoned, RNA)				

Pre-Test Conditions

- Local CUCM→DUT(s):7100 & 7101;SCCP:1000; SIP:2000;
- Remote CUCM→DUT:8000;
- Invalid DN:7777
- Call Waiting disabled for 7100 & 7100
- Device Profile: Busy Trigger set to 1 for DN: 7100 & 7101
- Voicemail disabled for 1000 & 2000;
- RPC is used to remotely control IP Phones with DN: 1000,2000;

Test Procedure	Expected Results
1. 7100 dials 1000→1000 answers 2. 7101 dials 7100 3. 7101 dials 1000 4. 7101 dials 2000 (RNA) 5. 7101 dials 7777 (Invalid DN) 6. 7101 dials 234-8000 7. 7101 goes on-hook before 8000 answers (Abandoned) 8. 7100 goes on-hook after 120 secs 9. Retrieve CDR from CUCM 10. Check Calling, Called, Duration, Origination & Termination Cause Codes	• Call establish between 7100 & 1000 with 2-way audio • 7101 hears busy tone for calls to 7100 & 1000 • 7101 hears ring back timeout for call to 2000 • 7100 hears reorder tone when it dialed invalid DN:7777 • 1 Call terminated normally • 5 unsuccessful call attempts • 6 CDR(s) retrieved • Selected fields in CDR match calls

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

9.6 Miscellaneous Tests

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

Test Case #	EP-51	**Category**	Miscellaneous Test: Codec (G722 & G729)	**RFC_Standard**	Y
Objective	Verify URI calls between DUT(s) & SIP endpoints for In-band Codec (G722, G729)				

- Local CUCM➔DUT(s):7100 (dutuser01@abc.inc) ; 7101(dutuser02@abc.inc);
  SCCP:1000 (cuser01@abc.inc); SIP:2000;cuser20@abc.inc;
- Remote CUCM ➔DUT:8000;
- 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 7100, 7101 with device pool=G722-dp
- Configure Speed Dial for 7100, 7101, 2000:
  - Device➔Phone➔7100➔Add new SD➔dutuser02@abc.inc
  - Device➔Phone➔7101➔Add new SD➔cuser20@abc.inc
  - Device➔Phone➔2000➔Add new SD➔dutuser02@abc.inc
  - Device➔Phone➔1000➔Add new SD➔dutuser01@abc.inc
- RPC is used to remotely control IP Phones with DN: 2000;

Test Procedure	**Expected Results**
1. 7100 hits Speed dial button 2. 7101 answers call 3. 7101 goes on-hook after 60s 4. 2000 hits Speed dial button 5. 7101 answers call 6. 2000 goes on-hook after 60s 7. 7101 hits Speed dial button 8. 2000 answers call 9. 7101 goes on-hook after 60s 10. 1000 hits Speed dial button 11. 7100 answers call 12. 7100 goes on-hook after 60s 13. Repeat steps 1-9 with device pool of G720-dp for 7100 & 7101 14. Retrieve CDR from CUCM Server 15. Check Calling, Called, Duration, Origination & Termination Cause Codes	• 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

Test Results: Comments	**P**	**F**	**N/A**	**N/S**	**N/T**	**B**
DUT GUI uses 'Favorites" option in place of "Speed Dial". Same functionality.	**X**					

<table>
<tr><td>Test Case #</td><td>EP-52</td><td>Category</td><td colspan="6">Miscellaneous Test: DUT display features</td><td>RFC_Standard</td><td>Y</td></tr>
<tr><td>Objective</td><td colspan="10">Verify different packetization period support on DUT(s) endpoints</td></tr>
<tr><td colspan="11">Pre-Test Conditions</td></tr>
<tr><td colspan="11">• Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
• Remote CUCM ➔DUT:8000;
• Configure Service Parameter: Preferred G.711 Millisecond Packet Size :10
• RPC is used to remotely control IP Phones with DN: 1000, 2000;</td></tr>
<tr><td colspan="4">Test Procedure</td><td colspan="7">Expected Results</td></tr>
<tr><td colspan="4">1. 7100 dials 7101➔7101 answers
2. 7100 goes on-hook after 60s
3. 7100 dials 234-8000➔8000 answers
4. 8000 goes on-hook after 60s
5. 7100 dials 1000➔1000 answers
6. 7100 goes on-hook after 60s
7. 2000 dials 7100➔7100 answers
8. 7100 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</td><td colspan="7">• Call establish between 7100 & 7101 with 2-way audio
• Call establish between 7100 & 8000 with 2-way audio
• Call establish between 7100 & 1000 with 2-way audio
• Call establish between 7100 & 2000 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</td></tr>
<tr><td colspan="5">Test Results: Comments</td><td>P</td><td>F</td><td>N/A</td><td>N/S</td><td>N/T</td><td>B</td></tr>
<tr><td colspan="5">DUT only supports ptime-20ms</td><td></td><td></td><td>X</td><td></td><td></td><td></td></tr>
</table>

<table>
<tr><td>Test Case #</td><td>EP-53</td><td>Category</td><td colspan="6">Miscellaneous Test: DUT Screen Features</td><td>RFC_Standard</td><td>Y</td></tr>
<tr><td>Objective</td><td colspan="10">Verify the features displayed on the screen of DUT</td></tr>
<tr><td colspan="11">Pre-Test Conditions</td></tr>
<tr><td colspan="11">• Local CUCM➔ DUT (s):7100 & 7101;</td></tr>
<tr><td colspan="4">Test Procedure</td><td colspan="7">Expected Results</td></tr>
<tr><td colspan="4">Check DUT:7100 phone display for:
• Missed Calls
• Placed Calls
• Received Calls
• Date & Time
• Clear Call History (Missed, Placed,Received)
• Redial or Dial from Call History List
• Softkeys for call features
• Multiple lines
• Edit Called #</td><td colspan="7">• Able to access all these features from the DUT's phone display</td></tr>
<tr><td colspan="5">Test Results: Comments</td><td>P</td><td>F</td><td>N/A</td><td>N/S</td><td>N/T</td><td>B</td></tr>
<tr><td colspan="5">DUT doesn't support multiple lines.</td><td>X</td><td></td><td></td><td></td><td></td><td></td></tr>
</table>

Test Case #	EP-54	Category	Miscellaneous Test: Long Duration Calls	RFC_Standard	Y
Objective	Verify long duration calls between DUT(s), SCCP, SIP and PSTN endpoints				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000;
- PSTN DN: 210-222-5400;
- RPC is used to remotely control IP Phones with DN: 1000, 2000, 2102225400;

Test Procedure	Expected Results
1. 7100 dials 7101➔7101 answers (Duration: 1 Hr.) 2. 2000 dials 234-8000➔ 8000 answers (Duration: 1 Hr) 3. Repeat step 1 by replacing Called DN:92102225400 4. Repeat step 2 by replacing the Calling DN: 1000 5. Retrieve CDR from CUCM 6. Check Calling, Called, Duration, Origination & Termination Cause Codes	• Call establish between 7100 & 7101 with 2-way audio • Call establish between 2000 & 8000 with 2-way audio • Call establish between 7100 & 2102225400 with 2-way audio • Call establish between 1000 & 8000 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

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

Test Case #	EP-55	Category	Miscellaneous Test: Cisco Phone Models	RFC_Standard	Y
Objective	Verify calls and mid-call features between DUT(s) and various Cisco IP Phone Models				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000;;
- Cisco Phone Models: 6961,8861, 8945, 7925, 9971, DX650
- For phone models not supported by RPC, Auto-Answer is enabled. (DX650)
- RPC is used to remotely control IP Phones

Test Procedure	Expected Results
1. 7100 dials 1100➔ 1100 answers➔7100 on-hook after 120s 2. 1100 dials 7100➔7100 answers➔1100 on-hook after 120s 3. 7101 dials 1100 ➔1100 answers➔7101 hits "Hold" after 20s 4. 7101 hits "Resume" after 20s➔1100 on-hook after 120s 5. 1100 dials 234-8000➔8000 answers➔8000 on-hook after 90s 6. 7100 dials 1100➔1100 answers➔ 1100 hits "Transfer" after 20s 7. 1100 dials 7101➔1100 hits "Transfer➔ 1100 on-hook 8. 7100 goes on-hook after 120s 9. 1100 dials 7100➔7100 answers➔7100 hits "Transfer" after 20s 10. 7100 dials 234-8000➔8000 answers 11. 7100 hits "Transfer" after 30s➔7100 on-hook 12. 8000 goes on-hook after 120s 13. 1100 dials 7101➔7101 answers 14. 7101 hits "Conference" after 30s➔7101 dials 234-8000 15. 8000 answers➔7101 hits "Conference" after 30s 16. 7101 goes on-hook after 120s 17. 1100 and 8000 goes on-hook after 200s 18. Repeat steps 1-17 by replacing DN:1100 with DN(s) of other Cisco phone models 19. Retrieve CDR from CUCM 20.Check Calling, Called, Duration, Origination & Termination Cause Codes	• 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 **Note:** Any Cisco IP Phone models not supported in RPC will have Auto Answer Turned On to test basic call functions only.

Test Results: Comments	P	F	N/A	N/S	N/T	B
DUT doesn't support conference.	X					
Test Results: Comments	**P**	**F**	**N/A**	**N/S**	**N/T**	**B**

Test Case #	EP-56	Category	Functional Test: Multiple Lines	RFC_Standard	Y
Objective	Verify DUT is able to handle calls and mid-call features on multiple lines				

Pre-Test Conditions

- Local CUCM➔DUT(s):7100 & 7101; SCCP:1000; SIP:2000;
- Remote CUCM ➔DUT:8000; SIP:6200; SCCP:5200;
- RPC is used to remotely control IP Phones with DN: 1000,2000;
- Assumption: DUT is Advanced 3rd Party SIP Endpoint with multiple lines

Test Procedure	Expected Results
1. Provision all the lines for both 7100 & 7101 Device➔Phone➔DN➔Line (DN Range: 7120 -7150) 2. Initiate calls on all the lines between 7100 & 7101 3. Calling & Called parties goes on-hook alternatively at random duration 4. Initiate intra-cluster & inter-cluster calls on all lines to SIP, SCCP & PSTN endpoints 5. Calling & Called parties goes on-hook alternatively at random Duration. 6. Initiate calls and perform mid-call features between these lines (Hold/Resume, Transfer, Conference, CFNA, CFB) 7. Retrieve CDR from CUCM 8. Check the Calling, Called, Duration, Origination & Termination Cause Codes	• 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

Test Results: Comments	P	F	N/A	N/S	N/T	B
DUT doesn't support multiple lines			X			

9.7 Basic call features using Expressway

<table>
<tr><td>Test Case #</td><td>EP-57</td><td>Category</td><td>Basic call features using Expressway</td><td>RFC_Standard</td><td></td></tr>
<tr><td>Objective</td><td colspan="5">Verify DUT is able to handle basic inbound call from external endpoint via Expressway</td></tr>
<tr><td colspan="6">Pre-Test Conditions</td></tr>
<tr><td colspan="6">• DUT is registered as internal user on CUCM
• Cisco Phone/client registered as external endpoint via Expressway-E</td></tr>
<tr><td colspan="3">Test Procedure</td><td colspan="3">Expected Results</td></tr>
<tr><td colspan="3">1. Initiate call from external endpoint to DUT
2. Answer the call in DUT
3. Disconnect the call from originating end</td><td colspan="3">• DUT starts ringing
• Verify audio connected successfully
• Verify call is disconnected successfully</td></tr>
</table>

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

<table>
<tr><td>Test Case #</td><td>EP-58</td><td>Category</td><td>Basic call features using Expressway</td><td>RFC_Standard</td><td></td></tr>
<tr><td>Objective</td><td colspan="5">Verify DUT is able to handle basic inbound call from external endpoint via Expressway</td></tr>
<tr><td colspan="6">Pre-Test Conditions</td></tr>
<tr><td colspan="6">• DUT is registered as internal user on CUCM
• Cisco Phone/client registered as external endpoint via Expressway-E</td></tr>
<tr><td colspan="3">Test Procedure</td><td colspan="3">Expected Results</td></tr>
<tr><td colspan="3">1. Initiate call from external endpoint to DUT
2. Answer the call in DUT
3. Disconnect the call from terminating end</td><td colspan="3">• DUT starts ringing
• Verify audio connected successfully
• Verify call is disconnected successfully</td></tr>
</table>

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

<table>
<tr><td>Test Case #</td><td>EP-59</td><td>Category</td><td>Basic call features using Expressway</td><td>RFC_Standard</td><td></td></tr>
<tr><td>Objective</td><td colspan="5">Verify DUT is able to place a basic outbound call to external endpoint via Expressway</td></tr>
<tr><td colspan="6">Pre-Test Conditions</td></tr>
</table>

• DUT is registered as internal user on CUCM • Cisco Phone/client registered as external endpoint via Expressway-E

Test Procedure	Expected Results
1. Initiate call from DUT to external endpoint 2. Answer the call in external endpoint 3. Disconnect the call from originating end	• External endpoint starts ringing • Verify audio connected successfully • Verify call is disconnected successfully

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

Test Case #	EP-60	Category	Basic call features using Expressway	RFC_Standard	
Objective	Verify DUT is able to place a basic outbound call to external endpoint via Expressway				
Pre-Test Conditions					
• DUT is registered as internal user on CUCM • Cisco Phone/client registered as external endpoint via Expressway-E					

Test Procedure	Expected Results
1. Initiate call from DUT to external endpoint 2. Answer the call in external endpoint 3. Disconnect the call from terminating end	• External endpoint starts ringing • Verify audio connected successfully • Verify call is disconnected successfully

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

Test Case #	EP-61	Category	Basic call features using Expressway	RFC_Standard	

Objective	Verify DUT is able to place an outbound call to external endpoint via Expressway on hold / resume

Pre-Test Conditions

- DUT is registered as internal user on CUCM
- Cisco Phone/client registered as external endpoint via Expressway-E

Test Procedure	Expected Results
1. Initiate call from DUT to external endpoint 2. Answer the call in external endpoint 3. Place the call on hold from DUT 4. Resume the call on DUT 5. Disconnect the call from originating end	• External endpoint starts ringing • Verify audio connected successfully • Verify the call is on hold state successfully • Verify the call is resumed successfully • Verify call is disconnected successfully

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

Test Case #	EP-62	Category	Basic call features using Expressway	RFC_Standard	
Objective	Verify DUT is able to place an inbound call from external endpoint via Expressway on hold / resume				

Pre-Test Conditions

- DUT is registered as internal user on CUCM
- Cisco Phone/client registered as external endpoint via Expressway-E

Test Procedure	Expected Results
1. Initiate call from external endpoint to DUT 2. Answer the call in DUT 3. Place the call on hold from DUT 4. Resume the call on DUT 5. Disconnect the call from originating end	• DUT starts ringing • Verify audio connected successfully • Verify the call is on hold state successfully • Verify the call is resumed successfully • Verify call is disconnected successfully

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

Test Case #	EP-63	Category	Basic call features using Expressway	RFC_Standard	
Objective	Verify DUT is able to transfer the call from a local phone to external phone registered via Expressway				

Pre-Test Conditions

- DUT is registered as internal user on CUCM
- Cisco Phone/client registered as local endpoint on CUCM
- Cisco Phone/client registered as external endpoint via Expressway-E

Test Procedure	Expected Results
1. Initiate call from local endpoint to DUT 2. Answer the call in DUT 3. DUT calls external endpoint 4. Answer the call in external endpoint 5. DUT transfers the first call with local endpoint to external endpoint 6. Disconnect the call from local endpoint	• DUT starts ringing • Verify audio connected successfully • Verify the first call is on hold state and external endpoint starts ringing • Verify the call is connected successfully • Verify transfer is successful • Verify call is disconnected successfully

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

Test Case #	EP-64	Category	Basic call features using Expressway	RFC_Standard	
Objective	Verify DUT is able to add an external endpoint registered via Expressway into a conference				

Pre-Test Conditions

- DUT is registered as internal user on CUCM
- Cisco Phone/client registered as local endpoint on CUCM
- Cisco Phone/client registered as external endpoint via Expressway-E

Test Procedure	Expected Results
1. Initiate call from DUT to local endpoint 2. Answer the call in Local endpoint 3. DUT escalates the call to conference by adding external endpoint 4. DUT hangs up and then local end point hangs up	• Local endpoint starts ringing • Verify audio connected successfully • Verify external endpoint is added to the conference successfully • Verify the call is disconnected successfully

Test Results: Comments	P	F	N/A	N/S	N/T	B
DUT doesn't support conference call.				X		

Test Case #	EP-65	Category	Basic call features using Expressway	RFC_Standard	

Objective	Verify DUT is able to forward an incoming call to external endpoint registered via Expressway							
Pre-Test Conditions								
• DUT is registered as internal user on CUCM • Cisco Phone/client registered as local endpoint on CUCM • Cisco Phone/client registered as external endpoint via Expressway-E • DUT is set for Call forward always to external endpoint								
Test Procedure		**Expected Results**						
1. Initiate call from local endpoint to DUT 2. Answer the call in external endpoint 3. Local endpoint hangs up		• External endpoint starts ringing • Verify audio connected successfully between local endpoint and external endpoint • Verify the call is disconnected successfully						
Test Results: Comments			**P**	**F**	**N/A**	**N/S**	**N/T**	**B**
			X					

9.8 SRST Failover Cases

Test Case #	EP-66	**Category**	SRST Failover Test: DUT Registers with SRST		**RFC_Standard**			
Objective	Verify DUT is able to register to SRST when CUCM call services fail							
Pre-Test Conditions								
• Local CUCM➔DUT(s):7100; • DUT is registered as internal user on CUCM • CUCM SRST configurations set and pointed towards SRST Go to System➔SRST➔Add New Name: SRST; IP Address: x.x.x.x; SIP Network/IP Address: x.x.x.x; Save Go to System➔Device Pool➔Default➔SRST Reference➔SRST; Save and Apply • SRST set with proper dial-peers and configurations								
Test Procedure			**Expected Results**					
1. Go to Cisco Unified Serviceability➔Tools➔Control Center – Feature Services➔cucmpub 2. Stop CallManager service			• Local Cisco Phone/client and DUT lose call services • Local Cisco Phone/client and DUT register with SRST, display that in Fallover Mode • Call services restored • DUT displays that it is registered with secondary proxy server					
Test Results: Comments			**P**	**F**	**N/A**	**N/S**	**N/T**	**B**
			X					

<table>
<tr><td>Test Case #</td><td>EP-67</td><td>Category</td><td>SRST Failover Test: Basic Call using SRST</td><td>RFC_Standard</td><td></td></tr>
<tr><td>Objective</td><td colspan="5">Verify DUT is able to make basic local calls when via SRST when CUCM call services are unavailable</td></tr>
<tr><td colspan="6">Pre-Test Conditions</td></tr>
<tr><td colspan="6">• SRST➔DUT(s):7100; SIP:2000
• DUT is registered with SRST
• Cisco Phone/client registered with SRST
• SRST configurations set and pointed towards CUCM
• SRST set with proper dial-peers and configurations</td></tr>
<tr><td colspan="3">Test Procedure</td><td colspan="3">Expected Results</td></tr>
<tr><td colspan="3">1. 2000 dials 7100➔7100 answers
2. 7100 on-hook after 30s
3. 7100 dials 2000➔2000 answers
4. 2000 on-hook after 30s</td><td colspan="3">• Call established between 2000 & 7100 with 2-way audio
• 7100 notified of incoming call (tone /display)
• 7100 answers incoming call
• Call established between 7100 & 2000 with 2-way audio
• 2000 notified of incoming call (tone/display)
• 2000 answers incoming call
• All calls routed through SRST</td></tr>
</table>

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

<table>
<tr><td>Test Case #</td><td>EP-68</td><td>Category</td><td>SRST Failover Test: Outbound Call to PSTN</td><td>RFC_Standard</td><td></td></tr>
<tr><td>Objective</td><td colspan="5">Verify DUT is able to make outbound calls to PSTN via SRST when CUCM call services are unavailable</td></tr>
<tr><td colspan="6">Pre-Test Conditions</td></tr>
<tr><td colspan="6">• SRST➔DUT(s):7100;
• DUT is registered with SRST
• SRST configurations set and pointed towards CUCM
• SRST set with proper dial-peers and configurations</td></tr>
<tr><td colspan="3">Test Procedure</td><td colspan="3">Expected Results</td></tr>
<tr><td colspan="3">1. Initiate call from DUT➔PSTN answers
2. DUT on-hook after 30s</td><td colspan="3">• Call establish between 7100 & PSTN with 2-way audio
• PSTN shows Caller ID of DUT
• All calls routed through SRST
• Verify audio connected successfully between local endpoint and external endpoint
• Verify the call is disconnected successfully</td></tr>
</table>

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

Test Case #	EP-69	Category	SRST Failover Test: Inbound Call from PSTN	RFC_Standard	
Objective	Verify DUT is able to receive inbound calls from PSTN via SRST when CUCM call services are unavailable				
Pre-Test Conditions					
• SRST➔DUT(s):7100; • DUT is registered with SRST • SRST configurations set and pointed towards CUCM • SRST set with proper dial-peers and configurations					
Test Procedure			**Expected Results**		
1. Initiate call from PSTN➔7100 2. PSTN on-hook after 30s			• Call established between PSTN & 7100 with 2-way audio • All calls routed through SRST • Call Verify audio connected successfully between local endpoint and external endpoint • Verify the call is disconnected successfully		

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

Test Case #	EP-70	Category	SRST Failover Test: DUT re-registers with CUCM	RFC_Standard	
Objective	Verify DUT is able to re-register with CUCM when call services are restored				
Pre-Test Conditions					
• SRST➔DUT(s):7100; • DUT is registered with SRST • SRST configurations set and pointed towards CUCM • SRST set with proper dial-peers and configurations					
Test Procedure			**Expected Results**		
1. Go to Cisco Unified Serviceability➔Tools➔Control Center – Feature Services➔cucmpub 2. Start CallManager service			• Local Cisco Phone/client and DUT re-register with CUCM • Call services restored • DUT displays that it is re-registered with primary proxy server		

Test Results: Comments	P	F	N/A	N/S	N/T	B
	X					

10 APPENDIX A: TEST RESULT MATRIX

Test Case #	P	F	NA	NS	NT	B	Comments
EP-1	X						
EP-2	X						
EP-3	X						
EP-4	X						
EP-5				X			
EP-6				X			
EP-7	X						
EP-8	X						
EP-9	X						
EP-10	X						
EP-11	X						
EP-12	X						
EP-13	X						
EP-14				X			
EP-15				X			
EP-16				X			
EP-17	X						
EP-18				X			
EP-19				X			
EP-20				X			
EP-21				X			
EP-22				X			
EP-23				X			
EP-24				X			
EP-25	X						
EP-26	X						
EP-27				X			
EP-28			X				
EP-29	X						
EP-30	X						
EP-31	X						
EP-32	X						
EP-33				X			
EP-34				X			
EP-35				X			
EP-36				X			
EP-37				X			
EP-38				X			
EP-39				X			
EP-40			X				
EP-41			X				

Test Case #	P	F	NA	NS	NT	B	Comments
EP-42			X				
EP-43			X				
EP-44			X				
EP-45			X				
EP-46	X						
EP-47	X						
EP-48	X						
EP-49	X						
EP-50	X						
EP-51	X						
EP-52			X				
EP-53	X						
EP-54	X						
EP-55	X						
EP-56			X				
EP-57	X						
EP-58	X						
EP-59	X						
EP-60	X						
EP-61	X						
EP-62	X						
EP-63	X						
EP-64				X			
EP-65	X						
EP-66							
EP-67							
EP-68							
EP-69							
EP-70							

===============**END OF DOCUMENT**==========

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