Ascom IP Dect was first tested on July 30st, 2014 with firmware version 7.1.3 on Call Manager 10.0. When testing we ran into fail test case 6.4.2, because of this the whole test failed and was not able to get its certification. Between tekVizion and Cisco we created a regression test plan for Ascom. Ascom fixed the previous issue and came out with IP Dect 7.2.7 firmware, this firmware was used for the regression test Call Manager 10.5. The regression test plan was tested on January 21st, 2015 and passed. Below you will see the original test report followed by the regression test report.

IVT Regression Test Plan and Report for Ascom IP Dect

Test Result	Pass
Test Date	January 21, 2015
Product Name	IP Dect
Product Version #	Firmware= 7.2.7
Call Manager Version X.X(x)	10.5.1
Partner Main Support Email	Karl-Magnus.Olsson@ascom.se

Revision History

Revision	Author	Date	Comment
2.3	Craig Newman	August 7th 2014	Updates after review with Cisco
2.4	Felipe Escobar	October 2, 2014	Regression Test Plan
2.5	Cherie Reed	October 15, 2014	Revised for regression and added Cisco required test cases
2.6	Felipe Escobar	January 21, 2015	Regression Test Report

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1 Executive Summary

The following summarizes tekVizion's findings:

- Test Case Failures:
  - 2.2.7: Call is disconnected 1800s after primary node goes down. The session timer expires value 1800s set by CUCM is never refreshed.

2 Introduction

This document is a regression test plan of Ascom IP DECT that was tested previously on July 30th, 2014. When previously tested the following test case failed:

Subscriber Outage: place station-to-station call, isolate Subscriber (disconnect network cable), place one more station-to-station call, reconnect Subscriber network cable, place another station-to-station call	Confirm all calls are completed.	**Fail**	**Failed to fallback: when the Subscriber(Primary to DUT) is down the active call stays up and after the call is disconnected, the DUT gets registered to secondary node but when primary comes back online the active call with seconday is getting disconnected and there is a down time before the calls are successful**

Ascom has created a fix for the issue identified. The purpose of this test plan is to validate resolution of the issue(s) identivied via an exception through Cisco approved regression testing. Upon completion, this document will be added as an addendum to the original report with notes referring to it and the exception made by Cisco in the Executive Summary section of the original report.

Note: All test specific information is contained in the original test document, this document contains regression cases and results only.

2.1 Basic Call Scenarios

The intention of this section is to verify the basic operation of the new version under test.

Test Case	Description	Expected Result	Pass/ Fail	Comments
Station to Station Calls				
2.1.1	DUT to DUT2, originator releases call	Two-way voice path, call released properly	**Pass**	
2.1.2	IP-DECT to IP-DECT, originator releases call (KPML)	Two-way voice path, call released properly	**Pass**	
2.1.3	DUT to CSP, originator releases call	Two-way voice path, call released properly	**Pass**	
2.1.4	DUT to CSIPP, originator releases call	Two-way voice path, call released properly	**Pass**	
PSTN Calls				
2.1.5	DUT to PSTN, originator releases call	Two-way voice path, call released properly	**Pass**	
2.1.6	IP-DECT to PSTN, originator releases call (TLS/SRTP)	Two-way voice path, call released properly	**Pass**	

Test Case	Description	Expected Result	Pass/ Fail	Comments
Call Forward All (CFA) (Note: Applicable to devices that send 3xx redirect)				
2.1.7	DUT to DUT2, Call forwarded to DUT3, Endpoint releases call	Two-way audio, call released properly	**Pass**	
3-Way Conference				
2.1.8	DUT to DUT2, Originator is the bridge(DUT bridges the call with DUT3) a) Originator drops out of the conference first b) DUT2 drops out of conference first c) DUT3 drops out of conference first	DUT2 receives TOH/silence DUT receives ring back, 3-way audio Other 2 parties remain on conference with 2-way audio.	**Pass**	

Test Case	Description	Expected Result	Pass/ Fail	Comments
Call Forward All (CFA) (Note: Applicable to devices that send 3xx redirect)				
2.1.9	DUT to DUT2, Call forwarded to DUT3, Endpoint releases call	Two-way audio, call released properly	**Pass**	Same as 2.1.7
2.1.10	DUT to DUT2, Call forwarded to DUT3, DUT abandons call	Terminator stops ringing, call released properly	**Pass**	
3-Way Conference				
2.1.11	DUT to DUT2, Originator is the bridge(DUT bridges the call with DUT3) d) Originator drops out of the conference first e) DUT2 drops out of conference first f) DUT3 drops out of conference first	DUT2 receives TOH/silence DUT receives ring back, 3-way audio Other 2 parties remain on conference with 2-way audio.	**Pass**	**Same as 2.1.8**

2.2 Specific Regression Tests

These tests are specific to the area of change.

Test Case	Description	Expected Result	Pass/ Fail	Comments
2.2.1	Primary Subscriber Outage: Place station-to-station call, isolate Subscriber (disconnect Primary UCM network cable) End Call. Make new call when device registers with Secondary UCM	Confirm call stays up until caller or called party disconnects. Upon call end, device registers with Primary UCM. Call completed after registration to Secondary UCM.	**Pass**	Note time to register to secondary UCM. On average 1 to 3 minutes.
2.2.2	Primary UCM Subscriber reinstated: Place station-to-station call while device is registered to secondary UCM While call is in conversation state – re-connect Primary UCM Sub. After Primary UCM has all previous devices registered and is fully back in service, disconnect call. Place new call.	Call stays in converstation state during primary recovery and until caller/called disconnect. After disconnect, device re-registers with primary. New call completes.	**Pass**	Note time to register to Primary UCM On average 1 to 3 minutes.
2.2.3	Primary and Secondary UCM failure. Disconnect Primary and secondary UCM from the network. Reconnect both. When device re-registers – make call.	Device loses registration. Device registers with first UCM online, and finally returns to Primary. Call completes normally.	**Pass**	

Test Case	Description	Expected Result	Pass/ Fail	Comments
2.2.4	Device Network Outage: Disconnect device from network. Disconnect Primary UCM from network. Connect Device to network. Make Call. Connect Primary UCM to network. End last call. Make new call	Device registers to Secondary UCM Call completes normally and stays in conversation state until user ends. New call completes.	**Pass**	Note time to register to UCM. On average 1 to 3 minutes.
2.2.5	Device Network Outage: Disconnect device from network. Disconnect Primary and Secondary UCM from network. Connect Device to network. Connect Secondary UCM to network. Make call Connect Primary UCM to network End first call Allow re-registration Make second call	Device registers to Secondary UCM Call completes normally and stays in conversation state until user ends. When Primary UCM is online and call is ended, device moves to Primary. New call completes.	**Pass**	Note time to register to UCM in both cases. On average 1 to 3 minutes.

Test Case	Description	Expected Result	Pass/ Fail	Comments
2.2.6	Create network bounce condition: Make call Drop, connect, drop, connect primary UCM while call is in conversation state. Wait until primary fully recovers and end call.	Call completes and is not ended until user ends.	**Pass**	
2.2.7	Long duration call – with Sub failure: Start call with device registered to Primary Sub. Disconnect Primary sub from network. Leave in this state for 1 hour. Reconnect Primary	Call stays in conversation state.	**Fail**	Call is disconnected 1800s after primary node goes down. The session timer expires value 1800s set by CUCM is never refreshed.
2.2.8	Long duration call – with Sub failure: Disconect primary Sub, allow device to register with secondary Sub. Start call with device registered to Secondary Sub. Re-sconnect Primary sub to network. Leave call in conversation state for 1 hour. End Call.	Call stays in conversation state. Device re-registers with primary Sub.	**Pass**	

3 Appendix

Email from Cisco that proves test case 2.2.7 is a fail test case but that the overall test is still a pass:

Re Re Certification Quote.msg

Detailed IVT Test Plan and Report for Ascom IP-Dect Phones with CUCM 10.0

Test Result	**Fail**
Test Date	**30th July 2014**
Product Name	**Ascom IP-DECT**
Product Version #	**7.1.3**
Call Manager Version X.X(x)	**10.0.1.10000-24**
Product Type (Billing, Voice Recording, phone apps etc.):	**SIP End Point**
API/Protocol(s) Used	**SIP**
Developer Services Contract #	**91326703**
Partner IVT Contact Name:	Gert Wallin
Partner IVT Contact Phone:	**+46 31 559514**
Partner IVT Contact Email:	gert.wallin@ascom.se
IVT Lab Location (EMEA or US):	**Richardson, USA**
Partner Main Support Number	+ 46 31 55 94 50
Partner Main Support Email	support@ascom.se

Revision History

Revision	Author	Date	Comment
1.0	Gerry Pearson	09/12/2008	Initial draft for Cisco review
1.1	Sonu Sijin	07/11/2012	Modified existing test cases to add more detailed description, added missing scenarios
1.2	Ameeta Thukral	07/12/2012	Added UCM9.0 feature - native call queuing related test scenarios.
1.3	Ameeta Thukral	07/17/2012	Incorporated internal review comments
1.4	Tony Nally/ Cherie Reed	08/09/2012	Test plan reviewed by Cisco
1.5	Ameeta Thukral/Sonu Sijin	08/17/2012	Incorporated the review comments
1.6	Neena Pemmaraju	08/20/2012	Updated template
1.7	Neena Pemmaraju	08/27/2012	Updated template based on feedback from Cisco
2.0	Ameeta Thukral	09/21/2012	Updating version to 2.0 based after Cisco review and approval.
2.1	Suresh Kadiyala	4/1/2013	Modified Native call queuing test cases.
2.2	Suresh Kadiyala	7/30/2014	Test Report.
2.3	Craig Newman	August 7th 2014	Updates after review with Cisco

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4 Introduction

This document is the detailed Interoperability Verification Test Plan and Report for Cisco Unified Communications Manager 10.0 and Ascom IP-Dect Phone 7.1.3.

4.1 Entry Criteria

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

4.2 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 whom is responsible for the problem (Cisco or the 3rd party product), the testing is considered unsuccessful.

Table **1 Defect Severity Level Description**

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

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

The following procedures are followed when testing fails:

- Preliminary analysis is made to determine the source of the problem. If the problem is related to a device under test, then the problem is reported to that partner. If the problem is deemed Cisco related, the problem will be reported to Cisco, but the partner is responsible to open a TAC case with Cisco developer services. Partner should provide the TAC case number to the test team so they can document it in the report.
- If testing can continue past this failure, the other test cases will be tested and verified for pass or fail. If the testing cannot progress past this problem, testing will be halted and a final test report submitted to Partner and Cisco.

- All problems and resolutions encountered during testing are documented in the final test report.
- If a severity 1 failure occurs, irrespective of whom is responsible for the problem (Cisco or the 3rd party product), the testing is considered unsuccessful.

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

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

5 Product Overview

The IP-DECT system from Ascom combines the VoIP world with the traditional wireless DECT solution in an innovative package. One big advantage is that you can have both packet data and high-quality voice connections on the same network and look forward to superb quality of service and excellent messaging capabilities in a secure radio environment. The wide range of Ascom IP-DECT handset meets the needs from any user, ranging from entry-level office handsets, purpose-built healthcare application handsets to robust handsets for industrial usage.

6 Executive Summary

Short summary of the test effort, summarizing tekVizion's findings during the testing

The following summarizes tekVizion's findings:

- Test Case Failures:
    - Fallback: Test case 6.4.2 fails. After a failover from the Primary Subscriber to the Secondary Subscriber, if a call remains in progress when the Primary Subscriber is restored it will be dropped. Users that complete their calls on the Secondary Subscriber will successfully establish future calls on the Primary server. Similarly, if a call is dropped on Fallback, a user's next call attempt will use the restored Primary after a short time-out. This is the current design of the DUT. After review with Cisco, this issue is deemed a Sev2 which results in an overall failure for this test cycle.

    - When DUT is configured with Digest authentication profile, Conference feature is failing: when conference is initiated from the DUT, the DUT sends a REFER and CUCM is responding with 401 unauthorized message and the DUT is not sending digest credentials in the following REFER to CUCM. Howerver conference feature worked successfully with regular non-secure profile. After review with Cisco, this issue is deemed a Sev3.

      This is a confirmed bug in the Ascom IP-DECT platform. However this an unusual/not preferred configuration. The Ascom IP-DECT device type in CUCM does not require an End user unless using security profile with digest authentication. To secure the communication it is preferred to use a security profile using SIP-TLS instead of digest authentication as this does not require adding and End user for every device.

- Features Not Supported:
    - Fallback: Test case 6.4.2 is not supported by Ascom. After a failover from the Primary Subscriber to the Secondary Subscriber, if a call remains in progress when the Primary Subscriber is restored it will be dropped. Users that complete their calls on the Secondary Subscriber will successfully establish future calls on the Primary server. Similarly, if a call is dropped on Fallback, a user's next call attempt will use the restored Primary after a short time-out. This is the current design of the DUT.

    - SIP URI Dialing
    - Mobile Handoff with Mobile communicator
    - Mid-call codec renegotiation

- Test Cases that are Not Applicable:
    - Multiline per phone
    - Call Forking
    - DUT as Hotline
    - Hlog key on DUT

- Test Cases that were Not Executed:
    - None

- Observations:
    - While not tied to a test case, an attempt to install the COP file on UCM 10.5 failed. Ascom has confirmed with Cisco that this is a generic issue with the installation of COP files on release 10.5. Cisco ticket 631068765 is open regarding this issue. The COP file was successfully installed using UCM 10.0.1.

    - Blind transfer: While doing Blind transfer from DUT to an invalid extension fails, the call is handed over back to the transferor (Originator) and the initial call is resumed.

    - Semi-Unattendend: While the transfer target phone is ringing, the transferee will hear MOH and once the Transfer target phone answers, the call will be connected.

    - On the DUT, Not possible to remove participants from the conference list.

7 Features Tested

The following features are tested as part of this test plan.

Unified Communications Manager Feature	RFC Reference	To Be Tested?
Call Hold and Resume	3261, 3264, 2327, 1889	Yes
Transfer Unattended	3261, 3264, 2327, 1889, 3515, 3420, 3265, 3892	Yes
Transfer Attended	3261, 3264, 2327, 1889, 3515, 3420, 3265, 3892, 3891	Yes
Call Forwarding All	3261, 3264, 2327, 1889	Yes
Call Forwarding No Answer	3261, 3264, 2327, 1889	Yes
Call Forwarding Busy	3261, 3264, 2327, 1889	Yes
Multiple Calls per Line	3261, 3264, 2327, 1889	Yes
Incoming Call Screening	3261, 3264, 2327, 1889, 3725	No
Outgoing Call Screening	3261, 3264, 2327, 1889, 3725	No
Calling and Connected Line ID	3261, 3264, 2327, 1889, Remote Party ID	Yes
Calling and Connected Name ID	3261, 3264, 2327, 1889, Remote Party ID	Yes
Message Waiting Indication	3261, 3264, 2327, 1889, 3842	Yes
Three-Way Conference Calling	3261, 3264, 2327, 1889	Yes
Call Forking	3261, 3264, 2327, 1889	No
Speed Dialing	3261, 3264, 2327, 1889	Yes
Multiple Lines per Phone	3261, 3264, 2327, 1889	No
SRST fallback		Yes

7.1 Items Not Tested

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

7.2 Assumptions

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

8 Test Environment

8.1 Administration, Testing and Debugging tools

Table 2 Administration, Testing and Debugging Tools

Product Name	Version	Type	Purpose	Units	Notes
Test Tools					
SIM Client					Lab Provided
3rd Party Tools					
N/A					
Debug Tools					
Wireshark	1.8.4	Protocol trace	Debug	1	Lab Provided

8.2 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 Call Manager	9.0	MCS7835- I3	Publisher and 2 Subscriber nodes	3	Lab Provided
Cisco 3800 (PSTN GW)	Version 12.4(13r)T	IOS	PSTN Gateway	1	Lab Provided

Product	Version	Type	Purpose	Units	Notes
3rd Party Products					
Ascom IP-DECT	6.1	SIP endpoint	Telephone	1base station, 4 handsets	Ascom Provided

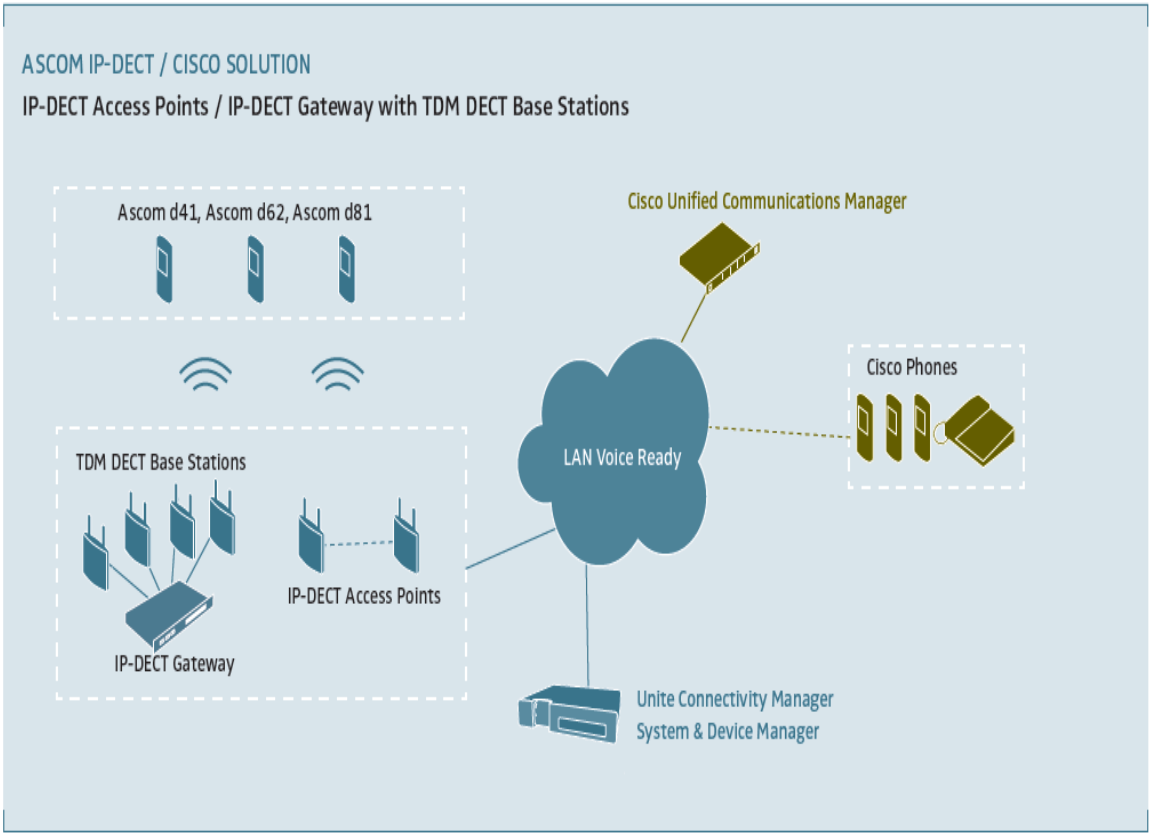
8.3 Cisco Phones

Table 4 Cisco Phones Information

Cisco Phone Model	Phone Firmware Version	Protocol	POE/ Power	Units	Notes
Cisco 7960	8.1(2.0)	SCCP	POE	1	Lab Provided
Cisco 7960	DSP Load ID 4.0(5.0)	SIP	POE	1	Lab Provided
Cisco7965	SIP 45.9-3-1 SR1-1S	SIP	POE	1	Lab Provided
Cisco 9971		SIP	POE	1	Lab Provided

8.4 Deployment Architecture

Figure 1 – Deployment Architecture



8.5 Test Environment Architecture

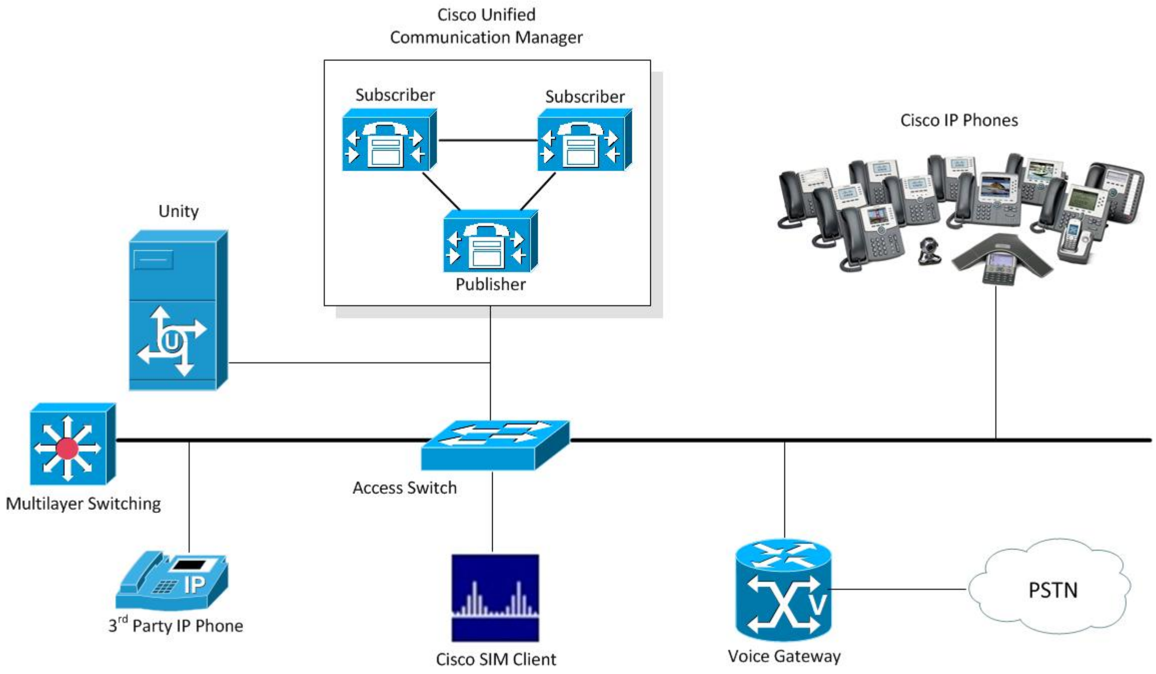


Figure 2 – Test Environment

9 Test Cases

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

Note: Unless otherwise noted, all tests will be run with a background load (80K BHCA of basic calls) on the CUCM.

Table 5 – Test Results Legend

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.

Note:

- DUT – Device Under Test
- CSP – Cisco Skinny Phone
- CSIPP – Cisco SIP Phone
- Treatment - Treatment may be an announcement which plays for example, 'The number you have dialed does not exist, please check the number and dial again ' or it could be a busy / disconnect tone, etc.

9.1 Basic Call Scenarios

The intention of this section is to verify that basic calls can be properly handled between the SIP Phone under test and Cisco Unified Communications Manager. This test includes the validation of the different call stages from setup, alerting, connecting, and tear down, as well as different call scenarios between end points, IP server local or remote extensions and calls to and from PSTN, Cisco SIP and SCCP phones.

Test Case	Description	Expected Result	Pass/ Fail	Comments
Station to Station Calls				
9.1.1	DUT to DUT2, originator releases call	Two-way voice path, call released properly	Pass	
9.1.1.1	IP-DECT to IP-DECT, originator releases call (KPML)	Two-way voice path, call released properly	Pass	
9.1.1.2	IP-DECT to IP-DECT, originator releases call (TLS/SRTP)	Two-way voice path, call released properly	Pass	
9.1.2	DUT to DUT2, originator abandons call	Terminator stops ringing, originator released properly	Pass	
9.1.3	DUT to DUT2, terminator releases	Two-way voice path, call released properly	Pass	
9.1.4	DUT to DUT2, terminator busy	Busy tone heard at originator	Pass	Call Waiting Disabled
9.1.5	DUT to DUT2, unanswered call	Ringing at terminator, ring back at originator, originator released properly	Pass	
9.1.6	DUT, call to unknown number(an invalid number)	Treatment heard at originator, originator released properly	Pass	Temp failed message on phone.
9.1.7	DUT to CSP, originator releases call	Two-way voice path, call released properly	Pass	
9.1.8	DUT to CSP, terminator releases call	Two-way voice path, call released properly	Pass	
9.1.8.1	IP-DECT to CSP, terminator releases call (TLS/SRTP)	Two-way voice path, call released properly	Pass	
9.1.9	CSP to DUT, originator abandons call	Terminator stops ringing, originator released properly	Pass	

Test Case	Description	Expected Result	Pass/ Fail	Comments
9.1.10	CSP to DUT, terminator releases	Two-way voice path, call released properly	Pass	
9.1.10.1	CSP to IP-DECT, terminator releases (TLS/SRTP)	Two-way voice path, call released properly	Pass	
9.1.11	DUT to CSP, terminator busy	Busy tone heard at originator	Pass	Call Waiting Disabled
9.1.12	DUT to CSP, unanswered call	Ringing at terminator, ring back at originator, originator released properly	Pass	
9.1.13	DUT to CSIPP, originator releases call	Two-way voice path, call released properly	Pass	
9.1.14	DUT to CSIPP, terminator releases call	Two-way voice path, call released properly	Pass	
9.1.14.1	IP-DECT to CSIPP, terminator releases call (TLS/SRTP)	Two-way voice path, call released properly	Pass	
9.1.15	CSIPP to DUT, originator abandons call	Terminator stops ringing, originator released properly	Pass	
9.1.16	CSIPP to DUT, terminator releases	Two-way voice path, call released properly	Pass	
9.1.16.1	CSIPP to IP-DECT, terminator releases (TLS/SRTP)	Two-way voice path, call released properly	Pass	
9.1.17	DUT to CSIPP, terminator busy	Busy tone heard at originator	Pass	Call Waiting Disabled
9.1.18	DUT to CSIPP, unanswered call	Ringing at terminator, ring back at originator, originator released properly	Pass	
SIP URI Dialing :CUCM 9.0 feature : Devices that support URI receive both DN and URI / Devices that don’t support URI receive the DN only				

Test Case	Description	Expected Result	Pass/ Fail	Comments
SIP URI Dialing: Intra-Cluster				
9.1.19	DUT to DUT2: SIP URI Dialing. Originator releases call.	DUT2 : receives both DUT DN and URI./ receive the DN only 2-way voice path established successfully. Call released successfully.	N/S	
9.1.20	DUT to CSIPP : SIP URI Dialing Originator releases call.	CSIPP : receives both DUT DN and URI./ receive the DN only 2-way voice path established successfully. Call released successfully.	N/S	
9.1.21	CSIPP to DUT : SIP URI Dialing Originator releases call.	CSIPP : receives both DUT DN and URI./ receive the DN only 2-way voice path established successfully. Call released successfully.	Pass	DUT receives only DN
SIP URI Dialing – Inter-cluster				
9.1.22	DUT to DUT2: SIP URI Dialing.- DUT2 in different cluster Originator releases call.	DUT2 : receives both DUT DN and URI./ receive the DN only 2-way voice path established successfully. Call released successfully.	N/S	

Test Case	Description	Expected Result	Pass/ Fail	Comments
9.1.23	DUT to CSIPP : SIP URI Dialing- CSIPP in different cluster Originator releases call.	CSIPP : receives both DUT DN and URI./ receive the DN only 2-way voice path established successfully. Call released successfully.	N/S	
9.1.24	CSIPP to DUT : SIP URI Dialing - DUT in different cluster Originator releases call.	DUT : receives both CSIPP DN and URI./ receive the DN only 2-way voice path established successfully. Call released successfully.	N/S	
PSTN Calls				
9.1.25	DUT to PSTN, originator releases call	Two-way voice path, call released properly	Pass	
9.1.25.1	IP-DECT to PSTN, originator releases call (TLS/SRTP)	Two-way voice path, call released properly	Pass	
9.1.26	DUT to PSTN, originator abandons call	Terminator stops ringing, originator released properly	Pass	
9.1.27	DUT to PSTN, terminator releases	Two-way voice path, call released properly	Pass	
9.1.27.1	IP-DECT to PSTN, terminator releases (TLS/SRTP)	Two-way voice path, call released properly	Pass	
9.1.28	DUT to PSTN, terminator busy	Busy tone heard at originator	Pass	
9.1.29	DUT to PSTN, unanswered call	Ringing at terminator, ring back at originator, originator released properly	Pass	

Test Case	Description	Expected Result	Pass/ Fail	Comments
9.1.30	DUT to PSTN, call to unknown number(an invalid number)	Treatment heard at originator, originator released properly	Pass	
9.1.31	PSTN to DUT, PSTN abandons call	Terminator stops ringing, originator released properly	Pass	
9.1.32	PSTN to DUT, terminator releases call	Two-way voice path, call released properly	Pass	
9.1.33	PSTN to DUT, terminator busy	Busy tone heard at originator	Pass	
9.1.34	PSTN to DUT, unanswered call	Ringing at terminator, ring back at originator, originator released properly	Pass	
DTMF Using G.711 (in band)				
9.1.35	DUT retrieves a voicemail, DUT releases call after sending DTMF tones	Voicemail retrieve successfully	Pass	Executed this by selecting Inband DTMF on the DUT and calling a PSTN DTMF recognition number. Heard tones from RTP stream.
9.1.36	DUT retrieves a voicemail, voicemail releases call after receiving DTMF tones	Voicemail retrieve successfully	Pass	Executed this by selecting Inband DTMF on the DUT and calling a PSTN DTMF recognition number. Heard tones from RTP stream.
DTMF Using RFC 2833 (out of band)				
9.1.37	DUT retrieves a voicemail, DUT releases call after sending DTMF tones	Voicemail retrieve successfully	Pass	
9.1.38	DUT retrieves a voicemail, voicemail releases call after receiving DTMF tones	Voicemail retrieve successfully	Pass	

9.2 Cisco Unified Communications Manager Feature Support

The goal of this section is to verify protocol interactions between the device under test and the Cisco Unified Communications Manager standards implementation. Focus is on feature call functionality, call control and other call information support interworking capabilities of the endpoint under test and the Cisco Unified Communications Manager version under test.

Test Case	Description	Expected Result	Pass/ Fail	Comments
Call Forward All (CFA) (Note: Applicable to devices that send 3xx redirect)				
9.2.1	DUT to DUT2, Call forwarded to DUT3, Endpoint releases call	Two-way audio, call released properly	Pass	
9.2.2	DUT to DUT2, Call forwarded to DUT3, DUT3 releases call	Two-way audio, call released properly	Pass	
9.2.3	DUT to DUT2, Call forwarded to DUT3, DUT abandons call	Terminator stops ringing, call released properly	Pass	
9.2.4	DUT to DUT2, Call forwarded to DUT3, DUT3 is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled.
9.2.5	PSTN to DUT, Call forwarded to DUT2, PSTN releases call	Two-way audio, call released properly	Pass	
9.2.6	PSTN to DUT, Call forwarded to DUT2, DUT2 releases call	Two-way audio, call released properly	Pass	
9.2.7	PSTN to DUT, Call forwarded to DUT2, PSTN abandons call	Terminator stops ringing, call released properly	Pass	
9.2.8	PSTN to DUT, Call forwarded to DUT2, DUT2 is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on DUT2.
9.2.9	DUT to CSP, Call forwarded to DUT2, DUT releases call	Two-way audio, call released properly	Pass	
9.2.10	DUT to CSIPP, Call forwarded to DUT2, DUT releases call	Two-way audio, call released properly	Pass	
9.2.11	DUT to CSP, Call forwarded to DUT2, DUT2 releases call	Two-way audio, call released properly	Pass	
9.2.12	DUT to CSIPP, Call forwarded to DUT2, DUT abandons call	Terminator stops ringing, call released properly	Pass	
9.2.13	DUT to CSP, Call forwarded to DUT2, DUT2 is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on DUT2

9.2.14	DUT to CSIPP, Call forwarded to DUT2, DUT2 is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on DUT2.
9.2.15	PSTN to DUT, Call forwarded to CSIPP, PSTN releases call	Two-way audio, call released properly	Pass	
9.2.16	PSTN to DUT, Call forwarded to CSP, CSP releases call	Two-way audio, call released properly	Pass	
9.2.17	PSTN to DUT, Call forwarded to CSIPP, PSTN abandons call	Terminator stops ringing, call released properly	Pass	
9.2.18	PSTN to DUT, Call forwarded to CSP, CSP is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on CSP
9.2.19	CSIPP to DUT, Call forwarded to DUT2, CSIPP abandons call	Terminator stops ringing, call released properly	Pass	
9.2.20	DUT to CSP, Call forwarded to CSSIP, CSSIP is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on CSIPP
9.2.21	DUT to CSIPP, Call forwarded to CSP, CSP is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on CSP
Call Forward if Busy (CFB)				
9.2.22	DUT to DUT2, Call forwarded to DUT3, Endpoint releases call	Two-way audio, call released properly	Pass	
9.2.23	DUT to DUT2, Call forwarded to DUT3, DUT3 releases call	Two-way audio, call released properly	Pass	
9.2.24	DUT to DUT2, Call forwarded to DUT3, DUT abandons call	Terminator stops ringing, call released properly	Pass	

9.2.25	DUT to DUT2, Call forwarded to DUT3, DUT3 is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on DUT3
9.2.26	PSTN to DUT, Call forwarded to DUT2, PSTN releases call	Two-way audio, call released properly	Pass	
9.2.27	PSTN to DUT, Call forwarded to DUT2, DUT2 releases call	Two-way audio, call released properly	Pass	
9.2.28	PSTN to DUT, Call forwarded to DUT2, PSTN abandons call	Terminator stops ringing, call released properly	Pass	
9.2.29	PSTN to DUT, Call forwarded to DUT2, DUT2 is busy	Originator hears busy tone, call released properly	Pass	
9.2.30	DUT to CSP, Call forwarded to DUT2, DUT releases call	Two-way audio, call released properly	Pass	
9.2.31	DUT to CSIPP, Call forwarded to DUT2, DUT releases call	Two-way audio, call released properly	Pass	
9.2.32	DUT to CSP, Call forwarded to DUT2, DUT2 releases call	Two-way audio, call released properly	Pass	
9.2.33	DUT to CSIPP, Call forwarded to DUT2, DUT abandons call	Terminator stops ringing, call released properly	Pass	
9.2.34	DUT to CSP, Call forwarded to DUT2, DUT2 is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on DUT2
9.2.35	DUT to CSIPP, Call forwarded to DUT2, DUT2 is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on DUT2.
9.2.36	PSTN to DUT, Call forwarded to CSIPP, PSTN releases call	Two-way audio, call released properly	Pass	
9.2.37	PSTN to DUT, Call forwarded to CSP, CSP releases call	Two-way audio, call released properly	Pass	

9.2.38	PSTN to DUT, Call forwarded to CSIPP, PSTN abandons call	Terminator stops ringing, call released properly	Pass	
9.2.39	PSTN to DUT, Call forwarded to CSP, CSP is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on DUT2
9.2.40	CSIPP to DUT, Call forwarded to DUT2, CSIPP abandons call	Terminator stops ringing, call released properly	Pass	
9.2.41	DUT to CSP, Call forwarded to CSSIP, CSSIP is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on CSIPP
9.2.42	DUT to CSIPP, Call forwarded to CSP, CSP is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on CSP
Call Forward if No Answer (CFNA)				
9.2.43	DUT to DUT2, Call forwarded to DUT3, Endpoint releases call	Two-way audio, call released properly	Pass	
9.2.44	DUT to DUT2, Call forwarded to DUT3, DUT3 releases call	Two-way audio, call released properly	Pass	
9.2.45	DUT to DUT2, Call forwarded to DUT3, DUT abandons call	Terminator stops ringing, call released properly	Pass	
9.2.46	DUT to DUT2, Call forwarded to DUT3, DUT3 is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on DUT3
9.2.47	PSTN to DUT, Call forwarded to DUT2, PSTN releases call	Two-way audio, call released properly	Pass	
9.2.48	PSTN to DUT, Call forwarded to DUT2, DUT2 releases call	Two-way audio, call released properly	Pass	
9.2.49	PSTN to DUT, Call forwarded to DUT2, PSTN abandons call	Terminator stops ringing, call released properly	Pass	

9.2.50	PSTN to DUT, Call forwarded to DUT2, DUT2 is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on DUT2
9.2.51	DUT to CSP, Call forwarded to DUT2, DUT releases call	Two-way audio, call released properly	Pass	
9.2.52	DUT to CSIPP, Call forwarded to DUT2, DUT releases call	Two-way audio, call released properly	Pass	
9.2.53	DUT to CSP, Call forwarded to DUT2, DUT2 releases call	Two-way audio, call released properly	Pass	
9.2.54	DUT to CSIPP, Call forwarded to DUT2, DUT abandons call	Terminator stops ringing, call released properly	Pass	
9.2.55	DUT to CSP, Call forwarded to DUT2, DUT2 is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on DUT2
9.2.56	DUT to CSIPP, Call forwarded to DUT2, DUT2 is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on DUT2
9.2.57	PSTN to DUT, Call forwarded to CSIPP, PSTN releases call	Two-way audio, call released properly	Pass	
9.2.58	PSTN to DUT, Call forwarded to CSP, CSP releases call	Two-way audio, call released properly	Pass	
9.2.59	PSTN to DUT, Call forwarded to CSIPP, PSTN abandons call	Terminator stops ringing, call released properly	Pass	
9.2.60	PSTN to DUT, Call forwarded to CSP, CSP is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on DUT2
9.2.61	CSIPP to DUT, Call forwarded to DUT2, CSIPP abandons call	Terminator stops ringing, call released properly	Pass	
9.2.62	DUT to CSP, Call forwarded to CSSIP, CSSIP is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on CSIPP

9.2.63	DUT to CSIPP, Call forwarded to CSP, CSP is busy	Originator hears busy tone, call released properly	Pass	Call Waiting Disabled on CSP
Call Hold and Resume				
9.2.64	DUT to DUT2. Originator Holds and resumes call	DUT2 hears TOH/silence 2-way audio resumes	Pass	
9.2.65	DUT to DUT2. Terminator Holds and resumes call	DUT hears TOH/silence, 2-way audio resumes	Pass	
9.2.66	DUT to DUT2. Originator Holds call to answer an incoming call	DUT2 hears TOH/silence, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.67	DUT to DUT2. Terminator Holds call to answer an incoming call	DUT hears TOH/silence, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.68	DUT to DUT2. Originator Holds to originate a second call	DUT2 hears TOH/silence, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.69	DUT to DUT2. Terminator Holds to originate a second call	DUT hears TOH/silence, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.70	DUT to DUT2. Originator Holds call, Terminator releases before retrieve	DUT2 hears TOH/silence, DUT2 leg properly released, DUT unable to retrieve, DUT properly released	Pass	
9.2.71	DUT to DUT2. Terminator Holds	DUT hears TOH/silence, DUT	Pass	

	call, Originator releases before retrieve	leg properly released, DUT2 unable to retrieve, DUT2 properly released		
9.2.72	PSTN to DUT, Terminator Holds and resumes call	PSTN hears silence, 2-way audio resumes	Pass	
9.2.73	PSTN to DUT, PSTN Holds and resumes call	DUT hears Silence/MOH, 2-way audio resumes	Pass	
9.2.74	PSTN to DUT, terminator Holds call to answer an incoming call	PSTN hears silence, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.75	PSTN to DUT, PSTN Holds call to answer an incoming call	DUT hears Silence/MOH, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.76	PSTN to DUT, terminator Holds to originate a second call	PSTN hears silence, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.77	PSTN to DUT, PSTN Holds to originate a second call	DUT hears Silence/ MOH, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.78	PSTN to DUT, terminator Holds call, PSTN releases before retrieve	PSTN hears silence, PSTN leg properly released, DUT unable to retrieve, DUT properly released	Pass	
9.2.79	PSTN to DUT, PSTN Holds call, End Point (DUT) release before retrieve	DUT hears silence/ MOH, DUT leg properly released, PSTN unable to retrieve, PSTN properly released	Pass	

9.2.80	DUT to PSTN, originator holds and resumes call	PSTN hears silence, 2-way audio resumes	Pass	
9.2.81	DUT to PSTN, PSTN Holds and resumes call	DUT hears silence/MOH, 2-way audio resumes	Pass	
9.2.82	DUT to PSTN, originator holds call to answer an incoming call	PSTN hears silence/, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.83	DUT to PSTN, PSTN Holds call to answer an incoming call	DUT hears silence/MOH, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.84	DUT to PSTN, originator holds to originate a second call	PSTN hears silence, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.85	DUT to PSTN, PSTN Holds to originate a second call	DUT hears silence/MOH, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.86	DUT to PSTN, DUT Holds call, PSTN releases before retrieve	PSTN hears silence, PSTN leg properly releas ed, DUT unable to retrieve, DUT properly released	Pass	
9.2.87	DUT to PSTN, PSTN Holds call, DUT release before retrieve	DUT hears silence/MOH, DUT leg properly released, PSTN unable to retrieve, PSTN properly released	Pass	
9.2.88	DUT to CSP. Originator Holds and resumes call	CSP hears silence, 2-way audio resumes	Pass	

9.2.89	DUT to CSP. Terminator Holds and resumes call	DUT hears MOH, 2-way audio resumes	Pass	
9.2.90	DUT to CSP. Originator Holds call to answer an incoming call	CSP hears silence, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.91	DUT to CSP. Terminator Holds call to answer an incoming call	DUT hears MOH, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.92	DUT to CSP. Originator Holds to originate a second call	CSP hears silence, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.93	DUT to CSP. Terminator Holds to originate a second call	DUT hears MOH, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.94	DUT to CSP. Originator Holds call, Terminator releases before retrieve	CSP hears silence, CSP leg properly released, DUT unable to retrieve, DUT properly released	Pass	
9.2.95	DUT to CSP. Terminator Holds call, Originator releases before retrieve	DUT hears MOH, DUT leg properly released, CSP unable to retrieve, CSP properly released	Pass	
9.2.96	CSP to DUT. Originator Holds and resumes call	DUT hears MOH, 2-way audio resumes	Pass	
9.2.97	CSP to DUT. Terminator Holds and resumes call	CSP hears silence, 2-way audio resumes	Pass	
9.2.98	CSP to DUT. Originator Holds call to answer an incoming call	DUT hears MOH, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	

9.2.99	CSP to DUT. Terminator Holds call to answer an incoming call	CSP hears silence, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.100	CSP to DUT. Originator Holds to originate a second call	DUT hears MOH, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.101	CSP to DUT. Terminator Holds to originate a second call	CSP hears silence, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.102	CSP to DUT. Originator Holds call, Terminator releases before retrieve	DUT hears MOH, DUT leg properly released, CSP unable to retrieve, CSP properly released	Pass	
9.2.103	CSP to DUT. Terminator Holds call, Originator releases before retrieve	CSP hears silence, CSP leg properly released, DUT unable to retrieve, DUT properly released	Pass	
9.2.104	DUT to CSIPP. Originator Holds and resumes call	CSIPP hears silence, 2-way audio resumes	Pass	
9.2.105	DUT to CSIPP. Terminator Holds and resumes call	DUT hears MOH, 2-way audio resumes	Pass	
9.2.106	DUT to CSIPP. Originator Holds call to answer an incoming call	CSIPP hears silence, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.107	DUT to CSIPP. Terminator Holds call to answer an incoming call	DUT hears MOH, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.108	DUT to CSIPP. Originator Holds to	CSIPP hears silence, 2-way audio on 2nd	Pass	

	originate a second call	call, 2nd call properly released, 2-way audio resumes		
9.2.109	DUT to CSIPP. Terminator Holds to originate a second call	DUT hears MOH, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.110	DUT to CSIPP. Originator Holds call, Terminator releases before retrieve	CSIPP hears silence, CSIPP leg properly released, DUT unable to retrieve, DUT properly released	Pass	
9.2.111	DUT to CSIPP. Terminator Holds call, Originator releases before retrieve	DUT hears MOH, DUT leg properly released, CSIPP unable to retrieve, CSIPP properly released	Pass	
9.2.112	CSIPP to DUT. Originator Holds and resumes call	DUT hears MOH, 2-way audio resumes	Pass	
9.2.113	CSIPP to DUT. Terminator Holds and resumes call	CSIPP hears silence, 2-way audio resumes	Pass	
9.2.114	CSIPP to DUT. Originator Holds call to answer an incoming call	DUT hears MOH, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.115	CSIPP to DUT. Terminator Holds call to answer an incoming call	CSIPP hears silence, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.116	CSIPP to DUT. Originator Holds to originate a second call	DUT hears MOH, 2-way audio on 2nd call, 2nd call properly released, 2-way audio resumes	Pass	
9.2.117	CSIPP to DUT. Terminator Holds to originate a second call	CSIPP hears silence, 2-way audio on 2nd call, 2nd call properly	Pass	

		released, 2-way audio resumes		
9.2.118	CSIPP to DUT. Originator Holds call, Terminator releases before retrieve	DUT hears MOH, DUT leg properly released, CSP unable to retrieve, CSP properly released	Pass	
9.2.119	CSIPP to DUT. Terminator Holds call, Originator releases before retrieve	CSIPP hears silence, CSP leg properly released, DUT unable to retrieve, DUT properly released	Pass	
Call Waiting				
9.2.120	DUT to DUT2, PSTN calls DUT Call waiting on Originator.	DUT indicates incoming call	Pass	
9.2.121	DUT to DUT2 PSTN calls DUT2 Call waiting on Terminator.	DUT2 indicates incoming call	Pass	
9.2.122	DUT to PSTN CSP/CSIPP calls DUT Call Waiting on Originator.	DUT indicates incoming call	Pass	
9.2.123	PSTN to DUT CSP/CSIPP calls DUT Call Waiting on Terminator.	DUT indicates incoming call	Pass	
9.2.124	DUT to CSP CSP/CSIPP calls DUT Call waiting on Originator.	DUT indicates incoming call	Pass	
9.2.125	CSP to DUT,	DUT indicates incoming call	Pass	

	CSP/CSIPP calls DUT Call waiting on Terminator.			
9.2.126	DUT to CSIPP, CSP/CSIPP calls DUT Call waiting on Originator.	DUT indicates incoming call	Pass	
9.2.127	CSIPP to DUT, CSP/CSIPP calls DUT Call waiting on Terminator.	DUT indicates incoming call	Pass	
Blind Call Transfer				
9.2.128	DUT to DUT2, Originator transfer to a second extension (DUT3)	DUT properly released, 2-way audio between DUT2 and extension	Pass	
9.2.129	DUT to DUT2, Originator failed to transfer call to a second extension (transfer call to an invalid number)	DUT properly released, DUT2 receives treatment, DUT2 properly released	Pass	Transfer to an invalid extension fails and the call is handed over back to the transferor (Originator) and the initial call is resumed.
9.2.130	DUT to DUT2, Originator transfer to a second extension (DUT3), release before answer	All legs properly released	Pass	
9.2.131	DUT to DUT2, Terminator transfer to a second extension (DUT3)	DUT2 properly released, 2-way audio between DUT and extension	Pass	
9.2.132	DUT to DUT2, Terminator failed to transfer call to a second extension(transfer call to an invalid number)	DUT2 properly released, DUT receives treatment, DUT properly released	Pass	Transfer to an invalid extension fails and the call is handed over back to the transferor (Terminator) and the initial call is resumed.
9.2.133	DUT to DUT2, Terminator transfer to	All legs properly released	Pass	

	a second extension(DUT3), Originator release before answer			
9.2.134	DUT to DUT2, Originator transfer to PSTN	DUT properly released, 2-way audio between DUT2 and PSTN	Pass	
9.2.135	DUT to DUT2, Originator failed to transfer call to PSTN (transfer call to an invalid PSTN number)	DUT properly released, DUT2 receives treatment, DUT2 properly released	Pass	Transfer to an invalid extension fails and the call is handed over back to the transferor (Originator) and the initial call is resumed.
9.2.136	DUT to DUT2, Originator transfer to PSTN, release before answer	All legs properly released	Pass	
9.2.137	DUT to DUT2, Terminator transfer to PSTN	DUT2 properly released, 2-way audio between DUT and PSTN	Pass	
9.2.138	DUT to DUT2, Terminator failed to transfer call to PSTN (transfer call to an invalid PSTN number)	DUT2 properly released, DUT receives treatment, DUT properly released	Pass	
9.2.139	DUT to DUT2, Terminator transfer to PSTN, Originator release before answer	All legs properly released	Pass	
9.2.140	PSTN to DUT, terminator transfer to second extension(DUT2)	DUT properly released, 2-way audio between PSTN and extension	Pass	
9.2.141	PSTN to DUT, terminator failed to transfer call(transfer call to an invalid number)	DUT properly released, PSTN receives treatment, PSTN properly released	Pass	Transfer call fails and the original call is resumed.
9.2.142	PSTN to DUT, terminator transfer(transfer call to DUT2), PSTN	All legs properly released	Pass	

	releases before call is answered			
9.2.143	DUT to PSTN, originator transfer to second extension(DUT2)	DUT properly released, 2-way audio between PSTN and extension	Pass	
9.2.144	DUT to PSTN, originator failed to transfer call (transfer call to an invalid number)	DUT properly released, PSTN receives treatment, PSTN properly released	Pass	
9.2.145	DUT to PSTN, originator transfer(transfer call to DUT2), PSTN releases before call is answered	All legs properly released	Pass	
9.2.146	DUT to CSP, Originator transfer to a second extension (DUT2)	DUT properly released, 2-way audio between CSP and extension	Pass	
9.2.147	DUT to CSP, Originator failed to transfer call to a second extension (DUT2)	DUT properly released, CSP receives treatment, CSP properly released	Pass	Transfer to an invalid extension fails and the call is handed over back to the transferor (Originator) and the initial call is resumed.
9.2.148	DUT to CSP, Originator transfer to a second extension(DUT2), release before answer	All legs properly released	Pass	
9.2.149	DUT to CSP, Originator transfer to PSTN	DUT properly released, 2-way audio between CSP and PSTN	Pass	
9.2.150	DUT to CSP, Originator failed to transfer call to PSTN(transfer call to an invalid PSTN number)	DUT properly released, CSP receives treatment, CSP properly released	Pass	Call hangs-up after the transfer.
9.2.151	DUT to CSP, Originator transfer to	All legs properly released	Pass	

	PSTN, release before answer			
9.2.152	DUT to CSIPP, Originator transfer to a second extension (DUT2)	DUT properly released, 2-way audio between CSIPP and extension	Pass	
9.2.153	DUT to CSIPP, Originator failed to transfer call to a second extension(transfer call to an invalid number)	DUT properly released, CSIPP receives treatment, CSIPP properly released	Pass	Transfer to an invalid extension fails and the call is handed over back to the transferor (Originator) and the initial call is resumed.
9.2.154	DUT to CSIPP, Originator transfer to a second extension(DUT2), release before answer	All legs properly released	Pass	
9.2.155	DUT to CSIPP, Terminator transfer to a second extension (DUT2)	CSIPP properly released, 2-way audio between DUT and extension	Pass	
9.2.156	DUT to CSIPP, Terminator failed to transfer call to a second extension(transfer call to an invalid extension)	CSIPP properly released, DUT receives treatment, DUT properly released	Pass	Call hangs-up after the transfer.
9.2.157	DUT to CSIPP, Terminator transfer to a second extension(DUT2), Originator release before answer	All legs properly released	Pass	
9.2.158	DUT to CSIPP, Originator transfer to PSTN	DUT properly released, 2-way audio between CSIPP and PSTN	Pass	
9.2.159	DUT to CSIPP, Originator failed to transfer call to PSTN(transfer call to an invalid PSTN number)	DUT properly released, CSIPP receives treatment, CSIPP properly released	Pass	Transfer to an invalid number fails and the call is handed over back to the transfer (Originator) and the initial call is resumed.

9.2.160	DUT to CSIPP, Originator transfer to PSTN, release before answer	All legs properly released	Pass	
9.2.161	DUT to CSIPP, Terminator transfer to PSTN	CSIPP properly released, 2-way audio between DUT and PSTN	Pass	
9.2.162	DUT to CSIPP, Terminator failed to transfer call to PSTN(transfer call to an invalid PSTN number)	CSIPP properly released, DUT receives treatment, DUT properly released	Pass	Call hangs-up after the transfer.
9.2.163	DUT to CSIPP, Terminator transfer to PSTN, Originator release before answer	All legs properly released	Pass	
9.2.164	DUT to DUT2, Originator transfer to a second extension (DUT3),DUT3 is busy	Originator hears busy tone, call released properly	Pass	Busy tone is heard on the transferor and the initial call is Resumed.
9.2.165	PSTN to DUT, DUT transfer to PSTN2	DUT properly released, 2-way audio between PSTN and PSTN2	Pass	
9.2.166	CSP to DUT, CSP transfer to CSIPP	CSP properly released, 2-way audio between DUT and CSIPP	**N/A**	Blind transfer not available in CSP
Consultative Call Transfer				
9.2.167	DUT to DUT2, Originator transfer to a second extension (DUT3)	DUT2 receives TOH/silence 2-way audio DUT to extension, DUT properly released, 2-way audio DUT2 to extension	Pass	
9.2.168	DUT to DUT2, Originator failed to transfer call to a second extension(transfer	DUT2 receives TOH/silence, DUT receives treatment, call successfully retrieved with 2-way	N/A	

	call to an invalid number), retrieve call	audio, failed leg properly released		
9.2.169	DUT to DUT2, Originator transfer to a second extension(DUT3), release before answer	DUT2 receives TOH/silence, 2-way audio DUT to extension, all call legs properly released	N/A	
9.2.170	DUT to DUT2, Terminator transfer to a second extension (DUT3)	DUT receives TOH/silence, 2-way audio DUT2 to extension, DUT2 properly released, 2-way audio DUT to extension	Pass	
9.2.171	DUT to DUT2, Terminator failed to transfer call to a second extension(transfer call to an invalid number), retrieve call	DUT receives TOH/silence, DUT2 receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	
9.2.172	DUT to DUT2, Terminator transfer to a second extension(DUT3), Originator release before answer	DUT receives TOH/silence, 2-way audio DUT2 to extension, all call legs properly released	N/A	Executed this in blind transfer
9.2.173	DUT to DUT2, Originator transfer to PSTN	DUT2 receives TOH/silence, 2-way audio DUT to PSTN, DUT properly released, 2-way audio DUT2 to PSTN	Pass	
9.2.174	DUT to DUT2, Originator failed to transfer call to PSTN(transfer call to an invalid PSTN number), retrieve call	DUT2 receives TOH/silence, DUT receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	
9.2.175	DUT to DUT2, Originator transfer to PSTN, release before answer	DUT2 receives TOH/silence, 2-way audio DUT to PSTN, all call legs properly released	**N/A**	Executed this in blind transfer

9.2.176	DUT to DUT2, Terminator transfer to PSTN	DUT receives TOH/silence, 2-way audio DUT2 to PSTN, DUT2 properly released, 2-way audio DUT to PSTN	Pass	
9.2.177	DUT to DUT2, Terminator failed to transfer call to PSTN(transfer call to an invalid PSTN number), retrieve call	DUT receives TOH/silence, DUT2 receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	
9.2.178	DUT to DUT2, Terminator transfer to PSTN, Originator release before answer	DUT receives TOH/silence, 2-way audio DUT2 to PSTN, all call legs properly released	**N/A**	Executed this in blind transfer
9.2.179	PSTN to DUT, terminator transfer to second extension(DUT2)	PSTN receives Silence, 2-way audio DUT to extension, DUT properly released, 2-way audio PSTN to extension	Pass	
9.2.180	PSTN to DUT, terminator failed to transfer call(transfer call to an invalid number)	PSTN receives silence, DUT receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	
9.2.181	PSTN to DUT, terminator transfer, PSTN releases before call is answered	PSTN receives silence, 2-way audio DUT to extension, all call legs properly released	**N/A**	Executed this in blind transfer
9.2.182	DUT to PSTN, originator transfer to second extension(DUT2)	PSTN receives silence, 2-way audio DUT to extension, DUT properly released, 2-way audio PSTN to extension	Pass	
9.2.183	DUT to PSTN, originator failed to transfer call(transfer	PSTN receives silence, DUT receives treatment, call successfully	Pass	

	call to an invalid number)	retrieved with 2-way audio, failed leg properly released		
9.2.184	DUT to PSTN, originator transfer(transfer call to DUT2), PSTN releases before call is answered	PSTN receives silence, 2-way audio DUT to extension, all call legs properly released	**N/A**	Executed this in blind transfer
9.2.185	DUT to CSP, Originator transfer to a second extension (DUT2)	CSP receives silence, 2-way audio DUT to extension, DUT properly released, 2-way audio CSP to extension	Pass	
9.2.186	DUT to CSP, Originator failed to transfer call to a second extension(transfer call to an invalid number), retrieve call	CSP receives silence, DUT receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	
9.2.187	DUT to CSP, Originator transfer to a second extension(DUT2), release before answer	CSP receives silence, 2-way audio DUT to extension, all call legs properly released	**N/A**	Executed this in blind transfer
9.2.188	DUT to CSP, Terminator transfer to a second extension (DUT2)	DUT receives MOH , 2-way audio DUT to extension, DUT properly released, 2-way audio CSP to extension	Pass	
9.2.189	DUT to CSP, Terminator failed to transfer call to a second extension(transfer call to an invalid number) , retrieve call	DUT receives MOH , DUT receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	
9.2.190	DUT to CSP, Terminator transfer to a second extension(DUT2),	DUT receives MOH , 2-way audio DUT to extension, all call legs properly released	**N/A**	Executed this in blind transfer

	Originator release before answer			
9.2.191	DUT to CSP, Originator transfer to PSTN	CSP receives silence, 2-way audio DUT to PSTN, DUT properly released, 2-way audio CSP to PSTN	Pass	
9.2.192	DUT to CSP, Originator failed to transfer call to PSTN(transfer call to an invalid PSTN number), retrieve call	CSP receives silence, DUT receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	
9.2.193	DUT to CSP, Originator transfer to PSTN, release before answer	CSP receives silence, 2-way audio DUT to PSTN, all call legs properly released	**N/A**	Executed this in blind transfer
9.2.194	DUT to CSP, Terminator transfer to PSTN	DUT receives MOH, 2-way audio CSP to PSTN, CSP properly released, 2-way audio DUT to PSTN	Pass	
9.2.195	DUT to CSP, Terminator failed to transfer call to PSTN(Transfer call to an invalid PSTN number), retrieve call	DUT receives MOH, CSP receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	
9.2.196	DUT to CSP, Terminator transfer to PSTN, Originator release before answer	DUT receives MOH, 2-way audio CSP to PSTN, all call legs properly released	**N/A**	Executed this in blind transfer
9.2.197	DUT to CSIPP, Originator transfer to a second extension (DUT2)	CSIPP receives silence, 2-way audio DUT to extension, DUT properly released, 2-way audio CSIPP to extension	Pass	
9.2.198	DUT to CSIPP, Originator failed to transfer call to a	CSIPP receives silence, DUT receives treatment,	Pass	

	second extension(transfer call to an invalid number), retrieve call	call successfully retrieved with 2-way audio, failed leg properly released		
9.2.199	DUT to CSIPP, Originator transfer to a second extension(DUT2), release before answer	CSIPP receives silence, 2-way audio DUT to extension, all call legs properly released	**N/A**	Executed this in blind transfer
9.2.200	DUT to CSIPP, Terminator transfer to a second extension (DUT2)	DUT receives MOH , 2-way audio DUT to extension, DUT properly released, 2-way audio CSIPP to extension	Pass	
9.2.201	DUT to CSIPP, Terminator failed to transfer call to a second extension(transfer call to an invalid number), retrieve call	DUT receives Silence , DUT receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	
9.2.202	DUT to CSIPP, Terminator transfer to a second extension(DUT2), Originator release before answer	DUT receives MOH , 2-way audio DUT to extension, all call legs properly released	**N/A**	Executed this in blind transfer
9.2.203	DUT to CSIPP, Originator transfer to PSTN	CSIPP receives silence, 2-way audio DUT to PSTN, DUT properly released, 2-way audio CSIPP to PSTN	Pass	
9.2.204	DUT to CSIPP, Originator failed to transfer call to PSTN(transfer call to an invalid PSTN number), retrieve call	CSIPP receives silence, DUT receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	
9.2.205	DUT to CSIPP, Originator transfer to PSTN, release before answer	CSIPP receives silence, 2-way audio DUT to PSTN, all call legs properly released	**N/A**	Executed this in blind transfer

9.2.206	DUT to CSIPP, Terminator transfer to PSTN	DUT receives MOH, 2-way audio CSIPP to PSTN, CSIPP properly released, 2-way audio DUT to PSTN	Pass	
9.2.207	DUT to CSIPP, Terminator failed to transfer call to PSTN (transfer call to an invalid PSTN number), retrieve call	DUT receives MOH, CSIPP receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	
9.2.208	DUT to CSIPP, Terminator transfer to PSTN, Originator release before answer	DUT receives MOH, 2-way audio CSIPP to PSTN, all call legs properly released	**N/A**	Executed this in blind transfer
9.2.209	DUT to DUT2, Originator transfer to a second extension (DUT3),DUT3 is busy	DUT2 hears busy tone, call released properly	Pass	
9.2.210	PSTN to DUT, DUT transfer to PSTN2	DUT properly released, 2-way audio between PSTN and PSTN2	Pass	
9.2.211	CSP to DUT, CSP transfer to CSIPP	CSP properly released, 2-way audio between DUT and CSIPP	Pass	
Semi - Unattended Transfer				
9.2.212	DUT to DUT2, Originator transfer to a second extension (DUT3)	DUT2 receives TOH/silence 2-way audio DUT to extension, DUT properly released, 2-way audio DUT2 to extension	Pass	While the transfer target phone is ringing, the transferee will hear MOH and once the Transfer target phone answers, the call will be connected.
9.2.213	DUT to DUT2, Originator failed to transfer call to a second extension(transfer call to an invalid number), retrieve call	DUT2 receives TOH/silence, DUT receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	Transfer call failed and the original call is Resumed.

9.2.214	DUT to DUT2, Originator transfer to a second extension(DUT3), release before answer	DUT2 receives TOH/silence, 2-way audio DUT to extension, all call legs properly released	Pass	
9.2.215	DUT to DUT2, Terminator transfer to a second extension (DUT3)	DUT receives TOH/silence, 2-way audio DUT2 to extension, DUT2 properly released, 2-way audio DUT to extension	Pass	
9.2.216	DUT to DUT2, Terminator failed to transfer call to a second extension(transfer call to an invalid number), retrieve call	DUT receives TOH/silence, DUT2 receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	Transfer call failed and the original call is Resumed.
9.2.217	DUT to DUT2, Terminator transfer to a second extension(DUT3), Originator release before answer	DUT receives TOH/silence, 2-way audio DUT2 to extension, all call legs properly released	Pass	
9.2.218	DUT to DUT2, Originator transfer to PSTN	DUT2 receives TOH/silence, 2-way audio DUT to PSTN, DUT properly released, 2-way audio DUT2 to PSTN	Pass	
9.2.219	DUT to DUT2, Originator failed to transfer call to PSTN(transfer call to an invalid PSTN number), retrieve call	DUT2 receives TOH/silence, DUT receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	Transfer call failed and the original call is Resumed.
9.2.220	DUT to DUT2, Originator transfer to PSTN, release before answer	DUT2 receives TOH/silence, 2-way audio DUT to PSTN, all call legs properly released	Pass	
9.2.221	DUT to DUT2, Terminator transfer to PSTN	DUT receives TOH/silence, 2-way audio DUT2 to PSTN, DUT2 properly	Pass	

		released, 2-way audio DUT to PSTN		
9.2.222	DUT to DUT2, Terminator failed to transfer call to PSTN(transfer call to an invalid PSTN number), retrieve call	DUT receives TOH/silence, DUT2 receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	Transfer call failed and the original call is Resumed.
9.2.223	DUT to DUT2, Terminator transfer to PSTN, Originator release before answer	DUT receives TOH/silence, 2-way audio DUT2 to PSTN, all call legs properly released	Pass	
9.2.224	PSTN to DUT, terminator transfer to second extension(DUT2)	PSTN receives silence, 2-way audio DUT to extension, DUT properly released, 2-way audio PSTN to extension	Pass	While the transfer target phone is ringing, the transferee will hear MOH and once the Transfer target phone answers, the call will be connected.
9.2.225	PSTN to DUT, terminator failed to transfer call(transfer call to an invalid number)	PSTN receives silence, DUT receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	Transfer call failed and the original call is Resumed.
9.2.226	PSTN to DUT, terminator transfer, PSTN releases before call is answered	PSTN receives silence, 2-way audio DUT to extension, all call legs properly released	Pass	
9.2.227	DUT to PSTN, originator transfer to second extension(DUT2)	PSTN receives silence, 2-way audio DUT to extension, DUT properly released, 2-way audio PSTN to extension	Pass	While the transfer target phone is ringing, the transferee will hear MOH and once the Transfer target phone answers, the call will be connected.
9.2.228	DUT to PSTN, originator failed to transfer call(transfer call to an invalid number)	PSTN receives silence, DUT receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	Transfer call failed and the original call is Resumed.

9.2.229	DUT to PSTN, originator transfer(transfer call to DUT2), PSTN releases before call is answered	PSTN receives silence, 2-way audio DUT to extension, all call legs properly released	Pass	
9.2.230	DUT to CSP, Originator transfer to a second extension (DUT2)	CSP receives silence, 2-way audio DUT to extension, DUT properly released, 2-way audio CSP to extension	Pass	While the transfer target phone is ringing, the transferee will hear MOH and once the Transfer target phone answers, the call will be connected.
9.2.231	DUT to CSP, Originator failed to transfer call to a second extension(transfer call to an invalid number), retrieve call	CSP receives silence, DUT receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	Transfer call failed and the original call is Resumed.
9.2.232	DUT to CSP, Originator transfer to a second extension(DUT2), release before answer	CSP receives silence, 2-way audio DUT to extension, all call legs properly released	Pass	
9.2.233	DUT to CSP, Terminator transfer to a second extension (DUT2)	DUT receives MOH, 2-way audio DUT to extension, DUT properly released, 2-way audio CSP to extension	Pass	After the transfer Transferee hears the ring back.
9.2.234	DUT to CSP, Terminator failed to transfer call to a second extension(transfer call to an invalid number) , retrieve call	DUT receives MOH, DUT receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	
9.2.235	DUT to CSP, Terminator transfer to a second extension(DUT2), Originator release before answer	DUT receives MOH, 2-way audio DUT to extension, all call legs properly released	Pass	

9.2.236	DUT to CSP, Originator transfer to PSTN	CSP receives silence, 2-way audio DUT to PSTN, DUT properly released, 2-way audio CSP to PSTN	Pass	
9.2.237	DUT to CSP, Originator failed to transfer call to PSTN(transfer call to an invalid PSTN number), retrieve call	CSP receives silence, DUT receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	
9.2.238	DUT to CSP, Originator transfer to PSTN, release before answer	CSP receives silence, 2-way audio DUT to PSTN, all call legs properly released	Pass	
9.2.239	DUT to CSP, Terminator transfer to PSTN	DUT receives MOH, 2-way audio CSP to PSTN, CSP properly released, 2-way audio DUT to PSTN	Pass	
9.2.240	DUT to CSP, Terminator failed to transfer call to PSTN(Transfer call to an invalid PSTN number), retrieve call	DUT receives MOH, CSP receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	
9.2.241	DUT to CSP, Terminator transfer to PSTN, Originator release before answer	DUT receives MOH, 2-way audio CSP to PSTN, all call legs properly released	Pass	
9.2.242	DUT to CSIPP, Originator transfer to a second extension (DUT2)	CSIPP receives silence, 2-way audio DUT to extension, DUT properly released, 2-way audio CSIPP to extension	Pass	
9.2.243	DUT to CSIPP, Originator failed to transfer call to a second extension(transfer	CSIPP receives silence, DUT receives treatment, call successfully retrieved with 2-way	Pass	Transfer call failed and the original call is Resumed.

	call to an invalid number), retrieve call	audio, failed leg properly released		
9.2.244	DUT to CSIPP, Originator transfer to a second extension(DUT2), release before answer	CSIPP receives silence, 2-way audio DUT to extension, all call legs properly released	Pass	
9.2.245	DUT to CSIPP, Terminator transfer to a second extension (DUT2)	DUT receives MOH, 2-way audio DUT to extension, DUT properly released, 2-way audio CSIPP to extension	Pass	After the transfer Transferee hears the ring back.
9.2.246	DUT to CSIPP, Terminator failed to transfer call to a second extension(transfer call to an invalid number), retrieve call	DUT receives MOH, DUT receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	
9.2.247	DUT to CSIPP, Terminator transfer to a second extension(DUT2), Originator release before answer	DUT receives MOH, 2-way audio DUT to extension, all call legs properly released	Pass	
9.2.248	DUT to CSIPP, Originator transfer to PSTN	CSIPP receives silence, 2-way audio DUT to PSTN, DUT properly released, 2-way audio CSIPP to PSTN	Pass	
9.2.249	DUT to CSIPP, Originator failed to transfer call to PSTN(transfer call to an invalid PSTN number), retrieve call	CSIPP receives silence, DUT receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	Transfer call failed and the original call is Resumed.
9.2.250	DUT to CSIPP, Originator transfer to PSTN, release before answer	CSIPP receives silence, 2-way audio DUT to PSTN, all call legs properly released	Pass	

9.2.251	DUT to CSIPP, Terminator transfer to PSTN	DUT receives MOH, 2-way audio CSIPP to PSTN, CSIPP properly released, 2-way audio DUT to PSTN	Pass	Ringback is head after doing transfer from CSIP phone
9.2.252	DUT to CSIPP, Terminator failed to transfer call to PSTN (transfer call to an invalid PSTN number), retrieve call	DUT receives MOH, CSIPP receives treatment, call successfully retrieved with 2-way audio, failed leg properly released	Pass	
9.2.253	DUT to CSIPP, Terminator transfer to PSTN, Originator release before answer	DUT receives MOH, 2-way audio CSIPP to PSTN, all call legs properly released	Pass	
3-Way Conference				
9.2.254	DUT to DUT2, Originator is the bridge(DUT bridges the call with DUT3) g) Originator drops out of the conference first h) DUT2 drops out of conference first i) DUT3 drops out of conference first	DUT2 receives TOH/silence DUT receives ring back, 3-way audio Other 2 parties remain on conference with 2-way audio.	A- Pass B- Pass C- Pass	
9.2.255	DUT to DUT2, Originator failed to bridged call to a DUT3(bridges the call to an invalid number), retrieve call	DUT2 receives TOH/silence, DUT receives treatment, call successfully retrieved with 2-way audio	Pass	
9.2.256	DUT to DUT2, Originator is the bridge(DUT bridges the call with DUT3), Terminator release before answer	DUT2 receives TOH/silence, DUT receives ring back, DUT2 properly released, 2-way audio DUT – DUT3 after answer	N/S	Conference can be originated only after the other party answers the call.

9.2.257	DUT to DUT2, Originator is the bridge(DUT bridges the call with DUT3), Originator cancel 3 way call, retrieve original call	DUT2 receives TOH/silence, DUT receives ring back, call successfully retrieved with 2-way audio	N/S	
9.2.258	DUT to DUT2, Terminator is the bridge(DUT2 bridges the call with DUT3) a) DUT2 drops out of the conference first b) DUT drops out of conference first c) DUT3 drops out of conference first	DUT receives TOH/silence, DUT2 receives ring back, 3-way audio Other 2 parties remain on conference with 2-way audio.	Pass	
9.2.259	DUT to DUT2, Terminator failed to bridged call(DUT2 bridges the call with an invalid number), retrieve original call	DUT receives TOH/silence, DUT2 receives treatment, call successfully retrieved with 2-way audio	Pass	DUT2 reports as Vacant and the original can be retrieved.
9.2.260	DUT to DUT2, Terminator bridged to a DUT3, Originator releases before answer	DUT receives TOH/silence, DUT2 receives ring back, DUT properly released, 2-way audio DUT2 – DUT3 after answer	N/S	Conference can be originated only after the other party answers the call.
9.2.261	DUT to DUT2, Originator bridge to PSTN a) DUT drops out of the conference first b) DUT2 drops out of conference first c) PSTN drops out of conference first	DUT2 receives TOH/silence, DUT receives ring back, 3-way audio Other 2 parties remain on conference with 2-way audio.	Pass	
9.2.262	DUT to DUT2, Originator failed to bridge call to	DUT2 receives TOH/silence, DUT receives treatment,	Pass	

	PSTN(DUT bridges the call with an invalid PSTN number), retrieve call	call successfully retrieved with 2-way audio		
9.2.263	DUT to DUT2, Originator bridge to PSTN, terminator releases before answer	DUT2 receives TOH/silence, DUT receives ring back, DUT2 properly released, 2-way audio DUT – PSTN after answer	N/S	Conference can be originated only after the other party answers the call.
9.2.264	DUT to DUT2, Terminator bridge to PSTN a) DUT2 drops out of the conference first b) DUT drops out of conference first c) PSTN drops out of conference first	DUT receives TOH/silence; DUT2 receives ring back, 3-way audio. Other 2 parties remain on conference with 2-way audio.	A-Pass B-Pass C-Pass	
9.2.265	DUT to DUT2, Terminator failed to bridge call to PSTN(DUT2 bridges the call with an invalid PSTN number), retrieve call	DUT receives TOH/silence, DUT2 receives treatment, call successfully retrieved with 2-way audio	Pass	
9.2.266	DUT to DUT2, Terminator bridge to PSTN, Originator release before answer	DUT receives TOH/silence, DUT2 receives ring back, DUT properly released, 2-way audio DUT2 – PSTN after answer	N/S	Conference can be originated only after the other party answers the call.
9.2.267	PSTN to DUT, terminator bridge to DUT2 a) DUT drops out of the conference first b) DUT2 drops out of conference first	PSTN receives silence, DUT receives ring back, 3-way audio Other 2 parties remain on conference with 2-way audio.	A-Pass B-Pass C-Pass	

	c) PSTN drops out of conference first			
9.2.268	PSTN to DUT, terminator failed to bridge call(DUT bridges the call with an invalid number)	PSTN receives silence, DUT receives treatment, call successfully retrieved with 2-way audio	Pass	
9.2.269	PSTN to DUT, terminator attempt to bridge(DUT bridges the call with DUT2), PSTN releases before call is answered	PSTN receives silence, DUT receives ring back, PSTN properly released, 2-way audio DUT – DUT2 after answer	N/S	Conference can be originated only after the other party answers the call.
9.2.270	DUT to PSTN, originator bridge to DUT2 a) DUT drops out of the conference first b) DUT2 drops out of conference first c) PSTN drops out of conference first	PSTN receives silence; DUT receives ring back, 3-way audio. Other 2 parties remain on conference with 2-way audio	A-Pass B-Pass C-Pass	
9.2.271	DUT to PSTN, originator failed to bridge call(DUT bridges the call with an invalid number)	PSTN receives silence, DUT receives treatment, call successfully retrieved with 2-way audio	Pass	
9.2.272	DUT to PSTN, originator attempts to bridge call(DUT bridges the call with DUT2), PSTN releases before call is answered	PSTN receives silence, DUT receives ring back, PSTN properly released, 2-way audio DUT – DUT2 after answer	N/S	Conference can be originated only after the other party answers the call.
9.2.273	DUT to CSP, Originator is the bridge (DUT bridges the call with DUT2)	CSP receives silence; DUT receives ring back, 3-way audio.	A-Pass	

	a) DUT drops out of the conference first b) CSP drops out of conference first c) DUT2 drops out of conference first	Other 2 parties remain on conference with 2-way audio	B- Pass C- Pass	
9.2.274	DUT to CSP, Originator failed to bridged call to a DUT2(DUT bridges the call with DUT2,an invalid number), retrieve call	CSP receives silence, DUT receives treatment, call successfully retrieved with 2-way audio	Pass	
9.2.275	DUT to CSP, Originator is the bridge(DUT bridges the call with DUT2), Terminator release before answer	CSP receives silence, DUT receives ring back, CSP properly released, 2-way audio DUT – DUT2 after answer	N/S	Conference can be originated only after the other party answers the call.
9.2.276	DUT to CSP, Originator is the bridge(DUT bridges the call with DUT2), Originator cancel 3 way call, retrieve original call	CSP receives silence, DUT receives ring back, call successfully retrieved with 2-way audio	N/S	
9.2.277	DUT to CSP, Terminator is the bridge(CSP bridges the call with DUT2) a) CSP drops out of the conference first b) DUT drops out of conference first c) DUT2 drops out of conference first	DUT receives MOH; CSP receives ring back, 3-way audio. Other 2 parties remain on conference with 2-way audio	A- Pass B- Pass C- Pass	
9.2.278	DUT to CSP, Terminator failed to bridged call(CSP	DUT receives MOH, CSP receives treatment, call	Pass	

	bridges the call to an invalid number), retrieve original call	successfully retrieved with 2-way audio		
9.2.279	DUT to CSP, Terminator bridged to a DUT2, Originator releases before answer	DUT receives MOH, CSP receives ring back, DUT properly released, 2-way audio CSP – DUT2 after answer	Pass	
9.2.280	DUT to CSP, Originator bridge to PSTN a) DUT drops out of the conference first b) CSP drops out of conference first c) PSTN drops out of conference first	CSP receives silence; DUT receives ring back, 3-way audio. Other 2 parties remain on conference with 2-way audio	A- Pass B- Pass C- Pass	
9.2.281	DUT to CSP, Originator failed to bridge call to PSTN(DUT bridge call to an invalid PSTN number), retrieve call	CSP receives silence, DUT receives treatment, call successfully retrieved with 2-way audio	Pass	
9.2.282	DUT to CSP, Originator bridge to PSTN, terminator releases before answer	CSP receives silence, DUT receives ring back, CSP properly released, 2-way audio DUT – PSTN after answer	N/S	Conference can be originated only after the other party answers the call.
9.2.283	DUT to CSP, Terminator bridge to PSTN a) DUT drops out of the conference first b) CSP drops out of conference first c) PSTN drops out of conference first	DUT receives silence; CSP receives ring back, 3-way audio. Other 2 parties remain on conference with 2-way audio	A- Pass B- Pass C- Pass	

9.2.284	DUT to CSP, Terminator failed to bridge call to PSTN(CSP bridge call to an invalid number), retrieve call	DUT receives silence, CSP receives treatment, call successfully retrieved with 2-way audio	Pass	
9.2.285	DUT to CSP, Terminator bridge to PSTN, Originator release before answer	DUT receives silence, CSP receives ring back, DUT properly released, 2-way audio CSP – PSTN after answer	Pass	
9.2.286	DUT to CSIPP, Originator is the bridge(DUT bridge call with DUT2) a) DUT drops out of the conference first b) CSIPP drops out of conference first c) DUT2 drops out of conference first	CSIPP receives silence; DUT receives ring back, 3-way audio. Other 2 parties remain on conference with 2-way audio	A- Pass B- Pass C- Pass	
9.2.287	DUT to CSIPP, Originator failed to bridged call to a DUT3(DUT bridge call with DUT3, an invalid number), retrieve call	CSIPP receives silence, DUT receives treatment, call successfully retrieved with 2-way audio	Pass	
9.2.288	DUT to CSIPP, Originator is the bridge(DUT bridge call with DUT2), Terminator release before answer	CSIPP receives silence, DUT receives ring back, CSIPP properly released, 2-way audio DUT – DUT2 after answer	N/S	Conference can be originated only after the other party answers the call.
9.2.289	DUT to CSIPP, Originator is the bridge(DUT bridge call with DUT2), Originator cancel 3	CSIPP receives silence, DUT receives ring back, call successfully	N/S	

	way call, retrieve original call	retrieved with 2-way audio		
9.2.290	DUT to CSIPP, Terminator is the bridge(CSIPP bridge call with DUT2) a) DUT drops out of the conference first b) CSIPP drops out of conference first c) DUT2 drops out of conference first	DUT receives MOH; CSIPP receives ring back, 3-way audio. Other 2 parties remain on conference with 2-way audio	A- Pass B- Pass C- Pass	
9.2.291	DUT to CSIPP, Terminator failed to bridged call(CSIPP bridge call to an unknown number), retrieve original call	DUT receives MOH, CSIPP receives treatment, call successfully retrieved with 2-way audio	Pass	
9.2.292	DUT to CSIPP, Terminator bridged to a DUT3, Originator releases before answer	DUT receives MOH, CSIPP receives ring back, DUT properly released, 2-way audio CSIPP – DUT2 after answer	Pass	
9.2.293	DUT to CSIPP, Originator bridge to PSTN a) DUT drops out of the conference first b) CSIPP drops out of conference first c) PSTN drops out of conference first	CSIPP receives silence; DUT receives ring back, 3-way audio. Other 2 parties remain on conference with 2-way audio	A- Pass B- Pass C- Pass	
9.2.294	DUT to CSIPP, Originator failed to bridge call to PSTN(DUT bridge call to an invalid PSTN number), retrieve call	CSIPP receives silence, DUT receives treatment, call successfully retrieved with 2-way audio	Pass	

9.2.295	DUT to CSIPP, Originator bridge to PSTN, terminator releases before answer	CSIPP receives silence, DUT receives ring back, CSIPP properly released, 2-way audio DUT – PSTN after answer	N/S	Conference can be originated only after the other party answers the call.
9.2.296	DUT to CSIPP, Terminator bridge to PSTN a) DUT drops out of the conference first b) CSIPP drops out of conference first c) PSTN drops out of conference first	DUT receives MOH; CSIPP receives ring back, 3-way audio. Other 2 parties remain on conference with 2-way audio	A- Pass B- Pass C- Pass	
9.2.297	DUT to CSIPP, Terminator failed to bridge call to PSTN(CSIPP bridge call to an invalid PSTN number), retrieve call	DUT receives silence, CSIPP receives treatment, call successfully retrieved with 2-way audio	Pass	
9.2.298	DUT to CSIPP, Terminator bridge to PSTN, Originator release before answer	DUT receives MOH, CSIPP receives ring back, DUT properly released, 2-way audio CSIPP – PSTN after answer	Pass	
9.2.299	DUT to DUT2, Originator is the bridge(DUT bridges the call with DUT3) ,DUT3 busy	DUT2 receives TOH/silence DUT receives busy tone, All calls released properly.	Pass	
9.2.300	PSTN to DUT, Terminator is the bridge(DUT bridges the call with PSTN2)	PSTN receives Silence; DUT receives ring back, 3-way audio.	Pass	
9.2.301	CSP to DUT, Originator is the	CSP receives Silence; DUT	Pass	

	bridge(CSP bridges the call with CSIPP)	receives ring back, 3-way audio.		
Calling Line Identification				
9.2.302	DUT to DUT2, Calling Line identification type I(basic calls)	Calling Line ID presented at terminator	Pass	
9.2.303	CSIPP to DUT2, Calling Line identification type I(basic calls)	Calling Line ID presented at terminator	Pass	
9.2.304	CSP to DUT2, Calling Line identification type I(basic calls)	Calling Line ID presented at terminator	Pass	
9.2.305	DUT to PSTN, Calling Line identification type I(basic calls)	Calling Line ID presented at terminator	Pass	
9.2.306	PSTN to DUT, Calling Line identification type I(basic calls)	Calling Line ID presented at terminator	Pass	
9.2.307	DUT to DUT2, Type II(call waiting) on terminator, call in progress between two end points	Calling Line ID presented at terminator	Pass	
9.2.308	DUT to DUT2, Type II (call waiting) on originator, call in progress between two end points	Calling Line ID presented at terminator	Pass	
9.2.309	DUT to DUT2, Type II(call waiting) on originator, call in progress between End point and PSTN	Calling Line ID presented at terminator	Pass	
9.2.310	DUT to DUT2, Type II(call waiting) on terminator, call in progress between PSTN and End point	Calling Line ID presented at terminator	Pass	

9.2.311	DUT to DUT2, Calling Line identification type I(basic call) Restricted	Calling Line ID *not* presented at terminator	Pass	
9.2.312	CSIPP to DUT2, Calling Line identification type I (basic call) Restricted	Calling Line ID *not* presented at terminator	Pass	
9.2.313	CSP to DUT2, Calling Line identification type I (basic call) Restricted	Calling Line ID *not* presented at terminator	Pass	
9.2.314	Multiline DUT (single line presented, others not) to DUT Line Presented (Calling party)	Calling Line ID presented at terminator	N/A	
9.2.315	Multiline DUT (single line presented, others not) DUT to DUT Name Presented (Calling Party)	Calling Name presented at terminator	N/A	
9.2.316	Multiline DUT (single line presented, others not) DUT to DUT line and Name Presented (Calling Party)	Calling Line ID and Calling Name presented at terminator	N/A	
9.2.317	Multiline DUT (single line presented, others not) DUT to DUT Line restricted (Calling Party)	Calling Line ID *not* presented at terminator	N/A	
9.2.318	Multiline DUT (single line presented, others not) DUT to DUT Name restricted (Calling Party)	Calling Name *not* presented at terminator	N/A	
9.2.319	Multiline DUT (single line presented, others not) DUT to DUT Line and Name restricted (Calling Party)	Calling Line ID and Name *not* presented at terminator	N/A	
9.2.320	Call Forward (DUT to DUT2, forwarded to DUT3)	Calling Line ID presented at DUT3	Pass	

9.2.321	Call Transfer (DUT to DUT2, originator transfer to DUT3)	Calling Line ID presented at DUT3	Pass	
9.2.322	Call Transfer (DUT to DUT2, terminator transfer to DUT3)	Calling Line ID presented at DUT3	Pass	
Calling Name Presentation				
9.2.323	DUT to DUT2, Calling Party Name	Calling Name presented at terminator	Pass	
9.2.324	DUT to DUT2, Calling Party Name Restricted	Calling Name *not* presented at terminator	Pass	
9.2.325	CSP to DUT, Calling Party Name	Calling Name presented at terminator	Pass	
9.2.326	CSP to DUT, Calling Party Name Restricted	Calling Name *not* presented at terminator	Pass	
9.2.327	CSIPP to DUT, Calling Party Name	Calling Name presented at terminator	Pass	
9.2.328	CSIPP to DUT, Calling Party Name Restricted	Calling Name *not* presented at terminator	Pass	
9.2.329	DUT to PSTN, Calling Party Name	Calling Name presented at terminator	Pass	
9.2.330	PSTN to DUT, Calling Party Name	Calling Name presented at terminator	Pass	
9.2.331	Multiline DUT (single line presented, others not) to DUT Line Presented (Calling party)	Calling Line ID presented at terminator	**N/A**	
9.2.332	Multiline DUT (single line presented, others not) DUT to DUT Name Presented (Calling Party)	Calling Name presented at terminator	**N/A**	

9.2.333	Multiline DUT (single line presented, others not) DUT to DUT line and Name Presented (Calling Party)	Calling Line ID and Calling Name presented at terminator	**N/A**	
9.2.334	Multiline DUT (single line presented, others not) DUT to DUT Line restricted (Calling Party)	Calling Line ID *not* presented at terminator	**N/A**	
9.2.335	Multiline DUT (single line presented, others not) DUT to DUT Name restricted (Calling Party)	Calling Name *not* presented at terminator	**N/A**	
9.2.336	Multiline DUT (single line presented, others not) DUT to DUT Line and Name restricted (Calling Party)	Calling Line ID and Calling Name *not* presented at terminator	**N/A**	
9.2.337	Call Forward (DUT to DUT2, forwarded to DUT3)	Calling Name presented at DUT3	Pass	
9.2.338	Call Transfer (DUT to DUT2, originator transfer to DUT3)	Calling Name presented at DUT3	Pass	
9.2.339	Call Transfer (DUT to DUT2, terminator transfer to DUT3)	Calling Name presented at DUT3	Pass	
Multiple Lines per Phone (Applicable to advanced 3rd part SIP devices only)				
9.2.340	DUT1 line 1 calls DUT2 line 1, DUT1 line 2 calls DUT2 line 2, etc. (all DUT1 lines occupied). Alternate between calls.	2-way audio on each call. Held calls receive MOH. Connected Line ID and Name display reflects current call.	**N/A**	
9.2.341	DUT line 1 calls CSP, DUT line 2 calls CSP, etc. (all DUT1 lines occupied). Alternate between calls.	2-way audio on each call. Held calls receive MOH. Connected Line ID and Name display reflects current call.	**N/A**	
9.2.342	DUT line 1 calls CSIPP, DUT line 2	2-way audio on each call. Held calls	**N/A**	

	calls CSIPP, etc. (all DUT1 lines occupied). Alternate between calls.	receive MOH. Connected Line ID and Name display reflects current call.		
9.2.343	DUT2 calls DUT1 line 1, CSP calls DUT1 line 2, CSIPP calls DUT1 line 3, PSTN calls DUT1 line 4. Alternate between calls. All originators release.	2-way audio on each call. Held calls receive MOH. Calling Line ID and Name display reflects current call. All calls properly released.	**N/A**	
9.2.344	DUT1 line 1 calls DUT2, DUT1 line 2 calls CSP, DUT1 line 3 calls CSIPP, and DUT1 line 4 calls PSTN. Alternate between calls. All originators release.	2-way audio on each call. Held calls receive MOH. Connected Line ID and Name display reflects current call. All calls properly released.	**N/A**	
9.2.345	Line Busy: CFWD Busy Line 1 to Line 2, CFWD Busy Line 2 to Line 3, Line 1 in call with Line 2, CSP calls Line 1	2-way audio between CSP and Line 3.	**N/A**	
9.2.346	CSP calls DUT line 1. CSIPP calls DUT line 2. Place line 1 on hold, answer line 2. Bring line 1 into conference.	3-way audio between parties.	**N/A**	
9.2.347	CSP calls DUT line 1. Blind transfer to line 2. Pick up line 2.	Line 1 properly released. 2-way audio with Line 2.	**N/A**	
9.2.348	MWI – Single line: leave VMAIL on line 2, retrieve and delete line 2 VMAIL	Line 2 message waiting is indicated, indicator cleared when VMAIL deleted	**N/A**	
9.2.349	MWI – All lines: leave VMAIL on all lines, retrieve and delete each	MWI consistent with voice mail state per line.	**N/A**	
Call Forking				

9.2.350	DUT1 to DUT2, DUT3 Call Forking – DUT2 Answers	2-way audio, DUT3 ring stop	**N/A**	
9.2.351	DUT1 to DUT2, DUT3 Call Forking – DUT3 Answers	2-way audio, DUT2 ring stop	**N/A**	
9.2.352	DUT1 to DUT2, CSIPP Call Forking – DUT2 Answers.	2-way audio, CSIPP ring stop	**N/A**	
9.2.353	DUT1 to DUT2, CSIPP Call Forking – CSIPP Answers.	2-way audio, DUT2 ring stop	**N/A**	
9.2.354	DUT1 to DUT2, CSP Call Forking – DUT2 Answers.	2-way audio, CSP ring stop	**N/A**	
9.2.355	DUT1 to DUT2, CSP Call Forking – CSP Answers.	2-way audio, DUT2 ring stop	**N/A**	
9.2.356	DUT1 to DUT2, DUT3 Call Forking, Race condition– DUT2 and DUT3 Answer	2-way audio, other phone ring stop	**N/A**	
9.2.357	DUT1 to DUT2, CSIPP Call Forking, Race condition– DUT2 and CSIPP Answer	2-way audio, other phone ring stop	**N/A**	
9.2.358	DUT1 to DUT2, CSP Call Forking, Race condition– DUT2 and CSP Answer	2-way audio, other phone ring stop	**N/A**	
9.2.359	DUT1 to DUT2, DUT3 Call Forking – DUT2 CFWD all to DUT3. DUT3 Answers	Only DUT3 rings, 2-way audio DUT1 to DUT3	**N/A**	
9.2.360	DUT1 to DUT2, DUT3 Call Forking – DUT2 Busy. DUT3 Answers	Only DUT3 rings, 2-way audio DUT1 to DUT3	**N/A**	
9.2.361	Call Forking – Multiline DUT	Two lines on DUT ring, 2-way audio on	**N/A**	Applicable to advanced 3rd party SIP devices only

		answered line, ring stop on other line		
9.2.362	**6.2.316** Call Forking – Multi call DUT (Call waiting)	Call waiting indication on in-use line, 2-way audio on answered line, call waiting indication ends when call answered	**N/A**	
Mobile Handoff				
9.2.363	Mobile Handoff: Configure a Cisco Unified Mobile Communicator and the DUT as shared lines. PSTN calls the shared line. Answer the call from mobile. Handoff the call to the DUT	Verify both the end points ring. Verify the call is handed off to DUT	**N/S**	
9.2.364	Mobile Handoff: Part 2 Configure a Cisco Unified Mobile Communicator and DUT as shared lines. PSTN calls the shared line. DUT answers the call. DUT hands off the call to mobile.	Verify the hand off works correctly.	**N/S**	
Message Waiting Indicator				
9.2.365	Call DUT, leave voice mail	MWI activated	Pass	
9.2.366	Retrieve only voice mail	MWI cleared	Pass	
9.2.367	Call DUT, leave 2 voice mails, retrieve one, hang up,	MWI activates with first voice mail, stays active after first	Pass	

	retrieve second voice mail, hang up	retrieve, clears after second retrieve		
9.2.368	Active call, second inbound call goes to VM, leave voice mail	MWI activates while in call	Pass	
Speed Dial				
9.2.369	Speed dial max digits (25)	INVITE sent with all 25 digits	Pass	
9.2.370	Configure and use a speed dial assigned to a 25 digit number, with external access number prefix (9).	INVITE sent with all digits	Pass	25 including prefix
9.2.371	Speed dials min digits (4 digit ext.):	INVITE sent with all digits	Pass	
9.2.372	Configure and use a speed dial to a 4 digit number	INVITE sent with all digits	Pass	
9.2.373	Speed dial no entry	Call does not originate	N/A	
9.2.374	Max Speed-dial entries. Use first and last entries, and one other	All speed dial entries populated and usable	Pass	
9.2.375	Exceed Max Speed Dial entries.	Entry n+1 rejected, all other entries remain intact and usable	N/A	
9.2.376	Re-assign existing speed dial to new number (25 digit to 4 digit)	Number successfully changed and usable	Pass	
9.2.377	Delete Speed Dial Entry.	Number no longer usable. Remaining entries unchanged.	Pass	
9.2.378	Reset phone	Speed dial configuration persists through reset	Pass	
9.2.379	Power cycle phone	Speed dial configuration persists through power cycle	Pass	

Hotline				
9.2.380	Hotline to DUT Steps: Configure CSP to be a Hotline Configure CSP to dial the DUT number Pick up the CSP, it should auto dial out DUT number	Verify the CSP will auto dial out Verify the call is not completed as DUT is not a Hotline phone.	**N/A**	
9.2.381	DUT to hotline Steps: Configure DUT to be a hotline device Configure DUT to dial out to CSIPP Pick up the DUT, it should not dial out to CSIPP	Verify DUT does not dial out to CSIPP Verify the call is not completed.	**N/A**	
Park/Pickup and Monitor **It only works when the park is placed by** 8961, 9951, or 9971 **(SIP phones only)**				
9.2.382	Park Monitor Steps: PSTN calls a 9951 Phone 9951 parks the call DUT1 picks up the call	Verify PSTN call is connected to 9951 phone Verify the call is parked Verify the Park Bubble is displayed on the 9951 phone. Verify DUT picks up the parked call Verify the Park bubble on the 9951 phone is cleared.	Pass	
9.2.383	Park and Pickup Steps: DUT1 calls DUT2 DUT2 parks the call 9951 picks up the call	Verify the call is parked Verify 9951 can pick up the call. Verify all calls are cleared.	Pass	

	DUT1 releases call			
Native Call Queuing – UCM 9.0 Feature				
9.2.384	DUT to hunt pilot, call queued and routed to alternate CSP2/CSIPP/DUT2 when maximum wait time condition is met	DUT hears ring back. After maximum wait time, queued call, calls lands on CSP2/CSIPP/DUT2. 2-way audio between DUT and CSP2/CSIPP/DUT2 MOH/TOH or IVR heard as long as call is in queue Verify queue status on CSP2/CSIPP by logging into hunt group	Pass	DUT is not configured with Hlog key
9.2.385	DUT to hunt pilot, call queued and routed to alternate CSP2/CSIPP/DUT2 when maximum callers in queue condition is met	DUT hears ring back. After maximum wait time, queued call, calls lands on CSP2/CSIPP/DUT2. 2-way audio between DUT and CSP2/CSIPP/DUT2 MOH/TOH or IVR heard as long as call is in queue Verify queue status on CSP2/CSIPP by logging into hunt group	Pass	DUT is not configured with Hlog key
9.2.386	DUT to hunt pilot, call queued and routed to alternate CSP2/CSIPP/DUT2 when no members available condition is met	DUT hears ring back. After maximum wait time, queued call, calls lands on CSP2/CSIPP/DUT2. 2-way audio between DUT and CSP2/CSIPP/DUT2 MOH/TOH or IVR heard as long as call is in queue Verify queue status on CSP2/CSIPP by	Pass	DUT is not configured with Hlog key

		logging into hunt group		
9.2.387	DUT to hunt pilot, call queued and routed to alternate DUT3 when maximum wait time condition is met	DUT hears ring back. After maximum wait time, queued call, calls lands on DUT3 2-way audio between DUT and DUT3 MOH/TOH or IVR heard as long as call is in queue Verify queue status on CSIPP2/CSP that are part of the hunt group by logging into hunt group	Pass	DUT is not configured with Hlog key
9.2.388	DUT to hunt pilot, call queued and routed to alternate DUT3 when maximum callers in queue condition is met	DUT hears ring back. After maximum wait time, queued call, calls lands on DUT3 2-way audio between DUT and DUT3 MOH/TOH or IVR heard as long as call is in queue Verify queue status on CSIPP2/CSP that are part of the hunt group by logging into hunt group	Pass	DUT is not configured with Hlog key.
9.2.389	DUT to hunt pilot, call queued and routed to alternate DUT3 when no members available condition is met	DUT hears ring back. After maximum wait time, queued call, calls lands on DUT3 2-way audio between DUT and DUT3 MOH/TOH or IVR heard as long as call is in queue Verify queue status on CSIPP2/CSP that are part of the hunt group by logging into hunt group	Pass	DUT is not configured with Hlog key.

9.2.390	PSTN to hunt pilot, call queued and routed to alternate CSIPP/CSP/DUT2 when maximum wait time condition is met	PSTN hears ring back. After maximum wait time, queued call, calls lands on CSIPP/CSP/DUT2 2-way audio between PSTN and CSIPP/CSP/DUT2 MOH/TOH or IVR heard as long as call is in queue Verify queue status on CSIPP2/CSP that are part of the hunt group by logging into hunt group	Pass	DUT is not configured with Hlog key.
9.2.391	PSTN to hunt pilot, call queued and routed to alternate CSIPP/CSP/DUT2 when maximum callers in queue condition is met	PSTN hears ring back. After maximum wait time, queued call, calls lands on CSIPP/CSP/DUT2 2-way audio between PSTN and DUT3 MOH/TOH or IVR heard as long as call is in queue Verify queue status on CSIPP2/CSP that are part of the hunt group by logging into hunt group	Pass	DUT is not configured with Hlog key.
9.2.392	PSTN to hunt pilot, call queued and routed to alternate CSIPP/CSP/DUT2 when no members available condition is met	PSTN hears ring back. After maximum wait time, queued call, calls lands on CSIPP/CSP/DUT2 2-way audio between PSTN and DUT3 MOH/TOH or IVR heard as long as call is in queue Verify queue status on CSIPP2/CSP that are part of the hunt	Pass	DUT is not configured with Hlog key.

		group by logging into hunt group		

Note: In all scenarios below involving DUT and CSIPP, the dialing will be SIP URI Dialing. If the third party endpoint does NOT support URI dialing, then DN will be used.

Consultative Transfer & TLS/SRTP				
6.3.316	CSP calls DECT1, DECT1 calls DECT2, DECT1 XFERs to CSP	CSP receives MOH, 2-way audio IP-DECT to IP-DECT1, IP-DECT2 properly released, 2-way audio IP-DECT2 to CSP	Pass	
6.3.317	CSIPP calls DECT1, DECT1 calls DECT2, DECT1 XFERs to CSIPP	CSIP receives MOH, 2-way audio IP-DECT1 IP-DECT2, IP-DECT properly released, 2-way audio IP-DECT2 CSIPP extension	Pass	
6.3.318	PSTN calls DECT1, DECT1 calls DECT2, DECT1 XFERs to PSTN	PSTN receives MOH, 2-way audio IP-DECT to IP-DECT2 , IP-DECT properly released, 2-way audio IP-DECT2 to PSTN	Pass	

9.3 System Control and Verification

These tests are executed to determine the impact on calls, the Communications Manager and the 3rd party application when combinations of the aforementioned fail by power failure or network connectivity problems. Testing robustness of the application through hardware and software fault insertion i.e. failover/failback.

Test Case	Description	Expected Result	Pass/ Fail	Comments
Registration and Digest Authentication (Basic)				
9.3.1	DUT Authenticated on Registration (name and password) (Positive)	DUT registers successfully	Pass	
9.3.2	DUT Authenticated on Registration (name and password) (Negative)	DUT registration rejected, retries	Pass	
9.3.3	DUT Authenticated on Origination	DUT resends INVITE with Authorization header, successfully originates call	Pass	
9.3.4	DUT Re-Registers before Registration Time Expires	Re-registration successful, DUT can originate calls	Pass	
9.3.5	Restart DUT phone remotely	DUT restarts and registers successfully	N/A	
9.3.6	DUT Multiline registration	All lines register successfully and can originate calls	N/A	Applicable to advanced 3rd party SIP devices only
9.3.7	Loses network connection then re-connected	DUT can originate calls after registration	Pass	

9.4 Negative Tests

These tests are executed to determine the impact on calls, the Call Manager and the 3rd party application when combinations of the aforementioned fail by power failure or network connectivity problems. Testing robustness of the application through hardware and software fault insertion i.e. failover/failback.

Test Case	Description	Expected Result	Pass/ Fail	Comments
9.4.1	Publisher outage: place station-to-station call, isolate Publisher (disconnect network cable), place one more station-to-station call, reconnect Publisher network cable, and place another station-to-station call.	Confirm all calls are completed.	**Pass**	
9.4.2	Subscriber Outage: place station-to-station call, isolate Subscriber (disconnect network cable), place one more station-to-station call, reconnect Subscriber network cable, place another station-to-station call	Confirm all calls are completed.	**Fail**	Failed to fallback: when the Subscriber(Primary to DUT) is down the active call stays up and after the call is disconnected, the DUT gets registered to secondary node but when primary comes back online the active call with seconday is getting disconnected and there is a down time before the calls are successful with the primary node.
9.4.3	IP-DECT Registers with SRST server when the CCM is unavailable		Pass	
9.4.4	IP-DECT re-Registered when CCM is back on service		Pass	
9.4.5	Basic call, IP-DECT uses SRST server when CCM is unavailable		Pass	
9.4.6	Call to PSTN using SRST server when CCM is unavailable		Pass	
9.4.7	Call from PSTN, call is terminated by SRST when CCM is unavailable		Pass	

9.5 Informational Tests

These tests are executed to verify specific information about the third-party product to Cisco. This is in relation to the IVT Questionnaire supplied by the vendor. *Test cases in this section will be selected or modified to reflect attributes of the device under test. Standard examples are:*

Test Case	Description	Expected Result	Pass/ Fail	Comments
Voice Codec Support				
9.5.1	G.711 µ-law Configure DUT to support only G711ulaw codec. Place call from/to DUT from DUT2/CSIP/CSIPP	2-way audio successfully established between DUT and DUT2/CSP/CSIPP with G711ulaw codec	Pass	
9.5.2	G.711 A-law Configure DUT to support only G711Alaw codec. Place call from/to DUT from DUT2/CSIP/CSIPP	2-way audio successfully established between DUT and DUT2/CSP/CSIPP with G711alaw codec	Pass	
9.5.3	G.723 Configure DUT to support only G723 codec. Place call from/to DUT from DUT2/CSIP/CSIPP	2-way audio successfully established between DUT and DUT2/CSP/CSIPP with G723 codec	Pass	Works between Ascom phones. N/T for Cisco phones
9.5.4	G.729 Configure DUT to support only G729 codec. Place call from/to DUT from DUT2/CSIP/CSIPP	2-way audio successfully established between DUT and DUT2/CSP/CSIPP with G729 codec	Pass	
9.5.5	Packetization period Configure DUT to support different p-times -10, 20, 30. Place call from/to DUT from DUT2/CSIP/CSIPP with p-time fixed at 20	2-way audio successfully established between DUT and DUT2/CSP/CSIPP with p-time negotiated.	Pass	**p-time 10 N/S**

Test Case	Description	Expected Result	Pass/ Fail	Comments
9.5.6	Mid-call codec renegotiation		N/S	Applicable to advanced devices only
General Phone Functions				
9.5.7	Phone display (missed calls, called numbers, received calls)		Pass	
9.5.8	All visible buttons and soft keys function as labeled		Pass	
General Dial Services				
9.5.9	Redial or dial from Call History		Pass	
9.5.10	Last Call Return		Pass	**Optional test case**

9.6 Performance/Load Tests

These tests are executed to determine the impact of the 3rd party software on the Cisco Call Manager's ability to process calls. Testing will also determine the outer limits of the application's ability to properly function under stress and perform characterizing measurements on the Call Manager.

Identify any modifications to Trace/Debug levels.

Background Load characteristics for all tests:

- 2500 phones per subscriber
- XX Callers and XX receivers
- 100 % Internal (SCCP)
- All configurations to support each load test must be completed and verified prior to the start of this test.
- Document Registered Users, BHCA and BHCC per call type, and dial tone delay metrics for each test in the Load Test Results table following this test section.

Table 6 – Performance Counters to Be Used

Performance Parameter Measured
Cisco Call Manager (Publisher)
Memory
Available Mbytes
Paging File
% Usage
Processor
% Processor Time
Logical Disk
Free Megabytes
SQL Server: Databases
Active Transactions

Performance Parameter Measured
Data File(s) Size (KB)
Log Cache Hit Ratio
Log File(s) Size (KB)
Flush Log Wait Time
Log Flush/sec
Log Growths
Percent Log Used
SQL Server: Cache Manager
Cache Hit Ratio
SQL Server: Memory Manager
Connection memory

Table 7 – Cisco Performance Pass/Fail Criteria

Description	**Cisco Criterion**
Call Completion Failures	Less than .001%
Call Drops	Less than .001%
Call Manager CPU Utilization	Equal to or less than 68%
Publisher CPU Utilization	Equal to or less than 68%
Memory	No increase trends
SQL Server	No persistent decrease in cache hit ratio when compared to baseline
SQL Server	No change in transaction log growth rate over baseline
Dial Tone Delay	250ms
Disk Usage	No significant increase trends
Network Outage Recovery	Recover in 10 minutes or less

Note: Any CPU pegs over 80% sustaining for 5 seconds or more and any sustained memory increases should be noted.

Test Case	Description	Expected Result	Pass/ Fail	Comments
6.5.1	Long duration Call Make a call and leave the call up for an hour.	Verify the call does not drop and the session timers are working correctly	Pass	
6.5.2	Continuous calls If the device supports auto answer, place the device in auto answer and make continues calls with a call hold time of 2 min for two hours.	Verify the all the calls are correctly handled.	Pass	
6.5.3	Continuous calls If the device does not support auto answer , place continuous ring and release the call after each ring for 2 hours Once the test is complete, place a call to the phone and confirm two way audio	Verify all the rings/calls are correctly handled. After the continuous calls DUT should continue to accept calls and display normal / expected behavior	N/A	DUT supports auto answer, so executed the above test case.

1 ASCOM Three-party services, In-Call/Out-of-Call Menu

1.1 Portable menu – Conferencing Main, add and remove participants

Test Case	Description	Expected Result	Pass/ Fail	Comments
7.1	To verify that it's possible to start a conference call and that participants can be added or leave conference call		Pass	Not possible to remove participants. Not implemented

Purpose

To verify that it's possible to start a conference call and that participants can be added or leave conference call.

9.6.1

Requirements

IPDECTR6-01.00020-00 Conferencing
IPDECTR6-01.00021-00 Conference Tone
IPDECTR6-01.00022-00 Remove last conference participant

Preconditions and configuration

- General pre-conditions.
- Ascom IP-DECT Devices are required (i.e. not Third-Party SIP Device)
- Requires Cisco license (Cisco version 7.1.5 or later)
- Conference call enabled and assigned a code in IP-DECT supplementary services
- 'Maximum Ad Hoc Conference' should be set to 5 in Cisco PBX
- Use the in-call menu item in the portable to hold and resume calls, to start a conference and add or remove participants from conference call.
- 5 portables subscribed – PP1, PP2, PP3, PP4, PP5

Test Instruction

Step	Action	Expected result
1	PP1 call PP2. PP2 must answer.	Call is connected
2	PP1 put PP2 on hold and call PP3. PP3 must answer.	Call is connected
3	Let PP1 start a CUCM hosted conference.	• PP1, PP2 and PP3 are added to a conference call. • Verify that a conference add warning tone is played to the participants. • Verify that that conversation can be made. • Voice quality must be acceptable.

		• Verify that PP1, PP2 and PP3 displays correct called party information.
4	Let PP1 put conference call on hold and call PP4. PP4 must answer.	Call is connected
5	Let PP1 add PP4 to conference call	• PP4 is added to conference call. • Verify that a conference add warning tone is played to the participants. • Verify that conversation can be made in all directions. • Voice quality must be acceptable. • Verify that PP1, PP2, PP3 and PP4 displays correct called party information.
6	Let PP1 put conference call on hold and call PP5. PP5 must answer.	Call is connected
7	Let PP1 add PP5 to conference call	• PP5 is added to conference call. • Verify that a conference add warning tone is played to the participants. • Verify that that conversation can be made in all directions. • Voice quality must be acceptable. • Verify that PP1, PP2, PP3, PP4 and PP5 displays correct called party information.
8	Let PP1 remove the last added conference participant (PP5)	• Verify that a conference remove warning tone is played to the participants. • PP5 is removed from conference call. • Call is disconnected on PP5 side. • Verify that that conversation can be made in all directions.
9	Let PP1 remove the last added conference participant (PP4)	• Verify that a conference remove warning tone is played to the participants. • PP4 is removed from conference call. • Call is disconnected on PP4 side. • Verify that that conversation can be made in all directions.
10	Let PP1 once again put conference call on hold, call PP4 and add it to conference call.	As stated
11	Let PP4 hang up	• Verify that a conference remove warning tone is played to the participants. • PP4 is removed from conference call. • Call is disconnected on PP4 side. • Verify that that conversation can be made in all directions.
12	Let PP3 hang up	• Verify that a conference remove warning tone is played to the participants.

		• Verify that call between PP1 and PP2 is still connected but conference call is ended.
13	In Cisco PBX set the 'Maximum Ad Hoc Conference' to 3 and save	• As stated
14	Let PP1 set up a conference call with PP2 and PP3	• Conference active
15	Try to add a fourth member to the conference	• Should not be accepted, since limit is set to 3 in Cisco
16	Repeat test with PP1 as fixed phone	• Outcome as previous steps

Note:

- By using Ascom phone: Steps 8 and 9 are not executed. Cannot remove participants from the conference list.
- Add warning tone was not heard, when having warning tone set to true in service parameters.
- By using fixed phone: all the steps are executed. Warning tone is also heard.

1.2 Portable menu – Conferencing, advanced Ad Hoc Conference

Test Case	Description	Expected Result	Pass/ Fail	Comments
7.2	To verify that if the organizer leaves the conference, the other members should be able to add more members to conference call.		Pass	

9.6.2

Purpose

To verify that if the organizer leaves the conference, the other members should be able to add more members to conference call.

Requirements

IPDECTR6-01.00020-00 Conferencing
IPDECTR6-01.00021-00 Conference Tone

Preconditions and configuration

General pre-conditions.
Ascom IP-DECT Devices are required (Not Third-Party SIP Device)
Requires Cisco license (Cisco version 7.1.5 or later)

Conference call enabled and assigned a code in IP-DECT supplementary services
'Advanced Ad Hoc Conference Enabled' should be set to True in Cisco PBX
Use the in-call menu item in the portable to hold and resume calls, to start a conference and add or remove participants from conference call.
4 portables subscribed – PP1, PP2, PP3, PP4

Test Instruction

Step	Action	Expected result
1	PP1 call PP2. PP2 must answer.	Call is connected
2	Let PP1 put PP2 on hold and call PP3. PP3 must answer.	Call is connected
3	Let PP1 start a CUCM hosted conference.	• PP1, PP2 and PP3 are added to a conference call. • Verify that a conference add warning tone is played to the participants. • Verify that that conversation can be made. • Voice quality must be acceptable. • Verify that PP1 and PP2 displays correct called party information.
4	Let PP1 put conference call on hold and call PP4. PP4 must answer.	Call is connected
5	Let PP1 add PP4 to conference call	• PP4 is added to conference call. • Verify that a conference add warning tone is played to the participants. • Verify that that conversation can be made in all directions. • Voice quality must be acceptable. • Verify that PP1, PP2, PP3 and PP4 displays correct called party information.
6	Let PP1 hang up.	• Verify that a conference remove warning tone is played to the participants. • PP1 is removed from conference call. • Call is disconnected on PP1 side. • Verify that that conversation can be made in all directions.
7	Let PP2 put conference on hold and call PP1. PP1 must answer.	Call is connected
8	Let PP2 add PP1 to conference call	• PP1 is added to conference call. • Verify that a conference add warning tone is played to the participants. • Verify that that conversation can be made in all directions. • Voice quality must be acceptable. • Verify that PP1, PP2, PP3 and PP4 displays correct called party information.
9	Let portables end conference call by all hanging up.	Conference call is ended.

10	Repeat test with PP1 as fixed phone	Outcome as previous steps

1.3 Portable menu – Meet me Conferencing

Test Case	Description	Expected Result	Pass/ Fail	Comments
7.3	To verify that it's possible to start a conference call and that participants can call the conference number and be added automatically.		Pass	

Purpose
To verify that it's possible to start a conference call and that participants can call the conference number and be added automatically.
Requirements
IPDECTR6-01.00020-00 Conferencing
IPDECTR6-01.00021-00 Conference Tone
Preconditions and configuration
General pre-conditions.
Ascom IP-DECT Devices are required (i.e. not Third-Party SIP Device)
Requires Cisco license (Cisco version 7.1.5 or later)
Conference call enabled and assigned a code in IP-DECT supplementary services
'Maximum Ad Hoc Conference' should be set to 5 in Cisco PBX
The conferencing numbers that can be used can be found in the in 'Call routing'/'Meet me number plan/pattern' in Cisco PBX
Use the out-of-call menu item in the portable to activate a conference call
5 portables subscribed – PP1, PP2, PP3, PP4, PP5

Test Instruction

Step	Action	Expected result
1	Let PP1 start a conferencing call from the out-of-call menu.	Conference call started

2	Let PP2 call the conference number	• PP1 and PP2 are added to a conference call. • Verify that a conference add warning tone is played to the participants. • Verify that that conversation can be made. • Voice quality must be acceptable. • Verify that PP1, PP2 and PP3 displays correct called party information.
3	Let PP3 call the conference number	• PP1, PP2 and PP3 are participating in the conference call. • Verify that a conference add warning tone is played to the participants. • Verify that that conversation can be made. • Voice quality must be acceptable. • Verify that PP1, PP2 and PP3 displays correct called party information.
4	Hang up on PP1	• Call is disconnected • Verify that conference call is still active for PP2 and PP3
5	Let PP1 call the conference number again	• PP1, PP2 and PP3 are participating in the conference call. • Verify that a conference add warning tone is played to the participants. • Verify that that conversation can be made. • Voice quality must be acceptable. • Verify that PP1, PP2 and PP3 displays correct called party information.
6	Hang up on PP1 and PP2	• Calls are disconnected • Verify that conference call is still active for PP3
7	Hang up on PP3	• Call is disconnected • Conference is ended
8	Repeat test with PP1 as fixed phone	• Outcome as previous steps

1.4 Portable menu – CCBS

Test Case	Description	Expected Result	Pass/ Fail	Comments
7.4	To verify that portable receives a call back when initiating the call back functionality in the in-call menu		Pass	

Purpose

To verify that portable receives a call back when initiating the call back functionality in the in-call menu

Requirements

IPDECTR6-01.00030-00 CCBS
IPDECTR6-01.00032-00 Call Back Confirmation
IPDECTR6-01.00033-00 Execution of Call Back

Preconditions and configuration

- General pre-conditions.
- If Ascom IP-DECT Devices are required during test, Cisco Licenses are needed (Cisco version 7.1.5 or later)
- If Third-Party SIP Device are required during test, Cisco licenses are not needed
- CCBS enabled in IP-DECT supplementary services
- Ensure that CW is inactive
- Use the in-call/out-of-call menu item to initiate call back
- 2 portables subscribed – PP1, PP2

Test Instruction

Step	Action	Expected result
1	Let PP1 call PP2 when PP2 is busy in another call	As stated
2	Let PP1 initiate the call back functionality	Verify that PP1 receives a call back confirmation message (and alert)
3	Let PP2 end call and become idle	• Verify that PP1 starts to alert and that a call back tone is played. • Verify that correct calling party information is displayed in PP1
4	Let PP1 answer the call	• Verify that PP2 receives a call back and starts to alert. • Verify that correct calling party information is displayed in PP2
5	Let PP2 answer call	• Call is connected and conversation can be made. • Voice quality must be acceptable. • Displays of portables show correct call party information.

Step	Action	Expected result
6	Repeat step 1-3 but this time decline the incoming call in step 4.	• PP1 becomes idle • Verify that PP2 does not receive a call back from PP1

Step	Action	Expected result

7	Repeat step 1-3 but this time do not answer the incoming call in step 4.	• PP1 becomes idle after some time • Verify that PP2 does not receive a call back from PP1
8	Repeat test with PP2 as fixed phone	Outcome as previous steps

1.5 Portable menu – CCNR

Test Case	Description	Expected Result	Pass/ Fail	Comments
7.5	To verify that portable receives a call back when initiating the call back functionality in the in-call menu		Pass	

Purpose

To verify that portable receives a call back when initiating the call back functionality in the in-call menu

Requirements

IPDECTR6-01.00031-00 CCNR
IPDECTR6-01.00032-00 Call Back Confirmation
IPDECTR6-01.00033-00 Execution of Call Back

Preconditions and configuration

- General pre-conditions.
- If Ascom IP-DECT Devices are required during test, Cisco Licenses are needed (Cisco version 7.1.5 or later)
- If Third-Party SIP Device are required during test, Cisco licenses are not needed
- CCNR enabled in IP-DECT supplementary services
- Ensure that CW is inactive
- Use the in-call/out-of-call menu item to initiate call back
- 2 portables subscribed – PP1, PP2

Test Instruction

Step	Action	Expected result
1	Let PP1 call PP2. Do not answer on PP2.	As stated

2	Let PP1 initiate the call back functionality	Verify that PP1 receives a call back confirmation message (and alert)
3	Let PP2 call another portable and shortly after hang up to become idle (will trigger call to PP1)	• Verify that PP1 starts to alert and that a call back tone is played. • Verify that correct calling party information is displayed in PP1
4	Let PP1 answer the call	• Verify that PP2 receives a call back and starts to alert. • Verify that correct calling party information is displayed in PP2
5	Let PP2 answer call	• Call is connected and conversation can be made. • Voice quality must be acceptable. • Displays of portables show correct call party information.

Step	Action	Expected result
6	Repeat step 1-3 but this time decline the incoming call in step 4.	• PP1 becomes idle • Verify that PP2 does not receive a call back from PP1

Step	Action	Expected result
7	Repeat step 1-3 but this time do not answer the incoming call in step 4.	• PP1 becomes idle after some time • Verify that PP2 does not receive a call back from PP1
8	Repeat test with PP2 as fixed phone	Outcome as previous steps

1.6 Portable menu – Repeated Call Back

Test Case	Description	Expected Result	Pass/ Fail	Comments
7.6	To verify that portable receives a call back when initiating the call back functionality in the in-call menu		Pass	

Purpose

To verify that portable receives a call back when initiating the call back functionality in the in-call menu

Requirements

IPDECTR6-01.00030-00 CCBS

IPDECTR6-01.00032-00 Call Back Confirmation
IPDECTR6-01.00033-00 Execution of Call Back

Preconditions and configuration

- General pre-conditions.
- If Ascom IP-DECT Devices are required during test, Cisco Licenses are needed (Cisco version 7.1.5 or later)
- If Third-Party SIP Device are required during test, Cisco licenses are not needed
- CCBS enabled in IP-DECT supplementary services
- Ensure that CW is inactive
- Use the in-call/out-of-call menu item to initiate call back
- 2 portables subscribed – PP1, PP2

Test Instruction

Step	Action	Expected result
1	Let PP1 call PP2 when PP2 is busy in another call	As stated
2	Let PP1 initiate the call back functionality	Verify that PP1 receives a call back confirmation message (and alert)
3	Let PP2 end call and become idle	• Verify that PP1 starts to alert and that a call back tone is played. • Verify that correct calling party information is displayed in PP1
4	Let PP2 be busy in another call once again	As stated
5	Let PP1 answer the call	• Verify that PP2 does not receive a call back and that a busy tone is heard in PP1 • Verify that correct calling party information is displayed in PP1
6	Let PP1 initiate the call back functionality	Verify that PP1 receives a call back confirmation message (and alert)
7	Let PP2 end call and become idle	• Verify that PP1 starts to alert and that a call back tone is played. • Verify that correct calling party information is displayed in PP1
8	Let PP1 answer the call	• Verify that PP2 receives a call back and starts to alert. • Verify that correct calling party information is displayed in PP2
9	Let PP2 answer call	• Call is connected and conversation can be made. • Voice quality must be acceptable. • Displays of portables show correct call party information.
10	Let portables end call	Call is ended
11	Repeat test with PP2 as fixed phone	Outcome as previous steps

1.7 Portable menu – New Call Back cancel old

Test Case	Description	Expected Result	Pass/ Fail	Comments
7.7			Pass	

Purpose

To verify that portable receives a call back when initiating the call back functionality in the in-call menu. Furthermore, if portable initiate call back and then initiate it again against a new handset, the call back against the first handset is cancelled.

Requirements

IPDECTR6-01.00030-00 CCBS
IPDECTR6-01.00032-00 Call Back Confirmation
IPDECTR6-01.00033-00 Execution of Call Back

Preconditions and configuration

- General pre-conditions.
- If Ascom IP-DECT Devices are required during test, Cisco Licenses are needed (Cisco version 7.1.5 or later)
- If Third-Party SIP Device are required during test, Cisco licenses are not needed
- CCBS enabled in IP-DECT supplementary services
- Ensure that CW is inactive
- Use the in-call/out-of-call menu item to initiate call back
- 3 portables subscribed – PP1, PP2, PP3
-

Test Instruction

Step	Action	Expected result
1	Let PP1 call PP2 when PP2 is busy in another call	As stated
2	Let PP1 initiate the call back functionality	Verify that PP1 receives a call back confirmation message (and alert)
3	Let PP1 call PP3 when PP3 is busy in another call	As stated
4	Let PP1 initiate the call back functionality	Verify that PP1 receives a call back confirmation message (and alert)
5	Let PP2 end call and become idle	Verify that PP1 does not start to alert since call back is cancelled
6	Let PP3 end call and become idle	• Verify that PP1 starts to alert and that a call back tone is played. • Verify that correct calling party information is displayed in PP1

7	Let PP1 answer the call	• Verify that PP3 receives a call back and starts to alert. • Verify that correct calling party information is displayed in PP1
8	Let PP3 answer the call	• Call is connected and conversation can be made. • Voice quality must be acceptable. • Displays of portables show correct call party information.
9	Let portables end call	Call is ended
10	Repeat test with PP2 as fixed phone	Outcome as previous steps

1.8 Portable menu – Cancel Call Back

Test Case	Description	Expected Result	Pass/ Fail	Comments
7.8	To verify is possible to cancel an activated Call Back		Pass	

Purpose

To verify is possible to cancel an activated Call Back

Preconditions and configuration

- General pre-conditions.
- If Ascom IP-DECT Devices are required during test, Cisco Licenses are needed (Cisco version 7.1.5 or later)
- If Third-Party SIP Device are required during test, Cisco licenses are not needed
- Call Back enabled in IP-DECT supplementary services
- Ensure that CW is inactive
- Use the in-call/out-of-call menu item to initiate call back
- 2 portables subscribed – PP1, PP2

Test Instruction

Step	Action	Expected result
1.	Let PP1 call PP2 when PP2 is busy in another call	As stated
2.	Let PP1 initiate the call back functionality	Verify that PP1 receives a call back confirmation message (and alert)
3.	Let PP1 deactivate the Call Back by stating the code found in suppl. Serv. in IP-DECT	
4.	Let PP2 end call and become idle	• Verify that PP1 does not start to alert since call back is cancelled

5.	Repeat test with PP2 as fixed phone	Outcome as previous steps

1.9 Portable menu – Abbreviated Dialing

Test Case	Description	Expected Result	Pass/ Fail	Comments
7.9	To verify that it's possible to enter an abbreviated number and initiate a call to the number that matches the abbreviated number		Pass	

Purpose

To verify that it's possible to enter an abbreviated number and initiate a call to the number that matches the abbreviated number

Requirements

IPDECTR6-01.00040-00 Abbreviated Dialing
IPDECTR6-01.00041-00 Abbreviated Number Configuration

Preconditions and configuration

- General pre-conditions.
- Ascom IP-DECT Devices are required during test
- Cisco Licenses are needed (Cisco version 7.1.5 or later)
- Abbreviated Dialing enabled in IP-DECT supplementary services
- Abbreviated number configured in Cisco PBX
- Use the in-call/out-of-call menu item to select the abbreviated number
- 2 portables subscribed – PP1, PP2

Test Instruction

Step	Action	Expected result
1	Let PP1 select out-of-call menu and select the abbreviated number belonging to PP2	PP2 starts alerting.
2	Let PP2 answer call	• Call is connected and conversation can be made. • Voice quality must be acceptable. • Displays of portables show correct call party information.

3	End call	Call ended
4	Repeat test with PP2 as fixed phone	Outcome as previous steps

1.10 Portable menu – Call Park

Test Case	Description	Expected Result	Pass/ Fail	Comments
7.10	To verify that it's possible to park an active call		Pass	

Purpose

To verify that it's possible to park an active call

Requirements

IPDECTR6-01.00060-00 Call Park
IPDECTR6-01.00061-00 Call Park Confirmation
IPDECTR6-01.00063-00 Call Park Retrieve

Preconditions and configuration

- General pre-conditions.
- Ascom IP-DECT Devices are required during test
- Cisco Licenses are needed (Cisco version 7.1.5 or later)
- Call Park enabled in IP-DECT supplementary services
- Call Park configured in Cisco PBX
- Use the in-call/out-of-call menu item to park a call
- 3 portables subscribed – PP1, PP2 and PP3

Test Instruction

Step	Action	Expected result
1	Let PP1 call PP2. PP2 must answer.	Call connected
2	Let PP1 park the active call with PP2	• Call is disconnected. • PP1 receives a popup display displaying information with the parking lot number usage. • PP2 hears music on hold and the park number is populated in the display
3	Let PP3 retrieve the parked call	• Call is connected and conversation can be made. • Voice quality must be acceptable. • Displays of portables show correct call party information.

4	Let PP3 park the active call with PP2	• Call is disconnected. • PP3 receives a popup display displaying information with the parking lot number usage.
5	Let PP1 retrieve the parked call	• Call is connected and conversation can be made. • Voice quality must be acceptable. • Displays of portables show correct call party information.
6	End call	Call ended
7	Let PP1 try to retrieve the call parked in step 4	Call cannot be connected since there is no call parked at that specific parking lot
8.	Repeat test with PP2 as fixed phone	Outcome as previous steps

1.11 Portable menu – Call Park during Call on Hold

Test Case	Description	Expected Result	Pass/ Fail	Comments
7.11	To verify that it's possible to park an active call while another call is put on hold		Pass	

Purpose

To verify that it's possible to park an active call while another call is put on hold

Requirements

IPDECTR6-01.00060-00 Call Park
IPDECTR6-01.00061-00 Call Park Confirmation
IPDECTR6-01.00063-00 Call Park RetrieveIPDECTR6-01.00010-00 Hold and resume
IPDECTR6-01.00011-00 Music on hold

Preconditions and configuration

- General pre-conditions.
- Ascom IP-DECT Devices are required during test
- Cisco Licenses are needed (Cisco version 7.1.5 or later)
- Call Park enabled in IP-DECT supplementary services
- Call Park configured in Cisco PBX
- Use the in-call/out-of-call menu item to park a call
- 4 portables subscribed – PP1, PP2, PP3 and PP4

Test Instruction

Step	Action	Expected result
1	Let PP1 call PP2. PP2 must answer.	Call connected
2	Let PP1 put call with PP2 on hold.	• PP2 is put on hold. • PP2 hears on hold music and the display indicates that it is on hold • Communication has been interrupted. Neither party can hear the other party.
3	Let PP1 call PP3. PP3 must answer.	Call connected
4	Let PP1 park the active call with PP3	• Call is disconnected. • PP1 receives a popup display displaying information with the parking lot number usage. • PP3 hears music on hold and the park number is populated in the display
5	Let PP1 retrieve the call on hold (PP2)	• Call is connected between PP1 and PP2 and conversation can be made. • Displays of portables show correct call party information.
6	Let PP4 retrieve the parked call	• Call is connected with PP3 and conversation can be made. • Voice quality must be acceptable. • Displays of portables show correct call party information.
7	End calls on all portables	Calls ended
8	Repeat test with PP3 as fixed phone	Outcome as previous steps

1.12 Portable menu – Directed Call PArk

Test Case	Description	Expected Result	Pass/ Fail	Comments
7.12	To verify that it’s possible to park a call at a specified parking lot. This is not possible to do with any call on hold		Pass	

Purpose

To verify that it’s possible to park a call at a specified parking lot. This is not possible to do with any call on hold.

Requirements

IPDECTR6-01.00062-00 Directed Call Park

IPDECTR6-01.00063-00 Call Park Retrieve

Preconditions and configuration

- General pre-conditions.
- If Ascom IP-DECT Devices are required during test, Cisco Licenses are needed (Cisco version 7.1.5 or later)
- If Third-Party SIP Devices are required during test, Cisco licenses are not needed
- Call Park configured in Cisco PBX (Directed Call Park Configuration)
- Use the in-call/out-of-call menu item to park a call
- 3 portables subscribed – PP1, PP2 and PP3.

Test Instruction

Step	Action	Expected result
1	Let PP1 call PP2. PP2 must answer.	• Call connected
2	Let PP1 put the call with PP2 on hold by manually selecting the item in the in-call menu.	• PP2 hears music on hold
3	Let PP1 call a specific parking lot number (allowed sequence should be stated in the PBX).	• Call with specific parking lot is connected
4	Transfer call with PP2 to the specified parking lot number by pressing R4.	• Call with PP1 is disconnected and handset becomes idle • PP2 has been parked. Parking lot number stated in PP2 display
	Let PP3 retrieve the parked call by stating the same parking lot number as in step 3 (*Retrieval prefix + parking lot number).	• Call is connected with PP2 and conversation can be made. • Voice quality must be acceptable. • Displays of portables show correct call party information.
4	End calls on all portables	Calls ended
5	Repeat test with PP2 as fixed phone	Outcome as previous steps

1.13 Portable menu – Call Pickup Own Group (non-auto mode)

Test Case	Description	Expected Result	Pass/ Fail	Comments
7.13	To verify that it's possible to pickup		Pass	No message sent to ascom device.

Test Case	Description	Expected Result	Pass/ Fail	Comments
	a call belonging to the same Call Pickup Group.			

Purpose

To verify that it's possible to pickup a call belonging to the same Call Pickup Group.

Requirements

IPDECTR6-01.00050-00 Own Group Pickup
IPDECTR6-01.00051-00 Non-auto Mode Pickup

Preconditions and configuration

- General pre-conditions.
- Ascom IP-DECT Devices are required during test
- Cisco Licenses are needed (Cisco version 7.1.5 or later)
- Call Pickup enabled in IP-DECT supplementary services
- 3 portables subscribed – PP1, PP2, PP3
- Use the in-call/out-of-call menu item
- All portables added to same group in Cisco PBX

Test Instruction

Step	Action	Expected result
1	Let PP1 call PP2. PP2 must not answer.	• PP2 starts alerting. • After a while a popup display message with information about the call is presented on PP3 together with an audio indication.
2	Let PP3 pickup the call with PP1	• Call is redirected to PP3 and starts to alert with correct calling party information.
3	Answer the incoming call on PP1	• Call is connected with PP1 and conversation can be made. • Voice quality must be acceptable. • Displays of portables show correct call party information.
4	End call	Call is disconnected
5	Let PP1 call PP2. PP2 must not answer.	PP2 starts alerting.
	Let PP3 pickup the call with PP1 before the popup display message is presented	Call is redirected to PP3 and starts to alert with correct calling party information.
6	Answer the incoming call on PP1	• Call is connected with PP1 and conversation can be made. • Voice quality must be acceptable.

		• Displays of portables show correct call party information.
7	End call	Call is disconnected
8	Repeat test with PP1 as fixed phone	Outcome as previous steps

1.14 Portable menu – Call Pickup Group (non-auto mode)

Test Case	Description	Expected Result	Pass/ Fail	Comments
7.14	To verify that it's possible to pickup a call belonging to a different Call Pickup Group		Pass	

Purpose

To verify that it's possible to pickup a call belonging to a different Call Pickup Group

Requirements

IPDECTR6-01.00053-00 Group Pickup
IPDECTR6-01.00051-00 Non-auto Mode Pickup

Preconditions and configuration

- General pre-conditions.
- Ascom IP-DECT Devices are required during test
- Cisco Licenses are needed (Cisco version 7.1.5 or later)
- Call Pickup enabled in IP-DECT supplementary services
- Use the in-call/out-of-call menu item
- 3 portables subscribed – PP1, PP2, PP3
- PP1 and PP2 added to Group 1 in Cisco PBX
- PP3 added to Group 2 in Cisco PBX

Test Instruction

Step	Action	Expected result
1	Let PP1 call PP2. PP2 must not answer.	PP2 starts alerting.
2	Let PP3 pickup the call with PP1 (enter Group Number in out-of-call Menu)	Call is redirected to PP3 and starts to alert with correct calling party information.
3	Answer the incoming call on PP1	• Call is connected with PP1 and conversation can be made. • Voice quality must be acceptable. • Displays of portables show correct call party information.

4	End call	Call is disconnected
5	Repeat test with PP1 as fixed phone	Outcome as previous steps

1.15 Portable menu – Call Pick up Other Group (non-auto mode)

Test Case	Description	Expected Result	Pass/ Fail	Comments
7.15	To verify that it's possible to pickup a call belonging to a Call Pickup Group associated with your own group.		Pass	

Purpose

To verify that it's possible to pickup a call belonging to a Call Pickup Group associated with your own group.

Requirements

IPDECTR6-01.00054-00 Other Group Pickup
IPDECTR6-01.00051-00 Non-auto Mode Pickup

Preconditions and configuration

- General pre-conditions.
- Ascom IP-DECT Devices are required during test
- Cisco Licenses are needed (Cisco version 7.1.5 or later)
- Call Pickup enabled in IP-DECT supplementary services
- 3 portables subscribed – PP1, PP2, PP3
- Use the in-call/out-of-call menu item
- PP1 and PP2 added to Group 1 in Cisco PBX
- PP3 added to Group 2 in Cisco PBX
- Make sure Group 1 and Group 2 are associated groups in Cisco PBX

Test Instruction

Step	Action	Expected result
1	Let PP1 call PP2. PP2 must not answer.	PP2 starts alerting.
2	Let PP3 pickup the call with PP1 (enter Group Number in out-of-call Menu)	Call is redirected to PP3 and starts to alert with correct calling party information.

3	Answer the incoming call on PP1	• Call is connected with PP1 and conversation can be made. • Voice quality must be acceptable. • Displays of portables show correct call party information.
4	End call	Call is disconnected
5	Repeat test with PP3 as fixed phone	Outcome as previous steps

1.16 Portable menu – Directed Call Pickup (non-auto group)

Test Case	Description	Expected Result	Pass/ Fail	Comments
7.16	To verify that it's possible to pickup a call belonging to a Call Pickup Group associated with your own group. It should be possible to enter the number of the ringing phone and pickup the call from that number.		Pass	

Purpose

To verify that it's possible to pickup a call belonging to a Call Pickup Group associated with your own group. It should be possible to enter the number of the ringing phone and pickup the call from that number.

Requirements

IPDECTR6-01.00055-00 Directed Call Pickup
IPDECTR6-01.00051-00 Non-auto Mode Pickup

Preconditions and configuration

- General pre-conditions.
- Ascom IP-DECT Devices are required during test
- Cisco Licenses are needed (Cisco version 7.1.5 or later)
- Call Pickup enabled in IP-DECT supplementary services
- 3 portables subscribed – PP1, PP2, PP3
- Use the in-call/out-of-call menu item
- PP1 and PP2 added to Group 1 in Cisco PBX
- PP3 added to Group 2 in Cisco PBX
- Make sure Group 1 and Group 2 are associated groups in Cisco PBX

Test Instruction

Step	Action	Expected result
1	Let PP1 call PP2. PP2 must not answer.	PP2 starts alerting.
2	Let PP3 pickup the call with PP1 (enter number on the phone that is ringing in out-of-call Menu)	Call is redirected to PP3 and starts to alert with correct calling party information.
3	Answer the incoming call on PP1	• Call is connected with PP1 and conversation can be made. • Voice quality must be acceptable. • Displays of portables show correct call party information.
4	End call	Call is disconnected
5	Repeat test with PP3 as fixed phone	Outcome as previous steps

1.17 Portable menu – Auto Mode Pickup

Test Case	Description	Expected Result	Pass/ Fail	Comments
7.17	To verify that it's possible to pick up a call in auto mode, i.e. to connect directly with the incoming call from portable		Pass	

Purpose

To verify that it's possible to pick up a call in auto mode, i.e. to connect directly with the incoming call from portable

Requirements

IPDECTR6-01.00052-00 Auto Mode Pickup

Preconditions and configuration

- General pre-conditions.
- Ascom IP-DECT Devices are required during test
- Cisco Licenses are needed (Cisco version 7.1.5 or later)
- Auto mode has to be set in Cisco since non-auto mode is default
- Call Pickup enabled in IP-DECT supplementary services
- Use the in-call/out-of-call menu item
- 3 portables subscribed – PP1, PP2, PP3
- PP1, PP2 and PP3 added to Group 1 in Cisco PBX

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Test Instruction

Step	Action	Expected result
1.	Let PP1 call PP2. PP2 must not answer.	PP2 starts alerting.
2.	Let PP3 pickup the call with PP1 (Auto Mode)	• Call is connected with PP1 and conversation can be made. • Voice quality must be acceptable. • Displays of portables show correct call party information.
3.	End call	Call is disconnected
4.	Repeat test with PP1 as fixed phone	Outcome as previous steps

10 Test Area – Shared line

10.1 (#1474.1) Configure Shared line using Ascom IP-DECT Device

10.1.1 Purpose

To verify that it is possible to configure Shared line with Ascom IP-DECT Device.

10.1.2 Requirements

- TBD (Jira IPDECT-226 and -401)

10.1.3 Preconditions and configuration

- General pre-conditions
- Ascom IP-DECT Devices are required (i.e. not Third-Party SIP Device)
- Requires Cisco license (Cisco version 7.1.5 or later) with COP-file (9.0v1 version)
- 1 Fixed phone (FP1) should be configured in CUCM
- 2 portables subscribed – PP1 and PP2

10.1.4 Test Instruction

Step	Action	Expected result
1.	Open Phone configuration for FP1 in CUCM (Device/Phone).	Association Information shows Line 1 – FP1.
2.	Add a new DN (Association Information for FP1). Fill in PP1 number in Directory Number field and Save.	PP1 information will automatically be filled in. Line 2 – PP1 can be seen in Association Information under Unassigned Associated items.
3.	Click on the button Modify Button Items (Association Information for FP1).	-
4.	Move Line 2- PP1 to Associated Items and Save.	Line 1 – FP1 and Line 2 – PP1 are now associated.
5.	Let PP2 make a call to PP1.	Verify that PP1 and FP1 are ringing (alerting indicates internal call).
6.	Let PP1 answer the call.	Call is connected between PP1 and PP2. FP1 should immediately stop alert when PP1 answers the call.
7.	Let PP2 end the call.	Call is ended.
8.	Pass	

10.2 (#1475.1) Basic call/Answer - Shared line using Ascom IP-DECT Device

10.2.1 Purpose

To verify that the feature Shared line can handle Basic call – Answer correctly. When portable phone answers a call fixed phone should stop alert and vice versa. Only the portable that answers the call should have a call in call list.

10.2.2 Requirements

- TBD (Jira IPDECT-226 and -401)

10.2.3 Preconditions and configuration

- General pre-conditions
- Ascom IP-DECT Devices are required (i.e. not Third-Party SIP Device)
- Requires Cisco license (Cisco version 7.1.5 or later) with COP-file (9.0v1 version)
- 1 Fixed phone (FP1) in CUCM
- 2 portables subscribed – PP1 and PP2
- Shared Line is configured for FP1 and PP1 (one extension).

10.2.4 Test Instruction

Step	Action	Expected result
1.	Let PP2 call PP1.	Both PP1 and FP1 should start to alert. Verify that PP1 and FP1 display calling party information.
2.	Let PP1 answer the call.	Call is connected between PP1 and PP2. Verify that both portables display correct call party information. FP1 should immediately stop alert when PP1 answers the call.
3.	Let PP2 end the call.	Call is ended.
4.	Check call list on PP1 and FP1.	Only PP1 should have PP2’s number in call list.
5.	Pass	

Step	Action	Expected result
6.	Let PP2 call PP1.	Both PP1 and FP1 should start to alert. Verify that PP1 and FP1 display calling party information.
7.	Let FP1 answer the call.	Call is connected between FP1 and PP2. Verify that both portables display correct call party information. PP1 should immediately stop alert when FP1 answer the call.
8.	Let PP2 end the call.	Call is ended.
9.	Check call list on PP1 and FP1.	Only FP1 should have PP2’s number in call list.
10.	Pass	

10.3 (#1476.1) Basic call/Decline - Shared line using Ascom IP-DECT Device

10.3.1 Purpose

To verify that the feature Shared line can handle Basic call - Decline correctly. When portable phone declines a call fixed phone should still alert. Both portable phone and fixed phone should have a call in call list.

10.3.2 Requirements

- TBD (Jira IPDECT-226 and -401)

10.3.3 Preconditions and configuration

- General pre-conditions
- Ascom IP-DECT Devices are required (i.e. not Third-Party SIP Device)
- Requires Cisco license (Cisco version 7.1.5 or later) with COP-file (9.0v1 version)
- 1 Fixed phone (FP1) in CUCM
- 2 portables subscribed – PP1 and PP2
- Shared Line is configured for FP1 and PP1 (one extension).

10.3.4 Test Instruction

Step	Action	Expected result
1.	Let PP2 call PP1.	Both PP1 and FP1 should start to alert.
2.	Let PP1 decline the call.	PP1 is idle. FP1 still alerts.
3.	Let FP1 answer the call (it's not possible to decline a call from a FP).	Call is connected between FP1 and PP2.
4.	Let FP1 end the call.	Call is ended.
5.	Check call list on PP1 and FP1.	PP1 and FP1 should have PP2 in call list.
6.	Pass	

10.4 (#1477.1) Basic call/No answer - Shared line using Ascom IP-DECT Device

10.4.1 Purpose

To verify that the feature Shared line can handle Basic call – No answer correctly. A missed/ignored call should be in Missed call list of both portable phone and fixed phone.

10.4.2 Requirements

- TBD (Jira IPDECT-226 and -401)

10.4.3 Preconditions and configuration

- General pre-conditions
- Ascom IP-DECT Devices are required (i.e. not Third-Party SIP Device)
- Requires Cisco license (Cisco version 7.1.5 or later) with COP-file (9.0v1 version)
- 1 Fixed phone (FP1) in CUCM
- 2 portables subscribed – PP1 and PP2
- Shared Line is configured for FP1 and PP1 (one extension).

10.4.4 Test Instruction

Step	Action	Expected result
1.	Let PP2 call PP1.	Both PP1 and FP1 should start to alert.
2.	Do not answer incoming call on PP1 or FP1.	Call is automatically ended after 3 minutes. Verify that call is in Missed call list of both PP1 and FP1.
3.	Pass	

10.5 (#1478.1) Basic call/Busy - Shared line using Ascom IP-DECT Device

10.5.1 Purpose

To verify that the feature Shared line can handle Basic call – Busy correctly. When portable phone is busy fixed phone should start to alert.

10.5.2 Requirements

- TBD (Jira IPDECT-226 and -401)

10.5.3 Preconditions and configuration

- General pre-conditions
- Ascom IP-DECT Devices are required (i.e. not Third-Party SIP Device)
- Requires Cisco license (Cisco version 7.1.5 or later) with COP-file (9.0v1 version)
- 1 Fixed phone (FP1) in CUCM
- 3 portables subscribed – PP1, PP2 and PP3
- Shared Line is configured for FP1 and PP1 (one extension).

10.5.4 Test Instruction

Step	Action	Expected result
1.	Let PP1 call PP2.	Call is connected between PP1 and PP2.
2.	Let PP3 call PP1.	FP1 alerts. Verify that FP1 displays calling party information (PP3). PP1 is still in call with PP2.
3.	Let FP1 answer the call.	Call is connected between PP3 and FP1.

		Verify that both portables display correct call party information.
4.	End all calls.	Calls are ended.
5.	Check call list on PP1 and FP1.	Only FP1 should have PP3's number in call list.
6.	Pass	

10.6 (#1479.1) Basic call/Hold - Shared line using Ascom IP-DECT Device

10.6.1 Purpose

To verify that the feature Shared line can handle Basic call – Hold correctly. Fixed phone should be able to retrieve a call on-hold from portable phone.

10.6.2 Requirements

- TBD (Jira IPDECT-226 and -401)

10.6.3 Preconditions and configuration

- General pre-conditions
- Ascom IP-DECT Devices are required (i.e. not Third-Party SIP Device)
- Requires Cisco license (Cisco version 7.1.5 or later) with COP-file (9.0v1 version)
- 1 Fixed phone (FP1) in CUCM
- 3 portables subscribed – PP1, PP2 and PP3
- Shared Line is configured for FP1 and PP1 (one extension).

10.6.4 Test Instruction

Step	Action	Expected result
1.	Let PP1 call PP2.	Call is connected between PP1 and PP2.
2.	Let PP1 put call on hold.	PP2 is put on hold. PP2 hears on hold music and the display indicates that it is on hold. Communication has been interrupted. Neither party can hear the other party.
3.	Let PP1 call PP3.	PP3 alerts. Displays of portables show correct call party information.
4.	Let PP3 answer the call.	Speech is connected. Displays of portables show correct call party information.
5.	Let FP1 pick up call which is on hold (PP2).	Call is connected between PP2 and FP1. Verify that both portables display correct call party information: • PP2 shows PP1 information due to the shared extension even if it is FP1 that resumed the call. • FP1 shows PP2 information.
6.	Let PP1 and PP3 end the call.	Call is ended between PP1 and PP3. PP1 and PP3 are in idle.

		FP1 and PP2 are still connected and portables still displays correct call party information (see Expected result 5).
7.	Let FP1 and PP2 end the call.	Call is ended.
8.	Check call list on PP1, PP2, PP3 and FP1.	PP1 should have PP2 in call list. PP2 should have PP1 in call list. PP3 should have PP1 in call list. FP1 should not have any entries in call list.
9.	Pass- FP1 has PP3 in call list	