

Detailed End Point IVT Test Plan and Report for **ASCOM i62** with **CUCM 12.0**



Test Date/ Result (Completed by Cisco or Authorized Test House)	18/01/18 - PASS
Partner Product Name	Ascom i62
Partner Product Type	Wireless SIP Phone
Partner Product Version #	6.0.1
Cisco Product Name	Cisco Unified Communications Manager
Cisco Product Version	CUCM 12.0
API/Protocol(s) Used	SIP
Date Testing Completed	December 21, 2017
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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 **<PARTNER COMPLETED>**

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 appliEPIe 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](#)

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	No test tools used for this test.
Can these tools and test scripts for these products be made available to support IVT	N/A
Are there proprietary test tools that could be made available to support IVT	No proprietary tools used for this test.

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	Appliance		
Disk	Appliance		
Memory	Appliance		
Max Users supported			

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**.

CUCM Configurations

- CUCM license for “Third-party SIP device (Basic)” implies some limitations, e.g. no Music-on-Hold (MoH) and lack of telephony features configurable from the handset etc.
- Add DUT device using “Third-party SIP device (Basic)” and assign user.
 - Handsets require unique and fictitious MAC addresses.
- Add End-User with user ID and password in CUCM.
 - Assign End-User to device configuration (digest user).

Ascom Configuration (Not Shown)

Refer to Ascom documentation for configuration details.

- Configure CUCM pub and/or subscriber nodes.
- Add device in UCM.
- Assign user and MAC address configured in CUCM.

Ascom i62 device Configured in CUCM

- Adding device and assign user to Ascom device. (Device-> Phone)
 - The Phone Type selected is Third-Party SIP device(Basic),
 - The MAC address used is unique and fictitious.
 - Digest user (created in End-User Configuration) is assign to user.

Association

Modify Button Items

1 Line [1] - 2601 (no partition)

Unassigned Associated Items

2 Line [2] - Add a new DN

Phone Type

Product Type: **Third-party SIP Device (Basic)**

Device Protocol: SIP

Real-time Device Status

Registration: Unknown

IPv4 Address: None

Device Information

Device is Active

Device is not trusted

MAC Address* 00013E11B460

Description SEP00013E11B460

Device Pool* Default [View Details](#)

Common Device Configuration < None > [View Details](#)

Phone Button Template* **Third-party SIP Device (Basic)**

Common Phone Profile* Standard Common Phone Profile [View Details](#)

Calling Search Space < None >

AAR Calling Search Space < None >

Media Resource Group List < None >

Location* Hub_None

AAR Group < None >

Device Mobility Mode* Default [View Current Device Mobility Settings](#)

Owner User Anonymous (Public/Shared Space)

Owner User ID* 2601

Mobility User ID < None >

Use Trusted Relay Point* Default

Always Use Prime Line* Default

Always Use Prime Line for Voice Message* Default

Geolocation < None >

Ignore Presentation Indicators (internal calls only)

Logged Into Hunt Group

Remote Device

Figure 1: Add DUT to CUCM Phone Configuration

Number Presentation Transformation	
Caller ID For Calls From This Phone	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)	
Remote Number	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)	
Protocol Specific Information	
BLF Presence Group *	Standard Presence group
MTP Preferred Originating Codec*	711ulaw
Device Security Profile *	Third-party SIP Device Basic - Standard SIP Non-S
Rerouting Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile View Details
Digest User	2601
<input type="checkbox"/> Media Termination Point Required	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> Require DTMF Reception	
MLPP and Confidential Access Level Information	
MLPP Domain	< None >
Confidential Access Mode	< None >
Confidential Access Level	< None >

Figure 2: Add DUT to CUCM Phone Configuration (Continue)

2. Add End User Configuration (User Management -> End User):

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | Go

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

End User Configuration | Related Links: Back to Find List Users | Go

Save | Delete | Add New

User Information

User Status: Enabled Local User

User ID*: 2601

Password: [Redacted] [Edit Credential](#)

Confirm Password: [Redacted]

Self-Service User ID: 2601

PIN: [Redacted] [Edit Credential](#)

Confirm PIN: [Redacted]

Last name*: Ascom

Middle name:

First name:

Display name:

Title:

Directory URI:

Telephone Number:

Home Number:

Mobile Number:

Pager Number:

Mail ID:

Manager User ID:

Department:

User Locale: < None >

Associated PC/Site Code:

Digest Credentials: [Redacted]

Confirm Digest Credentials: [Redacted]

User Profile: Use System Default("Standard (Factory Default) U: [View Details](#)

User Rank*: 1-Default User Rank

Service Settings

Home Cluster

Enable User for Unified CM IM and Presence (Configure IM and Presence in the associated UC Service Profile)

Include meeting information in presence(Requires Exchange Presence Gateway to be configured on CUCM IM and Presence server)

UC Service Profile: Use System Default("CUPS") [View Details](#)

Device Information

Controlled Devices: SEP00013E11B460

Available Profiles: 7971 Device profile, Cisco 7971 SIP, EM2017, EM3017, EM_1054

CTI Controlled Device Profiles:

Device Association
Line Appearance Association for Presence

Figure 3: Add End User for DUT

Extension Mobility

Available Profiles: 7971 Device profile, Cisco 7971 SIP, EM2017, EM3017, EM_1054

Controlled Profiles: [Empty list]

Default Profile: -- Not Selected --

BLF Presence Group*: Standard Presence group

SUBSCRIBE Calling Search Space: < None >

Allow Control of Device from CTI
 Enable Extension Mobility Cross Cluster

Directory Number Associations

Primary Extension: < None >

Mobility Information

Enable Mobility
 Enable Mobile Voice Access

Maximum Wait Time for Desk Pickup*: 10000

Remote Destination Limit*: 4

Remote Destination Profiles: [Empty list] [View Details](#)

Multilevel Precedence and Preemption Authorization

MLPP User Identification Number: [Empty field]

MLPP Password: [Empty field]

Confirm MLPP Password: [Empty field]

MLPP Precedence Authorization Level: Default

CAPF Information

Associated CAPF Profiles: [Empty list] [View Details](#)

Permissions Information

Groups: Standard CCM End Users, Standard CCM Phone Administration, Standard CCM Read Only, Standard CTI Allow Call Monitoring, Standard CTI Allow Call Park Monitoring [View Details](#)

Roles: Standard CCM Admin Users, Standard CCM End Users, Standard CCM Phone Management, Standard CCMADMIN Read Only, Standard CCMUSER Administration [View Details](#)

Conference Now Information

Enable End User to Host Conference Now

Meeting Number: 2601

Attendees Access Code: [Empty field]

Figure 4: Add End User for DUT (Continue)

5.2 Product Description

Ascom i62

The Ascom i62 is a VoWiFi handset that offers enterprise-grade telephony, messaging and personal alarm capabilities. It is based upon modern Wi-Fi technology, with support for 802.11a/b/g/n standards. The Ascom i62 is an extremely durable, feature rich, and is designed specifically to support mission critical communication where reliability is a must. Supported by Ascom's comprehensive interoperability program, the Ascom i62 integrates seamlessly with CUCM, as well as infrastructure from leading vendors.

Features

The Ascom i62 is an extremely durable, feature rich, and is designed specifically to support mission critical communication where reliability is a must. Supported by Ascom's comprehensive interoperability program, the Ascom i62 integrates seamlessly with CUCM, as well as infrastructure from leading vendors.

- High quality telephony
- Interactive messaging
- Personal alarm with a dedicated button and man-down alarm
- Push-To-Talk
- Location with WiFi access points and Ekahau
- Centralized Management
- WiFi standards 802.11 a/b/g/n, to operate in the existing WiFi infrastructure
- Integrates to CUCM via SIP, to provide a feature-rich telephony solution

5.3 Product Integration Diagram

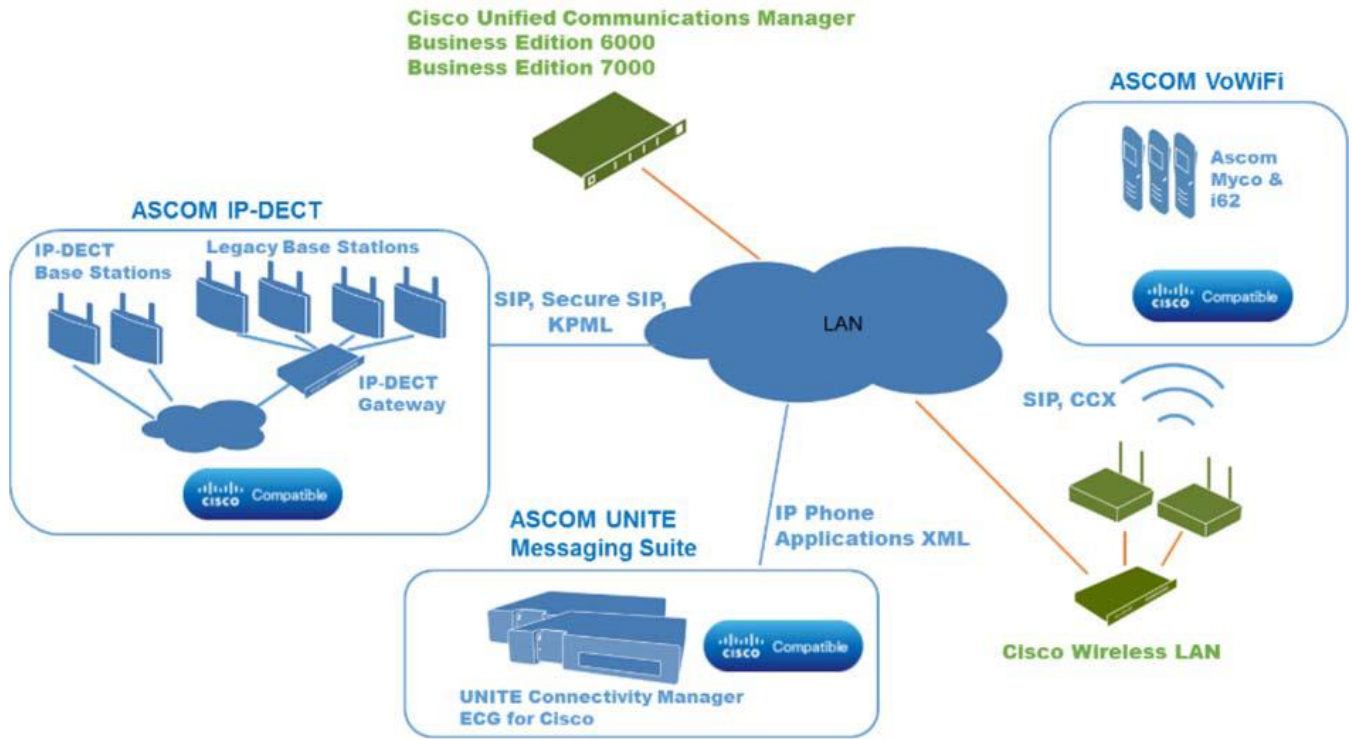


Figure 5: Ascom Integration Diagram

5.4 Product Integrated Use Cases

Ascom i62 device handsets registered and used for telephony services with CUCM Release 12.0.

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:

1. Speed dial,
2. Call Park,
3. Direct Call Transfer,
4. Shared Lines,
5. Meet-Me,
6. Call Back,
7. Barge/cBarge,
8. Authentication,
9. Join Across Lines,
10. Group Pickup,
11. DND,
12. iDivert,
13. Hotline

- Test Cases that are Not Applicable:

1. EP-27 Functional Test: Video Endpoints,
2. EP-28 Functional Test: Extension Mobility,
3. EP-40 Functional Test: Malicious Call,
4. EP-41-43 Functional Test: Mobile Connect,
5. EP-44 Functional Test: Mobile Voice Access (MVA),
6. EP-45 Functional Test: Enterprise Feature Access (EFA),
7. EP-52 Miscellaneous Test: DUT display features,
8. EP-56 Functional Test: Multiple Lines

- Test Cases that were Not Executed: None

- None

- Observations:
 1. There is no conference display indication seen on the handset devices. This is only a GUI issue.
 2. MWI (message waiting Indication) was tested with the i62 device with Cisco Unity Connection. When a voice mail message was deposited, the device displayed MWI notification. After the message was retrieved, MWI notification was removed.
 3. Ascom i62 devices support basic feature functionality. Feature softkeys for many of the advanced features are not implemented.

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					
None					
3rd Party Tools					
WireShark	2.2.5	application	Tool for packet capture of call traces.	1	
Debug Tools					
None					

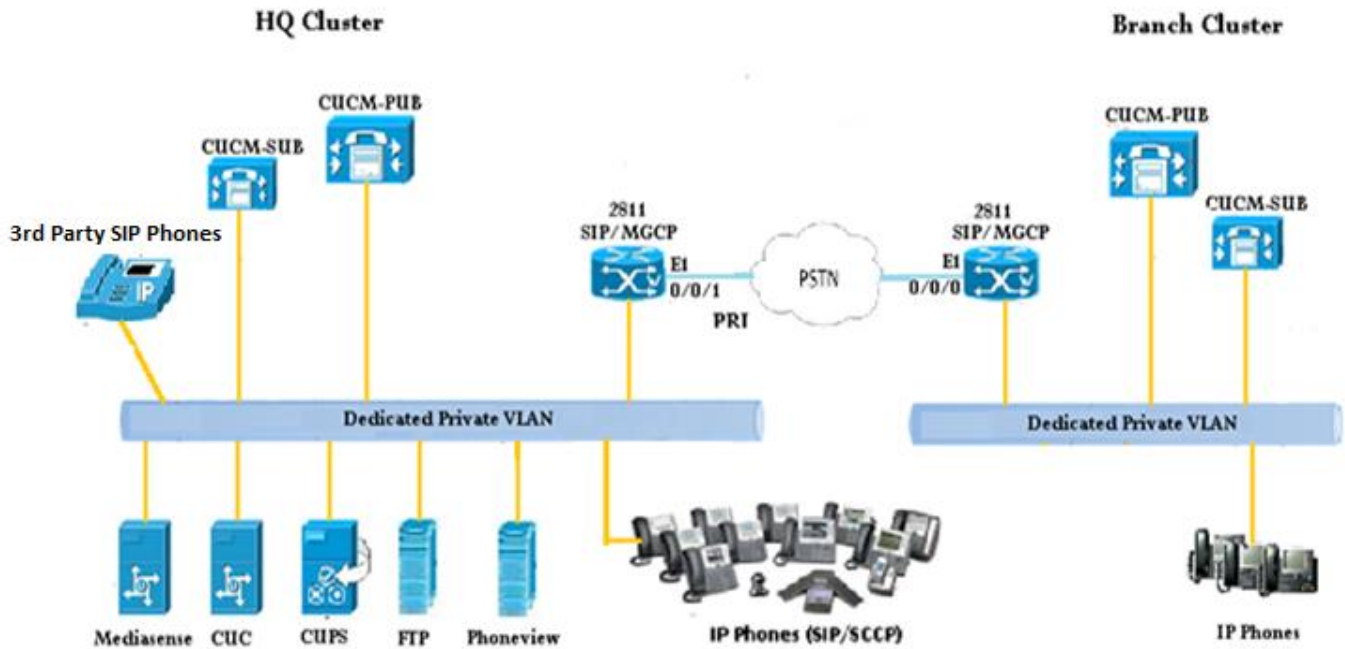
8.5 Equipment Requirements

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

Table 3. Test Lab Topology Components

Product	Version	Units	Description
CUCM	12.0	2 PUB & 2 SUB	HQ & Branch CUCM Clusters
CUPS	12.0	1	Cisco Presence Server
CUC	11.5	1	Cisco Unity Connection
Cisco 3845	12.4	1	PSTN Gateway
Jabber	11.8.2 Build 50390	1	Jabber client
IP Communicator	8.6.6.0	1	IP Communicator client
IP Phones	sip41.9-4-2SR3-1S	1	7961
	sccp41.9-4-2SR3-1S	1	7961
	sip9951.9-4-2SR2-2	1	9951
	sccp42.9-4-2SR3-1S	1	7962
	sip894x.9-4-2SR3-1	1	8945
	sipdx650.10-1-1-78	1	DX650
DUT(s)	6.0.1	3	Ascom i62 devices
Device Manager (UCM)	5.8.1	1	Ascom Device Manager

8.6 Lab Network Topology



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

**Run “Step 1_Endpoint_Registration” command to register endpoints.
Run “Step 2_Record_Connectivity_Validation” command after verifying the endpoints are registered with the correct network and load information.**

9.2 Integration Test


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

Test Case #	EP-1	Category	Connect→Validate	RFC_Standard	Y				
Objective	Verify 3 rd party endpoints (DUT) are registered in Call Manager successfully								
Pre-Test Conditions									
<ul style="list-style-type: none"> • 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 									
Test Procedure			Expected Results						
<ol style="list-style-type: none"> 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 <ul style="list-style-type: none"> > DUT:7100→dutuser01 > DUT:7101→dutuser02 > DUT:8000→rdutuser01 <p>Note:</p> <ul style="list-style-type: none"> • 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 			<ul style="list-style-type: none"> • 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 						
Test Results: Comments				P	F	N/A	N/S	N/T	B
				X					

Tests in this section requires RPC tool to remotely control Cisco IP Phones for manual calls. Follow instructions highlighted in green in each test case before execution. It is important to restore endpoints to it’s original state in some test cases before proceeding to next test case. Run “Step 3: Record_Functional_Test_Execution” command after executing all tests in this section
Retrieve CDR(s) in CUCM to validate calls

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										
<ul style="list-style-type: none"> 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) <ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 7100 dials 7101→7101 answers→7100 on-hook after 30s 7101 dials 1000→Select 1000 and answer call using RPC Select "Headphone" icon Enter "Play:AreYouThere.raw" & hit "Send" on the Command Line 7101 speaks "Testing1234"→1000 on-hook after 60s Repeat steps 2-5 with Calling DN:2000 & Called DN;7100 Repeat steps 2-5 with Calling DN:7101 & Called DN;2000 Calling & Called party release calls alternatively Retrieve CDR from CUCM Check Calling, Called, Duration, Origination & Termination Cause Codes 			<ul style="list-style-type: none"> 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

Test Case #	EP-3	Category	Functional Test: Inter-Cluster Call	RFC_Standard	Y				
Objective	Verify inter-cluster calls between DUT(s), SCCP and SIP endpoints								
Pre-Test Conditions									
<ul style="list-style-type: none"> Local CUCM → DUT(s): 7100 & 7101; Remote CUCM → DUT: 8000; SCCP: 5200; SIP: 6200; RPC is used to remotely control IP Phones: 5200, 6200; 									
Test Procedure			Expected Results						
<ol style="list-style-type: none"> 7100 dials 2348000 → 8000 answers → 8000 on-hook after 30s 7101 dials 2345200 → Select 5200 and answer call using RPC Select "Headphone" icon Enter "Play:AreYouThere.raw" and hit "Send" on the Command Line 7101 speaks "Testing1234" → 5200 on-hook after 60s Repeat steps 2-5 with Calling DN: 7101 & Called DN: 6200 Calling & Called party release calls alternatively Retrieve CDR from CUCM Check Calling, Called, Duration, Origination & Termination Cause Codes 			<ul style="list-style-type: none"> 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 						
Test Results: Comments				P	F	N/A	N/S	N/T	B
				X					

Test Case #	EP-4	Category	Functional Test: Off-Net Calls	RFC_Standard	Y				
Objective	Verify basic calls between DUT(s) and PSTN endpoints								
Pre-Test Conditions									
<ul style="list-style-type: none"> Local CUCM → DUT(s): 7100 & 7101; Remote CUCM → DUT: 8000; PSTN: 210-222-5400 (SIP); RPC is used to remotely control PSTN Phone: 2102225401; 									
Test Procedure			Expected Results						
<ol style="list-style-type: none"> 7100 dials 92102225400 → Select 2102225400 & answer using RPC Select "Headphone" icon Enter "Play:AreYouThere.raw" and hit "Send" on the Command Line 7100 speaks "Testing1234" → 2102225400 on-hook after 60s Repeat steps 1-4 with Calling DN: 2102225400 & Called DN: 94104447101 Retrieve CDR from CUCM Server Check Calling, Called, Duration, Origination & Termination Cause Codes <p>Note: Inbound & Outbound PSTN calls are dialed with a prefix of "9"</p>			<ul style="list-style-type: none"> 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 						
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
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Objective	Verify intra-cluster SIP URI calls between DUT and SIP endpoints								
Pre-Test Conditions									
<ul style="list-style-type: none"> 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: <ul style="list-style-type: none"> 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; <p>Note: Provision URI on device page: Device→Phone→DN→Line→URI (Known bug if provisioned via End User Page)</p>									
Test Procedure				Expected Results					
<ol style="list-style-type: none"> 7100 hits Speed Dial button 3→7101 answers 7101 goes on-hook after 30s 2000 hits Speed Dial button 3 →7100 answers 2000 goes on-hook after 30s 7101 hits Speed Dial button 3→2000 answers 7101 goes on-hook after 30s Retrieve CDR from CUCM Server Check Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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
Speed Dial is not supported by i62							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								
<ul style="list-style-type: none"> 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: <ul style="list-style-type: none"> 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; <p>Note: Provision URI on device page: Device→Phone→DN→Line→URI (Known bug if provisioned via End User Page)</p>								
Test Procedure			Expected Results					
<ol style="list-style-type: none"> 7100 hits Speed Dial button 3→8000 answers 8000 goes on-hook after 30s 7101 hits Speed Dial button 3→6200 answers 6200 goes on-hook after 30s 6200 hits Speed Dial button 3→7100 answers 7100 goes on-hook after 30s Retrieve CDR from CUCM Server Check Calling, Called, Duration, Origination & Termination Cause Codes 			<ul style="list-style-type: none"> 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
Speed Dial is not supported by i62						X		

Test Case #	EP-7	Category	Functional Test: CFA	RFC_Standard	Y
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Objective		Verify "CFA" calls between DUT(s), SCCP, SIP and PSTN endpoints									
Pre-Test Conditions											
<ul style="list-style-type: none"> 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): <ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 7100 dials 7101→8000 answers→7100 on-hook after 30s 8000 dials 4447100→1000 answers→1000 on-hook after 30s 7101 dials 2348000→6200 answers→7101 on-hook after 30s 7101 dials 2000→8000 answers 7100 dials 2345200→7100 on-hook - hears busy tone 7101 goes on-hook 7101 dials 92102225400→7100 answers 2102225400 goes on-hook after 30s 2102225400 dials 94104447101→8000 answers 8000 goes on-hook after 30 secs Retrieve CDR from CUCM Check Calling, Called, Duration, Origination & Termination Cause Codes <p>Note: Upon test completion, remove "CFA" feature for devices highlighted in yellow before proceeding to next test case</p> <p>CUCM Administration GUI: Device→Phone→DN→Line→Call Forward All →Destination→blank</p>				<ul style="list-style-type: none"> "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
In step 5, 162 phone call rang into second call appearance. Call was answered and ended.						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									
<ul style="list-style-type: none"> 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): <ul style="list-style-type: none"> 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						
<ol style="list-style-type: none"> 7100 dials 7101→7101 does not answer→8000 answers 7100 goes on-hook after 30s 8000 dials 4447100→7100 does not answer→1000 answers 1000 goes on-hook after 30s 7101 dials 2348000→8000 does not answer→6200 answers 7101 goes on-hook after 30s 7101 dials 2000→2000 does not answer→8000 does not answer 6200 answers call 8000 dials 5200→5200 does not answer→7101 does not answer 8000 goes on-hook - hears busy tone 6200 goes on-hook Retrieve CDR from CUCM Check Calling, Called, Duration, Origination & Termination Cause Codes <p>Note: Upon test completion, remove "CFNA" feature for devices highlighted in yellow before proceeding to next test case</p> <p>CUCM Administration GUI: Device→Phone→DN→Line→CFNA→Destination→blank</p>			<ul style="list-style-type: none"> 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											
<ul style="list-style-type: none"> 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 Enable CFB for DN(s): <ul style="list-style-type: none"> ➤ 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							
<ol style="list-style-type: none"> 1000 dials 7101→7101 answers 7100 dials 7101→8000 answers→7100 on-hook after 30s 5200 dials 8000→8000 answers 7100 dials 7101→7100 on-hook – hears busy tone 5200 goes on-hook 7100 dials 1000→8000 answers→7100 on-hook after 30s 7101 goes on-hook 2000 dials 1000→1000 answers 5200 dials 8000→8000 answers 7101 dials 1000→7101 on-hook – hears busy tone 1000 goes on-hook 1000 dials 6200→6200 answers 7100 dials 2346200→7101 answers→7101 on-hook after 30s 2000 dials 7101→7101 answers 7100 dials 2346200→7100 goes on-hook –hears busy tone 2000 & 6200 goes on-hook Retrieve CDR from CUCM Check Calling, Called, Duration, origination & termination cause codes matches the calls <p>Note: Upon test completion, remove "CFB" feature for devices highlighted in yellow before proceeding to next test case</p> <p>CUCM Administration GUI: Device→Phone→DN→Line→CFB→Destination→blank</p>				<ul style="list-style-type: none"> 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											
<ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 7100 dials 7101→7101 answers→ 7100 hits "Hold" after 20s 7100 hits "Resume" after 20s→7100 on-hook after 30s 7100 dials 234-8000→8000 answers 5200 dials 444-7100→7100 answers incoming call→8000 on-hold 7100 hits "Resume" after 60s→7100 on-hook after 30s 7100 dials 7101→7101 answers→7100 hits "Hold" after 30s 7100 dials 234-5200→5200 answers→5200 on-hook after 30s 7101 goes on-hook after 30s 7101 dials 7100→7100 answers→7101 hits "Hold" after 30s 7100 goes on-hook 10s later while call is on-hold 7101 goes-hook Repeat steps 1-2 for SCCP. Replace 7101 with 1000 Repeat steps 1-2 for SIP. Replace 7100 with 2000 Repeat steps 1-2 for PSTN. Replace 7101 with 92102225400 Retrieve CDR from CUCM Check Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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											
<ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 7100 dials 7101→7101 answers 8000 dials 444-7100→7100 answers incoming call 8000 goes on-hook after 30s 1000 dials 7101→7101 answers incoming call 1000 goes on-hook after 30s 7100 goes on-hook after 60s 7100 dials 1000→7100 answers 2000 dials 7100→7100 answers incoming call 2000 goes on-hook after 30s 7100 goes on-hook after 60s 6200 dials 444-7101→7101 answers 5200 dials 444-7101→7101 answers incoming call 5200 goes on-hook after 60s 6200 goes on-hook after 30s Retrieve CDR from CUCM Check Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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
I62 hear tone (internal)						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											
<ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 7100 dials 7101→7101 answers→7101 hits "Transfer" after 30s 7101 dials 234-8000→ 7101 hits "Transfer"→7101 is on-hook 8000 goes on-hook after 60s 7100 dials 7101→7101 answers→7100 hits "Transfer" after 30s 7100 dials 234-7777→7100 hits "Transfer"→7100 is on-hook 7101 goes on-hook – hears reorder tone 7100 dials 1000→1000 answers→7100 hits "Transfer" after 30s 7100 dials 7101→7100 hits "Transfer"→7100 is on-hook 1000 goes on-hook after 60s 7100 dials 2000→2000 answers→7100 hits "Transfer" after 30s 7100 dials 1000→7100 hits "Transfer"→7100 is on-hook 1000 goes on-hook after 20s 7101 dials 234-8000→8000 answers 7100 dials 2000→2000 answers→2000 hits "Transfer" after 30s 2000 dials 234-8000→2000 hits "Transfer"→2000 is on-hook 7100 goes on-hook - hears busy tone .Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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
						X					

Test Case #	EP-13	Category	Functional Test: Consult Transfer				RFC_Standard	Y			
Objective	Verify "Consult Transfer" calls between DUT(s), SIP, SCCP and PSTN endpoints										
Pre-Test Conditions											
<ul style="list-style-type: none"> 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; 											
Test Procedure				Expected Results							
<p>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</p> <p>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</p> <p>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</p> <p>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</p>				<ul style="list-style-type: none"> 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 							
Test Results: Comments						P	F	N/A	N/S	N/T	B
						X					

Test Case #	EP-14	Category	Functional Test: Conference Call	RFC_Standard	Y				
Objective	Verify Conference call between DUT(s), SIP, SCCP and PSTN endpoints								
Pre-Test Conditions									
<ul style="list-style-type: none"> 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; 									
Test Procedure			Expected Results						
<p>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</p> <p>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</p> <p>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</p> <p>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</p>			<ul style="list-style-type: none"> 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 						
Test Results: Comments				P	F	N/A	N/S	N/T	B
No MOH heard when i62 devices put calls on hold during conference. During last test call (call4), DUT device cancels call2 to 1000 to resume with call1.				X					

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											
<ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 7100 dials 2348000 → 8000 answers 7100 hits "Park" softkey after 10s 7101 dials park code: 3001 after 20s 8000 goes on-hook after 30s 7100 dials 1000 → 1000 answers 1000 hits "Park" softkey after 10s 7101 dials park code: 3001 after 20s 7100 goes on-hook after 30s 2000 dials 7101 → 7101 answers 7101 hits "Park" softkey after 10s 7100 dials park code: 3001 after 20s 2000 goes on-hook after 30s 2102225400 dials 7100 → 7100 answers 7100 hits the "Park" softkey after 10s 7100 dials park code: 3001 after 20s 2102225400 goes on-hook Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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 in i62 device.									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									
<ul style="list-style-type: none"> 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						
<ol style="list-style-type: none"> 7100 dials 234-8000→8000 answers 7100 hits "Park" softkey after 10s Do not pickup parked call for 60s 7100 is ringing→7100 answers 8000 goes on-hook after 30s 7100 dials 1000→1000 answers 1000 hits "Park" softkey after 10s Do not pickup parked call for 60s 1000 is ringing→1000 answers 7100 goes on-hook after 30s 2000 dials 7101→7101 answers 7101 hits "Park" softkey after 10s Do not pickup parked call for 60s 7101 is ringing→7101 answers 2000 goes on-hook after 30s Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 			<ul style="list-style-type: none"> 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 in i62 device.							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											
<ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 8000 dials 4442000→2000 answers 2000 hits "BLF" button for Assisted Directed Call Park after 20s 2000 goes on-hook 7101 dials *3011 to retrieve call when the BLF is flashing 8000 goes on-hook after 30s 2003 dials 234-8000→8000 answers 2003 hits "BLF" button for Assisted Directed Call Park after 20s 2003 goes on-hook 1000 hits dials *3011 to retrieve call when BLF is flashing 1000 goes on-hook after 30s Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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
Directed Call Park is not implemented in i62									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											
<ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 7101 dials 7100→7100 answers 7100 selects shared line:DN:1901 after 30s 1901 dials 234-8000→8000 answers Scroll to 1st call and hit select Scroll to 2nd call and hit select and hit "DirTfr" 1901 goes on-hook 7101 goes on-hook after 30s 7100 dials 7101→7101 answers 7100 selects shared line:DN:1901 after 30s 1901 dials 1000→1000 answers Scroll to 1st call and hit select Scroll to 2nd call and hit select and hit "DirTfr" 1901 goes on-hook 1000 goes on-hook after 30s 8000 dials 444-7100→7100 answers 7100 selects shared line:DN:1901 after 30s 1901 dials 2000→2000 answers Scroll to 1st call and hit select Scroll to 2nd call and hit select and hit "DirTfr" 1901 goes on-hook 2000 goes on-hook 7100 dials 234-8000 7100 selects shared line:DN:1901 after 30s 1901 dials 92102225400→2102225400 answers Scroll to 1st call and hit select Scroll to 2nd call and hit select and hit "DirTfr" 1901 goes on-hook 8000 goes on-hook Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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
Shared line is not implemented in i62									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									
<ul style="list-style-type: none"> 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						
<ol style="list-style-type: none"> 7100 dials 92102225400→2102225400 answers 7100 hits "Conference" after 30s 7100 dials 7101→7101 answers→7100 hits "Conference" after 30s 2000 dials 7101→7101 answers 2nd incoming call 2000 hits "Conference" after 20s 2000 dials 234-8000→8000 answers→2000 hits "Conference" after 20s 2000 dials 1000→1000 answers→2000 hits "Conference" after 20s 7101 selects conference 1 and hits the "Join" softkey 7101 goes on-hook after 60s 8000 goes on-hook after 70s All other participants ended call after 120s Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 			<ul style="list-style-type: none"> 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
				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									
<ul style="list-style-type: none"> 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						
<ol style="list-style-type: none"> 7101 goes off hook & hits "Meet-Me" softkey and dials 5555 1000 dials 5555 2000 dials 5555 8000 dials 444-5555 All 4 members go on-hook after 120 secs Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 			<ul style="list-style-type: none"> 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 <p>Note: Change the Calling Search Space for 7101→ None after test is complete</p>						
Test Results: Comments				P	F	N/A	N/S	N/T	B
"Meet-me" softkey is not implemented in i62							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											
<ul style="list-style-type: none"> 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										
<ol style="list-style-type: none"> 7100 dials 7101 → 7101 answers 8000 dials 444-7101 → 8000 hits "Callback" softkey and exits 7101 goes on-hook after 60s 8000 redials 444-7101 after callback alert 7101 answers → 7101 goes on-hook after 60s 7100 dials 1000 → 1000 answers 7101 dials 1000 → 7101 hits "Callback" softkey and exits 7100 goes on-hook after 60s 7101 redials 1000 after callback alert 1000 answers → 1000 goes on-hook after 60s 2000 dials 7100 → 7100 answers 7101 dials 2000 → 7101 hits "Callback" softkey and exits 7100 goes on-hook after 60s 7101 redials 2000 after callback alert 2000 answers → 7101 goes on-hook after 60s 7101 dials 92102225400 → 2102225400 answers 7100 dials 92102225400 → 7100 hits "Callback" softkey and exits 2102225400 goes on-hook after 60s 7100 redials 92102225400 after callback alert Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 	<ul style="list-style-type: none"> 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) retrieved Selected fields in CDR(s) match calls 										
Test Results: Comments						P	F	N/A	N/S	N/T	B
"Callback" softkey is not implemented in i62.									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											
<ul style="list-style-type: none"> Local CUCM → DUT(s): 7100 & 7101; SCCP: 1000; SIP: 2000; Remote CUCM → DUT: 8000; PSTN: 210-222-5400 Cluster-wide Service Parameters: <ul style="list-style-type: none"> Built In Bridge Enable → On Party Entrance Tone → True Device → Phone → DN: <ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 8000 dials 444-1901 1901 answers (Shared line on 7100) 7101 hits line 1901 and selects barge after 20s 8000 goes on-hook after 60s 7100 dials 1901 1901 answers (Shared line on 1000) 7101 hits line 1901 and selects barge after 20s 1000 goes on-hook after 60s 8000 dials 4441901 1901 answers (Shared line on 2000) 7100 hits line 1901 and selects barge after 20s 7100 goes on-hook after 60s 8000 goes on-hook after 70s 2102225400 dials 94104441901 1901 answers (Shared line on 7100) 7101 hits line 1901 and selects barge after 20s 2102225400 goes on-hook after 60s Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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
Shared line is not implemented in i62									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											
<ul style="list-style-type: none"> 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: <ul style="list-style-type: none"> Built In Bridge Enable → On Party Entrance Tone → True Device → Phone → DN: <ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 8000 dials 444-1901 → 1901 answers (Shared line on 7100) 7101 selects line 1901 after 20s 1901 goes on-hook after 60s (Shared line on 7100) 8000 goes on-hook after 80s 2102225400 dials 94104441901 → 1901 answers (Shared line on 1000) 2000 selects line 1901 after 20s 7100 selects line 1901 Select 1901 after 30s 1901 goes on-hook after 60s (Shared line on 7100) Remaining 3 parties go-hook after 120s Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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
Shared line is not implemented in i62									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									
<ul style="list-style-type: none"> 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						
<ol style="list-style-type: none"> 7100 dials 1901→1901 answers (Shared Line on 7101) 1901 hits “Hold” softkey after 30s 1901 hits “Resume” softkey after 30s 1901 goes on-hook after 80s 7100 dials 1901→1901 answers (Shared Line on 1000) 1901 hits “Hold” softkey after 30s 1901 hits “Resume” softkey after 30s 7100 goes on-hook after 80s 7100 dials 1901→1901 answers (Shared Line on 2000) 1901 hits “Hold” softkey after 30s 1901 hits “Resume” softkey after 30s 1901 goes on-hook after 80s 2001 dials 1901→1901 answers (Shared Line on 7101) 1901 hits “Hold” softkey after 30s 1901 hits “Resume” softkey after 30s 1901 goes on-hook after 80s Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 			<ul style="list-style-type: none"> Call establish between 7100 & 1901 with 2-way audio 7100 is On-Hold (tone or silence) 7100 & 1901 resume call Call terminate normally Call establish between 7100 & 1901 with 2-way audio 7100 is On-Hold (tone or silence) 7100 & 1901 resume call Call terminate normally Call establish between 7100 & 1901 with 2-way audio 7100 is On-Hold (tone or silence) 7100 & 1901 resume call Call terminate normally Call establish between 2001 & 1901 with 2-way audio 2001 is On-Hold (tone or silence) 2001 & 1901 resume call Call 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
Shared line is not supported by i62							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									
<ul style="list-style-type: none"> 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						
<ol style="list-style-type: none"> 7100 dials 1922 (Duration=30s) 1922 dials 7101 (Duration=30s) 8000 dials 444-1922 (Duration=30s) Calling and Called party goes on-hook alternatively Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 			<ul style="list-style-type: none"> 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					
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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									
<ul style="list-style-type: none"> 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: <ul style="list-style-type: none"> 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						
<ol style="list-style-type: none"> 7100 dials 1940 (Duration=30s) 1940 dials 7101 (Duration=30s) 8000 dials 444-1940 (Duration=30s) 1940 dials 234-8000 (Duration=30s) Calling and Called party goes on-hook alternatively Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 			<ul style="list-style-type: none"> 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									
<ul style="list-style-type: none"> Local CUCM→DUT(s): 7100 & 7101; Remote CUCM→DUT DN:8000; Video capable phone DN: 2003 									
Test Procedure			Expected Results						
<ol style="list-style-type: none"> 7100 dials 2003 (Duration=30s) 2003 dials 7101 (Duration=30s) 8000 dials 444-2005 (Duration=30s) Calling and Called party goes on-hook alternatively Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 			<ul style="list-style-type: none"> 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

Video call is not implemented in i62			X			
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Test Case #	EP-28	Category	Functional Test: Extension Mobility				RFC_Standard	N			
Objective	Verify DUT(s) supports "Extension Mobility" call										
Pre-Test Conditions											
<ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 7100 hits "Services" button and selects EM service 7100 logs in with "emuser01/123456" 1934 dials 1000→1000 answers →1934 on-hook after 30s 2000 dials 1934→1934 answers→2000 on-hook after 30s 7101 dials 1934→1934 answer→7101 on-hook after 30s 1934 dials 234-5200→5200 answers→1934 on-hook after 30s 6200 dials 444-1934→1934 answers→6200 on-hook after 30s 1934 dials 92102225400→2102225400 answers 2102225400 goes on-hook after 30s 1934 hits "Services" button and selects EM service 1934 logs out Retrieve CDR from CUCM Check Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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 supported in i62								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											
<ul style="list-style-type: none"> 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 (1st member-7101; 2nd member-1000; 3rd 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							
<ol style="list-style-type: none"> 7100 dials 3000→7101 answers→7100 on-hook after 60s 7101 dials 2000→2000 answers 7100 dials 3000→1000 answers→1000 on-hook after 60s 2000 goes on hook after 70s 7101 dials 1000→1000 answers 7100 dials 3000→2000 answers→7100 on-hook after 60s 2122225400 dials 4144443000→2000 answers 2000 goes on-hook after 60s 1000 goes on-hook Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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											
<ul style="list-style-type: none"> Local CUCM→DUT(s):7100 & 7101; SSCP:1000; SIP:2000; Remote CUCM →DUT:8000; SSCP 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							
<ol style="list-style-type: none"> 7101 stays off-hook to make it unavailable 7100 dials 3010→8000 answers→ 8000 on-hook after 60s 7100 dials 3012→1000 answers→ 1000 on-hook after 60s 7100 dials 3013→2000 answers→ 7100 on-hook after 60s 7100 dials 3014→2102225400 answers 2102225400 goes on-hook after 60s 7101 goes on-hook Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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											
<ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 7100 dials 3015→2000 answers 7101 dials 3015 8000 dials 444-3015 7100 goes on-hook after 200 secs Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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											
<ul style="list-style-type: none"> 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: <ul style="list-style-type: none"> > 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							
<ol style="list-style-type: none"> 7100 dials 7101→7101 answers→7100 on-hook after 30s 7100 dials 234-8000→8000 answers→8000 on-hook after 30s 7100 dials 1001→1001 answers→1001 on-hook after 30s 2001 dials 7101→7101 answers→7101 on-hook after 30s 1000 dials 7100→7100 answers→7100 on-hook after 30s 7101 dials 2000→2000 answers→2000 on-hook after 30s 7100 dials 2102225400→2102225400 answers 7100 goes on-hook after 30s Retrieve CDR from CUCM Check the Calling, Called, Duration, Secured Status, Origination & termination Cause Codes 				<ul style="list-style-type: none"> 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 <p>Note: callSecuredstatus in CDR = 1 callSecuredstatus in CDR = 0 for unsecured calls</p>							
Test Results: Comments						P	F	N/A	N/S	N/T	B
Authenticated Security Profiles not supported by i62									X		

Test Case #	EP-33	Category	Functional Test: Join Across Line				RFC_Standard	N			
Objective	Verify "Join Across Lines" calls between DUT(s), SCCP, SIP and PSTN endpoints										
Pre-Test Conditions											
<ul style="list-style-type: none"> 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; 											
Test Procedure				Expected Results							
<ol style="list-style-type: none"> 7100 dials 1901→1901 answers (Shared Line on 7101) 8000 dials 444-7101→ 7101 answers 7101 selects line 1901 and hits softkey "Join" 8000 goes on-hook after 120s 7100 dials 1901→1901 answers (Shared Line on 1000) 7101 dials 1000→1000 answers 1000 selects line 1901 and hits softkey "Join" 7101 goes on-hook after 120s 7100 dials 1901→1901 answers (Shared Line on 2000) 8000 dials 444-2000→2000 answers 2000 selects line 1901 and hits softkey "Join" 7100 goes on-hook after 120s 2102225400 dials 94104441901 1901 answers (Shared line on 7101) 8000 dials 444-7101→7101 answers 7101 selects line 1901 and hits softkey "Join" 2102225400 goes on-hook after 120s Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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 							
Test Results: Comments						P	F	N/A	N/S	N/T	B
"Join Across Lines" is not supported									X		

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											
<ul style="list-style-type: none"> Local CUCM→DUT(s):7100 & 7101; SCCP:1000; SIP:2000; Remote CUCM →DUT:8000; PSTN: 2102225400; Hotline Configuration to dial out 234-8000: <ul style="list-style-type: none"> 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: <ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 7100 goes off-hook→8000 rings & answers 7100 goes on-hook after 30s 1000 goes off-hook→8000 rings & answers 1000 goes on-hook after 30s 2000 goes off-hook→8000 rings & answers 8000 goes on-hook after 30s 7101 goes off-hook→2102225400 rings & answers 7101 goes on-hook after 30 Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes <p>Note: Upon completion of the test, change CSS for 7100, 7101, 1000 & 2000→None</p> <p>CUCM Administration: Device→Phone→DN→CSS→None</p>				<ul style="list-style-type: 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
Post dial is not implemented in i62									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											
<ul style="list-style-type: none"> 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); <ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 8000 dials 444-7100 1000 goes off-hook, hits "Group Pickup" softkey 1000 enters Sales Group_Pickup DN:3005 1000 goes on-hook after 60s 8000 dials 444-2000 7101 goes off-hook, hits "Group Pickup" softkey 7101 enters TAC Group_Pickup DN:3006 8000 goes on-hook after 60s 2102225400 dials 4104441000 7100 goes off-hook, hits "Group Pickup" softkey 7100 enters Sales Group_Pickup DN:3005 2102225400 goes on-hook after 60s Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes <p>Note: Upon completion of the test, change Call Pickup Group for 7100, 7101, 1000 & 2000→None</p> <p>CUCM Administration: Device→Phone→DN→Line→Call Pickup Group→None</p>				<ul style="list-style-type: 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" is not implemented in i62									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											
<ul style="list-style-type: none"> Local CUCM→DUT(s):7100 & 7101; SCCP:1000; SIP:2000; Remote CUCM →DUT:8000; PSTN: 2102225400; Enable DND on DN:7101 <ul style="list-style-type: none"> 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: <ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 7100 dials 7101→7100 goes on-hook after 5 secs 8000 dials 444-7101→8000 goes on-hook after 5 secs 1000 dials 7101→1000 goes on-hook after 5 secs 2000 dials 7101→2000 goes on-hook after 5 secs 2102225400 dials 94104447101→2102225400 goes on-hook Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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
DND is not implemented in i62									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											
<ul style="list-style-type: none"> Local CUCM→DUT(s):7100 & 7101; SCCP:1000; SIP:2000; Remote CUCM →DUT:8000; PSTN: 2102225400; Enable DND on:7100 <ul style="list-style-type: none"> 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: <ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 1001 dials 7100→7100 answers 7101 dials 7100→7100 hits "DND" softkey in connected state 7101 goes on-hook 8000 dials 444-7100→7100 hits "DND" in connected state 8000 goes on-hook 1000 dials 7100→7100 hits "DND" softkey in connected state 1000 goes on-hook 2000 dials 7100→7100 hits "DND" softkey in connected state 2000 goes on-hook 2102225400 dials 7100→7100 hits "DND" in connected state 1001 goes on-hook after 200s Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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
DND is not implemented in i62									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											
<ul style="list-style-type: none"> 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 <ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 7100 dials 7101→7101 hits "iDivert" softkey during ringing state 7100 leaves a voicemail and goes on-hook 8000 dials 444-7101→7101 hits "iDivert" softkey in ringing state 8000 goes on-hook without leaving a message 1000 dials 7101→7101 hits "iDivert" softkey during ringing state 1000 leaves a voicemail and goes on-hook 2000 dials 7101→7101 hits "iDivert" softkey during ringing state 2000 leaves a voicemail and goes on-hook 2102225400 dials 94104447101 7101 hits "iDivert" softkey during ringing state 2102225400 leaves a voicemail and goes on-hook Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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
iDivert" is not implemented in i62.									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											
<ul style="list-style-type: none"> 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 <ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 8000 dials 4447101→7100 hits "iDivert" softkey in ringing state 8000 leaves voicemail and goes on-hook 1000 dials 7101→7100 answers 7100 hits "iDivert" softkey during connected state (after 10s) 1000 leaves a voicemail and goes on-hook 2000 dials 7101→7100 answers 7100 hits "iDivert" softkey during connected state (after 20s) 2000 leaves a voicemail and goes on-hook 2102225400 dials 94104447101→7100 answers 2102225400 goes on-hook without leaving voicemail Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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
iDivert" is not implemented in i62									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											
<ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> 7100 dials 7101→7101 answers→7101 hits "MCID" softkey 7101 goes on-hook after 30s 8000 dials 444-7100→7100 answers→7100 hits "MCID" 8000 goes on-hook after 30s 1000 dials 7100→7100 answers==7100 hits "MCID" softkey 7100 goes on-hook after 30s 2000 dials 7100→7100 answers→7100 hits "MCID" softkey 2000 goes on-hook after 30s 2102225400 dials 94104447101→7101 answers 7101 hits "MCID" softkey 7101 goes on-hook after 30s Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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 is not implemented in i62.								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									
<ul style="list-style-type: none"> 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: <ul style="list-style-type: none"> ✚ Add Remote Destination Profile:Device→Device Settings→Remote Destination Profile→Add New→mobility_1_rdp <ul style="list-style-type: none"> ○ Userid:dutuser02 ○ Line:7101 ○ Add New Remote Destination: <ul style="list-style-type: none"> ✚ 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					
<ol style="list-style-type: none"> 7100 dials 7101→7101 answers→7101 hits "Mobility" softkey after 30s Select the option to send call to mobile device 8000 answers 7101 goes on-hook after 30s 8000 sends DTMF *74 after 30s for call hanoff 7101 answers 8000 goes on-hook 7101 goes on-hook after 30s Repeat steps 1-7, replace 7100 with SCCP:1000 Repeat steps 1-7, replace 7100 with SIP:2000 Repeat steps 1-7, replace 7100 with PSTN:2102225400 Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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
"Mobility" softkey is not implemented in i62.						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											
<ul style="list-style-type: none"> • 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 <ul style="list-style-type: none"> + Add Remote Destination Profile:Device→Device Settings→Remote Destination Profile→Add New→mobile2_rdp <ul style="list-style-type: none"> o Userid:dutuser01 o Line:7100 o Add New Remote Destination: <ul style="list-style-type: none"> + 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							
<ol style="list-style-type: none"> 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 				<ul style="list-style-type: none"> • 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
"Mobility" softkey is not implemented in i62.								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									
<ul style="list-style-type: none"> 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 <ul style="list-style-type: none"> Add Remote Destination Profile:Device→Device Settings→Remote Destination Profile→Add New→mobile1_rdp <ul style="list-style-type: none"> userid:rcuser21 Line:6201 Add New Remote Destination: <ul style="list-style-type: none"> 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						
<ol style="list-style-type: none"> 7100 dials 2346201→7101 answers 7101 sends DTMF *74" after 30s 6201 answers 7101 goes on-hook 6201 hits "Mobility" softkey after 30s 7101 answers 6201 goes on-hook 7100 goes on-hook after 60s Repeat steps 1-8 and replace the Calling DN:1000 Repeat steps 1-8 and replace the Calling DN:2000 Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 			<ul style="list-style-type: none"> 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
"Mobility" softkey is not implemented in i62.						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											
<ul style="list-style-type: none"> • Local CUCM → DUT(s): 7100 & 7101; SCCP: 1000; SIP: 2000; • Remote CUCM → DUT: 8000; • CUCM Service Parameter: <ul style="list-style-type: none"> ✚ 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: <ul style="list-style-type: none"> ✚ Add Remote Destination Profile: Device → Device Settings → Remote Destination Profile → Add New → mobile1_rdp <ul style="list-style-type: none"> ○ Userid: dutuser02 ○ Line: 7101 ○ Add New Remote Destination: <ul style="list-style-type: none"> ✚ 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							
<ol style="list-style-type: none"> 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 				<ul style="list-style-type: none"> • 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
Mobile voice access is not supported on i62.								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									
<ul style="list-style-type: none"> • Local CUCM → DUT(s):7100 & 7101; SCCP:1000; SIP:2000; • Remote CUCM → DUT:8000; • Service Parameter: <ul style="list-style-type: none"> ✚ 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: <ul style="list-style-type: none"> ✚ Add Remote Destination Profile: Device → Device Settings → Remote Destination Profile → Add New → mobile1_rdp <ul style="list-style-type: none"> ○ Userid:dutuser02 ○ Line:7101 ○ Add New Remote Destination: <ul style="list-style-type: none"> ✚ 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						
<ol style="list-style-type: none"> 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 			<ul style="list-style-type: none"> • 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
Advanced functionality is not applicable						X			

Tests in this section require manual calls.
Run Step_4_ "Record_Negative_Test_Execution" command after executing all tests.
Retrieve CDR(s) from CUCM to validate calls.

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									
<ul style="list-style-type: none"> 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						
<ol style="list-style-type: none"> 7100 dials 7101 → 7101 answers 2000 dials 234-8000 → 8000 answers Access CUCM-PUB server via SSH (Local Cluster) Enter CLI: utils system restart <CR> yes 1000 dials 7101 → 7101 answers 2nd incoming call Called party goes on-hook for all 3 calls Repeat steps 1-2, 5-6 after CUCM-PUB recovery Retrieve CDR from CUCM Check Calling, Called, Duration, Origination & Termination Cause Codes 			<ul style="list-style-type: none"> 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									
<ul style="list-style-type: none"> 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						
<ol style="list-style-type: none"> 7100 dials 7101→7101 answers 2000 dials 234-8000→8000 answers call Access CUCM-SUB server via SSH (Local Cluster) Enter CLI: utils system restart <CR> yes 1000 dials 7101 Called party goes on-hook for all 3 calls Repeat steps 1-2, after CUCM-SUB recovery Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 			<ul style="list-style-type: none"> 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
The active calls were preserved over the fail over and fail back. However the third call which is to be connected was unsuccessful due to Subscriber failure.				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									
<ul style="list-style-type: none"> 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						
<ol style="list-style-type: none"> 7100 dials 7101→7101 answers Unplug network EPlE from device DN:7100 Restore the network EPlE after 60s 2000 dials 7100→7100 answers 7100 goes on-hook after 60s Retrieve CDR from CUCM Check Calling, Called, Duration, Origination & Termination Cause Codes 			<ul style="list-style-type: none"> 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
Disconnected network to AP				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									
<ul style="list-style-type: none"> 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						
<ol style="list-style-type: none"> 7100 dials 7101→7101 answers Remove power cable from 7101 Restore power cable after 60s 2000 dials 7101→7101 answers call 7101 goes on-hook after 60s Retrieve CDR from CUCM Check Calling, Called, Duration, Origination & Termination Cause Codes 			<ul style="list-style-type: none"> 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
Removed Battery from DUT				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									
<ul style="list-style-type: none"> 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						
<ol style="list-style-type: none"> 7100 dials 1000→1000 answers 7101 dials 7100 7101 dials 1000 7101 dials 2000 (RNA) 7101 dials 7777 (Invalid DN) 7101 dials 234-8000 7101 goes on-hook before 8000 answers (Abandoned) 7100 goes on-hook after 120 secs Retrieve CDR from CUCM Check Calling, Called, Duration, Origination & Termination Cause Codes 			<ul style="list-style-type: none"> 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					

Tests in this section require manual calls.
Run "Step 5_Record_Miscellaneous_Test_Execution" command after executing all tests in this section. Retrieve CDR(s) from CUCM to validate calls.

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)								
<ul style="list-style-type: none"> • 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: <ul style="list-style-type: none"> ➤ 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								
<ol style="list-style-type: none"> 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 	<ul style="list-style-type: none"> • 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
				X					

Test Case #	EP-52	Category	Miscellaneous Test: DUT display features	RFC_Standard	Y				
Objective	Verify different packetization period support on DUT(s) endpoints								
Pre-Test Conditions									
<ul style="list-style-type: none"> 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; 									
Test Procedure			Expected Results						
<ol style="list-style-type: none"> 7100 dials 7101 → 7101 answers 7100 goes on-hook after 60s 7100 dials 234-8000 → 8000 answers 8000 goes on-hook after 60s 7100 dials 1000 → 1000 answers 7100 goes on-hook after 60s 2000 dials 7100 → 7100 answers 7100 goes on-hook after 60s Repeat steps 1-8 with Packet Size=20 Repeat steps 1-8 with Packet Size=30 Retrieve CDR from CUCM Server Check Calling, Called, Duration, Origination & Termination Cause Codes 			<ul style="list-style-type: none"> 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 						
Test Results: Comments				P	F	N/A	N/S	N/T	B
Ptime= 10ms is not supported on i62. Tested device with 20 and 30ms ptimes. Note change required in CM device.				X					

Test Case #	EP-53	Category	Miscellaneous Test: DUT Screen Features	RFC_Standard	Y				
Objective	Verify the features displayed on the screen of DUT								
Pre-Test Conditions									
<ul style="list-style-type: none"> Local CUCM → DUT (s):7100 & 7101; 									
Test Procedure			Expected Results						
Check DUT:7100 phone display for: <ul style="list-style-type: none"> 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 # 			<ul style="list-style-type: none"> Able to access all these features from the DUT's phone display 						
Test Results: Comments				P	F	N/A	N/S	N/T	B
i62 doesn't support Multiple lines or Softkeys for call features. Tested with noted limitations.				X					

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									
<ul style="list-style-type: none"> 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						
<ol style="list-style-type: none"> 7100 dials 7101→7101 answers (Duration: 1 Hr.) 2000 dials 234-8000→ 8000 answers (Duration: 1 Hr) Repeat step 1 by replacing Called DN:92102225400 Repeat step 2 by replacing the Calling DN: 1000 Retrieve CDR from CUCM Check Calling, Called, Duration, Origination & Termination Cause Codes 			<ul style="list-style-type: none"> 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									
<ul style="list-style-type: none"> 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								
<ol style="list-style-type: none"> 7100 dials 1100→ 1100 answers→7100 on-hook after 120s 1100 dials 7100→7100 answers→1100 on-hook after 120s 7101 dials 1100 →1100 answers→7101 hits "Hold" after 20s 7101 hits "Resume" after 20s→1100 on-hook after 120s 1100 dials 234-8000→8000 answers→8000 on-hook after 90s 7100 dials 1100→1100 answers→ 1100 hits "Transfer" after 20s 1100 dials 7101→1100 hits "Transfer"→ 1100 on-hook 7100 goes on-hook after 120s 1100 dials 7100→7100 answers→7100 hits "Transfer" after 20s 7100 dials 234-8000→8000 answers 7100 hits "Transfer" after 30s→7100 on-hook 8000 goes on-hook after 120s 1100 dials 7101→7101 answers 7101 hits "Conference" after 30s→7101 dials 234-8000 8000 answers→7101 hits "Conference" after 30s 7101 goes on-hook after 120s 1100 and 8000 goes on-hook after 200s Repeat steps 1-17 by replacing DN:1100 with DN(s) of other Cisco phone models Retrieve CDR from CUCM Check Calling, Called, Duration, Origination & Termination Cause Codes 	<ul style="list-style-type: none"> 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 <p>Note: Any Cisco IP Phone models not supported in RPC will have Auto Answer Turned On to test basic call functions only.</p>								
Test Results: Comments				P	F	N/A	N/S	N/T	B
Tested with DX650 and 8945 phones. Basic scenarios were previously covered with 9951, 7961, and 7962 phones.				X					

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											
<ul style="list-style-type: none"> 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							
<ol style="list-style-type: none"> Provision all the lines for both 7100 & 7101 Device→Phone→DN→Line (DN Range: 7120 -7150) Initiate calls on all the lines between 7100 & 7101 Calling & Called parties goes on-hook alternatively at random duration Initiate intra-cluster & inter-cluster calls on all lines to SIP, SCCP & PSTN endpoints Calling & Called parties goes on-hook alternatively at random Duration. Initiate calls and perform mid-call features between these lines (Hold/Resume, Transfer, Conference, CFNA, CFB) Retrieve CDR from CUCM Check the Calling, Called, Duration, Origination & Termination Cause Codes 				<ul style="list-style-type: none"> 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
Multiple line feature not implemented on i62.								X			

Run “Step 6_Complete_Submit” command after executing all tests in this Test Plan. Complete the Test Result Matrix in Appendix A. Provide exceptions, notes or issues in the comments section. Submit this completed Test Report to sb-ivt-submit@cisco.com

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			Speed Dial not supported by i62 device.
EP-6				X			Speed Dial not supported by i62 device.
EP-7	X						In step 5, i62 phone call rang into second call appearance. Call was answered and ended.
EP-8	X						
EP-9	X						
EP-10	X						
EP-11	X						i62 hear tone (internal)
EP-12	X						
EP-13	X						
EP-14	X						During last test call (call4), DUT device cancels call2 to 1000 to resume with call1.
EP-15				X			Call Park not implemented in i62 device.
EP-16				X			Call Park not implemented in i62 device.
EP-17				X			Call Park not implemented in i62 device.
EP-18				X			Shared Line not implemented in i62 device.
EP-19	X						
EP-20				X			Meet-me not implemented in i62 device.
EP-21				X			Callback not implemented in i62 device.
EP-22				X			Shared Line not implemented in i62 device.
EP-23				X			Shared Line not implemented in i62 device.
EP-24				X			Shared Line not supported by i62 device.
EP-25	X						
EP-26	X						
EP-27			X				Video not supported with i62 device.
EP-28			X				Extension mobility not supported in i62 device.
EP-29	X						
EP-30	X						
EP-31	X						
EP-32				X			Authenticated security profiles not supported by i62.
EP-33				X			Join Across Lines not implemented in i62.
EP-34				X			Post dial not implemented in i62.
EP-35				X			Group pickup not implemented in i62.
EP-36				X			DND is not implemented in i62.
EP-37				X			DND is not implemented in i62.
EP-38				X			iDivert not implemented in i62
EP-39				X			iDivert not implemented in i62
EP-40			X				MCID softkey not implemented in i62.
EP-41			X				Mobility softkey not implemented in i62.
EP-42			X				Mobility softkey not implemented in i62.
EP-43			X				Mobility softkey not implemented in i62.
EP-44			X				Mobile Voice Access not supported with i62 device.
EP-45			X				Advanced functionality is not applicable.
EP-46	X						
EP-47	X						The active calls were preserved over the fail over and fail back. However the third call which is to be connected was unsuccessful due to Subscriber failure

Test Case #	P	F	NA	NS	NT	B	Comments
EP-48	X						Disconnected network to AP
EP-49	X						Removed Battery from DUT
EP-50	X						
EP-51	X						
EP-52	X						Ptime=10ms is not supported on i62. Tested device with 20 and 30ms ptimes. Note change required in CM device.

Test Case #	P	F	NA	NS	NT	B	Comments
EP-53	X						i62 doesn't support Multiple lines or Softkeys for call features. Tested with noted limitations.
EP-54	X						
EP-55	X						
EP-56			X				Multiple lines functionality not implemented on i62.

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



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