IVT Detailed Test Plan and Report for Chronicall 3.12(4l) with CUCM & UCCX12.0

Test Result:	PASS
Test Date:	11/13/2019
Product Name:	Xima Chronicall
Product Version: (must be generally available)	3.12(4l)
Communications Manager Version:	12.0
UCCX Version:	12.0
Unity Connection Version:	12.0
IM & Presence Version:	12.0
Platform / OS Version:	Windows Server 2012 R2
Product Type:	CAB, Voice Recording and Wallboard Application
API/Protocol(s) Used:	HTTP, SIP
Developer Services Contract:	
Partner IVT Contact Name:	Elliott Roska
Partner IVT Contact Phone:	+1 (888) 944 – 9462
Partner IVT Contact Email:	eroska@ximasoftware.com
Partner Main Support Number:	+1 (888) 944 – 9462
IVT Lab Location (EMEA or US):	US

Revision History

Revision	Name	Date	Comments
1.0	tekVizion	29th Oct-2019	Initial Draft
2.0	Arun Kumar Somalingam	13- Nov-2019	Updated Test Results

Table of Contents

1 Introduction

This document is the detailed Interoperability Verification Test Plan and Report for **Cisco Unified Communications Manager 12.0** and **Chronicall 3.12(4l).**

1.2 Entry Criteria

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

1.3 Exit Criteria

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

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

Table **1 Defect Severity Level Description**

Severity Level		Description
1	Catastrophic	Common circumstance causes the entire system or a subsystem to stop working affects other areas/devices no workaround
2	Severe	Important functions are unusable, does not affect other areas/devices and no workaround
3	Moderate	minor feature doesn't work and has low impact

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

The following procedures are followed when testing fails:

- Preliminary analysis is made to determine the source of the problem. If the problem is related to a device under test, then the problem is reported to that partner. If the problem is deemed Cisco related, the problem will be reported to Cisco, but the partner is responsible to open a TAC case with Cisco developer services. Partner should provide the TAC case number to the test team so they can document it in the report.
- If testing can continue past this failure, the other test cases will be tested and verified for pass or fail. If the testing cannot progress past this problem, testing will be halted and a final test report submitted to Partner and Cisco.
- All problems and resolutions encountered during testing are documented in the final test report.
- If a severity 1 failure occurs, irrespective of whom is responsible for the problem (Cisco or the 3rd party product), the testing is considered unsuccessful.

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

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

2 Product Overview

Xima Chronicall is an all-purpose custom reporting and real-time display solution. It provides detailed call logs of the Cisco UCM into a single detailed reporting interface. Historical reports provide over 50 standard reports allowing to you report on agents, groups, numbers, queues and more. You can drill down into the calls through Chronicall's Call Detail View, displaying call data from the UCM.

2.2 What Are Tested

2.3 What Will Not Be Tested

2.4 Assumptions

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

3 Executive Summary

Briefly describe the objective of the testing, the versions used. Summarize the test results with any major issues or observations from the test.

The following summarizes tekVizion's findings:

- Test Case Failures:
  None
- Features Not Supported:
  5.3.81- Recording for calls between Encrypted IP phones
  5.3.82- Recording for calls between Encrypted and Non Secure IP phones
  5.3.91- Selective Silent Recording for a call when media source is Gateway Preferred
  5.3.92- Simultaneous Automatic Recording and Silent monitoring is supported for a call
  5.3.96- Silent Monitoring for a call Hold/Resume initiated by observed party
  5.3.97- Silent Monitoring for a call "Hold/Resume" initiated by an Observer
  5.3.106- Agent Disconnect Call / Wrap-up
  5.3.107- Customer Disconnect Call / Wrap-up

- Test Cases that are Not Applicable:
  None

- Test Cases that were Not Executed:
  None

- Observations:

1. Application can retrieve recording only from one publisher of the cluster. So the recording from Remote CUCM cannot be retrieved by the application.
2. When both phone A and B are monitoring phones and they are talking to each other, the application records and save it as one individual record.
3. Login states are not updated in the "Group Real-time" column in the application. Even when the agent is logged in, the application shows as logged out. However, the agent states can be seen under "Agent States" in the wallboard.
4. Wrap up reason cannot be seen in the application.

4 Test Environment

4.2 Administration, Testing and Debugging tools

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

Table 2 Administration, Testing and Debugging Tools

Product Name	Version	Type	Purpose	Units	Notes
Test Tools					
None					
3rd Party Tools					
None					
Debug Tools					
None					

4.3 Equipment Requirements

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

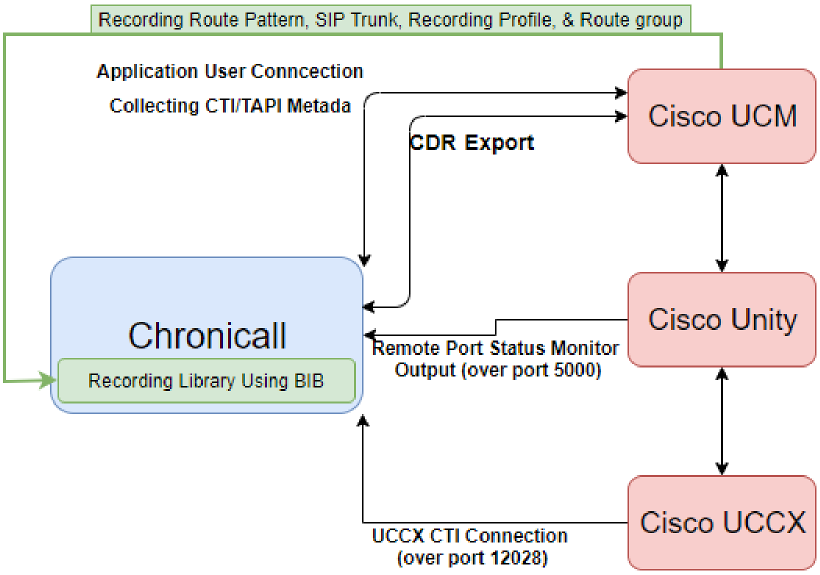
Table 3 Equipment and Product Information

Product	Version	Type	Purpose	Units	Notes
Cisco Products					
Cisco Unified CUCM	12.0	PBX	To register phones and manage calls both internal and externally	2	Lab Provided
Cisco Unity	12.0	Unity Connection	To setup voicemail feature in the phone.	1	Lab Provided
CUPS	12.0	IM and Presence	To register Cisco Jabber Clients and facilitate IM and Calling services	1	Lab Provided
Cisco Jabber	12.0	Client	To make calls and IM between Clients	2	Lab Provided
Cisco IP Communicator	8.6	Cisco Soft phone	To make calls between soft and physical endpoints	2	Lab Provided
3rd Party Products					
Xima Chronicall	3.12(4l)	CAB, Voice Recording and Wallboard Application	To collect Call Records and provides reports based on various requirements.	1	Customer Provided

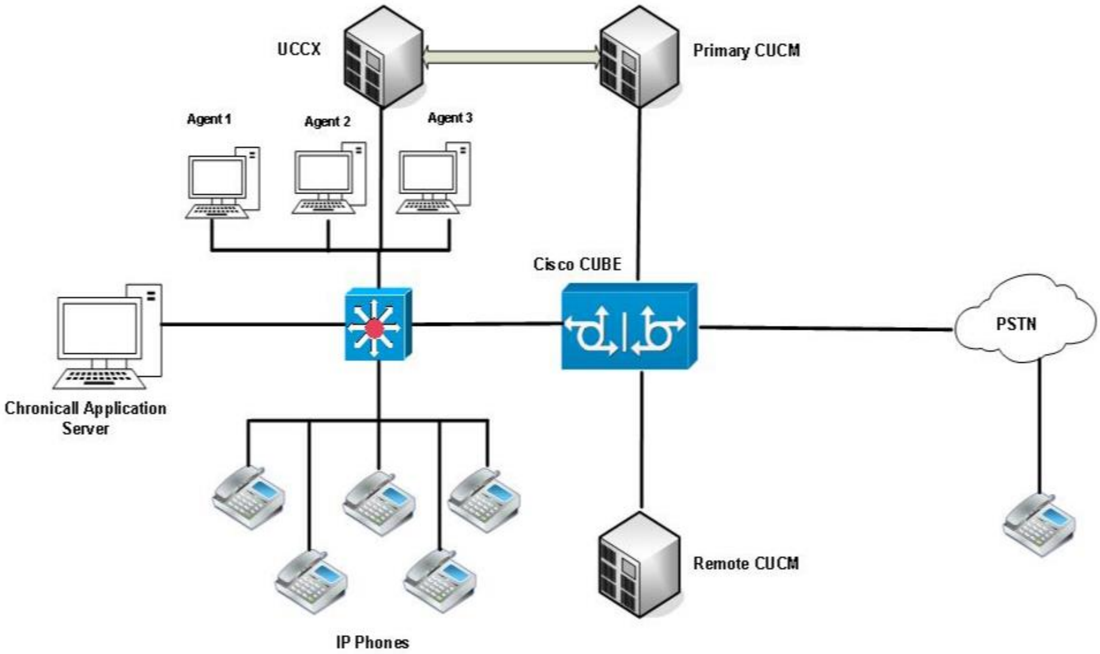
4.4 Cisco Phones

Cisco Phone Model	Phone Firmware Version	Protocol	POE/ Power	Units	Notes
Cisco 7965	SCCP45.9-4-2SR3-1S	SCCP	POE	2	Lab Provided
Cisco 7841	Sip78xx.12-1-1-12	SIP	POE	2	Lab Provided
Cisco 7942	SCCP42.9-4-2SR3-1S	SCCP	POE	1	Lab Provided

4.5 Deployment Architecture



4.6 Test Environment Architecture



5 Detailed Test Cases

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

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

5.2 Phase 1 Installation & Configuration

This phase included making sure the 3rd Party application installed correctly and that it registered correctly with the Communications Manager. These tests will also focus on clean installation, configuration and removal of any software components on the Communications Manager server(s).

5.2.1 Server application installation

Test Case Details	
Title	Server application installation
Description	• Installation of the server application. • Verification of the completeness and accuracy of the server application installation guide. • Documentation of any required software installation or configuration change on the Communications Manager.
Test Setup	
Procedure	• Read the 3rd party installation instructions. • **If 3rd party product is not turnkey:** On a designated client machine (not one of the Communications Manager machines), install the 3rd party product. Record length of time required for install. • Perform any configuration the 3rd party product according to the installation instructions. Document the various separate installation programs and manual configurations needed. • Check & document if the application is integrated with CUCM via FTP or SFTP
Expected Results	• Verify that via the partner application that the Server has installed correctly • Record any changes to the Communications Manager devices required by the 3rd party installation. • Confirm that there are no platform-agents present on the CUCM. • FTP or SFTP connection to the application is successful from CUCM.
Observations	PASS

5.2.2 Server Shutdown

Test Case Details	
Title	Server Shutdown
Description	Verify that that the server application shuts down properly.
Test Setup	None
Procedure	Follow installation guide (s): • Document how the product can be “shut down” • Validate that this works • Record the result.
Expected Results	As per stated in the procedure
Observations	PASS

5.3 Phase 2 Application Reliability Verification

Section: Call Accounting and Billing Tests

5.3.1 Inter-Cluster calls

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Inter-Cluster calls</td></tr>
<tr><td>Description</td><td>Verify application is able to retrieve CDRs for Inter-Cluster calls</td></tr>
<tr><td>Test Setup</td><td>

Application Server and CUCM clusters integrated successfully
CUCM Global Parameter Settings:

System➔Enterprise Parameters➔CDR File Time Interval➔1
System➔Service Parameters➔CDR Enabled Flag➔True
System➔Service Parameters➔CDR Log Calls with Zero Duration Flag➔True
System➔Service Parameters➔finalCalledPartyNumber➔True

Local CUCM➔SCCP: 1 phone ; SIP: 1 phone;
Remote CUCM➔SCCP: 1 phone ; SIP: 1 phone;
</td></tr>
<tr><td>Procedure</td><td>

Local SCCP Phone 1 dials Remote SCCP Phone 1
After 60 seconds, Local SCCP phone 1 goes on hook
Local SCCP Phone 1 dials Remote SIP Phone 1
After 60 seconds, Remote SIP phone 1 goes on hook
Local SIP Phone 1 dials Remote SIP Phone 1
After 60 seconds, Remote SIP phone 1 goes on hook
Local SIP Phone 1 dials Remote SCCP Phone 1
After 60 seconds, Remote SCCP phone 1 goes on hook
Retrieve CDR from application
Check the CDR fields
</td></tr>
<tr><td>Expected Results</td><td>

4 Calls established (talking state)
4 Calls terminated normally
4 Records retrieved
Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, duration fields in the CDR table for each call

<table>
<tr><th>CDR field</th><th>Call 1</th><th>Call 2</th><th>Call 3</th><th>Call 4</th></tr>
<tr><td>origCause_Value</td><td>16</td><td>0</td><td>16</td><td>0</td></tr>
</table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.2 Intra-Cluster calls

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Intra-Cluster calls</td></tr>
<tr><td>Description</td><td>Verify application is able to retrieve CDRs for Intra-Cluster calls</td></tr>
<tr><td>Test Setup</td><td>• Application Server and CUCM clusters integrated successfully
• CUCM Global Parameter Settings:
o System➔Enterprise Parameters➔CDR File Time Interval➔1
o System➔Service Parameters➔CDR Enabled Flag➔True
o System➔Service Parameters➔CDR Log Calls with Zero Duration Flag➔True
o System➔Service Parameters➔finalCalledPartyNumber➔True
• Local CUCM➔SCCP: 2 phones ; SIP: 2 phones;</td></tr>
<tr><td>Procedure</td><td>• SCCP Phone 1 dials SCCP Phone 2
• SIP Phone 1 dials SIP Phone 2
• Calling & Called party goes on-hook alternatively
• Retrieve CDR from application
• Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>• 2 Calls established (talking state)
• 2 Calls terminated normally
• 2 Records retrieved
• Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, duration fields in the CDR table for each call
<table>
<tr><th>CDR field</th><th>Call 1</th><th>Call 2</th></tr>
<tr><td>origCause_Value</td><td>16</td><td>0</td></tr>
</table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.3 CDRs for Off-Net calls

Test Case Details	
Title	CDRs for Off-Net calls
Description	Verify application is able to retrieve CDRs for Off-Net calls
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Local CUCM➔SCCP: 1 phone ; SIP: 1 phone; • PSTN➔ 2 phones ;
Procedure	• Local SCCP Phone 1 dials PSTN 1 • Local SIP Phone 1 dials PSTN 2 • Calling party goes on-hook • Retrieve CDR from application • Check the CDR fields

Expected Results	- 2 Calls established (talking state) - 2 Calls terminated normally - 2 Records retrieved - Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value=16, duration fields in the CDR table for each call
Observations	PASS

5.3.4 Called Party is busy (SIP)

Test Case Details	
Title	Called Party is busy (SIP)
Description	Verify CDR in Application when Called Party is busy (SIP)
Test Setup	- Application Server and CUCM clusters integrated successfully - **CUCM Global Parameter Settings** - Local CUCM➔SCCP: 1 phones ; SIP: 2 phones; - Disable VM & CW on SIP phone 2
Procedure	- Local SCCP Phone 1 dials Local SIP Phone 2 - Local SIP Phone 2 answers - Local SIP Phone 1 dials Local SIP Phone 2 - Local SCCP Phone 1 goes on-hook after 300s - Retrieve CDR from application - Check the CDR fields
Expected Results	- Call established between Local SCCP Phone 1 and Local SIP Phone 2 - Local SIP Phone 1 hears busy tone - Call terminated normally - 2 Records retrieved - Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, destCause_Value , duration fields in the CDR table for each call For call2, “destCause_Value = 17”
Observations	PASS

5.3.5 Called Party is busy (PSTN)

Test Case Details	
Title	Called Party is busy (PSTN)
Description	Verify CDR in Application when Called Party is busy (PSTN)
Test Setup	- Application Server and CUCM clusters integrated successfully - **CUCM Global Parameter Settings** - Local CUCM➔SCCP: 1 phone ; SIP: 1 phone; PSTN: 1 phone - Disable VM & CW on PSTN
Procedure	- Local SCCP Phone 1 dials PSTN Phone 1 - PSTN Phone 1 answers - Local SIP Phone 1 dials PSTN Phone 1

	• Local SCCP Phone 1 goes on-hook after 300s • Retrieve CDR from application • Check the CDR fields
Expected Results	• Call established between Local SCCP Phone 1 and PSTN Phone 1 • Local SIP Phone 1 hears busy tone • Call terminated normally • 2 Records retrieved • Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, destCause_Value, duration fields in the CDR table for each call For call2, “destCause_Value = 17”
Observations	PASS

5.3.6 Call to an invalid DN

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Call to an invalid DN</td></tr>
<tr><td>Description</td><td>Verify CDR in Application for a call to an invalid DN</td></tr>
<tr><td>Test Setup</td><td>• Application Server and CUCM clusters integrated successfully
• CUCM Global Parameter Settings
• Local CUCM➔SCCP: 1 phone ; SIP: 1 phone; PSTN: 1 phone(Invalid DN); Invalid DN : 9500(e.g)</td></tr>
<tr><td>Procedure</td><td>• Local SCCP Phone 1 dials 9500
• Local SCCP Phone 1 goes on-hook
• Local SIP Phone 1 dials PSTN Phone 1
• Local SIP Phone 1 goes on-hook
• Retrieve CDR from application
• Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>• Local SCCP Phone 1 and Local SIP Phone 1 hears reorder tone
• 2 Records retrieved
• Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, DestCause_Value, duration fields in the CDR table for each call
<table>
<tr><th>CDR field</th><th>Call 1</th><th>Call 2</th></tr>
<tr><td>origCause_Value</td><td>1</td><td>0</td></tr>
<tr><td>destCause_Value</td><td>0</td><td>1</td></tr>
<tr><td>duration</td><td>0</td><td>0</td></tr>
</table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.7 Call to an Unregistered IP Phone

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Call to an Unregistered IP Phone</td></tr>
<tr><td>Description</td><td>Verify CDR in Application for a call to an Unregistered IP Phone</td></tr>
<tr><td>Test Setup</td><td>Application Server and CUCM clusters integrated successfully
CUCM Global Parameter Settings
Local CUCM➔SCCP: 1 phone ; Unregistered IP Phone DN:1014</td></tr>
<tr><td>Procedure</td><td>Local SCCP Phone 1 dials Unregistered IP Phone DN
Local SCCP Phone 1 goes on-hook
Retrieve CDR from application
Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>Unregistered IP Phone DN hears reorder tone
1 Record retrieved
Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, destCause_Value, duration fields in the CDR table for each call
<table>
<tr><th>CDR field</th><th>Call 1</th></tr>
<tr><td>origCause_Value</td><td>27</td></tr>
<tr><td>destCause_Value</td><td>0</td></tr>
<tr><td>duration</td><td>0</td></tr>
</table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.8 Abandoned call

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Abandoned call</td></tr>
<tr><td>Description</td><td>Verify CDR in Application for an abandoned call</td></tr>
<tr><td>Test Setup</td><td>Application Server and CUCM clusters integrated successfully
CUCM Global Parameter Settings
Local CUCM➔SCCP: 1 phone ;</td></tr>
<tr><td>Procedure</td><td>Local SCCP Phone 1 goes Off-hook and On-hook quickly
Retrieve CDR from application
Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>Local SCCP Phone 1 abandoned call
1 Records retrieved
Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, destCause_Value, duration fields in the CDR table for each call
<table>
<tr><th>CDR field</th><th>Call 1</th></tr>
<tr><td>origCause_Value</td><td>16</td></tr>
<tr><td>destCause_Value</td><td>0</td></tr>
<tr><td>duration</td><td>0</td></tr>
</table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.9 CDR for short calls

Test Case Details	
Title	CDR for short calls
Description	Verify Application is able to retrieve CDR for short calls < 1 sec
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Local CUCM➔SCCP: 1 phone ; SIP: 1 Phone ; PSTN: 1 • Global parameter: "CdrLogCallsWithZeroDurationFlag"➔True
Procedure	• Local SIP Phone 1 dials Local SCCP Phone 1 • Local SCCP Phone 1 dials PSTN 1 • Local SIP Phone 1 dials PSTN 1 • Calling and Called party goes on-hook alternatively • Retrieve CDR from application • Check the CDR fields
Expected Results	• 3 short calls are established • 3 calls are terminated normally • 3 Records retrieved • Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, duration fields in the CDR table for each call
Observations	PASS

CDR field	Call 1	Call 2	Call 3
origCause_Value	16	16	0
duration	0	0	0

5.3.10 Call Hold/Resume On-Net call

Test Case Details	
Title	Call Hold/Resume On-Net call
Description	Verify CDR in Application for a Call Hold/Resume On-Net call
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Local CUCM➔SCCP: 2 phones ;
Procedure	• Local SCCP Phone 1 dials Local SCCP Phone 2 • Local SCCP Phone 2 answers • Local SCCP Phone 1 hits "Hold" softkey after 30s • Local SCCP Phone 1 hits "Resume" softkey after 30s • Local SCCP Phone 1 goes on-hook after 120s • Retrieve CDR from application • Check the CDR fields
Expected Results	• Call is established between Local SCCP Phone 1 & Local SCCP Phone 2 • Local SCCP Phone 2 is placed On-Hold • Local SCCP Phone 1 resumed the call after 30s

	• Call terminated normally • 1 Record retrieved • Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value = 16, duration fields in the CDR table for each call
Observations	PASS

5.3.11 Call Hold/Resume Off-Net call

Test Case Details	
Title	Call Hold/Resume Off-Net call
Description	Verify CDR in Application for a Call Hold/Resume Off-net call
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Local CUCM➔SCCP: 1 phone ; PSTN : 1
Procedure	• Local SCCP Phone 1 dials PSTN 1 • PSTN 1 answers • Local SCCP Phone 1 hits “Hold” softkey after 30s • Local SCCP Phone 1 hits “Resume” softkey after 30s • Local SCCP Phone 1 goes on-hook after 120s • Retrieve CDR from application • Check the CDR fields
Expected Results	• Call established between Local SCCP Phone 1 & PSTN 1 • PSTN 1 is placed On-Hold • Local SCCP Phone 1 resumed call after 30s • Call terminated normally • 1 Record retrieved • Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value= 16, duration fields in the CDR table for each call
Observations	PASS

5.3.12 CFA and CFNA enabled

Test Case Details	
Title	CFA and CFNA enabled
Description	Verify CDR in application for a call with CFA and CFNA enabled
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Local CUCM➔SCCP: 3 phones ; SIP: 1 Phone • Enable CFA on Local SIP Phone 1 (Device➔Phone➔ Local SIP Phone 1 ➔CFA➔ Local SCCP Phone 2) • Enable CFNA on Local SCCP Phone 2 (Device➔Phone➔ Local SCCP Phone 2 ➔CFNA➔ Local SCCP Phone 3)
Procedure	• Local SCCP Phone 1 dials Local SIP Phone 1 • Local SCCP Phone 2 does not answer

	• Local SCCP Phone 3 answers • Local SCCP Phone 1 goes on-hook after 60 secs • Retrieve CDR from application • Check the CDR fields
Expected Results	• Call forwarded to Local SCCP Phone 2 and phone rings • Call forwarded to Local SCCP Phone 3 and phone rings • Call established between Local SCCP Phone 1 & Local SCCP Phone 3 (talking state) • Call terminated normally • 1 Record retrieved • Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, lastRedirectRedirectReason = 2, duration fields in the CDR table for each call
Observations	PASS

5.3.13 CFB enabled

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>CFB enabled</td></tr>
<tr><td>Description</td><td>Verify CDR in Application for a call with CFB enabled</td></tr>
<tr><td>Test Setup</td><td>• Application Server and CUCM clusters integrated successfully
• **CUCM Global Parameter Settings**
• Local CUCM➔SCCP: 3 phones ; SIP: 1 Phone
• Enable CFB on Local SIP Phone 1 (Device➔Phone➔ Local SIP Phone 1 ➔CFB➔ Local SCCP Phone 3)</td></tr>
<tr><td>Procedure</td><td>• Local SCCP Phone 1 dials Local SIP Phone 1
• Local SIP Phone 1 answer
• Local SCCP Phone 2 dials Local SIP Phone 1
• Local SCCP Phone 3 answer
• Local SCCP Phone 3 goes on-hook after 60 secs
• Local SCCP Phone 1 goes on-hook after 60 secs
• Retrieve CDR from application
• Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>• Call established between Local SCCP Phone 1 and Local SIP Phone 1
• Call forwarded on busy to Local SCCP Phone 3
• Call established between Local SCCP Phone 2 & Local SCCP Phone 3 (talking state)
• Calls terminated normally
• 2 Record retrieved
• Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, lastRedirectRedirectReason, duration fields in the CDR table
<table>
<tr><th>CDR field</th><th>Call 1</th><th>Call2</th></tr>
<tr><td>lastRedirectRedirectReason</td><td>0</td><td>1</td></tr>
</table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.14 Verify CDR in Application for a CFNA call to a PSTN

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>CFNA enabled</td></tr>
<tr><td>Description</td><td>Verify CDR in Application for a call to a PSTN phone with CFNA enabled</td></tr>
<tr><td>Test Setup</td><td>• Application Server and CUCM clusters integrated successfully
• CUCM Global Parameter Settings
• Local CUCM➔SCCP: 1 phone ; SIP: 1 Phone ; PSTN: 1
• Enable CFNA on Local SCCP Phone 1 (Device➔Phone➔ Local SCCP Phone 1 ➔CFNA➔ PSTN 1)</td></tr>
<tr><td>Procedure</td><td>• Local SIP Phone 1 dials Local SCCP Phone 1
• Local SCCP Phone 1 does not answer
• PSTN 1 answer
• Local SIP Phone 1 goes on-hook after 60 secs
• Retrieve CDR from application
• Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>• Call forwarded to PSTN 1 as CFNA configured on SCCP phone 1
• Call established between Local SIP Phone 1 & PSTN 1 (talking state)
• Calls terminated normally
• 1 Record retrieved
• Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, lastRedirectRedirectReason, duration fields in the CDR table for each call
<table>
<tr><th>CDR field</th><th>Call 1</th></tr>
<tr><td>lastRedirectRedirectReason</td><td>2</td></tr>
</table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.15 Call Park

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Call Park</td></tr>
<tr><td>Description</td><td>Verify CDR on Application for a “Call Park” call</td></tr>
<tr><td>Test Setup</td><td>• Application Server and CUCM clusters integrated successfully
• CUCM Global Parameter Settings
• Local CUCM➔SCCP: 1 phone ; SIP: 2 Phones ;
• Call Park Code # 3001 provisioned in CUCM (Routing➔Call Park➔3001)</td></tr>
<tr><td>Procedure</td><td>• Local SCCP Phone 1 dials Local SIP Phone 1
• Local SIP Phone 1 answers
• Local SIP Phone 1 parks call using the “Park” soft key after 10s
• Local SIP Phone 2 dials park code:3001 after 20s
• Retrieve CDR from application
• Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>• Call established between Local SCCP Phone 1 & Local SIP Phone 1 (talking state)
• Local SCCP Phone 1 is parked</td></tr>
</table>

<table>
<tr><td></td><td>Local SIP Phone 2 picks up the parked call
Call established between Local SCCP Phone 1 & Local SIP Phone 2
Call terminated normally
2 Record retrieved
Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, lastRedirectRedirectReason, duration fields in the CDR table for each call
<table><tr><th>CDR field</th><th>Call 1</th><th>Call2</th></tr><tr><td>lastRedirectRedirectReason</td><td>0</td><td>8</td></tr></table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.16 Call Park Reversion

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Call Park Reversion</td></tr>
<tr><td>Description</td><td>Verify CDR on Application for a Call Park Reversion call</td></tr>
<tr><td>Test Setup</td><td>Application Server and CUCM clusters integrated successfully
CUCM Global Parameter Settings
Local CUCM➔SCCP: 1 phone ; SIP: 1 Phone ;
Call Park Code of 3001 provisioned in CUCM (Routing➔Call Park➔3001)
Call Park Reversion Timer Service Parameter➔60s</td></tr>
<tr><td>Procedure</td><td>Local SCCP Phone 1 dials Local SIP Phone 1
Local SIP Phone 1 answers
Local SIP Phone 1 parks call using the “Park” soft key after 40s
Do not pick up the parked call for 60s
Local SIP Phone 1 answers
Local SCCP Phone 1 goes on-hook after 170s
Retrieve CDR from application
Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>Call established between Local SCCP Phone 1 & Local SIP Phone 1 (talking state)
Local SCCP Phone 1 is parked
Park Reversion Timer expired
Local SIP Phone 1 is ringing
Call resumed between Local SCCP Phone 1 & Local SIP Phone 1
Call terminated normally
2 Record retrieved
Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, lastRedirectRedirectReason, duration fields in the CDR table for each call
<table><tr><th>CDR field</th><th>Call 1</th><th>Call2</th></tr><tr><td>lastRedirectRedirectReason</td><td>11</td><td>0</td></tr></table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.17 Consultative Transfer

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Consultative Transfer</td></tr>
<tr><td>Description</td><td>Verify CDR on Application for a Consultative Transfer call</td></tr>
<tr><td>Test Setup</td><td>Application Server and CUCM clusters integrated successfully
CUCM Global Parameter Settings
Local CUCM➔SCCP: 1 phone ; SIP: 1 Phone ; PSTN: 1</td></tr>
<tr><td>Procedure</td><td>Local SIP Phone 1 dials PSTN 1
PSTN 1 answers
Local SIP Phone 1 hits the “Transfer’ soft key after 80s
Local SIP Phone 1 dials Local SCCP Phone 1
Local SCCP Phone 1 answers
Local SIP Phone 1 hits the “Transfer” soft key after 20s
Local SIP Phone 1 goes on-hook
Local SCCP Phone 1 goes on-hook after 125s
Retrieve CDR from application
Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>Call established between Local SIP Phone 1 & PSTN 1 (talking state)
PSTN 1 is on-hold (MOH)
Call established between Local SIP Phone 1 & Local SCCP Phone 1
Local SIP Phone 1 is consulting with Local SCCP Phone 1
PSTN 1 is transferred to Local SCCP Phone 1
PSTN 1 & Local SCCP Phone 1 in talking state
Local SIP Phone 1 terminated normally
PSTN 1 & Local SCCP Phone 1 terminated normally
3 Record retrieved
Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCallTerminationOnBehalfOf, destCallTerminationOnBehalfOf Orig_Cause_Value, duration fields in the CDR table for each call
<table>
<tr><th>CDR field</th><th>Original Call CDR</th><th>Consultation Call CDR</th><th>Final Transferred Call CDR</th></tr>
<tr><td>origCallTerminationOnBehalfOf</td><td>10</td><td>10</td><td>10</td></tr>
<tr><td>destCallTerminationOnBehalfOf</td><td>10</td><td>10</td><td>12</td></tr>
</table></td></tr>
<tr><td>Observations</td><td>PASS
Expected 3 records. The application provides 2 records.
2nd and 3rd records are merged as one record.</td></tr>
</table>

5.3.18 Blind Transfer

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Blind Transfer</td></tr>
<tr><td>Description</td><td>Verify CDR on Application for a Blind Transfer call</td></tr>
<tr><td>Test Setup</td><td>• Application Server and CUCM clusters integrated successfully
• CUCM Global Parameter Settings
• Local CUCM➔SCCP: 1 phone ; SIP: 2 Phones ;</td></tr>
<tr><td>Procedure</td><td>• Local SCCP Phone 1 dials Local SIP Phone 1
• Local SIP Phone 1 answers
• Local SCCP Phone 1 hits the “Transfer’ softkey after 60s
• Local SCCP Phone 1 dials Local SIP Phone 2
• Local SCCP Phone 1 hits “Transfer” while Local SIP phone 2 is ringing
• Local SCCP Phone 1 goes on-hook
• Local SIP phone 2 answers
• Local SIP Phone 2 goes on-hook after 120s
• Retrieve CDR from application
• Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>• Call established between Local SCCP Phone 1 & Local SIP Phone 1 (talking state)
• Local SIP Phone 1 is on-hold (MOH)
• Local SCCP Phone 1 Blind transfers the call to Local SIP Phone 2
• Local SIP phone 1’s call is blind transferred to Local SIP phone 2
• All calls terminated normally
• 3 Record retrieved
• Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, origCallTerminationOnBehalfOf, duration fields in the CDR table for each call
<table>
<tr><th>CDR field</th><th>Original Call CDR</th><th>Consultation Call CDR</th><th>Final Transferred Call CDR</th></tr>
<tr><td>origCallTerminationOnBehalfOf</td><td>10</td><td>10</td><td>10</td></tr>
</table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.19 Direct Transfer from a shared line (SCCP)

Test Case Details	
Title	Direct Transfer from a shared line (SCCP)
Description	Verify CDR on Application for Direct Transfer call from a shared line (SCCP)
Test Setup	• Application Server and CUCM clusters integrated successfully • CUCM Global Parameter Settings • Local CUCM➔SCCP: 1 phone ; SIP: 2 Phones ; • Shared Line with DN:1901 on SCCP Phone 1 • SCCP phone 1 and SIP phone 3 shares the same DN 1901

Procedure	• Local SIP Phone 1 dials DN 1901 • 1901 answers (Shared line on SCCP phone 1) • Local SIP Phone 2 dials DN 1901 • 1901 answers second call (shared line on SCCP phone 1) • 1901 hits "Direct Transfer" soft key and selects first call after 30s and hits "Direct transfer". • 1901 goes on-hook • Local SIP Phone 2 goes on-hook after 120s • Retrieve CDR from application • Check the CDR fields
Expected Results	• Call established between Local SIP Phone 1 & Local SCCP Phone 1 1901 (talking state) • Local SIP Phone 1 is on-hold (MOH) • Call established between Local SIP phone 2 and 1901 • Local SIP phone 2 is directly transferred to SIP phone 1 • Call is established between Local SIP phone 1 and Local SIP phone 2 • All calls terminated normally • 3 Records retrieved • Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, origCallTerminationOnBehalfOf ,duration fields in the CDR table for each call
Observations	PASS

CDR field	Original Call CDR	Consultation Call CDR	Final Transferred Call CDR
origCallTerminationOnBehalfOf	10	10	10

5.3.20 Ad-Hoc Conference call

Test Case Details	
Title	Ad-Hoc Conference call
Description	Verify CDR on Application for an Ad-Hoc Conference call
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Local CUCM➔SCCP: 1 phone ; SIP: 1 Phone ; PSTN : 1
Procedure	• Local SIP Phone 1 dials Local SCCP Phone 1 • Local SCCP Phone 1 answers • Local SCCP Phone 1 hits "Conference" soft key after 60s • Local SCCP Phone 1 dials PSTN • PSTN Phone answers • Local SCCP Phone 1 hits "Conference" soft key after 20s • Local SCCP Phone 1 goes on-hook after 120s • The other parties goes on-hook after 180s • Retrieve CDR from application • Check the CDR fields
Expected Results	• Call established between Local SIP Phone 1 & Local SCCP Phone 1 (talking state)

	- Local SIP Phone 1 is on-hold (MOH) - Call established between Local SCCP Phone 1 & PSTN phone - Local SIP phone 1,Local SCCP phone 1 and PSTN phone are joined in a conference - Local SCCP phone 1 left the conference - Last 2 parties disconnected from conference bridge and connected like a regular call - All calls terminated normally - 6 Records retrieved - Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, origCallTerminationOnBehalf, duration fields in the CDR table for each call Also refer CDR values below.

CDR field	Original Call CDR	Setup Call CDR	Conference CDR1	Conference CDR2	Conference CDR3	Final CDR
origCallTerminationOnBehalf	4	4	4	12	4	4

Observations	**PASS** 1 record retrieved. When expanding the record in application, It shows the details of the call states and the corresponding CDR fields.

5.3.21 Ad-Hoc Conference calls are joined

Test Case Details	
Title	Ad-Hoc Conference calls are joined
Description	Verify CDR on Application when two Ad-Hoc Conference calls are joined
Test Setup	- Application Server and CUCM clusters integrated successfully - **CUCM Global Parameter Settings** - Local CUCM➔SCCP: 1 phone ; SIP: 2 Phones ; PSTN : 2 phones - CUCM Service parameter: Drop Ad Hoc Conference ➔ Never (Default)
Procedure	- Local SIP phone 1 dials Local SIP phone 2 - Local SIP phone 2 answers - Local SIP phone 2 hits "Conference" soft key after 30s - Local SIP phone 2 dials PSTN 1 - PSTN phone 1 answers - Local SIP phone 2 hits "Conference" soft key after 30s - Local SCCP phone 1 dials Local SIP phone 1 - Local SIP phone 1 answers incoming call - Local SCCP phone 1 hits "Conference" after 20s - Local SCCP phone 1 dials PSTN 2 - PSTN phone 2 answers - Local SCCP phone 1 hits "Conference" soft key after 20s - Local SIP phone 1 selects conference 1 and hits the "Join" soft key - Local SCCP phone 1 goes on-hook after 250s - Local SIP phone 1 goes on-hook after 300s

<table>
<tr><td></td><td>PSTN 2 goes on-hook after 350s
All other participants ended call after 420s
Retrieve CDR from application & CUCM
Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>Call established between Local SIP phone 2 & Local SIP phone 1 (talking state)
Local SIP phone 1 is placed on-hold (MOH)
Call established between Local SIP phone 2 & PSTN phone 1 (talking state)
All 3 participant joined in conference 1
Call established between Local SCCP phone 1 & Local SIP phone 1 (talking state)
Local SIP phone 1 is placed on-hold (MOH)
Call established between Local SCCP phone 1 & PSTN Phone 2 (talking state)
All 3 participants joined in conference 2
All participants in conference 1 & 2 are joined
Local SCCP phone 1 left conference
Local SIP phone 1 left conference
PSTN Phone 2 left conference
Local SIP phone 2 & PSTN phone 1 continued call
All calls terminated normally
12 Records retrieved
Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, origCallTerminationOnBehalfOf, lastRedirectRedirectReason, duration fields in the CDR table for each call
<table>
<tr><td>CDR field</td><td>Original Call CDR1</td><td>Setup Call CDR2</td><td>Conf. Call CDR3</td><td>Conf. Call CDR4</td><td>Conf. Call CDR5</td><td>Original Call CDR6</td></tr>
<tr><td>lastRedirectRedirectReason</td><td>0</td><td>0</td><td>98</td><td>98</td><td>98</td><td>0</td></tr>
<tr><td>CDR field</td><td>Setup Call CDR7</td><td>Conf. Call CDR8</td><td>Conf. Call CDR9</td><td>Conf. Call CDR10</td><td>Conf Call CDR11 (Join)</td><td>Conf CDR12</td></tr>
<tr><td>lastRedirectRedirectReason</td><td>0</td><td>98</td><td>98</td><td>98</td><td>4</td><td>98</td></tr>
<tr><td>CDR field</td><td>Final CDR</td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>lastRedirectRedirectReason</td><td>98</td><td></td><td></td><td></td><td></td><td></td></tr>
</table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.22 Meet-Me

Test Case Details	
Title	Meet-Me
Description	Verify CDR on Application for a Meet-Me Conference call
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Local CUCM➔SCCP: 1 phone ; SIP: 2 Phones ; • Create CTI_RP:Device➔CTI_RP➔DN:3002 • Assign Meet-Me CSS to Local SCCP Phone 1:Device➔Phone➔DN➔Local Phone 1's DN ➔CSS➔css_mm

Procedure	• Local SCCP Phone 1 hits the “Meet-Me” soft key and dials 3002 • Local SIP Phone 1 dials 3002 • Local SIP Phone 2 dials 3002 • All 3 members go on-hook after 300s • Retrieve CDR from application & CUCM • Check the CDR fields
Expected Results	• Local SCCP Phone 1 initiated a meet-me conference • Local SIP Phone 1 & Local SIP Phone 2 joined the meet-me conference bridge port • All 3 parties in conference (talking state) • Conference call terminated normally • 3 Records retrieved • Match th callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value=16, duration fields in the CDR table for each call
Observations	PASS

5.3.23 Call Waiting

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Call Waiting</td></tr>
<tr><td>Description</td><td>Verify CDR on Application for calls with Call Waiting active</td></tr>
<tr><td>Test Setup</td><td>• Application Server and CUCM clusters integrated successfully
• CUCM Global Parameter Settings
• Local CUCM➔SCCP: 2 phones ; SIP: 1 Phone ;</td></tr>
<tr><td>Procedure</td><td>• Local SIP Phone 1 dials Local SCCP Phone 1
• Local SCCP Phone 1 answers
• Local SCCP Phone 2 dials Local SIP Phone 1
• Local SIP Phone 1 answers incoming call
• Local SIP Phone 1 terminates 2nd call after 120s
• Local SIP Phone 1 resumes 1st call
• Local SCCP Phone 1 goes on-hook after 320s
• Retrieve CDR from application & CUCM
• Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>• Call established between Local SIP Phone 1 & Local SCCP Phone 1 (talking state)
• Local SCCP Phone 1 is On-Hold (MOH)
• Call established between Local SCCP phone 2 & Local SIP Phone 1 (talking state)
• 2nd call terminated normally
• 1st call resumed
• 1st call terminated normally
• 2 Records retrieved
• Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, DestCause_Value, duration fields in the CDR table for each call
<table>
<tr><th>CDR field</th><th>Call 1</th><th>Call 2</th></tr>
<tr><td>destCause_Value</td><td>16</td><td>0</td></tr>
</table>
</td></tr>
</table>

Observations	PASS

5.3.24 Callback

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Callback</td></tr>
<tr><td>Description</td><td>Verify CDR on Application for Callback calls</td></tr>
<tr><td>Test Setup</td><td>- Application Server and CUCM clusters integrated successfully
- CUCM Global Parameter Settings
- Local CUCM➔SCCP: 1 phone ; SIP: 2 Phones ;
- VM and CW disabled for all phones
- Callback softkey template assigned to all phones</td></tr>
<tr><td>Procedure</td><td>- Local SIP Phone 1 dials Local SIP Phone 2
- Local SIP Phone 2 answers
- Local SCCP Phone 1 dials Local SIP Phone 2
- Local SCCP Phone 1 hits “Callback” softkey and exits
- Local SIP Phone 1 goes on-hook after 120s
- Local SCCP Phone 1 dials Local SIP Phone 2 after the callback alert
- Local SIP Phone 2 answers
- Local SCCP Phone 1 goes on-hook after 120s
- Retrieve CDR from application & CUCM
- Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>- Call established between Local SIP Phone 1 & Local SIP Phone 2 (talking state)
- Local SCCP Phone 1 hears a busy tone
- Local SIP Phone 1 & Local SIP Phone 2 terminated call normally
- Local SCCP Phone 1 alerted of Local SIP Phone 2’s availability
- Call established between Local SCCP Phone 1 & Local SIP Phone 2 (talking state)
- Call terminated normally between Local SCCP Phone 1 & Local SIP Phone 2
- Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, origCallTerminationOnBehalfOf, duration fields in the CDR table for each call
<table>
<tr><th>CDR field</th><th>Call 1</th><th>Call 2-1</th><th>Call2-2</th><th>Call 3</th></tr>
<tr><td>origCallTerminationOnBehalfOf</td><td>12</td><td>20</td><td>20</td><td>12</td></tr>
</table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.25 Calls originated from a shared line

Test Case Details	
Title	Calls originated from a shared line
Description	Verify CDR on Application for calls originated from a shared line
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Local CUCM➔SCCP: 2 phones ; SIP: 1 Phone ; PSTN : 1 • Shared line DN:1906 is added to device with Local SCCP Phone 2 & Local SIP Phone 1
Procedure	• 1906(Shared line on SIP phone 1) dials Local SCCP Phone 1 • Local SCCP Phone 1 answers • Local SCCP Phone 1 goes on-hook after 60s • 1906(Shared line on Local SCCP Phone 2)dials PSTN • PSTN Phone answers • 1906 goes on-hook after 60s • Retrieve CDR from application & CUCM • Check the CDR fields
Expected Results	• Call established between 1906 & Local SCCP Phone 1 (talking state) • Call 1 terminated normally • Call established between 1906 & PSTN Phone (talking state) • Call 2 terminated normally • 2 Records retrieved • Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, duration fields in the CDR table for each call CDR field | Call 1 | Call 2 origCause_Value | 0 | 16
Observations	PASS

5.3.26 Calls terminated to a shared line

Test Case Details	
Title	Calls terminated to a shared line
Description	Verify CDR on Application for calls terminated to a shared line
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Local CUCM➔SCCP: 2 phones ; SIP: 2 Phones ; • Shared line DN:1907 is added to device with DN: Local SCCP Phone 1 & Local SIP Phone 2
Procedure	• Local SIP Phone 1 dials 1907 (Shared Line on SCCP phone 1) • 1907 answers • Local SIP Phone 1 goes on-hook after 60s • Local SCCP Phone 2 dials 1907 (Shared line on SIP phone 2) • 1907 answers

<table>
<tr><td></td><td>1907 goes on-hook after 60s
Retrieve CDR from application & CUCM
Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>Call established between Local SIP Phone 1 & 1907 (talking state)
Call 1 terminated normally
Call established between Local SCCP Phone 2 & 1907 (talking state)
Call 2 terminated normally
2 Records retrieved
Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, duration fields in the CDR table for each call
<table><tr><th>CDR field</th><th>Call 1</th><th>Call 2</th></tr><tr><td>origCause_Value</td><td>16</td><td>0</td></tr></table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.27 Call Hold/Resume on a shared line

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Call Hold/Resume on a shared line</td></tr>
<tr><td>Description</td><td>Verify CDR on Application for a Call Hold/Resume call on a shared line</td></tr>
<tr><td>Test Setup</td><td>Application Server and CUCM clusters integrated successfully
CUCM Global Parameter Settings
Local CUCM➔SCCP: 1 phone ; SIP: 1 Phone ;
Shared line DN:1908 is added to device with Local SCCP Phone 1</td></tr>
<tr><td>Procedure</td><td>Local SIP Phone 1 dials 1908(Shared line on SCCP phone 1)
1908 answers
1908 hits “Hold” soft key after 30s
1908 hits “Resume” soft key after 30s
1908 goes on-hook after 120s
Repeat test with Local SIP Phone 1 doing the “Hold/Resume”
Retrieve CDR from application & CUCM
Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>Call established between Local SIP Phone 1 & 1908 (talking state)
Call 1 terminated normally
Call established between 1908 & Local SIP Phone 1 (talking state)
Call 2 terminated normally
2 Records retrieved
Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, destCause_Value, duration fields in the CDR table for each call
<table><tr><th>CDR field</th><th>Call 1</th><th>Call 2</th></tr><tr><td>destCause_Value</td><td>0</td><td>16</td></tr></table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.28 MLPP Call

Test Case Details	
Title	MLPP Call
Description	Verify CDR on Application for an MLPP call
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Local CUCM➔SCCP: 3 phones ; • Configure MLPP domain: System➔MLPP➔Domain➔MLPP Domain➔ CAB001; • Configure MLPP on Phone devices: ➢ DN: Local SCCP Phone 1 (MLPP Domain: CAB001; MLPP Indication:On; MLPP Premption: Forceful;) ➢ DN: Local SCCP Phone 2 & Local SCCP Phone 3 (MLPP Domain: CAB001; MLPP Indication:On; MLPP Premption: Disabled;) • Configure partitions:exec ➔ css_exe; flash ➔css_flash; • Assign css_exec ➔ DN: Local SCCP Phone 2 ; css_flash➔DN: Local SCCP Phone 3; • Configure Translation Patterns: ➢ 90. Local SCCP Phone 1's DN (ex: 90.3125) with partition:exec and MLPP Precedence: Executive Overwrite; ➢ 90. Local SCCP Phone 1's DN (ex: 90.3125) with partition:flash and MLPP Precedence: Flash Overwrite;
Procedure	• Local SCCP Phone 3 dials 90. <Local SCCP Phone 1's DN> (Flash Overwrite) • Local SCCP Phone 1 answers • Local SCCP Phone 2 dials 90. <Local SCCP Phone 1's DN> (Executive Overwrite) after 3s • Local SCCP Phone 1 answers • Local SCCP Phone 1 goes on-hook after 120s • Retrieve CDR from application & CUCM • Check the CDR fields
Expected Results	• Local SCCP Phone 1 receives special precedence ringback & display • Call established between Local SCCP Phone 1 & Local SCCP Phone 3 • Local SCCP Phone 1 receives special precedence ringer and display • Local SCCP Phone 1 active call is pre-empted with the executive override call • Local SCCP Phone 1 answers the executive overwrite call • Call established between Local SCCP Phone 1 & Local SCCP Phone 2 (talking state) • Call from Local SCCP Phone 3 terminated • Local SCCP Phone 1 & Local SCCP Phone 2 terminated normally after 120s • 2 Records retrieved • Match the callingPartyNumber, OriginalCalledPartyNumber,

<table>
<tr><td></td><td>finalCalledPartyNumber, lastRedirectDn, origCause_Value, Dest_Cause_Value, OrigPrecedenceLevel ,duration fields in the CDR table for each call
<table><tr><th>CDR field</th><th>Call 1</th><th>Call 2</th></tr><tr><td>Orig_Cause_Value</td><td>8</td><td>0</td></tr><tr><td>Dest_Cause_Value</td><td>9</td><td>16</td></tr><tr><td>origPrecedenceLevel</td><td>0</td><td>0</td></tr></table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.29 Calls originating & terminating to a Jabber softphone

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Calls originating & terminating to a Jabber softphone</td></tr>
<tr><td>Description</td><td>Verify CDR on Application for calls originating & terminating to a Jabber softphone (Jabber for Windows)</td></tr>
<tr><td>Test Setup</td><td>• Application Server and CUCM clusters integrated successfully
• CUCM Global Parameter Settings
• Local CUCM➔SCCP: 1 phone ; SIP: 1 phone
• Jabber for Windows (Device➔Phone➔Add New➔CSFUSER:DN:1922-1923; End User:user1-2/123456)
• 2 Windows PC with Jabber clients installed</td></tr>
<tr><td>Procedure</td><td>• 1922 dials 1923 (Duration=30s)
• Local SCCP Phone 1 dials 1922 (Duration=30s)
• 1923 dials Local SIP Phone 1 (Duration=30s)
• Calling and Called party goes on-hook alternatively
• Retrieve CDR from application & CUCM
• Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>• 3 calls established
• 3 calls terminated
• 3 Records retrieved
• Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, duration fields in the CDR table for each call
<table><tr><th>CDR field</th><th>Call 1</th><th>Call 2</th><th>Call 3</th></tr><tr><td>origCause_Value</td><td>16</td><td>0</td><td>16</td></tr></table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.30 Extension Mobility

Test Case Details	
Title	Extension Mobility
Description	Verify CDR on Application for Extension Mobility call
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Local CUCM➔SCCP: 3 phones ; SIP: 1 phone; • Configure Extension Mobility on Local SCCP Phone 1 • Configure Extension Mobility on as DN##(EM User) • Configure Extension Mobility service on the CUCM
Procedure	• Local SCCP Phone 1 hits "Services" button and selects EM service • Local SCCP Phone 1 logs in with "psdtuser1/123456" • EMuser dials Local SCCP Phone 2's DN • Local SCCP Phone 2 answers • EMuser goes on-hook after 125s • Local SIP Phone 1 dials EMuser • EMuser rings and answers • Local SIP Phone 1 goes on-hook after 65s • Local SCCP Phone 3 dials EMuser • EMuser rings and answers • Local SCCP Phone 3 goes on-hook after 65s • EMuser hits "Services" button and selects EM service • EMuser logs out • Retrieve CDR from application & CUCM • Check the CDR fields
Expected Results	• Login successful – phone rebooted with EMuser's DN • 3 calls established (talking state) • All calls terminated normally • EMuser logs out and device rebooted to Local SCCP Phone 1 device profile • 3 Records retrieved • Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, duration fields in the CDR table for each call **CDR field** | **Call 1** | **Call 2** | **Call 3** origCause_Value | 16 | 16 | 16
Observations	**PASS** The Calling Party (EM user) is shown as Unlicensed in the application.

5.3.31 FAC

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>FAC</td></tr>
<tr><td>Description</td><td>Verify CDR on Application for FAC calls</td></tr>
<tr><td>Test Setup</td><td>• Application Server and CUCM clusters integrated successfully
• CUCM Global Parameter Settings
• Two CUCM Clusters In-Service
• Local CUCM -> SCCP : 1 phone ;SIP: 1 Phone ;PSTN 1
• Configure FAC(####) code associating to PSTN Trunk
o Create FAC Legal1; Code: FAC(####); Level=1 in CUCM (Routing➔Forced Authorization Codes➔Add New)
o Enable FAC(####) (Call Routing➔Route/Hunt➔Route Pattern➔9.XXXXXXXXXX Check FAC checkbox)</td></tr>
<tr><td>Procedure</td><td>• Local SCCP Phone 1 dials PSTN
• Caller enters FAC(####)#
• PSTN Phone 1 answers
• Local SCCP Phone 1 goes on-hook after 20s
• Retrieve CDR from application & CUCM
• Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>• Caller prompted to enter FAC
• Call established between Local SCCP Phone 1 & PSTN Phone 1 (talking state)
• Call terminated normally
• 1 Record retrieved
• Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, authCodeDescription ,authorizationLevel, duration fields in the CDR table for each call
<table>
<tr><th>CDR field</th><th>Call 1</th></tr>
<tr><td>authCodeDescription</td><td>Legal1</td></tr>
<tr><td>authorizationLevel</td><td>1</td></tr>
</table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.32 Incorrect FAC

Test Case Details	
Title	incorrect FAC
Description	Verify CDR on Application for calls with incorrect FAC
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Local CUCM -> SIP: 1 Phone ;PSTN: 1 • Configure FAC(####) code associating to PSTN Trunk o Create FAC Legal1; Code: FAC(####); Level=1 in CUCM (Routing➔Forced Authorization Codes➔Add New) o Enable FAC(####) (Call Routing➔Route/Hunt➔Route Pattern➔9.XXXXXXXXXX Check FAC checkbox)

<table>
<tr><td>Procedure</td><td>• Local SIP Phone 1 dials PSTN DN
• Caller enters incorrect FAC(####) code
• Local SIP Phone 1 goes on-hook
• Retrieve CDR from application & CUCM
• Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>• Caller prompted to enter FAC
• Call failed due to incorrect FAC
• 1 Record retrieved
• Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, authCodeDescription ,authorizationLevel ,duration fields in the CDR table for each call
<table>
<tr><th>CDR field</th><th>Call 1</th></tr>
<tr><td>authCodeDescription</td><td>Invalid Authorization Code</td></tr>
<tr><td>authorizationLevel</td><td>1</td></tr>
<tr><td>origCause_Value</td><td>3</td></tr>
</table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.33 Hunt Group

Test Case Details	
Title	Hunt Group
Description	Verify CDR on Application for Hunt Group calls
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Local CUCM Phones –SCCP : 1 Phone ; SIP : 1 Phone • Configure Hunt Group # On Local SIP Phone 1 as a member of hunt group.
Procedure	• Local SCCP Phone 1 dials hunt group # • Local SIP Phone 1 answers • Local SIP Phone 1 goes on-hook after 120s • Retrieve CDR from application & CUCM • Check the CDR fields
Expected Results	• Call routed to hunt group member Local SIP Phone 1 • Call established between Local SCCP Phone 1 & Local SIP Phone 1 (talking state) • Call terminated normally • 1 Record retrieved • Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, huntPilotDN, totalWaitTimeInQueue , duration fields in the CDR table for each call
Observations	PASS

5.3.34 Hunt Group calls when no members available

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Hunt Group calls when no members available</td></tr>
<tr><td>Description</td><td>Verify CDR on Application for Hunt Group calls when no members available</td></tr>
<tr><td>Test Setup</td><td>

Application Server and CUCM clusters integrated successfully
CUCM Global Parameter Settings
Local CUCM Phones –SCCP : 2 Phones ; SIP : 2 Phones
Configure Hunt Group # On Local SIP Phone 2 as a member of hunt group. queuing flag enabled, max. waiting timer=60s, route call to dest= Local SCCP Phone 2;

</td></tr>
<tr><td>Procedure</td><td>

Local SIP Phone 1 dials hunt group #
Local SIP Phone 2 answers
Local SCCP Phone 1 dials hunt group #
Local SCCP Phone 2 answers
Local SCCP Phone 1 goes on-hook after 60s
Local SIP Phone 1 goes on-hook after 600s
Retrieve CDR from application & CUCM
Check the CDR fields

</td></tr>
<tr><td>Expected Results</td><td>

Call routed to hunt group member Local SIP Phone
Call established between Local SIP Phone 1 & Local SIP Phone 2(talking state)
Hunt Group has no members available
Local SIP Phone 2 routed to alternate DN: Local SCCP Phone 2
Call established between Local SCCP Phone 1 & Local SCCP Phone 2 (talking state)
Both calls terminated normally
3 Records retrieved
Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, wasCallQueued, totalWaitTimeInQueue, duration fields in the CDR table for each call

<table>
<tr><th>CDR field</th><th>Call 1</th><th>Call 2 – CDR1</th><th>Call 2 – CDR2</th></tr>
<tr><td>wasCallQueued</td><td>0</td><td>1</td><td>1</td></tr>
<tr><td>totalWaitTimeInQueue</td><td>0</td><td>60</td><td>80</td></tr>
</table>
</td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.35 Hunt Group calls that exceeded maximum queue length

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Hunt Group calls that exceeded maximum queue length</td></tr>
<tr><td>Description</td><td>Verify CDR on Application for Hunt Group calls that exceeded maximum queue length</td></tr>
<tr><td>Test Setup</td><td>Application Server and CUCM clusters integrated successfully
CUCM Global Parameter Settings
Local CUCM Phones –SCCP : 2 Phones ; SIP : 2 Phones
Configure Hunt Group # On Local SIP Phone 2 as a member of hunt group. queuing flag enabled, max. route call to dest. disabled; max. # of callers in queue=1;</td></tr>
<tr><td>Procedure</td><td>Local SCCP Phone 1 dials hunt group #
Local SIP Phone 2 answers
Local SCCP Phone 2 dials hunt group #
Local SIP Phone 1 dials hunt group #
Local SCCP Phone 1 goes on-hook after 600s
Retrieve CDR from application & CUCM
Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>Call routed to hunt group member Local SIP Phone 2
Call established between Local SCCP Phone 1 & Local SIP Phone 2 (talking state)
Local SIP Phone 1 & Local SCCP Phone 2 are waiting in queue
Maximum number of callers in queue exceeded
Both calls were not terminated to hunt group
3 Records retrieved
Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, wasCallQueued, totalWaitTimeInQueue, duration fields in the CDR table for each call
<table>
<tr><th>CDR field</th><th>Call 1</th><th>Call 2 – CDR1</th><th>Call 2 – CDR2</th></tr>
<tr><td>wasCallQueued</td><td>0</td><td>1</td><td>1</td></tr>
<tr><td>totalWaitTimeInQueue</td><td>0</td><td>60</td><td>0</td></tr>
</table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.36 Calls between Encrypted IP Phones

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Calls Between Encrypted IP Phones</td></tr>
<tr><td>Description</td><td>Verify CDR on Application for calls between encrypted IP Phones</td></tr>
<tr><td>Test Setup</td><td>Application Server and CUCM clusters integrated successfully
CUCM Global Parameter Settings
Create Encrypted Phone Configuration file
Local CUCM Phones –SCCP : 1 Phone ; SIP : 1 Phone</td></tr>
</table>

	(Both phones registered with encrypted configuration file)
Procedure	• Local SCCP Phone 1 dials Local SIP Phone 1 • Local SIP Phone 1 answers • Local SCCP Phone 1 goes on-hook after 120s • Retrieve CDR from application & CUCM • Check the CDR fields
Expected Results	• Call established between Local SCCP Phone 1 & Local SIP Phone 1 (talking state) • Call terminated normally • 1 Record retrieved • Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, callSecureStatus,duration fields in the CDR table for each call CDR field: Call 1 callSecuredStatus: 2
Observations	PASS

5.3.37 Calls between Authenticated IP Phones

Test Case Details	
Title	Calls Between Authenticated IP Phones
Description	Verify CDR on Application for calls between authenticated IP Phones
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Create Authenticated Phone Configuration file • Local CUCM Phones –SCCP : 1 Phone ; SIP : 1 Phone (Both phones registered with Authenticated configuration file)
Procedure	• Local SCCP Phone 1 dials Local SIP Phone 1 • Local SIP Phone 1 answers • Local SCCP Phone 1 goes on-hook after 120s • Retrieve CDR from application & CUCM • Check the CDR fields
Expected Results	• Call established between Local SCCP Phone 1 & Local SIP Phone 1 (talking state) • Call terminated normally • 1 Record retrieved • Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, callSecureStatus, duration fields in the CDR table for each call CDR field: Call 1 callSecuredStatus: 1
Observations	PASS

5.3.38 Calls between Authenticated and encrypted IP Phones

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Calls between Authenticated and encrypted IP Phones</td></tr>
<tr><td>Description</td><td>Verify CDR on Application for calls between authenticated and encrypted IP Phones</td></tr>
<tr><td>Test Setup</td><td>• Application Server and CUCM clusters integrated successfully
• CUCM Global Parameter Settings
• Create Authenticated Phone Configuration file
• Configure SCCP Phone 1 as Encrypted Phone
• Configure SIP Phone 1 as Authenticated phone in CUCM</td></tr>
<tr><td>Procedure</td><td>• Local SCCP Phone 1 dials Local SIP Phone 1
• Local SIP Phone 1 answers
• Local SIP Phone 1 goes on-hook after 120s
• Retrieve CDR from application & CUCM
• Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>• Call established between Local SCCP Phone 1 & Local SIP Phone 1 (talking state)
• Call terminated normally
• Call terminated normally
• 1 Record retrieved
• Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, callSecureStatus, duration fields in the CDR table for each call
<table>
<tr><th>CDR field</th><th>Call 1</th></tr>
<tr><td>callSecuredStatus</td><td>1</td></tr>
</table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.39 Calls between authenticated and non-secure IP Phones

Test Case Details	
Title	Calls between authenticated and non-secure IP Phones
Description	Verify CDR on Application for calls between authenticated and non-secure IP Phones
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Create Authenticated Phone Configuration file • Configure SCCP Phone 1 & SIP Phone 1 & PSTN 1 as Non secure Phone • Configure SCCP Phone 2 & SIP Phone 2 as Authenticated phones in CUCM
Procedure	• Local SCCP Phone 2 dials Local SCCP Phone 1 • Local SCCP Phone 1 answers • Local SCCP Phone 1 goes on-hook after 120s • Local SIP Phone 1 dials Local SIP Phone 2

<table>
<tr><td></td><td>• Local SIP Phone 2 answers
• Local SIP Phone 2 goes on-hook after 120s
• Local SCCP Phone 2 dials PSTN 1
• PSTN 1 answers
• PSTN 1 goes on-hook after 120s
• Retrieve CDR from application & CUCM
• Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>• Call established between Local SCCP Phone 2 & Local SCCP Phone 1 (talking state)
• Call terminated normally
• Call established between Local SCCP Phone 1 & Local SIP Phone 2 (talking state)
• Call terminated normally
• Call established between Local SCCP Phone 2 & PSTN 1 (talking state)
• Call terminated normally
• 3 Records retrieved
• Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, callSecureStatus, duration fields in the CDR table for each call
<table>
<tr><th>CDR field</th><th>Call 1</th><th>Call 2</th><th>Call 3</th></tr>
<tr><td>callSecuredStatus</td><td>0</td><td>0</td><td>0</td></tr>
</table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.40 Calls between encrypted and non-secure IP Phones

Test Case Details	
Title	Calls between encrypted and non-secure IP Phones
Description	Verify CDR on Application for calls between encrypted and non-secure IP Phones
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Create Authenticated Phone Configuration file • Configure SCCP Phone 1 & SIP Phone 1 as Non secure Phone • Configure SCCP Phone 2 & SIP Phone 2 as Encrypted phones in CUCM
Procedure	• Local SCCP Phone 2 dials Local SIP Phone 1 • Local SIP Phone 1 answers • Local SCCP Phone 2 goes on-hook after 120s • Local SCCP Phone 1 dials Local SIP Phone 2 • Local SIP Phone 2 answers • Local SCCP Phone 1 goes on-hook after 120s • Retrieve CDR from application & CUCM • Check the CDR fields
Expected Results	• Call established between Local SCCP Phone 2 & Local SIP Phone 1 (talking state) • Call terminated normally • Call established between Local SCCP Phone 1 & Local SIP Phone 2

<table>
<tr><td></td><td>(talking state)
• Call terminated normally
• 2 Records retrieved
• Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, callSecureStatus, duration fields in the CDR table for each call
<table>
<tr><td>CDR field</td><td>Call 1</td><td>Call 2</td></tr>
<tr><td>callSecuredStatus</td><td>0</td><td>0</td></tr>
</table>
</td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.41 Join across Lines feature

Test Case Details	
Title	Join Across Lines feature
Description	Verify CDR on Application for calls that used Join Across Lines feature (SIP)
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Local CUCM Phones –SCCP : 2 Phones ; SIP : 2 phones • Configure shared line on to SIP phone 1 with SIP phone 2'sDN and set the Join Across Line On the SIP Phone 1
Procedure	• Local SCCP Phone 1 dials the shared line DN • Shared line answers (Shared Line on Local SIP Phone 1) • Local SCCP Phone 2 dials Local SIP Phone 1 • Local SIP Phone 1 answers • Local SIP Phone 1 selects the shared line and hits softkey "Join" • All parties end the conference after 120s • Retrieve CDR from application & CUCM • Check the CDR fields
Expected Results	• Call established between Local SCCP Phone 1 & shared line on Local SIP Phone 1 (talking state) • Shared Line on Local SIP Phone 1 is placed on-hold • Call established between Local SCCP Phone 2 & Local SIP Phone 2 (talking state) • All parties are joined in a conference • Conference call terminated normally • 5 Records retrieved • Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, LastRedirectRedirectOnBehalfOf, origCallTerminationOnBehalfOf, duration fields in the CDR table for each call

CDR field	Original Call CDR	Setup Call CDR	Conference CDR1	Conference CDR2	Conference CDR3
origCallTerminationOnBehalfOf	4	4	12	4	12
Orig_Cause_Value	393216	0	16	393216	16

Observations	PASS 2 Records Retrieved. When expanding the record in application, It shows the details of the call states and the corresponding CDR fields.

5.3.42 Group Pickup for On-Net calls

Test Case Details	
Title	Group Pickup for On-Net calls
Description	Verify CDR on Application for Group Pickup for On-Net calls
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Local CUCM Phones –SCCP : 1 Phone ; Sip : 5 Phones; • Configure 2 groups as Sales and TAC with the members of SIP Phone 2, SIP Phone 3, SIP Phone 4, SIP Phone 5 • Group : SALES (SIP Phone 2, SIP Phone 3) • Group : TAC (SIP Phone 4, SIP Phone 5) • Call Routing➔Call Pickup Group➔Add New➔Sales (DN #1;Visual Alert; Calling & Called party checked) • Call Routing➔Call Pickup Group➔Add New➔TAC (DN #2;Visual Alert; Calling & Called party checked) • Device➔Phone➔DN➔update Call Pickup Group to Sales for SIP Phone 2, SIP Phone 3; • Device➔Phone➔DN➔update Call Pickup Group to TAC for SIP Phone 4, SIP Phone 5;
Procedure	• local SIP Phone 1 dials Local SIP Phone 5 • Local SIP Phone 4 goes off-hook, hits “Group Pickup” softkey • Local SIP Phone 4 enters TAC group_pickup DN#2 • local SIP Phone 1 goes on-hook after 120s • local SCCP Phone 1 dials local SIP Phone 2 • local SIP Phone 3 goes off-hook, hits “Group Pickup” softkey • local SIP Phone 3 enters Sales group_pickup DN#1 • local SCCP Phone 1 goes on-hook after 120s • Retrieve CDR from application & CUCM • Check the CDR fields
Expected Results	• local SIP Phone 5 in alerting state • Call is established between local SIP Phone 1 & local SIP Phone 4 (talking state) • Call terminated normally • local SIP Phone 2 in alerting state • Call is established between local SCCP Phone 1 & local SIP Phone 2 (talking state) • Call terminated normally • 4 Records retrieved • Match the callingPartyNumber, OriginalCalledPartyNumber, origCallTerminationOnBehalfOf , lastRedirectRedirectOnBehalfOf ,finalCalledPartyNumber, lastRedirectDn, origCause_Value, duration fields in the CDR table for each call

CDR field	Original Call 1 - CDR1	Pickup CDR2	Original Call 2 - CDR1	Pickup CDR2
origCallTerminationOnBehalfOf	16	16	16	16
lastRedirectRedirectOnBehalfOf	0	16	0	16

Observations	PASS

5.3.43 Group Pickup for Off-Net calls

Test Case Details	
Title	Group Pickup for Off-Net calls
Description	Verify CDR on Application for Group Pickup for Off-net calls
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Local CUCM Phones – Sip : 4 Phones; PSTN 2 phones • Configure 2 groups as Sales and TAC with the members of SIP Phone 1, SIP Phone 2, SIP Phone 3, SIP Phone 4 • Group : SALES (SIP Phone 1, SIP Phone 2) • Group : TAC (SIP Phone 3, SIP Phone 4) • Call Routing➔Call Pickup Group➔Add New➔Sales (DN #1;Visual Alert; Calling & Called party checked) • Call Routing➔Call Pickup Group➔Add New➔TAC (DN #2;Visual Alert; Calling & Called party checked) • Device➔Phone➔DN➔update Call Pickup Group to Sales for SIP Phone 1, SIP Phone 2; Visual Alert; Calling & Called party checked • Device➔Phone➔DN➔update Call Pickup Group to TAC for SIP Phone 3, SIP Phone 4; Visual Alert; Calling & Called party checked
Procedure	• PSTN Phone 1 dials Local SIP Phone 4 • Local SIP Phone 3 goes off-hook, hits “Group Pickup” softkey • Local SIP Phone 3 enters TAC group_pickup DN#2 • PSTN Phone 1 goes on-hook after 120s • PSTN Phone 2 dials local SIP Phone 1 • local SIP Phone 2 goes off-hook, hits “Group Pickup” softkey • local SIP Phone 2 enters Sales group_pickup DN#1 • PSTN Phone 2 goes on-hook after 120s • Retrieve CDR from application & CUCM • Check the CDR fields
Expected Results	• local SIP Phone 4 in alerting state • Call is established between PSTN Phone 1 & local SIP Phone 3 (talking state) • Call terminated normally • local SIP Phone 1 in alerting state • Call is established between PSTN Phone 2 & local SIP Phone 1 (talking state) • Call terminated normally • 4 Records retrieved • Match the callingPartyNumber, OriginalCalledPartyNumber,

<table>
<tr><td></td><td>finalCalledPartyNumber, lastRedirectDn, origCause_Value, origCallTerminationOnBehalfOf , lastRedirectRedirectOnBehalfOf, duration fields in the CDR table for each call
<table>
<tr><th>CDR field</th><th>Original Call 1 - CDR1</th><th>Pickup CDR2</th><th>Original Call 2 - CDR1</th><th>Pickup CDR2</th></tr>
<tr><td>origCallTerminationOnBehalfOf</td><td>16</td><td>16</td><td>16</td><td>16</td></tr>
<tr><td>lastRedirectRedirectOnBehalfOf</td><td>0</td><td>16</td><td>0</td><td>16</td></tr>
</table>
</td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.44 Do Not Disturb Ringer Off

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>Do Not Disturb Ringer Off</td></tr>
<tr><td>Description</td><td>Verify CDR on Application for calls with Do Not Disturb Ringer Off enabled</td></tr>
<tr><td>Test Setup</td><td>

Application Server and CUCM clusters integrated successfully
CUCM Global Parameter Settings
Local CUCM Phones – SCCP : 1 Phone; SIP : 1 Phone
Enable DND on Local SCCP Phone1
➢ Service Parameters➔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 0f Local SCCP Phone1➔associate template to the device & enable DND device parameters (check DND; DND Option: Ringer Off; DND Incoming Call Alert: Flash Only)
➢ Activate DND on Local SCCP Phone1 via the softkey

</td></tr>
<tr><td>Procedure</td><td>

Local SIP Phone 1 dials Local SCCP Phone1
Local SIP Phone 1 goes on-hook after 5s
Retrieve CDR from application & CUCM
Check the CDR fields

</td></tr>
<tr><td>Expected Results</td><td>

Local SCCP Phone 1 flashes to indicate incoming call
Local SIP Phone 1 hears a ringback tone
Call terminated by Local SCCP Phone 1
1 Record retrieved
Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, duration fields in the CDR table for each call

<table>
<tr><th>CDR field</th><th>DND CDR1</th></tr>
<tr><td>Orig_Cause_Value</td><td>16</td></tr>
<tr><td>duration</td><td>0</td></tr>
</table>
</td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.45 Do Not Disturb Call Reject enabled

Test Case Details	
Title	Do Not Disturb Call Reject enabled
Description	Verify CDR on Application for calls with Do Not Disturb Call Reject enabled
Test Setup	• Application Server and CUCM clusters integrated successfully • CUCM Global Parameter Settings • Local CUCM Phones – SCCP : 2 Phones; SIP : 1 Phone • Enable DND on Local SIP Phone1 • Service Parameters➔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 0f Local SIP Phone1➔associate template to the device & enable DND device parameters (check DND; DND Option: Call Reject; DND Incoming Call Alert: Beep Only) • Activate DND on Local SIP Phone1 via the soft key
Procedure	• Local SCCP Phone 1 dials Local SIP Phone1 • Local SIP Phone 1 answers call • Local SCCP Phone 2 dials Local SIP Phone 1 • Local SIP Phone 1 hits the “DND” softkey during connected state • Local SCCP Phone 2 goes on-hook • Local SCCP Phone 1 goes on-hook after 130s • Retrieve CDR from application & CUCM • Check the CDR fields
Expected Results	• Call established between Local SCCP Phone 1 & Local SIP Phone 1 (talking state) • Local SIP Phone 1 hears the ringing of 2nd incoming call • CUCM rejects call with Reason:User Busy • Local SIP Phone 1 hears a beep for the 2nd call that was rejected • Call terminated by Local SCCP Phone 2 • 1st call terminated normally • 2 Records retrieved • Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, Dest_Cause_Value, duration fields in the CDR table for each call
	CDR field | **DND CDR1** | **DND CDR2** Dest_Cause_Value | 0 | 17
Observations	PASS

5.3.46 iDivert

<table>
<tr><th colspan="2">Test Case Details</th></tr>
<tr><td>Title</td><td>iDivert</td></tr>
<tr><td>Description</td><td>Verify CDR on Application for a call with iDivert activated</td></tr>
<tr><td>Test Setup</td><td>• Application Server and CUCM clusters integrated successfully
• CUCM Global Parameter Settings
• Local CUCM Phones – SCCP : 1 Phone; SIP : 2 Phones ;PSTN :1 Phone
• VM enabled on all phones
• Enable iDivert on Local SIP Phone 2
➢ Legacy Immediate Divert Service Parameter Set to True
➢ Configure 4 Service parameters to the appropriate setting (Refer to Administration Guide)
➢ Device➔Device Settings > Softkey Template, add iDivert to template (Connected, On Hold, and Ring states)
➢ Device➔Phone➔DN of Local SIP Phone 2 ➔associate template to the device</td></tr>
<tr><td>Procedure</td><td>• Local SIP Phone 1 dials Local SIP Phone 2
• Local SIP Phone 2 hits the “iDivert” softkey during ringing state
• Local SIP Phone 1 leaves a voicemail and goes on-hook
• Local SCCP Phone 1 dials Local SIP Phone 2
• Local SIP Phone 2 hits the “iDivert” softkey during ringing state
• Local SCCP Phone 1 goes on-hook without leaving a message
• PSTN Phone 1 dials Local SIP Phone 2
• Local SIP Phone 2 hits the “iDivert” softkey during ringing state
• PSTN Phone 1 leaves a voicemail and goes on-hook
• Retrieve CDR from application & CUCM
• Check the CDR fields</td></tr>
<tr><td>Expected Results</td><td>• Local SIP Phone 1 directed to Local SIP Phone 2 voicemail box
• Local SCCP Phone 1 directed to Local SIP Phone 2 voicemail box
• PSTN Phone 1 directed to Local SIP Phone 2 voicemail box
• All calls terminated normally
• Local SIP Phone 2 was able to retrieve voicemails from 3 parties
• 3 Records retrieved
• Match the callingPartyNumber, OriginalCalledPartyNumber, origCalledPartyRedirectOnBehalfOf, finalCalledPartyNumber, lastRedirectRedirectReason duration fields in the CDR table for each call
<table>
<tr><th>CDR field</th><th>Call-1 CDR</th><th>Call-2 CDR</th><th>Call-3 CDR</th></tr>
<tr><td>origCalledPartyRedirectOnBehalfOf</td><td>14</td><td>14</td><td>14</td></tr>
<tr><td>lastRedirectRedirectReason</td><td>50</td><td>50</td><td>50</td></tr>
</table></td></tr>
<tr><td>Observations</td><td>PASS</td></tr>
</table>

5.3.47 iDivert is activated on a connected call

Test Case Details	
Title	iDivert is activated on a connected call
Description	Verify CDR on Application when iDivert is activated on a connected call
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Local CUCM Phones – SCCP : 2 Phones; SIP : 1 Phone ; • VM enabled on all phones' VM Pilot:#### • Enable iDivert on Local SIP Phone 1 ➢ Legacy Immediate Divert Service Parameter Set to True ➢ Configure 4 Service parameters to the appropriate setting (Refer to Administration Guide) ➢ Device➔Device Settings > Softkey Template, add iDivert to template (Connected, On Hold, and Ring states) ➢ Device➔Phone➔DN of Local SIP Phone 1➔associate template to the device
Procedure	• Local SCCP Phone 1 dials Local SIP Phone 1 • Local SIP Phone 1 answers • Local SIP Phone 1 hits the "iDivert" softkey after 60s (Connected state) • Local SCCP Phone 1 leaves a voicemail and goes on-hook • Retrieve CDR from application & CUCM • Check the CDR fields
Expected Results	• Call established between Local SCCP Phone 1 & Local SIP Phone 1 (talking state) • Local SCCP Phone 1 directed to Local SIP Phone 1's voicemail box • Call terminated normally • Call terminated normally • 2 CDR Records retrieved • Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn,origCalledPartyRedirectOnBehalfOf , lastRedirectRedirectReason , lastRedirectRedirectOnBehalfOf , destCallTerminationOnBehalfOf , joinOnBehalfOf ,duration fields in the CDR table for each call
Observations	**PASS**

CDR field	**Original Call-1 CDR**	**Diverted Call-1 CDR**
origCalledPartyRedirectOnBehalfOf	0	14
lastRedirectRedirectReason	0	50
lastRedirectRedirectOnBehalfOf	0	14
destCallTerminationOnBehalfOf	14	14
joinOnBehalfOf	0	14

5.3.48 Calls originating & terminating to a softphone (IP Communicator)

Test Case Details	
Title	Calls originating & terminating to a softphone (IP Communicator)
Description	Verify CDR on Application for calls originating & terminating to a softphone (IP Communicator)
Test Setup	• Application Server and CUCM clusters integrated successfully • **CUCM Global Parameter Settings** • Local CUCM Phones – SCCP : 1 Phone; SIP : 1 Phone ; CIPC: 2 Clients
Procedure	• CIPC 1 dials CIPC 2 • Local SCCP phone 1 dials CIPC 1 • CIPC 2 dials Local SIP Phone 1 • Calling and Called party goes on-hook alternatively • Retrieve CDR from application & CUCM • Check the CDR fields
Expected Results	• 3 calls established • 3 calls terminated • 3 Records retrieved • Match the callingPartyNumber, OriginalCalledPartyNumber, finalCalledPartyNumber, lastRedirectDn, origCause_Value, duration fields in the CDR table for each call CDR field / Call 1 / Call 2 / Call 3 origCause_Value / 16 / 16 / 16
Observations	PASS

CDR field	Call 1	Call 2	Call 3
origCause_Value	16	16	16

5.3.49 Verify the application can detect periods of time when no call records are generated

Title	Verify the application can detect periods of time when no call records are generated
Description	Verify the application can detect periods of time when no call records are generated
Test Setup	• Application Server, CUCM cluster and SFTP/FTP Server integrated successfully • CUCM CDR dump set to 1 min
Procedure	• Do not generate any calls for at least twice the time period between successive exports to the application • 2. Check application if there were any CDRs generated during this idle call period
Expected Results	• No new CDRs in application during idle time
Observation	PASS

Section: Voice Recording Tests

Global Prerequisites:

- Call Recorder Application Server and CUCM integrated successfully
- Call Recorder Application UserID: recorduser /ciscopsdt /ciscopsdt
- SIP Trunk registered with Call Recorder Server
- Recording Profile: Device➔Device Settings➔Recording Profile➔Add New (record_profile)➔DN:9999
- Set CUCM Global Parameters :
  - ➢ System➔Service Parameters➔Built-in-Bridge➔On
  - ➢ System➔Service Parameter➔SIP Expiry Timer➔300000
  - ➢ System➔Location➔Region➔G722 & iLBC Codec Disabled if Recorder does not support these Codecs
  - ➢ System➔Service Parameters➔Feature-Recording
    - o Play Recording Notification Tone to Observed Target➔True
    - o Play Recording Notification Tone to Connected Target➔True

5.3.50 Intra-Cluster Calls

Test Case Details	
Title	Intra-Cluster Calls
Description	Verify Automatic Recording for Intra-Cluster calls
Test Setup	• Local CUCM ➔Sip phone 1 and Sip phone 2; • Associate Sip phone 1 to Call Recorder Application User • Enable Automatic Recording on Sip phone 1; ➢ Device➔Phone➔DN➔Line: Sip phone 1 Recording Option➔ Automatic Recording Enabled Recording Profile➔record profile Recording Media Source➔Phone Preferred
Procedure	• Sip phone 1 dials Sip phone 2 • After 30 seconds, Sip phone 1 goes on hook • Retrieve Recording from Application • Check Selected fields and playback recorded audio file
Expected Results	• Intra Cluster Call established (talking state) • Recording triggered for call • Call terminated normally • 1 Record retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS

5.3.51 Inter Cluster Calls

Test Case Details	
Title	Automatic Recording for Inter-Cluster calls
Description	Verify Automatic Recording for Inter-Cluster calls
Test Setup	• Local CUCM →Sip phone 1; Remote CUCM → Sip phone 2 ; • Associate Sip phone 1 to Call Recorder Application User • Enable Automatic Recording on Sip phone 1 & Sip phone 2 ➢ Device→Phone→DN→Line: ✚ Recording Option→ Automatic Recording Enabled ✚ Recording Profile→record_profile ✚ Recording Media Source→Gateway Preferred
Procedure	• Sip phone 1 dials Sip phone 2 • After 30 seconds, Sip phone 2 goes on hook • Sip phone 2 dials Sip phone 1 • After 30 seconds, Sip phone 1 goes on hook • Retrieve Recordings from application • Check selected fields & playback recorded audio files
Expected Results	• Inter –Cluster Call established (talking state) • Recording triggered for call • Calls terminated normally • 4 Records retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS Application can retrieve recording only from one publisher of the cluster. So the recording from Remote CUCM cannot be retrieved by the application.

5.3.52 Recording for Off-Net Calls

Test Case Details	
Title	Recording for Off-Net Calls
Description	Verify Automatic Recording for Off-Net calls
Test Setup	• Local CUCM→SIP: Sip phone 1 • PSTN 1; • Associate Sip phone 1 to Call Recorder Application User→ • Enable Automatic Recording on Sip phone 1; ➢ Device→Phone→DN→Line: ✚ Recording Option→ Automatic Recording Enabled ✚ Recording Profile→record_profile ✚ Recording Media Source→Gateway Preferred
Procedure	• Sip phone 1 dials PSTN 1 • After 30 seconds, PSTN 1 goes on hook

	• PSTN 1 dials Sip phone 1 • After 30 seconds, Sip phone 1 goes on hook • Retrieve Recordings from application • Check selected fields & playback recorded audio files
Expected Results	• Call is established between Sip phone 1 and PSTN 1 • Call is terminated normally • Call is established between PSTN 1 and Sip phone 1 • Call is terminated normally • 2 Recordings retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS

5.3.53 Recording disabled for a call to a busy line

Test Case Details	
Title	Recording disabled for a call to a busy line
Description	Verify Automatic Recording is disabled for a call to a busy line
Test Setup	• Local CUCM➔Sip phone 1, 2 and 3; PSTN 1; • Disable VM & CW on Sip phone 1 • Local CUCM➔Sip phone 1, 2 and 3; PSTN 1; • Disable VM & CW on Sip phone 1 • Associate Sip phone 1 and Sip phone 3 to Call Recorder Application User➔ • Enable Automatic Recording on Sip phone 1 and Sip phone 3 ➢ Device➔Phone➔DN➔Line: Recording Option➔ Automatic Recording Enabled Recording Profile➔record_profile Recording Media Source➔Gateway Preferred
Procedure	• Sip phone 1 dials Sip phone 2 • Sip phone 2 answers • Sip phone 3 dials Sip phone 1 • Sip phone 3 goes on-hook • Sip phone 1 goes on-hook after 120s • Sip phone 1 dials PSTN 1 • PSTN 1 answers • Sip phone 3 dials PSTN 1 • Sip phone 3 goes on-hook • Sip phone 1 goes on-hook after 120s • Retrieve Recordings from application • Check selected fields & playback recorded audio files
Expected Results	• Call established between Sip phone 1 and Sip phone 2 • Recording triggered for Call • Sip phone 3 hears busy tone • Recording is not triggered for Call-2 and terminated normally • Call-1 terminated normally

	• Call established between Sip phone 1 and PSTN 1 • Recording triggered for Call • Sip phone 2 hears busy tone and recording is not triggered for Call-3 • Call 4 terminated normally • 2 Records retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS

5.3.54 Recording for Call Hold/Resume Intra-cluster call

Test Case Details	
Title	Recording for Call Hold/Resume Intra –cluster call
Description	Verify Automatic Recording for Call Hold/Resume Intra-Cluster Call
Test Setup	• Local CUCM➔Sip phone 1 and Sip phone 2; • Associate Sip phone 2 to Call Recorder Application User • Enable Automatic Recording on Sip phone 2 ➢ Device➔Phone➔DN➔Line: Recording Option➔ Automatic Recording Enabled Recording Profile➔record_profile Recording Media Source➔Phone Preferred
Procedure	• Sip phone 1 dials Sip phone 2 • Sip phone 2 answers • Sip phone 2 hits “Hold” softkey after 30s • Sip phone 2 hits “Resume” softkey after 30s • Sip phone 2 goes on-hook after 60s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 1 & Sip phone 2 (talking state) • Recording triggered for call • Sip phone 1 is placed on-hold (MOH) • Recording stopped • Sip phone 2 resumed call • Recording resumed • Call terminated normally • 2 Records retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS

5.3.55 Recording for Call Hold/Resume Inter-Cluster call

Test Case Details	
Title	Recording for Call Hold/Resume Inter-Cluster call
Description	Verify Automatic Recording for Call Hold/Resume Inter-Cluster call
Test Setup	• Enable Recording in Remote CUCM • Local CUCM➔Sip phone 1; Remote CUCM➔Sip phone 2; • Recording enabled on Voice Gateway • Associate Sip phone 1 and Sip phone 2 to Call Recorder Application User for the respective CUCM clusters • Enable Automatic Recording on Sip phone 1 and Sip phone 2; ➢ Device➔Phone➔DN➔Line: Recording Option➔ Automatic Recording Enabled Recording Profile➔record_profile Recording Media Source➔Gateway Preferred
Procedure	• Sip phone 1 dials Sip phone 2 • Sip phone 2 answers • Sip phone 1 hits "Hold" softkey after 30s • Sip phone 1 hits "Resume" softkey after 30s • Sip phone 1 goes on-hook after 60s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 1 & Sip phone 2 (talking state) • Recording triggered for call • Sip phone 2 is placed on-hold • Recording stopped • Sip phone 1 resumed call • Recording resumed • Call terminated normally • 4 Records retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS Application can retrieve recording only from one publisher of the cluster. So the recording from Remote CUCM cannot be retrieved by the application.

5.3.56 Recording for Call Hold/Resume Off-Net call

Test Case Details	
Title	Recording for Call Hold/Resume Off-Net call
Description	Verify Automatic Recording for Call Hold/Resume Off-Net call
Test Setup	• Local CUCM➔Sip phone 1 and PSTN 1 • Recording enabled on Voice Gateway • Associate Sip phone 1 to Call Recorder Application User • Enable Automatic Recording on Sip phone 1:

	➢ Device➔Phone➔DN➔Line: Recording Option➔ Automatic Recording Enabled Recording Profile➔record_profile Recording Media Source➔Gateway Preferred
Procedure	- Sip phone 1 dials PSTN 1 - PSTN 1 answers - Sip phone 1 hits "Hold" softkey after 30s - Sip phone 1 hits "Resume" softkey after 30s - Sip phone 1 goes on-hook after 60s - Retrieve recording from application - Check selected fields and playback recorded audio file
Expected Results	- Call established between Sip phone 1 & PSTN 1(talking state) - Recording triggered for call - PSTN 1 is placed on-hold - Recording stopped - Sip phone 1 resumed call - Recording resumed - Call terminated normally - 2 Records retrieved - Recorded fields matched call & audio playback is successful
Observations	PASS

5.3.57 Recording for a monitored device when CFA and CFNA enabled-Internal User

Test Case Details	
Title	Recording for a monitored device when CFA and CFNA enabled-Internal User
Description	Verify Automatic Recording for a monitored device when CFA and CFNA enabled
Test Setup	- Local CUCM➔Sip phone 1, 2, 3, and 4; - Enable CFA on Sip phone 2:Device➔Phone➔Sip phone 2➔CFA➔Sip phone 3 - Enable CFNA on Sip phone 3:Device➔Phone➔Sip phone 3➔CFNA➔Sip phone 4 - Associate Sip phone 3, Sip phone 4 to Call Recorder Application User➔ - Enable Automatic Recording on Sip phone 3, Sip phone 4 ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔Phone Preferred
Procedure	- Sip phone 1 dials Sip phone 2 - Sip phone 3 does not answer - Sip phone 4 answers

	• Sip phone 1 goes on-hook after 60s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call forwarded to Sip phone 3 and phone rings (CFA) • Call forwarded to Sip phone 4 and phone rings (CFNA) • Call established between Sip phone 1 & Sip phone 4 (talking state) • Recording triggered for call • Call terminated normally • 1 Record retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS

5.3.58 Recording for a monitored device when CFA and CFNA enabled – PSTN

Test Case Details	
Title	Automatic Recording for a monitored device when CFA and CFNA enabled- PSTN
Description	Verify Automatic Recording for a monitored device when CFA and CFNA enabled- PSTN
Test Setup	• Local CUCM→Sip phone 1, 2, and 3 ; PSTN • Enable CFA on Sip phone 2:Device→Phone→Sip phone 2→CFA→Sip phone 3 • Enable CFNA on Sip phone 3:Device→Phone→Sip phone 3→CFNA→PSTN • Associate Sip phone 1 to Call Recorder Application User→ • Enable Automatic Recording on Sip phone 1 ➢ Device→Phone→DN→Line: ✚ Recording Option→ Automatic Recording Enabled ✚ Recording Profile→record_profile ✚ Recording Media Source→Gateway Preferred
Procedure	• Sip phone 1 dials Sip phone 2 • Sip phone 3 does not answer • PSTN answers • Sip phone 1 goes on-hook after 60s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call forwarded to Sip phone 3 and phone rings (CFA) • Call forwarded to PSTN and phone rings (CFNA) • Call established between Sip phone 1 & PSTN (talking state) • Recording triggered for call • Call terminated normally • 1 Record retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS

5.3.59 Recording for a call with CFB enabled to an unmonitored IP phone

Test Case Details	
Title	Recording for a call with "CFB" enabled to an unmonitored IP phone
Description	Verify Automatic Recording for a call with "CFB" enabled to an unmonitored IP phone
Test Setup	• Local CUCM➔Sip phone 1, 2 and 3; PSTN 1; • Recording enabled on Voice Gateway • Enable CFB on Sip phone 3:Device➔Phone➔Sip phone 3➔CFB➔Sip phone 2 • Associate Sip phone 3 to Call Recorder Application User➔recorduser/ciscopsdt • Enable Automatic Recording on Sip phone 3: ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔Gateway Preferred
Procedure	• Sip phone 1 dials Sip phone 3 • Sip phone 3 answers • PSTN 1 dials Sip phone 3 • Sip phone 2 answers • Sip phone 2 goes on-hook after 60s • Sip phone 1 goes on-hook after 300s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 1 & Sip phone 3 (talking state) • Recording triggered for call 1 • Call 2 forwarded on busy to Sip phone 2 • Call established between PSTN 1 & Sip phone 2 (talking state) • Recording not triggered for call 2 • Calls terminated normally • 1 Record retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS

5.3.60 Recording for a Call Park call

Test Case Details	
Title	Recording for "Call Park" Call
Description	Verify Automatic Recording for a "Call Park" call
Test Setup	• Local CUCM➔SIP: Sip phone 1, Sip phone 2-Sip phone 3; • Call Park Code 3001:Routing➔Call Park➔3001 • Associate Sip phone 1 to Call Recorder Application User • Enable Automatic Recording on Sip phone 1 : Recording Option➔ Automatic Recording Enabled Recording Profile➔record_profile

	Recording Media Source➔Phone Preferred
Procedure	• Sip phone 1 dials Sip phone 2 • Sip phone 2 answers • Sip phone 2 hits “Park” soft key after 10s • Sip phone 3 dials park code:3001 after 10s • Sip phone 1 goes on-hook after 60s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 1 & Sip phone 2 (talking state) • Recording triggered for call • Sip phone 2 is parked • Recording stopped • Sip phone 3 picks up parked call • Call established between Sip phone 1 & Sip phone 3 (talking state) • Recording resumed • Calls terminated normally • 2 Records retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS

5.3.61 Recording for a Call Park Reversion call

Test Case Details	
Title	Recording for a “Call Park Reversion” call
Description	Verify Automatic Recording for a “Call Park Reversion” call
Test Setup	• Local CUCM➔SIP: Sip phone 1; PSTN; • Call Park Code 3001:Routing➔Call Park➔3001 • Call Park Reversion Timer Service Parameter➔60s • Recording enabled on Voice Gateway • Associate Sip phone 1 to Call Recorder Application User • Enable Automatic Recording on Sip phone 1: ➢ Device➔Phone➔DN➔Line: Recording Option➔ Automatic Recording Enabled Recording Profile➔record_profile Recording Media Source➔Gateway Preferred
Procedure	• PSTN dials Sip phone 1 • Sip phone 1 answers • Sip phone 1 hits “Park” soft key after 10s • Do not pick up the parked call for 60s • Sip phone 1 answers • Sip phone 1 goes on-hook after 30s • Retrieve recording from application • Check selected fields and playback recorded audio file

Expected Results	• Call established between PSTN & Sip phone 1 (talking state) • Recording triggered for call • PSTN is parked • Park Reversion Timer expired • Sip phone 1 is ringing • Call resumed between PSTN & Sip phone 1 (talking state) • Recording resumed for call • Call terminated normally • 2 records retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS

5.3.62 Recording for an Assisted Directed Call Park Call

Test Case Details	
Title	Recording for an "Assisted Directed Call Park" call
Description	Verify Automatic Recording for an "Assisted Directed Call Park" call
Test Setup	• Local CUCM→SIP: Sip phone 1, Sip phone 2, & Sip phone 3; • 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 2 BLF→DN:3011 (Retrieval Prefix 21) • Associate Sip phone 3 to Call Recorder Application User • Enable Automatic Recording on Sip phone 3: ➢ Device→Phone→DN→Line: ✚ Recording Option→ Automatic Recording Enabled ✚ Recording Profile→record profile Recording Media Source→Phone Preferred
Procedure	• Sip phone 1 dials Sip phone 3 • Sip phone 3 answers • Sip phone 1 hits "BLF" button (Assisted Directed Call Park) after 40s • Sip phone 1 goes on-hook • Sip phone 2 dials prefix: 21 after 5s to retrieve call when BLF is flashing • Sip phone 3 goes on-hook after 120s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 1 & Sip phone 3 (talking state)

	• Recording triggered for call • Sip phone 3 is parked • Recording stopped • Sip phone 2 retrieved directed parked call • Call established between Sip phone 2 & Sip phone 3 (talking state) • Recording resumed • Calls terminated normally • 2 Records retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS

5.3.63 Direct Transfer from a Shared line (SCCP)

Test Case Details	
Title	Direct Transfer from a shared line (SCCP)
Description	Verify Automatic Recording for Direct Transfer call from a shared line (SCCP)
Test Setup	• Local CUCM➔SCCP phone 1 and 2 and 3 ; PSTN; • Shared Line (SCCP phone 3's DN) on SCCP phone 2; • Associate SCCP phone 2 and 3 to Call Recorder Application User➔recorduser/ciscopsdt • Enable Automatic Recording on SCCP phone 1 and 3: ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔Gateway Preferred
Procedure	• SCCP phone 1 dials SCCP phone 2 • SCCP phone 2 answers • SCCP phone 2 place the call on hold and dials PSTN • PSTN answers • SCCP phone 2 hits "Direct Transfer" soft key & selects first call after 30s and hits "direct transfer" softkey . • SCCP phone 2 goes on-hook • PSTN goes on-hook after 120 s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between SCCP phone 1 and 2 • Recording triggered for Call 1 • SCCP phone 1 is placed on hold(MOH) • Recording stopped for Call 1 • Call established between SCCP phone 2 and PSTN • Recording triggered for Call 2 • SCCP phone 1 is direct transferred to PSTN

	• SCCP phone 2 terminates normally • Recording between SCCP phone 2 and PSTN ended • SCCP 1 and PSTN are in talking state • Recording resumed for final call • All calls terminated normally • 4 Recordings retrieved • Recorded fields matched calls and audio playback is successful
Observations	PASS

5.3.64 Recording for Consultative Transfer Call

Test Case Details	
Title	Recording for Consultative Transfer Call
Description	Verify Automatic Recording for “Consultative Transfer” Call
Test Setup	• Local CUCM➔SIP: Sip phone 1 & Sip phone 2; PSTN; • Recording enabled on Voice Gateway • Associate Sip phone 2 & Sip phone 1 to Call Recorder Application User➔recorduser/ciscopsdt • Enable Automatic Recording on Sip phone 2 & Sip phone 1: ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record profile ✚ Recording Media Source➔Gateway Preferred
Procedure	• Sip phone 2 dials PSTN • PSTN answers • Sip phone 2 hits “Transfer’ softkey after 40s • Sip phone 2 dials Sip phone 1 • Sip phone 1 answers • Sip phone 2 hits “Transfer” softkey after 40s • Sip phone 2 goes on-hook • Sip phone 1 goes on-hook after 120s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 2 & PSTN (talking state) • Recording triggered for call 1 • PSTN is placed On-Hold • Recording stopped for call 1 • Call established between Sip phone 2 & Sip phone 1 (talking state) • Recording triggered for consult call • PSTN is transferred to Sip phone 1 • Sip phone 2 terminated normally • Recording ended for consult call • PSTN & Sip phone 1 in talking state • Recording triggered for transferred call • Transferred call terminated normally

	• 4 Records retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS

5.3.65 Recording for Consultative Transfer to a Jabber Client

Test Case Details	
Title	Recording for Consultative Transfer to a Jabber Client
Description	Verify Automatic Recording for “Consultative Transfer” to a Jabber Client
Test Setup	• Local CUCM➔SIP: Sip phone 1; Jabber (Windows):Jabber 1 (Credentials: juser/123456) • PSTN; • Associate Sip phone 1 & Jabber 1 to Call Recorder Application User • Enable Automatic Recording on Sip phone 1, Jabber 1 : ➢ Device➔Phone➔DN➔Line: Recording Option➔ Automatic Recording Enabled Recording Profile➔record profile Recording Media Source➔Gateway Preferred
Procedure	• PSTN dials Sip phone 1 • Sip phone 1 answers • Sip phone 1 hits “Transfer’ softkey after 60s • Sip phone 1 dials Jabber 1 • Jabber 1 answers call • Sip phone 1 hits “Transfer” softkey after 60s • Sip phone 1 goes on-hook • Jabber 1 goes on-hook after 240 s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between PSTN & Sip phone 1 (talking state) • Recording triggered for call 1 • PSTN is placed On-Hold (MOH) • Recording stopped for call 1 • Call established between Sip phone 1 & Jabber 1 (talking state) • Recording triggered for call • PSTN transferred to Jabber 1 • PSTN & Jabber 1 in talking state • Recording triggered for transferred call • Sip phone 1 terminated normally • Final call terminated normally • 4 Records retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS

5.3.66 Recording for Inter-Cluster Consultative Transfer Call

Test Case Details	
Title	Recording for Inter–Cluster “Consultative Transfer” Call
Description	Verify Automatic Recording for Inter-Cluster “Consultative Transfer” Call
Test Setup	• Local Cluster➔SIP:Sip phone 1 & Sip phone 2; Remote Cluster➔SIP:Sip phone 3 • Recording enabled on Voice Gateway • Associate Sip phone 1 & Sip phone 3 to Call Recorder Application User➔recorduser/ciscopsdt on their respective CUCM Clusters • Enable Automatic Recording on Sip phone 1 & Sip phone 3: ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔Phone Preferred
Procedure	• Sip phone 1 dials Sip phone 3 • Sip phone 3 answers • Sip phone 3 hits “Transfer” softkey after 60s • Sip phone 3 dials Sip phone 2 • Sip phone 2 answers • Sip phone 3 hits “Transfer” softkey after 60s • Sip phone 1 goes on-hook after 120s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 1 & Sip phone 3 (talking state) • Recording triggered for call 1 • Sip phone 1 is on-hold (MOH) • Recording stopped for Sip phone 3 and resumed for Sip phone 1 • Call established between Sip phone 3 & Sip phone 2 (talking state) • Recording triggered for call 2 & stopped after transfer • Call established between Sip phone 1 & Sip phone 2 • Recording triggered for transferred call • All calls terminated normally • 4 Records retrieved • Recorded fields matched calls & audio playback is successful
Observations	PASS Application can retrieve recording only from one publisher of the cluster. So the recording from Remote CUCM cannot be retrieved by the application.

5.3.67 Recording for an Ad-Hoc Conference Call

Test Case Details	
Title	Recording for an Ad-Hoc “Conference” Call
Description	Verify Automatic Recording for an Ad-Hoc “Conference” Call
Test Setup	• Local CUCM➔SIP: Sip phone 1, Sip phone 2; PSTN; • Recording enabled on Voice Gateway • CUCM Service parameter: Drop Ad Hoc Conference ➔ Never (Default) • Associate Sip phone 1 to Call Recorder Application User • Enable Automatic Recording on Sip phone 1: ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔Gateway Preferred
Procedure	• Sip phone 2 dials Sip phone 1 • Sip phone 1 answers • Sip phone 1 hits “Conference” softkey after 30s • Sip phone 1 dials PSTN • PSTN answers • Sip phone 1 hits “Conference” softkey after 20s • All participants goes on-hook after 120s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 2 & Sip phone 1 (talking state) • Recording triggered for call 1 • Sip phone 2 is placed On-Hold (MOH) • Recording stopped for call 1 • Call established between Sip phone 1 & PSTN (talking state) • Recording triggered for call 2 • All 3 parties joined in a conference • Recording triggered for conference call • All participants left conference • All calls terminated normally • 3 Records retrieved • Recorded fields matched calls & audio playback is successful
Observations	PASS

5.3.68 Recording when a participant is dropped from a Conference Call

Test Case Details

Title	Recording when a participant is dropped from a “Conference” Call
Description	Verify Automatic Recording when a participant is dropped from a “Conference” Call
Test Setup	• Local CUCM➔SIP: Sip phone 1-Sip phone 2, Sip phone 3; PSTN; • Recording enabled on Voice Gateway • CUCM Service parameter: Drop Ad Hoc Conference ➔ Never (Default) • Associate Sip phone 1 & Sip phone 2 to Call Recorder Application User➔recorduser/ciscopsdt • Enable Automatic Recording on Sip phone 1 & Sip phone 2: ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔Gateway Preferred
Procedure	• Sip phone 1 dials Sip phone 3 • Sip phone 3 answers • Sip phone 3 hits “Conference” softkey after 60s • Sip phone 3 dials PSTN • PSTN answers • Sip phone 3 hits “Conference” softkey after 30s • Sip phone 3 hits “Conference” softkey after 60s • Sip phone 3 dials Sip phone 2 • Sip phone 2 answers • Sip phone 3 hits “Conference” softkey after 30s • Sip phone 3 hits “ConfList” softkey after 10s • Sip phone 3 selects Sip phone 1 from the Conference List • Sip phone 3 hits “Drop” • Sip phone 3 goes on-hook after 60s • Sip phone 2 goes on-hook after 140s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 1 & Sip phone 3 (talking state) • Recording triggered for call 1 • Sip phone 1 is on-hold (MOH) • Recording stopped for call 1 • Call established between Sip phone 3 & PSTN (talking state) • All 3 parties joined in a conference • Recording triggered for conference call • Sip phone 1 & PSTN are placed On-Hold (MOH) • Recording stopped for conference call • Call established between Sip phone 3 & Sip phone 2 (talking state) • Recording triggered for call 3 • All 4 parties joined in a conference • Recording resumed for conference call • Sip phone 1 dropped from conference • Sip phone 3 left conference

	• Sip phone 2 &PSTN connected directly • Recording triggered for call 4 • All calls terminated normally • 5 Records retrieved • Recorded fields matched calls & audio playback is successful
Observations	PASS 4 records retrieved

5.3.69 Recording when two Ad-Hoc Conference calls are joined

Test Case Details	
Title	Recording when two Ad-Hoc Conference calls are joined
Description	Verify Automatic Recording when two “Ad-Hoc Conference” calls are joined
Test Setup	• Local CUCM➔SIP: Sip phone 1, Sip phone 2-Sip phone 3; PSTN: PSTN 1, PSTN 2; • Recording enabled on Voice Gateway • CUCM Service parameter: Drop Ad Hoc Conference ➔ Never (Default) • Associate Sip phone 2 to Call Recorder Application User➔recorduser/ciscopsdt • Enable Automatic Recording on Sip phone 2 ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔Gateway Preferred
Procedure	• Sip phone 2 dials Sip phone 3 • Sip phone 3 answers • Sip phone 3 hits “Conference” softkey after 60s • Sip phone 3 dials PSTN 1 • PSTN 1 answers • Sip phone 3 hits “Conference” softkey after 30s • Sip phone 1 dials Sip phone 2 • Sip phone 2 answers incoming call • Sip phone 1 hits “Conference” after 50s • Sip phone 1 dials PSTN 2 • PSTN 2 answers • Sip phone 1 hits “Conference” softkey after 20s • Sip phone 2 selects conference 1 and hits the “Join” softkey • All participants ended conference after 300s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 2 & Sip phone 3 (talking state) • Recording triggered for call 1 • Sip phone 2 is placed on-hold (MOH) • Recording stopped

	• Call established between Sip phone 3 & PSTN 1 (talking state) • All 3 participants joined in conference 1 • Recording triggered for conference-1 call • Call established between Sip phone 1 & Sip phone 2 (talking state) • Recording triggered for call 3 • Sip phone 2 is on-hold (MOH) • Recording stopped • Call established between Sip phone 1 & PSTN 2 (talking state) • All 3 participants joined in conference 2 • Recording triggered for conference-2 call • Participants in conference 1 & 2 are joined • Recording resumed for joined conference • Conference call terminated normally by all parties • 5 Records retrieved • Recorded fields matched calls & audio playback is successful
Observations	PASS

5.3.70 Recording for a Meet-Me Conference Call

Test Case Details	
Title	Recording for a “Meet-Me” Conference Call
Description	Verify Automatic Recording for a “Meet-Me” Conference Call
Test Setup	• Local CUCM➔SIP:Sip phone 1, Sip phone 2-Sip phone 3; • Configure Meet-Me:Call Routing➔Meet-Me➔Add New➔3002 [meet-me #] • Create CTI_RP:Device➔CTI_RP➔DN:3002 • Assign Meet-Me CSS to Sip phone 1, Sip phone 2 & Sip phone 3:Device➔Phone➔DN➔Sip phone 1➔CSS➔css_mm • Associate Sip phone 1 to Call Recorder Application User • Enable Automatic Recording on Sip phone 1: ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record_profile Recording Media Source➔Phone Preferred
Procedure	• Sip phone 1 hits the “Meet-Me” softkey and dials 3002 • Sip phone 2 dials 3002 • Sip phone 3 dials 3002 • All 3 members go on-hook after 300s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Sip phone 1 initiated a meet-me conference • Sip phone 2 & Sip phone 3 joined the meet-me conference bridge port • All 3 parties in conference (talking state) • Recording triggered for Meet-Me Conference • Conference call terminated normally

	• 1 Record retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS

5.3.71 Recording for a monitored IP phone when Call Waiting is Active

Test Case Details	
Title	Recording for a monitored IP phone when “Call Waiting” is Active
Description	Verify Automatic Recording for a monitored IP phone when “Call Waiting” is Active
Test Setup	• Local CUCM➔SIP: Sip phone 1 & Sip phone 2; PSTN; • Associate Sip phone 2 to Call Recorder Application User • Enable Automatic Recording on Sip phone 2: ➢ Device➔Phone➔DN➔Line: Recording Option➔ Automatic Recording Enabled Recording Profile➔record_profile Recording Media Source➔ Gateway Preferred
Procedure	• Sip phone 2 dials Sip phone 1 • Sip phone 1 answers call • PSTN dials Sip phone 2 after 10s • Sip phone 2 answers incoming call immediately • Sip phone 2 terminated 2nd call after 120s • Sip phone 2 resumed 1st call • Sip phone 1 goes on-hook after 200s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 2 & Sip phone 1 (talking state) • Recording triggered for call 1 • Sip phone 1 is placed On-Hold • Recording paused for call 1 • Call established between Sip phone 2 & PSTN (talking state) • Recording triggered for call 2 • Call 2 terminated normally • Call 1 resumed • Recording for call 1 resumed • Call 1 terminated normally • 3 Records retrieved • Recorded fields matched calls & audio playback is successful
Observations	PASS

5.3.72 Recording for calls using G729 Codec

Test Case Details	
Title	Recording for calls using G729 Codec

Description	Verify Automatic Recording for calls using G729 Codec
Test Setup	• Local Cluster➔ SIP: Sip phone 1 & Sip phone 2; Remote Cluster➔SIP: Sip phone 3 & Sip phone 4; • Assign a Device Pool with Region Codec➔G729 to DN: Sip phone 4 & Sip phone 2 • Associate Sip phone 1 & Sip phone 3 to Call Recorder Application User➔recorduser/ciscopsdt • Enable Automatic Recording on Sip phone 1 & Sip phone 3: ➢ Device➔Phone➔DN➔Line: Recording Option➔ Automatic Recording Enabled Recording Profile➔record_profile Recording Media Source➔ Gateway Preferred
Procedure	• Sip phone 1 dials Sip phone 4 • Sip phone 4 answers call • Sip phone 3 dials Sip phone 2 • Sip phone 2 answers • Sip phone 4 goes on-hook after 120s • Sip phone 2 goes on-hook after 180s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 1 & Sip phone 4 • Recording triggered for call 1 • Call established between Sip phone 3 & Sip phone 2 • Recording triggered for call 2 • All calls terminated normally • 2 Records retrieved • Recorded fields matched calls & audio playback is successful
Observations	PASS Application can retrieve recording only from one publisher of the cluster. So the recording from Remote CUCM cannot be retrieved by the application.

5.3.73 Recording for calls using G722.1 Codec

Test Case Details	
Title	Recording for calls using G722.1 Codec
Description	Verify Automatic Recording for calls using G722.1 Codec
Test Setup	• Local Cluster➔ SIP: Sip phone 1 & Sip phone 2; Remote Cluster➔SIP: Sip phone 3 & Sip phone 4; • Assign a Device Pool with Region Codec➔G722 to DN:Sip phone 4 & Sip phone 1 • Associate Sip phone 2 & Sip phone 4 to Call Recorder Application User➔recorduser/ciscopsdt • Enable Automatic Recording on Sip phone 1, Sip phone 2 & Sip phone 4: ➢ Device➔Phone➔DN➔Line:

	✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔ Gateway Preferred
Procedure	• Sip phone 2 dials Sip phone 4 • Sip phone 4 answers call • Sip phone 3 dials Sip phone 1 • Sip phone 1 answers • Sip phone 4 goes on-hook after 120s • Sip phone 1 goes on-hook after 180s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 2 & Sip phone 4 • Recording triggered for call 1 • Call established between Sip phone 3 & Sip phone 1 • Recording triggered for call 2 • All calls terminated normally • 3 Records retrieved • Recorded fields matched calls & audio playback is successful
Observations	PASS Application can retrieve recording only from one publisher of the cluster. So the recording from Remote CUCM cannot be retrieved by the application.

5.3.74 Recording for a Call Hold/Resume call on a Shared line

Test Case Details	
Title	Recording for a “Call Hold/Resume” call on a shared line
Description	Verify Automatic Recording for a “Call Hold/Resume” call on a shared line
Test Setup	• Local CUCM➔SIP: Sip phone 1, Sip phone 2, Sip phone 3; • Shared line DN: Sip phone 2 is added to device with DN: Sip phone 1 • Recording enabled on Voice Gateway • Privacy on Phones with shared lines➔Off • Associate Sip phone 2 to Call Recorder Application User • Enable Automatic Recording on Sip phone 2, Sip phone 3: ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔ Phone Preferred
Procedure	• Sip phone 3 dials Sip phone 2 • Sip phone 2 answers • Sip phone 2 hits “Hold” softkey after 30s • Sip phone 2 hits “Resume” softkey after 30s • Sip phone 2 goes on-hook after 120s • Sip phone 2 dials Sip phone 3 • Sip phone 3 answers

	• Sip phone 3 hits “Hold” softkey after 30s • Sip phone 3 hits “Resume” softkey after 30s • Sip phone 2 goes on-hook after 120s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 3 & Sip phone 2 (talking state) • Recording triggered for call • Sip phone 3 is placed On-Hold (MOH) • Recording stopped for Sip phone 2 but continues for Sip phone 3 • Call resumed between Sip phone 2 & Sip phone 3 (talking state) • Recording resumed for Sip phone 2 • Call established between Sip phone 3 & Sip phone 2 (talking state) • Recording triggered for call • Sip phone 2 is placed On-Hold (MOH) • Recording stopped for Sip phone 3 but continues for Sip phone 2 • Call resumed between Sip phone 2 & Sip phone 3 (talking state) • Recording resumed for Sip phone 3 • All calls terminated normally • 6 Records retrieved • Recorded fields matched calls & audio playback is successful
Observations	PASS Application generates 2 records for the whole scenario where the hold/resume records are merged into one record.

5.3.75 Recording for MLPP Call

Test Case Details	
Title	Recording for “MLPP” Call
Description	Verify Automatic Recording for “MLPP” Call
Test Setup	• Local CUCM➔:SCCP phone 1- SCCP phone 3; • Configure MLPP domain: System➔MLPP➔Domain➔MLPP Domain➔ CAB001; • Configure MLPP on Phone devices: ➢ DN:SCCP phone 1 (MLPP Domain: CAB001; MLPP Indication:On; MLPP Premption: Forceful;) ➢ DN:SCCP phone 2 & Sip phone 3 (MLPP Domain: CAB001; MLPP Indication:On; MLPP Premption: Disabled;) • Configure partitions:exec ➔ css_exe; flash ➔css_flash; • Assign css_exec ➔ DN: SCCP phone 2; css_flash➔DN:SCCP phone 3; • Configure Translation Patterns: 90.SCCP phone 1 with partition:exec and MLPP Precedence: Executive Overwrite; 90.SCCP phone 1 with partition:flash and MLPP Precedence: Flash Overwrite; • Associate 8003 to Call Recorder Application User • Enable Automatic Recording on SCCP phone 1: ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Automatic Recording Enabled

	✚ Recording Profile➔record_profile ✚ Recording Media Source➔ Phone Preferred
Procedure	• SCCP phone 3 dials SCCP phone 1 (Flash Overwrite) • SCCP phone 1 answers • SCCP phone 2 dials SCCP phone 1 (Executive Overwrite) after 3s • SCCP phone 1 answers • SCCP phone 2 goes on-hook after 120s • Retrieve recording from application • Check selected fields and playback recorded audio/video file
Expected Results	• SCCP phone 1 receives special precedence ringback & display • Call established between SCCP phone 1 & SCCP phone 3 • SCCP phone 1 receives special precedence ringer and display • SCCP phone 1 active call is pre-empted with the executive override call • SCCP phone 1 answers the executive overwrite call • Call established between SCCP phone 1 & SCCP phone 2 (talking state) • Call from SCCP phone 3 terminated • SCCP phone 1 & SCCP phone 2 terminated normally after 120s • 2 Records retrieved • Recorded fields match calls and Video/Audio file playback is successful
Observations	PASS

5.3.76 Recording for calls originating and terminating to a softphone (Jabber for windows)

Test Case Details	
Title	Recording for calls originating and terminating to a soft phone(jabber)
Description	Verify Automatic Recording for calls originating and terminating to a soft phone(jabber)
Test Setup	• Local CUCM➔:SIP: Sip phone 1; SIP: Sip phone 2; Jabber: Jabber 1-Jabber 2 (Credentials:juser01/123456 & juser02/123456) • PSTN:PSTN; • 2 Windows PC with Jabber clients installed • Associate Jabber 1-Jabber 2 to Call Recorder Application User • Enable Automatic Recording on Jabber 1-Jabber 2 ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔ Gateway Preferred
Procedure	• Jabber 1 dials Jabber 2 (Duration=30s) • Sip phone 1 dials Jabber 2 (Duration=30s) • Jabber 1 dials Sip phone 2 (Duration=30s) • PSTN dials Jabber 2 (Duration=30s)

	• Calling and Called party goes on-hook alternatively • Retrieve recording from application • Check selected fields and playback recorded audio/video file
Expected Results	• 4 calls established • 4 calls terminated • 4 Records retrieved • Recorded fields match calls and Video/Audio file playback is successful
Observations	PASS

5.3.77 Recording for Extension Mobility Call

Test Case Details	
Title	Recording for Extension Mobility Call
Description	Verify Automatic Recording for Extension Mobility Call
Test Setup	• Local CUCM➔SIP:Sip phone 1-Sip phone 2, Sip phone 3; • Extension Mobility Service activated & started • Extension Mobility Service provisioned➔Device➔Device Settings➔Phone Service➔Add New➔EM • Create Virtual Device Profile: Device➔Device Settings➔Device Profile➔Add New➔EM_1054 with DN:EM user(DN) • Extension Mobility enabled on EM_1054 device profile • Extension Mobility Service subscribed on 1054 & EM_1054 device profile • Create User/PIN: psdtuser1/123456; Associate device profile EM_1054 to user under Extension Mobility; EMCC checked; • Recording enabled on Voice Gateway • Associate Sip phone 2 & Sip phone 3 to Call Recorder Application User➔recorduser/ciscopsdt • Enable Automatic Recording on Sip phone 2 & Sip phone 3: ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔ Phone Preferred
Procedure	• Sip phone 1 hits “Services” button and selects EM service • Sip phone 1 logs in with “psdtuser1/123456” • EM user(DN) dials Sip phone 2 • Sip phone 2 answers • Sip phone 2 goes on-hook after 120s • Sip phone 3 dials EM user(DN) • EM user(DN) answers • Sip phone 3 goes on-hook after 60s • EM user(DN) hits “Services” button and selects EM service • EM user(DN) logs out • Retrieve recording from application • Check selected fields and playback recorded audio file

Expected Results	• Login successful – phone rebooted with DN:EM user(DN) • 2 calls established (talking state) • Recording triggered for both calls • Calls terminated normally • 2 Records retrieved • Recorded fields matched calls & audio playback is successful
Observations	PASS

5.3.78 Recording for FAC and CMC Calls

Test Case Details	
Title	Recording for “FAC” and “CMC” Calls
Description	Verify Automatic Recording for “FAC” and “CMC” Calls
Test Setup	• Local Cluster➔SIP: Sip phone 1; PSTN: PSTN 1 & PSTN 2; • Configure CMC Code:3004# (Routing➔Client Matter Codes➔Add New) • Enable CMC 3004# (Call Routing➔Route/Hunt➔Route Pattern➔9.211222XXXX CMC checked) • Configure FAC Legal1; Code:3003#; Level=1 in CUCM (Routing➔Forced Authorization Codes➔Add New) • Enable FAC 3003# (Call Routing➔Route/Hunt➔Route Pattern➔9.212222XXXX check FAC checkbox) • Associate Sip phone 1 to Call Recorder Application User➔recorduser/ciscopsdt • Recording enabled on Voice Gateway • Enable Automatic Recording on Sip phone 1: ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔ Gateway Preferred
Procedure	• Sip phone 1 dials PSTN 1 • Sip phone 1 enters CMC Code 3004# • PSTN 1 answers call • PSTN 1 goes on-hook after 30s • Sip phone 1 dials PSTN 2 • Sip phone 1 enters FAC Code 3003# • PSTN 2 answers call • PSTN 2 goes on-hook after 30s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Sip phone 1 is prompted to enter CMC code • Call established between Sip phone 1 & PSTN 1 (talking state) • Call terminates normally • Sip phone 1 is prompted to enter FAC code • Call established between Sip phone 1 & PSTN 2 (talking state) • Call terminates normally • 2 Records retrieved

	• Recorded fields matched calls & audio playback is successful
Observations	PASS

5.3.79 Recording for Hunt Group Calls

Test Case Details	
Title	Recording for “Hunt Group” Calls
Description	Verify Automatic Recording for “Hunt Group” Calls
Test Setup	• Local Cluster➔SIP: Sip phone 1, Sip phone 2; • Hunt Group Pilot 3000 (1 member-Sip phone 2), queuing flag enabled, max. waiting timer=60s, route call to dest=Sip phone 4; • Associate Sip phone 1 to Call Recorder Application User • Recording enabled on Voice Gateway • Enable Automatic Recording on Sip phone 1: ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔ Gateway Preferred
Procedure	• Sip phone 1 dials 3000 • Sip phone 2 answers call • Sip phone 1 goes on-hook after 30s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call routed to hunt group member Sip phone 2 • Call established between Sip phone 1 & Sip phone 2 (talking state) • Call terminated normally • 1 Record retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS

5.3.80 Recording for Hunt Group calls when no members are available

Test Case Details	
Title	Recording for Hunt Group calls when no members are available
Description	Verify Automatic Recording for Hunt Group calls when no members are available
Test Setup	• Local Cluster➔SIP: Sip phone 1-Sip phone 2, Sip phone 3 and 4 • Hunt Group Pilot 3000 (1 member-Sip phone 3), queuing flag enabled, max. waiting timer=60s, route call to dest=Sip phone 4; • Associate Sip phone 1to Call Recorder Application User • Recording enabled on Voice Gateway • Enable Automatic Recording on Sip phone 1& Sip phone 2: ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record_profile

	✚ Recording Media Source→ Phone Preferred
Procedure	• Sip phone 1dials 3000 • Sip phone 3 answers call (Duration=180s) • Sip phone 2 dials 3000 • Sip phone 4 answers call • Sip phone 2 goes on on-hook after 60s • Sip phone 1goes on-hook after 180s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call routed to hunt group member Sip phone 3 • Call established between Sip phone 1& Sip phone 3 (talking state) • Hunt Group has no members available • Sip phone 2 routed to alternate DN: Sip phone 4 • Call established between Sip phone 2 & Sip phone 4 (talking state) • Both calls terminated normally • 2 Records retrieved • Recorded fields matched calls & audio playback is successful
Observations	PASS

5.3.81 Recording for calls between Encrypted IP phones

Test Case Details	
Title	Recording for calls between Encrypted IP phones
Description	Verify Automatic Recording for calls between Encrypted IP phones
Test Setup	• Create Device Security Profile for Encrypted Phone • Configure Enterprise Parameter: Cluster Security Mode→Mixed Mode (1) • Local CUCM→SIP: Sip phone 1 & Sip phone 2; (Both phones registered with encrypted configuration file) • Associate Sip phone 1 & Sip phone 2 to Call Recorder Application User • Enable Automatic Recording on Sip phone 1 & Sip phone 2: ➢ Device→Phone→DN→Line: ✚ Recording Option→ Automatic Recording Enabled ✚ Recording Profile→record_profile ✚ Recording Media Source→ Phone Preferred
Procedure	• Sip phone 1 dials Sip phone 2 • Sip phone 2 answers • Sip phone 1 goes on-hook after 60s • Sip phone 2 dials Sip phone 1 • Sip phone 1 answers • Sip phone 2 goes on-hook after 60s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 1 & Sip phone 2 (talking state)

	• Recording triggered for both calls • Both calls terminated normally • 2 Records retrieved • Recorded fields matched calls & audio playback is successful
Observations	N/S Application does not support Encrypted (Secure) Recording.

5.3.82 Recording for calls between Encrypted and Non Secure IP phones

Test Case Details	
Title	Recording for calls between Encrypted and Non Secure IP phones
Description	Verify Automatic Recording for calls between Encrypted and Non Secure IP phones
Test Setup	• Create Device Security Profile for Encrypted Phones • Configure Enterprise Parameter: Cluster Security Mode➔Mixed Mode (1) • Local CUCM➔SIP: Sip phone 1 & Sip phone 2; (Encrypted); SIP: Sip phone 3, Sip phone 4; PSTN:PSTN; • Recording enabled on Voice Gateway • Associate Sip phone 1, Sip phone 2, Sip phone 3, Sip phone 4 to Call Recorder Application User • Enable Automatic Recording on Sip phone 1, Sip phone 2, Sip phone 3 & Sip phone 4: ➢ Device➔Phone➔DN➔Line: Recording Option➔ Automatic Recording Enabled Recording Profile➔record_profile Recording Media Source➔ Gateway Preferred
Procedure	• Sip phone 1 dials Sip phone 3 • Sip phone 3 answers • Sip phone 1 goes on-hook after 120s • Sip phone 4 dials Sip phone 2 • Sip phone 2 answers • Sip phone 2 goes on-hook after 120s • Sip phone 1 dials PSTN • PSTN answers • PSTN goes on-hook after 120s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 1 & Sip phone 3 (talking state) • Call established between Sip phone 4 & Sip phone 2 (talking state) • Call established between Sip phone 1 & PSTN (talking state) • Recording triggered for all calls • Calls terminated normally • 5 Records retrieved • Recorded fields matched calls & audio playback is successful
Observations	N/S Application does not support Encrypted (Secure) Recording.

5.3.83 Recording for Group Pickup On-Net Calls

Test Case Details	
Title	Recording for Group Pickup On-Net Calls
Description	Verify Automatic Recording for Group Pickup On-Net Calls
Test Setup	• Local CUCM:SIP:Sip phone 1-Sip phone 6; • Group Pickup configured on all phones; Group: Sales (DN:Sip phone 3 & Sip phone 4); Group:TAC (DN:Sip phone 5 & Sip phone 6); ➢ Call Routing➔Call Pickup Group➔Add New➔Sales (DN:3105;Visual Alert; Calling & Called party checked) ➢ Call Routing➔Call Pickup Group➔Add New➔TAC (DN:3106;Visual Alert; Calling & Called party checked) • Associate Sip phone 1, Sip phone 4, Sip phone 5 to Call Recorder Application User • Enable Automatic Recording on Sip phone 1 , Sip phone 4 & Sip phone 5: ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔ Phone Preferred
Procedure	• Sip phone 2 dials Sip phone 6 • Sip phone 5 goes off-hook, hits “Group Pickup” softkey • Sip phone 5 enters TAC group_pickup DN:3106 • Sip phone 2 goes on-hook after 120s • Sip phone 1 dials Sip phone 3 • Sip phone 4 goes off-hook, hits “Group Pickup” softkey • Sip phone 4 enters Sales group_pickup DN:3105 • Sip phone 1 goes on-hook after 120s • Retrieve recording from application • Check selected fields and playback audio file
Expected Results	• Sip phone 6 in alerting state • Call is established between Sip phone 2 & Sip phone 5 (talking state) • Recording triggered for call • Call terminated normally • Sip phone 3 in alerting state • Call is established between Sip phone 1 & Sip phone 4 (talking state) • Recording triggered for call • Calls terminated normally • 2 Records retrieved • Selected fields matched calls & audio playback is successful
Observations	PASS

5.3.84 Recording Group Pickup Off-Net Calls

Test Case Details	
Title	Recording Group Pickup Off-Net Calls
Description	Verify Automatic Recording for Group Pickup Off-Net Calls
Test Setup	• Local CUCM➔ SIP: Sip phone 1-Sip phone 2, Sip phone 3-Sip phone 4; PSTN: PSTN 1 - PSTN 2; • Group Pickup configured on all phones; Group: Sales (DN: Sip phone 1 & Sip phone 2); Group: TAC (DN: Sip phone 3 & Sip phone 4); ➢ Call Routing➔Call Pickup Group➔Add New➔Sales (DN:3105;Visual Alert; Calling & Called party checked) ➢ Call Routing➔Call Pickup Group➔Add New➔TAC (DN:3106;Visual Alert; Calling & Called party checked) • Recording enabled on Voice Gateway • Associate Sip phone 2 & Sip phone 3 to Call Recorder Application User • Enable Automatic Recording on Sip phone 2 & Sip phone 3: ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔ Gateway Preferred
Procedure	• PSTN 1 dials Sip phone 4 • Sip phone 3 goes off-hook, hits “Group Pickup” softkey • Sip phone 3 enters TAC group_pickup DN:3106 • Sip phone 3 goes on-hook after 120s • PSTN 2 dials Sip phone 1 • Sip phone 2 goes off-hook, hits “Group Pickup” softkey • Sip phone 2 enters Sales group_pickup DN:3105 • Sip phone 2 goes on-hook after 120s • Retrieve recording from application • Check selected fields and playback audio file
Expected Results	• Sip phone 4 in alerting state • Call is established between PSTN 1 & Sip phone 3 (talking state) • Recording triggered for call • Call terminated normally • Sip phone 1 in alerting state • Call is established between PSTN 2 & Sip phone 2 (talking state) • Recording triggered for call • Call terminated normally • 2 Records retrieved • Selected fields matched calls & audio playback is successful
Observations	PASS

5.3.85 Selective User Recording for a call when media Source is Gateway preferred

Test Case Details

Title	Verify Selective User Recording for a call when Media Source➔Gateway Preferred
Description	Verify Selective User Recording for a call when Media Source➔Gateway Preferred
Test Setup	• Local CUCM: SIP: Sip phone 1-Sip phone 2; PSTN:PSTN 1; • Recording enabled on Voice Gateway • Associate Sip phone 1 and Sip phone 2 to Call Recorder Application User • Enable Selective Recording on Sip phone 1and Sip phone 2 ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Selective Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔ Gateway Preferred
Procedure	• Sip phone 1 dials Sip phone 2 • Sip phone 2 answers • Sip phone 2 hits “Record” soft key after 10s • Sip phone 2 goes on-hook after 120s • Sip phone 1 dials PSTN 1 • PSTN 1 answers • Sip phone 1 hits “Record” soft key after 10s • PSTN 1 goes on-hook after 120s • PSTN 1 dials Sip phone 1 • Sip phone 1 answers • Sip phone 1 hits “Record” soft key after 10s • Sip phone 1 goes on-hook after 120s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• 3 calls established (talking state) • Recording triggered for all 3 calls • All calls terminated normally • 3 Records retrieved • Recorded fields matched calls & audio playback is successful
Observations	PASS

5.3.86 Selective User Recording for a call when media Source is Phone preferred

Test Case Details

Title	Verify Selective User Recording for a call when Media Source➔Phone Preferred
Description	Verify Selective User Recording for a call when Media Source➔ Phone Preferred
Test Setup	• Local CUCM: SIP: Sip phone 1-Sip phone 2,; PSTN:PSTN 1 • Recording enabled on Voice Gateway • Associate Sip phone 1 and Sip phone 2 to Call Recorder Application User • Enable Selective Recording on Sip phone 1and Sip phone 2 ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Selective Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔ Phone Preferred
Procedure	• Sip phone 1 dials Sip phone 2 • Sip phone 2 answers • Sip phone 1 hits "Record" soft key after 10s • Sip phone 1 goes on-hook after 120s • Sip phone 1 dials PSTN 1 • PSTN 1 answers • Sip phone 1 hits "Record" soft key after 10s • PSTN 1 goes on-hook after 120s • PSTN 1 dials Sip phone 1 • Sip phone 1 answers • Sip phone 1 hits "Record" soft key after 10s • Sip phone 1 goes on-hook after 120s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• 3 calls established (talking state) • Recording triggered for all 3 calls • All calls terminated normally • 3 Records retrieved • Recorded fields matched calls & audio playback is successful
Observations	PASS

5.3.87 Selective User Recording for Inter-Cluster Calls

Test Case Details

Title	Selective User Recording for Inter-Cluster Calls
Description	Verify Selective User Recording for Inter-Cluster Calls
Test Setup	• Local CUCM: SIP: Sip phone 1; Remote Cluster: SIP: Sip phone 2; PSTN: PSTN 1; • Associate Sip phone 1 & Sip phone 2 to Call Recorder Application User • Enable Selective Recording on Sip phone 1 & Sip phone 2; ➢ Device➔Phone➔DN➔Line: Sip phone 1 & Sip phone 2 ✚ Recording Option➔ Selective Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔ Gateway Preferred
Procedure	• Sip phone 1 dials Sip phone 2 • Sip phone 2 answers • Sip phone 2 hits "Record" soft key after 10s • Sip phone 2 hits "Stop Rec" soft key after 60s • Sip phone 2 goes on-hook after 120s • Sip phone 1 dials PSTN 1 • PSTN 1 answers • Sip phone 1 hits "Record" soft key after 10s • Sip phone 1 hits "Stop Rec" soft key after 60s • PSTN 1 goes on-hook after 120s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 1 & Sip phone 2 (talking state) • Recording triggered for call • Sip phone 2 phone displays "recording" • Soft key label changed to "Stop Rec" • Recording stopped • Call terminated normally • Call established between Sip phone 1 & PSTN 1 (talking state) • Recording triggered for call • Sip phone 1 phone displays "Recording" • Soft key label changed to "Stop Rec" • Recording stopped • Call 2 terminated normally • 2 Records retrieved • Recorded fields matched and audio file playback is successful
Observations	PASS Application fetches record only from Primary cluster, so the first record is not supported.

5.3.88 Selective User Recording for Hold/Resume Call

Test Case Details	
Title	Selective User Recording for “Hold/Resume” Call
Description	Verify Selective User Recording for “Hold/Resume” Call
Test Setup	• Local CUCM➔SIP: Sip phone 1; Remote CUCM➔SIP: Sip phone 2; • Associate Sip phone 1 to Call Recorder Application User➔ • Enable Selective Recording on Sip phone 1; ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Selective Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔ Gateway Preferred
Procedure	• Sip phone 1 dials Sip phone 2 • Sip phone 2 answers call • Sip phone 1 hits “Record” soft key after 10s • Sip phone 2 hits the “Hold” soft key after 30s • Sip phone 2 hits the “Resume” soft key after 30s • Sip phone 1 goes on-hook after 120s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 1 & Sip phone 2 (talking state) • Recording triggered for call • Sip phone 1 is placed On-Hold (MOH) • Recording continues • Call resumed • Call terminated normally • 1 Record retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS

5.3.89 Selective User Recording for a call on a Shared line

Test Case Details	
Title	Selective User Recording for a call on a Shared line
Description	Verify Selective User Recording for a call on a Shared line
Test Setup	• Local CUCM➔SIP: Sip phone 1, Sip phone 2, Sip phone 3; • Shared line DN: Sip phone 2 on DN: Sip phone 1 • Associate Sip phone 2 to Call Recorder Application User➔ • Enable Selective Recording on Sip phone 2: ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Selective Recording Enabled ✚ Recording Profile➔record_profile

	✚ Recording Media Source➔ Gateway Preferred
Procedure	• Sip phone 3 dials Sip phone 2 • Sip phone 2 answers • Sip phone 2 hits the “Record” soft key after 10s • Sip phone 2 goes on-hook after 120s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 3 & Sip phone 2 (talking state) • Recording triggered for call • Sip phone 2 phone displays “Recording” • Soft key label changed to “Stop Rec” • Call terminated normally • 1 Record retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS

5.3.90 Selective User Recording for a Hold/Resume call on a Shared line

Test Case Details	
Title	Selective User Recording for a “Hold/Resume” call on a shared line
Description	Verify Selective User Recording for a “Hold/Resume” call on a shared line
Test Setup	• Local CUCM➔SIP: Sip phone 1-Sip phone 2, Sip phone 3, Sip phone 4; • Shared line DN: Sip phone 3 on Sip phone 2 & Sip phone 4 • Privacy on Phones with shared lines➔Off • Associate Sip phone 3 to Call Recorder Application User • Enable Selective Recording on Sip phone 3: ➢ Device➔Phone➔DN➔Line: ✚ Recording Option➔ Selective Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔ Gateway Preferred
Procedure	• Sip phone 1 dials Sip phone 3 • Sip phone 3 (on DN: Sip phone 2) answers • Sip phone 3 hits the “Record” soft key after 10s • Sip phone 3 hits the “Hold” soft key after 30s • Sip phone 3 (on DN: Sip phone 4) hits the “Resume” soft key after 30s • Sip phone 1 goes on-hook after 110s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Call established between Sip phone 1 & Sip phone 3 (talking state) • Recording triggered for call • Sip phone 3 phone displays “Recording” • Soft key label changed to “Stop Rec”

	• Sip phone 1 is placed On-Hold • Recording stopped • Call resumed on shared line in another device (talking state) • Recording resumed when call resumed • Call terminated normally • 1 Records retrieved • Recorded fields matched call & audio playback is successful
Observations	PASS

Functional Test: Voice Recording

5.3.91 Selective Silent Recording for a call when Media Source is Gateway Preferred

Test Case Details	
Title	Selective Silent Recording for a call when Media Source is Gateway Preferred
Description	Verify Selective Silent Recording for a call when Media Source is Gateway Preferred
Test Setup	• Local CUCM: SIP: Sip phone 1-Sip phone 2; PSTN: PSTN; • Recording enabled on Voice Gateway • Associate Sip phone 1, Sip phone 2 to Call Recorder Application User • Enable Selective Recording on Sip phone 1, Sip phone 2 : ➢ Device➔Phone➔DN➔Line: Sip phone 1 & Sip phone 2 ✚ Recording Option➔ Selective Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔ Gateway Preferred • Silent Recorder supported in the 3rd Party Recorder Application
Expected Results	• 3 calls established (talking state) • Recording triggered for all 3 calls • All calls terminated normally • 3 Records retrieved • Recorded fields matched calls & audio playback is successful
Observations	N/S Application does not support Selective Silent Recording.

5.3.92 Simultaneous Automatic Recording and Silent monitoring is supported for a call

Test Case Details	
Title	Simultaneous Automatic Recording and Silent monitoring is supported for a call

Description	Verify Simultaneous Automatic Recording and Silent monitoring is supported for a call
Test Setup	• Local CUCM: SIP: Sip phone 1-Sip phone 2; • Observed Party➔Sip phone 2; Caller➔Sip phone 1; • Recording enabled on Voice Gateway • Associate Sip phone 2 to Call Recorder Application User • Enable Automatic Recording on Sip phone 2: ➢ Device➔Phone➔DN➔Line: Sip phone 2 ✚ Recording Option➔ Automatic Recording Enabled ✚ Recording Profile➔record_profile ✚ Recording Media Source➔ Phone Preferred • Configure CTI-Enabled Desktop or Phone-Based Application for Silent Monitoring (Observer); • Assign Monitoring Calling Search Space to Observer's line: Device➔Phone➔DN➔Monitoring Search Space➔css_mon
Procedure	• Sip phone 1 dials Sip phone 2 • Sip phone 2 answers • Observer invokes silent monitoring session • Sip phone 1 goes on-hook after 120s • Retrieve recording from application • Check selected fields and playback recorded audio file
Expected Results	• Calls established between Sip phone 1 & Sip phone 2 (talking state) • Recording triggered for call • Silent monitoring call initiated successfully • Original & monitored call terminated normally • 1 Record retrieved • Recorded fields matched call & audio playback is successful
Observations	N/S Application does not support Selective Silent Recording

5.3.93 Verify the ability to search recordings using various key fields of a call

Test Case Details	
Title	Ability to search recordings using various key fields of a call
Description	Verify the ability to search recordings using various key fields of a call
Test Setup	Global Settings • Administration Guide & User Guides for 3rd party Recording Application
Procedure	• Launch a search tool from the CTI-enabled Recorder • Search Call Recordings using the following key fields: • Originating Number • Terminating Number • Start Date/Time • End Date/Time

	• Calling Name • Called Name • Device Name • Tag Name • Media Type (Audio or Video)
Expected Results	• Call Recorder search tool launches successfully • Call recordings retrieved successfully using the respective key fields
Observations	PASS

5.3.94 Verify the ability to edit, delete and download recordings using the Recorder Application

Test Case Details	
Title	Ability to edit, delete and download recordings using the Application
Description	Verify the ability to edit, delete and download recordings using the Application
Test Setup	Global Settings • Administration Guide & User Guides for 3rd party Application
Procedure	• Retrieve all existing recordings in the application • Select all recordings for DN: Sip phone 1 • Edit the first recording for DN: Sip phone 1 • Delete 2 recordings from DN: Sip phone 1 • Select the option to download 2 recordings to a location of your choice
Expected Results	• All call recordings retrieve successfully • All call recordings for DN: Sip phone 1 retrieved • 1st recording for DN: Sip phone 1 edited successfully • Two call recordings for DN: Sip phone 1 deleted successfully • Two call recordings for DN: Sip phone 1 downloaded successfully
Observations	PASS Recordings could not be edited in application.

5.3.95 Verify the administrator has the ability to manage the services via the application

Test Case Details	
Title	Administrator has the ability to manage the services via the application
Description	Verify Administrator has the ability to manage the services via the application
Test Setup	Global Settings • Administration Guide & User Guides for 3rd party Application
Procedure	Using the service management tool, start, stop & check status of the services running in the system
Expected Results	Administrator is able to start, stop and check the status of all the

	services running in the system
Observations	PASS

5.3.96 Silent Monitoring for a call Hold/Resume initiated by observed party

Test Case Details	
Title	Silent Monitoring for a call Hold/Resume initiated by observed party
Description	Verify Silent Monitoring for a call Hold/Resume initiated by observed party
Test Setup	• Local CUCM➔SIP: Sip phone 1 – Sip phone 2; Remote CUCM➔SIP:Sip phone 3; • Observed Party➔Sip phone 1; Caller➔Sip phone 1; • Configure CTI-Enabled Desktop or Phone-Based Application for Silent Monitoring (Observer); • Register Observer 1 & Observer 2 IP Communicator as follows: ➢ Observer 1➔MAC: SEP000000001012; TFTP IP; DN ; ➢ Observer 2➔MAC: SEP000000001013; TFTP IP; DN;
Procedure	• Sip phone 3 dials Sip phone 1 • Sip phone 1 answers • Observer invokes silent monitoring session • Sip phone 2 dials Sip phone 1 after 30s • Sip phone 1 hits the “Hold” soft key to answer incoming call • Sip phone 2 goes on-hook after 60s • Sip phone 1 hits the “Resume” soft key • Sip phone 3 goes on-hook after 180s
Expected Results	• Call established between Sip phone 1 & Sip phone 3 (talking state) • Silent monitoring call initiated successfully • Sip phone 3 is placed On-Hold • Media streaming for monitored call stopped • Call established between Sip phone 2 & Sip phone 1 (talking state) • Call between Sip phone 1 & Sip phone 2 terminated normally • Sip phone 1 resumed call • Media streaming for monitored call resumed • Original & monitored calls terminated normally
Observations	N/S Application does not support Selective Silent Recording.

5.3.97 Silent Monitoring for a call Hold/Resume initiated by an Observer

Test Case Details	
Title	Silent Monitoring for a call “Hold/Resume” initiated by an Observer
Description	Verify Silent Monitoring for a call “Hold/Resume” initiated by an Observer

Test Setup	• Local CUCM➔SIP:Sip phone 1; Remote CUCM➔SIP:Sip phone 2; • Observed Party➔Sip phone 1; Caller➔Sip phone 2; • Configure CTI-Enabled Desktop or Phone-Based Application for Silent Monitoring (Observer); • Assign Monitoring CSS to Observer's line: Device➔Phone➔9210 & 9211➔Line➔Monitoring Search Space➔css_mon • Observer 1 & Observer 2 IPC MAC: SEP000000001012 & SEP000000001013; TFTP IP; DN: XXXX & XXXX;
Procedure	• Sip phone 2 dials Sip phone 1 • Sip phone 1 answers • Observer invokes silent monitoring session • Observer hits the "Hold" softkey after 30s • Observer hits the "Resume" softkey after 30s • Sip phone 1 goes on-hook after 180s
Expected Results	• Call established between Sip phone 1 & Sip phone 2 (talking state) • Silent monitoring call initiated successfully • Monitored session is placed on-hold (Silent) • Media streaming for monitored session stopped • Monitored session is resumed • Media streaming for monitored session resumed • Original & monitored
Observations	N/S Application does not support Selective Silent Recording

Section: Wallboard Tests

Major Parameters to be verified in the Agent and Queue view of Wallboard reports:

Agent View:

- Agent State
- Call Duration
- Wrap-up reason
- Agents login IDs
- Calls Handled
- Reason for state change
- Agent Name
- Maximum Time in Ready state
- Maximum Time in Not Ready state
- Total Time in Not Ready state
- Total Time in Ready state
- Start time of the call
- Phone number
- Contact disposition

Queue View:

- Average Talk Duration
- Average Time in Ready state
- Average Time in Not Ready state
- Agent Utilization in Ready
- Agent Utilization in Not Ready
- Calls offered (regardless of agents answer or not)
- Calls Handled (calls answered by agent)
- Average Talk Time
- Max Talk Time
- Total Talk Time
- Hold Time
- Calls Abandoned
- Duration of the call
- Total calls in progress & Type of call (Inbound/Outbound)
- Name of CSQs
- Number of calls in queue for each CSQ (Call Waiting)
- Elapsed wait time for the oldest call in the queue (Longest Call in Queue)
- Agents Talking
- No. of agents logged in and available in the queue

5.3.98 Agent States

Test Case Details	
Title	Agent States
Description	Verify that application displays the different agent states
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the Application
Procedure	Change the state of the agent to the following state • Login • Logout • Not Ready • Ready • Reserved() • Talking
Expected Results	1. Application displays the following fields in either queue or agent display appropriately • Respective agent state • Time duration that the agent is in that state
Observations	PASS Login states are not updated in the "Group Real-time" column in the

	application. Even when the agent is logged in, the application shows as logged out. However, the agent states can be seen under "Agent States" in the wallboard.

5.3.99 Agent CSQ Statistics

Test Case Details	
Title	Agent CSQ Statistics
Description	Verify that application displays Agent CSQ Statistics
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the application
Procedure	1. Login the Agents to their respective queues 2. Make calls to the queues 3. Allow the call to ring in the queue
Expected Results	1. Application displays the following fields in either queue or agent display appropriately • Agents login IDs • Name of CSQs • Number of calls in queue for each CSQ (Call Waiting) • Elapsed wait time for the oldest call in the queue (Longest Call in Queue) • Agents Logged In • Agents Talking Verify other applicable major parameters referred in 6.2.1.1
Observations	PASS

5.3.100 Skill Based Routing - Agent Answered

Test Case Details	
Title	Skill Based Routing
Description	Verify that application displays Skill Based Routing Data Parameters

Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the aplication 6. Skill Based Routing is configured with IVR options
Procedure	1. Make a call into the system 2. Caller receives IVR options to choose 3. Select a specific skill and let the call route to the appropriate queue 4. Make no agents available to answer the call in the queue and caller wait 5. After sometime, Agent becomes ready and answers the call 5. Agent in the queue disconnects the call
Expected Results	1. Application displays the following fields in either queue or agent display appropriately • IVR parameters • Total call duration • IVR duration • Call duration between agent and caller - • Wait time for caller before agent answers • All the agents status in the skill group while caller is awaiting any agent to answer the call Verify other applicable major parameters referred in 6.2.1.1
Observations	PASS IVR duration and parameters are not shown.

5.3.101 Call Details

Test Case Details	
Title	Call Details
Description	Verify that application displays call details
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the Application
Procedure	1. Make a call to the queue 2. Agent 1 answers the call 3. Agent 2 makes an outbound call 4. Answer call from Agent 2 5. Disconnect the calls
Expected Results	1. Application displays the following fields in either queue or agent display appropriately • Start time of the call • Duration of the call

	• Type of call (Inbound/Outbound) • Phone number • Contact disposition • Queue • No. of agents available and logged in in the queue Verify other applicable major parameters referred in 6.2.1.1
Observations	PASS

5.3.102 Agent Statistics for Active Call

Test Case Details	
Title	Agent Statistics for Call
Description	Verify that application displays the agent statistics during calls
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the application
Procedure	1. Make a call to queue 1 2. Let the call keep ringing in the queue 3. Make a call to queue 2 4. Agent 1 answers the call 5. Make a call to queue 2 after few seconds 6. Agent 2 answers the call 7. Disconnect the calls 8. Make another call to queue 1 9. Agent 1 answers 10. Agent 1 holds the call and resumes after few seconds 11. Agent 1 disconnects the call
Expected Results	1. Application displays the following fields in either queue or agent display appropriately • Calls offered (regardless of agents answer or not) • Calls Handled (calls answered by agent) • Average Talk Time • Max Talk Time • Total Talk Time • Hold Time • No. of agents available and logged in in the queue • Calls Abandoned Verify other applicable major parameters referred in 6.2.1.1
Observations	PASS

5.3.103 Skill Based Routing – Agent Not Answered

Test Case Details

Title	Skill Based Routing
Description	Verify that Application displays the Skill Based Routing Data Parameters
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the Application 6. Skill Based Routing is configured with IVR options
Procedure	1. Make a call into the system 2. Caller receives IVR options to choose 3. Select a specific skill and let the call route to the appropriate queue 4. Make no agents available to answer the call in the queue 5. IVR announces there are no agent available 6. Call disconnects automatically after announcement
Expected Results	1. AUT displays the following fields in either queue or agent display appropriately • IVR parameters • Total call duration • IVR duration • Call duration between first IVR and second IVR • Wait time for caller before IVR answers • All the agents status in the skill group while caller is awaiting any agent to answer the call Verify other applicable major parameters referred in 6.2.1.1
Observations	**PASS** IVR parameters and duration of IVR are not displayed

5.3.104 Agent Statistics for Ready/Not Ready

Test Case Details	
Title	Agent Statistics for Ready/Not Ready
Description	Verify that Application displays the agent statistics during Ready/Not Ready status
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the application
Procedure	1. Keep an agent in Not Ready state for 60 sec 2. Change the state to Ready and keep it for 60 sec 3. Change the state to Not Ready again and keep it for 60 sec 4. Change the state to Ready again
Expected Results	1. Application displays the following fields in either queue or agent display appropriately -

	• Average Time in Ready state • Maximum Time in Ready state • Total Time in Ready state • Average Time in Not Ready state • Maximum Time in Not Ready state • Total Time in Not Ready state • Agent Utilization in Ready • Agent Utilization in Not Ready • No. of agents logged in and available in the queue Verify other applicable major parameters referred in 6.2.1.1
Observations	PASS Not Ready state can be seen in DND Duration Agent utilization in ready and Not ready states are not shown.

5.3.105 Agent State Change Reason

Test Case Details	
Title	Agent State Change Reason
Description	Verify that Application displays the reason for agents changing states
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the Application
Procedure	1. Change an agent from Login to Logout 2. Select the reason when logout 3. Change an agent from Ready to Not Ready 4. Select the reason when Not Ready
Expected Results	1. Application displays the following fields in either queue or agent display appropriately • Agent Name • Agent State • Reason for state change • No. of agents logged in and available in the queue Verify other applicable major parameters referred in 6.2.1.1
Observations	PASS

5.3.106 Agent Disconnect Call / Wrap-up

Test Case Details	
Title	Agent Disconnect Call / Wrap-up
Description	Verify that Application displays the Agent Disconnect and Wrap-up reason

Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the Application
Procedure	1. Login an Agent 2. Make a call to the queue 3. Agent answers the call 4. Agent disconnects the call 5. Agent enters Wrap-up reason
Expected Results	1. Application displays the following fields in either queue or agent display appropriately • Agent State • Call Duration • Wrap-up reason • No. of agents logged in and available in the queue • Calls Handled • Average Talk Duration Verify other applicable major parameters referred in 6.2.1.1
Observations	N/S Wrap up reason cannot be seen in the application.

5.3.107 Customer Disconnect Call / Wrap-up

Test Case Details	
Title	Customer Disconnect Call / Wrap-up
Description	Verify that application displays the Customer Disconnect and Wrap-up reason
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the Application
Procedure	1. Login an Agent 2. Make a call to the queue 3. Agent answers the call 4. Customer disconnects the call 5. Agent enters Wrap-up reason
Expected Results	1. Application displays the following fields in either queue or agent display appropriately • Agent State • Call Duration • Wrap-up reason Verify other applicable major parameters referred in 6.2.1.1
Observations	N/S

	Wrap up reason cannot be seen in the application

5.3.108 Agent Statistics for Call on Hold

Test Case Details	
Title	Agent Statistics for Call on Hold
Description	Verify that Application displays the agent statistics call on hold
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the application
Procedure	1. Make a call to queue 1 2. Let the call keep ringing in the queue 3. Make a call to queue 2 4. Agent 1 answers the call 5. Make a call to queue 2 after few seconds 6. Agent 2 answers the call 7. Hold the call by Agent 1 8. Hold the call by Agent 2 9. Resume the calls 10. Disconnect the calls 11. Make another call to queue 1 12. Agent 1 answers 13. Agent 1 holds the call and resumes after few seconds 14. Agent 1 disconnects the call
Expected Results	1. Application displays the following fields in either queue or agent display appropriately • Calls offered (regardless of agents answer or not) • Calls Handled (calls answered by agent) • Average Hold time • Maximum Hold time • Total Hold time Some of the above may be shown as reports, if not displayed on real time. Verify other applicable major parameters referred in 6.2.1.1
Observations	PASS

5.3.109 Agent Statistics for Alternate Call

Test Case Details	
Title	Agent Statistics for Alternate Call
Description	Verify that Application displays the agent statistics for alternate call
Test Setup	1. UCCX is installed

	2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the Application
Procedure	1. Make a call to queue 1 2. Agent 1 answers the call 3. Agent 1 consults Agent 2 by extension 4. Alternate the call (Agent 1 in connection with the Caller) 5. Alternate the call (Agent 1 in connection with Agent 2) 6. Alternate the call (Agent 1 in connection with the Caller) 7. Agent 2 disconnects 8. Agent 1 disconnects the call
Expected Results	1. Application displays the following fields in either queue or agent display appropriately • Agent State Change • Call Duration • Hold count Verify other applicable major parameters referred in 6.2.1.1
Observations	PASS Hold count can be seen by adding Real time values (Basic receiving Agent Event) in the wallboard.

5.3.110 Call Abandon

Test Case Details	
Title	Call Abandon
Description	Verify that Application displays the call abandoned when ringing in the queue
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the Application
Procedure	1. Make a call to queue 1 2. After few rings, disconnect the call before agent answers
Expected Results	1. Application displays the following fields in either queue or agent display appropriately • Call Waiting in the Queue • Call Abandoned by Caller Verify other applicable major parameters referred in 6.2.1.1
Observations	PASS

5.3.111 Agent to Agent Blind Transfer

Test Case Details

Title	Agent to Agent Blind Transfer
Description	Verify that Application displays the Agent to Agent Blind Transfer
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the Application
Procedure	1. Make a call to queue 1 2. Agent 1 answers the call 3. Agent 1 blind transfers the call via route point to Agent 2 4. Agent 2 answers the call 5. Agent 2 disconnects the call
Expected Results	1. Application displays the following fields in either queue or agent display appropriately • Agents state • Blind transfer occurrence • Call duration with each Agent Verify other applicable major parameters referred in 6.2.1.1
Observations	PASS Blind transfer occurrence is not shown in application.

5.3.112 Agent to Agent Consult Transfer

Test Case Details	
Title	Agent to Agent Consult Transfer
Description	Verify that Application displays the Agent to Agent Consult Transfer
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the application
Procedure	1. Make a call to queue 1 2. Agent 1 answers the call 3. Agent 1 consult transfers the call via route point to Agent 2 4. Agent 1 completes the transfer 5. Agent 2 disconnects the call
Expected Results	1. Application displays the following fields in either queue or agent display appropriately • Agents state • Consult transfer occurrence -N • Call duration with each Agent Verify other applicable major parameters referred in 6.2.1.1
Observations	PASS Consult transfer occurrence is not shown in the application.

5.3.113 Agent to Agent Consultation

Test Case Details	
Title	Agent to Agent Consultation
Description	Verify that Application displays the Agent to Agent Consultation
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the Application
Procedure	1. Make a call to queue 1 2. Agent 1 answers the call 3. Agent 1 presses transfer button and calls Agent 2 via direct call 4. Agent 2 answers the call 5. After few seconds, Agent 2 disconnects the call 4. Agent 1 resumes the call with the caller 5. Agent 1 disconnects the call
Expected Results	1. Application displays the following fields in either queue or agent display appropriately • Agents state • Call duration between Agents • Call duration between Agent and Caller • Hold duration Consultation call count should not be populated in the total queue calls. Verify other applicable major parameters referred in 6.2.1.1
Observations	**PASS**

5.3.114 Agent to Agent Consultation and Caller Drop

Test Case Details	
Title	Agent to Agent Consultation and Caller Drop
Description	Verify that Application displays the Agent to Agent Consultation and Caller Drop
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the application
Procedure	1. Make a call to queue 1 2. Agent 1 answers the call 3. Agent 1 presses transfer button and calls Agent 2 via direct call 4. Agent 2 answers the call 5. Caller drops the call

	4. Agent 1 disconnects the call
Expected Results	1. Application displays the following fields in either queue or agent display appropriately • Agents state • Call duration between Agents • Call duration between Agent and Caller • Hold duration • Caller Drop while Agent 1 talking to Agent 2 Consultation call count should not be populated in the total queue calls. Verify other applicable major parameters referred in 6.2.1.1
Observations	PASS When the caller drops the call, the phone icon is disappeared near Agent which means the caller is dropped.

5.3.115 Agent Outbound Call and Hold

Test Case Details	
Title	Agent Outbound Call and Hold
Description	Verify that Application displays the agent outbound call and hold
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the Application
Procedure	1. Agent makes an outbound call 2. Caller answers 3. Agent 1 places the call on hold 4. Agent 1 retrieves the call 5. Agent 1 disconnects the call
Expected Results	1. Application displays the following fields in either queue or agent display appropriately • Agents state • Hold duration • Call Type • Call Duration • Wrap-up reason Verify other applicable major parameters referred in 6.2.1.1
Observations	PASS

5.3.116 Agent Outbound Call and Consultation/Reconnect

Test Case Details	
Title	Agent to Agent Consultation and Reconnect

Description	Verify that application displays Agent to Agent Consultation and Reconnect
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the Application
Procedure	1. Agent makes an outbound call 2. Caller answers 3. Agent 1 presses transfer button and calls Agent 2 via direct call 4. Agent 2 answers the call 5. Agent 1 reconnects the call 6. Agent 1 disconnects the call
Expected Results	1. Application displays the following fields in either queue or agent display appropriately • Agents state • Call duration between Agents • Call duration between Agent and Caller • Hold duration Verify other applicable major parameters referred in 6.2.1.1
Observations	**PASS** Call duration between Agent and caller is shown in Outbound Talking Duration field. It is paused when the agent holds the outbound call and the duration resumed when the call is resumed.

5.3.117 Add new agents to the queue and remove agents from the queue

Test Case Details	
Title	Add new agents to the queue and remove agents from the queue
Description	Verify that Application displays the data dynamically when agents are added and removed from a queue
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the Application
Procedure	1. UCCX Admin logs in to UCCX 2. Admin adds a new agent to the existing queue which is monitored by Wallboard 3. Admin removes an agent from the queue which is monitored by wallboard.
Expected Results	1. Application displays the queue and agent data with the updated UCCX changes dynamically. Verify other applicable major parameters referred in 6.2.1.1
Observations	**PASS**

	In Application, we need to perform a refresh users and groups.

5.3.118 Agent session ended forcefully

Test Case Details	
Title	Agent session ended forcefully
Description	Verify that Application displays the agent details properly when the session is ended forcefully
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 Queues 4. Each Queue is configured with minimum 1 Agent 5. Queue view and Agent view are selected in the Application
Procedure	1. Agent is logged in UCCX 2. User can able to see the agent details in the wallboard 3. Agent console is closed forcefully by killing the task from task manager which ends the session in UCCX
Expected Results	1. AUT displays the following fields in either queue or agent display appropriately • Agents state Verify other applicable major parameters referred in 6.2.1.1
Observations	PASS

5.4 Phase 3 - Negative Tests

5.4.1 Verify the CDR collection after a PUB failure recovery

Title	Verify the CDR collection after a PUB failure recovery
Description	Verify Application CDR collection after a PUB failure recovery at local site
Test Setup	• Application Server and CUCM integrated successfully
Procedure	• Make a call from SIP Phone 1 to SCCP Phone 1 and disconnect the call after 30 seconds. • Check the application is collecting CDRs • Make a call from SIP Phone 1 to SCCP Phone 1. • While the call is in progress, Login to local CUCM PUB • Restart the CUCM PUB • Disconnect the call after 120 seconds • Check if PUB is up and running. • Make a call from SIP Phone 1 to SCCP Phone 1 and disconnect the call after 30 seconds. • Check CDR collection when PUB is restored
Expected Results	• application is collecting CDRs for the completed call(Call 1) • CUCM PUB is restarted • application reports failure on connectivity to CUCM

	application reports failure recovery once the CUCM PUB is restored CDR is restored application is collecting CDRs of Call 2 &3 once the CUCM PUB is restored
Observations	PASS

5.4.2 Verify the CDR collection after a SUB failure recovery

Title	Verify the CDR collection after a SUB failure recovery
Description	Verify Application CDR collection is not affected by a SUB failure at local site
Test Setup	Application Server and CUCM integrated successfully
Procedure	Make a call from SIP Phone 1 to SCCP Phone 1 and disconnect the call after 30 seconds. Check the application is collecting CDRs Make a call from SIP Phone 1 to SCCP Phone 1. While the call is in progress, Login to local CUCM SUB Restart the CUCM SUB Disconnect the call after 120 seconds Check if SUB is up and running. Make a call from SIP Phone 1 to SCCP Phone 1 and disconnect the call after 30 seconds. Check CDR collection during SUB failure & restore
Expected Results	application is collecting CDRs for the completed call(Call 1) CUCM SUB is restarted application reports failure on connectivity to CUCM SUB CDR collection should not affected by SUB failure application is collecting CDRs of Call 2 & 3
	PASS

5.4.3 Verify the application recovers after a Network Failure

Title	Verify the application recovers after a Network Failure
Description	Verify the application recovers after a Network Failure
Test Setup	Application Server and CUCM integrated successfully
Procedure	Make a call from SIP Phone 1 to SCCP Phone 1 and disconnect the call after 30 seconds. Check the application is collecting CDRs Make a call from SIP Phone 1 to SCCP Phone 1. While the call is in progress, Bring down the network of the Application Disconnect the call after 120 seconds Bring up the network of the Application and make sure the connectivity is restored. Make a call from SIP Phone 1 to SCCP Phone 1 and disconnect the

	call after 30 seconds. - Check CDR collection during Application Network Failure
Expected Results	- application is collecting CDRs for the completed call(Call 1) - Network Interface of Application is down - application is collecting CDRs of Call 2 & 3 once the network interface is up
Observations	PASS

5.4.4 Verify Application is not affected by a SUB Failure at local site

Test Case Details	
Title	Verify Application is not affected by a SUB failure at local site
Description	Verify Application is not affected by a SUB failure at local site
Test Setup	- Application Server and CUCM integrated successfully - Automated Traffic Generator setup at local cluster with automatic recording enabled on traffic endpoints
Procedure	- Open Traffic Commands: - Run Utility_Start_Traffic and set duration➔15 mins. - Note the traffic start & stop time - Check the VR application is collecting audio files - Open Negative Tests Command: - Run Utility_Restart_CUCM - Select “Subscriber” from the dropdown box - Hit “Run” - Check if CUCM-SUB is up by pinging its IP from your lab pc - Check audio file collection during SUB failure & recovered phase - Compare VR audio file count with call statistics
Expected Results	- Background traffic running for 15 mins - VR should not be affected by SUB failure - Application audio file count match with traffic call statistics for that period
Observations	PASS

Section: Wallboard Testing

5.4.5 Application Reboot

Test Case Details	
Title	Application Reboot

Description	Verify that when application reboots and comes up it displays the statistics
Test Setup	1. UCCX is installed 2. Application is configured with UCCX
Procedure	1. Make simultaneous calls to queues 2. During the call in the queues, reboot AUT 3. Application restores after reboot
Expected Results	1. Application displays the Queue and Agent statistics after restore
Observations	PASS

5.5 Phase-4: Load Testing

5.5.1 Baseline Test

Test Case Details	
Title	CUCM Baseline Measurement
Description	Verify CUCM stability by turning the application down
Test Setup	Local CUCM with 10 physical endpoints registered 500 users for backend load.
Procedure	• Turn off the Application server, and remove any third party application service from IP phones • Using the Traffic Generator, generate a backend call load with 500 users registered with CUCM • Using the Traffic Generator, generate a call load to 10 physical phones and play the RTP for 30 seconds • Disconnect the call at the end of 30 seconds. • Generate the load for one hour
Expected Results	• Load successfully runs for 1 hour. • Collect the call statistics from the Traffic Generator. • Collect the perfmon logs from CUCM for the load period. • Verify that CPU usage of CUCM never went above 80% and that average CPU usage never went above 60% and record the results. • Compare the performance logs with the Baseline statistics and ensure no major deviation is found in the values collected. Ensure no negative impact is found in CUCM with the 3rd party application integrated.
Observations	Pass

5.5.2 8 Hour Load run

Test Number	4.3
Test Name	8 Hour Load run
Purpose	Verify application runs smoothly during an 8 hrs. stability run

<table>
<tr><td>Pre-requisite</td><td>• Application Server, CUCM cluster and SFTP/FTP Server integrated successfully
• Auto registration enabled in CUCM
• CDRs collection enabled in CUCM
• Automated Traffic Generator setup in CUCM</td></tr>
<tr><td>Procedure</td><td>• Start traffic generator for 480 min
• Note the traffic start & stop
• Check if CDRs are collected by application
• CDRS are received by the application during 8 hrs. run
• Compare CDR call counts with call statistics in the command output window at the end of traffic run
• Verify CDRs were received without any loss or duplication
• Application CDR count matched with traffic generator call count for that period
• Check the CPU & Memory usage during this 8 hrs. period</td></tr>
<tr><td>Expected Results</td><td>• Traffic running
• CDRS are received by the application during 8 hrs. run
• CDRs were received without any loss or duplication
• Application CDR count matched with traffic generator call count for that period</td></tr>
<tr><td>Observation</td><td>PASS
<table>
<tr><th>CUCM</th><th>Partition(Active)%CPU Time(Avg)</th><th>Memory(%Mem used)</th><th>Processor (Total)%CPU Time(Avg)</th></tr>
<tr><td>CUCM Pub</td><td>40%</td><td>65%</td><td>56%</td></tr>
<tr><td>CUCM Sub1</td><td>40%</td><td>60%</td><td>40%</td></tr>
</table></td></tr>
</table>

5.5.3 Verify Application runs smoothly during an 8 hour stability run

Test Case Details	
Title	Verify application runs smoothly during an 8 hr stability run
Description	Verify application runs smoothly during an 8 hr stability run
Test Setup	• Application Server and CUCM integrated successfully • Automated Traffic Generator setup at local cluster with automatic recording enabled on traffic endpoints.
Procedure	• Open LoadRun Tests Commands: • Run_Load_Test and set duration→480 mins • Note the traffic start & stop time • Check audio files are collected by VR application • Compare VR audio file counts with call statistics • Check the CPU & Memory usage during this 8 hr period
Expected Results	• Traffic running for 8 hours • Audio files generated by active recording during 8 hr run • Audio recordings were received without any loss or duplication • Application audio file count match with traffic call statistics for that period

Observations	PASS

Section: Wallboard tests

5.5.4 Performance/Load Baseline

Test Case Details	
Title	Performance/Load Baseline
Description	Verify the test environment statistics with a baseline configuration and with Application installed
Test Setup	1. UCCX is installed 2. Application is configured with UCCX 3. UCCX is configured with minimum 3 queues and minimum 300 Agents per queue (total agents should be 1000)
Procedure	1. Simulate simultaneous calls to all the 3 queues for 1 hour 2. Configure 5 sec ring time, 30 sec talk time
Expected Results	1. Application displays the following fields in either queue or agent display appropriately all the applicable data parameters for all the calls 2. Application provides report with all the details 3. There is no impact to Performance of UCCX and the Application 4. Document CPU, Memory Usage
Observations	PASS

5.6 Phase 5 Third Party Specific Scenarios

5.6.1 Change Wallboard password

Test Case Details	
Title	Change Wallboard password
Description	Verify that Application allows to change Wall board admin password
Test Setup	1. UCCX is installed with redundancy 2. Application is configured with UCCX
Procedure	1. Log in to application as admin 2. Change Admin password 3. Re-log in to AUT with new password
Expected Results	1. Application displays the data in Wallboard for users without any interruption 2. Admin could able to login to Application with new credentials successfully and view the data
Observations	PASS

5.6.2 Re-Sync the Wallboard with UCCX

Test Case Details	
Title	Re-Sync the Wallboard with UCCX
Description	Verify that Application could Resync data with UCCX
Test Setup	1. UCCX is installed with redundancy 2. Application is configured with UCCX
Procedure	1. Log in to Application as admin 2. Resync application with UCCX
Expected Results	1. Application displays Proper message to the users during Re-Sync 2. Re-Sync completed Successfully and users can view the Wallboard data without Successfully.
Observations	PASS

5.6.3 Change queue and data parameter of the wallboard view

Test Case Details	
Title	Change queue and data parameter of the wallboard view
Description	Verify that Application could Change queue and data parameter of the wallboard view dynamically when the corresponding template is modified by the admin.
Test Setup	1. UCCX is installed with redundancy 2. Application is configured with UCCX
Procedure	1. Log in to Application as admin 2. Change the queue associated to a wall board template and update the fields displayed in the wallboard template
Expected Results	1. Users should view the changes in the wallboard data dynamically in the same session.
Observations	PASS

5.6.4 Remove a user session via admin user

Test Case Details	
Title	Remove a user session via admin user
Description	Verify that Application admin could remove an user session
Test Setup	1. UCCX is installed with redundancy 2. Application is configured with UCCX
Procedure	1. Log in to Application as admin 2. Remove a user session
Expected Results	1. Admin should able to see the number of session n users connected to the session 2. Admin should able to successfully remove the user session

	3. User should not able to view the wallboard once the session is ended by the admin
Observations	PASS

5.6.5 Rename and queue and parameter value to be displayed in the Wallboard

Test Case Details	
Title	Rename and queue and parameter value to be displayed in the Wallboard
Description	Verify that Application admin could rename a queue or parameter displayed in the wallboard
Test Setup	1. UCCX is installed with redundancy 2. Application is configured with UCCX
Procedure	1. Log in to Application as admin 2. Rename a queue and a parameter in the Wallboard template
Expected Results	1. Admin should able to rename the queue and parameter successfully 2. Modification in one template should not affect the other templates and UCCX data. 3. User should able to view the change in the Wallboard dynamically.
Observations	PASS

6 Glossary

The following list describes specific acronyms and definitions for terms used throughout this document:

ACD	Automatic Call Distributor. A device that distributes calls to agents based on administratively settable rules.
AUT	Application Under Test
BHCA	Busy Hour Calls Attempted.
BHCC	Busy Hour Calls Completed.
DID	Direct Inward Dialing
DNIS	Dial Number Identification Service, the telephone number being dialed,

	same as called party number.
DSP	Digital Signal Processor
E1	32 64kpbs timeslots on a 2.048 Mbps serial interface
ICS	Integrated Communication System.
IP	Internet Protocol
ISDN	Integrated Services Digital Network
IVR	Interactive Voice Response
IVT	Interoperability Verification Testing
LAN	Local Area Network
MCS	Media Convergence Server.
MCU	Multi-point Control Unit
PSTN	Plain Old Telephone Service
PRI	Primary Rate Interface: ISDN interface to 64kbps D channel plus 23 (T1) or 30 (E1)B channels for voice or data.
PSTN	Public Switched Telephone Network
RAS	Registration Admission Status
RTCP	Real Time Transport Control Protocol
RTP	Real Time Transport Protocol
T1	24 64kpbs timeslots on a 1.544 Mbps serial interface
UM	Unified Messaging. A voice mail system that includes fax and email capabilities.
Agent Phone	A client application typically used in a Call Center environment, which allows graphical call control of the agent’s phone.
Call Center	A place where calls are answered and/or calls are made. A typical call center will have lots of people, called agents, answering phones. Outbound calls are typically made using a machine-automated process and inbound calls are typically answered by an IVR system before being placed in an on-hold queue to wait for a live agent.
Client App	A TAPI or JTAPI based program that allows call control through a Windows interface but does not actually handle termination for a CTI port device like a SoftPhone does.
JTAPI	Java Telephony API is a set of APIs for Java-based telephony control. JTAPI applications, like Java, are platform independent and depend on another API, TAPI in the case of Cisco, to control the actual telephony hardware.
Prompt quality	Like speech quality except that the call is between a phone and an automated system such as IVR.
SimClient	The name applied to a Cisco proprietary end point simulator. A SimClient machine can simulate the traffic of up to 1000 simultaneous calls. Used by Cisco, and Cisco partners, to stress test applications.
SoftPhone	Controls and handles media termination for a CTI port device. Typically refers to a software program, programmed in TAPI or JTAPI, which acts like a PBX phone. An example of a PBX phone is the Cisco 7960.
Speech quality	When grading the quality of sound as passed between two phones we call it speech quality.
TAPI	AKA Microsoft Windows Telephony API. TAPI is a standard group of

	Win32 APIs that allow communications applications to control telephony functions.
Auto Registration	Auto Registration is a feature of Communications Manager where a phone can be added to the network. The phone will get an IP address and TFTP address through DHCP, and find the Communications Manager through the .CNF provided information.
Call Forward Busy	Configurable feature that re-routes incoming calls to an alternate line when the first line is in use.
Call Forward No Answer	Configurable feature that re-routes incoming calls from one phone to another phone when the first phone is not answered after a certain number of rings.
Call Park	Call Park allows you to place a call on hold at an extension specified by your system administrator so that anyone on the IP Telephony network can retrieve it. For example, you could park a call at extension 3000. Anyone on the system can dial 3000 to retrieve the call.
Call Waiting	Call Waiting lets you receive a second incoming call on the same line without disconnecting the first call. When the second call arrives, you hear a brief call waiting indicator tone.
Conference	Conference allows you to connect three or more people into one phone conversation.
Configuration file	An unformatted ASCII file that stores initialization information for an application. For Cisco Communications Manager, files in the .cnf format define the parameters for Cisco IP Phone connection.
Group Pick Up	A feature that allows users to pick up incoming calls within their own group or within other call pickup groups by dialing the group call pickup number for that group.
Failover/Failback	Failover describes the action a 3rd party phone must take to re-register to the Secondary Communications Manager if the Primary Communications Manager is no longer available. Failback describes what happens when the Primary Communications Manager becomes available again.
Forward All	Forward All lets you set up a Cisco IP Phone so that all calls destined for that Cisco IP Phone automatically ring another phone. You can forward all calls to Cisco IP Phones or non-Cisco IP Phones. Forward All remains in effect until you cancel it.
Hold	Hold lets you store a call on the line for later retrieval. While a call is on hold, you can place another call or use any of the other features of the phone. While on hold, the caller hears an intermittent tone.
Manual Registration	Manual registration describes the process by which a user configures an IP telephone manually on the target Communications Manager. Manual registration also requires the following to be manually entered into the phone: IP, submask, DNS, TFTP, and Default Gateway.
Shared Line Extension	A shared line extension is an extension shared by two or more phones similar to a PSTN ‘party line’. However, unlike a ‘party line’, if one of the shared line IP phones makes or receives a call, the other phones cannot hear the conversation.
Speakerphone	You can use speakerphone to place and answer calls without using the

	handset. Use speakerphone with any other feature.
Transfer	Transfer allows you to send an existing call to another extension. You can cancel the transfer by pressing HOLD.