Test Results for Cisco Unified Communications System Release 8.5(1)

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Test Results for Cisco Unified Communications System Release 8.5(1) for Japanese - Add-on testing for Cisco Japan

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CHAPTER 1

Cisco Unified Communications System Test

Cisco Unified Communications System Test, an Integral part of the Enterprise Voice Solution Management, is a program that validates and tests, specified systems level solution for the various products and platforms in the Cisco Unified Communications System.

Cisco Unified Communications System Test, the systems integration layer, ensures that the Unified Communications components delivered across the various engineering teams when combined, improves the Unified Communications system software quality.This is achieved by testing the various components.

The requirements for Cisco Unified Communications System Test is derived based on the following:

- Popular customer scenarios
- Input from various Business Units, fields and Cisco Services.

The test bed architecture is build based on the Solution Reference Network Design (SRND), cross-section of product deployment models etc. The different types of testing carried out as part of Cisco Unified Communications System Test are:

- Interoperability/Compatibility
- Functionality
- Availability/Reliability/Stability
- Performance/Scalability/Capacity
- Usability, Serviceability
- Special focus area—CAP (Customer Assurance Program), Technical Assistance Center (TAC)
- Security

Cisco Unified Communications System Test for Japanese

Cisco Unified Communications System Test for Japanese, in turn is an add-on testing at the solution level, where the requirements gathered are specific to Japanese usage and market. The requirements are derived based on the following:

- Customer found defects in selected UC products
- High priority cases that are covered by the Cisco Unified Communications System Test team
- Inputs from SEs, TAC team of Cisco Japan

The test execution is carried out on selected UC products, which affect the Japanese segment and that are prioritized by SE of the Cisco Japan team. Japanese specific equivalents, such as, Japanese locale, ISDN Switch type being NTT, JPNP for Numbering Plan are implemented.

The objective of Cisco Unified Communications System Test for Japanese is to run a sub-set of system testing that is not covered by Cisco Unified Communications System Test and implement equivalents with Japanese environment such as Japanese OS, Localized application, select Cisco Compatible Products, and third party equipment.

In Cisco Unified Communications System Test for Japanese tests the following new features.

- Cisco Unified Communications Manager
- Cisco Unity Connection
- Cisco Unified Presence
- Cisco Unified Survivable Remote Site Telephony
- Cisco Unified Border Element
- Cisco Unified Communications Manager Express
- Cisco Unified IP Phone
- Cisco Unified Personal Communicator
- Cisco UC Integration™ for Microsoft Office Communicator
- Cisco Unified Contact Center Express
- Multi-Stage Upgrade
- Single-Stage Upgrade
- Regression Testing

Acronyms

Acronym	Description
AMWI	Audible Message Waiting Indicator
AAR	Automated Alternate Routing
ANAT	Alternate Network Address Translation
ACN	Alternate Contact Number
ACD	Automatic Call Distribution
BAT	Bulk Administrator tool
BLF	Busy Lamp Field
CAD	Cisco Agent Desktop
CAD BE	Cisco Agent Desktop Browser Edition
CAS	Channel Associated Signalling
CCD	Call Control Discovery
CDA	Cisco Desktop Administrator
CDR	Call Detail Record
CED	Caller Entered Digits
CFA	Call Forward All
CFB	Call Forward Busy
CFD	Customer Found Defect
CFNA	Call Forward No Answer
CIPC	Cisco Unified IP Communicator
CFNC	Call Forward No Coverage
CFUR	Call Forward Unregistered
CLI	Command Line Interface
CLID	Caller ID
CME	Cisco Unified Communications Manager Express
CSD	Cisco Supervisor Desktop
CSS	Calling Search Space
CSQ	Contact Service Queue
CTI	Computer Telephony Interface
CU	Cisco Unity
CUC	Cisco Unity Connection
CUP	Cisco Unified Presence
CUCM	Cisco Unified Communications Manager
CUPC	Cisco Unified Personal Communicator
CUPS	Cisco Unified Presence Server
DCR	Device and Credential Repository

Acronym	Description
DHCP	Dynamic Host Configuration Protocol
DN	Directory Number
DND	Do Not Disturb
DO	Delayed Offer
DPNSS	Digital Private Network Signaling System
DSCP	Differentiated Services Code Point.
EO	Early Offer
FXS	Foreign Exchange Station
GW	Gateway
HR	Historical Reporting
ICT	Intercluster trunk
IPMA	Cisco IP Manager Assistant
IPPA	IP Phone Agent
IPPM	IP Phone Messenger
ISDN	Integrated Services Digital Network
MGCP	Media Gateway Control Protocol
MOH	Music on hold
MWI	Message Waiting Indicator
NLP	Non Linear Processing
PCA	Personal Communication Assistant
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
RSS	Really Simple Syndication
QRT	Quality Report Tool
QSIG	Q-Signaling protocol
SAF	Service Advertisement Framework
SIP	Session Initiation Protocol
SME	Session Management Edition
SCCP	Skinny Client Control Protocol
SRST	Survivable Remote Site Telephony
SSL	Security Socket layer
TNP	The New Phone
TRP	Trust Relay Point
TUI	Telephony User Interface
UCS	Unified Computing System
UCCX	Cisco Unified Contact Center Express
UMG	Unified Messaging Gateway

Acronym	Description
VGW	Voice Gateway
VoIP	Voice over IP
VPIM	Voice Profile for Instant Messaging
VMN	Voice Mail Notification
WAN	Wide Area Network

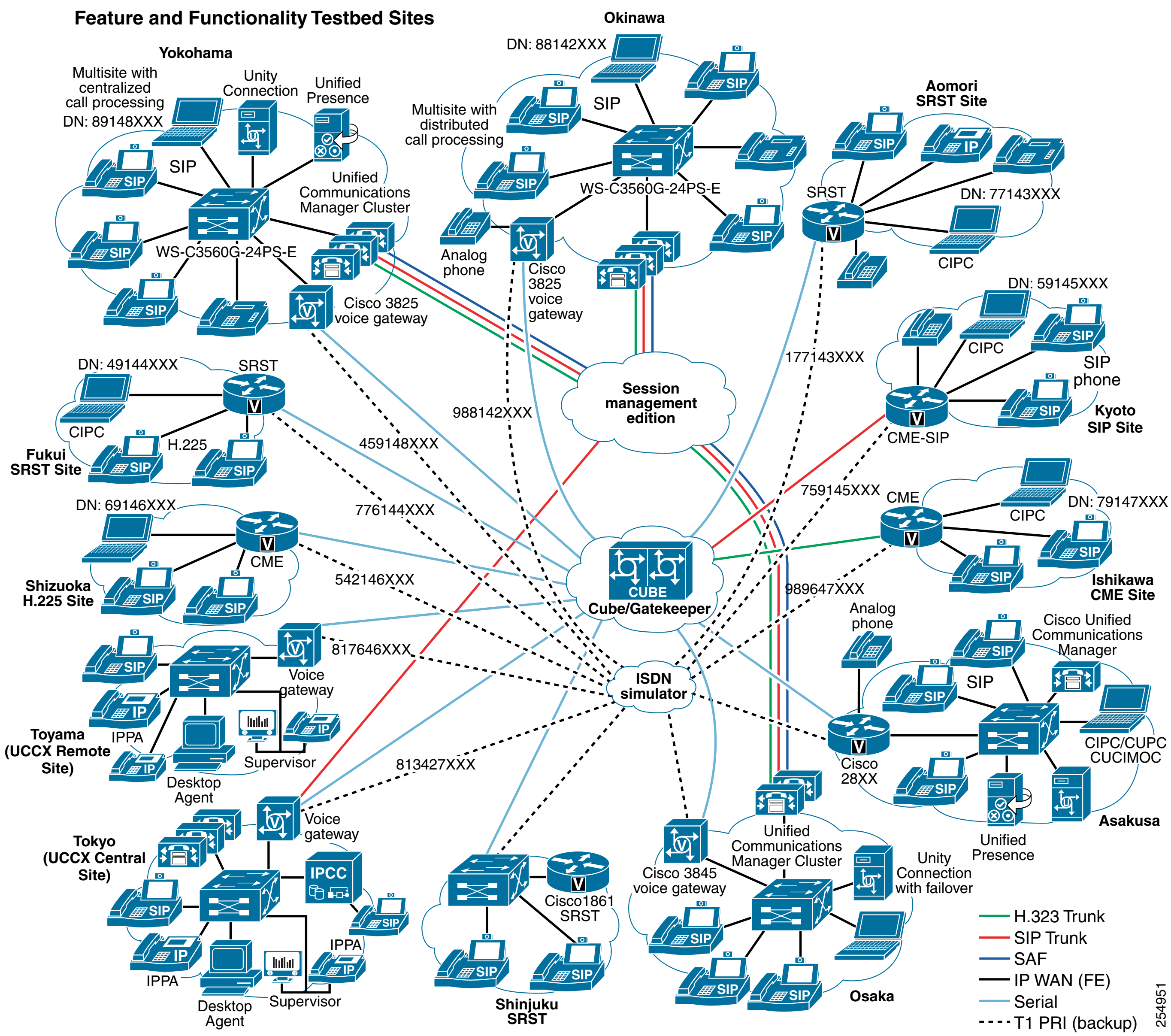
CHAPTER 2

Test Topology and Environment

This section gives information about the test topology, the supporting environment details, and the open caveats:

Test Topology

Figure 2-1 Topology in Use



Environment Matrix

Category	Component	Version	
Call Control	Cisco Unified Communications Manager	Version	8.5.1.10000-26
		Locale	JP(8.5.1.9902-56)
		Dial plan	dp-ffr.3-1-8.JP.cop.sgn
	Unified Survivable Remote Site Telephony (SRST)	Version	8.5
		IOS	15.1(3)T
	Cisco Unified Communications Manager Express	Version	8.5
		IOS	15.1(3)T
		Locale	JP (8.1.2.1)
Applications	Cisco Unified Presence	Version	8.5.1.10000-35
		Locale	JP(8.5.1.9902-150)
Voice Mail and Unified Messaging	Cisco Unity Connection	Version	8.5.1.10000-26
		Locale	JP(8.5.0.0-181)
Endpoints and Clients	Cisco Unified IP Phones		
	SIP 3911		8-1-2SR1
	6921		9-1-1-0
	6941		9-1-1-0
	6961		9-1-1-0
	7961		9-1-1SR1S
	7961G		9-1-1SR1S
	7975		9-1-1SR1S
	7985		cmterm_7985.4-1-7-0
	8961		sip8961.9-1-1
	9951		sip9951.9-1-1
	9971		sip9971.9-1-1
	Cisco Unified Personal Communicator		8.5.1.17660
	UC Integration for Microsoft Office Communicator		8.5.98.16872
	Cisco Supervisor Desktop		8.5.1 JP
	Cisco Agent Desktop		8.5.1 JP
	IP Phone Agent		8.5.1 JP
	Cisco Agent Desktop Browser Edition		8.5.1 JP
Communications Infrastructure	Cisco IOS Voice and Data Gateways	IOS	15.1(3)T
Client	Operating System	Win-XP	Windows XP - SP2 (Japanese)
	Browser	IE	IE 8
Contact Center	Cisco Unified Contact Center Express		8.5.1

What's New?

Cisco Unified Presence 8.5 and Cisco Unified Personal Communicator 8.5 have been used in this solution testing.

The following table describes the New Features in Cisco Unified Communications System Release 8.5(1):

Table 2-1 New Features

New Feature	Description
Single Inbox	Synchronization of voice messages in Cisco Unity Connection and Microsoft Exchange mailboxes.
Service Advertisement Framework (SAF)	Cisco Service Advertisement Framework (SAF) is a network-based, scalable, bandwidth-efficient approach to service advertisement and discovery.
Sesssion Management Engine (SME)	Cisco Unified Communications Manager Session Management Edition extends collaboration applications such as unified messaging, mobility, TelePresence, social networking, and web applications (using Web 2.0 interfaces) to every user in the network. Unified applications are deployed at the network core, so that users in multivendor PBX can use the centrally deployed applications.
Cisco Unified Communications Manager Express and Cisco Unified Border Element	Features such as IP Address Trusted Authentication, TLS support in Unified IP Phone 6900 Series and Unified IP Phone 9900 Series are tested as part of this release. IP Phones 6900 Series and IP Phones 9900 Series support for Unified SRST and Unified CME is tested.
Cisco UC Integration™ for Microsoft Office Communicator	Single Sign-on is supported.

Open Caveats

Open caveats describe the possible unexpected behaviors that you may encounter in Release 8.5(1) of Unified Communications System.

Defect ID	Defect Title
CSCtk62187	Unable to upload scripts through Application Wizard of UCCX
CSCtk67772	ICD states are disabled. Unable to change the state (UCCX CAD)

Limitations

1. Forced Authorization Code is not supported in SIP Phones.
2. Private Voicemails can be forwarded from the Outlook Web Client for Exchange 2010. (The inability to forward private messages is because of Cisco Unity Connection using the best effort approach. In this case, the third-party client (MS-Outlook) that is not in the scope of control is allowing the forward).
3. Localization for Unified IP Phone 6900 Series and Unified IP Phone 9900 Series will be supported by Cisco Unified Communications Manager Express 8.6, which will be released in March 2011.

CHAPTER 3

Test Results Summary

This section lists the various features, the test cases under each feature and the test results. The following features are tested:

- Cisco Unified Communications Manager
- Cisco Unity Connection
- Cisco Unified Presence
- Cisco Unified Border Element
- Cisco Unified Communications Manager Express
- Cisco Unified IP Phone
- Cisco Unified Personal Communicator
- Cisco UC Integration™ for Microsoft Office Communicator
- Cisco Unified Contact Center Express
- Multi-Stage Upgrade
- Single-Stage Upgrade
- Regression Testing

Cisco Unified Communications Manager

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.CUCM.D.001	Cisco Unified Communications Manager	Endpoints displayed in the Route Group Configuration page	Verify that all the 250 endpoints are displayed successfully in the Route Group Configuration page.		Passed	
UCJ851S.CUCM.D.002	Cisco Unified Communications Manager	Generate User Report	Verify that the BAT generates User Report for all users.		Passed	
UCJ851S.CUCM.D.003	Cisco Unified Communications Manager	Insert phones with Intercom lines	Verify that phones with intercom lines are inserted successfully using BAT.		Passed	
UCJ851S.CUCM.D.004	Cisco Unified Communications Manager	Insert FAC by overriding Authorization Code Name	Verify that the FAC overrides using BAT successfully.		Passed	
UCJ851S.CUCM.D.013	Cisco Unified Communications Manager	Reboot time after switch version	Verify that the Cisco Unified Communications Manager does not take a long time to reboot after a switch version.		Passed	
UCJ851S.CUCM.D.017	Cisco Unified Communications Manager	Intercluster calls over a SIP trunk	Verify that intercluster calls over a SIP trunk, connects different versions of Cisco Unified Communications Manager successfully.	Cisco Unified IP Phone -> Cisco Unified Communications Manager -> SIP Trunk -> Cisco Unified Communications Manager -> Cisco Unified IP Phone	Passed	
UCJ851S.SRST.U.001	Cisco Unified Survivable Remote Site Telephony	Authentication and encryption support in SRST mode	Verify that the communication and media between the Unified IP Phone 6900 Series, Unified IP Phone 9900 Series and Cisco Unified Communications Manager Express is secure.	Unified IP Phone 6900 Series -> SRST -> Unified IP Phone 6900 Series	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SRST.U.002	Cisco Unified Survivable Remote Site Telephony	Conference in SRST mode	Verify that conferences can be initiated from an Unified IP Phone 9900 Series, when it is in fallback mode.	Unified IP Phone 6900 Series 1, Unified IP Phone 6900 Series 2 -> SRST -> Conference	Passed	
UCJ851S.SRST.U.003	Cisco Unified Survivable Remote Site Telephony	Placing a PSTN call in SRST mode	Verify that a PSTN call can be initiated from an Unified IP Phone 9900 Series, when it is in the fallback mode.	Unified IP Phone 6900 Series -> SRST -> PSTN phone	Passed	
UCJ851S.SRST.U.004	Cisco Unified Survivable Remote Site Telephony	Shared lines	Verify that the SRST can support shared lines between Unified IP Phone 6900 Series.	PSTN -> SRST -> Unified IP Phone 6900 Series	Passed	
UCJ851S.SRST.U.005	Cisco Unified Survivable Remote Site Telephony	Direct Transfer	Verify that SRST can support Direct Transfer between Unified IP Phone 6900 Series.	Unified IP Phone 6900 Series 1 -> SRST-> Unified IP Phone 6900 Series 2 ,Direct Transfer->Unified IP Phone 6900 Series 3	Passed	
UCJ851S.SAF.U.001	SAF	Hold and Resume a call through a SIP SAF trunk	Verify that the Hold and Resume feature for a call from an Cisco Unified Communications Manager cluster A to an Cisco Unified Communications Manager cluster B through a SIP-SAF trunk works successfully.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager Cluster A -> SIP SAF trunk -> Cisco Unified Communications Manager Cluster B -> Cisco Unified IP Phone B	Passed	
UCJ851S.SAF.U.002	SAF	Hold and Resume a call through a H.323 SAF trunk	Verify that the Hold and Resume feature for a call from an Cisco Unified Communications Manager cluster A to an Cisco Unified Communications Manager cluster B through a H.323-SAF trunk works successfully.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager Cluster A -> H.323 SAF trunk -> Cisco Unified Communications Manager Cluster B -> Cisco Unified IP Phone B	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SAF.U.003	SAF	Call Transfer through a SIP-SAF trunk	Verify that a call tranfser from an Cisco Unified Communications Manager cluster A to an Cisco Unified Communications Manager cluster B through a SIP-SAF trunk works successfully.	Cisco Unified IP Phone 1A -> Cisco Unified Communications Manager Cluster A -> Cisco Unified IP Phone 1B -> Transfer -> SIP SAF trunk -> Cisco Unified Communications Manager Cluster B -> Cisco Unified IP Phone 2A	Passed	
UCJ851S.SAF.U.004	SAF	Call Transfer through a H.323-SAF trunk	Verify that a call transfer from an Cisco Unified Communications Manager cluster A to an Cisco Unified Communications Manager cluster B through a H.323-SAF trunk works successfully.	Cisco Unified IP Phone 1A -> Cisco Unified Communications Manager Cluster A -> Cisco Unified IP Phone 1B -> Transfer -> H.323-SAF trunk -> Cisco Unified Communications Manager Cluster B -> Cisco Unified IP Phone 2A	Passed	
UCJ851S.SAF.U.005	SAF	Add a new SAF edge forwarder in the SAF network	Verify that configuring a new SAF forwarder, which is located in CME and joining it to the SAF network, does not affect other learning routes.		Passed	
UCJ851S.SAF.U.006	SAF	Call to an Cisco Unified IP Phone in the newly added SAF edge forwarder	Verify that a call to an Cisco Unified IP Phone, which is in the newly added SAF edge forwarder, from an Cisco Unified Communications Manager cluster is successful.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SAF.U. 007	SAF	Remove a SAF edge forwarder from the SAF network	Verify that the DN patterns are not advertised after the removal of a SAF forwarder from the network. The other valid learned routes are also not affected by the removal of the forwarder.		Passed	
UCJ851S.SAF.U. 008	SAF	WAN link between an edge and transit forwarder flapping	Verify that the learned routes are not affected after flapping the WAN link between an edge and a transit forwarder.		Passed	
UCJ851S.SAF.U. 009	SAF	Transfer the RSVP SAF call from cluster A to cluster B	Verify that a call configured with RSVP between two clusters is transferred successfully through a SIP SAF trunk.	Cisco Unified IP Phone 1 (in RSVP location) -> Cisco Unified Communications Manager A -> SAF trunk -> Cisco Unified Communications Manager B -> Cisco Unified IP Phone 2 (in RSVP location) -> Transfer -> Cisco Unified IP Phone 3 (in RSVP location)	Passed	
UCJ851S.SAF.U. 010	SAF	Post a VoiceMail in an Cisco Unified IP Phone in cluster B through a SIP SAF trunk	Verify that a VoiceMail is successfully posted in an Cisco Unified IP Phone, which is present in cluster B, through a SIP SAF trunk.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager A -> SIP SAF trunk -> Cisco Unified Communications Manager B -> Cisco Unified IP Phone B -> VoiceMail -> Cisco Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SAF.U.011	SAF	Post a VoiceMail in a RSVP enabled Cisco Unified IP Phone through a SIP SAF trunk.	Verify that a VoiceMail is successfully posted to a RSVP enabled Cisco Unified IP Phone through a SIP SAF trunk.	Cisco Unified IP Phone A (RSVP Enabled) -> Cisco Unified Communications Manager A -> SIP SAF trunk -> Cisco Unified Communications Manager B -> Cisco Unified IP Phone B (RSVP Enabled) -> VoiceMail -> Cisco Unity Connection	Passed	
UCJ851S.SAF.U.012	SAF	SIP SAF call with the CfdAll enabled in the Called Party	Verify that a call to an Cisco Unified IP Phone with the CfdAll enabled is forwarded successfully.	Cisco Unified IP Phone -> Cisco Unified Communications Manager 1 -> SAF trunk -> Cisco Unified Communications Manager 2 -> Cisco Unified IP Phone -> CfdAll -> Cisco Unified IP Phone C	Passed	
UCJ851S.SAF.U.013	SAF	SAF call with the RSVP enabled end-to-end.	Verify that a call through a SIP SAF trunk (end-to-end RSVP enabled) is successful.	IP Phone (RSVP Enabled) -> Cisco Unified Communications Manager 1 -> e2e RSVP SIP SAF trunk -> Cisco Unified Communications Manager 2 -> IP Phone (RSVP Enabled)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SAF.U.014	SAF	RSVP SIP SAF call with the CfdAll enabled in the Called Party	Verify that a call to an Cisco Unified IP Phone with the CfdAll enabled in a RSVP location is forwarded successfully.	Cisco Unified IP Phone (RSVP Enabled) -> Cisco Unified Communications Manager 1 -> SAF trunk -> Cisco Unified Communications Manager 2 -> Cisco Unified IP Phone(RSVP Enabled) -> CfdAll -> Cisco Unified IP Phone C	Passed	
UCJ851S.SAF.U.015	SAF	Conference through a SIP SAF trunk	Verify that a conference through a SIP SAF trunk is successful.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager A -> Cisco Unified IP Phone B -> Conference -> SIP SAF trunk -> Cisco Unified Communications Manager B -> Cisco Unified IP Phone C	Passed	
UCJ851S.SAF.U.016	SAF	Swap the SIP and H.323 SAF trunks in the Advertising service	Verify that calls can be placed successfully after swapping the SIP and H.323 SAF trunks.		Passed	
UCJ851S.SAF.U.017	SAF	Conference through a H.323 SAF trunk	Verify that a conference through a H.323 SAF trunk is successful.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager A -> Cisco Unified IP Phone B -> Conference -> H.323 SAF trunk -> Cisco Unified Communications Manager B -> Cisco Unified IP Phone C	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SAF.U.018	SAF	Call Park through a H.323 SAF trunk	Verify that a H.323 SAF trunk call is parked and retrieved successfully.	Cisco Unified IP Phone 1A -> Cisco Unified Communications Manager Cluster A -> H.323 SAF trunk -> Cisco Unified Communications Manager Cluster B -> Cisco Unified IP Phone 1B -> Park -> Cisco Unified IP Phone 2B -> Park retrieval	Passed	
UCJ851S.SAF.U.019	SAF	Call Park through a SIP SAF trunk	Verify that a SIP SAF trunk call is parked and retrieved successfully.	Cisco Unified IP Phone 1A -> Cisco Unified Communications Manager Cluster A -> SIP SAF trunktrunk -> Cisco Unified Communications Manager Cluster B -> Cisco Unified IP Phone 1B -> Park -> Cisco Unified IP Phone 2B -> Park retrieval	Passed	
UCJ851S.SAF.U.020	SAF	Manual summarization using the Cisco Unified Border Element	Verify that an Cisco Unified Border Element is used manually to summarize SAF advertisements from one SAF AS and re-advertise them to another AS.	Cisco Unified IP Phone -> Cisco Unified Communications Manager -> H.225 trunk -> Cisco Unified Communications Manager -> Cisco Unified IP Phone	Passed	
UCJ851S.SAF.U.021	SAF	Loss of Connectivity between an Advertising Client and SAF Forwarder	Verify that any change to the advertised DN is pushed to the Forwarder, after the connectivity between the advertising client and SAF forwarder is restored.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SAF.U.022	SAF	Hold and Resume a call through a SME, which is between two leaf clusters over a SAF and H.323 trunk	Verify that the Hold and Resume feature for a call from a leaf cluster A to a leaf cluster B through a SME is successful. A SAF trunk is configured between leaf cluster A and SME. A H.323 or SIP trunk is configured between leaf cluster B and SME.	Cisco Unified IP Phone -> Cisco Unified Communications Manager A -> SIP SAF trunk -> SME -> SIP/H.323 trunk -> Cisco Unified Communications Manager B -> Cisco Unified IP Phone	Passed	
UCJ851S.SAF.U.023	SAF	Ad-hoc conference in the Cisco Unified IP Phones from a leaf cluster A to leaf cluster B through a SAF trunk	Verify that a three-way ad-hoc conference in Cisco Unified IP Phones from leaf cluster A to leaf cluster B through a SAF trunk is successful.	Cisco Unified IP Phone -> Cisco Unified Communications Manager A -> H.323 SAF trunk -> SME -> SIP/H.323 trunk -> Cisco Unified Communications Manager B -> Cisco Unified IP Phone (Or conf bridge)	Passed	
UCJ851S.SAF.U.024	SAF	Post a VoiceMail in an end-to-end enabled RSVP SAF call through a SME	Verify that a VoiceMail is successfully posted in an End to End RSVP enabled Cisco Unified IP Phone through SIP SAF trunk and SME.	Cisco Unified IP Phone 1 (in RSVP location) -> Cisco Unified Communications Manager A -> SAF trunk -> SME -> SIP trunk -> Cisco Unified Communications Manager B -> Cisco Unified IP Phone 2 (in RSVP location)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SAF.U.025	SAF	RSVP call through a SME between two leaf clusters over a SAF and H.323 trunk	Verify that an RSVP enabled call from a leaf cluster A to a leaf cluster B through a SME is successful. SAF trunk is configured between Leaf Cluster A and SME. H.323 trunk is configured between Leaf Cluster B and SME.	Cisco Unified IP Phone 1 (in RSVP location) -> Cisco Unified Communications Manager A -> SAF trunk -> SME -> H.323 trunk -> Cisco Unified Communications Manager B -> Cisco Unified IP Phone 2 (in RSVP location)	Passed	
UCJ851S.SAF.U.026	SAF	Loss of Connectivity between Service Advertisement Framework Forwarders.	Verify that the clients are able to maintain the connectivity with their forwarders when the connectivity between the two Service Advertisement Framework forwarders is lost.		Passed	
UCJ851S.SME.U.001	SME	Hold and Resume a call through a SME, which is between two leaf clusters over a SIP trunk.	Verify that the Hold and Resume feature for a call from a leaf cluster A to a leaf cluster B through a SME is successful. SIP trunks are configured between SME and Leaf clusters.	Cisco Unified IP Phone -> Cisco Unified Communications Manager (Yokohama) -> SIP trunk -> SME -> SIP trunk -> Cisco Unified Communications Manager B (Okinawa) -> Cisco Unified IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SME.U.002	SME	Transfer a call through a SME, which is between two leaf clusters over an ICT and SIP trunk	Verify that a call from a leaf cluster A to a leaf cluster B through a SME is transferred to the another Cisco Unified IP Phone in leaf cluster B. An ICT trunk is configured between leaf cluster A and SME. A SIP trunk is configured between leaf cluster B and SME.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager (Yokohama) -> ICT trunk -> SME -> SIP trunk -> Cisco Unified Communications Manager (Okinawa) -> Cisco Unified IP Phone B -> Transfer -> Cisco Unified IP Phone C.	Passed	
UCJ851S.SME.U.003	SME	Blind Transfer a call through a SME, which is between two leaf clusters over an ICT trunk.	Verify that a call from a leaf cluster A to a leaf cluster B through a SME is Blind transferred to another Cisco Unified IP Phone present in the originating cluster. An ICT trunk is configured between the leaf clusters and SME.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager (Yokohama) -> ICT trunk -> SME -> ICT trunk -> Cisco Unified Communications Manager (Okinawa) -> Cisco Unified IP Phone B -> Blind Transfer -> ICT trunk -> SME -> ICT trunk -> Cisco Unified Communications Manager (Yokohama) -> Cisco Unified IP Phone C	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SME.U.004	SME	Consult Transfer a call through a SME, which is between two leaf clusters over an ICT trunk	Verify that a call from a leaf cluster A to a leaf cluster B through a SME is Consult transferred to the another Cisco Unified IP Phone present in the originating cluster. An ICT trunk is configured between the leaf clusters and SME.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager (Yokohama) -> ICT trunk -> SME -> ICT trunk -> Cisco Unified Communications Manager (Okinawa) -> Cisco Unified IP Phone B -> Consult Transfer -> ICT trunk -> SME -> ICT trunk -> Cisco Unified Communications Manager (Yokohama) -> Cisco Unified IP Phone C	Passed	
UCJ851S.SME.U.005	SME	Blind Transfer a call through a SME, which is between two leaf clusters over a SIP and ICT trunk	Verify that a call from a leaf cluster A to a leaf cluster B through a SME is Blind transferred to another Cisco Unified IP Phone present in the originating cluster. A SIP trunk is configured between leaf cluster A and SME. An ICT trunk is configured between leaf cluster B and SME.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager (Yokohama) -> SIP trunk -> SME -> ICT trunk -> Cisco Unified Communications Manager (Okinawa) -> Cisco Unified IP Phone B -> Blind Transfer -> ICT trunk -> SME -> SIP trunk -> Cisco Unified Communications Manager(Yokohama) -> Cisco Unified IP Phone C.	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SME.U .007	SME	Transfer a call through a SME, which is between two leaf clusters over a SIP and ICT trunk	Verify that a call from a leaf cluster A to a leaf cluster B through a SME is transferred to another Cisco Unified IP Phone in leaf cluster B. A SIP trunk is configured between leaf cluster A and SME. An ICT trunk is configured between leaf cluster B and SME.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager (Yokohama) -> SIP trunk -> SME -> ICT trunk -> Cisco Unified Communications Manager (Okinawa) -> Cisco Unified IP Phone B -> Transfer- -> Cisco Unified IP Phone C	Passed	
UCJ851S.SME.U .014	SME	A call from an EMCC IP Phone is consult transferred to another Cisco Unified IP Phone in the visiting cluster through a SME	Verify that a call from an EMCC logged Cisco Unified IP Phone to another Cisco Unified IP Phone in the visiting cluster performs consult transfer successfully. The call that originates from the home cluster traverses through a SIP trunk to a SME and then connects the visiting cluster over another SIP trunk.	EMCC Cisco Unified IP Phone -> Cisco Unified Communications Manager (Home cluster) -> SIP trunk -> SME -> SIP trunk -> Cisco Unified Communications Manager (Visiting cluster) -> Cisco Unified IP Phone 1 -> Consult Transfer -> Cisco Unified IP Phone 2.	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SME.U.015	SME	Call Park and retrieve through a SME, which is configured between two leaf clusters over a secure SIP trunk.	Verify that a call from a Cisco Unified IP Phone in leaf cluster A to a Cisco Unified IP Phone in leaf cluster B through a SME, over a secure SIP trunk is parked and then retrieved from another Cisco Unified IP Phone in cluster B.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager A (Yokohama) -> secure SIP trunk -> SME -> secure SIP trunk -> Cisco Unified Communications Manager B (Okinawa) -> Cisco Unified IP Phone B -> Call Park(2008) -> Cisco Unified IP Phone C -> Dials 2008.	Passed	
UCJ851S.SME.U.016	SME	Blind Transfer to Hunt List through a SME, over a secure SIP QSIG trunk.	Verify that a call from a Cisco Unified IP Phone in leaf cluster A to a Cisco Unified IP Phone in leaf cluster B through a SME, over a secure SIP trunk is Blind transferred to a Hunt List in cluster A.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager A (Yokahama) -> secure SIPT -> SME -> secure SIPT -> Cisco Unified Communications Manager B (Okinawa) -> Cisco Unified IP PhoneB -> Blind Transfer -> secure SIPT -> SME -> secure SIPT -> Cisco Unified Communications Manager A (Yokohama) -> Hunt List	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SME.U.017	SME	Blind Transfer a call through a SME, which is between leaf cluster A and CME over a SIP trunk.	Verify that a call from a Cisco Unified IP Phone in leaf cluster A to a Cisco Unified IP Phone in CME B through a SME, is Blind transferred to another Cisco Unified IP Phone present in the originating cluster.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager A (Yokohama) -> SIPT -> SME -> SIPT -> CUBE -> SIPT -> CME -> Cisco Unified IP Phone B -> Blind Transfer -> SIPT -> CUBE -> SIPT -> SME -> SIPT -> Cisco Unified Communications Manager A (Yokohama) -> Cisco Unified IP Phone C	Passed	
UCJ851S.SME.U.018	SME	Consult Transfer a call through a SME, which is between leaf cluster A and CME over a SIP trunk	Verify that a call from a Cisco Unified IP Phone in leaf cluster A to a Cisco Unified IP Phone in CME B through a SME, is Consult transferred to another Cisco Unified IP Phone present in the originating cluster.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager A (Yokohama) -> SIPT -> SME -> SIPT -> CUBE -> SIPT -> CME -> Cisco Unified IP Phone B -> Consult Transfer -> SIPT -> CUBE -> SIPT -> SME -> SIPT -> Cisco Unified Communications Manager A (Yokohama) -> Cisco Unified IP Phone C	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SME.U .020	SME	Transfer to MGCP PSTN when the WAN between leaf cluster and SME is down	Verify that a call from leaf cluster A to leaf cluster B is established over PSTN. The call is then Blind transferred from leaf cluster B to MGCP PSTN.	Cisco Unified IP Phone -> Cisco Unified Communications Manager A (Yokohama) -> SIPT -> SME -> PSTN -> Cisco Unified Communications Manager B (Okinawa) -> SIPT -> SME -> PSTN	Passed	
UCJ851S.SME.U .021	SME	EMCC user calls an EMCC user present in a visiting cluster, over a secure SIP trunk through an SME	Verify that an EMCC Phone makes a call to an EMCC user present in the visiting cluster and performs consult transfer successfully.	EMCC Cisco Unified IP Phone B -> Cisco Unified Communications Manager A (Yokohama) -> secure SIPT -> SME -> secure SIPT -> Cisco Unified Communications Manager B (Okinawa) -> EMCC Cisco Unified IP Phone A	Passed	
UCJ851S.SME.U .022	SME	Call from PSTN through a H.323 GW in SME toan UCCX agent and transfer the call to another agent	Verify that an inbound call from a H.323 gateway in SME can be routed to an UCCX agent and then transferred to another agent successfully.	PSTN Phone -> H.323 GW -> SME -> SIPT -> Cisco Unified Communications Manager -> UCCX -> Cisco Unified Communications Manager -> CAD Agent 1 -> Transfer -> CAD Agent 2	Passed	
UCJ851S.SME.U .024	SME	Conference Chaining between two clusters through a SME, over a SIP trunk	Verify that two conferences are chained successfully that is between two clusters through a SME over a SIP trunk.	Cisco Unified IP Phone -> Cisco Unified Communications Manager 1 -> SIPT -> SME -> SIPT -> Cisco Unified Communications Manager 2 -> Conf Bridge	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SME.U.025	SME	Blind Transfer over a dual stack SIPT with the ANAT (EO) enabled through an SME to a dual stack phone	Verify that a call from a local cluster to a remote cluster over a dual stack SIP trunk, through a SME is blind transferred to a dual stack phone.	DS Cisco Unified IP Phone -> Cisco Unified Communications Manager (Yokohama) -> SIPT (DS, IPv6 media, IPv4 and IPv6 Signalling, Early Offer) -> SME -> SIPT -> (DS, IPv6 media, IPv4 and IPv6 Signalling, Early Offer) -> Cisco Unified Communications Manager (Okinawa) -> DS Cisco Unified IP Phone -> Xfer (Blind) -> DS Cisco Unified IP Phone	Passed	
UCJ851S.SME.U.027	SME	Call to Hunt Pilot with broadcast distribution over a dual stack SIPT through a SME	Verify that a call to a hunt pilot with IPv4, dual stack, and softphones in the line group can be routed successfully through a SME.	DS Cisco Unified IP Phone -> Cisco Unified Communications Manager (Yokohama) -> SIPT (DS, IPv6 media, IPv4 and IPv6 Signalling, Early Offer) -> SME -> SIPT -> (DS, IPv6 media, IPv4 and IPv6 Signalling, Early Offer) -> Cisco Unified Communications Manager (Okinawa) -> Cisco Unified IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SME.U.030	SME	Adhoc conference that involves dual stack SIP trunk and ICT trunks through a SME and a PSTN phone	Verify that an adhoc conference can be placed successfully through a SME, which involves a dual stack SIP trunk, inter-cluster trunk, and a PSTN phone. The SME is configured as the centralized PSTN break out site.	Cisco Unified IP Phone 1 -> Cisco Unified Communications Manager (Yokohama) -> SIPT (DS, IPv6 media, IPv4 and IPv6 Signalling, Early Offer) -> SME -> ICT -> Cisco Unified Communications Manager (Okinawa) -> Cisco Unified IP Phone 2; Conference -> Cisco Unified IP Phone 1 -> Cisco Unified Communications Manager (Yokohama) -> SIPT -> (DS, IPv6 media, IPv4 and IPv6 Signalling, Early Offer) -> SME -> SIPT (DS, IPv6 media, IPv4 Sig, Early Offer) -> DS SIP GW -> PSTN	Passed	
UCJ851S.SME.U.031	SME	Hold and Resume a call that involves a TRP, early offer, and delayed offer trunks to SME	Verify that the Hold and Resume feature for a call works successfully, when one trunk to SME is enabled for early offer and the other trunk from SME to the remote cluster is enabled for delayed offer.	Cisco Unified IP Phone 1 -> Cisco Unified Communications Manager -> SIPT (DS, IPv6 media, IPv4 Sig, Early Offer) -> SME -> SIPT -> Cisco Unified Communications Manager -> Cisco Unified IP Phone 2; Cisco Unified IP Phone 1 -> Hold	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SME.U .032	SME	Consult transfer an IPv6 call to SME through early offer and delayed offer trunks	Verify that a call can be transferred (Consult) when one trunk to SME is enabled for early offer and the other trunk from SME to the remote cluster is enabled for delayed offer.	Cisco Unified IP Phone 1 -> Cisco Unified Communications Manager (Yokohama) -> SIPT (DS, IPv6 media, IPv4 Sig, Early Offer) -> SME -> SIPT (DS, IPv6 media, IPv4 Sig, Delayed Offer) -> Cisco Unified Communications Manager (Okinawa) -> Cisco Unified IP Phone 2 -> Xfer (Consult) -> Cisco Unified Communications Manager -> SIPT (DS, IPv6 media, IPv4 Sig, Delayed Offer) -> SME -> SIPT (DS, IPv6 media, IPv4 Sig, Early Offer) -> DS SIP GW -> PSTN	Passed	
UCJ851S.SME.U .036	SME	Call Pickup over a dual stack SIP trunk through a SME	Verify that PSTN calls from SME can be answered successfully using the PickUp softkey.	Cisco Unified IP Phone 1 -> Cisco Unified Communications Manager (Yokohama) -> SIPT (DS, IPv6 media, IPv4 Sig, Early Offer) -> SME -> SIPT (DS, IPv6 media, IPv4 Sig, Early Offer) -> Cisco Unified Communications Manager (Okinawa) -> Cisco Unified IP Phone 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SME.U.040	SME	Invoke MTP in SME site for a call from SIP Gateway, due to codec mismatch	Verify that media resources are invoked when there is a need in the SME site.	PSTN Phone -> SIP GW -> SME -> Xcoder -> UC Application	Passed	
UCJ851S.SME.U.041	SME	Call from PSTN through a H.323 GW in SME to UCCX over SIP trunk	Verify that inbound calls from a H.323 gateway in SME can be routed to UCCX successfully.	PSTN Phone -> H.323 GW -> SME -> SIPT -> Cisco Unified Communications Manager -> UCCX -> Cisco Unified Communications Manager -> CAD Agent	Passed	
UCJ851S.SME.U.042	SME	Early offer call to Cisco Unity Connection through a SME using SIP integration	Verify that Cisco Unity Connection supports early offer calls through SME using SIP integration.	Cisco Unified IP Phone -> Cisco Unified Communications Manager -> SIPT (Early offer) -> SME -> SIPT (Early Offer) -> Cisco Unity Connection	Passed	
UCJ851S.SME.U.043	SME	Supervise transfer a call to a phone registered with the Cisco Unified Communications Manager from PSTN	Verify that Cisco Unity Connection can supervise transfer a call from PSTN to a phone registered to Cisco Unified Communications Manager.	PSTN Phone -> SIP GW -> SME -> SIPT -> Cisco Unity Connection -> Supervise Xfer -> SME -> SIPT -> Cisco Unified Communications Manager -> Cisco Unified IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.SME.U.044	SME	Cisco Unity Connection in SME provides Voicemail service to phones present in the two clusters	Verify that Cisco Unity Connection in SME provides Voicemail service to users in the two Cisco Unified Communications Manager clusters.	Cisco Unified IP Phone 1 -> Cisco Unified Communications Manager -> ICT/SIPT -> SME -> ICT/SIPT -> Cisco Unified Communications Manager -> Cisco Unified IP Phone 2 -> CFA/CFB/CFNA -> Cisco Unified Communications Manager -> ICT/SIPT -> SME -> Cisco Unity Connection	Passed	
UCJ851S.SME.U.045	SME	Cisco Unity Connection SCCP integration with SME	Verify that calls can be made to Cisco Unity Connection in SME from phones registered with Cisco Unified Communications Manager.	Cisco Unified IP Phone -> Cisco Unified Communications Manager -> SIPT -> SME -> SCCP -> Cisco Unity Connection -> Release transfer -> SME -> SIPT -> Cisco Unified Communications Manager -> Cisco Unified IP Phone	Passed	

Cisco Unity Connection

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.CUC.U.05	Cisco Unity Connection	Cisco Unity Connection SCCP Integration (IPv6 Enabled in Cisco Unity Connection and Cisco Unified Communications Manager)	Verify that the SCCP Integration works successfully with dual stack.	Cisco Unified IP Phone -> Cisco Unified Communications Manager -> Cisco Unified IP Phone -> CFNA -> SCCP Cisco Unity Connection	Passed	
UCJ851S.CUC.U.06	Cisco Unity Connection	Diversion header support for multiple call forwards in a dual stack environment	Verify that the Cisco Unity Connection dual stack integration can support more than one diversion header with an early offer.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> Forward -> Cisco Unified IP Phone C -> SCCP -> Cisco Unity Connection	Passed	
UCJ851S.CUC.U.07	Cisco Unity Connection	Message notification through a dual stack SIP integration	Verify that the message notification through a dual stack SIP integration is successful.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager -> IP Phone B -> CFNA -> Cisco Unity Connection	Passed	
UCJ851S.CUC.U.10	Cisco Unity Connection	Transcription services (Speech View) for dual stack configuration	Verify that speech view can be invoked, when the media is IPv6 for a call to Cisco Unity Connection.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> CFNA -> Cisco Unity Connection	Passed	
UCJ851S.CUC.U.11	Cisco Unity Connection	Transfer an IPV6 call to an alternate contact number	Verify that Cisco Unity Connection can transfer an IPV6 call to an alternate contact number successfully.		Passed	
UCJ851S.CUC.U.15	Cisco Unity Connection	Accessing email present in an external message store for an IPv6 call	Verify that subscribers can call Cisco Unity Connection and hear the emails present in the exchange successfully.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.CUC.U.17	Cisco Unity Connection	Single Inbox	Verify that the Single Inbox feature works successfully when the Cisco Unified Communications Manager and Cisco Unity Connection are integrated using IPv6 (SCCP/SIP).	Cisco Unified IP Phone A -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> CFNA -> Cisco Unity Connection	Passed	
UCJ851S.CUC.U.18	Cisco Unity Connection	PSTN access from a dual stack SIP Gateway to Cisco Unity Connection	Verify that Cisco Unity Connection can inter-operate seamlessly with Cisco Unity Connection. The SIP Gateway and Cisco Unity Connection support IPv4 and IPv6.	Analog Phone -> SIP Gateway -> Cisco Unified Communications Manager -> Cisco Unified IP Phone -> CFNA -> Cisco Unity Connection	Passed	
UCJ851S.CUC.U.19	Cisco Unity Connection	Voice Mail for user	Verify that a Voicemail is sent successfully when the dual stack is enabled.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> CFNA -> Cisco Unity Connection	Passed	
UCJ851S.CUC.U.20	Cisco Unity Connection	Reply to a voicemail by placing a call, when the media is IPv6	Verify that the Cisco Unity Connection responds to a message by placing a call. Ensure that the media for the call is IPv6.		Passed	
UCJ851S.CUC.U.21	Cisco Unity Connection	Addressing and deleting a message through a TUI	Verify that the user can listen to Voicemails and delete them through a TUI, when IPv6 is enabled.		Passed	
UCJ851S.CUC.U.22	Cisco Unity Connection	Single Inbox - Secure message synchronization with MS-Exchange 2010	Verify that Cisco Unity Connection can synchronize messages that are marked as secure by the sender.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> CFNA -> Cisco Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.CUC.U.23	Cisco Unity Connection	Single Inbox - New and saved messages with MWI indicator	Verify that the MWI is turned ON when the saved message is marked as new. The MWI status should be updated based on the actions specified in the email client.		Passed	
UCJ851S.CUC.U.24	Cisco Unity Connection	Single Inbox - Synchronizing voice message with attachments	Verify that Cisco Unity Connection can synchronize voice message with the attachments successfully.		Passed	
UCJ851S.CUC.U.25	Cisco Unity Connection	Single Inbox - Deleting voicemails from MS-Exchange 2010, using Email client	Verify that Voicemails can be deleted successfully from MS-Exchange 2010, using Email client.		Passed	
UCJ851S.CUC.U.26	Cisco Unity Connection	Single Inbox - Marking voice messages urgent	Verify that the status of a Voicemail marked as urgent is updated in the email client.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> CFNA -> Cisco Unity Connection	Passed	
UCJ851S.CUC.U.26	Cisco Unity Connection	Single Inbox - Marking voice messages urgent	Verify that the status of a Voicemail marked as urgent is updated in the email client.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> CFNA -> Cisco Unity Connection	Passed	
UCJ851S.CUC.U.27	Cisco Unity Connection	Single Inbox - Read receipts for synchronized voicemails	Verify that read receipts are synchronized with the email client.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> CFNA -> Cisco Unity Connection	Passed	
UCJ851S.CUC.U.28	Cisco Unity Connection	Single Inbox-Message synchronization when the primary Connection server in the cluster is down	Verify that synchronization happens successfully, when the primary connection server is down.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.CUCM.D.014	Cisco Unity Connection	Redirecting a Conference party to Voicemail	Verify that redirecting a Conference party to Voicemail does not fail.		Passed	
UCJ851S.CUC.D.001	Cisco Unity Connection	Text to Speech	Verify that the Text to speech feature with display name works successfully.	Cisco Unified IP Phone A -> Unified Call Manager -> Cisco Unified IP Phone B -> Cisco Unity Connection.	Passed	
UCJ851S.CUC.D.002	Cisco Unity Connection	Cisco Unity Connection License	Verify that the Cisco Unity Connection supports Unity License.		Passed	
UCJ851S.CUC.D.003	Cisco Unity Connection	Connection SMTP Server	Verify that the SMTP Server sends and receives messages successfully.		Passed	
UCJ851S.CUC.D.004	Cisco Unity Connection	Post Greeting Recorded message in a single site	Verify that the Post Greeting in a single site is played correctly.	Cisco Unified IP Phone A -> Unified Call -> Cisco Unified IP Phone B -> Cisco Unity Connection	Passed	
UCJ851S.CUC.D.005	Cisco Unity Connection	Post Greeting Recorded Message in a multi-site	Verify that the Post Greeting in a multi-site is played correctly.	Cisco Unified IP Phone A -> Unified Call Manager A -> ICT -> Unified Call Manager B -> Cisco Unified IP Phone B -> Cisco Unity Connection.	Passed	
UCJ851S.CUC.D.006	Cisco Unity Connection	Call Routing rule - Forwarded Routing rule	Verify that the Forwarded Call Routing rule in Cisco Unity Connection works successfully.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> Cisco Unity Connection. Cisco Unified IP Phone C -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> Cisco Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.CUC .D.007	Cisco Unity Connection	Call Routing rule - Direct Routing Rule	Verify that the Direct Call Routing rule in Cisco Unity Connection works successfully.	Cisco Unified IP Phone A -> Cisco Unity Connection. Cisco Unified IP Phone B -> Cisco Unity Connection.	Passed	
UCJ851S.CUC .D.008	Cisco Unity Connection	Bulk Administration Tool	Verify that a csv file is imported successfully using BAT .		Passed	
UCJ851S.CUC .D.009	Cisco Unity Connection	SMTP Timeout	Verify that the number of seconds to wait for SMTP response is configurable.		Passed	
UCJ851S.CUC .D.010	Cisco Unity Connection	Voice Enabled Directory Handler - Direct Routing	Verify that the Voice Enabled Directory Handler works successfully for Direct routing.		Passed	
UCJ851S.CUC .D.011	Cisco Unity Connection	Voice Enabled Directory Handler -Forward Routing	Verify that the Voice Enabled Directory Handler works successfully for Forwarded routing.		Passed	
UCJ851S.CUC .D.012	Cisco Unity Connection	Voice Enabled Directory Handler - Custom Enabled Routing	Verify that the Voice Enabled Directory Handler works successfully for custom enabled greeting.		Passed	
UCJ851S.CUC .D.043	Cisco Unity Connection	Move a message from deleted items to the Inbox in the Cisco Unity Connection Mailbox.	Verify that Cisco Unity Connection and Exchange Synchronization are working correctly.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> CFNA -> Cisco Unity Connection	Passed	
UCJ851S.CUC .D.044	Cisco Unity Connection	Move a secure message to a folder in an email client	Verify that a secure message is moved successfully to a folder in an email client.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> CFNA -> Cisco Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.CUC.D.045	Cisco Unity Connection	Move a private message to a folder in an email client	Verify that a private message is moved successfully to a folder in an email client.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> CFNA -> Cisco Unity Connection	Passed	
UCJ851S.CUC.D.046	Cisco Unity Connection	Forward a private message from an email client	Verify that a private message is forwarded successfully from an email client.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> CFNA -> Cisco Unity Connection	Passed	
UCJ851S.CUC.D.047	Cisco Unity Connection	Move a Voicemail to any other folder other than an inbox folder in an email client	Verify that a Voicemail is moved successfully to any other folder other than an inbox folder in an email client.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> CFNA -> Cisco Unity Connection	Passed	
UCJ851S.CUC.D.048	Cisco Unity Connection	Move a Voicemail to a private folder in an email client	Verify that a Voicemail is moved successfully to a private folder in an email client.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> CFNA -> Cisco Unity Connection	Passed	
UCJ851S.CUC.D.049	Cisco Unity Connection	Move the Unified Messaging user exchange Mailbox to a different database in the Exchange	Verify that synchronization is successful, when the Unified Messaging user exchange Mailbox is moved to a different database in the MS Exchange.		Passed	
UCJ851S.CUC.D.050	Cisco Unity Connection	Unified Messaging User Message Length	Verify that the Unified Messaging users Message length setting is working successfully.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.CUC.D.051	Cisco Unity Connection	Unified Messaging users mailbox restriction	Verify that Unified Messaging user mailbox restriction is working successfully.		Passed	
UCJ851S.CUC.D.052	Cisco Unity Connection	Importing user from LDAP	Verify that the extension field is editable for LDAP users.		Passed	

Cisco Unified Presence

ID	Feature teated	Case Title	Description	Call Component Flow	Status	Defects
UC851F. CUP.001	Cisco Unified Presence	Cisco Unified Presence upgrade	Verify the following after the two Cisco Unified Presence cluster peers are upgraded from 8.0 to 8.5: • The sub-cluster logical topology remains intact and High availability is not enabled immediately. • The user buddy lists remain intact and intra and inter-cluster presence functions properly.	Cisco Unified Personal Communicator 1 -> Cisco Unified Presence Server 1 -> WAN -> Cisco Unified Presence Server 2 -> Cisco Unified Personal Communicator 2	Passed	
UC851F. CUP.002	Cisco Unified Presence	HA over WAN client failover during manual fallback	Verify that the Cisco Unified Presence node failure in the CoW deployment, results in all clients and communication to failover automatically to another Cisco Unified Presence node located across the WAN with a 80ms delay. The clients fallback to the original configuration when the node is UP and manual fallback is initiated.	Cisco Unified Personal Communicator 1 -> Cisco Unified Presence Server 1; After failover: Cisco Unified Personal Communicator 1 -> WAN -> Cisco Unified Presence Server 2	Passed	
UC851F. CUP.003	Cisco Unified Presence	Co-located High availability client failover during manual fallback	Verify that the Cisco Unified Presence node failure, results in all clients and communication to failover automatically to another Cisco Unified Presence node. The clients fallback to the original configuration using a manual system fallback.	Cisco Unified Personal Communicator 1 -> Cisco Unified Presence Server 1; After failover: Cisco Unified Personal Communicator 1 -> Cisco Unified Presence Server 2	Passed	

ID	Feature teated	Case Title	Description	Call Component Flow	Status	Defects
UC851F. CUP.004	Cisco Unified Presence	CoW user rebalance	Verify that the user rebalance is successful when all the users are assigned from one node to another node in a CoW setup.	Cisco Unified Personal Communicator 1 -> Cisco Unified Presence Server 1; After failover: Cisco Unified Personal Communicator 1 -> Cisco Unified Presence Server 2	Passed	
UC851F. CUP.005	Cisco Unified Presence	Inter-cluster peer fail over	Verify that the inter-cluster functionality is successful when the server fails over to the backup due to inter-cluster peer node failure.		Passed	
UC851F. CUP.006	Cisco Unified Presence	Do Not Disturb with Shared line	Verify that the Do Not Disturb feature works successfully, when the secondary phone unregisters from Cisco Unified Communications Manager.		Passed	
UC851F. CUP.007	Cisco Unified Presence	Presence status in the Presence viewer page	Verify that the Presence Viewer displays the buddy presence status correctly, when the user ID in the contact list has spaces.		Passed	
UC851F. CUP.008	Cisco Unified Presence	Adding a new contact using the Unified Personal Communicator	Verify that a new contact is added through the Unified Personal Communicator. The added contact is not case sensitive.		Passed	

ID	Feature teated	Case Title	Description	Call Component Flow	Status	Defects
UC851F. CUP.009	Cisco Unified Presence	Unified Persence Server GUI display for a Cisco Unified Personal Communicator end user with a space in his user id	Verify that the end user ID with space is correctly displayed in the Unified Persence Server GUI.		Passed	
UC851F. CUP.010	Cisco Unified Presence	Status of SIP Publish model in the troubleshooter page.	Verify that the status of the SIP Publish model in the troubleshooter page is displayed without any error.		Passed	
UC851F. CUP.011	Cisco Unified Presence	Buddy list with mixed characters.	Verify that a buddy with mixed characters is added succesfully in the buddy list. The buddy status is also updated in the Unified Persence Server succesfully.		Passed	
UC851F. CUP.012	Cisco Unified Presence	Inter domain Federation	Verify that the user can enable or disable the Audible Alert Notification from an Cisco Unified IP Phone Manager or web GUI.		Passed	
UC851F. CUP.013	Cisco Unified Presence	Adding custom messages in the Cisco Unified Personal Communicator	Verify that custom messages are added successfully in the Cisco Unified Personal Communicator.		Passed	
UC851F. CUP.014	Cisco Unified Presence	Cisco Unified Presence authentication with the user having the same Directory number in different partitions	Verify that the Cisco Unified Presence authentication is successful for a user having the same Directory number in different partitions.		Passed	

ID	Feature teated	Case Title	Description	Call Component Flow	Status	Defects
UC851F.CUP.015	Cisco Unified Presence	Checkmark in Cisco Unified Presence or Unified Personal Communicator Licensed column	Verify that the Cisco Unified Presence or Cisco Unified Personal Communicator Licensed column is updated successfully.		Passed	
UC851F.CUP.016	Cisco Unified Presence	Cisco Unified Personal Communicator device versions	Verify that the device versions are correctly listed in the Cisco Unified Presence viewer.		Passed	
UC851F.CUP.017	Cisco Unified Presence	Connectivity to Unified Presence server when SIP proxy domain has space	Verify that the Cisco Unified Personal Communicator connects to the Unified Presence server successfully, when the SIP proxy domain has spaces.		Passed	
UC851F.CUP.018	Cisco Unified Presence	Instant Message	Verify that Instant Messages can be sent and received immediately without any failure or time delay.		Passed	
UC851F.CUP.019	Cisco Unified Presence	Outbound Call using the soft key	Verify that an outbound call is made successfully using the Dial soft key in Cisco Unified Personal Communicator.		Passed	

Cisco Unified Border Element

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.CUBE.U.001	Cisco Unified Border Element	Non-RSVP to RSVP and RSVP to non-RSVP call flows	Verify the non-RSVP to RSVP and RSVP to non-RSVP call flows with the Cisco Unified Border Element.	Cisco Unified Communications Manager1-> Unified SIP Proxy -> UnifiedBorder Element -> Cisco Unified Communications Manager Express 1; Cisco Unified Communications Manager Express 1 -> UnifiedBorder Element -> Cisco Unified Communications Manager Express 2	Passed	
UCJ851S.CUBE.U.002	Cisco Unified Border Element	Non-RSVP to RSVP and RSVP to non-RSVP call flows (DO-DO and DO-EO)	Verify the non-RSVP to RSVP and RSVP to non-RSVP call flows with the Cisco Unified Border Element (DO-DO and DO-EO).	Cisco Unified Communications Manager -> Unified SIP Proxy -> Cisco Unified Border Element -> Cisco Unified Communications Manager Express 1; Cisco Unified Communications Manager Express 1 -> Cisco Unified Border Element -> Cisco Unified Communications Manager Express 2	Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.CUBE.U.003	Cisco Unified Border Element	Non-RSVP to RSVP and RSVP to non-RSVP call flows (FS-EO)	Verify the non-RSVP to RSVP and RSVP to non-RSVP call flows with the Cisco Unified Border Element (FS-EO).	Cisco Unified Communications Manager 1 -> Unified SIP Proxy -> Cisco Unified Border Element -> Cisco Unified Communications Manager Express 1; Cisco Unified Communications Manager Express 1 -> Cisco Unified Border Element -> Cisco Unified Communications Manager Express 2	Passed	
UCJ851S.CUBE.U.004	Cisco Unified Border Element	Voice class codec transcoding and handling mid call codec changes	Verify the voice class codec transcoding of Cisco Unified Border Element.	Cisco Unified Communications Manager Express -> Cisco Unified Border Element -> Session Manager Edition	Passed	

Cisco Unified Communications Manager Express

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.CME.U.001	Cisco Unified Communications Manager Express	Direct Transfer	Verify that the Unified IP Phone 6900 Series can register with the Unified Communcations Manager Express and Direct Transfer across lines is supported.	Unified IP Phone 6900 Series -> Cisco Unified Communications Manager Express -> PSTN phone	Passed	
UCJ851S.CME.U.002	Cisco Unified Communications Manager Express	Authentication and encryption support	Verify that the communication and media between the Unified IP Phone 6900 Series, Unified IP Phone 9900 Seriess, and Cisco Unified Communications Manager Express is secure.	Unified IP Phone 6900 Series -> Cisco Unified Communications Manager Express -> Unified IP Phone 9900 Series	Passed	
UCJ851S.CME.U.003	Cisco Unified Communications Manager Express	Conference across lines	Verify that conference across lines can be performed from Unified IP Phone 9900 Series, which is registered to the Cisco Unified Communications Manager Express router support.	Unified IP Phone 6900 Series-> Cisco Unified Communications Manager Express -> Conference	Passed	
UCJ851S.CME.U.004	Cisco Unified Communications Manager Express	Shared lines	Verify that the Cisco Unified Communications Manager Express can support shared lines between RT and Unified IP Phone 6900 Series.	PSTN -> Cisco Unified Communications Manager Express -> RT and Unified IP Phone 6900 Series	Passed	
UCJ851S.CME.U.005	Cisco Unified Communications Manager Express	A VoIP call from a SRST to an Cisco Unified Communications Manager Express with no IP address trusted authentication	Verify that a VoIP call is placed from a SRST to an Cisco Unified Communications Manager Express with no IP address trusted authentication.	Cisco Unified IP Phone 3 -> SRST -> Cisco Unified Communications Manager Express –> Cisco Unified IP Phone 1, Cisco Unified IP Phone 2	Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.CME. U.006	Cisco Unified Communications Manager Express	A VoIP call from a SRST to an Cisco Unified Communications Manager Express, which has the Gateway configuration and no RAS dial peer	Verify that a VoIP call is placed from a SRST to an Cisco Unified Communications Manager Express, which has the Gateway configuration and no RAS dial peer.	Cisco Unified IP Phone 3 -> SRST -> Cisco Unified Communications Manager Express (remote) –> Cisco Unified IP Phone 1, Cisco Unified IP Phone 2	Passed	
UCJ851S.CME. U.007	Cisco Unified Communications Manager Express	A VoIP call from a SRST to an Cisco Unified Communications Manager Express, which has the Gateway configuration and no RAS dial peer, is blocked	Verify that a VoIP call placed from a SRST to an Cisco Unified Communications Manager Express, which has the Gateway configuration and no RAS dial peer, is blocked.	Cisco Unified IP Phone 3 -> SRST -> Cisco Unified Communications Manager Express (remote) – > Cisco Unified IP Phone 1, Cisco Unified IP Phone 2	Passed	
UCJ851S.CME. U.008	Cisco Unified Communications Manager Express	A VoIP call from a SRST to an Cisco Unified Communications Manager Express, which has no Gateway configuration and RAS dial peer.	Verify that a VoIP call is placed from a SRST to an Cisco Unified Communications Manager Express, which has no Gateway configuration and RAS dial peer.	Cisco Unified IP Phone 3 -> SRST -> Cisco Unified Communications Manager Express (remote) – >Cisco Unified IP Phone 1, Cisco Unified IP Phone 2	Passed	
UCJ851S.CME. U.009	Cisco Unified Communications Manager Express	A VoIP call from a SRST to an Cisco Unified Communications Manager Express, which has no Gateway configuration and the RAS dial peer, is blocked.	Verify that a VoIP call placed from a SRST to an Cisco Unified Communications Manager Express, which has no Gateway configuration and the RAS dial peer, is blocked.	Cisco Unified IP Phone 3 -> SRST -> Cisco Unified Communications Manager Express (remote) –> Cisco Unified IP Phone 1, Cisco Unified IP Phone 2	Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.CME.U.010	Cisco Unified Communications Manager Express	A VoIP call from a SRST to an Cisco Unified Communications Manager Express, which no Gateway configuration and no RAS dial peer	Verify that a VoIP call is placed from a SRST to an Cisco Unified Communications Manager Express which no Gateway configuration and no RAS dial peer.	Cisco Unified IP Phone 3 -> SRST -> Cisco Unified Communications Manager Express (remote) – >Cisco Unified IP Phone 1, Cisco Unified IP Phone 2	Passed	
UCJ851S.CME.U.011	Cisco Unified Communications Manager Express	A VoIP call from a SRST to an Cisco Unified Communications Manager Express, which has no Gateway configuration and no RAS dial peer, is blocked.	Verify that a VoIP call placed from a SRST to an Cisco Unified Communications Manager Express, which has no Gateway configuration and no RAS dial peer, is blocked.	Cisco Unified IP Phone 3 -> SRST -> Cisco Unified Communications Manager Express (remote) –>Cisco Unified IP Phone 1, Cisco Unified IP Phone 2-	Passed	
UCJ851S.CME.U.012	Cisco Unified Communications Manager Express	A VoIP call from an Cisco Unified Communications Manager through an Cisco Unified Communications Manager Gateway to a Cisco Unified Communications Manager Express with no IP address trusted authentication.	Verify that a VoIP call is placed from an Cisco Unified Communications Manager through an Cisco Unified Communications Manager Gateway to a Cisco Unified Communications Manager Express with no IP address trusted authentication.	Cisco Unified IP Phone 3 -> Cisco Unified Communications Manager -> Cisco Unified Communications Manager Gateway -> Cisco Unified Communications Manager Express (remote) -> Cisco Unified IP Phone 1, Cisco Unified IP Phone 2	Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.CME.U.013	Cisco Unified Communications Manager Express	A VoIP call from an Cisco Unified Communications Manager through an Cisco Unified Communications Manager Gateway to an Cisco Unified Communications Manager Express, which has the Gateway configuration and no RAS dial peer.	Verify that a VoIP call is placed from an Cisco Unified Communications Manager through an Cisco Unified Communications Manager Gateway to an Cisco Unified Communications Manager Express, which has the Gateway configuration and no RAS dial peer.	Cisco Unified IP Phone 3 -> Cisco Unified Communications Manager -> Cisco Unified Communications Manager Gateway -> Cisco Unified Communications Manager Express (remote) -> Cisco Unified IP Phone 1, Cisco Unified IP Phone 2	Passed	
UCJ851S.CME.U.014	Cisco Unified Communications Manager Express	A VoIP call from an Cisco Unified Communications Manager through an Cisco Unified Communications Manager Gateway to an Cisco Unified Communications Manager Express, which has the Gateway configuration and no RAS dial peer, is blocked.	Verify that a VoIP call placed from an Cisco Unified Communications Manager through an Cisco Unified Communications Manager Gateway to an Cisco Unified Communications Manager Express, which has the Gateway configuration and no RAS dial peer, is blocked.	Cisco Unified IP Phone 3 -> Cisco Unified Communications Manager -> Cisco Unified Communications Manager Gateway -> Cisco Unified Communications Manager Express (remote) -> Cisco Unified IP Phone 1, Cisco Unified IP Phone 2	Passed	
UCJ851S.CME.U.015	Cisco Unified Communications Manager Express	A VoIP call from an Cisco Unified Communications Manager through an Cisco Unified Communications Manager Gateway to an Cisco Unified Communications Manager Express with no Gateway configuration and RAS dial peer	Verify that a VoIP call is placed from an Cisco Unified Communications Manager through an Cisco Unified Communications Manager Gateway to an Cisco Unified Communications Manager Express, which has no Gateway configuration and RAS dial peer.	Cisco Unified IP Phone 3 -> Cisco Unified Communications Manager ->Cisco Unified Communications Manager gateway -> Cisco Unified Communications Manager Express (remote) -> Cisco Unified IP Phone 1, Cisco Unified IP Phone 2	Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.CME.U.016	Cisco Unified Communications Manager Express	Placing a call from Cisco Unified Communications Manager via Cisco Unified Communications Manager Gateway to Cisco Unified Communications Manager Express with blocked ip address trusted authentication	Verify that the VoIP call placed from Cisco Unified Communications Manager via Cisco Unified Communications Manager Gateway to a Cisco Unified Communications Manager Express, which has no Gateway configuration and RAS dial peer, is blocked.	Cisco Unified IP Phone 3 -> Cisco Unified Communications Manager ->Cisco Unified Communications Manager gateway -> Cisco Unified Communications Manager Express (remote) -> Cisco Unified IP Phone 1, Cisco Unified IP Phone 2	Passed	
UCJ851S.CME.U.017	Cisco Unified Communications Manager Express	A VoIP call from an Cisco Unified Communications Manager through an Cisco Unified Communications Manager Gateway to an Cisco Unified Communications Manager Express , which has no Gateway configuration and no RAS dial peer.	Verify that a VoIP call is placed from an Cisco Unified Communications Manager through an Cisco Unified Communications Manager Gateway to an Cisco Unified Communications Manager Express , which has no Gateway configuration and no RAS dial peer.	Cisco Unified IP Phone 3 -> Cisco Unified Communications Manager -> Cisco Unified Communications Manager gateway -> Cisco Unified Communications Manager Express (remote) -> Cisco Unified IP Phone 1, Cisco Unified IP Phone 7	Passed	
UCJ851S.CME.U.018	Cisco Unified Communications Manager Express	A VoIP call placed from an Cisco Unified Communications Manager through an Cisco Unified Communications Manager Gateway to a Cisco Unified Communications Manager Express, which has no Gateway configuration and no RAS dial peer, is blocked.	Verify that a VoIP call placed from Cisco Unified Communications Manager through an Cisco Unified Communications Manager Gateway to a Cisco Unified Communications Manager Express, which has no Gateway configuration and no RAS dial peer, is blocked.	Cisco Unified IP Phone 3 ->Cisco Unified Communications Manager -> Cisco Unified Communications Manager gateway -> Cisco Unified Communications Manager Express (remote) -> Cisco Unified IP Phone 1, Cisco Unified IP Phone 2	Passed	

Cisco Unified IP Phone

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.Phone.D.001	Cisco Unified IP Phone	IP address of the Cisco Unified IP Phone after changing the reservation address in the DHCP Server	Verify that the IP address of the Cisco Unified IP Phone is updated successfully after changing the reservation address in the DHCP Server.	Cisco Unified IP Phone1 -> Unified CallManager -> Cisco Unified IP Phone2	Passed	
UCJ851S.Phone.D.002	Cisco Unified IP Phone	Privacy lamp behavior for BLF SD in a Cisco Unified IP Phone	Verify that the Privacy lamp glows for the BLF SD in a Unified phone, when it is configured for a particular user.	Cisco Unified IP Phone1 -> Cisco Unified Communications Manager	Passed	
UCJ851S.SRST.U.005	Cisco Unified IP Phone	Call park number response time	Verify the response time of a call park number when the user answers the call immediately.	IP Phone1 - CUCM - IP Phone2 (Call Park to the number configured)	Passed	
UCJ851S.Phone.D.008	Cisco Unified IP Phone	Dial or Ring tone behavior when the Call Back feature is enabled	Verify that the Cisco Unified IP Phone plays the ring tone when the Call Back feature is enabled.	Cisco Unified IP Phone A -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B (Call Back)	Passed	
UCJ851S.Phone.D.009	Cisco Unified IP Phone	Behavior of the phone display name	Verify that the last character of the label field in the Cisco Unified IP Phone is displayed.		Passed	
UCJ851S.Phone.D.010	Cisco Unified IP Phone	Cisco Unified IP Phone load information	Verify that the Cisco Unified IP Phone load information is updated when the firmware changes from SCCP to SIP.		Passed	

Cisco Unified Personal Communicator

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC850F.CUPC.001	Cisco Unified Personal Communicator	Outbound call in softphone mode does not make the Cisco Unified Personal Communicator inaccessible	Verify that a two-way voice call is established, for an outbound call from an Cisco Unified Personal Communicator, which is in the softphone mode.		Passed	
UC850F.CUPC.002	Cisco Unified Personal Communicator	Pick a Parked call from an Cisco Unified Personal Communicator application (shared line).	Verify that a parked call from an Cisco Unified Personal Communicator is picked.		Passed	
UC850F.CUPC.003	Cisco Unified Personal Communicator	Placing a call from the contact list	Verify that an Cisco Unified Personal Communicator provides an option to place a call from the contact list.		Passed	
UC850F.CUPC.004	Cisco Unified Personal Communicator	Cisco Unified Personal Communicator Presence status	Verify that the changed status of the Cisco Unified Personal Communicator Presence is updated.		Passed	
UC850F.CUPC.005	Cisco Unified Personal Communicator in softphone mode	Transfer a video call to another Video phone	Verify that the caller ID of the called party is displayed successfully in the Video phone after a blind transfer of the video call.		Passed	
UC850F.CUPC.006	Cisco Unified Personal Communicator	Call to an external PSTN phone	Verify that an outbound call to an external PSTN phone through H.323 gateway is successful.		Passed	
UC850F.CUPC.007	Cisco Unified Personal Communicator	Calling party information displayed in corresponding locale	Verify that the Calling party information is displayed in the corresponding locale successfully.		Passed	
UC850F.CUPC.008	Cisco Unified Personal Communicator	Initiating Instant Messaging from Recent Communications module	Verify that all valid Cisco Unified Presence users are able to initiate Instant Messages from the Recent communications module successfully.		Passed	
UC850F.CUPC.009	Cisco Unified Personal Communicator	Cisco Unified Personal Communicator Login	Verify that the Cisco Unified Personal Communicator Login is successful.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC850F.CUPC.010	Cisco Unified Personal Communicator	Instant messaging	Verify that the user can send an Instant message successfully.		Passed	
UC850F.CUPC.011	Cisco Unified Personal Communicator	Pick up an inbound ringing conference call from a Cisco Unified Personal Communicator.	Verify that the Cisco Unified Personal Communicator can pick up a conference call.		Passed	
UC850F.CUPC.012	Cisco Unified Personal Communicator	Video call (soft phone mode)	Verify that the Cisco Unified Personal Communicator can make inbound and outbound video calls in Soft phone mode successfully.		Passed	
UC850F.CUPC.013	Cisco Unified Personal Communicator	Cisco Unified Personal Communicator login with a long username or domain name	Verify that the Cisco Unified Personal Communicator login with a long username or domain name is successful.		Passed	
UC850F.CUPC.014	Cisco Unified Personal Communicator	Video call (Desk phone mode)	Verify that the Cisco Unified Personal Communicator can make inbound and outbound video calls in Desk phone mode successfully.		Passed	
UC850F.CUPC.015	Cisco Unified Personal Communicator	Cisco Unified Personal Communicator Music On Hold	Verify that music is played successfully, when a call is put on hold in the Cisco Unified Personal Communicator if MOH is configured.		Passed	
UC850F.CUPC.016	Cisco Unified Personal Communicator	Cisco Unified Personal Communicator Registration	Verify that the Cisco Unified Personal Communicator Registration is successful when the network is restored.		Passed	
UC850F.CUPC.017	Cisco Unified Personal Communicator	Cisco Unified Personal Communicator Call termination	Verify that the Cisco Unified Personal Communicator Call is terminated when the Cisco Unified Personal Communicator is closed.		Passed	
UC850F.CUPC.018	Cisco Unified Personal Communicator	Cisco Unified Personal Communicator and DeskPhone	Verify that the Cisco Unified Personal Communicator and DeskPhone share the same or different Directory number.		Passed	
UC850F.CUPC.019	Cisco Unified Personal Communicator	DTMF signals	Verify that a DTMF signal is heared for a key press, during IVR instructions, in the Cisco Unified Personal Communicator.		Passed	

Cisco UC Integration™ for Microsoft Office Communicator

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ85F.CSF.U.001	UC Integration for Microsoft Office Communicator	The user calls a IP Communicator	Verify that the UC integration for Microsoft Office Communicator, is able to make a call to a DND enabled IP Communicator.	UC Integration for Microsoft Office Communicator 1 -> Unified Communications Manager -> Unified IP Communicator	Passed	
UCJ85F.CSF.U.002	UC Integration for Microsoft Office Communicator	The user calls a Cisco Unified Personal Communicator	Verify that the UC integration for Microsoft Office Communicator, is able to make a call to a DND enabled Cisco Unified Personal Communicator.	UC Integration for Microsoft Office Communicator 1 -> Cisco Unified Communications Manager -> Cisco Unified Personal Communicator	Passed	
UCJ85F.CSF.U.003	UC Integration for Microsoft Office Communicator	The user calls a DND enabled SCCP Phone	Verify that the UC integration for Microsoft Office Communicator, is able to make a call to a DND enabled SCCP Phone.	UC Integration for Microsoft Office Communicator 1 -> Cisco Unified Communications Manager -> SCCP Phone	Passed	
UCJ85F.CSF.U.004	UC Integration for Microsoft Office Communicator	Call transfer to an alternate device	Verify that the Transfer feature in the UC Integration for Microsoft Office Communicator works successfully and the call connectivity to the alternate device is successful.		Passed	
UCJ85F.CSF.U.005	UC Integration for Microsoft Office Communicator	Call park in softphone mode	Verify that the user can park an active call in the UC Integration for Microsoft Office Communicator, when it is in the softphone mode.		Passed	
UCJ85F.CSF.U.006	UC Integration for Microsoft Office Communicator	Call park in deskphone mode	Verify that the user can Park an active call in the UC Integration for Microsoft Office Communicator, when it is in thedeskphone mode.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ85F.CSF.U.007	UC Integration for Microsoft Office Communicator	Single Sign-on: UC Integration for Microsoft Office Communicator user with the LDAP setting "User must change password at next logon" enabled is able to reset the password and register with the UC application on login.	Verify that the UC Integration for Microsoft Office Communicator user is able to reset the password and change the password on next login. Once logged the user registers with all the UC applications.		Passed	
UCJ85F.CSF.U.008	UC Integration for Microsoft Office Communicator	Call from UC Integration for Microsoft Office Communicator to a SCCP Phone with the shared line configured	Verify that call connectivity is successful, while making a call from UC Integration for Microsoft Office Communicator to an SCCP phone, which is registered to the Cisco Unified Communications Manager and has the shared line configured.	UC Integration for Microsoft Office Communicator -> Cisco Unified Communications Manager -> SCCP Phone A, SCCP Phone B	Passed	

Cisco Unified Contact Center Express

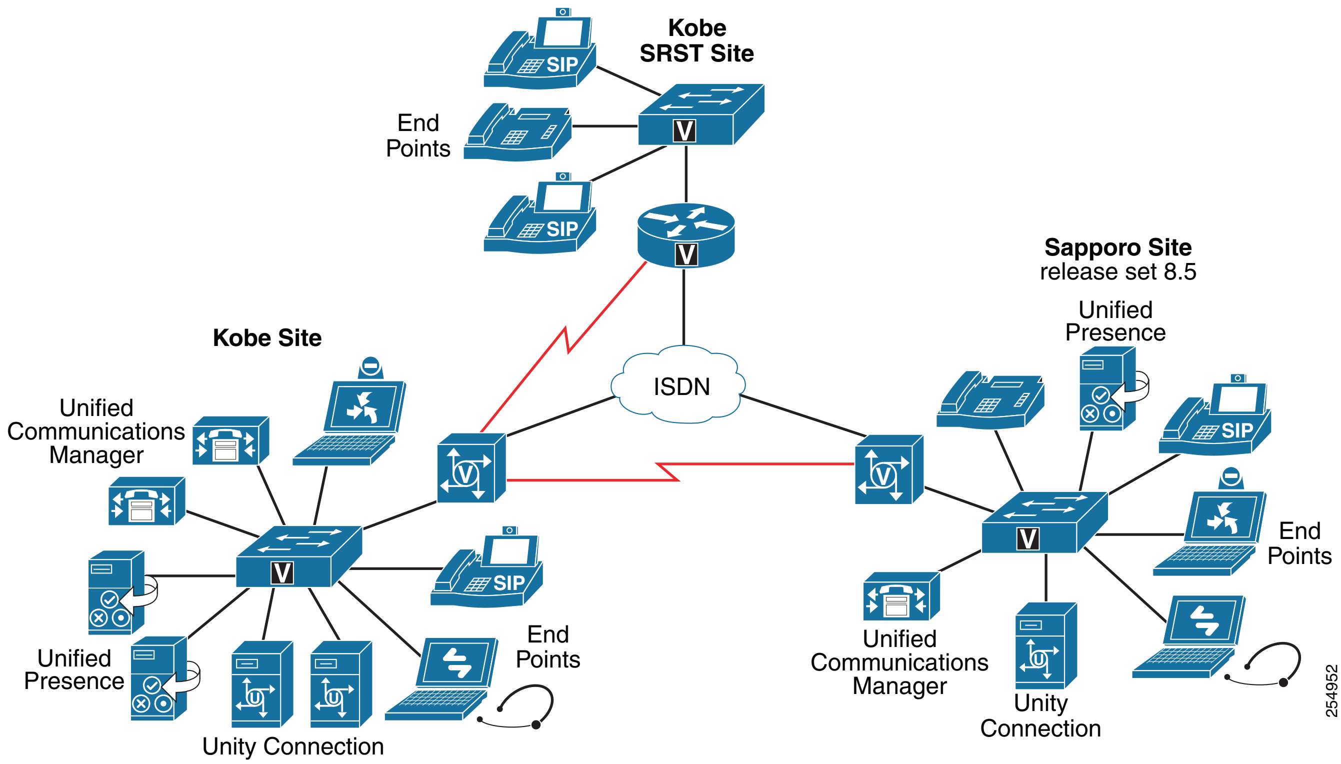
ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.ACD.001	UCCX	Agent-based routing	Verify that an Agent-based routing call is successful.	Cisco Unified IP Phone -> Gateway -> Cisco Unified Communications Manager -> Cisco Unified Contact Center Express -> CAD Agent	Passed	
UCJ851S.CAD.018	UCCX	Chat during an active call	Verify that an Agent can chat with the Supervisor during an active call.		Passed	
UCJ851S.HR.005	UCCX	Agent Login Logout Activity Report	Verify that the generation of Agent Login Logout Activity Report is successful.		Passed	
UCJ851S.CBE.006	UCCX	CAD-BE agent state verification	Verify the correctness of the agent state in CAD-BE.		Passed	
UCJ851S.CBE.009	UCCX	CAD-BE conference call between two different teams	Verify that the CAD-BE can make a conference call to a CAD-BE agent in a different team.	PSTN Phone -> Voice Gateway -> Cisco Unified Communications Manager -> Cisco Unified Contact Center Express -> CAD BE (conference with a different team)	Passed	
UCJ851S.CSD.001	UCCX	Intercept feature in CSD	Verify that the intercept feature in CSD is successful.	PSTN Phone -> Voice Gateway -> Cisco Unified Communications Manager -> Cisco Unified Contact Center Express -> CAD -> CSD	Passed	
UCJ851S.CSD.004	UCCX	Barge-in feature	Verify that the Barge-in feature is successful.	PSTN Phone -> Voice Gateway ->Cisco Unified Communications Manager -> Cisco Unified Contact Center Express -> CAD BE -> CSD	Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.ACD.005	UCCX	CED resource routing with a Unified IP Phone 6900 Series	Verify that the Caller entered digits (CED) routing is successful.	Cisco Unified IP Phone -> Gateway -> Unified Communciations Manager -> Cisco Unified Contact Center Express -> CAD Agent	Passed	
UCJ851S.CSD.007	UCCX	Not ready state from CSD to a CAD agent	Verify that the Not Ready state change from CSD to a CAD agent is successful.		Passed	
UCJ851S.HR.002	UCCX	Aborted Rejected Call Detail Report	Verify that the extraction of Aborted Rejected call report history is successful.		Passed	
UCJ851S.IPPA.003	UCCX	IPPA agent state verification from a Supervisor desktop	Verify that the IPPA agent state change viewed from a Supervisor desktop is successful.		Passed	
UCJ851S.OO.001	UCCX	Outbound option	Verify that outbound calls are working properly.	Cisco Unified Contact Center Express -> CAD -> Gateway -> PSTN Phone	Passed	
UCJ851S.OO.003	UCCX	Preview Outbound option in a centralized deployment	Verify that the working of two different campaigns in parallel is successful.	Cisco Unified Contact Center Express -> CAD -> PSTN Phone	Passed	
UCJ851S.CAD.001	UCCX	Team message	Verify that team messages are received by all agents.		Passed	
UCJ851S.HA.003	UCCX	UCCX High Availability basic call flow with the CAD Agent in a remote site	Verify that the Cisco Unified Contact Center Express HA feature for a basic CAD call flow that is with an agent in a remote site is successful.	PSTN Phone -> Voice Gateway -> Cisco Unified Communications Manager -> Cisco Unified Contact Center Express -> Voice Gateway -> CAD agent	Passed	
UCJ851S.CBE.001	UCCX	Logout provides alert for an agent during an on-going call using a CAD-BE.	Verify that the logout operation provides an alert for an agent during an on-going call using the CAD-BE.		Passed	
UCJ851S.SBR.002	UCCX	Skill based routing functionality when the agents are busy	Verify that the skill based routing functionality works successfully when the agents are busy.	PSTN Phone -> Voice Gateway -> Cisco Unified Communications Manager -> Cisco Unified Contact Center Express -> Agents	Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.CBE.003	UCCX	Transfer feature in CAD-BE	Verify that the transfer feature in CAD-BE works successfully.	PSTN Phone -> Voice Gateway -> Cisco Unified Communications Manager -> Cisco Unified Contact Center Express -> CAD BE -> (transfer)-> CAD BE	Passed	
UCJ851S.SPT.001	UCCX	Upload scripts through Application Wizard of UCCX	Verify that the scripts are uploaded successfully using the Application wizard in UCCX.		Failed	CSCtk62187
UCJ851S.CAD.019	UCCX	Agent state change after an unattended call	Verify that the agent can change his status to Ready after an unattended call.	PSTN Phone -> Voice Gateway -> Cisco Unified Communications Manager -> Cisco Unified Contact Center Express -> CAD agent	Failed	CSCtk67772

Upgrade

Figure 3-1 Upgrade Topology



Multi-Stage Upgrade

Upgrade 4.5 through 7.1(3) to 8.5(1)

Environment matrix of Upgrade 4.5

Product/Component	Base Release Set	Intermediate Release set	Target Release set
Cisco Unified Communications Manager	4.3(1) -> 4.3(2)	7.1(3)	8.5.1
BARS(MCS Backup System)	4.0.15	—	—
DMA(Data Migration Assistant)	7.1(3)	—	—
Cisco Unified Communications Manager Locale	cm-locale-ja_JP-4.3.1.3000 and cm-locale-combined_network-4.3.1.3000	cm-locale-ja_JP-7.1.3.2000-1.cop.sgn	cm-locale-ja_JP-8.5.1.9902-34.cop.sgn
Cisco Unity Connection	—	7.1(3)	8.5.1
Cisco Unity Connection Locale	—	uc-locale-ja_JP-7.1.2.0-139.cop.sgn	uc-locale-ja_JP-8.5.0.0-128.cop
Cisco Unified Survivable Remote Site Telephony	3.3	8.0	8.5
Cisco Unified Survivable Remote Site Telephony IOS	12.3(14) T	15.0(1)M XA	15.1.3 T
IOS (Voice gateways 2801)	12.4(13d)	15.0(1)M	15.1.3 T
Access Switch (3750)	12.2-35.SE5.bin	12.2(44)SE6	12.2(53)SE2
IP Communicator	2.0(1)	7.0(3)	7.0(3)

Test Results

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.UPGRADE.U.011	Upgrade	Upgrade Cisco Unified Communications Manager Publisher 4.5	Verify the successful upgrade of Cisco Unified Communications Manager 8.5(1)		Passed	
UCJ851S.UPGRADE.U.012	Upgrade	Upgrade Cisco Unified Communications Manager Subscriber 4.5	Verify the successful upgrade of Cisco Unified Communications Manager 8.5(1)		Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.UPGRADE.U.016	Upgrade	Upgrade to Cisco Unity Connection 7.1(3)	Verify the successful upgrade for release set 8.5(1) Cisco Unity Connection		Passed	
UCJ851S.UPGRADE.U.018	Upgrade	Upgrade SRST endpoints	Verify the successful upgrade of release set 8.5(1) SRST endpoints		Passed	

Upgrade 5.1 through 7.1(5) to 8.5(1)

Environment matrix of Upgrade 5.1

Product/Component	Base Release Set	Intermediate Release set	Target Release set
Cisco Unified Communications Manager	5.1(3)	7.1(5)	8.5.1
Cisco Unified Communications Manager Locale	cm-locale-ja_JP-5.1.1.2000-1.cop.sgn	cm-locale-ja_JP-7.1.3.2000-1.cop.sgn	cm-locale-ja_JP-8.5.1.9902-34.cop.sgn
Cisco Unity Connection	2.1(2)	7.1(5)	8.5.1
Cisco Unity Connection Locale	uc-locale-ja_JP-6.1.1.0-362.cop.sgn	uc-locale-ja_JP-7.1.2.0-139.cop.sgn	uc-locale-ja_JP-8.5.0.0-128.cop
Cisco Unified Survivable Remote Site Telephony	4.0(2)	8.0	8.5
Cisco Unified Survivable Remote Site Telephony IOS	12.4(11)T3	15.0(1)M XA	15.1.3 T
IOS (Voice gateways 2801)	12.4(15)T4	15.0(1)M	15.1.3T
Access Switch (3750)	12.2(35)SE5	12.2(44)SE6	12.2(53)SE2
IP Communicator	2.0(1)	7.0(3)	7.0(3)

Test Results

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.UPGRADE.U.46	Upgrade	Upgrade Cisco Unified Communications Manager Publisher 5.x	Verify the successful upgrade of Cisco Unified Communications Manager 8.5(1)		Passed	
UCJ851S.UPGRADE.U.47	Upgrade	Upgrade Cisco Unified Communications Manager Subscribers 5.x	Verify the successful upgrade of Cisco Unified Communications Manager 8.5(1)		Passed	
UCJ851S.UPGRADE.U.53	Upgrade	Upgrade to Cisco Unity Connection 2.1(2) primary	Verify the successful upgrade for release set 8.5(1) Cisco Unity Connection		Passed	
UCJ851S.UPGRADE.U.54	Upgrade	Upgrade to Cisco Unity Connection 2.1(2) Secondary	Verify the successful upgrade for release set 8.5(1) Cisco Unity Connection		Passed	
UCJ851S.UPGRADE.U.57	Upgrade	Upgrade SRST endpoints	Verify the successful upgrade of Release set 8.5(1) SRST endpoints.		Passed	

Single-Stage Upgrade

Upgrade from 6.1.(5) to 8.5(1)

Environment matrix of Upgrade 6.1

Product/Component	Base Release Set	Target Release set
Cisco Unified Communications Manager	6.1(5)	8.5.1
Cisco Unified Communications Manager Locale	cm-locale-ja_JP-6.1.3.9902-1142.cop.sgn	cm-locale-ja_JP-8.5.1.9902-34.cop.sgn
Cisco Unity Connection	2.1(2)	8.5.1
Cisco Unity Connection Locale	uc-locale-ja_JP-6.1.1.0-362.cop.sgn	uc-locale-ja_JP-8.5.0.0-128.cop
Cisco Unified Survivable Remote Site Telephony	4.1	8.5
Cisco Unified Survivable Remote Site Telephony IOS	12.4(15)T4	15.1.3T
IOS (Voice gateways 2801)	12.4(15)T4	15.1.3T
Access Switch (3750)	12.2(35)SE5	12.2(53)SE2
IP Communicator	2.1	7.0(3)

Test Results

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.UPGRADE.U.002	Upgrade	Upgrade Cisco Unified Communications Manager Publisher 6.x	Verify the successful upgrade of Cisco Unified Communications Manager 8.5(1)		Passed	
UCJ851S.UPGRADE.U.003	Upgrade	Upgrade Cisco Unified Communications Manager Subscribers 6.x	Verify the successful upgrade of Cisco Unified Communications Manager 8.5(1)		Passed	
UCJ851S.UPGRADE.U.018	Upgrade	Upgrade Cisco Unity Connection 2.1(2) primary	Verify the successful upgrade for Cisco Unity Connection 8.5(1)		Passed	
UCJ851S.UPGRADE.U.019	Upgrade	Upgrade Cisco Unity Connection 2.1(2) Secondary	Verify the successful upgrade for Cisco Unity Connection 8.5(1)		Passed	
UCJ851S.UPGRADE.U.022	Upgrade	Upgrade SRST endpoints	Verify the successful upgrade of release set 8.5(1) SRST endpoints		Passed	

Upgrade from 7.1(3) to 8.5(1)

Environment matrix of Upgrade 7.1

Product/Component	Base Release Set	Target Release set
Cisco Unified Communications Manager	7.1(3)	8.5.1
Cisco Unified Communications Manager Locale	cm-locale-ja_JP-7.1.3.2000-1.cop.sgn	cm-locale-ja_JP-8.5.1.9902-34.cop.sgn
Cisco Unity Connection	7.1(3)	8.5.1
Cisco Unity Connection Locale	uc-locale-ja_JP-7.1.2.0-139.cop.sgn	uc-locale-ja_JP-8.5.0.0-128.cop
Cisco Unified Presence	7.0(5)	8.0
Cisco Unified Presence Locale	ps-locale-ja_JP-7.0.4.1000-1.cop.sgn	ps-locale-ja_JP-8.5.1.9902-138.cop.sgn
Cisco Unified Survivable Remote Site Telephony	8.0	8.5
Cisco Unified Survivable Remote Site Telephony IOS	15.0(1)M XA	15.1.3T
IOS (Voice gateways 2801)	15.0(1)M	15.1.3T
Access Switch (3750)	12.2(44)SE6	12.2(53)SE2
IP Communicator	7.0(3)	7.0(3)
Cisco Unified Personal Communicator	7.0(2)	8

Test Results

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.UPGRADE.U.024	Upgrade	Upgrade Cisco Unified Communications Manager Publisher 7.x	Verify the successful upgrade of Cisco Unified Communications Manager 8.5(1)		Passed	
UCJ851S.UPGRADE.U.025	Upgrade	Upgrade Cisco Unified Communications Manager Subscribers 7.x	Verify the successful upgrade of Cisco Unified Communications Manager 8.5(1)		Passed	
UCJ851S.UPGRADE.U.028	Upgrade	Upgrade Cisco Unified Presence 7.x	Verify the successful upgrade of Cisco Unified Presence 8.5(1)		Passed	
UCJ851S.UPGRADE.U.031	Upgrade	Upgrade Cisco Unity Connection 7.1(3) primary	Verify the successful upgrade of Cisco Unity Connection 8.5(1)		Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ851S.UPGRADE.U.032	Upgrade	Upgrade Cisco Unity Connection 7.1(3) Secondary	Verify the successful upgrade for Cisco Unity Connection 8.5(1)		Passed	
UCJ851S.UPGRADE.U.035	Upgrade	Upgrade SRST endpoints	Verify the successful upgrade of release set 8.5(1) SRST endpoints		Passed	

Regression Testing

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ713F.CUCM.D.001	Cisco Unified Communications Manager	Call routing in newly added gateways (members) in a Route Group	Verify that the communication and media between the Unified IP Phone 6900 Series, Unified IP Phone 9900 Series and Cisco Unified Communications Manager Express is secure.		Passed	
UCJ713F.CUCM.D.002	Cisco Unified Communications Manager	CFNA in a shared line	Verify that CFNA in a shared line works successfully.		Passed	
UCJ713F.CUCM.D.003	Cisco Unified Communications Manager	Privacy feature in the Shared Line	Verify that the Privacy feature in the shared line works successfully.		Passed	
UCJ713F.CUCM.D.004	Cisco Unified Communications Manager	Call Pickup in different partitions	Verify that Call Pickup works successfully, when the Call Pickup groups are in different partition but in the same calling search space.	Cisco Unified IP Phone -> Cisco Unified Communications Manager -> Cisco Unified IP Phone -> Call Pick up	Passed	
UCJ713F.CUCM.D.005	Cisco Unified Communications Manager	cBarge after call hold and resume	Verify that cBarge feature after call hold and resume works successfully.	Cisco Unified IP Phone -> Unified Communications Manager -> Cisco Unified IP Phones -> cBarge	Passed	
UCJ713F.CUCM.D.006	Directed Call Park	Directed call park to a remote phone	Verify that Directed Call Park feature works successfully between a remote phone and main site phone. After timeout the call is retrieved with the Reversion number configured in the partition.	Remote Cisco Unified IP Phone -> Cisco Unified Communications Manager -> Cisco Unified IP Phone -> Directed Call Park	Passed	
UCJ713F.CUCM.D.007	Cisco Unified Communications Manager	Privacy settings in shared lines	Verify that "privacy on hold toggling" setting in the shared line works successfully.		Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ713F.CUCM.D.008	Cisco Unified Communications Manager	Auto Pickup feature works with the Privacy feature and Privacy on hold in a shared line	Verify that the Auto Pickup feature works successfully with the Privacy feature and Privacy on hold in a shared line.		Passed	
UCJ713F.CUCM.D.009	Cisco Unified Communications Manager	Auto Pickup feature in a shared line	Verify that the Auto Pickup feature works successfully in a shared line.		Passed	
UCJ713F.CUCM.D.010	Cisco Unified Communications Manager	Call park with service parameters configured	Verify that the call park with the following service parameters is configured successfully: • Display timer is 0 • Caller ID display priority enabled is set to True		Passed	
UCJ713F.CUCM.D.011	Directed Call Pickup	Directed Call PickUp with Group PickUp in a different partition	Verify that Directed Call Pick up works successfully, when the PickUp Groups are configured with a partition and CSS.		Passed	
UCJ713F.CUCM.D.012	Directed Call Pickup	Directed Call Pickup when multiple calls are available for pickup	Verify that Directed Call Pickup works successfully, when multiple calls are available for pickup.		Passed	
UCJ713F.CUCM.D.013	MOH	MOH for call park and blind transfer	Verify that MOH is played during a Call Park or Blind Transfer. Verify that the appropriate status message appears in the Cisco Unified IP Phone.	Cisco Unified IP Phone -> Cisco Unified Communications Manager -> Cisco Unified IP Phone -> Call Park and Blind Transfer	Passed	
UCJ713F.CUCM.D.014	Cisco Unified Communications Manager	Ring back tone in Cisco Unified IP Phones during a call transfer.	Verify that ring back tone is heard during a Call Transfer successfully.		Passed	
UCJ713F.CUCM.D.015	Cisco Unified Communications Manager	Blind Transfer across SIP trunks	Verify Blind transfer across SIP trunks works successfully.		Passed	
UCJ713F.CUCM.D.016	Cisco Unified Communications Manager	Busy Tone with SIP Phone	Verify that the busy tone is heard for a SIP phone, when the called party is busy.		Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ713F.CUCM.D.017	Cisco Unified Communications Manager	Mutual Hold and Resume feature	Verify that the mutual Hold and Resume feature works successfully during a conference.		Passed	
UCJ713F.CUCM.D.018	Cisco Unified Communications Manager	Multiple SIP phones with shared line	Verify that multiple SIP phones with a shared line works successfully.		Passed	
UCJ713F.CUCM.D.019	Cisco Unified Communications Manager	Adhoc Inter-Cluster Conference with SIP Phones	Verify that a six-party ad hoc inter-cluster conference is established using SIP phones and the conference is still active after the originator is dropped from the conference.		Passed	
UCJ713F.CUCM.D.020	Cisco Unified Communications Manager	Long duration call between SIP Phones	Verify that a 30-minute long duration call is maintained successfully between SIP phones.		Passed	
UCJ713F.CUCM.D.021	Cisco Unified Communications Manager	Call Pickup Priority	Verify that the Call Pickup feature works successfully based on the priority set in the "OtherGroup" list, for inbound calls from mulitple groups.		Passed	
UCJ713F.CUCM.D.022	Cisco Unified Communications Manager	One touch call pickup	Verify that one touch Call Pickup using a softkey works successfully.		Passed	
UCJ713F.CUCM.D.023	Cisco Unified Communications Manager	Group Pickup	Verify that the one touch Group Pickup using a softkey works successfully.		Passed	
UCJ713F.CUCM.D.0024	Cisco Unified Communications Manager	SIP-to-SIP intra-cluster call in a shared line	Verify that the directory number is displayed for a SIP-to-SIP intra-cluster call, when a SIP call is placed to a shared line directory number.		Passed	
UCJ713F.CUCM.D.025	Cisco Unified Communications Manager	Call Park for intra-cluster calls between encrypted SIP phones	Verify that the Call Park feature works successfully for intra-cluster calls between encrypted SIP phones.		Passed	

Related Documentation

Cisco Unified Communications Manager Documentation Guide

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/docguide/7_1_3/dg713.html

http://www.cisco.com/en/US/partner/docs/voice_ip_comm/cucm/srnd/8x/uc8x.html

http://www.cisco.com/en/US/partner/docs/voice_ip_comm/cucm/docguide/8_5_1/dg851.html

Cisco Unified Communications System Documentation

http://www.cisco.com/en/US/docs/voice_ip_comm/uc_system/unified/communications/system/ucstart.htm

Cisco Unified Communications System Description

http://www.cisco.com/en/US/docs/voice_ip_comm/uc_system/UC7.1.3/system_description/SD713.pdf

SAF Configuration Guide

http://www.cisco.com/go/saf

SME Guide

http://www.cisco.com/en/US/partner/products/ps10661/index.html

CHAPTER 4

System Test Results for IP Telephony: Cisco Unified Communications System Release 8.5(1)

This section lists the various features, the test cases under each feature and the test results. The following features are tested:

- Cisco Emergency Responder
- Cisco IME
- Codec Protocols
- DPNSS Conversion
- Gateways
- IP Communicator
- QSIG
- Quality of Service
- Reliability, Load
- RSVP
- Service Advertisement Framework
- Session Management Edition
- UC Integration
- Unified Border Element
- Unified CM Business Edition
- Unified CM Express
- Unified Communications Manager
- Unified Contact Center Express
- Unified MeetingPlace
- Unified Mobility
- Unified PC
- Unified Presence
- Unified SIP Proxy
- Unified SRST
- Unity

- Unity Connection
- Unity Express
- Video Telephony
- Regression Tests

Cisco Emergency Responder

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR61.CER.102	Cisco Emergency Responder	E911 Call Handling When Both Cisco Emergency Responder and SRST are Capable of Routing E911 Calls to PSAP	Verify that Cisco Emergency Responder (Cisco Emergency Responder) can route E911 calls from a branch Phone when both Cisco Emergency Responder and Unified SRST routers are configured to route the E911 call to a local Public Safety Answering Point (PSAP).		Passed	
SR61.CER.102	Cisco Emergency Responder	E911 Call Handling When Both Cisco Emergency Responder and SRST are Capable of Routing E911 Calls to PSAP	Verify that Cisco Emergency Responder can route E911 calls from a branch Phone when both Cisco Emergency Responder and Unified SRST routers are configured to route the E911 call to a local Public Safety Answering Point (PSAP).		Passed	
UC700IF.CER.101	Cisco Emergency Responder	Link Layer Discovery Protocol - Media Endpoint Discovery (LLDP-MED) and CDP-Enabled TNP Phones Dialing E911 Call	Verify that Cisco Emergency Responder (Cisco Emergency Responder) can track the location of TNP phones enabled with both Link Layer Discovery Protocol - Media Endpoint Discovery (LLDP-MED) and Cisco Discovery Protocol (CDP) and those 911 calls from these phones are routed to the local Public Safety Answering Point (PSAP).		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC700IF.CER.101	Cisco Emergency Responder	LLDP-MED and CDP-Enabled TNP Phones Dialing E911 Call	Verify that Cisco Emergency Responder can track the location of TNP phones enabled with both Link Layer Discovery Protocol - Media Endpoint Discovery (LLDP-MED) and Cisco Discovery Protocol (CDP) and those 911 calls from these phones are routed to the local Public Safety Answering Point (PSAP).		Passed	
UC802IF.CER.101	Cisco Emergency Responder	JTAPI Over WAN	Verify the validation of Cisco Emergency Responder in Clustering over WAN (CoW) deployment such as the Cisco Emergency Responder node registering CTI ports and route points to Unified Communications Manager over WAN.		Passed	
UC802IF.CER.101	Cisco Emergency Responder	JTAPI Over WAN	Verify the validation of Cisco Emergency Responder in CoW deployment such as the Cisco Emergency Responder node registering CTI ports and route points to Unified Communications Manager over WAN.		Passed	

Cisco IME

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF. IME.002	Cisco IME	Cisco IME Audio call from Cluster with Inline Adaptive Security Appliance (ASA) to Another Cluster with Off-Path ASA	Verify a Cisco IME audio call from cluster with inline ASA to another cluster with off-path ASA.	Originating Phone->IME->Cisco Unified Communications Manager->Off-path ASA->SIP Trunk->Off-path ASA->Cisco Unified Communications Manager->IME->Terminating Phone	Passed	
UC802EF. IME.005	Cisco IME	Cisco IME Video Call from Cluster with Inline ASA to Another Cluster with Off-Path ASA	Verify a Cisco IME video call from cluster with inline ASA to another cluster with off-path ASA.	Originating Video Phone->IME-> Cisco Unified Communications Manager->Inline ASA->SIP Trunk->Off-path ASA-> Cisco Unified Communications Manager->IME->Terminating Video Phone	Passed	
UC802EF. IME.009	Cisco IME	Cisco IME Call with Audio Escalation to Video	Verify Cisco IME call with audio escalation to video.	Originating Phone->IME-> Cisco Unified Communications Manager->SIP Trunk-> Cisco Unified Communications Manager->IME->Terminating Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF. IME.010	Cisco IME	Cisco IME Call with Hold/Resume	Verify Cisco IME call interaction with Hold/Resume.	Originating Phone->IME-> Cisco Unified Communications Manager->SIP Trunk-> Cisco Unified Communications Manager->IME->Terminating Phone (Hold/Resume)	Passed	
UC802EF. IME.011	Cisco IME	Cisco IME Call Transfer	Verify Cisco IME call interaction with call transfers.	Originating Phone->IME-> Cisco Unified Communications Manager->SIP Trunk-> Cisco Unified Communications Manager->IME->Terminating Phone->Transfer ->Destination Phone	Passed	
UC802EF. IME.012	Cisco IME	Cisco IME Call Conference	Verify Cisco IME call interaction with conference calls.	Originating Phone->IME-> Cisco Unified Communications Manager->SIP Trunk-> Cisco Unified Communications Manager->IME->Terminating Phone->Conf-> Destination Phone	Passed	
UC802EF. IME.018	Cisco IME	Cisco IME Call gets Transferred to Unified CME Site Aggregated by Unified SIP Proxy	Verify if a Cisco IME call can be transferred to a Unified CME site aggregated by Unified SIP Proxy.	Originating Phone->IME->Unified CM->SIP Trunk->Unified CM->IME->Terminating Phone->Xfer->SIP Trunk->Cisco Unified SIP Proxy->Unified CME->Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF. IME.019	Cisco IME	Cisco IME Call get Transferred to Unified Communications Manager-Interoperability Site	Verify if a Cisco IME call can be transferred to a phone in interoperability-site with Unified Communications Manager 7.x version.	Originating Phone->IME->Unified CM->SIP Trunk->Unified CM->IME->Terminating Phone->Xfer->QSIG ICT->Unified CM 7.x->Phone	Passed	
UC802EF. IME.020	Cisco IME	Cisco IME Call Transferred to PSTN Phone	Verify if a Cisco IME call can be transferred to a PSTN phone.	Originating Phone->IME->Unified CM->SIP Trunk->Unified CM->IME->Terminating Phone->Xfer->Unified CM->PSTN Gateway->PSTN Phone	Passed	
UC802EF. IME.021	Cisco IME	Cisco IME Call Transferred to PBX Phone	Verify if a Cisco IME call can be transferred to a PBX phone.	Originating Phone->IME->Unified CM->SIP Trunk->Unified CM->IME->Terminating Phone->Xfer->Unified CM->QSIG PBX->PBX Phone	Passed	
UC802IF.IME.203	Cisco Intercompany Media Enigine (IME)	Video Call Over Unified B2B Trunk From TRP Enabled Endpoint	Verify that video calls are supported over IME trunks for a TRP enabled endpoint.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.IME.207	Cisco Intercompany Media Enigine (IME)	PSTN Fallback With Unified B2B	Verify the fallback to PSTN feature of Unified B2B when the quality of the voice call deteriorates. Also verify the fallback under various sensitivity settings for different codec's using different call quality condition.		Passed	
UC802IF.IME.220	Cisco IME	Unified B2B Calls to Phone With Mobile Connect Enabled	Verify that Mobile Connect feature works with Unified B2B.		Passed	
UC851EF.IME.001	Cisco IME	Cisco IME Call Between Unified Communications Manager Leaf Clusters through Unified Session Manager Edition	Verify a Cisco IME call between Unified Communications Manager leaf clusters through Unified Session Management Edition.	IP Phone->Unified CM 1 (Leaf)->SIP ICT (QSIG)->SME->ASA->IME Trunk->ASA->Unified CM 2 (Leaf)->IP Phone	Passed	
UC851EF.IME.002	Cisco IME	Cisco IME Video Call Between Unified Communications Manager Leaf Clusters via Unified Session Manager Edition	Verify Cisco IME video call between Unified Communications Manager leaf clusters through Unified Session Management Edition.	Video IP Phone->Unified CM 1 (Leaf)->SIP ICT (QSIG)->SME->ASA->IME Trunk->ASA->Unified CM 2 (Leaf)->Video IP Phone	Passed	
UC851EF.IME.003	Cisco IME	Call Forwards: Cisco IME Call Between Unified Communications Manager leaf clusters via Unified Session Manager Edition	Verify the behavior of call forwards in Cisco IME calls between Unified Communications Manager leaf clusters via Unified Session Management Edition.	IP Phone->Unified CM 1 (Leaf)->SIP ICT (QSIG)->SME->ASA->IME Trunk->ASA->Unified CM 2 (Leaf)->IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF. IME.004	Cisco IME	Call Transfers: Cisco IME Call Between Unified Communications Manager Leaf Clusters via Unified Session Manager Edition	Verify the behavior of call transfers in Cisco IME calls between Unified Communications Manager leaf clusters via Unified Session Management Edition.	IP Phone->Unified CM 1 (Leaf)->SIP ICT (QSIG)->SME->ASA->IME Trunk->ASA->Unified CM 2 (Leaf)->IP Phone	Passed	
UC851EF. IME.005	Cisco IME	Call Conference: Cisco IME Calls Between Unified Communications Manager Leaf Clusters through Unified Session Manager Edition	Verify the behavior of call conference in Cisco IME calls between Unified Communications Manager leaf clusters through Unified Session Management Edition.	IP Phone->Unified CM 1 (Leaf)->SIP ICT (QSIG)->SME->ASA->IME Trunk->ASA->Unified CM 2 (Leaf)->IP Phone	Passed	
UC851EF. IME.006	Cisco IME	Shared Lines: Cisco IME Call Between Unified Communications Manager Leaf Clusters through Unified Session Manager Edition	Verify the behavior of shared lines in Cisco IME calls between Unified Communications Manager leaf clusters through Unified Session Management Edition.	IP Phone (Shared line)->Unified CM 1 (Leaf)->SIP ICT (QSIG)->SME->ASA->IME Trunk->ASA->Unified CM 2 (Leaf)->IP Phone	Passed	
UC851EF. IME.007	Cisco IME	Cisco IME Negative Scenarios	Verify the behavior of Cisco IME calls failing back to PSTN when WAN links are degraded/congested.	IP Phone->Unified CM 1->ASA->IME Trunk->ASA->Unified CM 2 ->IP Phone; Call fall backs to PSTN after degradation.	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.IME.008	Cisco IME	Cisco IME and SAF Interoperability Scenario	"Verify the Cisco IME and SAF interoperability scenario on making a call from a Unified CM cluster1 phone to Unified CM cluster2 through Unified Session Management Edition cluster via SAF enabled SIP trunk and verify if the Cisco IME route is created in Session Management Edition cluster. Verify if a CDR record is generated on making a second call from cluster1 to cluster2 over IME.	"IP Phone->Unified CM 1 (Leaf)->SAF enabled Trunk (SIP ICT - QSIG)->SME->ASA->IME Trunk->ASA->Unified CM 2 (Leaf)->IP Phone	Passed	
UC851EF.IME.009	Cisco IME	IME & SAF interworking on Cisco Unified Session Manager Edition	Verify IME & SAF interworking on Cisco Unified Session Manager edition.	IP Phone->Unified CM 1->SIP ICT (SAF)->SME Unified CM->IME Trunk->Unified CM 2->IP Phone	Passed	
UC851EF.IME.010	Cisco IME	IME Call Flow from Cluster with Inline ASA to Cluster with Offpath ASA	Verify IME call flow from cluster with inline ASA to cluster with offpath ASA.	IP Phone->Unified CM 1->SIP ICT->SME Unified CM->Inline ASA->IME Trunk->Offpath ASA->Unified CM 2->IP Phone	Passed	

Codec Protocols

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC85IIR. CDC.501	Codec Protocols	iLBC Codec for Unified IP Phone 69xx series: Hold and Resume	Verify that during a call between Phone A and Phone B, Hold/Resume on either Phone A or Phone B is succesful three times.		Passed	
UC85IIR. CDC.502	Codec Protocols	iLBC Codec for Unified IP Phone 6921: Calls between Two Endpoints	Verify that calls between two endpoints are successful. Phone A calls Phone B and Phone B calls Phone A.		Passed	
UC85IIR. CDC.503	Codec Protocols	iLBC Codec for Unified IP Phone 6941: Calls between Two Endpoints	Verify that calls between two endpoints are successful. Phone A calls Phone B and Phone B calls Phone A.		Passed	
UC85IIR. CDC.601	Codec Protocols	iLBC Codec for Unified IP Phone 6941 series: Multiple Calls on Multiple Lines on Unified IP Phone	Verify that multiple calls on multiple lines on a Unified IP Phone are successful. EpAsstnt1 makes 1st call to EpFeature. EpAsstnt2 makes 2nd call to EpFeature on a different line. EpFeature can switch the 1st call and 2nd call correctly.		Passed	
UC85IIR. CDC.602	Codec Protocols	iLBC Codec for Unified IP Phone 6961 series: Multiple Calls on Multiple Lines on Unified IP Phone	Verify that multiple calls on multiple lines on a Unified IP Phone are successful. EpAsstnt1 makes 1st call to EpFeature. EpAsstnt2 makes 2nd call to EpFeature on a different line. EpFeature can switch the 1st call and 2nd call correctly.		Passed	
UC85IIR. CDC.603	Codec Protocols	iLBC Codec for Unified IP Phone 69xx series: Call Forward All	Verify call forward all feature. EpAsstnt1 calls EpFeature and EpFeature forwards all calls to EpAsstnt2		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC85IIR. CDC.604	Codec Protocols	iLBC Codec for Unified IP Phone 69xx series: Call Forward Busy	Verify call forward busy feature. EpAsstnt1 calls EpFeature. When the EpFeature busy, the call is forwarded to EpAsstnt2.		Passed	
UC85IIR. CDC.605	Codec Protocols	iLBC Codec for Unified IP Phone 69xx series: Call Forward No Answer	Verify call forward no answer feature. EpAsstnt1 calls EpFeature. EpFeature rings, but does not answer. The call is forwarded to EpAsstnt2.		Passed	
UC85IIR. CDC.606	Codec Protocols	iLBC Codec for Unified IP Phone 69xx series: Same Group <normal> Pickup by Pickup Softkey	Verify same group <normal> pickup by Pickup softkey.Setup EpFeature and EpAsstnt2 in the same pickup group and Unified Communications Manager service parameter <Auto Call Pickup Enabled> as false. EpAsstnt1 calls EpAsstnt2 . While EpAsstnt2 is ringing, EpFeature pushes PickUp softkey. After ringing stops, EpFeature goes off-hook to connect the call.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC85IIR. CDC.607	Codec Protocols	iLBC Codec for Unified IP Phone 69xx series: Group Pick Up by GPickUp Softkey	Veify group pickup by GPickUp softkey. Setup EpFeature and EpAsstnt2 in the different pickup group and <Auto Call Pickup Enabled> as false in Unified Communications Manager. EpAsstnt1 calls EpAsstnt2 . While EpAsstnt2 is ringing, EpFeature pushes GPickUp softkey and dials the pick code. After ringing stops, EpFeature goes off hook and connects the call.		Passed	
UC85IIR. CDC.608	Codec Protocols	iLBC Codec for Unified IP Phone 69xx Series: Group Pick Up by OPickUp Softkey.	Verify <normal> other group pickup by OPickUp softkey. Setup EpAsstnt2 in an associate pickup group with EpFeature and Unified Communications Manager service parameter <Auto Call Pickup Enabled> as false. EpAsstnt1 calls EpAsstnt2. During EpAsstnt2 ringing, EpFeature pushes OPickUp softkey. After the ringing, EpFeature goes off-hook to connect the call.		Passed	

DPNSS Conversion

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802.DPN.001	DPNSS interworking	Call Back in RDX Extension	Verify if RDX extension can invoke call back when free against IP-Phone in Legacy Unified Communications Manager Cluster.	PBX Ph1->PBX->Westell->Unified CM->ICT(QSIG)->Unified CM->SCCP/SIP Ph1	Passed	
UC802.DPN.003	DPNSS interworking	Call Forward with DPNSS Phone	Verify if RDX extension can initiate a call which is forwarded by SCCP Phone, and transfers to Unified CME and Unified IP Phone on Unified CME back to the originating RDX extension.	PBX Ph1->PBX->Westell->Unified CM->CDG Central SCCP Ph1->Transfer->Unified CME->SCCP Ph1->CFA->Unified CM->Westell->PBX->PBX Ph1	Passed	
UC802.DPN.005	DPNSS interworking	Call Forward in iSDX Extension	Verify if iSDX extension can initiate a call which is forwarded by SIP Phone in CDG Remote, SIP Phone on Cisco SIP Proxy Server (CSPS) and M1 extension back to the originating iSDX extension.	PBX Ph1->PBX->Westell->Unified CM->CDG Remote SIP Ph1->iDivert->SIP Trunk->CSPS->SIP Ph1->Westell->PBX->PBX Ph2->Unified CM->Westell->PBX->PBX Ph1	Passed	

Gateways

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. GTW.001	Analog Gateway - Security	Secure Cisco IPS Manager Express (IME) Phone Call to Analog Phone's Remote Destination, Park Call back to Analog Phone	Verifies the ability to place an IME call from one secure analog phone to another in a different cluster that has an SNR mobile device configured. Verifies the ability to answer the call from the mobile device, then dial *74 to park the call back to the analog phone, and accept the call at the analog phone.	Secure analog ph->Secure GW->ABI Unified CM->Offpath ASA->Secure IME Trunk->Inline ASA->MSP Unified CM->(Secure PSTN GW->Mobile phone) Xfer to (Secure GW->Secure analog phone)	Passed	
UC851IF. GTW.002	Analog Gateway - Security	Non-secure IME Phone Call, Transfer to Non-secure Endpoint	Verifies the ability to place an IME call from a secure analog phone to a non-secure Unified IP Phone in another cluster, verify audio, hold/resume the call, and transfer the call from the Unified IP Phone to another non-secure Unified IP Phone.	Secure analog phone (Xfer to analog phone)->Secure gateway->ABI Unified CM->Offpath ASA->IME Trunk->Inline ASA->MSP Unified CM->Unified IP Phone	Passed	
UC851IF. GTW.003	Analog Gateway - Security	Secure Analog Phone Placing 911 Call (CER)	Verify that the 911 call placed from an analog phone located behind a secure gateway is routed to the correct PSAP with the correct ELIN. Verify that PSAP callback works and that the media setup with the gateway is successful.	PSAP->PSTN->Gateway->Unified CM->CER->Unified CM->Secure Gateway->Analog phone (and reverse callflow)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. GTW.004	Analog Gateway - Security	Analog Phone Calling Analog Phone via PSTN, Call Transferred to Unified IP Phone (1)	Verifies if ABI analog phone calling MSP analog phone via PSTN (SIP -> MGCP), call is transferred to Unified IP Phone.	Analog phone->Secure Gateway->ABI Unified CM->SIP PSTN gateway->PSTN->MSP MGCP gateway->MSP Unified CM->(Secure gateway->Secure analog phone) Xfer to->(Unified IP Phone);	Passed	
UC851IF. GTW.005	Analog Gateway - Security	Analog Phone Calling Analog Phone via PSTN, Call Sent to Mobile Destination	Verifies the ability to configure an analog phone behind a VG224 with an SNR mobile destination. Verifies if an ABI analog phone calling MSP analog phone via PSTN (H.323 -> MGCP), call is sent to remote destination.	Analog phone->Secure Gateway->ABI Unified CM->H.323 PSTN gateway->PSTN->MSP MGCP gateway->MSP Unified CM->(Secure gateway->Secure analog phone) Xfer to (local PSTN gateway->Cell phone);	Passed	
UC851IF. GTW.006	Analog Gateway - Security	Secure Meet-me Conference	Verifies if secure media is set up, when a secure analog phone creates a meet-me conference, and another secure analog phone joins this meet-me conference over a secure SIP trunk.	Secure analog phone->Unified CM1->Secure SIP Trunk->Unified CM2->Meet-me Secure conference bridge	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. GTW.009	Analog Gateway - Security	Secure SIP Session Management Edition call, Analog Phone Initiates Ad-hoc Conference	Verifies the ability to place a secure end-to-end call from an analog phone in one cluster to an analog phone in another cluster, and have the analog phone initiate an ad-hoc conference with a secure Unified IP Phone.	Analog phn->Secure GW->Unified CM1->Secure SIPT->SME->Secure SIPT->Unified CM2->Secure GW->Analog phn; Analog phn->Secure GW->Unified CM1->Secure SIPT->SME->Secure SIPT->Unified CM2->Secure conf bridge	Passed	
UC851IF. GTW.010	Analog Gateway - Security	Secure H.323 Session Management Edition Call, Analog Phone Transfers Call and Answered by Non-secure Endpoint	Verifies the ability to place a secure end-to-end call from an analog phone in one cluster to an analog phone in another cluster, have the analog phone transfer the call, and then answer the call with a non-secure Unified IP Phone.	Analog ph->Secure GW->CUCM1->Secure H.323 trunk->SME->Secure H.323 trunk->CUCM2->Secure GW->Analog phone; transfer->Analog phone->Secure GW->CUCM1->Secure H.323 trunk->SME->Secure H.323 trunk->CUCM2->Unified IP Phone	Passed	
UC851IF. GTW.011	Analog Gateway - Security	Secure Analog Phone Leaving and Receiving Voicemails from Session Management Edition-centralized Unity Connection	Verifies the use of an analog phone behind secure SCCP gateway to leave and receive voicemails from Unity Connection (located in an Session Management Edition cluster).	Analog phn->Secure GW->Unified CM1->Secure H.323 trunk->SME->SIPT -->Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. GTW.012	Analog Gateway - Security	Secure end-to-end Fax over Session Management Edition	Verifies the ability to send a secure end-to-end fax between two clusters via Session Management Edition, when both fax machines are located behind secure gateways.	Fax machine->Secure Gateway->Unified CM1->ICT->SME ->ICT->Secure Gateway->Fax machine	Passed	
UC851IF. GTW.013	Analog Gateway - Security	Analog Phone behind Secure Gateway Calling Over SAF Trunk (SIP)	Verify the ability to place a call from a secure analog phone over a SIP SAF trunk to an Unified IP Phone. Verify audio, hold/resume, and hang up the call.	Analog phone->Secure gateway->Unified CM1->SIP SAF Trunk->Unified CM2->Unified IP Phone	Passed	
UC851IF. GTW.014	Analog Gateway - Security	Analog Phone behind Secure Gateway Calling Over SAF Trunk (H.323)	Verify the ability to place a call from a secure analog phone over an H.323 SAF trunk to an Unified IP Phone. Verify audio, hold/resume, and hang up the call.	Analog phone->Secure gateway->Unified CM1->H.323 SAF Trunk->Unified CM2->Unified IP Phone	Passed	
UC851IF. GTW.015	Analog Gateway - Security	Secure end-to-end Fax (SCCP Gateway and SIP Gateway)	Verifies the ability to send a secure end-to-end fax between two clusters, when one fax machine is located behind a secure SCCP gateway and the other is located behind a secure SIP gateway.	Fax machine->Secure SCCP Gateway->Unified CM1->ICT->Unified CM2->Secure SIP gateway->Fax machine	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. GTW.016	Analog Gateway - Security	Secure end-to-end fax (SCCP gateway and MGCP gateway)	Verifies the ability to send a secure end-to-end fax between two clusters, when one fax machine is located behind a secure SCCP gateway and the other is located behind a secure MGCP gateway.	Fax machine->Secure SCCP Gateway->Unified CM1->ICT->Unified CM2->Secure MGCP gateway->Fax machine	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. GTW.017	Analog Gateway - Security	cBarge with Analog Phone into ICT Call between two Non-secure Unified IP Phones	Verify the ability to place a non-secure call between two Unified IP Phones across a SIP trunk, when one of the Unified IP Phones has a shared line and the other device on the line is an analog phone between a secure SCCP gateway, used to cBarge into the call. Verify if a shared conference bridge is invoked and non-secure media negotiation is successful.	Unified IP Phone->Unified CM1->SIP trunk->Unified CM2->Conf Bridge <- Unified IP Phone and Analog Phone	Passed	
UC851IF. GTW.018	Analog Gateway - Security	cBarge with Secure Analog Phone into ICT Call between two Secure Unified IP Phones	Verify the ability to place a secure call between two Unified IP Phones across a SIP trunk, when one of the Unified IP Phones has a shared line and the other device on the line is an analog phone between a secure SCCP gateway, used to cBarge into the call. Verify if a shared conference bridge is invoked and secure media negotiation is successful.	Secure Unified IP Phone->Unified CM1->Secure SIP trunk->Unified CM2->Secure Conf Bridge <- Secure Unified IP Phone and Secure Analog Phone	Passed	

IP Communicator

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.IPP.001	ICT call flow	Cisco IP Communicator/ Cisco Unified Video Advantage->ICT-> Video phone, Repeat with audio phone (SIP & SCCP)	Verify the Cisco IP Communicator (CIPC)/Cisco Unified Video Advantage (CUVA) interactions with trunks and endpoints.	IP Communicator/ Unified Video Advantage->Unified CM->ICT Trunk->Unified CM->Endpoint	Passed	
UC851IF.IPP.002	ICT call flow	Video phone->ICT->Cisco IP Communicator/ Cisco Unified Video Advantage, repeat with audio phone (SIP & SCCP)	Verify the Cisco IP Communicator/ Cisco Unified Video Advantage interactions with trunks and endpoints.	Endpoint->Unified CM->ICT Trunk->Unified CM->Cisco IP Comunicator/Cisco Unified Video Advantage	Passed	
UC851IF.IPP.003	Conference	Cisco IP Communicator/ Cisco Unified Video Advantage starts Ad-hoc conference and Cisco IP Communicator/ Cisco Unified Video Advantage joins Ad-hoc conference via ICT (SCCP)	Verify the Cisco IP Communicator/ Cisco Unified Video Advantage interactions with trunks, endpoints, and Ad-hoc conference.	Cisco IP Communicator/Cisco Unified Video Advantage->Unified CM->ICT Trunk->Unified CM->Cisco IP Communicator/Cisco Unified Video Advantage	Passed	
UC851IF.IPP.004	Unified SRST	Cisco IP Communicator/ Cisco Unified Video Advantage in SRST site, site becomes isolated (SCCP)	Verify the Cisco IP Communicator/ Cisco Unified Video Advantage interactions with SRST	Cisco IP Communicator-> Unified CME-> Endpoint	Passed	

QSIG

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701EF. QSG.024	Call forwarding and IPC	Calling Line ID Restriction from a QSIG PBX Phone through ICT to a Unified Communications Manager SCCP phone	Verify the calling and called line and name restriction on a call from a QSIG trunk to Unified Communications Manager and an Inter Cluster trunk to a Unified Communications Manager SCCP phone.	PBX Ph1->PBX ->QSIG Trunk->Unified CM->ICT(QSIG)->Unified CM->SCCP Ph1	Passed	
UC712EF. QSG.003	IPV6	Inter PBX Call through Unified Communications Manager Clusters	Verify that QSIG PBX can call to other PBX phones using SIP Gateway via Unified CM	PBX ph1->QSIG trunk->Unified CM->QSIG ICT ->Unified CM->QSIG Trunk->PBX ph1->CFNA->QSIG Trunk->Unified CM->Unity	Passed	
UC851EF. QSG.001	IME with QSIG over SIP	IME calls over QSIG enabled SIP trunks with Callback	Verify that callback works over QSIG enabled IME trunks.		Passed	
UC851EF. QSG.002	PSTN interworking	IME calls over QSIG enabled SIP trunks with Callback and Call Forward All (CFA)			Passed	
UC851EF. QSG.003	QSIG PBX over SIP Gateway	QSIG PBX (ECMA and ISO) calls using QSIG enabled SIP Gateway	Verify that QSIG PBX can call to other PBX phones using SIP Gateway via Unified CM	PBX Ph1->PBX(ECMA)->Gateway->Unified CM(ISO)->Gateway->PBX(ISO)-> PBX Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF. QSG.004	QSIG PX to SIP Gateway	QSIG PBX (ECMA and ISO) calls using QSIG enabled SIP Gateway.	Verify that callback with CFA works with QSIG PBX calls	Variation1:PBX Ph1->PBX(ECMA)->Gateway->CUCM(ISO)->Gateway->PBX(ISO)->PBX Ph1->CFA->PBX Ph2 Variation2: PBX Ph1->PBX(ECMA)->Gateway->CUCM(ECMA)->Gateway->PBX(ISO)->PBX Ph1->CFA->PBX Ph2	Passed	
UC851EF. QSG.005	QSIG PBX to Unified CM	QSIG PBX (ECMA and ISO) calls using QSIG enabled SIP Gateway	Verify that QSIG PBX can call to a Unified CM over SIP ICT	Variation1:PBX Ph1->PBX(ECMA)->Gateway->CUCM(ISO)->ASA->SIP ICT->CUCM(ECMA)->Ph2 Variation2: PBX Ph1->PBX(ISO)->Gateway->CUCM(ECMA)->ASA->SIP ICT->CUCM(ISO)->Ph2 Variation3:Verify Call back	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.QSG.006	Interaction with Unity	Interaction with Unity	Verify that a PBX phone calls to Unity is successful. Verify with Early offer trunks.	1) PBX ph1->QSIG trunk->CUCM->SCCP ph1->CFNA->SIP ICT(qsig) ->CUCM->Unity 2) PBX ph1->QSIG trunk->CUCM->SCCP ph1->CFA->SIP ICT(qsig)->CUCM->Unity 3) PBX ph1->QSIG trunk->CUCM->SCCP ph1->CFB->QSIG ICT ->CUCM->Unity	Passed	
UC851EF.QSG.007	Interaction with DPNSS PBX	Interaction with Callback and Digital Private Network Signaling System (DPNSS) PBX	Verify Callback with DPNSS PBX	SCCP Ph1->Unified CM->SIP (qsig)ICT->Unified CM->SIP(qsig) ICT->Unified CM ->QSIG trunk->VG30D->PBX Ph1	Failed	
UC851EF.QSG.008	Interaction with SIP Proxy	Unified CM Phone Books Callback on a SIP Proxy Phone	Verify Callback with CUSP	SCCP Ph1->Unified CM->SIP (qsig)trunk->CUSP->CME->QSIG trunk->PBX->Ph1	Passed	
UC851EF.QSG.009	Path Replacement	Path Replacement and Blind Transfer	Verify Path replacement on qsig enabled SIP trunks between three Unified CM clusters.	Ph1->Unified CM 1->SIP (qsig)ICT->Unified CM 2->Ph1->CFB->Ph2->XFER->SIP(qsig) ICT->Unified CM 3->Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.QSG.010	Path Replacement	Path Replacement in Trombone call and VoiceMail	Verify Path replacement on qsig enabled SIP trunks between three Unified CM clusters for a trombone call.		Passed	
UC851EF.QSG.011	Path Replacement	Path Replacement with Extension mobility and Call Forward No Answer (CFNA)	Verify Path replacement on qsig enabled SIP trunks between three Unified CM clusters with EM,CFNA		Passed	
UC851EF.QSG.012	Path Replacement	Path Replacement involving QSIG PBX and Unity	Verify Path replacement with QSIG PBX	Variaton1:Ph1->CUCM 1->SIP(qsig) Gateway->PBX->PBX Ph1->XFER->MGCP(qsig) Gateway ->CUCM 1->Unity Variation2: Ph1->CUCM 1->SIP(qsig) Gateway->PBX->PBX Ph1->XFER->SIP (qsig) Gateway->CUCM 1->Unity	Passed	
UC851EF.QSG.015	Path Replacement	Call Forward by Reroute with Extension Mobility and IME.	Verify Call forward on qsig enabled IME trunks between three Unified CM clusters with EM	EMA1->Unified CM 1->IME trunk>Unified CM 2->Ph1->CFNA->AnnexM1->UnifiedCM3->VM	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.QSG.016	Call forward by reroute in IME	Path Replacement in Trombone Call involving QSIG PBX and a Two Line Unified IP Phone	Verify Path replacement with IME for a two line Unified IP Phone.	Ph1->CUCM 1->IME trunk->CUCM 2->Ph1_L1->CFB ->Ph1_L2->XFER->QSIG Trunk->PBX Ph1->XFER->CUCM2->IME trunk->CUCM 1->Ph2	Passed	
UC851EF.QSG.017	Path Replacement	Call Diversion by Reroute involving Digital Private Network Signaling System (DPNSS) PBX and a two line Unified IP Phone	Verify Call diversion by reroute over IME trunk	Ph1->CUCM 1->IME trunk->CUCM 2->Ph1_L1->CFB ->Ph1_L2->CFNA->QSIG Trunk->VG30D->PBX Ph1->CFNA->VG30D->SIP(qsig)Gateway Trunk->CUCM 1->Ph2	Passed	
UC851EF.QSG.018	Path Replacement	Interaction of Call Forward and Call Transfer with a Third Party Operator	Verify that a call from a Third party Operator console to a QSIG PBX phone is transferred to another QSIG PBX phone and call forwarded on no answer via SIP ICT to a Unified CM SIP Phone	OP Cons1->Unified CM->QSIG Trunk->PBX->PBX Ph1->XFER_C->PBX Ph2->CFNA->Unified CM->SIP (QSIG) ICT->Unified CM->SIP Ph1	Passed	
UC851EF.QSG.019	Third Party op console and SIP(qsig) trunk	Interworking with PSTN network and SIP (QSIG) Trunk	Verify Call forwarding scenario involving a call over a SIP trunk ,QSIG-ICT and MGCP Gateway to pots	PSTN Ph 1->SIP(BRI/PRI)->Unified CM1->SCCP Ph1 CFx->SIP ICT(QSIG)->Unified CM2->SCCP/SIP Ph2->CFB->MGCP (BRI/PRI)->PSTN Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.QSG.020	Calling Line ID Restriction	Calling Line ID Restriction from QSIG PBX Phones	Verify Calling Line ID Restriction from a QSIG PBX phone via ICT to a CCM SCCP phone	PBX Ph1->PBX->SIP(QSIG) Gateway->Unified CM->SIP ICT(qsig)->Unified CM->SCCP Ph1	Passed	
UC851EF.QSG.021	CFNA with CME and SIP trunks	Interworking with Unified CME	Verify CFNA and transfer interaction with CME, QSIG PBX ,MGCP Gateway ,ICT and IPMA manager	SCCP Ph1->CME->IPIP Gateway (H323)->GK->CUCM->SIP(qsig) Gateway->PBX->PBX Ph1->XFER_C->CUCM->ATA->ATA Ph1->CFNA->CUCM->SIP(ICT)(QSIG)->CUCM->IPMA Mgr Ph1	Passed	
UC851EF.QSG.023	Call forwarding	Multiple Call Forwarding	Verify Multiple Call forwarding over SIP proxy ,Unified CM and ICT	SIP Ph1->CME->CUSP->SIP (qsig) trunk->Unified CM->SIP Ph2->CFA->SIP ICT (QSIG)->Unified CM->SCCP Ph1->CFA->SIP ICT (QSIG)->Unified CM->SCCP Ph2->CFA->SIP ICT(QSIG)->Unified CM->SIP Ph2	Passed	
UC851EF.QSG.024	Session Management Edition	QSIG PBX and IP communicator	Verify Interaction of Call forward and Transfer with Unified CM, SIP Phone, QSIG PBX, and IP communicator.		Passed w/exception	CSCti30318
UC851EF.QSG.027	Session Management Edition	Path Replacement over Annex M1 and SIP Trunks	Verify path replacement in a Trombone call over AnnexM1 and SIP Trunks.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.QSG.028	Session Management Edition	Path Replacement over SIP Trunks	Verify Path replacement over SIP trunks in Session Management Edition.		Passed	
UC851EF.QSG.029	Session Management Edition	Path Replacement over SIP Trunks	Verify path replacement in a Trombone call over SIP trunks in .Session Management Edition.		Passed	
UC851EF.QSG.030	Session Management Edition	Interaction of Session Management Edition with Unity	Verify Centralized Unity in Session Management Edition.	SIP Ph1->CUCM1(leaf)->SIP(qsig) ICT->SME->SIP(qsig)ICT-> CUCM2(leaf)->SCCP ph1->CFNA/CFB/CFA/iDivert->SIP(qsig) ICT->SME->Unity	Passed	
UC851EF.QSG.031	Session Management Edition	PBX and Session Management Edition Interoperability	Verify PBX calls from leaf.	Ph1->PBX->MGCP(qsig) Gateway->Unified CM1(leaf)->SIP(qsig)SME->SIP(qsig)ICT->Unified CM2(leaf)->ph1	Passed w/exception	CSCti30318

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.QSG.032	Session Management Edition	Interaction with SIP Trunks with Call Transfer Remote	Verify call transfer over Session Management Edition when AnnexM1 trunk is a delayed offer trunk.	"Variation1:SIP Ph1->CUCM1(leaf)->AnnexM1->SME->SIP (qsig)trunk->CUCM3(leaf)->SCCP Ph1->Transfer (blind-remote)->SIP (qsig) trunk->SME->AnnexM1->CUCM1(leaf)->SCCP ph1 Variation2:SIP Ph1->CUCM1->Transfer (attended-local)->SIP trunk->CUCM2_Tand->SIP trunk ->CUCM3->SCCP ph1	Passed	
UC851EF.QSG.033	Session Management Edition	Interaction with SIP Trunks with Call Hold/Resume	Verify Call Hold/Resume over Session Management Edition.	"Variation1:SIP Ph1->CUCM1(leaf)->SIP (qsig) trunk->SME->SIP (qsig) trunk->CUCM3(leaf)->SCCP Ph1->Hold/Resume->SIP (qsig) trunk->SME->SIP (qsig) trunk ->CUCM1(leaf)->SCCP ph1 Variation2:SIP Ph1->Hold/Resume(Local MOH) CUCM1->SIP trunk->SME->SIP trunk->CUCM3->SCCP Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.QSG.034	Session Management Edition	Interaction of two Session Management Editions with a static trunk between them.	Verify path replacement with SAF trunks that become the shortest route than the static route between Session Management Edition to Session Management Edition calls, and inbound trunks to Session Management Edition are delayed offer.	"Stage1:SIP Ph1->CUCM1(leaf)->SIP(qsig)trunk->SME1->SIP (qsig) trunk->SME2->SIP(qsig) trunk->CUCM2(leaf)->SCCP ph1 Stage2:SIP Ph1->CUCM1(leaf)->SAF SIP (qsig) trunk->SME2->SIP(qsig) trunk->CUCM2(leaf)->SCCP ph1	Passed	
UC851EF.QSG.035	Session Management Edition	Interaction of Session Management Edition and Path Replacement	Verify path replacement with H.225 and SIP trunks over Session Management Edition and AnnexM1 Trunks are delayed offer.	"Stage1:SIP Ph1->CUCM1(leaf)->SIP(qsig)trunk->SME->SIP (qsig) trunk->CUCM2(leaf)->SCCP ph1->Xfer->SME->AnnexM1 trunk->CUCM3(leaf)->SCCP ph1 Stage2:SIP Ph1->CUCM1(leaf)->AnnexM1 trunk->CUCM2(leaf)->SCCP ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.QSG.036	Session Management Edition	Session Management Edition Routing Decision based on Dialed Number Analysis	Verify PSTN breakout calls using digit manipulation on Session Management Edition.	"Stage1:Ph1->REM1->CUCM1(leaf)->SIP(qsig)ICT->SME->SIP(qsig)ICT->CUCM1(leaf)->REM2->PSTN->Ph1 Stage2:Ph1->REM2->CUCM1(leaf)->SIP(qsig)ICT->SME->SIP(qsig)ICT->CUCM1(leaf)->REM2->PSTN->Ph1 Stage3:Ph1->REM1->CUCM1(leaf)->SIP(qsig)ICT->SME->SIP(qsig)ICT->CUCM1(leaf)->REM1->PSTN->Ph1	Passed	
UC851EF.QSG.037	Session Management Edition	Session Management Edition Routing Decision based on Calling Number Analysis	Verify PSTN breakout calls using geolocations on Session Management Edition.	"Stage1:Ph1->REM1->CUCM1(leaf)->SIP(qsig)ICT->SME->SIP(qsig)ICT->CUCM1(leaf)->REM2->PSTN->Ph1 Stage2:Ph1->REM2->CUCM1(leaf)->SIP(qsig)ICT->SME->SIP(qsig)ICT->CUCM1(leaf)->REM2->PSTN->Ph1 Stage3:Ph1->REM3->CUCM1(leaf)->SIP(qsig)ICT->SME->Gateway->PSTN-Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.QSG.060	Path Replacement	Path Replacement involving QSIG PBX and Unity	Verify Path replacement with QSIG PBX	Variaton1:Ph1->CUCM 1->SIP(qsig) Gateway->PBX->PBX Ph1->XFER->MGCP(qsig)Gateway->CUCM 1->Unity Variation2: Ph1->CUCM 1->SIP(qsig) Gateway->PBX->PBX Ph1->XFER->SIP(qsig) Gateway->CUCM 1->Unity	Passed	
UC851EF.QSG.062	QSIG PBX over SIP Gateway	QSIG PBX (ECMA and ISO) calls using QSIG enabled SIP Gateway	Verify that QSIG PBX can call to other PBX phones using SIP Gateway via Unified Communications Manager	PBX Ph1->PBX(ECMA)->Gateway->CUCM(ISO)->Gateway->PBX(ISO)->PBX Ph1	Passed	
UC851EF.QSG.063	QSIG PBX over SIP Gateway	QSIG PBX (ECMA and ISO) calls using QSIG enabled SIP Gateway	Verify that QSIG PBX can call to other PBX phones using SIP Gateway via Unified Communications Manager	PBX Ph1->PBX(ECMA)->Gateway->Unified CM(ISO)->Gateway->PBX(ISO)->PBX Ph1	Passed	
UC851EF.QSG.064	Session Manager	Interaction of Session Management Edition with Unity	Verify Unity in Session Management Edition outbound early offer trunks and inbound delay offer trunks.	SIP Ph1->CUCM1(leaf)->SIP(qsig) ICT->SME->SIP(qsig)ICT->CUCM2(leaf)->CUCIRTX->CFNA/CFB/CFA/iDivert->SIP(qsig) ICT->SME->Unity	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF. QSG.065	Session Manager	Interaction with SIP Trunks with call Transfer Remote	Verify Call Transfer over Session Management Edition when AnnexM1 trunk is a delayed offer trunk.	Variation1:SIP Ph1->CUCM1(leaf)->AnnexM1->SME->SIP (qsig)trunk->CUCM3(leaf)->SCCP Ph1->Transfer (blind-remote)->SIP (qsig) trunk->SME->AnnexM1->CUCM1(leaf)->SCCP ph1 Variation2:SIP Ph1->CUCM1->Transfer (attended-local)->SIP trunk->CUCM2_Tand->SIP trunk ->CUCM3->SCCP ph1	Passed	
UC851EF. QSG.066	Session Manager	Interaction with SIP Trunks with Call Hold/Resume	Verify Call Hold/Resume over Session Management Edition.	Variation1:SIP Ph1->CUCM1(leaf)->SIP (qsig) trunk->SME->SIP (qsig) trunk->CUCM3(leaf)->SCCP Ph1->Hold/Resume->SIP (qsig) trunk->SME->SIP (qsig) trunk ->CUCM1(leaf)->SCCP ph1 Variation2:SIP Ph1->Hold/Resume(Local MOH) CUCM1->SIP trunk->SME->SIP trunk->CUCM3->SCCP Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF. QSG.067	QSIG PBX over SIP Gateway	QSIG PBX (ECMA and ISO) Calls using QSIG enabled SIP Gateway	Verify that callback with CFA works with QSIG PBX calls	Variation1:PBX Ph1->PBX(ECMA)->Gateway->CUCM(ISO)->Gateway->PBX(ISO)->PBX Ph1->CFA->PBX Ph2 Variation2: PBX Ph1->PBX(ECMA)->Gateway->CUCM(ECMA)->Gateway->PBX(ISO)->PBX Ph1->CFA->PBX Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.QSG.068	QSIG PBX to Unified CM	QSIG PBX (ECMA and ISO) calls using QSIG enabled SIP Gateway	Verify that QSIG PBX can call to a Unified Communications Manager over SIP ICT	Variation1:PBX Ph1->PBX(ECMA)->Gateway->CUCM(ISO)->ASA->SIP ICT->CUCM(ECMA)->Ph2 Variation2: PBX Ph1->PBX(ISO)->Gateway->CUCM(ECMA)->ASA->SIP ICT->CUCM(ISO)->Ph2 Variation3:Verify Call back	Passed	
UC851EF.QSG.071	Interaction with Unity	Interaction with Unity	Verify that a PBX phone calls to Unity is successful. Verify with Early offer trunks.	1) PBX ph1->QSIG trunk->CUCM->SCCP ph1->CFNA->SIP ICT(qsig) ->CUCM->Unity 2) PBX ph1->QSIG trunk->CUCM->SCCP ph1->CFA->SIP ICT(qsig)->CUCM->Unity 3) PBX ph1->QSIG trunk->CUCM->SCCP ph1->CFB->QSIG ICT ->CUCM->Unity	Passed	

Quality of Service

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF. QOS.001	E2E RSVP	E2E RSVP Call Between Phones in two Unified Communications Manager Clusters	Verify an E2E RSVP call between phones in two Unified Communications Manager clusters.	Originating Phone->RSVP Agent->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2->RSVP Agent->Terminating Phone	Passed	
UC802EF. QOS.004	E2E RSVP	E2E RSVP Call Between a Phone in a Remote Branch of one Cluster to a Phone in Another Cluster Where Location is Set as Hub-None	Verify an E2E RSVP call between a phone in a remote branch of one cluster to a phone in another cluster where location is set as hub-none.	Originating Phone->Remote RSVPAgent->Remote Branch->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2 (hub_none)->RSVP Agent->Terminating Phone	Passed	
UC802EF. QOS.006	E2E RSVP	E2E RSVP Call from an Unified IP Phone in one Cluster to a Remote FXS Phone in Another Cluster Registered to a Remote Branch (SIP Gateway) which has Pre-conditions Enabled	Verify an E2E RSVP call from an Unified IP Phone in one cluster to a remote FXS phone in another cluster that is registered to a Remote Branch (SIP Gateway) which has pre-conditions enabled.	IP Phone->RSVPAgent ->Unified CM 1->SIP ICT->Unified CM 2->SIP Trunk->Remote SIP Gateway->FXS Phone	Passed	
UC802EF. QOS.009	E2E RSVP	E2E RSVP Call Between Unified IP Phone to FXS Phone on Same Remote Site	Verify an E2E RSVP call between Unified IP Phone to FXS phone on the same remote site.	Remote IP Phone->Cisco Unified Communications Manager->SIP Trunk->Remote FXS Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF. QOS.014	E2E RSVP	E2E RSVP Call from Phone in Unified Communications Manager Cluster to Unified Communications ManagerE Phone Aggregated by Unified SIP Proxy	Verify an E2E RSVP call from a phone in Unified CM cluster to a Unified CME phone aggregated by Unified SIP Proxy.	Originating Phone->RSVPAgent ->Unified CM Cluster 1->SIP Trunk->Cisco Unified SIP Proxy->Unified CME->RSVPAgent->Terminating Phone	Passed	
UC802EF. QOS.017	E2E RSVP	Supplementary Services in an E2E RSVP Call Between Central Site Phones in two Unified Communications Manager Clusters	Verify supplementary services in an E2E RSVP call between central site phones in two Unified Communications Manager clusters. Verify for Hold/Resume, Transfer and Conference.	Originating Phone->RSVP Agent->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2->RSVP Agent->Terminating Phone	Passed	
UC802EF. QOS.025	E2E RSVP	E2E RSVP Call from IP Communicator in Central Site to Unified CME phone via Unified SIP Proxy	Verify E2E RSVP call from IP Communicator in central site to Unified CME phone via Unified SIP Proxy.	CIPC->Cisco Unified Communications Manager->SIP Trunk->Cisco Unified SIP Proxy->Unified CME->IP Phone	Passed	
UC802EF. QOS.026	E2E RSVP	E2E RSVP Call from Unified Personal Communicator in one Cluster to another Unified Personal Communicator in a Different Cluster	Verify the E2E RSVP call from Unified Personal Communicator in one cluster to another Unified Personal Communicator in a different cluster.	Unified Personal Communicator->Cisco Unified Presence->Cisco Unified Communications Manager->SIP Trunk->Cisco Unified Communications Manager->Cisco Unified Presence->Unified Personal Communicator	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF. QOS.027	E2E RSVP	E2E RSVP Call from Remote Unified IP Phone to PSTN Phone Over SIP PRI	Verify E2E RSVP call from remote Unified IP Phone to PSTN phone over SIP PRI.	Remote IP Phone->Cisco Unified Communications Manager 0> SIP Trunk->SIP PRI Gateway->PSTN Phone	Passed	
UC802EF. QOS.029	E2E RSVP	E2E RSVP and Unified Enterprise Attendant Console Interoperability	Verify an E2E RSVP call when a call is transferred by Unified Enterprise Attendant Console to a phone in another cluster.	IP Phone->Unified Enterprise Attendant Console (CUEAC)->IP Phone->RSVP Agent->Cisco Unified Communications Manager 1->SIP ICT->Cisco Unified Communications Manager 2->RSVP Agent->IP Phone	Passed	
UC802EF. QOS.901	E2E RSVP	Basic E2E RSVP Video Intercluster Call	"Verify that video EPs can call from one cluster to another cluster.			
Both the clusters invoke only one central RSVP agent per cluster.	"Unified Video Advantage->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G	Passed				

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF. QOS.902	E2E RSVP	E2E RSVP Video Call Between Remote and Central Phones Between Two Different Clusters	Verify that remote video EP can call from one cluster to other cluster's central video EP when Cluster 1 and Cluster 2 must invoke their remote RSVP agents.	Remote UC Integration™ for Microsoft Office Communicator->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G	Passed	
UC802EF. QOS.903	E2E RSVP	E2E RSVP Video Intercluster Call and Transfer to the Remote Site of Called Cluster	Verify that video EP can call from one cluster to other cluster. Both the clusters invoke only one central RSVP agent per cluster. Verify the ability to transfer the call to the remote Video EP of the second cluster.	RT Video->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G->XFER->remote UC Integration™ for Microsoft Office Communicator	Passed	
UC802EF. QOS.904	E2E RSVP	E2E RSVP Video Intercluster Call and Conference with Remote Site of Calling Cluster	Verify that Video EP can call from one cluster to other cluster when both the clusters invoke only one central RSVP agent per cluster and the second phone initiates a conference with a remote video EP in first cluster.	7985G->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G->CNF->Unified CM cluster 1->remote 7985G	Passed	
UC802EF. QOS.905	E2E RSVP	E2E RSVP Video Call Between Central Phones of Two Different Clusters and Location is set as Hub-None for Both the End-Points	Verify that video EP can call from one cluster to another when both clusters invoke only one central RSVP agent per cluster. verify that the location is set as Hub-None for both the EPs.	RT Video->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->UC Integration™ for Microsoft Office Communicator	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF. QOS.001	E2E RSVP SIP Preconditions Unified Border Element	E2E RSVP Audio Call Between Unified Communications Manager 7.x and 8.x Clusters	Verify E2E RSVP audio calls between Unified Communications Manager 7.x and 8.x clusters.	IP Phone->CCM 7.x->SIP Trunk->Unified Border Element->SIP Trunk->Cisco Unified Communications Manager 8.0->RSVP Agent->IP Phone	Passed	
UC851EF. QOS.002	E2E RSVP SIP Preconditions Unified Border Element	E2E RSVP Calls Between Unified Communications Manager 7.x and 8.x Clusters with Location as Hub-None	Verify E2E RSVP call between Unified Communications Manager 7.x and 8.x clusters with location as hub-none for the end-points.	IP Phone->Cisco Unified Communications Manager 7.x->SIP Trunk->Unified Border Element->SIP Trunk->Cisco Unified Communications Manager 8.0->RSVP Agent->IP Phone (Hub_None)	Passed	
UC851EF. QOS.003	E2E RSVP SIP Preconditions Unified Border Element	E2E RSVP Calls Between Remote Phone in Unified Communications Manager 7.x Cluster and Central Phone in 8.x Cluster	Verify E2E RSVP calls between remote phone in Unified Communications Manager 7.x cluster and central phone in 8.x cluster.	Remote IP Phone->Cisco Unified Communications Manager 7.x->SIP Trunk->Unified Border Element->SIP Trunk->Cisco Unified Communications Manager 8.0->RSVP Agent->IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.QOS.004	E2E RSVP SIP Preconditions Unified Border Element	E2E RSVP Calls Between Remote Phones in Unified Communications Manager 7.x and 8.x Clusters	Verify E2E RSVP calls between remote phones in Unified Communications Manager 7.x and 8.x clusters.	Remote IP Phone->Cisco Unified Communications Manager 7.x->SIP Trunk->Unified Border Element->SIP Trunk->Cisco Unified Communications Manager 8.0->RSVP Agent->Remote IP Phone	Passed	
UC851EF.QOS.005	E2E RSVP SIP Preconditions Unified Border Element	Reservation Failure in E2E RSVP Calls	Verify PSTN fallback in E2E RSVP calls when reservation failure occurs.	Stage1: IP Phone->Cisco Unified Communications Manager 7.x->SIP Trunk->Unified Border Element->SIP Trunk->Cisco Unified Communications Manager 8.0->RSVP Agent->IP Phone; After reservation failure: IP Phone->Cisco Unified Communications Manager 7.x->PSTN Gateway->CCM 8.0->IP Phone	Passed	
UC851EF.QOS.006	E2E RSVP SIP Preconditions Unified Border Element	Supplementary Services in E2E RSVP Calls Between Unified Communications Manager 7.x and CCM 8.x Clusters	Verify supplementary services in E2E RSVP calls between Unified Communications Manager 7.x and Cisco Unified Communications Manager 8.x clusters.	IP Phone->Cisco Unified Communications Manager 7.x->SIP Trunk->Unified Border Element->SIP Trunk->Cisco Unified Communications Manager 8.0->RSVP Agent->IP Phone	Passed	

Reliability, Load

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EL.REL.001	Reliability	Basic IP to IP Intra Cluster Calls	Verify that IP to IP calls in a large size cluster of 3800 Busy-hour Call Attempts (BHCA) are successful for 120 hours.	SCCP Ph1->CUCM->SCCP Ph 2	Passed	
UC851EL.REL.002	Reliability	IP to IP Calls over PSTN	Verify that IP to PSTN to IP calls in a large size cluster of 7200 BHCA are successful for 120 hours.	SCCP Ph 1->CUCM->MGCP g/w->CUCM->SCCP Ph2 SIP Ph 1->CUCM->MGCP g/w->CUCM->SIP Ph2	Passed	
UC851EL.REL.003	Reliability	IP to IP Calls over Inter-Cluster Trunk	Verify that IP to IP inter-cluster calls with supplementary services (hold, blind transfer, consultative transfer) in a large size cluster with 720 BHCA are successful for 120 hours.	SCCP Ph 1->CUCM->ICT Trunk1->CUCM->SCCP Ph2	Passed	
UC851EL.REL.004	Reliability	IP to IP over Inter-Cluster Trunk Call Checking Interoperability with Version 7.1.3 of Unified Communications Manager	Verify that IP to IP inter-cluster calls with supplementary services (hold, blind transfer, consultative transfer) in a large size cluster with 1,350 BHCA are successful for 120 hours.	IP Ph1->CUCM->IP Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EL. REL.005	Reliability	IP to IP Calls between Unified Communications Manager Express and Unified Communications Manager Cluster over Inter-Cluster Trunk	Verify that IP to IP calls between Unified Communications Manager Express and Unified Communications Manager clusters with 4,000 BHCA over Inter-Cluster Trunks are successful for 120 hours.	SCCP Ph1->CUCM->GK ->IPIPGW->GK-> CME->SCCP Ph2	Passed	
UC851EL. REL.006	Reliability	Remote IP to Remote IP Over PSTN Calls	Verify that IP to PSTN to IP calls in a remote branch office with 7,800 BHCA are successful for 120 hours.	REM SCCP Ph 1->CUCM->MGCP/H.323/SIP g/w->PSTN->MGCP/H.323/SIP g/w->CUCM->REM SCCP Ph2	Passed	
UC851EL. REL.007	Reliability	IP to IP RSVP-enabled WAN Calls	Verify that RSVP-enabled IP to IP calls from remote branch office to central site with 2500 BHCA over WAN are successful for 120 hours.	REM SCCP Ph1->CUCM->RSVP Agent 1 (Remote)->RSVP Agent 2(Central site)->CUCM->SCCP Ph 2; SCCP Ph1->CUCM->RSVP Agent 2(Central site)->RSVP Agent 1(Remote)->CUCM->REM SCCP Ph 2	Passed	
UC851EL. REL.008	Reliability	Calls to Unity Connection Voicemail	Verify that Unity Connection Voicemail calls of 3600 BHCA are successful for 120 hours.	Stage1: SCCP Ph1->CUCM1->SCCPPh2->CFNA->VOICE MAIL ; Stage2: SCCPPh2->VOICE MAIL->RETRIEVAL	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EL. REL.009	Reliability	Five Day Load Run of 36000 BHCA(All Call Flows) in a Large Size Cluster	Verify that different call flows (IP to IP, IP to PSTN to IP, Unity Connection call flow) are running successfully for 120 hours with 36000 BHCA.		Passed	
UC851EL. REL.010	Reliability	Basic IP to IP Intra-cluster Calls in Medium Size Cluster	Verify that IP to IP intra-cluster calls in a medium size cluster of 2440 BHCA are successful for 120 hours.	SCCP Ph1->CUCM->SCCP Ph 2	Passed	
UC851EL. REL.011	Reliability	IP to IP Calls over PSTN in Medium Size Cluster	Verify that IP to PSTN to IP calls in a medium size cluster of 5700 BHCA are successful for 120 hours.	SCCP Ph 1->CUCM->MGCP g/w->CUCM->SCCP Ph2; SIP Ph 1->CUCM->MGCP g/w->CUCM->SIP Ph2	Passed	
UC851EL. REL.012	Reliability	IP to IP Calls Over Inter-Cluster Trunk in Medium Size Cluster	Verify that IP to IP inter cluster calls with supplementary services(hold, blind transfer, consultative transfer) in a medium size cluster 720 BHCA are successful for 120 hours.	SCCP Ph 1->CUCM->ICT Trunk1->CUCM->SCCP Ph2	Passed	
UC851EL. REL.013	Reliability	IP to IP Calls between Communications Manager Express and Call Manager over Inter-Cluster Trunk	Verify that IP to IP calls between Call Manager Express (CME) to Unified Communications Manager of 1800 BHCA are successful for 120 hours.	SCCP Ph1->CUCM->GK->IPIPGW->GK->CME->SCCP Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EL. REL.014	Reliability	Calls to Unity Connection Voicemail in Medium Size Cluster	Verify that Unity Connection calls of 600 BHCA are successful for 120 hours.	Stage1: SCCP Ph1->CUCM1->SCCPPh2->CFNA->VOICE MAIL; Stage2: SCCPPh2->VOICE MAIL->RETRIEVAL	Passed	
UC851EL. REL.015	Reliability	IP to IP ICT Call Going Through Tandem Cluster	Verify that IP to IP inter cluster calls with supplementary services(hold, blind transfer, consultative transfer) involving third cluster of 850 BHCA are successful for 120 hours.	SCCP Ph 1->CUCM1->ICT Trunk->CUCM2->ICT Trunk2->CUCM3->SCCP Ph2	Passed	
UC851EL. REL.016	Reliability	IP to IP ICT Call Checking Interoperability with Previous Version Call Manager	Verify that IP to IP inter cluster calls with supplementary services(hold, blind transfer, consultative transfer) in a medium size cluster 720 BHCA are successful for 120 hours.	SCCP Ph 1->CUCM->ICT Trunk1->CUCM->SCCP Ph2	Passed	
UC851EL. REL.017	Reliability	Five Day Load Run of 12000 BHCA (All Call Flows) in Medium Size Cluster	Verify that different call flows (IP to IP, IP to PSTN to IP, Unity Connection call flow) are running successfully for 120 hours with 12000 BHCA.		Passed	
UC851EL. REL.018	Reliability	Basic IP to IP Intra Cluster Calls in Langley Cores	Verify that IP to IP calls in Langley Cores of 1000 BHCA are successful for 120 hours.	SCCP Ph1->Langley Cores CUCM->SCCP Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EL. REL.019	Reliability	IP to IP calls over PSTN in Langley Unified Communications Manager Cores	Verify that IP to PSTN to IP calls in Langley Cores Unified Communications Manager of 1040 BHCA are successful for 120 hours.	SCCP Ph 1->Langley Cores CUCM->MGCP g/w->Langley Cores CUCM->SCCP Ph2 SIP Ph 1->CUCMBE->MGCP g/w->CUCMBE->SIP Ph2	Passed	
UC851EL. REL.020	Reliability	Remote IP to Remote IP Over PSTN Calls	Verify that IP to PSTN to IP calls in a Remote Branch Office of 360 BHCA are successful for 120 hours.	REM SCCP Ph 1->Langley Cores CUCM->MGCP/H.323/SIP g/w->PSTN->MGCP/H.323/SIP g/w->Langley Cores CUCM->REM SCCP Ph2	Passed	
UC851EL. REL.021	Reliability	IP to IP RSVP WAN Calls	Verify that RSVP enabled IP to IP calls from Remote Branch Office to Central site Over WAN of 360 BHCA are successful for 120 hours.	REM SCCP Ph1->Langley Cores CUCM->RSVP Agent 1 (Remote)->RSVP Agent 2(Central site)->CUCMBE->SCCP Ph 2;SCCP Ph1->CUCMBE->RSVP Agent 2(Central site)->RSVP Agent 1(Remote)->Langley Cores CUCM->REM SCCP Ph 2	Passed	
UC851EL. REL.022	Reliability	Calls to Unity Connection Voicemail	Verify that Unity Connection calls of 600 BHCA are successful for 120 hours.	Stage1: SCCP Ph1->Langley Cores CUCM->SCCPPh2->CFNA->VOICEMAIL; Stage2: SCCPPh2->VOICE MAIL->RETRIEVAL	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EL. REL.023	Reliability	Five Day Load Run of 3400 BHCA(All Call Flows) on Langley Cores 4 Apps deploymen t Site	Verify that Different call flows (IP to IP, IP to PSTN to IP, Unity Connection call flow) are running successfully for 120 hours with 3400 BHCA.		Passed	
UC851EL. REL.024	Reliability	IP to IP Inter-cluste r Trunk Call Going Through Session Manageme nt Edition Cluster using QSIG/SIP SAF Trunk with E2E RSVP	Verify that IP to IP inter cluster calls with supplementary services (Basic, Hold, Transfer, Conference) through Session Management Edition cluster of 4000 BHCA are successful with E2E RSVP over SAF enabled QSIG over SIP Trunks for 120 hrs.	SCCP Ph1->CUCM->(Q SIG/SIP) SAF enabled ICT Trunk->CUCM-S ME->(QSIG/SIP) SAF enabled ICT Trunk->CUCM->S CCP Ph2	Passed	
UC851EL. REL.025	Reliability	IP to IP Inter-cluste r Trunk Call Going Through Session Manageme nt Edition Cluster using QSIG/SIP SAF Trunks	Verify that IP to IP inter cluster calls through Session Management Edition cluster of 4000 BHCA are successful over SAF enabled QSIG/SIP Trunks for 120 hrs.	SCCP Ph1->CUCM->(Q SIG/SIP) SAF enabled ICT Trunk->CUCM-S ME->(QSIG/SIP) SAF enabled ICT Trunk->CUCM->S CCP Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EL.REL.026	Reliability	IP to IP Inter-cluster Trunk Call Going through Session Management Edition Cluster Using QSIG/SIP Static Trunks	Verify that IP to IP inter cluster calls with supplementary services (Basic, Hold, Transfer, Conference) through Session Management Edition cluster of 5000 BHCA are successful over QSIG over SIP Trunks for 120 hrs.	SCCP Ph1->CUCM->(QSIG/SIP) ICT Trunk->CUCM-SME->(QSIG/SIP) ICT Trunk->CUCM->SCCP Ph2	Passed	
UC851EL.REL.027	Reliability	IP to IP Inter-cluster Trunk Call Going Through Session Management Edition Cluster from Remote Site to Another Remote Site with E2E RSVP	Verify that IP to IP inter cluster E2E RSVP calls with supplementary services (Basic, Hold, Transfer, Conference)from a remote to another remote through Session Management Edition Cluster for 120 hrs.	SCCP Ph1->Remote->CUCM->QSIG/SIP Trunk->CUCM-SME->QSIG/SIP Trunk->CUCM->Remote->SCCP Ph2	Passed	
UC851EL.REL.028	Reliability	IP to IP Inter-cluster Trunk Call Going Through Session Management Edition Cluster using QSIG/SIP to SAF Enabled QSIG/SIP Trunks	Verify that IP to IP inter cluster calls via Session Management Edition cluster of 4000 BHCA are successful QSIG/SIP Static to QSIG/SIP SAF Trunks for 120 hrs.	SCCP Ph1->CUCM->QSIG/SIP ICT Trunk->CUCM-SME->QSIG/SIP SAF enabled ICT Trunk->CUCM->SCCP Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EL.REL.029	Reliability	IP to IP Inter-cluster Trunk Call to Check Interoperability Going Through Session Management Edition Cluster Using H.323 annex1/QSIG trunks	Verify that IP to IP inter cluster calls through Session Management Edition cluster to a Call Manager cluster running on lower version of 5000 BHCA are successful over H.323 annex1/QSIG Trunks for 120 hrs.	SCCP Ph1->CUCM->QSIG ICT Trunk->CUCM-SME->QSIG/SIP ICT Trunk->CUCM->SCCP Ph2	Passed	
UC851EL.REL.030	Reliability	Calls to Centralized Unity Connection Voicemail on the Same from Leaf Clusters	Verify that Unity Connection Calls of 2000 BHCA are successful when leaf nodes will access the centralized Unity Connection Server on Session Management Edition for depositing and retrieving voice mails for 120 hours.	Stage1: SCCP Ph1->CUCM1->SCCPPh2->CFNA->SIP Trunks->CUCM-SME->VOICE MAIL Stage2: SCCPPh2->VOICE MAIL->RETRIEVAL	Passed	
UC851EL.REL.031	Reliability	IP to IP Calls over PSTN from the Gateway on the Session Management Edition Cluster	Verify that IP to PSTN to IP calls in a medium size cluster of 2500 BHCA are successful when leaf node access PSTN from the gateway residing at central Session Management Edition site.	SCCP Ph 1->CUCM->QSIG Trunk->CUCM-SME->MGCP g/w->CUCM-SME->QSIG Trunk->CUCM->SCCP Ph2	Passed	
UC851EL.REL.033	Reliability	Session Management Edition IP-IP Calls with 30000 BHCA	Verify that IP to IP calls placed in Unified Communication Manager CME Centralized cluster with 30000 BHCA are successful on different QSIG & QSIG/SIP Trunks for 120 hours.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IL.REL.001	Reliability	IP to Analog Calls	Verify that calls from a Unfiied IP phone to an Analog (6624/VG224/VG248) phone are successful.		Passed	
UC851IL.REL.002	Reliability	IP to IP Calls	Verify that calls from a Unified IP phone to another IP phone are successful.		Passed	
UC851IL.REL.004	Reliability	IP to Inter-Cluster Trunk to Unified Communications Manager Express to IP Calls	Verify that calls from an Unified IP Phone to Unified Communications Manager Express Unified IP Phone over Inter-Cluster Trunk are successful.		Passed	
UC851IL.REL.005	Reliability	IP to Unified Contact Center Express Calls	Verify that calls from an Unified IP Phone to a Unified Contact Center Express agent phone are successful.		Passed	
UC851IL.REL.006	Reliability	IP to PSTN to IP Calls	Verify that calls from an Unified IP Phone to another Unified IP Phone over PSTN are successful.		Passed	
UC851IL.REL.007	Reliability	IP to Inter-Cluster Trunks to IP Calls	Verify that calls from an Unified IP Phone in one cluster to an Unified IP Phone in another cluster, over Inter-Cluster Trunks are successful.		Passed	
UC851IL.REL.008	Reliability	IP to Unified MeetingPlace Calls	Verify that calls from an Unified IP Phone to Unified MeetingPlace are successful.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IL. REL.009	Reliability	IP to Unity Connection Calls	Verify that calls from an Unified IP Phone to Unity Connection Voicemail are able to deposit and retrieve voicemail successfully.		Passed	
UC851IL. REL.010	Reliability	IP to Unity Calls	Verify that calls from an Unified IP Phone to Unity Voicemail are able to deposit and retrieve voicemail successfully.		Passed	

RSVP

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. RAC.006	RSVP	E2E RSVP with EMCC	Verify that an Extension Mobility Cross Cluster (EMCC) logged in phone can make an E2E RSVP call across a SIP trunk that has RSVP enabled.	TP->ABI CCM->MSP CCM->UC end point	Passed	
UC851IF. RAC.009	Call Admission Control	Failure of RSVP Reservation when using Mixed Trunks	Verifies that RSVP will fail for calls made using mixed trunks.	UC->ABI CCM->SIP Trunk->SME->H.323 Trunk->MSP CCM->UC	Passed	
UC851IF. RAC.010	Call Admission Control	SIP Preconditions and Fallback to Local RSVP	Verifies that fallback to local RSVP happens successfully for video calls.	Cisco UC->ABI CCM->SIP Trunk->MSP CCM->UC	Passed	
UC851IF. RAC.011	Call Admission Control	RSVP with Shared Line Video End Points	Verifies if RSVP for video calls work with shared lines between a main site phone and SRST phone.	TP->ABI CCM->ABI SRST1->UC; TP->ABI CCM->ABI SRST2->UC	Passed	
UC851IF. RAC.012	Call Admission Control	E2E RSVP with Session Management Edition	Verify that E2E RSVP between two Unified Communications Manager clusters passing through Session Management Edition.	TP->ABI CCM->Sip Trunk->MSP CCM->TP; TP->RSVPAgent1->ConfRSVPAgent->ConfBridge; UC->RSVPAgent2->ConfRSVPAgent->ConfBridge	Passed	

Service Advertisement Framework

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.001	Service Advertisement Framework for Distributed PSTN Fall Back	Unified Communications Manager Advertises SRST PSTN Prefix Information to SRST Sites in SAF Network	Verify that SAF CCD on Unified CM advertises the SRST sites DN ranges and its corresponding'To PSTN' prefix to the SRST sites via SAF enabled SIP trunk in the network by making a PSTN call from SRST 1 to SRST 2 when WAN connectivity from SRST sites to Unified CM are down (All branch routers in SRST mode). Verify the ability to make a PSTN call from SRST 2 to SRST 1, and by dialing VOIP number instead of direct PSTN Number from phone 1 in SRST1 to Phone 2 in SRST2, and from Phone 1 in SRST2 to Phone 2 in the SRST1 site.	Stage 1:Ph 1->SRST 1->PSTN->SRST 2->Ph 2; Stage 2: Ph 1->SRST 2->PSTN->SRST 1->Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.004	Service Advertisement Framework for Distributed PSTN Fall Back	Unified Communications Manager advertises SRST PSTN Prefix Information to other Unified Communications Manager Sites in SAF Network	1.Verify SAF CCD on the Unified CM1 advertises its SRST1 site DN ranges and its corresponding 'To PSTN' prefix to Unified CM2 cluster via SAF enabled SIP trunk in the network. 2.Verify if Phone 1 on Unified CM2 can call phone 2 on Unified CM1 via SAF enabled SIP trunk between Unified CM1 and Unified CM2. 3. Verify if a PSTN call can be made from Unified CM2 cluster to SRST site (in Unified CM1 site) when WAN connectivity between Unified CM1 and Unified CM2 is down and assuming that there is no connectivity problem between the Unified CM1 and SRST1 site. 4. Verify Phone 1 in Unified CM1 in SRST mode can call phone 2 in Unified CM2 via PSTN. 5. Bring back the WAN connection between the Unified CM1 and Unified CM2 and make a call from Phone 1 in SRST1 Unified CM1 to Unified CM2 central phone.	"Stage 1:Ph 1->Unified CM2->SAF enabled SIP Trunk->Unified CM1->Ph 2;Stage 2:Ph 1->Unified CM2->MGCP Gateway->PSTN->SRST 1->Unified CM1->Ph 2 Stage 3:Ph 1->Unified CM 1->SRST 1->PSTN->MGCP Gateway->Unified CM 2->Ph 1 Stage 5:Ph 1->Unified CM1->Unified CM2->Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.007	Service Advertisement Framework for Distributed PSTN Fall Back	Unified Communications Manager1 Advertises DN Information to Other Clients in the SAF Network in Load-Balancing Mode	Verify the SAF CCD on Unified CM1 on SAF network advertises its own site DN pattern and its reachability information to Unified CM2 cluster sites via SAF enabled SIP Trunk and H.225 enabled SAF between the Unified CM1 and Unified CM2. Verify the SAF CCD on Unified CM2 on SAF network advertises its own DN pattern and its reachability information to Unified CM1 cluster via SAF enabled SIP trunk. Verify if the Unified CM1 SAF trunks are used in load balancing mode by making first VOIP call from Unified CM2 to Unified CM1, by making second VOIP call from Unified CM2 to Unified CM 1, by making third VOIP call from the Unified CM2 to Unified CM1, and all calls are successful. Verify the End-to-End RSVP reservation in the above scenarios.	Stage 1:Ph 1->Unified CM2-> SIP precondition enabled SAF Trunk ->Unified CM1 ->Ph 2 Stage 2:Ph 1->Unified CM2 ->H.225 Trunk (SAF enabled Trunk)->Unified CM1 ->Phone2 Stage 3:Ph 1->Unified CM2->SIP precondition enabled SAF Trunk->Unified CM1 ->Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.011	Call Control Discover connection advertisement & learning	SAF - Forwarders is Redundancy Mode in Service Advertisement Framework	"Verify the ability to make the first VOIP call from Unified CM2 to Unified CM1, then bring the forwarders SAFF1 and SAFF3 down. Verify the Service Advertisement Forwarder redundancy functionality given that forwarders SAFF1 and SAFF2, SAFF3 and SAFF4 are acting in active/active mode for the Service Advertisement for Unified CM1 and Unified CM2 respectively. Modify the DN pattern on the Unified CM 1 and advertise it by CCD service on Unified CM1. Verify by making a second VOIP call from Unified CM2 to Unified CM1, making sure it uses the forwarders SAFF2 and SAFF4 to learn the DN patterns and reachability information, then bring back the SAFF1 and SAFF3 as Active mode, verify by making third VOIP call from Unified CM2 to Unified CM1 again make sure again it uses forwarders SAFF3 and SAFF4.	"Stage 1:Phone1->Unified CM2->SAF enabled Trunk->Unified CM1->Phone 2 Stage 2:Phone1->Unified CM2 (using Unified CM1 modified DN pattern by SAF)->SAF enabled Trunk->Unified CM1->Phone 2Stage 3: Phone 1->Unified CM 2->SAF enabled SIP Trunk->Unified CM 1->Phone 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.012	Service Advertisement Framework for TCO	Video Over SIP-SAF Trunks	Verify the SAF CCD on Unified CM1 advertisement reaches to Unified CM2 SAF enabled trunk between Unified CM1 and Unified CM2. Verify a video call from the Unified CM1 to Unified CM2, check the End-to-End RSVP reservation in this scenario, and then reduce the bandwidth on the Unified CM1 such that it should allow only one Video call. Verify the ability to make two consecutive video calls from Unified CM2 to Unified CM1 by reducing the bandwidth such that it allows only audio part of video call. Verify the ability to make a video call from Unified CM2 to Unified CM1 on reducing the bandwidth such that it rejects both the audio and video calls.	Video Ph 1->Unified CM1->SIP Trunk(RSVP enabled SAF Trunk)->Unified CM 2->Video Ph 2;Video Ph 2->Unified CM1->SIP Trunk(RSVP enabled SAF Trunk)->Unified CM2->Video Ph3;Video Ph 3->Unified CM2->SIP Trunk(RSVP enabled SAF Trunk)->Unified CM1->Video Ph4(only Audio part)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.016	Call Control Discover connection advertisement & learning	SAF - Forwarder Failure Results in Alternate PSTN Routing	Verify SAF Forwarder failure results in PSTN routing on Unified CM1 by SAF CCD service advertise its DN pattern and "To PSTN" reachability information through SAF enabled SIP trunk created between the Unified CM1 and Unified CM2. Verify SAFF1 is acting for the Service Advertisement for Unified CM1, and SAFF2 is acting for the Service Advertisement for the Unified CM2 by making first VOIP call from Unified CM2 to Unified CM1, and then bring down SAFF1 in the network ensuring that on Unified CM2 it marks the DN pattern it learned as down and starts the age-out timer. verify that after the Age-out Timer of DN pattern is learnt, second call from Unified CM2 to Unified CM1 is made, and it is routed through the PSTN network. Verify the ability to bring back SAFF1 in Unified CM1 site and modify the DN Pattern on Unified CM1 and advertise it in the SAF network, by making a VOIP call from Unified CM2 to Unified CM1.	Stage 1:Ph 1->Unified CM2->SAF enabled SIP Trunk->Unified CM 1->Ph 2 Stage 2:Ph 1 - Unified CM2->MGCP Gateway->PSTN->MGCP Gateway->Unified CM 1->Ph 2 Stage 3:Ph 1->Unified CM2->SAF enabled SIP Trunk->Unified CM 1->Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.022	Service Advertisement Framework for TCO RSVP Support for Cisco Call Admission Control	Unified Communications Manager Client Advertises its Own DN Pattern to Unified Communications ManagerE Client in SAF Network	Verify if Unified CM on Service Advertisement Framework (SAF) network advertises its own site DN pattern and its reachability information to other Unified CME sites through SIP preconditions enabled SAF Trunk by making a VOIP call from the Unified CME to Unified CM.	Variation 1 :Phone 1->Unified CME->SIP Precondition enabled SAF Trunk->Unified CM->Phone 2;Variation 2: Phone 1->Unified CM->SIP Precondition enabled SAF Trunk->Unified CME->Phone 2	Passed	
UC802EF.SAF.024	Service Advertisement Framework for TCO RSVP Support for Cisco Call Admission Control	Interworking of SAF Learnt Route and Static Route	Verify if static route interworks with the route learnt from the Service Advertisement Framework network for the same destined service on a Unified CM by making VOIP call from the Unified CM to Unified CME by varying the learnt partition and CSS on the Unified CM. Verify the interworking of SAF learnt route and static route on the Unified CME to Unified CM (vice versa call flow).	Variation1 : Phone 1->Unified CM->SIP Precondition enabled SAF Trunk->Unified CME->Phone 2;Variation2 : Phone 1->Unified CME->SIP Precondition enabled SAF Trunk->Unified CM->Phone 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.027	Service Advertisement Framework for TCO	Unified CME Client Advertises DN Pattern to Unified CME Client in the SAF Network Through SIP and H.225	Verify if a call can be made from a Unified CME registered to Unified SIP Proxy to a Unified CME client that is associated with H.225 Trunk in the Unified CM cluster. Verify when Unified CME1 is registered to Unified SIP Proxy and Unified CME2 to a CME gatekeeper, a call made fails initially since there is no route pattern in Unified CME1 to reach Unified CME2. Verify that after Unified CME1 advertises its modified DN range to Unified CME2 Cluster, a call made from Unified CME1 to Unified CME2 will work.	Variation1:Ph1->Unified CME1->SAF trunk->Unified CME2->Ph2 (advertisement);Variation 2: Ph1->Unified CME2->H.225 trunk->Unified CME1->Ph2	Passed	
UC802EF.SAF.030	Service Advertisement Framework for TCO	Unified CME Client Advertises its own DN pattern and "To PSTN" prefix to SRST Client in the SAF Network	Verify if Unified CME on Service Advertisement Framework (SAF) network advertises its own site DN pattern and "To PSTN" prefix and its reachability information to SRST client in the SAF network.	Unified CM is down: Variation 1:Phone 1->Unified CME->PSTN->SRST->Phone 2; Variation 2: Phone1->SRST->PSTN->Unified CME->Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.SAF.101	Service Advertisement Framework-CCD	Load Balancing of Calls to Remote Unified Communications Manager Cluster Advertising Same HostedDN From Unified Communications Manager Express	Verify that when a trunk in the advertising cluster is assigned to two Unified Communications Managers, Unified Communications Manager Express receives the same pattern twice, one for each Unified Communications Manager node. Verifies if calls from Unified Communications Manager Express to the advertised DN should alternate between the two Unified Communications Manager nodes.	Unified IP Phone->Unified CME->H.225 Trunk->ASA->Unified CM->ASA->Unified IP Phone	Passed	
UC802IF.SAF.103	Service Advertisement Framework-CCD	Co-Existing With Static Routes on Unified Communications Manager Express	Verify when Service Advertisement Framework is advertising a pattern that matches a statically configured Dial Peer, then routes from Unified Communications Manager Express to the advertised DN should be prioritized appropriately.	Unified IP Phone->Unified CME->H.225 Trunk->Unified CME->Unified IP Phone	Passed	
UC802IF.SAF.106	Service Advertisement Framework-CCD	Geophysical Location Based Advertisement and Service Advertisement Framework Redundancy	Verify that patterns are advertised based on the Hosted DN configuration in Clustering over WAN (CoW). Verify that redundancy is available for Service Advertisement Framework.	Unified IP Phone->Unified CME->H.225 Trunk->Unified CM->Unified IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.SAF.107	Service Advertisement Framework-CCD	PSTN Failover Due to Bandwidth Over Subscription on Calling Cluster	Verify the bandwidth control for calls over Service Advertisement Framework trunks are controlled using Locations based Call Admission Control (CAC). Verify that calls failover to PSTN when sufficient bandwidth is not available.	Unified IP Phone1->Unified CM->H.225 Trunk->Unified CM->Unified IP Phone2; Unified IP Phone1->Xfer->Unified CM->PSTN->Unified IP Phone3	Passed	
UC802IF.SAF.108	Service Advertisement Framework-CCD	PSTN Failover Due to Bandwidth Over Subscription on Called Cluster	Verify that the bandwidth control for calls over Service Advertisement Framework trunks are controlled using Locations based CAC. Verify that calls failover to PSTN when sufficient bandwidth is not available.	Unified IP Phone1->Unified CM->H.225 Trunk->Unified CM->Unified IP Phone2; Unified IP Phone3->Unified CM->PSTN->Unified IP Phone4	Passed	
UC802IF.SAF.109	Service Advertisement Framework-CCD	Service Advertisement Framework Aware H.323 Gateway Providing Unified SRST Services	Verify that Router when acting as a H.323 gateway can learn advertised DN's and route incoming calls based on the learnt pattern. Verify if the same router also routes calls to PSTN when the WAN link fails and the PSTN call is established based on the "prefix alias-dn-steering" configuration.	Unified IP Phone1->Unified CME->PSTN->H.323 GW->Unified CM->Unified IP Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.SAF.110	Service Advertisement Framework-CCD	Service Advertisement Framework Functioning on Unified SRST Router	Verify that calls to central site phone uses the PSTN even when the WAN link is up but the phones are still registered to the Cisco Unified Survivable Remote Site Telephony (Unified SRST) router because of connection monitor timer.	Unified IP Phone1->SRST->PSTN->Unified CM->Unified IP Phone2	Passed	
UC802IF.SAF.111	Service Advertisement Framework-CCD	Lost Connectivity Between Service Advertisement Framework Forwarders	Verifies the behavior when connectivity between two Service Advertisement Framework forwarders is lost, but clients are able to actively maintain connectivity with their forwarders.		Passed	
UC802IF.SAF.112	Service Advertisement Framework-CCD	Lost Connectivity Between Advertising Client and Service Advertisement Framework Forwarder	Verify the behavior when client lose connectivity to the Service Advertisement Framework (SAF) forwarders. Verify if any change to the advertised DN is pushed to the Forwarder once connectivity is restored.	Unified IP Phone->Unified CME->H.225 Trunk->Unified CM->Unified IP Phone	Passed	
UC802IF.SAF.113	Service Advertisement Framework-CCD	Age-Out Timer Expiry and PSTN Flush Out Timer Expiry	Verify that the patterns are marked down when connectivity to SAF Forwarder is lost. Verify DN or patterns are flushed out after age-out timer expiry and PSTN routes are also deleted after PSTN age-out expiry.	Unified IP Phone->Unified CME->H.225 Trunk->Unified CM->Unified IP Phone	Passed	
UC802IF.SAF.114	Service Advertisement Framework-CCD	Manual Summarization Using Unified Border Element	Verify that Unified Border Element can be used manually to summarize SAF Advertisements from one SAF AS and re-advertise these into another AS.	Unified IP Phone->Unified CM->H.225 Trunk->Unified CM->Unified IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.SAF.123	Service Advertisement Framework-CCD	Incoming SAF Call Terminated on Remote Destination Phone and Routed to Desk Phone	Verifies that an incoming call through SAF trunk can be answered on a mobile phone, it's remote destination. Verify the ability to move the call to Desk phone by disconnecting the call at mobile phone and resuming the call at Desk phone. Verify the ability to move call to and fro between desk phone and Mobile phone.	Unified Contact Center Express Node 1->ASA->Clustering over WAN->ASA->Unified Contact Center Express Node 2	Passed	
UC802IF.SAF.126	Service Advertisement Framework-CCD	Failure of Active Unified Communications Manager When SAF Calls are Active	Verify that when SAF calls are active, fail the active Unified Communications Manager by shutting down the port. Verify that the active calls stay and subsequent SAF calls are successful.	Unified Contact Center Express Node 1->ASA->Clustering over WAN->ASA->Unified Contact Center Express Node 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SAF.001	Service Advertisement Framework for TCO	Unified Communications Manager-Session Management Edition advertises the Leaf Cluster DN Information	Verify the SAF CCD on Unified Communications Manager-Session Management Edition advertises leaf clusters DN pattern and "To PSTN" prefix information to other leaf clusters through SAF enabled SIP trunk between Unified Communications Manager-Session Management Edition and leaf clusters in the SAF network.	Ph1->Unified CM2->SAF-SIP ICT->Unified CM-SME->SIP ICT->Unified CM1->Ph2; Ph1->Unified CM1->SAF-SIP trunk->Unified CM-SME->SIP ICT->Unified CM2->Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SAF.002	Service Advertisement Framework for TCO	Interworking of SAF Learnt Route and Static Route	Verify the SAF CCD on Unified Communications Manager-Session Management Edition advertises the leaf clusters DN pattern and its reachability information to other leaf clusters through SAF enabled SIP trunk between the Unified Communications Manager-Session Management Edition and leaf clusters in the SAF network. Verify Inter working of SAF learnt route and Static route on Unified Communications Manager-Session Management Edition by configuring static SIP trunk, to route the calls from Unified Communications Manager-Session Management Edition to Unified Communications Manager1 and from Unified Communications Manager-Session Management Edition to Unified Communications Manager2.	Stage3:Ph1(CSS->Static partition)->Unified CM2->SIP ICT->Unified CM-SME->SIP ICT->Unified CM1->Ph3;stage4:Ph1(CSS->SAF partition)->Unified CM2->SAF enabled SIP trunk->SME - Unified CM->SIP ICT->Unified CM1->Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SAF.003	Service Advertisement Framework for TCO	Unified Communications Manager-Session Management Edition advertises the Leaf Cluster DN Information- E2E RSVP support	Verify the SAF CCD on Unified CM-Session Management Edition advertises leaf clusters DN pattern and "To PSTN" prefix information to other leaf clusters through SIP precondition enabled SAF trunk between Unified CM-Session Management Edition and leaf clusters in the SAF network. Verify the ability to route the calls from Unified CM-Session Management Edition to Unified CM1 and from Unified CM-Session Management Edition to Unified CM2 on configuring Unified CM-Session Management Edition static SIP precondition enabled trunk.	Stage1:Ph1->Unified CM2->SIP precondition enabled SAF trunk->Unified CM-SME->static SIP precondition enabled trunk->Unified CM1->Ph2; Stage2:Ph1->Unified CM1->SIP precondition enabled SAF trunk->Unified CM-SME->static SIP precondition enabled trunk->Unified CM2->Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851 EF.SA F.004	Service Advertise ment Framework for TCO	DN advertisements on Unified Communications Manager-Sessio n Management Edition as well on Leaf cluster	Verify the SAF CCD on Unified CM-Session Management Edition advertises the leaf clusters DN pattern and "To PSTN" prefix information to other leaf clusters through SAF enabled trunk between Unified CM-Session Management Edition and leaf clusters in the SAF network. Verify the SAF CCD on leaf cluster advertises its own DN pattern and "To PSTN" prefix information to the SAF network.	Stage2:Ph1- >Unified CM1->SAF enabled SIP trunk->Unifi ed CM-SME-> SAF enabled SIP trunk->Unifi ed CM2->Ph2 Stage3:Ph1- >Unified CM1->SAF enabled SIP trunk->Unifi ed CM-SME-> SAF enabled SIP trunk->Unifi ed CM1->Ph2	Passed	
UC851 EF.SA F.005	Service Advertise ment Framework for TCO RSVP Support for Cisco Call Admission Control	DN advertisements on Unified Communications Manager-Sessio n Management Edition as well on Leaf cluster with E2E RSVP	Verify the SAF CCD on Unified CM-Session Management Edition advertises the leaf clusters DN pattern and "To PSTN" prefix information to other leaf clusters through SIP precondition enabled SAF enabled trunk between Unified CM-Session Management Edition and leaf clusters in the SAF network. Verify the SAF CCD on leaf cluster advertisements its own DN pattern and "To PSTN" prefix information to the SAF network.	Stage2:Ph1- >Unified CM1->SAF- SIP(QSIG) trunk(E2E)- >Unified CM-SME1- >SAF-SIP(q sig) trunk(e2e)-> Unified CM2->Ph;St age3:Ph1-> Unified CM1->SAF- SIP(QSIG)tr unk(E2E)-> Unified CM-SME1- >SAF-SIP(QSIG) trunk(E2E)- >Unified CM1->Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SAF.006	Service Advertisement Framework for Distributed PSTN Fall Back	SAF-PSTN fallback on SAF trunk-Unified Communications Manager-Session Management Edition Cluster Down	Verify the SAF CCD on Unified CM-Session Management Edition advertises leaf clusters DN pattern and "To PSTN" prefix information to other leaf clusters in the SAF network. Verify the SAF-PSTN fallback on SAF trunk, incase Unified CM-Session Management Edition cluster down.	Stage3:Ph1->Unified CM2->MGCP Gateway->PSTN N/W->MGCP Gateway->Unified CM1->Ph2; Stage4:Ph1->Unified CM1->MGCP Gateway->PSTN N/W->MGCP->Unified CM2->Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SAF.007	Service Advertisement Framework for Distributed PSTN Fall Back	PSTN fallback on Static SIP trunk-Unified Communications Manager-Session Management Edition Cluster Down	Verify the SAF CCD on Unified CM-Session Management Edition advertises the leaf clusters DN pattern and its reachability information to other leaf clusters through SAF enabled SIP trunk between the Unified CM-Session Management Edition and leaf clusters to the SAF network. Verify the ability to configure static SIP trunk on Unified CM-Session Management Edition to route the calls from Unified CM-Session Management Edition to Unified CM1 and from Unified CM-Session Management Edition to Unified CM2. Verify if both leaf clusters (Unified CM1 and Unified CM2) have pstn trunk as second option in the trunk route group, while the static SIP ICT as first preference, and set the static partition as first preference in the Calling Search Space-partition search order in both clusters (Unified CM1 and Unified CM2).	Stage1:Ph1->Unified CM2->static SIP ICT->Unified CM-SME(timeout);Stage2:Ph1->Unified CM2->MGCP Gateway->PSTN N/W->MGCP->Unified CM1->ph2; Stage3:Ph1->Unified CM1->static SIP ICT->Unified CM-SME(timeout);Stage4:Ph1->Unified CM1->MGCP Gateway->PSTN N/W->MGCP->Unified CM2-Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SAF.008	Service Advertisement Framework for Distributed PSTN Fall Back	SAF-PSTN fallback on SAF Trunk-Leaf Cluster Down-phones Registered to SRST	Verify the SAF CCD on Unified CM-Session Management Edition advertises leaf clusters DN pattern and "To PSTN" prefix information to other leaf clusters in the SAF network. Verify SAF CCD on leaf cluster advertises its own DN pattern and "To PSTN" prefix information to SAF network. verify if SAF CCD on Unified CM1 (leaf cluster) advertises the SRST1 DN pattern and "To PSTN' prefix to the SAF network. Verify SAF-PSTN fallback on SAF trunk when Leaf cluster is down and phones in the remote site are registered to SRST.	Stage3:Ph1->Unified CM2->SAF enabled SIP trunk->Unified CM-SME->SIP Gateway->PSTN N/W->SRST-1->Ph2;Stage4: Ph1->SRST-1->PSTN N/W->SIP Gateway->Unified CM-SME->Static SIP ICT->Unified CM2->Ph2	Passed	
UC851EF.SAF.009	Service Advertisement Framework for Distributed PSTN Fall Back	PSTN Fallback on Static Trunk-Leaf Cluster Down-phones Registered to SRST	Verify the SAF CCD on Unified CM-Session Management Edition advertises the leaf clusters DN pattern and its reachability information to other leaf clusters through SAF enabled SIP trunk between the Unified CM-Session Management Edition and leaf clusters in the SAF network. Verify PSTN fallback on static trunk- incase leaf cluster is down and phones are registered to SRST.	Stage3:Ph1->SRST-1->PSTN N/W->SIP Gateway->Unified CM-SME->Static SIP ICT->Unified CM2->Ph2;	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SAF.010	Service Advertisement Framework for Distributed PSTN Fall Back	SAF PSTN Fallback by Originating Cluster-Leaf Cluster Down -phones Registered to SRST	Verify the SAF CCD on Unified CM-Session Management Edition advertises the leaf clusters DN pattern and its reachability information to other leaf clusters through SAF enabled SIP trunk between the Unified CM-Session Management Edition and leaf clusters in the SAF network. Verify SAF PSTN fallback by the originating cluster when Leaf cluster is down and phones are registered to SRST.	Stage2:Ph1->Unified CM2->MGCP Gateway->PSTN N/W->MGCP Gateway->SRST-1->Ph1;Stage3:Ph1->SRST-1->SIP Gateway->Unified CM-SME->static SIP ICT->Unified CM2->Ph1	Passed	
UC851EF.SAF.011	Service Advertisement Framework for Distributed PSTN Fall Back	SAF-PSTN fallback on SAF trunk- Unified Communications Manager-Session Management Edition WAN link failure between Unified Communications Manager-Session Management Edition and Leaf Cluster	Verify the SAF CCD on Unified CM-Session Management Edition advertises leaf clusters DN pattern and "To PSTN" prefix information to the SAF network. Verify SAF-PSTN fallback on SAF trunk when WAN link fails between Unified CM-Session Management Edition and Leaf cluster. Verify SAF CCD on leaf cluster advertises its own DN pattern and "To PSTN" prefix information to SAF network.	Stage3:Ph1->Unified CM2->SAF enabled SIP trunk->Unified CM-SME->SAF enabled SIP ICT->Unified CM1->Ph2;stage4:Ph1->Unified CM1->SAF enabled SIP trunk->Unified CM-SME->SAF enabled SIP ICT->Unified CM2->Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SAF.012	Service Advertisement Framework for Distributed PSTN Fall Back	SAF to Non-SAF--PSTN fallback on Unified Communications Manager-Session Management Edition--WAN link failure	Verify the SAF CCD on Unified CM-Session Management Edition advertises the non-SAF Leaf Cluster(Unified CM4) DN pattern and "To PSTN" prefix information to the SAF network through SAF enabled SIP trunks between the Unified CM-Session Management Edition and other leaf clusters(SAF enabled).Verify SAF to non-SAF -PSTN fallback on Session Management Edition cluster when WAN connectivity fails between Unified CM-Session Management Edition and Unified CM4(non-SAF Leaf Cluster)	Stage5:Ph1->Unified CM1->SAF enabled SIP trunk->Unified CM-SME->static SIP trunk->Unified CM4(timeout);Stage6:Ph1->Unified CM1->SAF enabled SIP trunk->Unified CM-SME->SIP Gateway->PSTN N/W->MGCP Gateway->Unified CM4->Ph1	Passed	
UC851EF.SAF.013	Service Advertisement Framework for Distributed PSTN Fall Back	SAF to Non SAF- PSTN fallback on Originating cluster -- WAN link failure	Verify the SAF CCD on Unified CM-Session Management Edition advertises the non SAF leaf cluster(Unified CM4) DN pattern and "To PSTN" prefix information to the SAF network through SAF enabled SIP trunk between the Unified CM-Session Management Edition and other leaf clusters(SAF enabled).Verify the SAF to non-SAF ---PSTN fallback on originating cluster when the WAN connection failure between Unified CM-Session Management Edition and non-SAF Leaf cluster.	Stage5:Ph1->Unified CM1->SAF enabled SIP trunk->Unified CM-SME->static SIP trunk->Unified CM4(timeout);Stage6:Ph1->Unified CM1->MGCP Gateway->PSTN N/W->MGCP Gateway->Unified CM4->Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851 EF.SA F.014	Call Control Discovery: connection advertisement & learning	BLF indication on leaf cluster	Verify SAF CCD on Unified CM-Session Management Edition cluster advertises leaf clusters DN pattern and "To PSTN" prefix information to other leaf clusters in the SAF network. Verify BLF indication on the phones located in the leaf cluster.	Stage2:Ph1->Unified CM3->Ph2; Stage3:Ph1->Unified CM1->SAF enabled SIP trunk->Unified CM-SME1->Static SIP trunk->Unified CM-SME2->Static SIP trunk->Unified CM3->Ph2(Ph1 should get the BLF indication)	Passed	
UC851 EF.SA F.015	Q.SIG over Service Advertisement Framework SIP Trunk	Interworking of Q.SIG over SAF Trunks and static QSIG trunks	Verify the SAF CCD on the Unified CM advertises leaf clusters DN pattern and "To PSTN" prefix information to other leaf clusters in the SAF network. Verify interworking of Q.SIG over SAF trunks and Q.SIG over static trunks.	Stage3:Ph1->PBX1->QSIG PRI trunk->MGCP Gateway->Unified CM3->SAF enabled SIP trunk->Unified CM-SME->QSIG SIP ICT->Unified CM1->MGCP Gateway->QSIG PRI trunk->PBX->Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SAF.016	Q.SIG over Service Advertisement Framework SIP Trunk	Interworking of the QSIG tunneling over SAF trunks	Verify the interworking of QSIG over SAF enabled SIP trunk and QSIG over SAF enabled H.323 trunk. Verify supplementary services -Call back and call forwarding.	Stage1:Ph1->Unified CM1->SAF enabled SIP(QSIG) ICT->Unified CM-SME1->SAF enabled H.323(QSIG) ICT->Unified CM2->Ph2; Stage2:Ph1->Unified CM2->SAF enabled H.323(QSIG)ICT->Unified CM-SME1->SAF enabled SIP(QSIG)->Unified CM1->Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SAF.017	Q.SIG over Service Advertisement Framework SIP Trunk	Path Replacement and Blind Transfer over SAF Trunk	Verify path replacement on QSIG tunneled SAF enabled SIP trunks between leaf Unified CM clusters (via Session Management Edition cluster). Verify SAF CCD advertises its leaf clusters DN pattern and "To PSTN" prefix information to other leaf clusters via SAF enabled SIP trunk between Session Management Edition cluster and its leaf clusters, to the SAF network on Session Management Edition clusters (Unified CM-Session Management Edition1, Unified CM-Session Management Edition2).	Stage1:Ph1->Unified CM1->SAF enabled SIP (QSIG) trunk->Unified CM-SME1->static SIP(QSIG)ICT->Unified CM2-Ph1->CFB->Ph2->XFER->Unified CM2->SAF enabled SIP(QSIG) trunk->Unified CM-SME1->static SIP(QSIG) ICT->Unified CM-SME2->static SIP ICT->Unified CM3->Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SAF.018	Q.SIG over Service Advertisement Framework SIP Trunk	Path Replacement in Trombone Call	Verify Path replacement on QSIG tunneled SAF enabled SIP trunks through Session Management Edition cluster for a trombone call.	Stage1:Ph1->Unified CM1->SAF enabled SIP(QSIG) ICT->Unified CM-SME1->SIP(QSIG) ICT->Unified CM2->Ph1->CFNA->Ph2->XFER->Unified CM2->SAF enabled SIP (QSIG)ICT->Unified CM-SME1->SIP(QSIG) ICT->Unified CM1->Ph2	Passed	
UC851EF.SAF.019	Q.SIG over Service Advertisement Framework SIP Trunk	Interworking of QSIG PBX (ECMA and ISO) and SIP Gateway and call over SAF trunk	Verify that QSIG PBX call to a Unified Communications Manager over SAF enabled SIP(QSIG) ICT	Stage2:PBX Ph1->PBX(ISO)->QSIG PRI Trunk->SIP(QSIG) Gateway->Unified CM1(ISO)->SAF enabled SIP(QSIG) ICT->Unified CM-SME(ECMA)->static SIP(QSIG) ICT->Unified CM2(ECMA)->Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SAF.020	Q.SIG over Service Advertisement Framework SIP Trunk	Interaction of QSIG of Unified Communications Manager and DPNSS(PBX)	Verify the interaction of QSIG of Unified Communications Manager and DPNSS(PBX). Verify Callback with DPNSS PBX.	Stage2:Ph->DPNSS PBX->Vg30D(ISO)->SIP (QSIG)Gateway->Unified CM3(ECMA)->SAF enabled SIP (QSIG) ICT->Unified CM-SME2(ECMA)->static SIP(qsig) ICT->Unified CM-SME1(ECMA)->static SIP(QSIG) ICT->Unified CM1(ISO)->SCCP Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SAF.021	Q.SIG over Service Advertisement Framework SIP Trunk	Interworking of QSIG of Unified Communications Manager and ISR	Verify call forwarding scenario involving a call over SAF enabled SIP(QSIG) trunk, SIP Gateway and MGCP Gateway to pots.	Stage2: PSTN Ph1->MGCP(BRI/PRI)->Unified CM2->SAF-SIP(QSIG) ICT->Unified CM-SME1->Static SIP(QSIG) ICT->Unified CM1->SCCP Ph1->CFB->XFER->Unified CM1->SIP(BRI/PRI)->PSTN Ph1	Passed	
UC851EF.SAF.024	Call Control Discover connection advertisement & learning	Video over E2E RSVP Enabled SAF Trunk through Session Management Edition cluster	Verify the SAF CCD on Unified CM-Session Management Edition advertises leaf clusters DN pattern and "To PSTN" prefix information to other leaf clusters through SIP precondition enabled SAF trunk between Unified CM-Session Management Edition and leaf clusters in the SAF network. Verify video over E2E reservation over SAF trunk through Unified CM-Session Management Edition cluster.	Stage4: Video Ph1->Unified CM1->SAF(E2E)trunk->Unified CM-SME1->SIP (E2E) trunk->Unified CM2-Video Ph2 => audio call	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SAF.025	Q.SIG over Service Advertisement Framework SIP Trunk	SAF-PSTN Fallback on SAF Trunk-E2E RSVP Reservation Fails	Verify the SAF CCD on Unified CM-Session Management Edition advertises leaf clusters DN pattern and "To PSTN" prefix information to other leaf clusters through SIP precondition enabled SAF trunk between Unified CM-Session Management Edition and leaf clusters in the SAF network. Verify the SAF CCD on leaf cluster advertisements its own DN pattern and "To PSTN" prefix information to the SAF network. Verify SAF-PSTN fallback on SAF trunk- incase E2E RSVP reservation fails.	Stage3:Ph1->Unified CM1->SAF(E2E) trunk->Unified CM-SME1->SAF enabled SIP(E2E) trunk->Unified CM2->Ph2;stage4:Ph1->Unified CM1->SAF(E2E) trunk->Unified CM-SME1->SIP Gateway->PSTN->MGCP->Unified CM2->Ph2	Passed	
UC851EF.SAF.026	Service Advertisement Framework for TCO RSVP Support for Cisco Call Admission Control	Hold/Resume-on SAF Trunks with E2E RSVP Support	Verify the SAF CCD on Unified CM-Session Management Edition advertises the leaf clusters DN pattern and "To PSTN" prefix information to the leaf clusters through SIP precondition enabled SAF enabled trunk between Unified CM-Session Management Edition and leaf clusters in the SAF network. Verify SAF CCD on leaf cluster advertises its own DN pattern and "To PSTN" prefix information to SAF network. Verify MoH reservations and E2E Reservations are successful.	Stage1:Ph1->Unified CM1->SAF-SIP(QSIG)trunk(E2E)->Unified CM-SME1->SAF-SIP(QSIG) trunk(E2E)->Unified CM2->Ph2->Hold/Resume->SAF-SIP(QSIG)trunk(E2E)->Unified CM-SME1->SAF-SIP(QSIG) trunk(E2E)->Unified CM1-SCCP Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SAF.027	Service Advertisement Framework for TCO RSVP Support for Cisco Call Admission Control	Adhoc Conf on SAF Trunks with E2E RSVP Support	Verify the SAF CCD on Unified CM-Session Management Edition advertises the leaf clusters DN pattern and "To PSTN" prefix information to other leaf clusters through SIP precondition enabled SAF enabled trunk between Unified CM-Session Management Edition and leaf clusters in the SAF network. Verify SAF CCD on leaf cluster advertises its own DN pattern and "To PSTN" prefix information to SAF network. Verify the Adhoc conference call through Session Management Edition cluster.	Stage1:Ph1->Unified CM1-SAF-SIP(QSIG)trunk(E2E)->Unified CM-SME1->SAF-SIP(QSIG) trunk(E2E)->Unified CM2->Ph2->CONF->SAF-SIP(QSIG) trunk(E2E)->Unified CM-SME1->SAF-SIP(QSIG) trunk(E2E)->Unified CM1-SCCP ph2	Passed	
UC851EF.SAF.028	Service Advertisement Framework for TCO RSVP Support for Cisco Call Admission Control	QSIG Callback Testing on SAF trunks with E2E RSVP support	Verify the SAF CCD on Unified CM-Session Management Edition advertises the leaf clusters DN pattern and "To PSTN" prefix information to other leaf clusters through SIP precondition enabled SAF enabled trunk between Unified CM-Session Management Edition and leaf clusters in the SAF network. Verify SAF CCD on leaf cluster advertises its own DN pattern and "To PSTN" prefix information to the SAF network. Verify the following supplementary services on SAF trunks with E2E RSVP support.	Stage1:Ph1->Unified CM2->SAF-SIP(QSIG)trunk(E2E)->Unified CM-SME1->SAF-SIP(QSIG) trunk(E2E)->Unified CM1->Ph2; Stage2:Ph1->Unified CM1->SAF-SIP(QSIG) trunk(E2E)->Unified CM-SME1->SAF-SIP(QSIG) trunk(E2E)->Unified CM2->Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SAF.029	Service Advertisement Framework for TCO	Message Waiting Indicator on SAF trunks	Verify Message Waiting Indicator on QSIG tunneled SAF enabled SIP trunk through Session Management Edition cluster.	Stage1:Ph1->Unified CM1->SAF enabled SIP(QSIG) ICT->Unified CM-SME1->SIP(QSIG) ICT->Unified CM2->Ph1->CFNA->Static SIP(QSIG) ICT->Unified CM-SME1->Unity connection(message deposit);Ph1->Unified CM2->SAF enabled SIP(QSIG) ICT->Unified CM-SME1->Unity connection (message retrieval)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SAF.030	Service Advertisement Framework for TCO	SIP Early Offer on SAF Trunks	Verify the SAF CCD on Unified CM-Session Management Edition advertises leaf clusters DN pattern and "To PSTN" prefix information to other leaf clusters through SAF enabled SIP trunk between Unified CM-Session Management Edition and leaf clusters in the SAF network. Verify the ability to configure static SIP trunk on Unified CM-Session Management Edition to route the calls from Unified CM-Session Management Edition to Unified CM1 and from Unified CM-Session Management Edition to Unified CM2. Verify SIP early offer on SAF trunks.	Stage2:SCCP Ph1->CUCM1->SAF enabled SIP(qsig) ICT->CUCM-SME1->SIP(qsig) ICT->CUCM2->SCCP Ph2;Stage3; SCCP Ph1->CUCM1->SAF enabled SIP(qsig) ICT->CUCM-SME1->SIP(qsig) ICT->CUCM2->SIP Ph2	Passed	
UC851IF.SAF.001	Unified Communications Manager Session Management EditionService Advertisement Framework	Hold/Resume when Phone in Leaf Cluster A Calls Phone in Leaf Cluster B (Session Management Edition advertising through SIP)	Verify the ability to place a video call from leaf cluster A to leaf cluster B via a SAF trunk, given that the route is advertised by Session Management Edition via SIP with the route destination IP corresponding to the Session Management Edition cluster, and hold/resume the call.	Unified IP Phone->Unified CM A->SIP SAF Trunk->SME->SIP/H.323 trunk->Unified CM B->Unified IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.SAF.002	Unified Communications Manager Session Management EditionService Advertisement Framework	Initiate Conference when Phone in Leaf Cluster A Calls Phone in Leaf Cluster B (Session Management Edition advertising through H.323)	Verify the ability to place a video call from leaf cluster A to leaf cluster B via a SAF trunk, given that the route should be advertised by Session Management Edition via H.323 with the route destination IP corresponding to the Session Management Edition cluster. Verifies the ability to initiate a three way ad-hoc conference with another phone in the same cluster.	Unified IP Phone->Unified CM A->H.323 SAF Trunk->SME->SIP/H.323 trunk->Unified CM B->Unified IP Phone (Or conf bridge)	Passed	
UC851IF.SAF.003	Unified Communications Manager Session Management EditionService Advertisement Framework	Leaf cluster PSTN Fallback when Session Management Edition Unreachable (Session Management Edition advertising through SIP)	Verifies if the leaf cluster routes call through PSTN, on placing a call from leaf cluster A to leaf cluster B via a SAF trunk when the Session Management Edition node is unreachable, given that the route should be advertised by Session Management Edition via SIP with the route destination IP corresponding to the Session Management Edition cluster.	Unified IP Phone->Unified CM A->SIP SAF trunk->SME (timeout); Unified IP Phone->Unified CM A->PSTN gateway->PSTN->PSTN gateway->Unified CM B->Unified IP Phone	Passed	
UC851IF.SAF.004	Unified Communications Manager Session Management EditionService Advertisement Framework	Leaf Cluster PSTN Fallback when Session Management Edition Unreachable (Session Management Edition advertising through H.323)	Verifies if the leaf cluster routes the call via PSTN, on placing a call from leaf cluster A to leaf cluster B through a SAF trunk when the Session Management Edition node is unreachable, given that the route should be advertised by Session Management Edition via H.323 with the route destination IP corresponding to the Session Management Edition cluster.	2w	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.SAF.005	Unified Communications Manager Session Management EditionService Advertisement Framework	Session Management Edition PSTN Fallback when Leaf Cluster Unavailable (Session Management Edition advertising via SIP)	Verifies the ability to place a call from leaf cluster A to leaf cluster B through an Session Management Edition SAF trunk when the terminating node in leaf cluster B is unreachable. Verifies if the route should be advertised by Session Management Edition through SIP with the route destination IP corresponding to the Session Management Edition cluster, and Session Management Edition should route the call through PSTN.	Unified IP Phone->Unified CM A->SIP SAF Trunk->SME->SIP/H.323 trunk->Unified CM B (timeout); Unified IP Phone->Unified CM A->SIP SAF Trunk->SME->PSTN gateway->PSTN->PSTN gateway->Unified CM B->Unified IP Phone	Passed	
UC851IF.SAF.006	Unified Communications Manager Session Management EditionService Advertisement Framework	Session Management Edition PSTN Fallback when Leaf Cluster Unavailable (Session Management Edition advertising via H.323)	Verify the ability to place a call from leaf cluster A to leaf cluster B through an Session Management Edition SAF trunk when the terminating node in leaf cluster B is unreachable. Verifies if the route should be advertised by Session Management Edition through H.323 with the route destination IP corresponding to the Session Management Edition cluster, and Session Management Edition should route the call through PSTN.	Unified IP Phone->Unified CM A->H.323 SAF Trunk->SME->SIP/H.323 trunk->Unified CM B (timeout); Unified IP Phone->Unified CM A->H.323 SAF Trunk->SME->PSTN gateway->PSTN->PSTN gateway->Unified CM B->Unified IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.SAF.008	Service Advertisement Framework	Add New SAF Edge Forwarder (CME Site)	Verifies the ability to configure a new SAF forwarder located at a Unified Communications Manager Express location and join it to the SAF network, have it advertise new phone numbers, and verify that other SAF-enabled locations are able to place calls to the new site. Verify other learned routes are not affected by the joining of the new forwarder.		Passed	
UC851IF.SAF.009	Service Advertisement Framework	Remove SAF Edge Forwarder (CME Site)	Verify the routes to a SAF forwarder removed from the network are no longer advertised. Verify that other valid learned routes are not affected by the forwarder's removal.		Passed	
UC851IF.SAF.010	Service Advertisement Framework	WAN Link Between Edge and Transit Forwarder Flapping	Verify that ABI's learned routes are not affected on flapping the WAN link between edge and forwarder SAF routers at the ABI location. Verify that other locations' learned routes to ABI are not affected.		Passed	
UC851IF.SAF.011	Service Advertisement Framework	Failover between edge forwarder and redundant transit forwarder links	Verify that failover between edge forwarder and redundant forwarder works.		Passed	
UC851IF.SAF.012	Service Advertisement Framework Unified Communications Manager RSVP	RSVP SAF Call Between Cluster A and Cluster B, Transfer the Call	Verify the ability to place a video SIP SAF call between two clusters which are configured for RSVP. Verify on RSVP agents that the reservation is created end-to-end, the bandwidth in the phone locations is deducted. Verify if the RSVP reservation remains on call transfer to another phone.	Phone 1 (in RSVP location)->Unified CM A->SAF trunk->Unified CM B->Phone 2 (in RSVP location)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.SAF.013	Service Advertisement Framework Unified Communications Manager RSVP	RSVP SAF Call Through Session Management Edition (RSVP Enabled End-to-end), Leave a Voicemail	Verify the ability to place a video SIP SAF call between two clusters which are configured for RSVP through Session Management Edition. Verify on RSVP agents the reservation is created end-to-end and the bandwidth in the phone locations has been deducted. Verify if the phone call allow to ring out and leave a voicemail in the user's Unity Connection mailbox.	Phone 1 (in RSVP location)-> Unified CM A->SAF trunk->SME ->SIP Trunk->Unified CM B->Phone 2 (in RSVP location)	Passed	
UC851IF.SAF.014	Service Advertisement Framework Unified Communications Manager RSVP	Mixed Trunks with Local RSVP and SAF	Verify if local RSVP is enabled on each leaf cluster, on placing a SAF video call through Session Management Edition. Verify if one call leg is SIP, the other call leg is H.323, and local reservations are created on both sides.	Phone 1 ->Unified CM A->SIP SAF trunk->SME ->H.323 SAF trunk->Unified CM B->Phone 2	Passed	
UC851IF.SAF.015	Service Advertisement Framework	RSVP SAF Call to Unified Communications Manager Express and Call Transfer	Verify the ability to place a video SAF call from a Unified Communications Manager cluster to a Unified Communications Manager Express site. Configure RSVP end-to-end, and verify on RSVP agents that the reservation is created end-to-end and the bandwidth in the phone locations is deducted. verify the end-to-end reservations remain, on call transfer.	Phone 1->Unified CM A->SIP SAF Trunk->Unified CME->Phone 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.SAF.016	Service Advertisement Framework	RSVP SAF Call to Unified Border Element, Phone Set to CfwdAll	Verify the ability to place a SAF call from a Unified Communications Manager cluster to a Unified Communications Manager Express site through Unified Border Element (called phone is set to CFwdAll to another Unified IP Phone located at the CME site). Verify on RSVP agents that the reservation is created end-to-end and the bandwidth in the phone locations is deducted, on configuring RSVP end-to-end.	Phone 1->Unified CM A->SIP SAF Trunk->Unified Border Element->SIP SAF Trunk->Unified CME->Phone 2	Passed	
UC851IF.SAF.017	Service Advertisement Framework	H.323 SAF Call: Verify Local RSVP Reservations and Park the Call	Verify on each cluster RSVP agent that local RSVP reservations are created on placing a video H.323 SAF call between two clusters. Verify a reservation is not created on end-to-end. Verify reservation still remains on parking a call and have another phone in the same cluster answer.	Phone 1->Unified CM A->H.323 SAF Trunk->Unified CM B->Phone 2	Passed	
UC851IF.SAF.018	Service Advertisement Framework	End-to-end H.323 SAF Call via Session Management Edition, Verify Local RSVP Reservations	Verify local RSVP reservations are created on each cluster RSVP agent on placing a video H.323 SAF call between two leaf clusters through Session Management Edition. Verify a reservation is not created on end-to-end when call is on hold/resume.	Phone 1->Unified CM A->H.323 SAF Trunk->SME->H.323 trunk->Unified CM B->Phone 2	Passed	
UC851IF.SAF.050	Service Advertisement Framework	Resetting SAF Client Registrations through Unified Communications Manager	Verify that various command flows on the Unified Communications Manager do not adversely affect the client registration of the Unified Communications Manager with the SAF forwarder(s).		Failed	CSCtj65451

Session Management Edition

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF .SME.101	Session Management EditionE2E RSVP SIP Preconditions	E2E RSVP Calls Between Unified Communications Manager Leaf Clusters through Session Management Edition Cluster	Verify E2E RSVP calls between Unified Communications Manager leaf clusters through Session Management Edition cluster.	IP Phone->RSVP Agent->Cisco Unified Communications Manager 1 (Leaf)->SIP ICT (QSIG)->SME->SIP ICT (QSIG)->Cisco Unified Communications Manager 2 (Leaf)->RSVP Agent->IP Phone	Passed	
UC851EF .SME.102	Session Management EditionE2E RSVP SIP Preconditions	E2E RSVP Calls Between Dual-Stack Leaf Cluster and Other Leaf Cluster via Session Management Edition	Verify E2E RSVP calls between dual-stack leaf cluster and other leaf cluster via Session Management Edition.	IP Phone->Cisco Unified Communications Manager 1 (Dual Stack-Leaf)->SIP ICT (QSIG)->RSVP Agent->SME->SIP ICT (QSIG)->Cisco Unified Communications Manager 2 (Leaf)->RSVP Agent->IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SME.103	Session Management EditionE2E RSVP SIP Preconditions	E2E RSVP Calls Between Remote Phone from Dual-Stack Leaf Cluster to Phone on Another Leaf Cluster through Session Management Edition	Verify E2E RSVP calls between remote phones from dual-stack leaf cluster to a phone on another leaf cluster through Session Management Edition.	Remote IP Phone->Cisco Unified Communications Manager 1 (Dual Stack-Leaf)->SIP ICT (QSIG)->RSVP Agent->SME->SIP ICT (QSIG)->Cisco Unified Communications Manager 2 (Leaf)->RSVP Agent->IP Phone	Passed	
UC851EF.SME.104	Session Management EditionE2E RSVP SIP Preconditions	Reservation Failure in E2E RSVP Calls through Session Management Edition Clusters	Verify PSTN fallback in E2E RSVP calls through Session Management Edition clusters.	IP Phone->RSVP Agent->Cisco Unified Communications Manager 1 (Leaf)->SIP ICT (QSIG)->SME->SIP ICT (QSIG)->Cisco Unified Communications Manager 2 (Leaf)->RSVP Agent->IP Phone; IP Phone->Cisco Unified Communications Manager 1(Leaf)->SIP ICT (QSIG)->SME->PSTN Gateway->Cisco Unified Communications Manager 2 (Leaf)->IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SME.105	Session Management EditionE2E RSVP SIP Preconditions	Negative Scenarios in E2E RSVP Calls	Verify the negative scenarios in E2E RSVP calls like low bandwidth resulting in failure of RSVP reservations.	IP Phone->RSVP Agent->Cisco Unified Communications Manager 1 (Leaf)->SIP ICT (QSIG)->SME->SIP ICT (QSIG)->Cisco Unified Communications Manager 2 (Leaf)->RSVP Agent->IP Phone	Passed	
UC851EF.SME.106	Session Management Edition E2E RSVP SIP Preconditions	E2E RSVP supplementary call flows through Session Management Edition clusters	Verify E2E RSVP supplementary call flows through Session Management Edition clusters.		Passed	
UC851EF.SME.50	Session Management Edition	Callback over Session Management Edition	Verify callback over AnnexM1 and SIP trunks.	Ph1->Unified CM(leaf)->AnnexM1 ICT->SME->SIP(qsig)ICT->Unified CM(leaf)->ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.SME.001	Unified Communications Manager Service Advertisement Framework IME SME	Cluster A Calling Cluster B over SAF/SIP and IME through Session Management Edition	Verify the ability to place a video call from an 89/9900 Phone89/9900 Phone in cluster A to an 89/9900 Phone in cluster B via Session Management Edition. Verify if the call from cluster A traverses a SIP SAF trunk to Session Management Edition, the Session Management Edition connects the call to cluster B via IME, and hold/resume the call.	Unified IP Phone->Unified CM A->SIP SAF trunk->SME->IME trunk->Unified CM B->Unified IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.SME.002	Unified Communications Manager Service Advertisement Framework IME SME	Cluster A Calling Cluster B over SAF/H.323 and IME through Session Management Edition	Verify the ability to place a call from an RT Lite phone in cluster A to a 7985 in cluster B via Session Management Edition. Verify if the call from cluster A traverses an H.323 SAF trunk to Session Management Edition, and the Session Management Edition connects the call to cluster B via IME, when the called party transfers the call to another phone in the same cluster.	Unified IP Phone->Unified CM A->H.323 SAF trunk->SME->IME trunk->Unified CM B->Unified IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.SME.003	Unified Communications Manager IME SME	Cluster A Calling Cluster B Over SIP Trunk and IME through Session Management Edition	Verify the ability to place a video call from a 7985 endpoint in cluster A to an 89/9900 Phone in cluster B via Session Management Edition. Verify if the call from cluster A traverses a SIP trunk to Session Management Edition, the Session Management Edition connects the call to cluster B via IME, and the called party initiates a 3 way video conference with a 7985 in the same cluster.	Unified IP Phone->Unified CM A->SIP trunk->SME->IME trunk->Unified CM B->Unified IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. SME.004	Unified Communications Manager IME SME	Cluster A Calling Cluster B over H.323 Trunk and IME through Session Management Edition	Verify the ability to place a video call from a TNP endpoint in cluster A to an RT Lite phone in cluster B via Session Management Edition. Verify if the call from cluster A traverses an H.323 trunk to Session Management Edition, and then Session Management Edition connects the call to cluster B via IME, and the calling party transfers the call to another phone over a SIP trunk.	Unified IP Phone1->Unified CM A->H.323 trunk->SME->IME trunk->Unified CM B->Unified IP Phone2 after Xfer IP Phone3->Unified CM B->SIP trunk->Unified CM A->H.323 trunk->SME->IME trunk->Unified CM B->IP Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.SME.005	Unified Communications Manager IME SME	Session Management Edition-centralized IME Call, Calling Party Hands Off to Mobile Phone using Local PSTN Gateway	Verify the ability to place a call from an RT pro endpoint in cluster A to an RT Lite phone in cluster B via Session Management Edition. Verify if the call from cluster A traverses a SIP trunk to Session Management Edition, the Session Management Edition connects the call to cluster B via IME, and the called party then hands off the call to a mobile device.	Unified IP Phone->Unified CM A->SIP trunk->SME->IME trunk->Unified CM B->IP Phone; after hand-off Unified IP Phone->Unified CM A->SIP trunk->SME->IME trunk->Unified CM B->PSTN gateway->PSTN ->mobile phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. SME.007	Unified Communications Manager IME Session Management EditionUnified Contact Centre Express	Session Management Edition-centralized IME Call to CAD Agent	Verify the ability to place a call from a TNP endpoint in cluster A to a Unified CCX pilot number in cluster B via Session Management Edition. Verify if the call from cluster A traverses a H.323 trunk to Session Management Edition, and then Session Management Edition connects the call to cluster B via IME.	Unified IP Phone->Unified CM A->SIP trunk->SME->IME trunk->Unified CM B->CAD agent phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.SME.008	Unified Communications Manager IME SME	Session Management Edition-centralized IME Conference Call	Verify the ability to start a video call between two 89/9900 Phones in Cluster A. Verify if the call from cluster A traverses an SIP trunk to Session Management Edition, and the Session Management Edition connects the call to cluster B via IME on conferencing a third user in cluster B through Session Management Edition.	Video conf bridge->Unified CM A->SIP trunk->SME->IME trunk->Unified CM B->Unified IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. SME.009	Unified Communications Manager IME Session Management EditionUnity Connection	Session Management Edition-centralized IME Call, Leave Voicemail on UnityCon	Verify the ability to start a video call between two 89/9900 Phones in Cluster A. Verify if the call from cluster A traverses an SIP trunk to Session Management Edition, and the Session Management Edition connects the call to cluster B via IME on conferencing a third user in cluster B through Session Management Edition.	Unified IP Phone->Unified CM A->H.323 trunk->SME->IME trunk->Unified CM B->SIP trunk->Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.SME.010	Unified Communications Manager SME	H.323 Fast Start/SIP Early Offer Inter-op through Session Management Edition with 3-way Video Conference	Verify the ability to place a video call from a video phone in Cluster A to a phone in cluster B, place the call on hold and place another video call to cluster B. Verify if calls between cluster A and cluster B is routed through Session Management Edition, with H.323 fast start on one call leg, and SIP Early Offer on the second call leg. Verify fast start and early offer procedures are invoked.	Conf bridge->Unified CM A->H.323/fast start trunk->SME->SIP/early offer trunk->Unified CM B->Unified IP Phone1 Unified IP Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.SME.013	Unified Communications Manager Session Management EditionIME	IME Incoming Call to Session Management Edition Routed via H.323 Trunk to Pickup Group	Verify the ability to place an IME call to a call pickup line from cluster A to Session Management Edition. Verify that when the call arrives at Session Management Edition, it is routed to cluster B via an H.323 trunk, and the call is answered on one of the ringing phones in the group pickup.	Unified IP Phone->Unified CM A->IME->SME->H.323 trunk->Unified CM B->IP Phone after fallback: IP Phone->Unified CM A->PSTN Gateway->PSTN ->PSTN gateway->SME->H.323 trunk->Unified CM B->Unified IP Phone	Passed	
UC851IF.SME.014	Unified Communications Manager Session Management EditionIME	IME Incoming Call to Session Management Edition Routed through SIP Trunk to Shared Line, cBarge	Verify the ability to place an IME call to a shared line from cluster A to Session Management Edition. Verify when the call arrives at Session Management Edition, it is routed to cluster B via a SIP trunk, the call is answered and on the other shared line device, barge the call using the cBarge soft key.	Unified IP Phone->Unified CM A->IME->SME->SIP trunk->Unified CM B->IP Phone (conf bridge for cBarge) after fallback: IP Phone->Unified CM A->PSTN Gateway->PSTN ->PSTN gateway->SME->SIP trunk->Unified CM B->Conf bridge	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.SME.101	Unified Communications Manager Service Advertisement Framework IME SME	Call Park and Retrieve via Session Management Edition Secure SIP Trunk to Secure H.225 Trunk	Verify if a call placed from a Secure Phone in Cluster A to a Secure Phone in Cluster B through Session Management Edition (Secure SIP Trunk to Secure H.225 Trunk) is parked and then retrieved from a non-secure phone in cluster B.	Unified IP Phone->Unified CM A->SIPT->SME->H.225 ICT->Unified CM B->Unified IP Phone	Passed	
UC851IF.SME.102	Unified Communications Manager QSIG SME	Transfer to Hunt List via Session Management Edition Secure SIP QSIG Trunk to Secure SIP QSIG Trunk	Verify if a call placed from a secure phone in cluster A to a non-secure phone in cluster B through Session Management Edition (Secure SIP Trunk Secure SIP Trunk) is Blind Transferred to a Hunt List in Cluster A.	Unified IP Phone->Unified CM A->SIPT->SME->H.225 ICT->Unified CM B->Unified IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. SME.103	Unified Communications Manager Session Management EditionCME	Transfer from CME via Session Management Edition Secure SIP Trunk to Secure H.225 Trunk	Verify if a call placed from a secure phone in cluster A to a secure phone in Unified Communications Manager Express B through Session Management Edition (Secure SIP Trunk to Secure SIP Trunk) is Blind Transferred to secure phone in cluster C (Secure SIP Trunk to Secure H.225 ICT).	Unified IP Phone->Unified CM A->SIPT->SME ->SIPT->Unified BE->Cisco Unified SIP Proxy->SIPT-> Unified CME->Unified IP Phone	Passed	
UC851IF. SME.104	Unified Communications Manager Session Management EditionCME	Transfer from CME via Session Management Edition Secure SIP Trunk to Non-Secure Phone	Verify if a call placed from a secure phone in cluster A to a secure phone in Unified Communications Manager Express B through Session Management Edition (Secure SIP Trunk to Secure SIP Trunk) is consultatively transferred back to a non-Secure phone in cluster A.	Unified IP Phone->Unified CM A->SIPT->SME ->SIPT->Unified Border Element->Cisco Unified SIP proxy->SIPT-> Unified CME->Unified IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. SME.105	Unified Communications Manager SME	Conference between Session Management Edition Leaf Nodes while One Session Management Edition SUB is down in SRV	Verify the ability to establish a conference between Session Management Edition Leaf Nodes while 1 Session Management Edition SUB is down in SRV.	Unified IP Phone->Unified CM A->SIPT->SME ->SIPT->Unified CM B->Conf Bridge	Passed	
UC851IF. SME.106	Unified Communications Manager SME	Call Routes to PSTN via MGCP GW when not enough Bandwidth on IME Trunk	Verify if call routes to PSTN via MGCP GW when not enough bandwidth is available on IME trunk.	Unified IP Phone->Unified CM A->SIPT->SME ->PSTN->Unified CM B	Passed	
UC851IF. SME.107	Unified Communications Manager SME	Secure EMCC Phone Transfers to Centralized MGCP PSTN via Session Management Edition QSIG SIP Trunk	Verify if secure Extension Mobility Cross Cluster (EMCC) Phone consult transfers a call from a second secure EMCC Phone via Session Management Edition QSIG SIP trunk to an MGCP gateway in the centralized PSTN.	Unified IP Phone->Unified CM A->SIPT->SME ->PSTN	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. SME.108	Unified Communications Manager SME	Intercluster Call over Session Management Edition to Agent's Voice Mail is Re-queued to CRS-CSQ by Call Transfer Option in a Centralized Unity Connection	Verify if intercluster call over Session Management Edition SIP trunk to Unified CCX agent's voice mail is re-queued to Unified CCX trigger by Call Transfer option in a centralized Unity Connection.	Unified IP Phone->Unified CM 2->SIPT->SME->SIPT->Unified CM 1->Unified IP Phone	Passed	
UC851IF. SME.109	Unified Communications Manager SME	Leave Voice Mail on Cisco Unity Express after Call Transfer over Session Management Edition SIP Trunk	Verify the ability to leave voice mail on Cisco Unity Express after call is transferred over Session Management Edition SIP trunk to remote branch office.	Unified IP Phone->Unified CM 1->SIPT->SME->SIPT->Unified CM 2->Cisco Unity Express	Passed	
UC851IF. SME.110	Unified Communications Manager SME	Conference Chaining between Two Clusters via Session Management Edition SIP Trunk to H.225 Trunk	Verify the ability to chain two conference between two Clusters via Session Management Edition SIP trunk to H.225 trunk	Unified IP Phone->Unified CM 1->SIPT->SME->SIPT->Unified CM 2->Conf Bridge	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. SME.701	Unified Session Manager	Blind Transfer over DS SIPT's with Alternative Network Address Types (ANAT) (EO) enabled via Session Management Edition to a DS phone	Verify the ability to call with IPv6 media from local cluster to a remote cluster over DS SIP trunks via Session Management Edition blind transferred to a DS phone.	DS Unified IP Phone-->Unified CM->SIPT (DS IPv6 media IPv4 Sig Early Offer)-->SME-->SIPT-->(DS IPv6 media IPv4 Sig Early Offer)-->Unified CM->DS Unified IP Phone-->Xfer (Blind)-->DS Unified IP Phone	Passed	
UC851IF. SME.702	Unified Session Manager	Mid-call Video for an IPv6 Call through Session Management Edition Transferred to Cisco UC Integration™ for Microsoft Office Communicator	Verify that video is turned on for a call which is originally established as an audio only IPv6 call via Session Management Edition	DS Unified IP Phone w/CUVA -->Unified CM->SIPT (DS IPv6 media IPv4 Sig Early Offer)-->SME-->SIPT-->(DS IPv6 media IPv4 Sig Early Offer)->Unified CM->DS Unified IP Phone w/ CUVA->Xfer-> UC Integration™ for Microsoft Office Communicator	Passed	
UC851IF. SME.703	Unified Session Manager	Call to Hunt Pilot with Broadcast Distribution over Dual Stack SIPT through Session Management Edition	Verify that call to a hunt pilot with IPv4 only, dual stack and soft phones in the line group can be routed successfully through Session Management Edition.	DS Unified IP Phone -->Unified CM->SIPT (DS IPv6 media IPv4 Sig ANAT) -->SME -->SIPT -->(DS IPv6 media IPv4 Sig ANAT) -->Unified CM->Unified IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. SME.704	Unified Session Manager	Call Park Over Dual Stack SIP Trunks through Session Management Edition	Verify that call can be parked using an IPv4 Phone incoming over a DS SIPT from Session Management Edition and the parked call can be answered again using a DS phone.	DS Unified IP Phone -->Unified CM->SIPT (DS IPv6 media IPv4 Sig) -->SME -->SIPT -->(DS IPv6 media IPv4 Sig Early Offer) -->Unified CM->IPv4 Phone -->Park/Retrieve -->DS Unified IP Phone	Passed	
UC851IF. SME.706	Unified Session Manager	Adhoc Conference Involving Dual Stack SIP Trunk and ICT Trunks through Session Management Edition and a PSTN Phone	Verify that an adhoc conference can be placed successfully through Session Management Edition that involves a dual stack SIP trunk as well as an inter-cluster trunk and a PSTN phone where Session Management Edition is configured as centralized PSTN break out site.	Unified IP Phone1-->Unified CM-->SIPT (DS EO)-->SME-->ICT-->Unified CM-->Unified IP Phone2; Conference-->Unified IP Phone1-->Unified CM-->SIPT-->(DS EO)-->SME-->SIPT (DS EO)-->DS SIP GW-->PSTN	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. SME.707	Unified Session Manager	Hold/Resume Involving a Trusted Relay Point (TRP) and both Early Offer and Delayed Offer Trunks to Session Management Edition	Verify that a call is placed on hold and resumed when one trunk to Session Management Edition is enabled for early offer and the other trunk from Session Management Edition to the remote cluster is delayed offer.	Unified IP Phone1 -->Unified CM -->SIPT (DS IPv6 media IPv4 Sig Early Offer) -->SME -->SIPT -->Unified CM -->Unified IP Phone2; Unified IP Phone1 -->Hold;	Passed	
UC851IF. SME.708	Unified Session Manager	Consult Transfer of an IPv6 Call Over Early Offer and Delayed Offer Trunks to Session Management Edition	Verify that a call with IPv6 media can be transferred (Consult) when one trunk to Session Management Edition is enabled for early offer and the other trunk from Session Management Edition to remote the cluster is delayed offer	Unified IP Phone1->Unified CM->SIPT (DS Early Offer)->SME->SIPT (DS Delayed Offer)->Unified CM->Unified IP Phone2 -->Xfer (Consult)->Unified CM->SIPT (DS Delayed Offer)->SME->SIPT (DS Early Offer)->DS SIP GW->PSTN	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.SME.709	Unified Session Manager	Call Transfer to Mobile Network from Cisco Mobility Client through Session Management Edition Centralized PSTN Breakout	Verify that Cisco mobility client can transfer the call to the mobile phone through centralized Session Management Edition PSTN breakout when the phone registered to Unified Communications Manager is dual stack and the SIP GW is dual stack, given that the media for the call from the dual stack phone to the SIP GW is IPv6.	DS Unified IP Phone1->Unified CM->Mobile Client; Mobile Client->Xfer to Cell phone; DS Unified IP Phone1->Unified CM -->SIPT (DS IPv6 media IPv4 Sig Early Offer)->SME->SIPT (DS IPv6 media IPv4 Sig Early Offer)->PSTN->Cell phone	Passed	
UC851IF.SME.710	Unified Session Manager	Call from CME to Unified Communications Manager through Session Management Edition over a Dual Stack SIP Trunk Forwarded to another Cluster through Session Management Edition over ICT	Verifies to ensure that a call from Unified Communications Manager Express to Unified Communications Manager through Session Management Edition forwarded to a second cluster is successful.	Unified IP Phone1->Unified CME->SIPT->SME->SIPT (DS IPv6 media IPv4 Sig Early Offer)->Unified CM->Unified IP Phone2->CFA->Unified CM->SIPT (DS IPv6 media IPv4 Sig Early Offer)->SME->ICT->Unified CM->Unified IP Phone3	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.SME.711	Unified Session Manager	Encrypted Call Across Session Management Edition with Dual Stack SIP Trunks	Verifies to ensure that an encrypted call can be placed between two clusters through Session Management Edition where the SIP trunks between Session Management Edition and the clusters are dual stack.	Unified IP Phone1 -->Unified CM -->SIPT (DS IPv6 media IPv4 Sig Early Offer TLS Encrypted) -->SME -->SIPT (DS IPv6 media IPv4 Sig Early Offer Encrypted TLS) -->Unified CM -->Unified IP Phone2	Passed	
UC851IF.SME.751	Unified Session Manager	Call from PSTN through a Dual Stack SIP GW in Session Management Edition to Dual Stack Phone answered by Dual Stack Phone using PickUp	Verify that PSTN calls from Session Management Edition can be answered using PickUp softkey and the media negotiated is IPv6.	Unified IP Phone1 -->Unified CM -->SIPT (DS IPv6 media IPv4 Sig Early Offer TLS Encrypted) -->SME -->SIPT (DS IPv6 media IPv4 Sig Early Offer Encrypted TLS) -->Unified CM -->Unified IP Phone2	Passed	
UC851IF.SME.752	Unified Session Manager	Conference Call via Secure Early Offer SIP trunk to Secure H.323 ICT with Secure Centralized SIP Gateway	Verify that a secure call between an early offer secure SIP trunk and a secure H.323 ICT is able to conference in a party through a centralized secure SIP PSTN gateway.	Unified IP Phone1 -->Unified CM -->SIPT (Secure Early Offer) -->SME -->ICT (Secure) -->Unified CM -->Unified IP Phone2 -->SIPT (Secure) -->SME -->SIPT (Secure) -->PSTN	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.SME.753	Unified Session Manager	Encrypted Call Involving Secure MGCP Gateway in Session Management Edition	Verify that calls to PSTN via secure MGCP gateway in Session Management Edition is successful.	Unified IP Phone1 -->Unified CM -->SIPT (DS IPv6 media IPv4 Sig Early Offer) -->SME -->secure MGCP GW -->PSTN	Passed	
UC851IF.SME.754	Unified Session Manager	Secure Conference Involving Secure Phones and PSTN	Verify that a secure conference can be held between two Unified IP Phones in different clusters and a PSTN phone where the gateway is in Session Management Edition site.	Phn1->Unified CM->Sec Conf; Phn2->Unified CM->SIPT (DS EO Secure)->SME->SIPT (DS EO Encrypted)->Unified CM->Sec Conf; PSTN Phone->Secure SIP GW->SIPT (DS EO Encrypted)->SME->SIPT (EO Encrypted)->Unified CM->Sec Conf	Passed	
UC851IF.SME.755	Unified Session Manager	Invoking MTP in Session Management Edition Site for a Call from SIP GW due to Codec Mismatch	Verify that media resources can be invoked when there is a need in Session Management Edition site.	PSTN Phone -->SIP GW -->SME -->Xcoder -->UC Application	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. SME.756	Unified Session Manager	Call from PSTN through a H.323 GW in Session Management Edition to Unified CCX over an Early Offer Enabled SIP Trunk	Verifies to ensure that calls inbound from a H.323 gateway in Session Management Edition can be routed to Unified CCX successfully where the SIP trunk between Session Management Edition and Unified Communications Manager is configured with early offer.	PSTN Phone -->H.323 GW -->SME -->SIPT (DS IPv6 media IPv4 Sig Early Offer) -->Unified CM -->Unified CCX -->Unified CM -->Unified IP Phone Agent	Passed	
UC851IF. SME.771	Unified Session Manager	Early Offer Call to Unity Connection with SIP Integration in Session Management Edition	Verify that Unity Connection supports early offer calls with SIP integration.	Unified IP Phone -->Unified CM -->SIPT (Early offer) -->SME -->SIPT (Early Offer) -->Unity Connection	Passed	
UC851IF. SME.772	Unified Session Manager	Centralized Unity Connection in Session Management Edition - Supervise Transfer to a Phone Registered to Unified Communications Manager for a Call from PSTN	Verify that Unity Connection can supervise the transfer of a call from PSTN to a phone registered to Unified Communications Manager.	PSTN Phone -->SIP GW -->SME -->SIPT -->Unity Connection -->Supervise Xfer -->SME -->SIPT (DS IPv6 media IPv4 Sig) -->Unified CM -->Unified IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.SME.773	Unified Session Manager	Unity Connection in Session Management Edition Providing Voicemail Services to Phones in Two Clusters	Verify that Unity Connection in Session Management Edition can provide voicemail service to users in two Unified Communications Manager clusters.	Unified IP Phone1->Unified CM->Trunk->SME->Trunk->Unified CM->Unified IP Phone2->CFNA->Unified CM->Trunk->SME->Connection; Unified IP Phone2->Unified CM->Trunk->SME->Trunk->Unified CM->IP Phone1->CFB->Unified CM->Trunk->SME->Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.SME.774	Unified Session Manager	Unity Connection Secure SCCP Integration with Session Management Edition	Verify that Encrypted Calls can be placed to Unity in Session Management Edition from Phones Registered to Unified Communications Manager.	Unified IP Phone (secure) -->Unified CM -->SIPT (Early offer secure) -->SME -->SCCP (secure) -->Unity Connection -->Release transfer -->SME -->SIPT (Early Offer secure) -->Unified CM -->Unified IP Phone (secure)	Passed	
UC851IF.SME.775	Unified Session Manager	Unity Connection Secure SIP Integration with Session Management Edition	Verify that encrypted calls can be placed to Unity in Session Management Edition from phones registered to Unified Communications Manager.	Unified IP Phone (secure) -->Unified CM -->SIPT (Early offer secure) -->SME -->SIP (secure) -->Unity Connection -->Release transfer -->SME -->SIPT (Early Offer secure) -->Unified CM -->Unified IP Phone (secure)	Passed	

UC Integration

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CSF.002	CSF UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator with Unified IP Phones 9971/9951/8961 (deskphone) makes Video Call to Unified IP Phone 7985 in Another Cluster over QSIG ICT	Verify if a UC Integration™ for Microsoft Office Communicator in deskphone mode can make a video call to a Unified IP Phone 7985 over QSIG ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->QSIG ICT->Unified CM->7985	Passed	
UC802EF.CSF.003	CSF UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator with Unified IP Phones 9971/9951/8961 (deskphone) makes a Video Call to Unified IP Phones 9971/9951/8961 in Another Cluster over SIP ICT	Verify if a UC Integration™ for Microsoft Office Communicator in deskphone mode can make a video call to Unified IP Phones 9971/9951/8961 over SIP ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->SIP ICT->Unified CM->89/9900 Phone	Passed	
UC802EF.CSF.010	CSF UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator (softphone) on Remote Site in SRST Mode with Audio	Verify the behavior of UC Integration™ for Microsoft Office Communicator (softphone) on a remote site in Unified SRST mode with audio.	89/9900 Phone->Unified CM->Remote Branch->UC Integration™ for Microsoft Office Communicator (SRST)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CSF.012	CSF UC Integration™ for Microsoft Office Communicator	Visual Voicemail with UC Integration™ for Microsoft Office Communicator in Softphone Mode	Verify visual voicemail with UC Integration™ for Microsoft Office Communicator in softphone mode.	UC Integration™ for Microsoft Office Communicator->Unified CM->Unity Connection	Passed	
UC802EF.CSF.015	CSF UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator Video Call from Remote Phone to Central Site H.323 Endpoint	Verify UC Integration™ for Microsoft Office Communicator video call from remote phone to central site H.323 endpoint.	UC Integration™ for Microsoft Office Communicator->Remote Branch->Unified CM->H.323 Video Endpoint	Passed	
UC802EF.CSF.020	CSF UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator (Softphone) Video Call to H.323 MCU in Different Cluster over QSIG ICT	Verify UC Integration™ for Microsoft Office Communicator (softphone) video call to a H.323 MCU in a different cluster over QSIG ICT.	UC Integration™ for Microsoft Office Communicator (Softphone)->Unified CM->QSIG ICT->Unified CM->H.323-MCU	Passed	
UC802EF.CSF.021	CSF UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator (softphone) Video Call to H.323 Video Endpoint in Different Cluster over SIP ICT	Verify UC Integration™ for Microsoft Office Communicator (softphone) video call to H.323 video endpoint in different cluster over SIP ICT.	UC Integration™ for Microsoft Office Communicator (Softphone)->Unified CM->SIP ICT->Unified CM->H.323 Video endpoint	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CSF.023	CSF UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator Video Conference Call Across Different Clusters over SIP ICT	Verify UC Integration™ for Microsoft Office Communicator Video Conference Call Across Different Clusters over SIP ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->SIP ICT->Unified CM->UC Integration™ for Microsoft Office Communicator->CONF->UC Integration™ for Microsoft Office Communicator	Passed	
UC802EF.CSF.024	CSF UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator Video Call to Unified Video Advantage and IP Communicator in Different Cluster over QSIG ICT	Verify UC Integration™ for Microsoft Office Communicator video call to Unified Video Advantage and IP Communicator in different cluster over QSIG ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->QSIG ICT->Unified CM->CIPC + CUVA	Passed	
UC802EF.CSF.033	CSF UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator Video Call from Remote Phone to Central Site H.323 Endpoint	Verify UC Integration™ for Microsoft Office Communicator video call from remote phone to central site H.323 endpoint.	UC Integration™ for Microsoft Office Communicator->Remote Branch->Unified CM->H.323 Video Endpoint	Passed	
UC851EF.CSF.001	UC Integration™ for RTX	UC Integration™ (Softphone) for RTX Inter-cluster Video Call to Third Party Skinny Endpoint	Verify if UC Integration™ for RTX (softphone) from one cluster can make a video call to a third party Tandberg SCCP endpoint in another cluster over inter cluster trunks.	UC Integration™ for RTX->Unified CM1->SIP(QSIG) ICT->Unified CM2->SCCP video endpoint	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF .CSF.002	UC Integration™ for RTX	Escalation of Audio Call to Video Call in Softphone Mode When Calling Unified IP Phone 9900 Series is in Another Cluster.	Verify if UC Integration™ for RTX (softphone) from one cluster can make an audio call to Unified IP Phone 9900 series in another cluster and then can escalate to video.	UC Integration™ for RTX->Unified CM1->Annex M1 ICT->Unified CM2->89/9900 Phone	Passed	
UC851EF .CSF.003	UC Integration™ for RTX	Escalation of Audio Call to Video Call in Deskphone Mode When Calling UC Integration™ for Microsoft Office Communicator in Another Cluster	Verify if UC Integration™ deskphone) for RTX can make an inter-cluster call to UC Integration™ for MOC and can escalate the audio call to video call.	UC Integration™ for RTX->Unified CM1->SIP(QSIG) ICT->Unified CM2->UC Integration™ for Microsoft Office Communicator	Passed	
UC851EF .CSF.004	UC Integration™ for RTX	Inter-cluster Video Conference with Unified IP Phone 7985 and Unified IP Phone 9900/8900 series	Verify if UC Integration™ (softphone) for RTX can make an inter-cluster video call to Unified IP Phone 9900/8900 series and can join an Unified IP Phone 7985 in another cluster to the conference.	UC Integration™ for RTX->Unified CM1->Annex M1 ICT->Unified CM2->89/9900 Phone	Passed	
UC851EF .CSF.005	UC Integration™ for RTX	Hold/Retrieve from Shared Line with Unified IP Phone 8900/9900 Series in Another Cluster	Verify if UC Integration™ (deskphone) for RTX can make an inter-cluster video call to an Unified IP Phone 8900/9900 series ,can put the call on hold and retrieve it from a shared line.	UC Integration™ for RTX->Unified CM1->SIP (QSIG) ICT->Unified CM2->89/9900 Phone.	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.CSF.006	UC Integration™ for RTX	Fall Back to Unified SRST When Unified Communications Manager goes Down	Verify if UC Integration™ for RTX registers to Unified SRST when Unified Communications Manager goes down and if basic call functionality is available.	SCCP Phone 1->Unified CM->Remote branch->UC Integration™ for RTX (SRST)	Passed	
UC851EF.CSF.007	UC Integration™ for RTX	Voicemail in UC Integration™ for RTX (Softphone) Mode	Verify voicemail retrieval with Unity Connection and message waiting indication in UC Integration™ for RTX (softphone) mode.	UC Integration™ for RTX->Unified CM->Unity Connection	Passed	
UC851EF.CSF.008	UC Integration™ for RTX	Third party H.323 Endpoint with Gatekeeper Video Call to UC Integration™ (Softphone) for RTX	Verify if third party H.323 endpoint with gatekeeper can make a video call to UC Integration™ (softphone) for RTX.	H.323 video endpoint->Unified CM->UC Integration™ for RTX	Passed	
UC851EF.CSF.009	UC Integration™ for RTX	Inter Cluster Video Call to IP Communicator and Unified Video Advantage	Verify if UC Integration™ (softphone) for RTX can make an inter-cluster video call to IP Communicator and Unified Video Advantage.	UC Integration™ for RTX->Unified CM1->Annex M1 ICT->Unified CM2->Cisco IP Communicator + Cisco Unified Video Advantage	Passed	
UC851EF.CSF.010	UC Integration™ for RTX	Video Call from Remote Site UC Integration™ (softphone) for RTX to Central Site UC Integration™ for MOC.	Verify if UC Integration™ (softphone) for RTX in a remote site can make a video call to central site UC Integration™ for MOC.	UC Integration™ for RTX (Remote)->Unified CM->Central UC Integration™ for Microsoft Office Communicator	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.CSF.011	UC Integration™ for RTX	Central Site Unified Personal Communicator 8.0 Client Video Call to UC Integration™ (softphone) for RTX in Remote Site.	Verify if the central site Unified Personal Communicator 8.0 client can make a video call to UC Integration™ (softphone) for RTX in remote site.	Central Excession->Unified CM->UC Integration™ for RTX (Remote)	Passed	
UC851EF.CSF.012	UC Integration™ for RTX	UC Integration™ (Softphone) for RTX Call to PBX Phone in Interoperability Site	Verify if UC Integration™ (softphone) for RTX can make a call to a PBX phone in interoperability site.	UC Integration™ for RTX->Unified CM1->SIP ICT(QSIG)->Unified CM2->QSIG Trunk->PBX phone	Passed	
UC851EF.CSF.013	UC Integration™ for RTX	Audio Conference with QSIG PBX and PSTN Phones	Verify if UC Integration™ (softphone) for RTX can make a conference call with PSTN and QSIG PBX phone.	UC Integration™ for RTX> Unified CM->PSTN gateway->PSTN ->Conf->QSIG Trunk->PBX phone	Passed	
UC851EF.CSF.014	UC Integration™ for RTX	UC Integration™ for RTX Failover to PSTN When WAN is Down	Verify if UC Integration™ for RTX calls go through PSTN to another cluster when it detects that WAN is down or if there is insufficient bandwidth.	SCCP Ph1->Unified CM->MGCP PRI Gateway->PSTN ->UC Integration™ for RTX	Passed w/exception	CSCtj25015

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.CSF.015	UC Integration™ for RTX	Escalation of Audio Call to Video When call is Transferred to a Video Phone	UC Integration™ for RTX in remote site calls central SCCP phone, call is transferred to UC Integration™ for Microsoft Office Communicator after which video should be sent two ways.	UC Integration™ for RTX (remote)->Unified CM->SCCP Ph 1->Transfer->UC Integration™ for Microsoft Office Communicator	Passed	
UC851EF.CSF.016	UC Integration™ for RTX	PSTN Call to H.320 Endpoint from UC Integration™ for RTX	Verify if UC Integration™ for RTX can make a PSTN call to a H.320 Endpoint	UC Integration™ for RTX->Unified CM->PSTN gateway->PSTN->H.320 endpoint	Passed	
UC851EF.CSF.017	UC Integration™ for RTX	Inter-Cluster Call over Cisco IME	Verify if UC Integration™ for RTX can make an Intercluster Call to an Unified IP Phone 7985 over Cisco IME.	UC Integration™ for RTX->Unified CM 1->ASA->IME Trunk->ASA->Unified CM 2->7985	Passed	
UC851EF.CSF.018	UC Integration™ for RTX	Intercluser Video Call over Cisco IME After Transfer from UC Integration™ for MOC.	Verify if UC Integration™ for MOC can make an inter-cluster call to an SCCP phone which is then transferred to UC Integration™ for RTX in remote branch.	UC Integration™ for Microsoft Office Communicator->Unified CM1->ASA->IME Trunk->ASA->Unified CM2->SCCP ph1->Xfer->IME trunk->ASA->Unified CM1->Remote Branch->UC Integration™ for RTX	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.CSF.019	UC Integration™ for RTX	UC Integration™ for RTX Video Call Between Remote Sites	Verify if UC Integration™ for RTX in a remote branch can make a video call to UC Integration™ for RTX in another remote site.	UC Integration™ for RTX (Remote1)->Unified CM->UC Integration™ for RTX (Remote 2)	Passed	
UC851EF.CSF.020	UC Integration™ for RTX	Inter-Cluster Ad-Hoc Video Conference with IP Communicator and Unified Video Advantage and Third Party H.323 Endpoint	"Verify UC Integration™ for RTX Inter-Cluster Ad-Hoc Video Conference with IP Communicator and Unified Video Advantage			
and Third Party H.323 Endpoint.	"Cisco IP Communicator + Unified Video Advantage->Unified CM->UC Integration™ for RTX->Conf->H.323 video endpoint	Passed				
UC851IF.CSF.001	UC Integration™ for Microsoft Office Communicator	Single Sign-on: UC Integration™ for Microsoft Office Communicator with Access to all Servers on Log in to the Laptop using Secure LDAP Implementation	Verify that Single Sign-on feature works in a secure LDAP environment.	UC Integration™ for Microsoft Office Communicator -->Unified CM LDAP CUC MP	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.CSF.002	UC Integration™ for Microsoft Office Communicator	Single Sign-on: UC Integration™ for Microsoft Office Communicator user's password is reset and set to change the password on next user log in	Verify that Single Sign on feature works after the password is reset.	UC Integration™ for Microsoft Office Communicator -->Unified CM LDAP CUC MP	Passed	
UC851IF.CSF.007	UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator Tethered to RT-Std Phone and Setting up a Conference Call	Verify if an Adhoc video conference can be set up involving Tandberg video end points, RT-Pro phone, and Life-size video phone, when UC Integration™ for Microsoft Office Communicator is running on a laptop tethered to RT-Std phone, .	UC Integration™ for Microsoft Office Communicator -->Unified CM1----<sip> -->Unified CM2 -->MXE---TP	Passed	
UC851IF.CSF.009	UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator in Adhoc conference between LifeSize Video Terminal, Tandberg Video Terminal, and Sony Video terminal	Verify the ability of UC Integration™ for Microsoft Office Communicator to establish Adhoc conference between LifeSize Video Terminal, Tandberg Video Terminal, and Sony Video terminal.	UC Integration™ for Microsoft Office Communicator -->Unified CM1 -->MXE---->TP Excession Tandberg H.323 Sony	Failed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.CSF.012	UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator in Deskphone Mode in scheduled video conference with MXE in path	Verifies the UC Integration™ for Microsoft Office Communicator in deskphone mode tethered to a Guiness Phone in scheduled video conference with MXE in path.	UC Integration™ for Microsoft Office Communicator -->Unified CM1-->MP WebEx	Failed	CSCti85801
UC851IF.CSF.013	UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator in Desk Phone Mode and Control of Failed Call Attempt	Verifies the UC Integration™ for Microsoft Office Communicator in desk phone mode and control of desk phone when outgoing call is failed.	UC Integration™ Microsoft Office Communicator1 UC Integration™ for Microsoft Office Communicator2 UC Integration™ for Microsoft Office Communicator3-->WebEx	Failed	CSCti96812
UC851IF.CSF.014	UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator Parks an Incoming Active Call through IME Trunk and Retrieves the Call from an IPV6 Enabled Phone	Verify that supplementary services can be invoked using UC Integration™ for Microsoft Office Communicator	iPhone Client<--UC Integration™ for Microsoft Office Communicator1 <-----Unified CM1----<IME trunk>----Unified CM2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. CSF.016	UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator Answering Incoming Call Through Session Management Edition and Transfer to 89/9900 Phone Across SIP Trunk	Verifies the UC Integration™ for Microsoft Office Communicator answering incoming call through Session Management Edition and transferring to 89/9900 Phone across SIP trunk.	UC Integration™ for Microsoft Office Communicator1 <-----Unified CM1----<H.323 SME trunk>----Unified CM2	Passed	
UC851IF. CSF.017	UC Integration™ for Microsoft Office Communicator	Visual Voice Mail Indication in UC Integration™ for Microsoft Office Communicator and Downloading and Playing secure VM	Verify secure voicemail scan be downloaded using Visual Voicemail	UC Integration™ for Microsoft Office Communicator1-----Unified CM1----CUC	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. CSF.018	UC Integration™ for Microsoft Office Communicator	Visual VM indication in UC Integration™ for Microsoft Office Communicator and download and play VPIM VM	Verify voicemail received over VPIM can be downloaded and played	UC Integration™ for Microsoft Office Communicator1-----Unified CM1----CUC	Passed	
UC851IF. CSF.019	UC Integration™ for Microsoft Office Communicator	An IPV6 Enabled Phone parks an Incoming Call through Session Management Edition and the Call is retrieved from UC Integration™ for Microsoft Office Communicator	Verify if an IPV6 enabled phone parks an incoming call through SIP Session Management Edition and the call is retrieved from UC Integration™for Microsoft Office Communicator and normal video is resumed.	IPPhone<--UC Integration™ for Microsoft Office Communicator1 <-----Unified CM1----<SME trunk>----Unified CM2	Passed	

Unified Border Element

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.UBE.001	Unified Border Element - Media Antitromboning	Media Antitromboning	Verifies if using click to dial application transfers the call from application binary interface (ABI) to MSP via SIP trunk to Unified Border Element.	9971 - MSP Unified CM ---- SIP Trunk--- Unified Border Element --- SIP Trunk ----ABI Unified CM ---Call Transfer - MSP -- Unified CM 9971 --- SIP trunk --- Unified Border Element --- SIP trunk --- SME Unified CM ---- CTS 1000	Passed	
UC851IF.UBE.003	Unified Border Element - CUSM Interoperability	Call Statistics in SIP BYE for Unified Communications Manager CDR	Verify Cisco Unified Service Monitor (CUSM) is able to display MoS Score when SIP call traverses Unified Border Element.	9971 --- ABI Unified CM ---- Sip trunk ---Unified Border Element ---- SIP trunk --- MSP Unified CM ----ABI Unified CM -- CDR --- ABI Unified Service Monitor	Passed	
UC851IF.UBE.004	Unified Border Element - Media Flow	Support Media Flow around on a Unified Communications Manager Controlled SIP Call from a Branch Phone to PSTN	Verify SRST SIP Phone is able to talk to PSTN phone and media on Unified Border Element is set to Flow Around.	ABI-SRST--- SIP Phone --- SIP trunk --- Unified Border Element --- SIP trunk ---ABI Unified CM ----T1 PSTN	Passed	
UC851IF.UBE.005	Unified Border Element - Transcoding	Transcoding for SS to EO	Verify transcoding is invoked for codec mismatch for H.323 - SIP Call.	79xx - ABI Unified CM --- H.323 GW ----Unified Border Element --- SIP trunk --- Unified CME --79xx	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.UBE.006	Unified Border Element - Video Support	Video Escalation and de escalation	Verify if the audio can escalate to video when call is placed through Unified Border Element.	9971 ---MSP Unified CM -- SIP trunk ---- Unified Border Element --- SIP Trunk -- ABI --UC Integration™ for Microsoft Office Communicator	Passed	
UC851IF.UBE.007	PR1-A10	Non RSVP to RSVP and RSVP to Non RSVP Call Flows with Unified Border Element (EO-EO)	Verifies Non RSVP to RSVP and RSVP to Non RSVP call flows with Cisco Unified Border Element (Unified Border Element).	Unified CM1->Cisco Unified SIP Proxy->Unified Border Element-->Unified CME1; Unified CME1->Unified Border Element->Unified CME2	Passed	
UC851IF.UBE.008	PR1-A10	Non RSVP to RSVP and RSVP to Non RSVP Call Flows with Unified Border Element (DO-DO and DO-EO)	Verifies Non RSVP to RSVP and RSVP to Non RSVP call flows with Unified Border Element(DO-DO and DO-EO).	Unified CM1->Cisco Unified SIP Proxy->Unified Border Element-->Unfiied CME1; Unified CME1->Unified Border Element->Unified CME2	Passed	
UC851IF.UBE.009	PR1-A10	Non RSVP to RSVP and RSVP to Non RSVP Call Flows with Unified Border Element (FS-EO)	Verifies Non RSVP to RSVP and RSVP to Non RSVP call flows with Unified Border Element(DO-DO and DO-EO).	Unified CM1->Cisco Unified SIP Proxy->Unified Border Element-->Unified CME1; Unified CME1->Unified Border Element->Unified CME2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. UBE.010	PR1-A10	Non RSVP to RSVP and RSVP to Non RSVP Call Flows with Unified Border Element (FS-EO)	Verifies Non RSVP to RSVP and RSVP to Non RSVP call flows with Unified Border Element(DO-DO and DO-EO).	Unified CM1->Cisco Unified SIP Proxy->Unified Border Element-->Unified CME1; Unified CME1->Unified Border Element->Unified CME2	Passed	
UC851IF. UBE.011	PR1-A10	Voice Class Codec Transcoding and Handling Mid Call Codec Changes of Unified Border Element	Verifies voice class codec transcoding of Unified Border Element.	Unified CME->Unified Border Element-->SME	Passed	
UC851IF. UBE.012	PR1-A10	Mid Call Codec changes for Audio Calls Escalated to video calls with Unified Border Element	Verifies the mid call codec change.	UC->ABI CCM->Unified Border Element->ABI CCM->UC Integration™ for Microsoft Office Communicator	Passed	

Unified CM Business Edition

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF .SMB.085	Unity Connection	Voicemail for Remote Phones Over SIP/PRI PSTN Gateway	Verify voicemail deposit and retrieval for remote phones over SIP/PRI PSTN gateway.		Passed	
UC851EF .SMB.001	Unified Communications Manager	UC Integration™for RTX Transfers Call to IP Communicator	Verify that a remote Unified IP Phone 6900 series can call a central UC Integration™for RTX and CUCI-RTX UC Integration™for RTX can successfully transfer the call to a remote IP Communicator.	Rem1 RT-Lite Phone->Unified CM->Cen CUCI-RTX->XFER_B->Rem2 Cisco IP Communicator	Passed	
UC851EF .SMB.002	Unified Communications Manager	Central Cisco IP Communicator Consult Transfers Calls to Remote UC Integration™for RTX.	Verify that central IP Communicator can call central VG224 POTS phone and VG224 POTS Phone can successfully consult transfer to remote UC Integration™for RTX.	Cen Cisco IP Communicator->Unified CM->Cen VG224 POTS Ph1->XFER_C->Rem1 CUCI-RTX	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF .SMB.003	Unified Communications Manager	Adhoc Conferencing Using Unified IP Phone 7937, remote UC Integration™for RTX and remote Unified IP Phone 69xx.	Verify that adhoc conferencing is established among Central Unified IP Phone 7937, remote UC Integration™for RTX and remote Unified IP Phone 69xx using the centralized resources in 2901 PSTN gateway with remote UC Integration™for RTX initiating the adhoc conference.	Central 7937->Unified CM->Rem2 CUCI-RTX ; Rem2 CUCI-RTX-> CNF->Unified CM->Rem1 RT-Lite Phones	Passed	
UC851EF .SMB.004	Unified Communications Manager	Meet-me Conferencing Using Unified IP Phone 69xx, remote UC Integration™for RTX , remote Unified IP Phone 7937 , remote IP Communicator and VG224 POTS Phone	Verify that meet-me conferencing is established among Central Unified IP Phone 69xx series, remote UC Integration™for RTX and remote Unified IP Phone 7937 , remote IP Communicator and VG224 POTS Phone using centralized resources in the 2901 PSTN gateway.	Central VG224 POTS Ph1->meet-me Rem1 CIPC->meet-me	Passed	
UC851EF .SMB.005	Unified Communications Manager	PSTN Call from Remote Unified IP Phone 69xx series to the PSTN Phone Through Centralized 2901 MGCP PRI	Verify that the PSTN call from a remote Unified IP Phone 69xx series to PSTN Phone goes through the Centralized 2901 MGCP PRI (PSTN gateway).	Rem1 RT-Lite->Unified CM->Cen MGCP PRI->PSTN-> PSTN Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SMB.006	Unified Communications Manager	PSTN Call from Remote UC Integration™for RTX to PSTN Phone Through Centralized 2901 MGCP PRI	Verify that the PSTN call from a remote UC Integration™for RTX to the PSTN Phone goes through the centralized 2901 MGCP PRI (PSTN gateway).	Rem2 CUCI-RTX->Unified CM->Cen MGCP PRI->PSTN->PSTN Ph2	Passed	
UC851EF.SMB.007	Unified Communications Manager	Depositing Voicemail from PSTN Phone to remote UC Integration™for RTX Over Centralized 2901 PSTN Gateway	Verify that the PSTN phone calls remote UC Integration™for RTX phone over the centralized 2901 PSTN gateway which on CFNA goes to voicemail. Verify if PSTN Phone deposits the voicemail to remote UC Integration™for RTX and UC Integration™for RTX retrieves the voicemail.		Passed	
UC851EF.SMB.008	Unified Communications Manager	PSTN Call from PSTN Phone over Centralized 2901 PSTN Gateway to Remote Unified IP Phone 7937 and then CFB to VG224 POTS Phone	Verify that the PSTN phone calls the remote Unified IP Phone 7937 which on CFB goes to central VG224 POTS phone.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF .SMB.009	Unified Communications Manager	Hold/Resume on Shared Lines of UC Integration™for RTX and Unified IP Phone 7937.	Verify that the call from an Unified IP Phone 6900 phone to shared lines of UC Integration™for RTX and Unified IP Phone 7937. Verify the ability to hold the call from Unified IP Phone 7937 and resume the call from UC Integration™for RTX.	Rem1 RT-Lite Ph1-> Unified CM->Cen CUCI-RTX and 7937 (shared lines) 7937->answers and puts on hold Cen CUCI-RTX-> resumes	Passed	
UC851EF .SMB.010	Unified Communications Manager	Callback Notification from UC Integration™for RTX to IP Communicator	Verify that when the remote IP Communicator calls central UC Integration™for RTX and UC Integration™for RTX was busy on another call , remote IP Communicator presses callback and when UC Integration™for RTX becomes available callback notification is sent to remote IP Communicator.	Cen CUCI-RTX-> becomes available and callback notification is sent to Rem1 Cisco IP Communicator	Passed	
UC851EF .SMB.011	Unified Communications Manager	Extension Mobility on IP Communicator and Unified IP Phone 6900 series	Verify the extension mobility behavior on IP Communicator and Unified IP Phone 6900 series.	Rem Cisco IP Communicator->Unified CM->EM;	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.SMB.012	Unified Communications Manager	Auto Attendant Requests User to Dial Extension and Transfers Call to Matching Extensions	Verify that the remote UC Integration™for RTX dials the Auto Attendant DN and AutoAttendant requests the user to dial the extension and UC Integration™for RTX dials the Central IP Communicator and transfers the call successfully to Central IP Communicator.	Rem2 CUCI-RTX-> Unified CM->AA AA->request the user to dial the extension (Rem2 CUCI-RTX dials Cent CIPC DN) AA->XFER-> Cent Cisco IP Communicator	Passed w/ exception	CSCtj11088
UC851EF.SMB.013	Unified Communications Manager	AutoAttendant Plays Corresponding Greeting and Terminates Call for Calls Made After Business Hours	UC Integration™(soft phone) for RTX makes a conference call with PSTN and QSIG PBX phone.		Passed	
UC851EF.SMB.014	Unified Communications Manager	AutoAttendant Requests User to Re-enter Extension When no Matching Entry Found	UC Integration™ for RTX call should go through PSTN to another cluster when it detects WAN is down or of insufficient bandwidth.	Cen RT-Lite Ph1->Unified CM ->AA AA->request the user to dial the extension (Cen RT-Lite Ph1 dials invalid extension) AA->asks the user to re-enter the extension as there is no matching entry found	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF .SMB.015	Unified Communications Manager	AutoAttendant Support for Simultaneous Calls	"UC Integration™ for RTX in remote site calls central SCCP phone, call is transferred to UC Integration™ for MOC after which video should be sent two ways.	"Cen CUCI-RTX-> Unified CM->AA Rem1 Cisco IP Communicator->Unified CM->AA Rem2 RT-Lite Ph1->Unified CM->AA (check for AA supporting simultaneous calls)	Passed	
UC851EF .SMB.016	Unified Communications Manager	Immediate Response Handling by AutoAttendant	UC Integration™ for RTX makes a PSTN call to an H.320 endpoint	Rem2 CUCI-RTX-> Unified CM->AA AA plays the greeting and during that when a key is pressed - AA should process immediate key press responses.	Passed	
UC851EF .SMB.017	Unified Communications Manager	Check for UI on Browsers like IE 8.0 and Later, Firefox 3.5 and Later , Safari.	UC Integration™ for RTX makes an intercluster call to a 7985 over IME.	Navigate few pages and check for different browser support on IE 8.0 and later- Firefox 3.0 and later - safari browser	Passed	
UC851EF .SMB.018	Unified Communications Manager	Backup and Restore of Unified Communications Manager	"UC Integration™ for MOC makes an inter-cluster call to SCCP phone which is then transferred to remote branch UC Integration™ for RTX.	"Unified CM->Backup and Restore	Passed	

Unified CM Express

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.CME .108.1	Supplementary Service - Unified Border Element	Blind Transfer from the SIP Network to H.323 Network Involving IP-to-IP Gateway	Verify the ability to make a call from a Gatekeeper controlled Unified Communications Manager Express SCCP Unified IP Phone through an IP-to-IP Gateway (IPIPGW) to a Unified Communications Manager Express SIP Unified IP Phone via SIP trunk. The call transferred (blind) to a SIP Unified IP Phone registered to another Unified Communications Manager Express and transferred back (blind) to the originating Unified Communications Manager Express (the same cluster as called party) SCCP Unified IP Phone.		Passed	
SR60.CME .108.10	Supplementary Service	Release Transfer with Unity Connection integrated to Unified Communicati ons Manager Express.	Verify if a call from a Gatekeeper controlled Unified Communications Manager Express SCCP Unified IP Phone through an IP-to-IP Gateway to Unity Connection integrated to Unified Communications Manager Express is successful and that Unity Connection is able to transfer the call.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.CME .108.11	Supplementary Service	Call between H.323 and SIP Site via IP-to-IP Gateway involving a MGCP Gateway	Verify if a call from a PSTN phone to an Unified IP Phone registered to Unified Communications Manager Express can be call forwarded to a remote Unified Communications Manager Express phone, and be forwarded to Unity Express on no answer.		Passed	
SR60.CME .108.12	Supplementary Service	Call between H.323 and SIP Site via IP-to-IP Gateway involving a SIP Gateway	Verify if a call from a Unified IP Phone registered to Unified Communications Manager Express to a remote Unified Communications Manager Express phone can be transferred to a PSTN phone through a SIP Gateway.		Passed	
SR60.CME .108.13	Supplementary Service	Call from a PSTN Phone through a H.323 Gateway forwarded to a Shared Line	Verify if a call from a PSTN phone to an Unified IP Phone registered to Unified Communications Manager Express can be forwarded to a remote Unified Communications Manager Express phone configured for shared line.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.CME .108.13	Basic Call Flow	Call from PSTN Phone via H.323 Gateway Forwarded to Shared Line	Verify that a call from a PSTN phone to an IP Phone registered to Unified CME can be forwarded to a remote Unified CME phone configured for shared line.		Passed	
SR60.CME .108.2	Supplementary Service	Blind Transfer from H.323 Network to SIP Network involving IP-to- IP Gateway.	Verify the ability to make a call from a SIP Unified IP Phone registered to Unified Communications Manager Express via SIP Trunk and IP-to-IP Gateway to a Gatekeeper controlled Unified Communications Manager Express SCCP IP Unified IP Phone. The call transferred (blind) to a SCCP Unified IP Phone registered to another Unified Communications Manager Express and transferred back (blind) to the originating Unified Communications Manager Express (the same cluster as called party) SIP Unified IP Phone.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.CME.108.3	Supplementary Service	Consultative Transfer from SIP Network to H.323 Network involving IP-to-IP Gateway	Verify the ability to make a call from a Gatekeeper controlled Unified Communications Manager Express SCCP Unified IP Phone through a IPIPGW to a Unified Communications Manager Express SIP Unified IP Phone via SIP trunk. The call transferred (consultative) to a SIP Unified IP Phone registered to another Unified Communications Manager Express and transferred back (consultative) to the originating Unified Communications Manager Express (the same cluster as called party) SCCP Unified IP Phone.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.CME .108.4	Supplementary Service	Consultative transfer from H.323 network to SIP network involving IP-to-IP Gateway	Verify the ability to make a call from a SIP Unified IP Phone registered to Unified Communications Manager Express via SIP Trunk and IP-to-IP Gateway to a Gatekeeper controlled Unified Communications Manager Express SCCP IP Unified IP Phone. The call transferred (consultative) to a SCCP Unified IP Phone registered to another Unified Communications Manager Express and transferred back (consultative) to the originating Unified Communications Manager Express (the same cluster as called party) SIP Unified IP Phone.		Passed	
SR60.CME .108.5	Supplementary Service	Hold and Resume where the Call is Placed on Hold on the SIP Network	Verify if a call from a Gatekeeper controlled Unified Communications Manager Express SCCP Unified IP Phone through a IP-to-IP Gateway to a Unified Communications Manager Express SIP Unified IP Phone via SIP trunk can be placed on hold and resumed.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.CME.108.6	Supplementary Service	Hold and Resume where the call is placed on hold on the H.323 Network	Verify if a call from a SIP Unified IP Phone registered to Unified Communications Manager Express via SIP Trunk and IP-to-IP Gateway to a Gatekeeper controlled Unified Communications Manager Express SCCP Unified IP Phone can be placed on hold and resumed.		Passed	
SR60.CME.108.7	Unified CME Conference	Adhoc Conference involving IP-to-IP Gateway and Unified Communications Manager	Verify that an Adhoc conference can be established by a Unified Communications Manager Express SCCP Unified IP Phone with a SIP Unified IP Phone registered to Unified Communications Manager Express across an IPIPGW and a SIP Phone registered to the Cisco Unified Communications Manager.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.CME .108.8	Unified CME Conference	Adhoc Conference involving IP-to-IP Gateway and PSTN Phone	Verify that an Adhoc conference can be established by a Unified Communications Manager Express SIP Unified IP Phone with a SCCP Unified IP Phone registered to Unified Communications Manager Express across a IPIPGW and a PSTN Phone.		Passed	
SR60.CME .108.9	Supplementary Service	Call Forward across IP-to-IP Gateway involving two CMEs	Verify if a call from a Gatekeeper controlled Unified Communications Manager Express SIP Unified IP Phone through a IPIPGW to a Unified Communications Manager Express SIP Unified IP Phone via SIP trunk with 'Call Forward All' set on the called party to a SCCP Phone registered to the same Unified Communications Manager Express as the called party is successful.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC702EF.CME.005	Unified Communications Manager Express	Call Forward All from QSIG PBX on Unified Communications Manager Express to Cisco IP Manager Assistant (IPMA) Manager	Verify if a call from a SCCP phone in Unified CME via Unified Communications Manager to QSIG PBX phone on Unified Communications Manager is Call Forwarded All to a QSIG PBX phone connected to Unified Communications Manager Express which in turn has call forward busy to IPMA Manager.	SCCP Ph1->Unified CME->H.323 Trunk->GK->Unified CM->QSIG Trunk->PBX Ph1->CFA->QSIG Trunk->Unified CM->IP-to-IP Gateway(H.323)->Unified CME->QSIG Trunk->PBX Ph1->CFA->QSIG Trunk->Unified CME->IP-to-IP Gateway(H.323)->Unified CM->IPMA	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC702EF. CME.015	Unified Communications Manager Express	Ad-Hoc Conference on Unified Communications Manager Express with Digital Private Network Signaling System (DPNSS) PBX Phone and IPMA Manager Phone	Verify the ad-Hoc conference setup by local Unified Communications Manager Express Phone with DPNSS PBX phone and IPMA Manager Phone.	1: SCCP Ph1->Unified CME->IPIPGateway(H.323)->Unified CM->QSIG Trunk->Westell Gateway->DPNSS PBX Ph1 2: SCCP Ph1->Unified CME->CNF->Unified CME->IPIPGateway(H.323)->Unified CM->IPMA Manager 3: SCCP Ph1->Unified CME->CNF->DPNSS PBX Ph1 & IPMA Manager phone	Passed	
UC702EF. CME.028	Unified Communications Manager Express	Meet Me Conference between QSIG PBX Phone on Unified Communications Manager Express and DPNSS PBX Phone	Verify the Meet Me conference feature on Unified Communications Manager Express between two Unified Communications Manager Express phones (SCCP and QSIG PBX phone connected to Unified Communications Manager Express) and a DPNSS PBX phone.	1 : SCCP Ph1->Unified CME->CNF_MM 2: PBX ph1->QSIG Trunk->Unified CME->CNF_MM 3: DPNSS PBX Ph1->Westell->QSIG Trunk->Unified CM->IP-to-IP Gateway (H.323)->Unified CME->CNF_MM	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC702EF.CME.156	Unified Communications Manager Express	Shared line cBarge and Privacy support on Unified Communications Manager Express with Consult Transfer to Unified Communications Manager Video Phone	Verify if a call from QSIG PBX phone (Phone C) to SCCP Unified Communications Manager Express shared line phone (Phone A) is barged from another Unified Communications Manager Express shared line SCCP phone (Phone B) and if the call is consult transferred to a Unified Communications Manager video phone (Phone D) from Unified Communications Manager Express phone (Phone A).	1: PBX Ph1->QSIG Trunk->Unified CM->GK->IP-to-IP Gateway->GK->Unified CME->SCCP Phone1(SL) 2: SCCP Ph2 (SL)->Unified CME->cBarge->SCCP Ph1 (SL)->XFER_C->Unified Unified CME->GK->IP-to-IP Gateway->GK->Unified CM->SCCP Video Ph1	Passed	
UC851EF.CME.001	Unified CME	Unified IP Phone 9900/8900/6900 Series Support in Unified CME	Verify that RT-Lite Phone in Unified Communications Manager calls a CME RT-Lite Phone over H225 Trunk and then RT-Lite Phone on CME blind transfers the call to another CME 89/9900 Phone	RT-LitePh1->Unified CM->GK1->IP-to-IP Gateway->GK2->CME->RT-LitePh2->XFER_B->RTSIPPh1(Secure)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF. CME.002	Unified CME	Callpark Using Unified IP Phone 9900/8900 Series in Unified CME	Verify that RT-Lite Phone in Unified Communications Manager calls a CME RT Phone over H225 Trunk and then RT Phone on CME parks the call and the parked call is retrieved by the 89/9900 Phone in the same CME	RT-LitePh1->Unified CM->GK1->IP-to-IP Gateway->GK2->CME->RTSIPPh1->Parkscall RTSIPPh2->retrieves call	Passed	
UC851EF. CME.003	Unified CME	Unified IP Phone 9900/8900 Series Support in Unified CME	Verify that RT Phone in Unified Communications Manager calls a 89/9900 Phone in CME over CUSP and 89/9900 Phone in CME has CFWDALL to another 89/9900 Phone in CME	RTSIPPh1->Unified CM->SIPTRUNK->Cisco Unified SIP Proxy->Unified CME->RTSIPPh2->Cfwd ALL->Cisco Unified SIP Proxy->Unified CME->RT Lite	Passed	
UC851EF. CME.004	Unified CME	Adhoc Conferencing Using Unified IP Phone 9900/8900 Series in Unified CME	Verify that the adhoc conferencing is established between 89/9900 Phones in Unified Communications Manager, 89/9900 Phones in CME, RT-Lite Phones in CME and TNP Phones in CME. Adhoc conferencing is initiated by 89/9900 Phone in CME		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF. CME.005	Unified CME	Barge Using Unified IP Phone 9900/8900 Series in Unified CME	Verify that the barging is done successfully using 89/9900 Phones in CME when a call is made from Unified Communications Manager 89/9900 Phone over H225 Trunk. 89/9900 Phones are shared line phones.	RT SIP Ph3 ->Barges	Passed	
UC851EF. CME.006	Unified CME	Meet-me Conferencing Using Unified IP Phone 9900/8900 Series in Unified CME	Verify that meet-me conferencing is established using Unified IP Phone 9900/8900 series, Unified IP Phone 6900 series in Unified Communications Manager and Unified CME.		Passed	
UC851EF. CME.008	Unified CME	Media Flow Between Unified CM Unified IP Phones 9900/8900 After Call Transfer from Unified CME	Verify that media flows through Cisco Unified SIP Proxy Unified CME even after the call is transferred to Unified CME Unified IP Phones 9900/8900.	RTSIPPh1-> Unified CM->SIPTRUNK->Cisco Unified SIP Proxy->Unified CME->RTSIPPh2 RTSIPPh2-> CME->XFER_C->CUSP->SIPTRUNK->Unified CM->RTSIPPh3	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.CME.801	Unified Communications Manager Express	Direct Transfer across lines using RT Lite phone registered to Unified Communications Manager Express	Verifies to ensure that RT Lite can register with Unified Communications Manager Express and direct transfer across lines is supported.	RT Lite -->Unified CME -->PSTN	Passed	
UC851IF.CME.802	Unified Communications Manager Express	Authentication and encryption support for RT and RT Lite phones in Unified Communications Manager Express	Verifies if RT Lite and 89/9900 Phones can be registered to Unified Communications Manager Express, and the communication and media between the phones and Unified Communications Manager Express is secure.	RT Lite -->Unified CME -->89/9900 Phone	Failed	CSCtj77805
UC851IF.CME.803	Unified Communications Manager Express	Conference across lines using 89/9900 Phone in SRST mode	Verify if conference across lines can be performed from 89/9900 Phones registered to the SRST router support.	RT Lite -->Unified CME -->Conference	Passed	
UC851IF.CME.804	Unified Communications Manager Express	Authentication and encryption support for RT and RT Lite phones in SRST mode	Verifies to ensure that RT Lite and 89/9900 Phones can be registered to Unified Communications Manager Express and the communication and media between the phones and Unified Communications Manager Express is secure.	RT Lite -->Unified CME -->89/9900 Phone	Failed	CSCtj77805

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. CME.805	Unified Communications Manager Express	89/9900 Phone registered to Unified Communications Manager Express calling a phone registered to Unified Communications Manager through Session Management Edition and Unified Border Element	Verify if calls can be placed to a phone registered to Unified Communications Manager from a 89/9900 Phone registered to Unified Communications Manager Express through Session Management Edition and Unified Border Element.	89/9900 Phone -->Unified CME -->SIPT -->Unified Border Element -->SIPT -->SME -->SIPT -->Unified CM -->Unified IP Phone1 -->Xfer -->Unified IP Phone2	Passed	
UC851IF. CME.806	Unified Communications Manager Express	Shared Lines between RT Lite and 89/9900 Phones Registered to Unified Communications Manager Express	Verify that Unified Communications Manager Express can support shared lines between RT and RT Lite phones.	PSTN -->Unified CME -->RT and RT Lite Phones	Passed	
UC851IF. CME.807	Unified Communications Manager Express	Conferencing using an 89/9900 Phone in Unified Communications ManagerE-SRST mode	Verify that conferences can be initiated from an 89/9900 Phone when it is in fallback mode.	89/9900 Phone Unified IP Phone1 Unified IP Phone2 -->Unified CME -->Conference	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. CME.808	Unified Communications Manager Express	Placing a Call to PSTN using RT Lite Phone in SRST mode	Verify that PSTN call can be initiated from an 89/9900 Phone when it is in fallback mode.	RT Lite -->SRST -->PSTN	Passed	
UC851IF. CME.809	Unified Communications Manager Express	Placing a call to Unity Connection in Session Management Edition through Unified Border Element by entering FAC	Verify that calls can be allowed and restricted based on Forced Authorization Codes (FAC).	89/9900 Phone -->Unified CME -->SIPT -->SME -->SCCP -->Unity Connection	Passed	

Unified Communications Manager

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF. CCM.007	IPV6	Dual-stack Phone Operation Based on Media and Signaling Preference Settings	Verify a call from the Non DS cluster to the DS Phone on the DS cluster when Call Forward All (CFwdAll) is set to DS remote phone on the DS cluster over SIP Gateway(DS).	Stage1: SCCP Ph1->Unified CM->QSIG ICT->Unified CM(DS)->SCCP Ph2 (DS); Stage2: SCCP Ph2(DS)->CfwdD ALL->PSTN Gateway (SIP Gateway DS)->PSTN->PSTN Gateway (SIP Gateway DS)->Unified CM(DS)->Rem SCCP Ph3(DS)	Passed	
UC712EF. CCM.023	IPV6	Location Based CAC with IPv4/IPv6 Interworking	Verify if a user can place a call from a SCCP phone A to SIP Phone B which are in different locations and later initiate consultative transfer from SIP Phone B to SCCP Phone C with speech path established between Phone A & Phone B and consultative transfer fails with prompt " Not enough bandwidth".	Rem SCCP Ph A (DS) > Unified CM(DS) > SIP Ph B > Xfer_C > Unified CM> CAC > Rem SCCP Ph C	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF. CCM.024	IPV6	Intercom Call from SCCP (IPv4) to SCCP (DS) Phone	Verify is a user can make an intercom call from SCCP phone to another remote SCCP phone in different site when the target SCCP phone gets the tone and goes on speaker mode with audio muted, and after "talk" softkey is pressed two ways audio is established by the target phone.	Rem SCCP Ph 1 (v4) > Intercom > Unified CM(DS) > Rem SCCP Ph 2 (DS)	Passed	
UC712EF. CCM.027	IPV6	Unity Connection Call Failing Due to Unavailable Bandwidth	Verify that voice mail deposit can not be done when policy is mandatory and when there is not enough bandwidth available for unity connection system from target phone.	Rem SCCP Ph A (ds) > Unified CM(DS) > Rem SCCP Ph B > CFNA > Unified CM(DS) > CAC > Unity Connection VM#	Passed	
UC712EF. CCM.033	Unified CMBE 6000	Conferencing During Load Balancing Condition with IPv4/IPv6 Interworking	Verify if SIP IPv4 phone calls SCCP (DS) phone in one remote, SIP IPv4 phone calls SCCP (DS) phone in another remote site which is in active-active / active-standby mode. Verify when remote 3a goes down, connected call can be handled by remote 3b without any interruption. Verify that having established the call between SIP IPv4 and SCCP (DS), SIP IPv4 phone completes conference by pressing conf softkey.	SIP ipv4 > Unified CM > Rem SCCP ph	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CCM.005	Unified CMBE 6000	IM session on Unified IP Phones, Central CAD agent, Unified IP Phones 9971/9951/8961 and Unified Personal Communicator	Verify if IM from Unified IP Phones 9971/9951/8961 and SIP Phone acting as CAD agent in central sites to Unified Personal Communicator in central site is possible and also check the presence status of all buddies. Verify if IM from SIP/SCCP (IPPM clients) to Unified IP Phones 9971/9951/8961 and central site SIP Phone acting as CAD agent is possible and also check the presence status of all contacts.	Variation 1 :RT SIP Ph1->CUP->SCCP Ph/SIP Ph/Cisco Unified Personal Communicator Variation 2: SIP Ph1 (CAD)->Unified CUCM->UCCX->Unified CUCM->CUP->SCCP Ph/SIP Ph/CUPC	Passed	
UC802EF.CCM.009	Unified CMBE 6000	Inter-Working with Unified Attendant Server Console	"Verify the interworking of Unified Attendant Server and Unified Personal Communicator.	"PSTN->FXO->REM->Unified CMBE->ARC ->Unified CMBE->CUP->Unified Personal Communicator	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF. CCM.010	Unified CMBE 6000	Unified IP Phones 6921/6941/6961 in SRST mode	Verify the interworking of Unified Attendant Server and Unified Personal Communicator. PSTN call transfer from Unified Attendant Server to Unified Personal Communicator in desk phone mode through FXO. The call lands on the Unified Attendant Server and the Unified Attendant Server console checks the presence status of the user to which the call has to be transferred. Unified Personal Communicator in desk phone mode picks up the call.	Rem RT-Lite SCCP Ph1->SRST1->PSTN Gateway->PSTN->PSTN Gateway->SRST 2->Rem SCCP Ph2;Rem SCCP Ph2->CfwdALL->PSTN Gateway->PSTN Ph1	Passed	
UC802EF. CCM.012	Unified CMBE 6000	Unified IP Phones 9971/9951/8961 Interworking with Unified CCX	Verify if a call made from a PSTN phone to a SCCP phone is transferred to Unified CCX CAD agent with a Unified IP Phones 9971/9951/8961.	PSTN Ph1->MGCP PRI->Unified CMBE->SCCP Ph1->XFER_B->Unified CMBE->UCCX->Unified CMBE->RT SIP Ph1(CAD)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CCM.014	Unified CMBE 6000	Hunt List using Unified IP Phones 9971/9951/8961	Verify if a call made from a Central SCCP Phone to a Central Unified IP Phone 9971/9951/8961 is unanswered, the call goes to all members in the hunt group until it is answered depending upon the algorithm configured. Verify if the hunt group has a combination of Unified IP Phones 9971/9951/8961, Unified IP Phones 6921/6941/6961s and Unified Personal Communicator.	SCCP Ph1->Unified CMBE->RT SIP Ph1; RT SIP Ph1->Don't answer; The call Should go to all the members in the hunt group until it is answered depending upon the algorithm configured.	Passed	
UC802EF.CCM.015	Unified CMBE 6000	Depositing VM from Unified IP Phones 6921/6941/6961 using Unity Connection to Unified IP Phones 9971/9951/8961	Verify if a PSTN call can be made from the Unified IP Phones 6921/6941/6961 to a remote Unified IP Phone 9971/9951/8961 over H.323/SIP gateways.	Stage 1: RT-Lite SCCP Ph1->Unified CMBE->PSTN Gateway->PSTN->PSTN Gateway->Unified CMBE->Rem RT SIP Ph1->CFNA->UNC; Stage 2: RT-Lite SCCP Ph1->Deposits a VM; Stage 3: Rem RT SIP Ph1->Retrieves a VM	Passed	
UC802IF.CCM.021	Unified Communications Manager	iPhone Client Joining Unified MeetingPlace Internal Meeting by Receiving VOIP Call and Handoff Call to Deskphone	Verify that a iPhone client can join a Unified MeetingPlace hosted meeting by answering call back. When meeting in progress iPhone handoff the call to deskphone.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CCM.170	Unified Communications Manager	E911 Call Handling when Users Log in Across Cluster Using Extension Mobility Cross Cluster (EMCC) Feature	Verify if the E911 call from users logged in using EMCC feature in visiting Unified Communications Manager cluster is routed to local PSAP.		Passed	
UC802IF.CCM.170	Unified Communications Manager	E911 Call Handling When Users Logs in Across Cluster Using EMCC Feature	Verify that the E911 call from users logged in using EMCC feature in visiting Unified Communications Manager cluster is routed to local Public Safety Answering Point (PSAP).		Passed	
UC802IF.CCM.203	Unified Communications Manager	iPhone Client Receiving a Cisco IME Call and Setting Up Adhoc Conference	Verify that iPhone client can answer an incoming IME call and then setup an adhoc conference.		Passed	
UC802IF.CCM.204	Unified Communications Manager	iPhone Client Setting Up an IME Call and Transferring the Call to PSTN	Verify that iPhone client can set up an inter enterprise IME call and then transfer the call to a PSTN destination through H.323 gateway.		Passed	
UC802IF.CCM.205	Unified Communications Manager	Visual VM Indication at iPhone Client, Downloading and Playing VM Using Unity Connection	Verify that a iPhone client can receive Visual VM indication and can dial into Unity Connection VM server and check VM, and the DTMF is set for kpml.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CCM.207	Unified Communications Manager	iPhone Client Downloading and Playing Voice Profile for Internet Mail (VPIM) Forwarded and Replied Voice Mail Messages When Voice Mail Server is Cisco Unity Connection	Verify iPhone clients get Voice Mail alerts for Voice Profile for Internet Mail (VPIM) forwarded and replied voicemails and see visual Voice Mail indication and it can download and play these Voice Mail messages.		Passed	
UC802IF.CCM.211	Unified Communications Manager	Call Pick Up for iPhone Client	Verify if iPhone client and Unified IP Phones 9971, 9951, and 8961 are in same call pickup group. Verify that Unified IP Phones 9971, 9951, and 8961 can pick the call for iPhone client and vice versa (iPhone Client can pick up the call for Unified IP Phones 9971, 9951, and 8961 if supported).		Passed	
UC802IF.CCM.303	Cisco Unity Connection	Enhanced Message Waiting Indicator (eMWI) for Extension Mobility Across Cluster	Verify that Enhanced Message Waiting Indicator (eMWI) works for Extension Mobility across cluster.		Passed	
UC802IF.CCM.500.1	Unified Communications Manager	Unified Communications Manager Clustering over WAN (CoW) Deployment Model	Verify and validate the CoW Deployment Model.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CCM.500.2	Unified Communications Manager	Unified Communications Manager CoW Deployment Model	Verify and validate the CoW Deployment Model.		Passed	
UC802IF.CCM.500.3	Unified Communications Manager	Unified Communications Manager CoW Deployment Model	Verify and validate the CoW Deployment Model.		Passed	
UC802IF.CCM.500.4	Unified Communications Manager	Unified Communications Manager CoW Deployment Model	Verify and validate the CoW Deployment Model.		Passed	
UC802IF.CCM.525	Unified IP Phone	IP Phone 6911 and Unified Video Advantage Interworking	Verify that Unified IP Phone 6911 can interwork with Unified Video Advantage and use as a video phone.		Passed	
UC802IF.CCM.600	Unfied Communications Manager Extension Mobility Cross Cluster	Extension Mobility Cross Cluster Attend Home Cluster Unified MeetingPlace Conference, Primary Home Unified Communications Manager Failure	Verify that user is able to log into a visiting cluster using Extension Mobility Cross Cluster and attend a meeting place conference hosted on the home cluster and the home cluster's primary Unified Communications Manager fails.		Passed	
UCS712IF.CCM.101	Dual Stack Support	Blind Transfer of an IPV6 call to IPCCX Phone Agent	Verify the ability to call from a remote cluster across a dual stack SIP trunk to dual stack SCCP phone. Verify if the call is blind transferred from the SCCP phone to IPCCX with a dual stack phone as agent.	SCCP (v6/v4)->CUCM->SIPT (ANAT-on) (add mode v6/v4) (v6 sig pref/media pref)->CUCM->SCCP (v6/v4)->bxfer->IPCCX->SCCP Phone (v6/v4) IP Phone Agent	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF. CCM.106	Dual Stack Support	PSTN Call to a Dual Stack VG224 and Call Answered by Group PickUp	Verify the ability to call from PSTN to dual stack VG224. Verify if the call is answered by another phone using Group Pick Up.	PSTN Phone -->SIP GW (v6/v4) -->SIPT (v6 sig/media pref)-->Unified CM -->VG224 GW (v6/v4) -->FXS phone -->GPickup -->FXSPhone	Passed	
UCS712IF. CCM.107	Dual Stack Support	Call Over a Dual Stack SIPT Forwarded over ICT to Unified Communications Manager Express	Verify if the call is forwarded on busy to Unified Communications Manager Express over ICT, when a call is placed from a dual stack phone to another dual stack phone where the SIPT is configured to support v4 and v6 but the media and signaling preference has been set to v6.	SCCP Phone (v4/v6) -->Unified CM -->SIPT (v4/v6) -->Unified CM -->SCCP Phone (v4/v6) -->CFB -->ICT -->Unified CME -->SIP Phone	Passed	
UCS712IF. CCM.108	Dual Stack Support	Call to MOC client over a SIPT trunk with MTP required checked but ANAT enabled	Verify if a call is placed from a dual stack phone to a MOC client over a SIPT trunk with MTP required checked and ANAT is on.	SCCP (v6/v4) -->Unified CM -->SIPT (v4/v6) (v6 sig/media pref) (MTP) (ANAT on) -->Unified CM -->MTP -->MOC	Passed	
UCS712IF. CCM.114	Dual Stack Support	Call from a Dual Stack Phone to Cisco Unity Express AA Transferred to Another Dual Stack Phone	Verify if calls can be placed from a dual stack phone to CTI devices such as Cisco Unity Express.	SCCP (v6/v4) -->Unified CM -->CTI -->Cisco Unity Express -->AA -->Unified CM -->SCCP (v6/v4)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF. CCM.116	Dual Stack Support	CER Support for Dual Stack Endpoints	Verify that CER can discover dual stack phones and route 911 calls from those phones.	SCCP (v6/v4) -->Switch -->CER; SCCP (v6/v4) -->Unified CM -->CER -->GW -->PSAP	Passed	
UCS712IF. CCM.119	Dual Stack Support	Dual stack SCCP phone behind Unified SRST Gateway	Verify that dual stack phones can fallback to SRST router when the WAN link or connectivity to Unified Communications Manager is down. Verify that the dual stack phone can register back to Unified Communications Manager went it becomes reachable.	SCCP (v6/v4) (secure) -->Unified CM; WAN link Failure SCCP (v4) (Secure) -->SRST/SIPGW; Place a call; WAN link restored SCCP (v6/v4) (secure) -->Unified CM	Passed	

Unified Contact Center Express

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701IF.CRS.100.1	Unified Contact Center Express	Monitor Presence Status and Establish Chat Session with Session Management Edition (Testcase Id 1)	Verify if CAD Desktop can monitor presence status and establish chat session with non-agent Session Management Edition using Cisco Unified Personal Communicator (CUPC)/IPPM.		Passed	
UC701IF.CRS.100.1	Unified Contact Center Express	Monitor Presence Status and Establish Chat Session with Subject Matter Expert (Session Management Edition)	Verify that the CAD Desktop can monitor presence status and establish chat session with non-agent Session Management Edition using Unified Personal Communicator /IPPM.		Passed	
UC701IF.CRS.101.1	Unified Contact Center Express	Click to Dial, Transfer, Conference (Testcase Id 1)	Verify if CAD Desktop can click to dial, transfer or conference the Subject Matter Expert (Session Management Edition).		Passed	
UC701IF.CRS.101.1	Unified Contact Center Express	Click to Dial, Transfer, Conference	Verify that the CAD Desktop can click to dial, transfer or conference the Session Management Edition.		Passed	
UC701IF.CRS.103.1	Unified Contact Center Express	Personal Contacts (Testcase Id 1)	Verify if CAD desktop can show Cisco Unified Personal Communicator (CUPC) Agent's buddies as Personal Contacts in CAD Chat Selection Window.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701IF.CRS.103.1	Unified Contact Center Express	Personal Contacts	Verify that the CAD Desktop can show Unified Personal Communicator Agent's buddies as Personal Contacts in CAD Chat Selection Window.		Passed	
UC802EF.CRS.001	Unified Contact Center Express	Unified Contact Center Express with High-Availability	Verify Unified CCX with high-availability when Inbound call from a PSTN phone to a Cisco Agent Desktop agent with SCCP phone in the central site when one of the Unified CCX server is down.	PSTN Ph1->MGCP Gateway->Unified CM->Unified CCX->SCCP Ph1 (CAD)	Passed	
UC802EF.CRS.003	Unified Contact Center Express	Outbound Call from Cisco Agent Desktop SCCP Remote Phone over MGCP FXO	Verify if outbound call from a Cisco Agent Desktop SCCP phone in a remote site to a PSTN phone over MGCP FXO remote gateway is successful.	Unified CCX->Unified CM->SCCP Ph1 (CAD)->MGCP FXO->PSTN Ph1	Passed	
UC802EF.CRS.004	Unified Contact Center Express	Outbound Call from Cisco Agent Desktop Unified IP Phone 6941	Verify if an outbound call from a Cisco Agent Desktop Unified IP Phone 6941 to a PSTN phone is successful.	Unified CCX->Unified CM->6941 Phone(CAD)->MGCP PRI->Rem PSTN Ph	Passed	
UC802EF.CRS.009	Unified Contact Center Express	Outbound Call from a Cisco Agent Desktop SCCP Phone	Verify if an outbound call from a Cisco Agent Desktop SCCP phone in a remote site to a PSTN phone over SIP PRI remote gateway is successful.	Unified CCX->Unified CM->SCCP Ph1 (CAD)->SIP PRI->PSTN Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CRS.010	Unified Contact Center Express	Unified CCX with Unity Connection as Front-End	Verify if Unified CCX with Unity connection can function as front-end.	PSTN Ph1->MGCP PRI->Unified CM->Unity Connection->XFR->Unified CCX->9971 Phone (CAD)	Passed	
UC802EF.CRS.011	Unified Contact Center Express	Unified CCX with Cisco Unity as Back-End	Verify if Unified CCX with Cisco Unity can function as back-end.	PSTN Ph1->MGCP PRI->Unified CM->Unified CCX->SCCP Ph1->CFNA->Unity	Passed	
UC802EF.CRS.012	Unified Contact Center Express	QSIG PBX Phone Call to Unified CCX Agent through ICT	Verify if the call from a QSIG PBX phone to Unified CCX through ICT is transferred to a Cisco Agent Desktop with an SCCP phone in the central site.	PBX Ph1->QSIG Trunk->Unified CM->ICT (QSIG)->Unified CM->Unified CCX->Unified CM->6941 Phone (CAD)	Passed	
UC802EF.CRS.017	Unified Contact Center Express	Agent Gets Stuck in Reserved When Call is Blind-Transferred from Cisco Unity to Unified CCX	Verify if the agent gets stuck in Reserved when a call is blind-transferred from Cisco Unity to Unified CCX.	SCCP Ph1->Unified CM->Unity->XFR_B->Unified CCX->SCCP Ph2 (CAD)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CRS.017	Unified Contact Center Express	Agent Gets Stuck in Reserved When Call is Blind-Transferred from Cisco Unity to Unified CCX	Verify if the agent gets stuck in Reserved when a call is blind-transferred from Cisco Unity to Unified CCX.	SCCP Ph1->Unified CM->Unity->XFR_B->Unified CCX->SCCP Ph2 (CAD)	Passed	
UC802EF.CRS.018	Unified Contact Center Express	Agent Cannot Hang Up Call After Transfer to an Extension	Verify if an agent is not able to hang up a call after transferring to an extension.	PSTN Ph1->MGCP PRI->Unified CM->Unified CCX->SCCP Ph1 (CAD)->XFR_C->SCCP Ph2	Passed	
UC802EF.CRS.020	Unified Contact Center Express	Non-ICD Transferred Call to Agent Disconnected from Source	Verify if a non-ICD transferred call to an agent gets disconnected from source side.	SCCP Ph1->Unified CM->ICT (QSIG)->Unified CM->SCCP Ph1 (CAD)	Passed	
UC802EF.CRS.022	Unified Contact Center Express	Extension Mobility of Unified IP Phone A in Unified IP Phone Agent B	Verify if Unified IP Phone Agent A can perform an Extension Mobility in IP agent B to accept the queued call.	Stage 1:9971 Ph (CAD)->EM->SCCP Ph1(CAD);Stage 2:SIP Ph->Unified CM->Unified CCX->9971 Ph(CAD)	Passed w/exception	CSCti77146
UC802IF.CRS.105.1	Unified Contact Center Express	Validate Island Mode Operation of Unified Contact Centre Express	Verify and validate the island mode moderation of Unified Contact Centre Express deployed in High Availability over WAN when Unified Contact Centre Express node is active due to WAN link failure.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. CCX.001	Outbound Dialer	Predictive Outbound IVR Based Campaign Call Classification	Verifies that Predictive Outbound Interactive Voice Response (IVR) correctly classifies voice, answering machine (touchtone and unity).	Unified Contact Centre Express->Unified CM->PSTN gateway->CustomerPhone (CPA determines voice)->Inline Transfer->Outbound Trigger->IVR	Passed	
UC851IF. CCX.002	Outbound Dialer	Predictive IVR Campaign Leaves Message in Unity Phone	Verifies that Predictive IVR campaign leaves a message in Unity for 89/9900 Phone.	Unified Contact Centre Express->PSTN GW->MSP Unified CM->89/9900 Phone->Unity connection	Passed	
UC851IF. CCX.003	Outbound Dialer	Predictive IVR campaign failover in COW deployment	Verifies the Predictive IVR campaign failover in Cow deployment.	Unified CCX Dialer->Unified CM->PSTN Gateway->Customer Phone (CPA determines voice)->Inline Transfer->Unified Contact Centre Express IVR	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.CCX.004	Outbound Dialer	Call Classification by the dialer when IVR Ports are Exhausted	Verifies the call classification by the dialer when IVR ports are exhausted.	Unified CCX Dialer->Unified CM->PSTN Gateway->Customer Phone (CPA determines voice)->Inline Transfer->Unified CCX IVR	Passed	
UC851IF.CCX.005	Outbound Dialer	Verifies IVR based Outbound dialer can make calls to PSTN through Session Management Edition	Verifies that IVR based Outbound dialer can make calls to PSTN through Session Management Edition.	ABI Unified CCX->ABI CCM->SIP Trunk->SME->PSTN->Customer	Passed	
UC851IF.CCX.007	Outbound Dialer	IVR Predictive/Progressive Outbound dialer dialing out customers through SIP Trunk	Verifies that Unified CCX is associated with Unified Communications Manager Cluster1, when all Agents are in Cluster 2. Verifies if an outbound campaign is initiated to the contact in Enterprise 2. The call is placed via SIP Trunk.	ABI Unified CCX->ABI CCM->SIP Trunk(MTP Required)->MSPCCM->Unified IP Phone	Passed	

Unified MeetingPlace

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.MP.106.1	Unified MeetingPlace	Test Endpoints Audio CODECS With Software Media Server	Verify that a user can establish meeting place conferences with WebEx as the web conference provider utilizing a software media server. Join the conference using multiple endpoint models configured to different supported audio CODECS.		Passed	
UC802IF.MP.107.1	Unified MeetingPlace	Test Endpoints Audio CODECS With Hardware Media Server	Verify that a user can establish a meeting place conference with WebEx as the web conference provider utilizing a hardware media server. Join the conference using multiple endpoint models configured to the different supported audio codecs.		Passed	

Unified Mobility

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713IF. CUM.001	Unified Mobility	Secure VM Download when VM Servers and Unity Connection in Active-Active Mode	Verify if an iPhone Unified Mobile Communicator client can download the Voicemail from paired Unity Connection server when the Voicemail servers and Unity Connection is in Active-Active mode.		Passed	
UC713IF. CUM.002	Unified Mobility	Secure VM Download when VM Servers and Unity Connection in Digital Networking Mode	Verify if an iPhone Unified Mobile Communicator client can download Voicemail when the Voicemail servers are Unity Connection in Digital networking mode.		Passed	
UC713IF. CUM.007	Unified Mobility	Unity Connection in Active-Active Configuration	Verify if an iPhone Unified Mobile Communicator client can download a reply Voicemail from Unity Connection in active-active mode with one of them down.		Passed	
UC713IF. CUM.015	Unified Mobility	Callback on Unified MeetingPlace Only Meeting	Verify that Unified Mobile Communicator gets the meeting list for all types of meetings and call back works for Unified MeetingPlace only meetings.		Passed	

Unified PC

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.EXC.004	Unified Presence	Cisco Unity: Visual Voicemail Capabilities for Nonsecure Messages	Verify for a Cisco Unity subscriber that message notifications are received when a new nonsecure voicemail arrives, that the user can retrieve and listen to the voicemail, perform operations on the message, and then call the user back who left the voicemail.		Passed	
UC802IF.EXC.009	Unified MeetingPlace Unified CM Unified PC Unified Presence	Soft Phone Escalates Incoming Call to Video and Merge with Outgoing Video Call	Verify if a user can set up an audio call from a SIP IP Phone 9900 series video endpoint in another cluster via SIP trunk to Unified Personal Communicator 8.0 in soft phone mode. Verify the ability to escalate the call to video, set up an outgoing video call from the same Unified Personal Communicator 8.0 client to SCCP (Unified Video Advantage) endpoint, and check for bi-directional voice path given that both the calls merge.		Passed	
UC802IF.EXC.010	Unified MeetingPlace Unified CM Unified PC Unified Presence	Soft Phone Escalate Incoming Call to Video and Merge with Incoming Video Call	Verify if a user can set up an audio call from a SIP IP Phone 9900 series video endpoint in another cluster via H.323 trunk to Unified Personal Communicator 8.0 in soft phone mode. Verify the ability to escalate the call to video, set up an incoming video call from another SIP endpoint (IP Phone 7985) to the same Unified Personal Communicator 8.0 client, and check for bi-directional voice path given that both the calls merge.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.EXC.014	Unified CM Unified Mobility Unified PC Unified Presence	Dusting with Soft Phone Mode, Hand-Off to Mobility Device	Verify if a call can be handed off to a mobility device from Unified Personal Communicator 8.0 soft phone.		Passed	
UC802IF.EXC.022	Unified CM Unified Personal Communicator Unified Presence	Soft Phone Escalate Incoming Call to Video and Merge with Incoming Video Call	Verify if a user can set up an audio call from a SIP video endpoint in another cluster through SIP trunk to Unified Personal Communicator 8.0 in soft phone mode. Verify the ability to escalate the call to video, set up an incoming video call from another SIP endpoint (7985) to the same Unified PC 8.0 client, and check for bi-directional voice path given that both the calls merge.		Passed	

Unified Presence

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60. CUP.0 22	Interoperability	Extension Mobility Wireless Cisco Unified IP Phone 7921 Setting Do Not Disturb (DND)	Verify that the presence status of a user is available when he or she is logged into an Extension Mobility wireless Cisco Unified IP Phone 7921 and the Unified CallManager administrator sets Do Not Disturb (DND).		Passed w/ Exception	Unified IP Phone 7921 Wireless Phone does not support Extensi on Mobilit y Cross Cluster (EMCC)
UC802 EF.CU P.002	Presence IM	Call from Unified Personal Communicator in Desk Phone to UC Integration™ for Microsoft Office Communicator	Verify CUCI MOC and Unified Presence interworking	Unified Personal Communicator->Unified Presence->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	
UC802 EF.CU P.003	Unified Presence Node Failure	Failover of Unified Presence Server and Unified Communications Manager Node in Cluster	Verify failover in Unified Presence cluster	RT/Video Ph->REM->WAN->Unified Presence->Unified CM->SCCP Ph/SIP Ph/ Unified Personal Communicator	Passed	
UC802 EF.CU P.005	Unified Presence/RSVP	Video phones, Unified IP Phones 9971/9951/8961, Unified Personal Communicator Make Video Calls with RSVP	Verify Video call with RSVP enabled between Cisco Unified Personal Communicator and Remote 89/9900 Phone	RT/Video Ph->REM->WAN->Unified Presence->Unified CM->SCCP Ph/SIP Ph/ Unified Personal Communicator	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CUP.008	IM Presence in ARC	Interworking with Unified Attendant Server	Verify presence and call transfer with Unified Presence and ARC	Stage1: PSTN->FXO->REM->Unified CM->ARC;Stage2: ARC->Unified CM->Unified Presence>CUPC/Rasputin	Passed	
UC802EF.CUP.402	Presence in ARC	Interworking with Unified Attendant Server	Verify presence and call transfer with Unified Presence and ARC		Passed	
UC802EF.CUP.403	Cisco Unified Personal Communicator with Video	Video phones, Unified IP Phones 9971/9951/8961, Unified Personal Communicator 8.0 Make Video Calls with RSVP	Verify Video call with RSVP enabled between Cisco Unified Personal Communicator and Remote 89/9900 Phone		Passed	
UC802EF.CUP.407	Cisco Unified Personal Communicator with Video	Third Party Video Interoperability	Verify Video call		Passed	
UC802EF.CUP.500	Cisco Unified Personal Communicator and Unity connection	Unified Personal Communicator 8.0 and Unity Connection	Verify Voice mail for Cisco Unified Personal Communicator endpoint		Passed	
UC802EF.CUP.501	Cisco Unified Personal Communicator and Unity connection	Unified Personal Communicator 8.0 and Unity Connection	Verify Voice mail for Cisco Unified Personal Communicator endpoint		Passed	
UC802EF.CUP.804	Cisco Unified Personal Communicator with Video	Video Phones, Unified IP Phones 9971/9951/8961, Unified Personal Communicator 8.0 are in Video Conference	Verify video call		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.CUP.001	Cisco Unified Presence	Cisco Unified Presence Upgrade	Verify that the sub-cluster logical topology remains intact on upgrading two Cisco Unified Presence cluster peers from 8.0 to 8.5, and that HA is NOT enabled immediately after the upgrade. Verify that user buddy lists remain intact and intra/inter-cluster presence is functioning properly post-upgrade.	CUPC 1->CUPS 1->WAN->CUPS 2->CUPC 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851 IF.CU P.002	Cisco Unified Presence Cisco Unified Personal Communicator	High Availability (HA) over WAN client failover (active/active) , manual fallback	Verify if all clients and communication are able to failover automatically to another Cisco Unified Presence node located across the WAN with 80ms delay, when a Unified Presence node fails in a CoW deployment. Verify that the clients are able to fallback to original configuration when the node comes back online, a manual fallback is initiated, and users that are not moved in the failover are unaffected.	CUPC 1->CUPS 1; After failover: CUPC 1->WAN->CUP S 2	Passed	
UC851 IF.CU P.003	Cisco Unified Presence Cisco Unified Personal Communicator	Co-located HA client failover, manual fallback	Verify that when a Cisco Unified Presence node fails, all clients and communication are able to failover automatically to another Cisco Unified Presence node. Verify that the clients are able to fallback to original configuration via a manual system fallback.	CUPC 1->CUPS 1; After failover: CUPC 1->CUPS 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.CUP.004	Cisco Unified Presence Cisco Unified Personal Communicator	CoW User Rebalance	Verifies the ability to assign all users from one node to another node in a CoW setup, and then perform a user rebalance.	CUPC1->CUPS1; After failover: CUPC1->CUPS2	Passed	
UC851IF.CUP.005	Cisco Unified Presence Cisco Unified Personal Communicator Unified Communications Manager	Aggregated presence with Exchange 2010 calendaring	Verifies if Cisco Unified Presence displays the correct aggregated presence information for the user based on the user's Exchange calendar status and presence status of the user's other devices, upon making changes to the user's calendar.	CUPS->MS Exchange CUPS->Unified CM	Passed	
UC851IF.CUP.006	Cisco Unified Presence Cisco Unified Personal Communicator	Mailbox Located on Exchange 2010 Server Over WAN	Verify if Cisco Unified Presence handles redirection to a user's calendar information, when calendar sharing is enabled, but the user's mailbox is located on an Exchange server other than the one configured as the Presence gateway (the other server is located over WAN).	CUPS->MS Exchange->WAN->MS Exchange2 CUPC->CUPS	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.CUP.008	Cisco Unified Presence Cisco Unified Personal Communicator	CoW Data Center Failover (Unified Communications Manager and Cisco Unified Presence)	Verify if a user configured for MSP currently accessing MSP Unified Presence and Unified Communications Manager in a device pool which also lists System Event Archive (SEA) Unified Communications Manager, fails over and the soft phone can still make calls and IM/presence via the SEA Unified Communications Manager after bringing down both the MSP Unified Presence and Unified Communications Manager.	CUPC 1->CUPS 1; After failover: CUPC 1->CUPS 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851 IF.CU P.009	Cisco Unified Presence	Two Day Sustained 80ms Delay with CoW/HA	Verify that there are no issues over a two day period with an HA-enabled CoW sub cluster on simulating 80ms delay over the CoW link. (For example, SRM does not initiate a failover during this time period, no core dumps due to network delay issues, etc.) .	CUP node 1->WAN (80ms RTT)->CUP node 2	Passed	
UC851 IF.CU P.010	Cisco Unified Presence Cisco Unified Personal Communicator	HA over WAN Client Failover (Active/Stand by), Manual Fallback	Verify that when a Cisco Unified Presence node fails in a CoW deployment, all clients and communication are able to failover automatically to another Cisco Unified Presence node located across the WAN with 80ms delay. Verify that the clients are able to fallback to original configuration when the node comes back online and a manual fallback is initiated.	CUPC 1->CUPS 1; After failover: CUPC 1->WAN->CUP S 2	Passed	

Unified SIP Proxy

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712IF. CUSP.004	Unified SIP Proxy Interoperability with PSTN	Unified CME Phone to PSTN Phone via Unified SIP Proxy	Verify that Unified Communications Manager phone can call PSTN Phone via Unified SIP Proxy.		Passed	
UC712IF. CUSP.006	Unified SIP Proxy Supplementary Services Interopearbility with Unity Voicemail	Supplementary Services via Unified SIP Proxy	"Verify that Unified CME Phone can call Unified Communications Manager phone and perform the following: • Call Transfer • Hold and resume • Call Forward			

Unified SRST

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC85IIR. SRS.500e	Unified eSRST	Unified Messaging Gateway/ Unified eSRST Provisioning	Verify that Unified Messaging Gateway/Unified eSRST provision from Unified Messaging Gateway GUI is successful, when Unified eSRST is in pre-provisioning configuration mode.		Passed	
UC85IIR. SRS.501c	Basic Call Flow	Unified CME/Unified SRST Maps Unified Communications Manager Calls Between Endpoints	Verify whether Unified CME/Unified SRST can map Unified Communications Manager config for calls between two endpoints. Phone A calls Phone B and Phone B calls Phone A both when WAN is up and WAN is down.		Passed	
UC85IIR. SRS.501e	Basic Call Flow	Unified Messaging Gateway/Unified CME/Unified eSRST Maps Unified Communications Manager Calls Between Endpoints	Verify whether Unified Messaging Gateway/Unified CME/Unified eSRST can map Unified Communications Manager config for calls between two endpoints. Phone A calls Phone B and Phone B calls Phone A both when WAN is up and WAN is down.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC85IIR. SRS.502c	Basic Call Flow	Unified CME/Unified SRST Maps Unified Communications Manager Config for Hold and Resume	Verify whether Unified CME/Unified SRST can map Unified Communications Manager Config for the hold and resume feature. During a call between Phone A and Phone B hold/resume Phone A three times both when WAN is up and WAN is down.		Passed	
UC85IIR. SRS.502e	Basic Call Flow	Unified Messaging Gateway/Unified CME/Unified eSRST Maps Unified Communications Manager Config for Hold and Resume	Verify whether Unified Messaging Gateway/Unified CME/Unified eSRST can map Unified Communications Manager Config for the hold and resume feature. During a call between Phone A and Phone B hold/resume Phone A three times both when WAN is up and WAN is down.		Passed	
UC85IIR. SRS.601c	Basic Call Flow	Unified CME/Unified SRST Maps Unified Communications Manager Config for Call Forward All	Verify whether Unified CME/Unified SRST can map Unified Communications Manager config for call forward all feature. EpAsstnt1 (assistant endpoint 1) calls EpFeature (feature applied endpoint) and EpFeature forwards all calls to EpAsstnt2 (assistant endpoint 2) both when WAN is up and WAN is down.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC85IIR. SRS.601e	Basic Call Flow	Unified Messaging Gateway/Unified CME/Unified eSRST Maps Unified Communications Manager Config for Call Forward All	Verify whether Unified Messaging Gateway/Unified CME/Unified eSRST can map Unified Communications Manager config for call forward all feature. EpAsstnt1 (assistant endpoint 1) calls EpFeature (feature applied endpoint) and EpFeature forwards all calls to EpAsstnt2 (assistant endpoint 2) both when WAN is up and WAN is down.		Passed	
UC85IIR. SRS.602c	Basic Call Flow	Unified CME/Unified SRST can map Unified Communications Manager config for call forward busy feature.	Verify whether Unified CME/Unified SRST can map Unified Communications Manager config for call forward busy feature. EpAsstnt1 calls EpFeature. when EpFeature busy the call is forwarded to EpAsstnt2 both when WAN is up and WAN is down.		Passed	
UC85IIR. SRS.602e	Basic Call Flow	Unified Messaging Gateway/Unified CME/Unified eSRST Maps Unified Communications Manager Config for Call Forward Busy	Verify whether Unified Messaging Gateway/Unified CME/Unified eSRST can map Unified Communications Manager config for call forward busy feature. EpAsstnt1 calls EpFeature. when EpFeature busy the call is forwarded to EpAsstnt2 both when WAN is up and WAN is down.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC85IIR. SRS.603c	Basic Call Flow	Unified CME/Unified SRST Maps Unified Communications Manager Config for Call Forward No Answer	Verify whether Unified CME/Unified SRST can map Unified Communications Manager config for call forward no answer feature. EpAsstnt1 calls EpFeature and EpFeature ringing with no answer. The call is forwarded to EpAsstnt2 both when WAN is up and WAN is down.		Passed	
UC85IIR. SRS.603e	Basic Call Flow	Unified Messaging Gateway/Unified CME/Unified eSRST Maps Unified Communications Manager Config for Call Forward No Answer	Verify whether Unified Messaging Gateway/Unified CME/Unified eSRST can map Unified Communications Manager config for call forward no answer feature. EpAsstnt1 calls EpFeature and EpFeature ringing with no answer. The call is forwarded to EpAsstnt2 both when WAN is up and WAN is down.		Passed	
UC85IIR. SRS.604c	Basic Call Flow	Unified CME/Unified SRST Maps Unified Communications Manager Config for Same Group <normal> Pickup	Verify whether Unified CME/Unified SRST can map Unified Communications Manager config for same group <normal> pickup. EpAsstnt1 calls EpAsstnt2 . During EpAsstnt2 ringing EpFeature do same group pickup to connect the call both when WAN is up and WAN is down.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC85IIR. SRS.604e	Basic Call Flow	Unified Messaging Gateway/Unified CME/Unified eSRST Maps Unified Communications Manager Config for Same Group <normal> Pickup	Verify whether Unified Messaging Gateway/Unified CME/Unified eSRST can map Unified Communications Manager config for same group <normal> pickup. EpAsstnt1 calls EpAsstnt2 . During EpAsstnt2 ringing EpFeature do same group pickup to connect the call both when WAN is up and WAN is down.		Passed	
UC85IIR. SRS.605c	Basic Call Flow	Unified CME/Unified SRST Maps Unified Communications Manager Config for Different Group <normal> Pickup	Verify whether Unified CME/Unified SRST can map Unified Communications Manager config for different group <normal> pickup. EpAsstnt1 calls EpAsstnt2 . During EpAsstnt2 ringing. EpFeature push GPickUp softkey and dials the pick code. After ringing EpFeature goes offhook and connect the call both when WAN is up and WAN is down.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC85IIR. SRS.605e	Basic Call Flow	Unified Messaging Gateway/Unified CME/Unified eSRST Maps Unified Communications Manager Config for Different Group <normal> Pickup	Verify whether Unified Messaging Gateway/Unified CME/Unified eSRST can map Unified Communications Manager config for different group <normal> pickup. EpAsstnt1 calls EpAsstnt2 . During EpAsstnt2 ringing. EpFeature push GPickUp softkey and dials the pick code. After ringing EpFeature goes offhook and connect the call both when WAN is up and WAN is down.		Passed	
UC85IIR. SRS.606c	Basic Call Flow	Unified CME/Unified SRST Maps Unified Communications Manager Config for Direct Pick Up by GPickUp Softkey and Dialing DN	Verify whether Unified CME/Unified SRST can map Unified Communications Manager config for Direct pick up by GPickUp softkey and dialing DN. EpAsstnt1 calls EpAsstnt2 . EpAsstnt2 ringing but not answer. EpFeature pushes GPickUp softkey and enters DN of phone EpAsstnt2 ; EpFeature and EpAsstnt1 should be connected both when WAN is up and WAN is down.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC85IIR.SRS.606e	Basic Call Flow	Unified Messaging Gateway/Unified CME/Unified eSRST Maps Unified Communications Manager Config for Direct Pick Up by GPickUp Softkey and Dialing DN	Verify whether Unified Messaging Gateway/Unified CME/Unified eSRST can map Unified Communications Manager config for Direct pick up by GPickUp softkey and dialing DN. EpAsstnt1 calls EpAsstnt2 . EpAsstnt2 ringing but not answer. EpFeature pushes GPickUp softkey and enters DN of phone EpAsstnt2 ; EpFeature and EpAsstnt1 should be connected both when WAN is up and WAN is down.		Passed	
UC85IIR.SRS.607c	Basic Call Flow	Unified CME/Unified SRST Maps Unified Communications Manager Config for Call Park	Verify whether Unified CME/Unified SRST can map Unified Communications Manager config for call Park feature. During call with epAsstnt1; EpFeature pushes Park softkey to hold the call and gets a park extension number. EpAsstnt2 dials the park number and retrieves the call both when WAN is up and WAN is down.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC85IIR.SRS.607e	Basic Call Flow	Unified Messaging Gateway/Unified CME/Unified eSRST Maps Unified Communications Manager Config for Call Park	Verify whether Unified Messaging Gateway/Unified CME/Unified eSRST can map Unified Communications Manager config for call Park feature. During call with epAsstnt1; EpFeature pushes Park softkey to hold the call and gets a park extension number. EpAsstnt2 dials the park number and retrieves the call both when WAN is up and WAN is down.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC85IIR. SRS.608c	Basic Call Flow	Unified CME/Unified SRST Maps Unified Communications Manager Config for Hunt Group and Line Group	Verify whether Unified CME/Unified SRST can map Unified Communications Manager config for hunt group and line group feature. EpFeature calls hunt pilot number. both EpAsstnt1 and EpAsstnt2 are in the line group for hunt group. EpAsstnt1 has higher priority than EpAsstnt2 . When EpAsstnt1 doesn't answer the call. EpAsstnt2 get ringing tone and answer the call.		Passed	
UC85IIR. SRS.608e	Basic Call Flow	Unified Messaging Gateway/Unified CME/Unified eSRST Maps Unified Communications Manager Config for Hunt Group and Line Group	Verify whether Unified Messaging Gateway/Unified CME/Unified eSRST can map Unified Communications Manager config for hunt group and line group feature. EpFeature calls hunt pilot number. both EpAsstnt1 and EpAsstnt2 are in the line group for hunt group. EpAsstnt1 has higher priority than EpAsstnt2 . When EpAsstnt1 doesn't answer the call. EpAsstnt2 get ringing tone and answer the call.		Passed	

Unity

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC700IF. UNI.301	Cisco Unity	Follow-Me Home transfer with Cross-server Transfer Set to Release Transfer	Verify that Unity subscribers can alter how calls are transferred to desired endpoints.		Passed	
UC802IF. UNI.103	Cisco Unity	Enhanced Message Waiting Indicator (eMWI) for Extension Mobility Across Cluster	Verify that Enhanced Message Waiting Indicator (eMWI) works for Extension Mobility across cluster.		Passed w/ Exception	eMWI feature works correctly for Unified IP Phones 89xx/99xx series except 8961. See CSCtk193 39.
UC802IF. UNI.104	Cisco Unity	Support of eMWI Over Inter Cluster SIP Trunks	Verify that eMWI works over inter cluster CIP trunks.		Passed	
UC802IF. UNI.112	Cisco Unity	Unity in UMR mode - Partner Exchange Server Unavailable for Long Duration	Verify that Cisco Unity can handle calls when it looses connectivity to the partner exchange server for more than 24 hours.		Passed	
UC802IF. UNI.113	Cisco Unity	Primary Unified Communications Manager Server Unavailable to Cisco Unity Server	Verify that Cisco Unity can handle calls when it looses connectivity to the primary Unified Communications Manager server.		Passed	

Unity Connection

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.UNC.002	Unity Connection	Visual Voicemail with Unity Connection	Verify visual voicemail capability on Unified IP Phones (8961/9951/9971) with Unity Connection.	IP Phones (8961/9951/9971)->Cisco Unified Communications Manager->Unity Connection	Passed	
UC802EF.UNC.003	Unity Connection	Enhanced Message Waiting Indicator (MWI) with Unity Connection	Verify enhanced MWI capability on Unified IP Phones (8961/9951/9971) with Unity Connection.	89/9900 Phone->Cisco Unified Communications Manager->Unity Connection	Passed	
UC802EF.UNC.006	Unity Connection	Voicemail for QSIG PBX Phone in Unified Communications Manager Cluster	Verify deposit and retrieval of a message for a QSIG PBX phone in Unified Communications Manager cluster.	Phone->Cisco Unified Communications Manager->QSIG PBX->PBX Phone->CFNA->Unity Connection	Passed	
UC802EF.UNC.016	Unity Connection	Voicemail for Remote Phones Over MGCP/PRI PSTN Gateway	Verify voicemail deposit and retrieval for remote phones over MGCP/PRI PSTN gateway.		Passed	
UC802IF.UNC.202	Visual Voicemail	Updating list With Visual Voicemail on Cisco UC Integration™ for Microsoft Office Communicator	Verifies that visual voicemail list is updated on arrival of new message while list is displayed.	Unity Connection->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	
UC802IF.UNC.205	Visual Voicemail	Securing Visual Voicemail With Cisco UC Integration™ for Microsoft Office Communicator	Verify that security (HTTPS) can be enabled for visual voicemail.	Unity Connection->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF. UNC.206	Visual Voicemail	Download Direct and Forwarded Voicemails From Unity Connection Server2 When Server1 is Down	Verifies that Unified IP Phone 8900 and 9900 series can successfully download direct and forwarded voicemails from currently active Unity Connection server when the primary Unity Connection server in active-active cluster is down where the voice mail is originally deposited.	89/9900 Phone->Unified CM->Unity Connection	Passed	
UC802IF. UNC.208	Visual Voicemail	Message Actions With Visual Voicemail on Unified IP Phone 8900 and 9900 series	Verify that message actions such as play, pause, rewind, mark as new, delete, reply, reply to Instant Messages (IM) can be performed.	Unity Connection->Unified CM->89/9900 Phone	Passed	
UC802IF. UNC.209	Visual Voicemail	Retrieve, Reply and Send a Secure Message Using Unified IP Phone 8900 and 9900 series	Verify if messages that have been marked secure can be retrieved and replied to. Verify if it is possible to send a voicemail and mark it secure.	Unity Connection->Unified CM->89/9900 Phone	Passed	
UC802IF. UNC.210	Visual Voicemail	Securing Visual Voicemail With Unified IP Phone 8900 and 9900 series	Verify that security (HTTPS) can be enabled for visual voicemail.	Unity Connection->Unified CM->89/9900 Phone	Passed	
UC802IF. UNC.215	Visual Voicemail	Send Message to User on Digitally Networked Server Using Unified IP Phone (8900 and 9900) Series	Verify that messages can be sent using Visual Voice Mail (VVM) to a subscriber on a Unity Connection server that is digitally networked to the server on which the VVM user is.	Unity Connection->Unified CM->89/9900 Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. UNC.001	Unity Connection	Support for Authentication and Encryption with Dual Stack SCCP Integration	Verify that calls with encrypted media can be placed to dual stack (IPv4/IPv6) unity connection that is integrated to Unified Communications Manager.	DS Unified IP Phone->Unified CM -->SIPT (DS ANAT Secure Early offer) -->Unified CM -->Unified IP Phone -->CFNA -->(SCCP) Unity Connection	Passed	
UC851IF. UNC.002	Unity Connection	Support for Authentication and Encryption with Dual Stack SIP Integration	Verify that calls with encrypted media can be placed to Dual Stack (IPv4/IPv6) Unity Connection that is integrated to Unified Communications Manager.	DS Unified IP Phone->Unified CM -->SIPT (DS ANAT Secure Early offer) -->Unified CM -->Unified IP Phone -->CFNA -->(SIP DS ANAT Secure Early offer) Unity Connection	Passed	
UC851IF. UNC.003	Unity Connection	Supervised Transfer to Secure Phone from a Dual Stack Unity Connection - SCCP Integration	Verify that Unity Connection can successfully supervise transfer a call to a DS Unified IP Phone.	Secure DS Unified IP Phone->Unified CM -->SIPT (DS ANAT Secure Early offer) -->Unified CM -->SCCP (secure) -->Unity Connection -->Supervised Xfer -->secure DS Unified IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. UNC.004	Unity Connection	Release Transfer to a Phone from a Dual Stack Unity Connection - SIP Integration	Verify that Unity Connection can successfully release transfer a call to a DS Unified IP Phone.	DS Unified IP Phone->Unified CM -->SIPT (DS ANAT Early offer) -->Unified CM -->SIP (secure) -->Unity Connection -->Release Xfer -->DS Unified IP Phone	Passed	
UC851IF. UNC.005	Unity Connection	Unity Connection SIP Integration with Early Offer Enabled on Unified Communications Manager	Verify that Unity Connection supports Early offer.	Phn1 -->Unified CM -->SIPT -->Unified CM -->SIPT (DS Early offer) -->Unity Conneciton	Passed	
UC851IF. UNC.006	Unity Connection	Diversion Header Support for Multiple Call Forwards in a Dual Stack Environment	Verify that Unity Connection dual stack integration can support more than one diversion header with early offer.	Phone -->Unified CM -->SIPT/ICT -->SME -->SIPT/ICT -->Unified CM -->Phone1 -->CFA -->Phone2 -->CFNA -->Unity Connection	Passed	
UC851IF. UNC.007	Unity Connection	Message Notification over Dual Stack SIP Integration to a DS Phone	Verify that message notification works for dual stack SIP integration with Unified Communications Manager.	Unity Connection -->DS SIPT -->Unified CM -->SIPT/ICT -->SME -->SIPT/ICT -->Unified CM -->Phone	Passed	
UC851IF. UNC.008	Unity Connection	Sending a Message to Remote VPIM Location from a Dual Stack Phone	Verify that a message can be addressed to a remote VPIM location from a dual stack phone when the media is IPv6.	Unity Connection -->VPIM -->UMG -->VPIM -->Cisco Unity Express	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. UNC.009	Unity Connection	Sending Broadcast Messages from a DS Unified IP Phone	Verify that a broadcast message can be sent when the media is IPv6	DS Unified IP Phone -->Unified CM -->Unity Connection	Passed	
UC851IF. UNC.010	Unity Connection	Visual Voice Mail when the Call Setup for Recording Voicemail is IPv6	Verify that visual voicemail works fine when the media negotiated when leaving a voicemail was IPv6.	DS Unified IP Phone -->Unified CM -->Unity Connection	Passed	
UC851IF. UNC.011	Unity Connection	Transfer to Alternate Contact Number for an IPv6 Call	Verifies to ensure that Unity Connection can transfer the call to a DS phone when the caller input for a digit is set to alternate contact number.	DS Unified IP Phone -->Unified CM -->SCCP/SIPT DS -->Unity Connection -->SCCP/SIPT DS -->Unified CM -->DS Unified IP Phone	Passed	
UC851IF. UNC.012	Unity Connection	Addressing a Message from Directory Handler to a Digitally Networked System for an IPv6 call	Verifies to ensure that a caller can look-up a subscriber in a digitally networked system and address a message where the media for the call is IPv6.	DS Unified IP Phone -->Unified CM -->Unity Connection1 -->Unity Connection2	Passed	
UC851IF. UNC.013	Unity Connection	Cross Server Transfer for an IPv6 Call	Verifies to ensure that Unity Connection can successfully hand off the call to a networked Unity Connection server using the cross server transfer feature.	DS Unified IP Phone -->Unified CM -->Unity Connection1 -->Unity Connection2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. UNC.014	Unity Connection	Cross Server Sign in for an IPv6 Call	Verifies to ensure that Unity Connection can successfully hand off the call to a networked Unity Connection server using the cross server sign in feature.	DS Unified IP Phone -->Unified CM -->Unity Connection1 -->Unity Connection2	Passed	
UC851IF. UNC.015	Visual Voicemail	Using VMO to Send a Voicemail	Verifies to ensure that subscribers can retrieve voicemails sent using Cisco Unity ViewMail for Outlook (VMO), can retrieve it over a DS trunk.	DS Unified IP Phone -->Unified CM -->Unity Connection	Passed	
UC851IF. UNC.017	Unity Connection - Dual Stack	Addressing a Message to a Remote CCI System from a Dual Stack Unity Connection	Verify that a message can be sent to a subscriber in Copy Control Information (CCI) system from a Unity Connection server that supports both IPv4 and IPv6.	DS Unified IP Phone -->Unified CM -->DS Unity Connection1 -->CCI -->DS Unity Connection2	Passed	
UC851IF. UNC.018	Unity Connection - Dual Stack	PSTN Access from a Dual Stack SIP GW to Unity Connection that supports both IPv4 and IPv6	Verifies to ensure that Unity Connection can inter operate seamlessly with Unity Connection where both SIP GW and Unity Connection support IPv4 and IPv6.	PSTN Phone -->DS SIP GW -->SIPT (DS ANAT) -->Unified CM -->SIPT (DS ANAT)	Passed	
UC851IF. UNC.019	Unity Connection - Dual Stack	Invoking Transcoder for a Call to Unity Connection	Verifies that a transcoder gets allocated dynamically when needed, for a call to Unity Connection.	PSTN Phone -->PSTN GW (IPv4 only) -->Unified CM -->Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF. UNC.020	Unity Connection - Dual Stack	Replying to a Voicemail by Placing a Call where the Media is IPv6	Verify that Connection can respond to a message by placing a call, ensuring that the media for the call is IPv6.	DS Unified IP Phone -->Unified CM -->DS Unity Connection -->Xfer -->Unified CM -->DS Unified IP Phone	Passed	
UC851IF. UNC.021	Unity Connection - Dual Stack	Addressing and Deleting a Message through Voice User Interface (VUI) when the Media for the Call is IPv6	Verifies to ensure that the caller user experience is unchanged when the media negotiating the call is IPv6.	DS Unified IP Phone -->Unified CM -->DS Unity Connection	Passed	
UC851IF. UNC.101	Unity Connection - Single Inbox	Single Inbox - Secure Message Synchronizatio n with Exchange 2010	Verify that Unity Connection does not synchronize messages that are marked as secure by the sender, and only a text message stating that message is secure is displayed.	DS Unified IP Phone -->Unified CM -->DS Unity Connection	Passed	
UC851IF. UNC.102	Unity Connection - Single Inbox	Single Inbox - New and Saved Messages with MWI Indicator	Verify that when a saved message is marked new in the email client then Message Waiting Indication (MWI) is turned on, given that MWI status should be updated based on the actions in the email client.	DS Unified IP Phone -->Unified CM -->DS Unity Connection	Passed	
UC851IF. UNC.103	Unity Connection - Single Inbox	Single Inbox - Synchronizing Voice Message with Attachments	Verify that Unity Connection can synchronize voice message with attachments.	DS Unified IP Phone -->Unified CM -->DS Unity Connection	Passed	
UC851IF. UNC.104	Unity Connection - Single Inbox	Single Inbox - Deleting Voicemails from Exchange 2010	Verify that voicemails can be deleted from Exchange 2010.	DS Unified IP Phone -->Unified CM -->DS Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.UNC.105	Unity Connection - Single Inbox	Single Inbox - Marking Voice Messages Urgent	Verifies to ensure that voicemails marked as urgent reflect the status correctly when synchronized with Exchange.	DS Unified IP Phone -->Unified CM -->DS Unity Connection	Passed	
UC851IF.UNC.106	Unity Connection - Single Inbox	Single Inbox - Read Receipts for Synchronized Voicemails	Verify that read receipts are also synchronized with Exchange.	DS Unified IP Phone -->Unified CM -->DS Unity Connection	Passed	
UC851IF.UNC.107	Unity Connection - Single Inbox	Single Inbox - Message synchronization when the primary connection server in the cluster is down.	Verify that synchronization happens even when the primary Connection server is down	DS Unified IP Phone -->Unified CM -->DS Unity Connection	Passed	
UCS712IF.UNC.101	Unity Connection Unity Express Unified Messaging Gateway	Forwarding a Fax message from Unity Conneciton to Cisco Unity Express (CUE) over Unified Messaging Gateway and back.	Verify that a fax received by Cisco Gateway can be relayed to Unity Connection through Cisco Fax Server. Verify that the connection subscriber can forward the message to Cisco Unity Express over Unified Messaging Gateway (VPIM) and Cisco Unity Express subscriber can in turn forward back the message to Unity Connection subscriber over a VPIM network with UMG in between.	Fax -->PSTN -->IOS Gateway -->CFS -->Unity Connection <-->VPIM <-->UMG <-->VPIM <-->Cisco Unity Express	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF.UNC.101	Cisco Unity Connection Cisco Unity Express Unified Messaging Gateway	Forwarding a Fax message from Cisco Unity Connection to Cisco Unity Express over Unified Messaging Gateway and Back	Verify that a fax received by Cisco Gateway can be relayed to Cisco Unity Connection via Cisco Fax Server. Verify that the Cisco Unity Connection subscriber can forward the message to Cisco Unity Express over Unified Messaging Gateway (VPIM) and Cisco Unity Express subscriber can in turn forward back the message to Cisco Unity Connection subscriber over a VPIM network with Unified Messaging Gateway in between.		Passed	
UCS712IF.UNC.102	Unity Connection Unity Express Unified Messaging Gateway	Cisco Unity Express Subscriber Replying to a Fax Message from Unity Conneciton to Cisco Unity Express over Unified Messaging Gateway	Verify that a fax received by Cisco Gateway can be relayed to Unity Connection via Cisco Fax Server. Verify that the connection subscriber can forward the message to Cisco Unity Express over Unified Messaging Gateway (VPIM) and Cisco Unity Express subscriber can in turn reply back to that message.	Fax -->PSTN -->IOS Gateway -->CFS -->Unity Connection <-->VPIM <-->UMG <-->VPIM <-->CUE	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF .UNC.106	Unity Connection Unity Express Unified Messaging Gateway	Forwarding a fax to a System Distribution List in UMG	Verify that fax message forwarded to SDL is received by each member in the SDL.	Fax -->PSTN -->IOS Gateway -->CFS -->Unity Connection <-->VPIM <-->UMG <-->VPIM <-->Cisco Unity Express	Passed	
UCS712IF .UNC.202	Unity Connection Unity Express	Cisco Unity Express Subscriber Replying to a Fax Message from Unity Conneciton to Cisco Unity Express over VPIM	Verify that a fax received by Cisco Gateway can be relayed to Unity Connection through Cisco Fax Server. Verify that the Connection subscriber can forward the message to Cisco Unity Express over VPIM and Cisco Unity Express subscriber can in turn reply back to that message.	Fax -->PSTN -->IOS Gateway -->CFS -->Unity Connection <-->VPIM <-->Cisco Unity Express	Passed	
UCS712IF .UNC.202	Cisco Unity Connection Cisco Unity Express	Cisco Unity Express Subscriber Reply to Fax Message from Cisco Unity Conneciton to Cisco Unity Express over VPIM	Verify that a fax received by Cisco Gateway can be relayed to Cisco Unity Connection via Cisco Fax Server and the Cisco Unity Connection subscriber can forward the message to Cisco Unity Express over VPIM and the Cisco Unity Express subscriber can in turn reply back to that message.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF.UNC.301	Unity Connection Unity Express Unified Messaging Gateway	Forwarding a Fax message from Cisco Unity Express to Unity Conneciton over Unified Messaging Gateway and back	Verify that a fax received by Cisco Gateway can be relayed to Cisco Unity Express via Cisco Fax Server. Verify that the Cisco Unity Express subscriber can forward the message to Unity Connection over Unified Messaging Gateway (VPIM) and Connection subscriber can in turn forward back the message to Cisco Unity Express subscriber over a VPIM network with Unified Messaging Gateway in between.	Fax -->PSTN -->IOS Gateway -->CFS -->CUE <-->VPIM <-->UMG <-->VPIM <-->Unity Connection	Passed	
UCS712IF.UNC.304	Unity Connection Unity Express Unified Messaging Gateway	Forwarding a fax to a System Distribution List in UMG	Verify that fax message forwarded to SDL is received by each member in the SDL.	Fax -->PSTN -->IOS Gateway -->CFS -->Cisco Unity Express <-->VPIM <-->UMG <-->VPIM <-->Unity Connection	Passed	
UCS712IF.UNC.501	Unity Connection	Fax Call Forwarded to Unity Connection in Connect First Mode for Single Number Fax	Verify that fax message is successfully delivered to the subscriber when the call is forwarded to Unity Connection when the gateway is configured in Connection-first fax detection mode.	Fax -->PSTN -->IOS Gateway -->SMTP -->Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF .UNC.501	Cisco Unity Connection	Fax Call Forwarded to Unity Connection In Connect First Mode - Single Number Fax	Verify that a fax message is successfully delivered to the subscriber when the call is forwarded to Unity Connection. The gateway is configured in Unity Connection-first fax detection mode.		Passed	
UCS712IF .UNC.502	Unity Connection	Fax Call Put On Hold for Single Number Fax	Verify that fax message is successfully delivered to the subscriber when the call is answered by the phone and is put on hold.	Fax -->PSTN -->IOS Gateway -->SMTP -->Unity Connection	Passed	
UCS712IF .UNC.504	Unity Connection	Fax Call Disconnect by user for Single Number Fax Connect First Mode	Verify if the fax message is successfully delivered to the subscriber when the call is answered by the phone and is disconnected.	Fax -->PSTN -->IOS Gateway -->SMTP -->Unity Connection	Passed	
UCS712IF .UNC.505	Unity Connection	Single Number Fax - Listen first mode - Fax call	Verify if the fax message is successfully delivered to the subscriber when the call is answered by the phone and is disconnected.	Fax -->PSTN -->IOS Gateway -->SMTP -->Unity Connection	Passed	
UCS712IF .UNC.505	Cisco Unity Connection	Single Number Fax - Listen First Mode - Fax Call	Verify that the fax message is successfully delivered to the subscriber when the call is answered by the phone and is disconnected.		Passed	

Unity Express

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF. CUE.002	Cisco Unity Express	Configure Multiple Greetings Through CLI in Cisco Unity Express	Verify if multiple greetings like Busy, Closed, Internal, Vacation, Meeting and Extended Absence greetings through CLI in Cisco Unity Express can be configured to make a call from QSIG PBX Phone to Unified CME SIP Phone which has Call Forward No Answer (CFNA) to Cisco Unity Express voicemail.	PBX Ph1->QSIG PBX->Unified CM->SIP Trunk->Cisco Unified SIP Proxy->Unified CME->SIP Ph1-> CFNA->Cisco Unity Express	Failed	CSCtj23133
UC802EF. CUE.003	Cisco Unity Express	Depositing Voicemail to the Unified CME Phones Over Unified SIP Proxy from a Unified CM Unified IP Phones 8961/9951/9971	Verify that the Unified IP Phones 8961/9951/9971 are able to successfully deposit a voice mail to the Unified CME phone over Unified SIP Proxy.	RT Pro/Biz/Std + Ph1 ->Unified CM->SIP Trunk->Unified SIP Proxy->Unified CME->Ph2 ->CFNA->Cisco Unity Express	Failed	CSCtj23133

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CUE.004	Cisco Unity Express	Depositing Voicemail to Unified CME SIP Phones over Unified SIP Proxy from a PBX Phone	Verify that PBX phones are able to successfully deposit a voice mail to the Unified CME SIP Phones over Unified SIP proxy.	PBX Ph1->PBX->VG30D->MGCP Gateway->Unified CM->SIP Trunk->Unified SIP Proxy->Unified CME->SIP Ph1 (First Line)->CFNA->SIP Ph1 (Second Line)->CFNA->Cisco Unity Express	Failed	CSCtj23133
UC802IF.CUE.100	Cisco Unity Express Visual Voicemail	Cisco Unity Express Subscriber Listens to Message via VoiceView Express	Verifies that the user can listen to voicemail messages via VoiceView xpress feature of Cisco Unity Express.		Passed	
UC802IF.CUE.101	Cisco Unity Express Visual Voicemail	Cisco Unity Express Subscriber Sends Message via VoiceView Express	Verify that Cisco Unity Express subscriber can send message to another subscriber via voice view express.		Passed	
UC802IF.CUE.102	Cisco Unity Express Visual Voicemail	Cisco Unity Express Subscriber Sends Broadcast Message	Verify that Cisco Unity Express subscriber can send broadcast message via VoiceView Express.		Passed w/ Exception	CSCtj87186

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CUE.752	Cisco Unity Express-SRSV	Using Multiple Cisco Survivable Remote Site Voicemail Unified Messaging Gateway to Support Multiple SRSV-Cisco Unity Express	Verify that multiple SRSV-Unified Messaging Gateway can be used with one Unified Communications Manager cluster to provision multiple SRSV-Cisco Unity Express locations where each SRSV-UMG is assigned with a subset of available SRSV-Cisco Unity Express.		Passed	
UC802IF.CUE.756	Cisco Unity Express-SRSV	Provisioning and Functioning for Unified SRST with SIP Phones	Verify that SRSV feature can operate seamlessly in SIP Unified SRST deployment.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CUE.762	Cisco Unity Express-SRSV	Custom Holiday Schedule and Holiday Greeting	Verify that Holiday Greeting is played by SRSV-Cisco Unity Express when in fallback mode.		Passed	
UC802IF.CUE.766	Cisco Unity Express-SRSV	Addressing Messages to Distribution List Containing Members from Digitally Networked Cisco Unity Connection	Verify that messages can be addressed to distribution lists from SRSV-Cisco Unity Express and they are successfully delivered on WAN link restoration.	IP Phone->SRST->SIP->Cisco Unity Express; IP Phone->Unified CM->SCCP/SIP->Connection	Passed	

Video Telephony

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
CO10. VID.00 5	Video	SIP Non-Video to SCCP Video Call Transfer to SCCP Video Endpoint Inter-Cluster SIP Trunk	Verify the call transfer from SIP (TNP) non-video endpoint, transfering a SCCP Video endpoint to another SCCP Video capable endpoint across inter-cluster SIP Trunk.		Passed	
CO10. VID.02 5	Video	IP Communicator and CVTA With Mid-Call Video Inter-Cluster SIP Trunk	Verify that Cisco IP Communicator with CVTA endpoint can call another video endpoint while CVTA application is down. After call connects CVTA application is started and mid-call video is established through inter-cluster SIP Trunk.		Passed	
UC701 EF.VID .008	Video Conference using IPVC	Reservationless Videoconference on 3545 through H.323 Interface	Verify if a user can make reservationless videoconference using an IP Videoconferencing (IPVC) 3545 through H.323 interface and calling users are H.320 terminal, with video endpoints supporting H.323 and SCCP.	Stage1: H.320->3545 (Gateway)-> Unified CM->3545 Stage2: H.323 Video Ph1->Unified CM 3545 Stage3: Rem SCCP Video Ph2->Unified CM->3545	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701 EF.VID .009	Video Conference using IPVC	Reservationless Videoconference through H.323 Interface	Verify if a user can make reservationless videoconference using an Multipoint Control Unit (MCU) 3545 through H.323 interface and calling user is H.320 terminal, and video endpoints supporting H.323 and SCCP.	Stage1: H.320->3545(Gateway)->Unified CM->3545 Stage2: H.323 Ph1->Unified CM->3545 Stage3: Rem SCCP Ph1->Unified CM->3545	Passed	
UC702 EF.VID .030	Video Conference using IPVC	Reservationless Conference Using Cisco 3545 Involving H.320 Endpoints and Remote Third Party Video SCCP Endpoint	Verify the reservationless video conference call using Cisco 3545 involving H.320 endpoint and Remote third party SCCP video phone.	Stage1: H.320 Video Ph1->PSTN->Unified CM->3545; Stage 2: 3rd Party Video SCCP Ph2->Unified CM-> 3545; Stage 3: SCCP Video Ph3->Unified CM->3545	Passed	
UC702 EF.VID .031	Video Conference using IPVC	Reservationless Videoconference through H.323 using IPVC 3545 with Unified Video Advantage Phone	Verify if a user can make a reservationless videoconference through H.323 using IPVC 3545 when the calling users are Unified Video Advantage phones and SCCP video phones.	Stage1: SCCP Ph1 (Unified VA)->Unified CM->3545 Stage2: Rem SCCP Video Ph2->Unified CM->3545 Stage3: Rem SCCP Video Ph3 (Unified VA)->Unified CM->3545	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC702 EF.VID .040	Video Conference using IPVC	Adhoc Video Conference Using IPVC 3545 with SCCP and IP Communicator Endpoints	Verify if a user can make Adhoc video conference using an IPVC 3545 involving SCCP and IP Communicator endpoints.	Stage1: SCCP Video ph1->Unified CM->H.323 Video ph2 Stage2: H.323 Video ph2->Unified CM->3545(MCU)->Unified CM->IP Communicator (Unified VA)	Passed	
UC702 EF.VID .041	Video Conference using IPVC	Continuous Presence and Transrating on MCU 3545 using H.323 Interface Involving Third party Video Phone	Verify if a user can make reservationless conference using MCU 3545 and enable continuous presence involving third party video phone.	Stage1: SCCP Video Ph1->Unified CM->3545; Stage2: Rem SCCP Video Ph2->Unified CM->3545; Stage3: H.323 Video Ph3->Unified CM->3545; Stage4: 3rd Party SCCP Video Ph4->Unified CM->3545	Passed	
UC851 EF.VID .020	Telepresence	Video Calls Over Session Manager Using H.320 phone and 89/9900 Phone	Verify video calls over session manager using H.320 phone and 89/9900 Phone	9971/9951 Ph1->Unified CM1->SIP->SME->SIP->Unified CM2->IPVC->H.320 PSTN EP	Passed	
UC851 EF.VID .021	Telepresence	Video Calls Over Session Manager using H.320 phone and 89/9900 Phone	Verify video calls over session manager using H.320 phone and 89/9900 Phone	9971/9951 Ph1->Unified CM1->ANNEXM1->SME->SIP->Unified CM2->IPVC->H.320 PSTN EP	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851 EF.VID .022	Telepresence	Video Calls Over Session Manager Using H.320 phone and 89/9900 Phone	Verify video calls over session manager using H.320 phone and 89/9900 Phone	9971/9951 Ph1->Unified CM1->SIP-> SME->ANNEXM1->Unified CM2->IPVC->H.320 PSTN EP	Passed	
UC851 EF.VID .023	Telepresence	Video Calls Over Session Manager Using H.320 Phone and Unified Video Advantage phone	Verify video calls over session manager using H.320 phone and Cisco Unified Video Advantage pro phone.	Cisco Unified Video Advantage-> Unified CM1->SIP-> SME->SIP-> Unified CM2->IPVC->H.320 PSTN EP	Passed	
UC851 EF.VID .024	Session Management Edition	Video Calls Over Session Manager Using H.320 phone and Unified Video Advantage Phone	Verify video calls over session manager using H.320 phone and Unified Video Advantage phone.	Unified Video Advantage-> Unified CM1->SIP-> SME->ANNEXM1->Unified CM2->IPVC->H.320 PSTN EP	Passed	
UC851 EF.VID .025	Session Management Edition	Video Calls Over Session Manager Using H.320 phone and Unified Video Advantage phone	Verify video calls over session manager using H.320 phone and Unified Video Advantage phone.	H.320 Pstn Ep->IPVC-> Unified CM1->ANNEXM1->SME->SIP->Unified CM2->Unified Video Advantage	Passed	
UC851 EF.VID .026	Session Management Edition	Video Calls Over Session Manager controlled by RSVP	Verify video calls over session manager controlled by RSVP.	7985 ph1->Unified CM1(RSVP)->SIP->SME->SIP->(RSVP) Unified CM2->7985 Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851 EF.VID .027	Session Management Edition	Video Calls Over Session Manager controlled by RSVP	Verify video calls over session manager controlled by RSVP.	UC Integration™ for Microsoft Office Communicator ph1->Unified CM1(RSVP)->SIP->SME->SIP->(RSVP) Unified CM2->UC Integration™ for Microsoft Office Communicator ph1	Passed	
UC851 EF.VID .028	Session Management Edition	Video Escalation Over Session Manager Edition	Verify video call escalation over session manager edition.	7985 SCCP Ph1->Unified CM1->SIP-> SME->SIP-> Unified CM2->7961 ph1->Xfer_C->9971/9951 Ph1	Passed	
UC851 EF.VID .031	Session Management Edition	Video De-escalation Over Session Manager Edition	Verify video call de-escalation over Session Management Edition.	7985 SCCP Ph1->Unified CM1->SIP-> SME->SIP-> Unified CM2->9971/9951 Ph1->Xfer_C ->7961 Ph1	Passed	
UC851 EF.VID .032	Session Management Edition	Video De-escalation Over Session Manager Edition	Verify video call de-escalation over Session Management Edition.	7985 SCCP Ph1->Unified CM1->SIP-> SME->ANNEXM1->Unified CM2->9971/9951 Ph1->Xfer_C ->7961 Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851 EF.VID .033	Session Management Edition	Video De-escalation Over Session Manager Edition	Verify video call de-escalation over session manager edition.	7985 SCCP Ph1->Unified CM1->ANNEXM1->SME->SIP->Unified CM2->9971/9951 Ph1->Xfer_C->7961 Ph1	Passed	
UC851 EF.VID .034	Session Management Edition	Video Call transfer Over Session Manager Edition	Verify that the video call transfer is working over Session Management Edition.	7985 SCCP Ph1->Unified CM1->SIP-> SME->SIP-> Unified CM2->Unified Personal Communicator->Xfer_C-> ANNEXM1-> Unified CM1->UC Integration™ for Microsoft Office Communicator	Failed	
UC851 EF.VID .035	Telepresence	Video Call Hold Over Session Manager Edition	Verify that the video call hold is working over Session Management Edition.	Unified Video Advantage ph1-> Unified CM1->SIP-> SME->SIP-> Unified CM2->7985 SIP ph1	Passed	
UC851 EF.VID .036	Session Management Edition	Video Call Hold Over Session Manager Edition	Verify that the video call hold is working over Session Management Edition.	Unified Video Advantage ph1->Unified CM1->SIP-> SME->ANNEXM1->Unified CM2->7985 SCCP ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851EF.VID.037	Telepresence	Video Call Resume Over Session Manager Edition	Verify that the video call resume is working over Session Management Edition.	Unified Video Advantage ph1->Unified CM1->SIP->SME->SIP->Unified CM2->7985 SIP ph1	Passed	
UC851EF.VID.038	Session Management Edition	Video Call Resume Over Session Manager Edition	Verify that the video call resume is working over Session Management Edition.	7962 Unified Video Advantage ph1->Unified CM1->SIP->SME->ANNEXM1->Unified CM2->7985 SCCP ph1	Passed	
UC851EF.VID.045	Session Management Edition	RSVP Agent is Getting Rebooted when the E2E Video RSVP is in Active	Verify that when the RSVP agent in either side of the phone is rebooted during the active E2E video RSVP call, the bandwidth reserved for the call should be released properly.	7985 Ph1->Unified CM1->Sip Trunk->Unified CM2->7985 Ph2	Passed w/exception	CSCtj99237
UC851IF.VID.004	Video Interoperability	Point to Point Video Call between MXP 1700 and Life Size over SAF trunk	Verify video call between MXP 1700 registered with ABI Unified Communications Manager and life size SIP video registered to MSP Unified Communications Manager.	MXP 1700 -->ABI Unified CM -->SAF SIP Trunk -->Lifesize -->MSP Unified CM	Passed	
UC851IF.VID.005	Video Interoperability	Point to Point Call Between SCCP TD 1000 and UC Endpoint over H.323 Trunk	Verify video call between SCCP TD 1000 registered to MSP Unified Communications Manager and UC endpoint (9971, 9951 ,CSF) registered to ABI Unified Communications Manager over H.323 Trunk.	TD 1000 --> SCCP--->MSP Unified CM ---->H.323 ---->ABI Unified CM -->9971	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.VID.007	Video Interoperability	H.225 Video between Unified Communications Manager and Unified Communications Manager Express	Verify Video between UC endpoint registered to ABI Unified Communications Manager and 7985 registered to Unified Communications Manager Express over H.225 trunk.		Passed	
UC851IF.VID.021	Video Interoperability	Multipoint Conference with CTS 500 , CTS 3000 and TD 1000 through MXE and CTMS	Verify multipoint conference with CTS 1000 registered in Session Management Edition cluster , TD 1000 and CTS 3000 via MXE and CTMS.	CTMS --- Conference --- MXE ---CTS 1000 --- H.323 --- TD 1000 ---CTS 3000	Passed	
UC851IF.VID.023	Video Interoperability	Point to Point Call from a UC endpoint to CTS 1000 via MXE and SAF and then Transfer the Call to Polycom over SIP trunk	Verify if video is established between UC endpoint and Polycom VSX 7000 when the video call is transferred over SIP trunk.		Failed	CSCtj06789
UC851IF.VID.024	Video Interoperability	Call Transfer when Redirecting CSS is Not Configured for Third Party Video Phone	Verifies to ensure that the original call remains connected when call transfer fails due to redirecting CSS for a Tandberg E20.	89/9900 Phone -->Unified CM -->SIPT -->Unified CM -->E20	Failed	CSCti25708
UC851IF.VID.040	Video Interoperability	Video over Wireless	Verify quality of video when 9971 registered to Unified Communications Manager through wireless conferences.	9971 --- ABI Unified CM -- H.225 Trunk --- MSP Unified CM --- 9971 --- Conference from ABI Unified CM ---H.225 --- 9971	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.VID.047	Video Interoperability	Video Call Over SIP / H.323 SAF Trunk	Verify if Session Management Edition site can interwork H.323 / SIP video when the routes learned is through different protocol.	9971 --- ABI Unified CM --- SIP Trunk ---- SME Site --- H.323 Site ---MSP Unified CM---9971	Passed	
UC851IF.VID.050	Video Interoperability	Bandwidth Limitation : Call Fall Back to Audio	Verify if the call from video enabled endpoints tries to setup audio when bandwidth is restricted across WAN.	CUPC ---- ABI Unified CM --- SIP trunk --- MSP Unified CM --- 9971 ---CIPC with CUVA --- ABI Unified CM --- MSP Unified CM --- 9971	Passed	
UC851IF.VID.051	Video Interoperability	Call from 9971 Registered in ABI Unified Communications Manager to E20 Video Endpoint Â Registered to VCS Control	Verify point to point call between 9971 registered to ABI Unified Communications Manager and E20 SIP endpoint registered to VCS.	9971-- ABI Unified CM ---SIP trunk --- VCS -E20	Passed	
UC851IF.VID.052	Video Interoperability	Call from CTS 1000 Registered to Unified Communications Manager to C20 H.323 Video Endpoint Behind VCS	Verify video call can be placed between CTS 1000 and C20 H.323 Video endpoint registered to VCS.	9971 -- ABI Unified CM --- SIP trunk --- C20 - H.323 ---VCS	Passed	
UC851IF.VID.053	Video Interoperability	Verify Point to Point Call between Polycom in H.323 mode and MOVi client registered to VCS as SIP endpoint	Verify video call is established end to end between polycom registered to ABI Unified Communications Manager in H.323 mode and MOVi client registered in SIP mode.	Polycom -- Video GK - RAS Agg ---ABI Unified CM --- SIP Trunk --- MOVi	Failed	CSCtj38859

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.VID.054	Video Interoperability	Point to Point call from E20 registered to VCS to E20 registered to Unified Communications Manager	Verify the video call can be placed and put on hold by E20 registered to Unified Communications Manager and the call resumes back to video.	E20 --- ABI Unified CM --- SIP TRUNK ---E20 -- VCS	Passed	
UC851IF.VID.055	Video Interoperability	Call Transfer between E20 and C20	Verify video call can be transferred from E20 registered to Unified Communications Manager to C20 registered to VCS.	E20 -- ABI Unified CM -- SIP trunk -- VCS -- C20 -- E20 TRANSFER ---SIP trunk --- TD 1700 MXP -- VCS	Passed	
UC851IF.VID.057	Video Interoperability	Conference with VCS and CTS endpoints via MXE and CTMS	Verify conference between VCS and CTS 1000 via MXE and CTMS.	E20 -VCS --- SIP trunk --SME -- SIP trunk -- MXE -- CUCM --CTMS -- Conference --CTS 1000 -- CUCM--- MXE --- SME --SIP Trunk--CTMS - Conference	Passed	
UC851IF.VID.058	Video Interoperability	Scheduled Conference using Codian MCU	Verify if E20 , C20 registered to VCS and E20 registered to Unified Communications Manager are able to join Scheduled conference using Codian MCU bridge.	C20 E20 E20 ---- H.323 DN --- Codian MCU	Passed	
UC851IF.VID.059	Video Interoperability	Adhoc Conference Using Codian MCU and VCS Multiway Feature	Verify if the endpoints can join the conference bridge using multiway feature.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.VID.060	Video Interoperability	E20 Behind VCS Interoperability with Voicemail System	Verify if endpoint registered to VCS can deposit voicemail to E20 registered to VCS.	C20 --- VCS ----E20 -- CFWD -- Unity connection	Passed	
UC851IF.VID.061	Video Interoperability	E20 Behind VCS Calling IPCC Agent	Verify if E20 registered to VCS is able to call to IPCC agent and two way video is established between E20 and IPCC video enabled Agent.	E20 ---VCS -- SIP trunk --- IPCC Pilot ----IPCC Agent	Passed	
UC851IF.VID.062	Video Interoperability	H.323 VCS Endpoint calling CER	Verify if H.323 endpoint behind VCS is able to call CER.	C20 ---VCS -- SIP trunk --- CER	Passed	
UC851IF.VID.063	Video Interoperability	Successfully Transfer Call Placed Between VCS and Unified Communications Manager to another UC endpoint	Verify the ability to call from TD 1000 registered to VCS to 9971 over SIP trunk which is call forwarded to 9971 Â in MSP Unified Communications Manager. Verify if video is established between transferee and transferred end point.	TD 1000 ---VCS -- SIP trunk --- 9971 ---Unified CM -- Call forward --- MSP Unified CM --9971	Passed	
UC851IF.VID.065	Video Interoperability	Video Call between MOVi behind VCS and 9971 in SRST Location	Verify Video Call Quality from MOVi Client Registered to VCS to 9971 registered to SRST Site	E20 ---VCS -- SIP trunk --- Unified CM --- WAN --- SRST --- 9971	Passed	
UC851IF.VID.1000	Video Interoperability	8961+Unified Video Advantage Interoperability with CTS through MXE	To test if MXE allows an 8961 end point with Unified Video Advantage camera to Interoperability with CTS 1000.	8961+CUVA -MSP Unified CM->DEN Unified CM(10.4.101.11)->MXE(10.4.101.16)->DEN Unified CM->ABI Unified CM->CTS 1000	Failed	CSCtj10392

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.VID.1001	Video Interoperability	MXE with using Session Management Edition as a Tandem Cluster	Verifies if MXE can be introduced in the call path on a leaf cluster for UC-TP Interoperability.	89/9900 Phone->MSP Unified CM->MXE->MSP Unified CM->SIPT->DEN Unified CM(SME)->ASIPT->ABI Unified CM-CTS 1000	Passed	
UC851IF.VID.1199	Video Interoperability	MOVi Client Interoperability with MXE	MOVi client dials into a CTMS meeting on tandem cluster through MXE.	MOVI->VCS->DEN Unified CM->SIPT->MXE->SIPT->DEN Unified CM->SIPT->CTMS	Passed	
UC851IF.VID.551	Video Interoperability	Video Call between TD 1700 MXP to 7985 with TRP On	Verify video call is established between TD 1700 MXP to 7985 when TRP is enabled on 7985.	Polycom -- Video GK -RAS Agg ---ABI Unified CM --- SIP Trunk --- MOVi	Passed	
UC851IF.VID.552	Video Interoperability	C20 , E20 Behind Unified Communications Manager Establish Conference with Codian Conference Bridge	Verify C20 , E20 registered as SIP endpoint to call manager can schedule the conference and attend the conference.	Polycom -- Video GK -RAS Agg ---ABI Unified CM --- SIP Trunk --- MOVi	Passed	
UC851IF.VID.701	Video Interoperability	Call from Third Party Video Endpoint to Audio Only Phone	Verify point to point video call from Lifesize Express to an audio only phone works.	Life Size -->Unified CM -->79xx	Failed	CSCti19593

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.VID.702	Video Interoperability	Call from Third Party Endpoint to Cisco IP Communicator over SIP Trunk	Verify if point to point video call from Life Size to Cisco IP Communicator works, and there is audio and video for the call.	Life Size -->Unified CM -->SIPT -->Unified CM -->Cisco IP Communicator	Passed	
UC851IF.VID.703	Video Interoperability	Call transfer from a Tandberg Video Endpoint to Lifesize involving 9971	Verify Call Transfer is Successful from a Tandberg E20 to a Lifesize Video Endpoint.	RT 9971 -->Unified CM -->Tandberg E20 -->Xfer -->Lifesize Express 220	Passed	
UC851IF.VID.817	Video Interoperability	Multiparty Conference using Life-size , CTS 3000 and Unified Presence over SIP Trunk	Verify multiparty conference between LifeSize registered to MSP Unified Communications Manager and CTS 3000 registered to ABI Unified Communications Manager and Cisco Unified Personal Communicator registered to MSP Unified Communications Manager.	CTS 3000 --- CTMS --- conference : LifeSize and Unified Personal Communicator	Passed	
UC851IF.VID.836	Video Interoperability	911 call from TD E20	Verify whether the TD E20 is able to able call local emergency	E20 -->Unified CM -->CER	Passed	
UC851IF.VID.837	Video Interoperability	CTS 3000 , CTS 1000 and EMCC Phone Video conference	Verify if the EMCC phone is able to join video conference initiated by Cisco TelePresence Multipoint Switch (CTMS).	CTMS : Conference --- EMCC Phone Registered to ABI Cluster residing in MSP CTS 3000 and CTS 1000 in SME cluster	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC851IF.VID.849	Video Interoperability	Network Congestion : Tolerance to Video Packet Loss	Verify if the audio call stays up when delay is introduced over the WAN Link while the parties are connected to meeting place conference.	9971 --- MSP Unified CM ---ABI ---Meeting place	Passed	
UC851IF.VID.866	Video Interoperability	Point to Point call from Unified Personal Communicator to E20 Registered to VCS	Verify that there is two-way video for a point to point call between Unified Personal Communicator and E20 registered to VCS.	CUPC -->Unified CM -->SIPT -->VCS -->E20	Passed	

Regression Tests

Folders	Pending	Blocked	Passed	Pass w/ X	Failed	Dropped	Total
NA Automated Regression Tests	0 (0.0%)	0 (0.0%)	1,436 (97.2%)	2 (0.1%)	29 (2.0%)	11 (0.7%)	1478
CCM-BASIC	0 (0.0%)	0 (0.0%)	228 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	228
CCM-CFWD	0 (0.0%)	0 (0.0%)	33 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	33
CCM-CONF	0 (0.0%)	0 (0.0%)	60 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	60
CCM-EMOB	0 (0.0%)	0 (0.0%)	7 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	7
CCM-INTER	0 (0.0%)	0 (0.0%)	18 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	18
CCM-MISC	0 (0.0%)	0 (0.0%)	86 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	86
CCM-SHARED	0 (0.0%)	0 (0.0%)	32 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	32
CCM-XFER	0 (0.0%)	0 (0.0%)	44 (97.8%)	0 (0.0%)	1 (2.2%)	0 (0.0%)	45
CME-BASIC	0 (0.0%)	0 (0.0%)	14 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	14
CME-CFWD	0 (0.0%)	0 (0.0%)	21 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	21
CME-CONF	0 (0.0%)	0 (0.0%)	38 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	38
CME-MISC	0 (0.0%)	0 (0.0%)	13 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	13
CME-XFER	0 (0.0%)	0 (0.0%)	27 (96.4%)	0 (0.0%)	1 (3.6%)	0 (0.0%)	28
CUE	0 (0.0%)	0 (0.0%)	14 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	14
ENDPOINTS	0 (0.0%)	0 (0.0%)	2 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	2
FAILOVER	0 (0.0%)	0 (0.0%)	18 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	18
FAXMOD	0 (0.0%)	0 (0.0%)	40 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	40
GW-SIP	0 (0.0%)	0 (0.0%)	8 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	8
ICT	0 (0.0%)	0 (0.0%)	26 (96.3%)	0 (0.0%)	1 (3.7%)	0 (0.0%)	27
INTEROP	0 (0.0%)	0 (0.0%)	39 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	39
IPCCX	0 (0.0%)	0 (0.0%)	79 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	79
IPMA	0 (0.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	0
MP	0 (0.0%)	0 (0.0%)	8 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	8
MPE	0 (0.0%)	0 (0.0%)	31 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	31
QOS	0 (0.0%)	0 (0.0%)	61 (63.5%)	0 (0.0%)	24 (25.0%)	11 (11.5%)	96
SECURITY	0 (0.0%)	0 (0.0%)	53 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	53
SRST	0 (0.0%)	0 (0.0%)	47 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	47
UNC	0 (0.0%)	0 (0.0%)	45 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	45
UNITY	0 (0.0%)	0 (0.0%)	88 (97.8%)	0 (0.0%)	2 (2.2%)	0 (0.0%)	90
VIDEO	0 (0.0%)	0 (0.0%)	44 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	44
WAN	0 (0.0%)	0 (0.0%)	6 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	6
Auto Express	0 (0.0%)	0 (0.0%)	88 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	88
WIRELESS	0 (0.0%)	0 (0.0%)	3 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	3

Folders	Pending	Blocked	Passed	Pass w/ X	Failed	Dropped	Total
New for UC 8.0(2)	0 (0.0%)	0 (0.0%)	26 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	26
New for UC 8.5(1)	0 (0.0%)	0 (0.0%)	60 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	60
EUEM Automated Regression Tests	0 (0.0%)	0 (0.0%)	119 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	120
CCM-BASIC	0 (0.0%)	0 (0.0%)	1 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	1
CCM-CFWD	0 (0.0%)	0 (0.0%)	2 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	2
CCM-CONF	0 (0.0%)	0 (0.0%)	2 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	2
CCM-XFER	0 (0.0%)	0 (0.0%)	4 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	4
CME-BASIC	0 (0.0%)	0 (0.0%)	4 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	4
CME-CONF	0 (0.0%)	0 (0.0%)	2 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	2
CUE	0 (0.0%)	0 (0.0%)	8 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	8
FAILOVER	0 (0.0%)	0 (0.0%)	1 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	1
GW-SIP	0 (0.0%)	0 (0.0%)	1 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	1
UNC	0 (0.0%)	0 (0.0%)	13 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	13
UNITY	0 (0.0%)	0 (0.0%)	1 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	1
E2E RSVP	0 (0.0%)	0 (0.0%)	15 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	15
SMB	0 (0.0%)	0 (0.0%)	28 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	28
SRST	0 (0.0%)	0 (0.0%)	6 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	6
QSIG	0 (0.0%)	0 (0.0%)	3 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	4
IPv6	0 (0.0%)	0 (0.0%)	12 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	12
Logical Partition	0 (0.0%)	0 (0.0%)	17 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	17
RT-Phones	0 (0.0%)	0 (0.0%)	5 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	5