Test Results for Cisco Unified Communications System Release 8.6(1a) for Japanese

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Japan Headquarters
Cisco Systems G.K
Midtown Tower Building 9-7-1
Akasaka, Minato-Ku
Tokyo 107-6227
Japan
http://www.cisco.com
Tel: +81-3-6434-6500
Fax: +81-3-6434-6501

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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.

Note The current release focusses on testing the UC components in UCS infrastructure. Refer to Environment Matrix for version details.

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

- Cisco Unified Communications Manager
- Cisco Unity Connection
- Cisco Unified Presence
- Cisco Unified Border Element
- Cisco Unified Survivable Remote Site Telephony
- Cisco Unified Communications Manager Express
- Cisco Unified IP Phone
- Cisco Unified Personal Communicator
- CUCI-LYNC (CUCIMOC)
- Cisco Unified Contact Center Express
- Analog TelePhone Adapter
- Cisco IP Communicator
- Upgrade

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
CoW	Clustering over WAN
CSD	Cisco Supervisor Desktop
CSS	Calling Search Space
CSQ	Contact Service Queue
CTI	Computer Telephony Interface
CU	Cisco Unity
CUC	Cisco Unity Connection
CUCI-LYNC	Cisco UC Integration for LYNC
CUP	Cisco Unified Presence
CUCM	Cisco Unified Communications Manager
CUCIMOC	Cisco UC Integration for Microsoft Office Communicator

Acronym	Description
CUPC	Cisco Unified Personal Communicator
CUPS	Cisco Unified Presence Server
DCR	Device and Credential Repository
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
HA	High Availability
ICT	Inter-cluster 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
NTP	Network Time Protocol
POTS	Plain Old Telephony System
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

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

CHAPTER 2

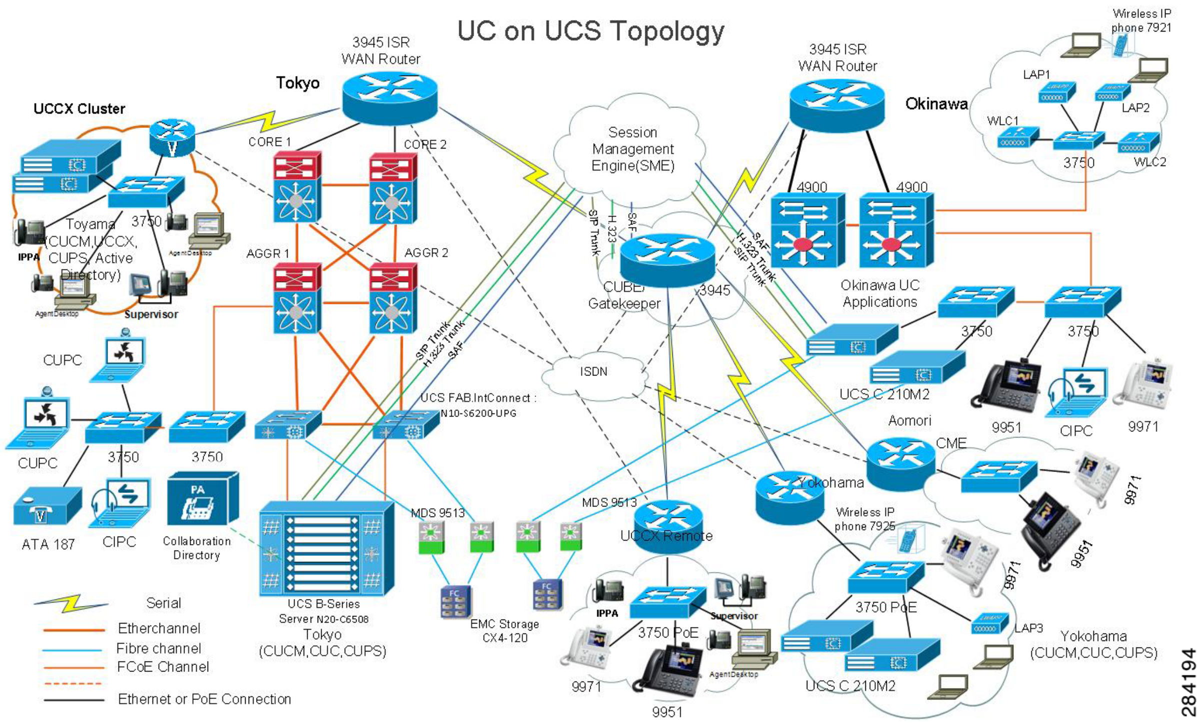
Test Topology and Environment

This chapter gives information on the following sections:

- Test Topology
- Environment Matrix
- What's New?
- Open Caveats

Test Topology

Figure 2-1 Topology in Use



Environment Matrix

Category	Component	Version	
Call Control	Cisco Unified Communications Manager	Version	CUCM-8.6.1.21004-1
		Locale	JP(8.6.1.1000-1)
		Dial plan	dp-ffr.3-1-9.JP.cop.sgn
	Cisco Unified Survivable Remote Site Telephony (SRST)	Version	8.6.1
		IOS	15.1(4)M1
	Cisco Unified Communications Manager Express	Version	8.6.1
		IOS	15.1(4)M1
		Locale	JP(8.6.2.4)
Applications	Cisco Unified Presence	Version	8.6.1.10000-34
		Locale	JP(8.6.1.9902-175)
	Collaboration Directory	Version	2.0.1
Contact Center	Cisco Unified Contact Center Express	Version	8.5.1.11001-35
Contact Center Endpoints	Cisco Supervisor Desktop	Version	8.5.1
	Cisco Agent Desktop	Version	8.5.1
	Cisco Agent Desktop Browser Edition	Version	8.5.1
	IP Phone Agent		
Voice Mail and Unified Messaging	Cisco Unity Connection	Version	CUC-8.6.1ES3.21004-3
		Locale	JP(8.6.1.1-2)

Category	Component	Version	
Endpoints and Clients	Cisco Unified IP Phones		
	6921		SCCP69xx.9-2-1-0/SIP69xx.9-2-1-0
	6941		SCCP69xx.9-2-1-0/SIP69xx.9-2-1-0
	6961		SCCP69xx.9-2-1-0/SIP69xx.9-2-1-0
	7961G		SCCP41.9-2-1S
	7975		SCCP75.9-2-1S
	7985		cmterm_7985.4-1-7-0
	8961		sip8961.9-2-1
	9951		sip9951.9-2-1
	9971		sip9971.9-2-1
	7925G		CP7925G-1.4.1SR1
	7921G		CP7921G-1.4.1SR1
	7937G		apps37sccp.1-4-4-0
	Cisco IP Communicator		cipc-Admin-ffr.8-6-1-0
	ATA 187		ata187.9-2-1-0
	Cisco Unified Personal Communicator		8.5.1.18771
	Cisco UC Integration for Microsoft Lync		8.5.3
Communications Infrastructure	Cisco IOS Voice and Data Gateways	IOS	15.1(4)M1
UCS	UCS Infrastructure Bundle		ucs-k9-bundle-infra.1.4.3i.A
	UCS Manager		ucs-k9-bundle-b-series.1.4.3i.B.bin
	ESXi host on Blade one	Blade Server-1	ESXi 4.1
	Nexus 1000V	Nexux 1kV	nexus-1000v-mz.4.0.4.SV1.3b.bin
	VCenter Server	Laptop	ESX 4.1
	MDS Switch	M9500	m9500-sf2ek9-mz.5.0.4.bin sup-1 m9500-sf2ek9-kickstart-mz.5.0.4.bin sup-2
Client	Operating System	Win-XP	Windows XP - SP3 (Japanese)
		Windows 7	Windows 7 - SP1 (Japanese)
	Browser	IE	IE 8
Wireless			
Controller	Wireless LAN controller 4402	7.0.116.0	
	Wireless LAN controller 5508	7.0.116.0	

Category	Component	Version	
Applications	Wireless Control system	7.0.172.0	
	Cisco ACS	4.2	
Access Point	Wireless Access point 1142	12.4	
	Wireless Access point 35XX	12.4	
Client	OS	Windows XP SP 2 Japanese	
	Browser	IE	7.0 Japanese
		Mozilla Firefox	4.0 Japanese

What's New?

The following table describes the new features in Cisco Unified Communications System release 8.6(1a):

Table 2-1 New Features

New Feature	Description
Partitioned Intra domain federation feature overview	In the partitioned model, a single domain has multiple servers, each of which manages a non-overlapping set of users. For each user in the domain, their presence data, policy and IM handling reside on a single server.
Cross Cluster Redirect	User who connects to any CUP node will be redirected to his correct home node whether that is in the same or a different cluster.
Localization Support for SIP IP Phones in CME 8.6	Adds localization support for SIP IP Phones.
Video in CME 8.6	Video and camera support for Cisco Unified IP Phones 9951 and 9971 in CME 8.6

Open Caveats

Open caveats describe the possible unexpected behaviors that you may encounter in release 8.6(1a) of Cisco Unified Communications System.

Defect ID	Defect Title
CSCtq61347	In 6900 Phones no IPv6 option is available in network settings after applying Japanese locale
CSCtq67517	gh-sip.jar missing in locale-ja_JP-Japanese-8.6.2.4.tar
CSCtr32471	Insert of Phones/Users in BAT does not associate Primary Extension for Users
CSCtq58601	In 6900 series IP Phones BLFspeed dial Button displays in English
CSCtr32784	Issue regarding Personal Directory Logout for 69xx phones
CSCtr51513	Conference Message is showing in English instead of Japanese
CSCtr52906	"Private" displays on the Calling Party instead of Conference
CSCtr56948	Number is showing in ENGLISH language instead of JAPANESE.
CSCtr40396	Blank Display in the Analog Phone Connected to ATA
CSCtq58578	In 6900 Phone Single number reach option displays in English
CSCtt21815	CUC 8.6 Japanese option is not shown in System Default Language

Limitations

1. 6945 IP Phones will be supported in CME 8.8
2. ATA 187 not supported in CME 8.6

Known Issues

Defect ID	Defect Title
CSCtj32839	MX TImeout happens after RESUME a call.
CSCtl03266	CUCM:JPN:7937: XML error occurs on Extension Mobility

CHAPTER 3

Test Results Summary

This chapter contains the following sections:

- Cisco Unified Communications Manager
- Cisco Unity Connection
- Cisco Unified Presence
- Cisco Unified Border Element
- Cisco Unified Survivable Remote Site Telephony
- Cisco Unified Communications Manager Express
- Cisco Unified IP Phone
- Cisco Unified Personal Communicator
- CUCI-LYNC (CUCIMOC)
- Cisco Unified Contact Center Express
- Analog TelePhone Adapter
- Cisco IP Communicator
- vMotion
- Upgrade
- Regression Testing
- Related Documentation

Cisco Unified Communications Manager

Logical ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UC861S.CUCM.D.101	BAT	BAT: Primary Extension field on the user page associated to DN when BAT file with partition and Phone Line Template DN with partition	Verify that Primary Extension field on the user page is associated to the directory number after adding users and phones through BAT tool when BAT file with partition and Phone Line Template DN with partition		Passed	
UC861S.CUCM.D.103	BAT	BAT: Primary Extension field on the user page associated to DN when BAT file without partition and Phone Line Template DN with partition	Verify that Primary Extension field on the user page is associated to the directory number after adding users and phones through BAT tool when BAT file without partition and Phone Line Template DN with partition		Passed	
UC861S.CUCM.D.104	BAT	BAT: Primary Extension field on the user page associated to DN when BAT file with partition and Phone Line Template DN without partition	Verify that Primary Extension field on the user page is associated to the directory number after adding Users and Phones through BAT tool when BAT file with Partition and Phone Line Template DN without partition		Passed	
UC861S.CUCM.D.105	Dial Plan	Upgrade Japanese dial plan	Verify that Japanese dial plan upgrade is successful		Passed	
UC861S.CUCM.D.106	Dial Plan	Upgrade Japanese dial plan to the same version	Verify that Cisco Unified Communication Manager does not allow to upgrade Japanese dial plan to the same version		Passed	

Logical ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UC861S.CUCM.D.107	Hold and Resume	Hold and Resume the Inter cluster Video call on 7985 IP Phone	Verify that hold and resume on active video call between 9971 and 7985 IP Phone works successfully	7985 IP Phone -> Cisco Unified Communication Manager 1 -> SIP EO Trunk -> Cisco Unified Communication Manager 2 -> 9971 IP Phone7985 IP Phone -> Hold and Resume	Passed	
UC861S.CUCM.D.108	Hold and Resume	Hold and Resume the Inter cluster Video call on 9971 IP Phone.	Verify that hold and resume on active video call between 9971 IP Phone works successfully	9971 IP Phone A -> Cisco Unified Communication Manager 1 -> SIP EO Trunk -> Cisco Unified Communication Manager 2 -> 9971 IP Phone B9971 IP Phone A -> Hold and Resumes	Passed	

Logical ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UC861S.CUCM.D.109	Hold and Resume	Hold and Resume the Inter cluster Video call on 9951 IP Phone.	Verify that hold and resume on active video call between 8961 and 9971 IP Phone works successfully	9951 IP Phone A -> Cisco Unified Communication Manager 1 -> SIP EO Trunk -> Cisco Unified Communication Manager 2 -> 9971 IP Phone B9951 IP Phone A -> Hold and Resume	Passed	
UC861S.CUCM.D.110	IPV6	Call between IPv4 and IPv6 endpoints (EO).	Verify that call between IPv4 and IPv6 endpoints through Early Offer enabled SIP Trunk with MTP required is successful		Passed	
UC861S.CUCM.D.111	IPV6	Call between IPv4 and IPv6 endpoints (DO).	Verify that call between IPv4 and IPv6 endpoints through Delayed offer enabled SIP Trunk with MTP required is successful		Passed	
UC861S.CUCM.D.113	BAT	Add a new phone using the BAT Phone Template	Verify that the new phone is added successfully by using the BAT Phone Template which is subscribed to phone service		Passed	
UC861S.CUCM.D.114	BAT	Add a new phone using the BAT Phone Template	Verify that the new phone is added successfully by using the BAT Phone Template with Extension Mobility option enabled		Passed	
UC861S.CUCM.D.115	BAT	Add a new phone using the BAT Phone Template	Verify that the new phone is added successfully by using the BAT Phone Template with Do Not Disturb option enabled		Passed	

Logical ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UC861S.CUCM.D.116		Cisco Unity Connection service activation page	Verify that AXL/UXL services are not available in the Cisco unity connection service activation page		Passed	
UC861S.CUCM.D.117		“Utils Diagnose test” command in Cisco Unified Communications Manager CLI	Verify that “Utils Diagnose test” command in Cisco Unified Communications Manager CLI does not result in any error for tomcat_sessions.		Passed	
UC861S.CUCM.D.118		“Utils Diagnose test” command in Cisco Unified Communications Manager CLI	Verify that “Utils Diagnose test” command in Cisco Unified Communications Manager CLI does not result in any error for validate_network.		Passed	
UC861S.CUCM.D.119		“Utils Diagnose fix” command in Cisco Unified Communications Manager CLI	Verify that “Utils Diagnose fix” command in Cisco Unified Communications Manager CLI does not result in any error.		Passed	
UC861S.CUCM.D.120		Validate length for the fields Organization, Unit and Location	Verify that length validation works successfully for the fields Organization, Unit and Location while installing Cisco Unified Communications Manager		Passed	
UC861S.CUCM.D.121		Network Services on Cisco Unified Communications Manager serviceability page	Verify that network services are up and running fine in Cisco Unified Communications Manager migrated from MCS to UCS		Passed	
UC861S.CUCM.D.122		Feature Services on Cisco Unified Communications Manager serviceability page	Verify that feature services are up and running fine in Cisco Unified Communications Manager migrated from MCS to UCS.		Passed	
UCJ86IF.CUCM.001	RTMT	RTMT alerts after upgrade Cisco Unified Communications Manager from 8.5(1) to 8.6(1a)	Verify that RTMT tool shows some alerts after upgrading Cisco Unified Communications Manager from 8.5(1) to 8.6(1a)		Passed	

Logical ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UCJ86IF.CUCM.002	Cisco Unified Communications Manager	Resources displaying status while creating new resource called User Web pages	Verify that all resources and options displayed before adding another resource in a new configured role information		Passed	
UCJ86IF.CUCM.004	Cisco unified serviceability	Cisco Serviceability Reporter service on Cisco Unified Communications Manager Serviceability page	Verify that Cisco Serviceability Reporter service is up and running fine in Cisco Unified Communications Manager upgraded from 8.5 to 8.6(1a)		Passed	
UCJ86IF.CUCM.005	Cisco Unified Communications Manager	Choosing an appropriate time schedule to partition when time schedule numbers crosses more than 250	Verify that the **Find** button on partition page to select an appropriate time schedule works fine when time schedule numbers reach the limit		Passed	
UCJ86IF.CUCM.006	Cisco Unified Communications Manager	Change in SIP sequence after upgrading Cisco Unified Communications Manager to 8.6(1a)	Verify that Call Manager sends 180 ringing to ISR successfully when Unified IP Phone A (connected to ISR) calls Unified IP Phone B		Passed	
UCJ86IF.CUCM.009	Upgrade	Database replication for newly installed Publisher using DRS(Disaster Recovery System)	Verify that Database replication occurs successfully in Cisco Unified Communications Manager migrated from MCS to UCS		Passed	
UCJ86IF.CUCM.010	Disaster Recovery System	Manual back up for CCM Feature	Verify that manual back up of all the components of CCM feature is successful using Disaster Recovery System		Passed	
UCJ86IF.CUCM.011	Disaster Recovery System	Manual back up for CDR-CAR Feature	Verify that manual back up of all the components of CDR-CAR feature is successful using Disaster Recovery System		Passed	
UCJ86IF.CUCM.012	Disaster Recovery System	Scheduled back up for CDR-CAR Feature	Verify that scheduled back up of all the components of CDR-CAR feature is successful using Disaster Recovery System		Passed	

Logical ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UCJ86IF.CUCM .013	Disaster Recovery System	Scheduled back up for CCM Feature	Verify that scheduled back up of all the components of CCM feature is successful using Disaster Recovery System		Passed	
UCJ86IF.CUCM .014	Upgrade	CMI (Cisco Messaging Interface) service availability status after upgrading Cisco Unified Communications Manager to 8.6(1a)	Verify that CMI (Cisco Messaging Interface) service in service name list is available after upgrading Cisco Unified Communications Manager from 8.5(1)to 8.6(1a)		Passed	
UCJ86IF.CUCM .0015	Upgrade	Error level trace when CMI (Cisco Messaging Interface) service failure	Verify that the failure errors in the traces at error level are printed when CMI (Cisco Messaging Interface) service fails to start, after upgrading Cisco Unified Communications Manager from 8.5(1) to 8.6(1a)		Passed	
UCJ86IF.CUCM .018	Cisco Unified Communi cations Manager	Display status for Called Party Name	Verify that Called Party Name displays successfully when Analog Phone calls Unified IP phone	PSTN -> GW -> (H323) -> CUCM -> (SIP or SCCP) -> Unified IP phone	Passed	
UCJ86IF.CUCM .019	Upgrade	Locale Upgrade for Cisco Unified Communications Manager	Verify that FTP server locates files to upgrade locale if folder contains another file name with special characters		Passed	
UCJ86IF.CUCM .203	Extension Mobility	Locale status when user logs into Unified IP Phone using Extension Mobility	Verify that Japanese locale is shown correctly on Unified IP Phone when a user logs in using Extension Mobility		Passed	
UCJ86IF.CUCM .204	BLF Dpark	BLF Dpark status when Unified IP Phone park and retrieves call	Verify that BLF Dpark button turn off on Unified IP Phone A after call is parked and retrieved using BLF Dpark button		Passed	

Logical ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UCJ86IF.CUCM .205	Transfer	Consultative transfer to CUPC desk phone	Verify that consultative transfer occurs successfully between 6941 Unified IP Phone,9971 Unified IP Phone and CUPC desk phone		Passed	
UCJ86IF.CUCM .206	Unified IP Phone	Audio status when video call between two IP Phones from different clusters	Verify that audio is good when video call is initiated between two 9971 IP Phones placed in different clusters if audio bandwidth is sufficient and video bandwidth is lacking		Passed	
UCJ86IF.CUCM .207	Unified IP Phone	Audio status when video call between Unified IP Phone and CUPC(soft phone) from different cluster	Verify that audio is good when video call is initiated between 9971 Unified IP Phone and CUPC (soft phone) placed in different clusters if audio bandwidth is sufficient and video bandwidth is lacking		Passed	
UCJ86IF.CUCM .208	Unified IP Phone	IP Phone detection process for SRST after Cisco Unified Communications manager fails	Verify that Unified IP Phone detection process for SRST takes more time than normal when Detect Unified CM connection Failure set to Delayed in Cisco Unified Communications manager fails situation		Passed	
UCJ86IF.CUCM .209	Unified IP Phone	Call from 7961 Unified IP Phone after upgrading Cisco Unified Communications Manager from 8.5(1) to 8.6(1a)	Verify that 7961 Unified IP Phone is able to call other Unified IP Phones after upgrading Cisco Unified Communications manager from 8.5(1) to 8.6(1a)		Passed	
UCJ86IF.CUCM .210	Fast Dial	Call from 7975 Unified IP Phone using Fast dial	Verify that call from 7975 Unified IP Phone to any other Unified IP Phone works fine using Fast Dial		Passed	
UC861S.cucm.T .101	Personal directory	Login to Unified IP Phone Personal Directory – 99XX IP Phone	Verify that user is able to login to Personal Directory after exiting from the same in 99xx series Unified IP Phone.		Passed	

Logical ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UC861S.cucm.T.102	Personal directory	Login to Unified IP Phone Personal Directory – 69XX IP Phone	Verify that user is able to login to Personal Directory after exiting from the same in 69XX series Unified IP Phone		Passed	
UC861S.cucm.T.103	MOH	Unicast Music on Hold on IP Phone	Verify that IP Phone configured for Unicast Music on Hold hears hold tone		Passed	
UC861S.cucm.T.104	MOH	Multicast Music on Hold on IP Phone	Verify that IP Phone configured for Multicast Music on Hold hears hold tone		Passed	
UC861S.cucm.T.105	MOH	Music on Hold to off-net user (MGCP gateway) placed on hold	Verify that an analog phone connected through MGCP gateway hears Music on Hold, when the phone has been placed on hold.		Passed	
UC861S.cucm.T.106	MOH	Music on Hold to off-net user (H323 gateway) placed on hold	Verify that an analog phone connected through H323 gateway hears Music on Hold, when the phone has been placed on hold.		Passed	
UC861S.cucm.T.107	Transfer	Call transfer to a busy phone over H323 gateway	Verify that calling party hears busy tone when called party transfers the call to a busy phone		Passed	
UC861S.cucm.T.108	Transfer	Call transfer to a busy phone over MGCP gateway	Verify that calling party gets busy tone when called party transfers the call to a busy phone		Passed	
UC861S.CUCM.T.006	DRS	Manual Backup using DRS	Verify that Manual Backup of Cisco Unified Communication Manager using DRS is completed successfully		Passed	
UC861S.CUCM.T.007	DRS	Deletion of Backup device in a backup schedule.	Verify that Backup device in a backup schedule cannot be deleted.		Passed	
UC861S.CUCM.T.008	DRS	Restoring Publisher using DRS	Verify that restoring Cisco Unified Communication Manager Publisher data using DRS is successful		Passed	

Logical ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UC861S.CUCM.T.009	DRS	Restoring Subscriber with rebuild using DRS	Verify that restoring Cisco Unified Communication Manager Subscriber data using DRS is successful		Passed	
UC86IF.CUCM.U.001	Conference	ConfList Feature for Adhoc conference	Verify that the conflist softkey shows all the connected participants for an Adhoc conference in Unified IP Phone 9971/9951 and Unified IP Phone 7961.		Passed	
UC86IF.CUCM.U.002	Conference	ConfList Feature with Remove participants from Connected Conference	Verify that using the Remove softkey feature in the conference controller we can remove participants from connected conference.		Passed	
UC86IF.CUCM.U.003	Conference	Conflist for Connected Conference with maximum entries	Verify that the ConfList softkey shows all the participants in a Connected Conference.		Passed	
UC86IF.CUCM.U.004	Conference	ConfList feature for Conference in a shared line	Verify that the conference controller, configured with shared line is able to hold and resume the conference, which is from a remote shared line.		Passed	
UC86IF.CUCM.U.005	Conference	ConfList feature when Cisco Unified Communications Manager service is shutdown	Verify that the ConfList softkey shows the call preservation status of the Unified IP Phone when Cisco Unified Communications Manager service is shutdown.		Passed	
UC86IF.CUCM.U.006	Conference	ConfList for Park	Verify that the conference controller is able to park and retrieve the conference from another Unifed IP phone. The confList softkey displays the parked and retrieved status accordingly.		Passed	

Logical ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UC86IF.CUCM.U.007	Conference	ConfList with Park reversion timer	Verify that after the park reversion timer expires for a parked conference, the conference controller is able to join back the conference and the confList softkey shows the updated rejoined status.		Passed	
UC86IF.CUCM.U.008	Conference	ConfList Feature with DN change	Verify that the ConfList softkey does not display the changed Directory number of the Unified IP Phone in a conference.		Passed	
UC86IF.CUCM.U.009	Conference	Advanced consultative (adhoc) conference	Verify that the ConfList softkey displays the remove and update softkeys for all the participants in advanced consultative conference.		Passed	
UC86IF.CUCM.U.010	Conference	ConfList Feature for Conference Transfer	Verify that the conference controller is able to transfer the conference to a call Forwarded activated Unified IP phone and the confList softkey displays the updated conference status.		Passed	
UC86IF.CUCM.U.011	Cisco Unified Communications Manager	Auto Pickup feature in a shared line	Verify that the Auto Pickup feature works successfully in a shared line.		Passed	
UC86IF.CUCM.U.012	Call Park	Assisted BLF Dpark for a transferred call	Verify the following scenario: • A call is parked using Assisted BLF Dpark in Unified IP Phone A. • The Unified IP Phone B transfers the call to another Unified IP Phone 9971/9951. • The Unified IP Phone A retrieves the transferred call		Passed	

Logical ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UC86IF.CUCM.U.013	Conference	ConfList for Conference across trunk	Verify that the conference controller is able to remove a participant, who is registered to a different Cisco Unified Communications Manager present in a different cluster and connected through a trunk.		Passed	
UC86IF.CUCM.U.014	Conference	ConfList Feature for Conference within the same cluster	Verify that the conference controller is able to remove a participant, who is registered to different Cisco Unified Communications Manager present in the same cluster.		Passed	
UC86IF.CUCM.U.015	Conference	ConfList with Park reversion timer	Verify that after the Park reversion timer expires for a parked conference, the conference controller is able to join back the conference and the confList/softkey shows updated rejoined status.		Passed	
UC86IF.CUCM.U.016	Call Park	Assisted BLF Dpark with Idivert	Verify that a call reverts to configured DN successfully after reversion timer expires and then IDivert the call. Verify the MWI status.		Passed	
UC86IF.CUCM.U.017	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 IPPhone -> Cisco Unified Communications Manager -> Cisco Unified IP Phone -> Call Park and Blind Transfer	Passed	

Logical ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UC86IF.CUCM.U.018	Call Park	Assisted BLF Dpark on same Dpark code simultaneously on different node	Verify the call park behavior in Assisted BLF Dpark, when the Dpark code is pressed simultaneously on different nodes.		Passed	
UC86IF.CUCM.U.019	Call Park	Assisted BLF Dpark and revert the call in Meet me conference	Verify that a call is parked using Assisted BLF Dpark and revert the call in Meet me conference after reversion timer expires.		Passed	
UC86IF.CUCM.U.020	Cisco Unified Communications Manager	MOH should be available to others when Phone A does Call Park and Blind Transfer	Verify whether MOH is available to others when Phone A does Call Park and Blind Transfer		Passed	
UC86IF.CUCM.U.021	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	
UC86IF.CUCM.U.022	Conference	ConfList Feature for a chained conference	Verify that the ConfList softkey displays the conference status, when the meet-me and consultative conference is barged.		Passed	
UC86IF.CUCM.U.023	Call Park	Assisted BLF Dpark with parked user is disconnected	Verify the BLF Dpark button behaviour for a parked call, when it is disconnected.		Passed	

Logical ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UC86IF.CUCM.U.024	Call Park	Assisted BLF Dpark for a transferred call and call reversion	Verify the following scenario: • A call is parked using Assisted BLF Dpark in Unified IP Phone A. • The Unified IP Phone B transfers the call to another Unified IP Phone 9971/9951. • The call is reverted to Unified IP Phone A after reversion timer expires		Passed	
UC86IF.CUCM.U.025	Cisco Unified Communications Manager	Privacy settings in shared lines	Verify that privacy on hold toggling setting in the sharedline works successfully.		Passed	
UC86IF.CUCM.U.026	Call Park	Call revert to configured DN, after the reversion timer expires	Verify that a call reverts to configured DN successfully after reversion timer expires.		Passed	
UC86IF.CUCM.U.027	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	
UC861S.CUCM.T.501	Call Pickup	6921 IP phone picks up the call with large pickup group number	Verify that 6921 Unified IP Phone is able to pick up the incoming call on a PickUp group with large pick up number using Group pickup.	Unified IP Phone A -> Unified IP Phone B (Call Pickup group 1,5555555555) -> Unified IP Phone C (6921, Call pick up group 2) -> Gpickup -> 5555555555	Passed	

Logical ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UC861S.CUCM.T.502	Call Pickup	9951 IP phone picks up the call with large pickup group number	Verify that 9951 Unified IP Phone is able to pick up the incoming call on a PickUp group with large pick up number using Group pickup.	Unified IP Phone A -> Unified IP Phone B (Call Pickup group 1,5555555555) -> Unified IP Phone C (9951, Call pick up group 2) -> Gpickup -> 5555555555	Passed	
UC861S.CUCM.T.503	Call Pickup	7937 IP phone picks up the call with large pickup group number	Verify that 7937 Unified IP Phone is able to pick up the incoming call on a PickUp group with large pick up number using Group pickup.	Unified IP Phone A -> Unified IP Phone B (Call Pickup group 1,5555555555) -> Unified IP Phone C (7937, Call pick up group 2) -> Gpickup -> 5555555555	Passed	
UC861S.CUCM.D.102	BAT	BAT: Primary Extension field on the user page associated to DN when BAT file without partition and Phone Line Template DN without partition	Verify that Primary Extension field on the user page is associated to the directory number after adding users and phones through BAT tool when BAT file without partition and Phone Line Template DN without partition		Failed	CSCtr32471
UC861S.cucm.T.109	Personal Directory Logout for Unified IP Phone 99XX	Personal Directory	Verify that user is able to logout from Personal Directory and login again to Personal Directory using credentials in 99XX Unified IP Phones		Failed	CSCtr32626

Logical ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UC861S.cucm.T.110	Personal Directory Logout for Unified IP Phone 69XX	Personal Directory	Verify that user is able to logout from Personal Directory and again login to Personal Directory using credentials in 69XX Unified IP Phones		Failed	CSCtr32784

Logical ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UC861S.CUCM.D.123	SME	Conference through SME	Verify that Conference displays in Japanese successfully when IPPF Phone and Unified IP Phone conferences another Unified IP Phone through SME	Analog Phone A -->IPPF--> Cisco Unified Communications Manager (8.6(1a))--> Unified IP Phone B (conference)-->SME--> Cisco Unified Communications Manager (8.6(1a)) --> Cisco Unified Communications Manager (8.6(1a) --> Unified IP Phone C	Failed	CSCtr51513

Logical ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UC861S.CUCM.D.124	Conference	Conference through Inter-Cluster trunk	Verify that conference message displays successfully when Unified IP Phone A calls Unified IP Phone B through SME and Unified IP Phone B conferences Unified IP Phone C	Unified IP Phone A --> Cisco Unified Communications Manager A --> ICT trunk --> SME Cisco Unified Communications Manager --> ICT --> Cisco Unified Communications Manager B --> Unified IP Phone B --> Conf --> ICT/SIP --> Cisco Unified Communications Manager A --> Unified IP PhoneC	Failed	CSCtr52906

Cisco Unity Connection

ID	Feature teated	Case Title	Description	Call Component Flow	Status	Defects
UCJ86IF.CUC.001	Cisco Unity Connection	Unity Connection Bulk Edit changes	Verify that Bulk Edit feature will modify fields which are not selected for intended changes		Passed	
UCJ86IF.CUC.002	Cisco Unity Connection	Voicemail should play sender’s voice name	Verify that voice mail should play sender’s name	Unified IP Phone A -> Unified Communications Manager -> Phone B	Passed	
UCJ86IF.CUC.003	Cisco Unity Connection	Opening Greeting on early digit	Verify that Opening Greeting should not play on early digit	Unified IP Phone A -> unified communications -> Phone B send voice mail. Login in Phone B change pin	Passed	
UCJ86IF.CUC.004	Cisco Unity Connection PCA	User can change password in PCA (Personal communications assistant)	Verify User can change Password in PCA		Passed	
UCJ86IF.CUC.005	Cisco Unity Connection	Secure LDAP (Light Weight Directory Access Protocol)	Verify that LDAP with secure authentication is running sucessfully or not		Passed	
UCJ86IF.CUC.006	Cisco Unity Connection	Applying software as patch Unity Connection	Verify that Software Patch uploads on Unity connection properly.		Passed	
UCJ86IF.CUC.007	Cisco Unity Connection	Double quote in display name	Verify that user can send a message if the disply name have one double quote.		Passed	
UCJ86IF.CUC.008	Cisco Unity Connection	Unity Connection Security Vulnerabilities	Verify that a vulnerability exists in SSL communications when clients are allowed to connect using no authentication algorithm		Passed	
UCJ86IF.CUC.009	Cisco Unity Connection	Viewmail For Outlook 8.5.4 with Outlook max connections	Verify that Outlook and Exchange works fine after Viewmail For Outlook is installed		Passed	

ID	Feature teated	Case Title	Description	Call Component Flow	Status	Defects
UCJ861F.CUC .010	Cisco Unity Connection	Cisco Unity Connection Manager core dump	Verify that one of the thread is able to perform a port group reset operation when another thread is trying to bring the voice ports in-service via a successful ping.	1. Create a SIP Trunk between CUCM and Unity Connection -> In the Unity Connection SA go to Port Group and press the Reset button while the system is idle. -> Repeat the step 2 while there is a opening greeting playing. -> Repeat the step 2 & 3 for ~8-10 times. -> After some time repeatedly press the Reset button. -> In the RTMT captured the Crash Dump - connection conversation manager -> From the SSH run the command "utils core active list"	Passed	

ID	Feature teated	Case Title	Description	Call Component Flow	Status	Defects
UCJ86IF.CUC.011	Cisco Unity Connection	Cisco Unity Connection 8.6(1a) -Calling with unsupported audio format	Verify that user cannot upload 8khz, 22khz, 44khz .wav files via the view mail inbox new message form	Verify that user can not upload 8khz, 22khz and 44khz .wav files via the inbox new message form. Steps to verify: 1.In Cisco Unity Connection Administration, go to a page on which the Media Master appears. ->On the Media Master Options menu, select Playback & Recording. ->In the Playback & Recording Settings dialog box, select a playback device and a recording device. ->If you chose the phone as the recording and playback device in Step 3, the Active Phone Number is set by default to your primary extension. To specify a different phone number, enter it in the Other Number field. ->Select OK. Note: Please make sure PCM files are in Mono (not Stereo)."	Passed	

Cisco Unified Presence

ID	Feature teated	Case Title	Description	Call Component Flow	Status	Defects
UCJ861F.CUP.001	Cisco Unified Presence	Instant Messaging between Cisco Unified Personal Communicator 8.5 and Microsoft Office Communicator(MOC)	Verify instant messaging between Cisco Unified Personal Communicator 8.5 and Microsoft Office Communicator (MOC)		Passed	
UCJ861F.CUP.002	Cisco Unified Presence	Group Chat between Cisco Unified Personal Communicator 8.5 and Microsoft Office Communicator(MOC)	Verify that Group Chat is not supported between Cisco Unified Personal Communicator 8.5 and Microsoft Office Communicator(MOC)		Passed	
UCJ861F.CUP.003	Cisco Unified Presence	Presence status translation between Cisco Unified Presence and Microsoft Office Communicator	Verify that presence status translation between Cisco Unified Presence and Microsoft Office Communicator occurs properly.		Passed	
UCJ861F.CUP.004	Cisco Unified Presence	Contact Search Capability	Verify that the user is able to search for any other user within the enterprise regardless of provision with Cisco Unified Personal Communicator or Microsoft Office Communicator.		Passed	
UCJ861F.CUP.005	Cisco Unified Presence	Cisco Unified Personal Communicator Cross cluster redirect with HA enabled in home node's sub-cluster	Verify that client login request to different cluster is redirected to user's Home node.		Passed	
UCJ861F.CUP.006	Cisco Unified Presence	Cisco Unified Personal Communicator Cross cluster redirect with HA disabled in home node's sub-cluster	Verify that client login request to different cluster is redirected to user's Home node.		Passed	
UCJ861F.CUP.007	Cisco Unified Presence	Cisco Unified Personal Communicator Cross cluster redirect 8.6 cluster to user's 8.5 Home node.	Verify that client login request to 8.6 cluster is redirected to user's 8.5 Home node.		Passed	

ID	Feature teated	Case Title	Description	Call Component Flow	Status	Defects
UCJ861F.CUP.008	Cisco Unified Presence	DRS restore Cisco Unified Presence image into VM image	DRS restore Cisco Unified Presence image taken from MCS into UCS C210 M2 (VM image)	In MCS Sever CUP DRS -> Backup -> Manual Backup -> <DeviceName> and in UCS Server CUP DRS -> Restore -> Restore Wizard -> <DeviceName>	Passed	
UCJ861F.CUP.009	Cisco Unified Presence	Enabling System Automatically Manages Primary Group Chat Server Aliases	Verify enabling the "System Automatically Manages Primary Group Chat Server Aliases"	CUP Administrator -> Messaging -> Group Chat and Persistent Chat Settings	Passed	
UCJ861F.CUP.010	Cisco Unified Presence	Version Switch back to 7.x from 8.5.x or 8.6.x	Verify Cisco Unified Personal Communicator 7 is able to login even after doing version switch back to 7.x from 8.5.x or 8.6.x		Passed	
UCJ861F.CUP.011	Cisco Unified Presence	Upgrading Cisco Unified Presence Subscriber	Verify that Cisco Unified Presence Subscriber is not failing to upgrade after the publisher has been upgraded and rebooted		Passed	
UCJ861F.CUP.012	Cisco Unified Presence	User logging-in after restarting the "Cisco XCP services"	Verify that the Cisco Unified Personal Communicator Users are able to login after restarting the "Cisco XCP Services"		Passed	
UCJ861F.CUP.013	Cisco Unified Presence	Subscriber Cisco Unified Presence Network Service Page services list.	Verify that in a Subscriber Cisco Unified Presence Network Service Page list of all the services is displayed.		Passed	
UCJ861F.CUP.014	Cisco Unified Presence	Publisher Cisco Unified Presence Network Service Page services list.	Verify that in a Publisher Cisco Unified Presence Network Service Page is showing list of all the services.		Passed	
UCJ861F.CUP.015	Cisco Unified Presence	Media Check and installation.	Verify that installation proceeds and finishes successfully after doing the Media check.		Passed	

ID	Feature teated	Case Title	Description	Call Component Flow	Status	Defects
UCJ861F.CUP.016	Cisco Unified Presence	Media Check with new build	Verify that media check succeeds with newly downloaded build file.		Passed	
UCJ861F.CUP.017	Cisco Unified Presence	Unified Presence upgrade	Upgrade two Unified Presence cluster peers from 8.5 to 8.6. Verify that the sub-cluster logical topology remains intact and that High Avaliability is NOT enabled immediately after the upgrade. Verify that user buddy lists remain intact and intra/inter-cluster presence is functioning properly post-upgrade.		Passed	
UCJ861F.CUP.018	Cisco Unified Presence	High Avaliability over WAN client failover, manual fallback	Verify that when a Unified Presence node fails in a Clustering Over WAN (CoW) deployment, all clients and communication are able to failover automatically to another 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.		Passed	
UCJ861F.CUP.019	Cisco Unified Presence	Co-located High Avaliability client failover, manual fallback	Verify that when a Unified Presence node fails, all clients and communication are able to failover automatically to another Unified Presence node. Verify that the clients are able to fallback to original configuration via a manual system fallback.		Passed	
UCJ861F.CUP.020	Cisco Unified Presence	Clustering Over WAN(CoW) user rebalance	Assign all users from one node to another node in a Clustering Over WAN (CoW) setup. Then, perform a user rebalance.		Passed	

ID	Feature teated	Case Title	Description	Call Component Flow	Status	Defects
UCJ861F.CUP.021	Cisco Unified Presence	Inter-cluster peer fail over	Fail an inter-cluster peer node. Verify that inter-cluster functionality is retained when the server fails over to the backup.		Passed	
UCJ861F.CUP.022	Cisco Unified Presence	DND with Shared line	Verify DND ia working as expected when secondary phone unregisters from Unified Communications Manager		Passed	
UCJ861F.CUP.023	Cisco Unified Presence	Presence status in Presence viewer page	Verify Presence Viewer displays correct buddy presence when contact list contains userID with space		Passed	
UCJ861F.CUP.024	Cisco Unified Presence	Status of SIP Publish model on troubleshooter page.	Verify that the Status of SIP Publish model on troubleshooter page shows ok symbol.		Passed	
UCJ861F.CUP.025	Cisco Unified Presence	Checkmark in Unified Presence/Unified Presence Communicator Licensed column	Checkmark in Unified Presence/Unified Presence Communicator Licensed column		Passed	
UCJ861F.CUP.026	Cisco Unified Presence	Verifying Instant Message	Verify Instant Message for a particular period of time without any IM failure.		Passed	
UCJ861F.CUP.027	Cisco Unified Presence	Restart time for Unified Presence	Verify the time taken for restarting a Unified Presence.		Passed	
UCJ861F.CUP.028	Cisco Unified Presence	Sub cluster status information	Verify that the Status of the subcluster is displayed in correct localized language.		Passed	
UCJ861F.CUP.029	Cisco Unified Presence	VM Details in Admin Splash	Verify that in Unified Presence Admin Splash Screen, below System Version Information, the hardware information of the Virtual Machine is displayed.		Passed	
UCJ861F.CUP.030	Cisco Unified Presence	Adding a new contact via Unified Presence Communicator	Verify the case insensitiveness of the contacts while adding a new contact via Unified Presence Communicator.		Passed	

Cisco Unified Border Element

ID	Feature teated	Case Title	Description	Call Component Flow	Status	Defects
UCJ861S.CUBE.U.001	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 IP Phone (SIP) -> Cisco Unified Communications Manager Express -> Cisco Unified BorderElement -> Session Management Edition -> Cisco Unified IP Phone	Passed	
UCJ861S.CUBE.U.002	Cisco Unified Border Element	Invoke Transcoder in Cisco Unified Communication Manager Express, when there is codec mismatch	Verify the codec mismatch for the following call: • A call from a SIP Phone that is registered to Cisco Unified Communications Manager goes to Cisco Unified Border Element through a SIP trunk. • From Cisco Unified Border Element the call goes to Cisco Unified Communications Manager Express and then to SCCP Phone	Cisco Unified IP Phone (SIP) -> Cisco Unified Communications Manager-> Cisco Unified Border Element -> Cisco Unified Communications Manager Express -> SCCP IP Phone	Passed	
UCJ861S.CUBE.U.003	Cisco Unified Border Element	High Density transcoding for Delayed Offer to Early Offer	Verify that transcoding is invoked for a codec mismatch in a H.323 - SIP call.	Unified IP Phone (SIP) -> Cisco Unified Communications Manager -> H.323 Gateway -> Cisco Unified Border Element -> SIP trunk -> Cisco Unified Communications Manager Express -> Cisco Unified IP Phone	Passed	

Cisco Unified Survivable Remote Site Telephony

ID	Feature teated	Case Title	Description	Call Component Flow	Status	Defects
UCJ861S.SRST.U.025	Cisco Unified Survivable Remote Site Telephony	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 a Cisco Unified Communications Manager Express with no IP address trusted authentication.	Cisco Unified IPPhone 3 -> SRST -> Cisco Unified Communications Manager Express –> Cisco Unified IP Phone 1, Cisco Unified IP Phone 2	Passed	
UCJ861S.SRST.U.026	Cisco Unified Survivable Remote Site Telephony	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 a 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	
UCJ861S.SRST.U.027	Cisco Unified Survivable Remote Site Telephony	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 a 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	
UCJ861S.SRST.U.028	Cisco UnifiedSurvivable Remote SiteTelephony	A VoIP call from a SRST to an Cisco Unified Communications	Verify that a VoIP call is placed from a SRST to a 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	

ID	Feature teated	Case Title	Description	Call Component Flow	Status	Defects
UCJ861S.SRST.U.029	Cisco Unified Survivable Remote Site Telephony	A VoIP call froma SRST to an Cisco Unified Communications Manager Express, which has no Gatewayconfiguration and the RAS dial peer, is blocked.	Verify that a VoIP call placed from a SRST to a 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	
UCJ861S.SRST.U.030	Cisco Unified Survivable Remote Site Telephony	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 a 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	
UCJ861S.SRST.U.031	Cisco Unified Survivable Remote Site Telephony	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 a 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	

Cisco Unified Communications Manager Express

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ861S.CME.U.001	Cisco Unified Communications Manager Express	Unicast Paging on 6921 IP Phone	Verify the Unicast paging from Unified Communications Manager Express on 6921 IP Phones, by Sending Unicast paging from 6921 IP Phone and see whether the other phones automatically turn on the speakerphone after receiving Unicast paging.		Passed	
UCJ861S.CME.U.002	Cisco Unified Communications Manager Express	Unicast Paging on 7925 IP Phone	Verify the Unicast paging from Unified Communications Manager Express on 7921 IP Phones, by Sending Unicast paging from 7921 IP Phone and see whether the other phones automatically turn on the speakerphone after receiving Unicast paging.		Passed	
UCJ861S.CME.U.003	Cisco Unified Communications Manager Express	Meet-Me Conference in IP Phone 6921	Verify that Meet-Me Conference functionality works fine in 6921 IPPhones.		Passed	
UCJ861S.CME.U.004	Cisco Unified Communications Manager Express	Adding Hunt Group Setting Via Unified Communications Manager Express GUI	Verify the Hunt group added via Unified Communications Manager Express works fine		Passed	
UCJ861S.CME.U.005	Cisco Unified Communications Manager Express	Conference across lines in IP Phone 6900 Series	Verify that conference across lines can be performed from Unified IP Phone 6900 Series, which is registered to the Unified Communications Manager Express router support		Passed	
UCJ861S.CME.U.006	Cisco Unified Communications Manager Express	Conference across lines in IP Phone 9900 Series	Verify that conference across lines can be performed from Unified IP Phone 9900 Series, which is registered to the Unified Communications Manager Express router support.		Passed	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ861S.CME.U.007	Cisco Unified Communications Manager Express	Blind Transfer	Verify blind transfer works successfully with Unified Communications Manager Express router support.		Passed	
UCJ861S.CME.U.008	Cisco Unified Communications Manager Express	Call forward busy Via GUI	Verify Call Forward Busy works successfully when configured via Unified Communications Manager Express GUI		Passed	
UCJ861S.CME.U..010	Cisco Unified Communications Manager Express	Call Park	Verify Call Park works successfully when configured via Unified Communications Manager Express GUI		Passed	
UCJ861S.CME.U.011	Cisco Unified Communications Manager Express	Adhoc-Conferencing	Verify Adhoc-Conferencing works successfully with Unified Communications Manager Express router support.		Passed	
UCJ861S.CME.U.012	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 Unified Communications Manager Express is secure.	Unified IP Phone 6900 Series -> Unified Communications Manager Express -> Unified IP Phone 9900 Series	Passed	
UCJ861S.CME.U.013	Cisco Unified Communications Manager Express	Direct Transfer in Unified IP Phone 6900 Series	Verify that the Unified IP Phone 6900 Series can register with the Unified Communcations Manager Express and Direct Transfer across lines supported.	Unified IP Phone 6900 Series -> Unified Communications Manager Express -> PSTN phone	Passed	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ861S.CME.U.014	Cisco Unified Communications Manager Express	Direct Transfer in Unified IP Phone 9900 Series	Verify that the Unified IP Phone 9900 Series can register with the Unified Communcations Manager Express and Direct Transfer across lines supported.	Unified IP Phone 9900 Series -> Unified Communications Manager Express -> PSTN phone	Passed	
UCJ861S.CME.U.015	Cisco Unified Communications Manager Express	Multiple Consult Transfer in Unified IP Phone 6900 Series	Verify that the Unified IP Phone 6900 Series can register with the Unified Communcations Manager Express and Multiple Consult Transfer across lines supported.		Passed	
UCJ861S.CME.U.016	Cisco Unified Communications Manager Express	Multiple Consult Transfer Unified IP Phone 9900 Series	Verify that the Unified IP Phone 9900 Series can register with the Unified Communcations Manager Express and Multiple Consult Transfer across lines supported.		Passed	
UCJ861S.CME.U.017	Cisco Unified Communications Manager Express	Shared lines	Verify that the Unified Communications Manager Express can support shared lines between Unified IP Phone 9971/9951 and Unified IP Phone 6900 Series.	PSTN -> Unified CommunicationsManager Express -> Unified IP Phone 9971/9951 and Unified IP Phone 6900 Series	Passed	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ861S.CME.U.018	Cisco Unified Communications Manager Express	A VoIP call from Cisco Unified Communications Manager through Cisco Unified Communications Manager Gateway to Cisco Unified Communications Manager Express with no IP address trusted authentication	Verify that a VoIP call is placed from Cisco Unified Communications Manager through Cisco Unified Communications Manager Gateway to 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	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ861S.CME.U.019	Cisco Unified Communications Manager Express	A VoIP call from Cisco Unified Communications Manager through Cisco Unified Communications Manager Gateway to Cisco Unified Communications Manager Express, which has the Gateway configuration and no RAS dial peer.	Verify that a VoIP call is placed from Cisco Unified Communications Manager through Cisco Unified Communications Manager Gateway to 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	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ861S.CME.U.020	Cisco Unified Communications Manager Express	A VoIP call from Cisco Unified Communications Manager through Cisco Unified Communications Manager Gateway to Cisco Unified Communications Manager Express, which has the Gateway configuration and no RAS dial peer, is blocked.	Verify that a VoIP call placed from Cisco Unified Communications Manager through Cisco Unified Communications Manager Gateway to 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	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ861S.CME.U.021	Cisco Unified Communications Manager Express	A VoIP call from Cisco Unified Communications Manager through Cisco Unified Communications Manager Gateway to Cisco Unified Communications Manager Express with no Gateway configuration and RAS dial peer	Verify that a VoIP call is placed from Cisco Unified Communications Manager through Cisco Unified Communications Manager Gateway to Cisco Unified Communications Manager Express, which has no Gatewayconfiguration 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	
UCJ861S.CME.U.022	Cisco Unified Communications Manager Express	A VoIP call placed from Cisco Unified Communications Manager through Cisco Unified Communications Manager Gateway to Cisco Unified Communications Manager Express, which has no Gateway configurationand no RAS dial peer, is blocked.	Verify that a VoIP call placed from Cisco Unified Communications Manager through Cisco Unified Communications Manager Gateway to a configuration and no RAS dial peer, is blocked. Verify that a VoIP callplaced from Cisco Unified Communications Manager through a Cisco Unified Communications Manager Gateway to a Cisco Unified Communications Manager Express, which has no Gateway configuration and no RAS dial peer, is blocked.		Passed	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ861S.CME.U.023	Cisco Unified Communications Manager Express	A VoIP call from a Cisco Unified Communications Manager through a Cisco Unified Communications Manager Gateway to a Cisco Unified Communications Manager Express, which has no Gateway configurationand no RAS dial peer.	Verify that a VoIP call is placed from a Cisco Unified Communications Manager through a Cisco Unified Communications Manager Gateway to a 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) -> CiscoUnified IP Phone 1, Cisco Unified IP Phone 7	Passed	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ861S.CME.U.0124	Cisco Unified Communications Manager Express	Placing a call from Cisco Unified Communications Manager via Cisco Unified Communications ManagerGateway 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 IPPhone 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	
UCJ861S.CME.U.0125	Cisco Unified Communications Manager Express	Apply Japanese Locale in Cisco Unified Phone 8961	Verify that Japanese locale is applied to Cisco Unified Phone Successfully.		Failed	CSCtq67517
UCJ861S.CME.U.0126	Cisco Unified Communications Manager Express	Check the BLF SPEED DIAL BUTTON option in 6900 Series Cisco Unified IP Phone after applying japanese locale.	Verify that BLF DIAL BUTTON option is displayed succesfully in Japanese .		Failed	CSCtq58601
UCJ861S.CME.U.0127	Cisco Unified Communications Manager Express	Check the Single Number Reach options in 6900 Series Cisco Unified IP Phone after applying japanese locale.	Verify that Single Number Reach option is displayed succesfully in Japanese.		Failed	CSCtq58578
UCJ861S.CME.U.0128	Cisco Unified Communications Manager Express	IPV6 Option in Cisco Unified IP Phone 6900 after Applying Japanese Locale.	Verify that IPV6 option is shown correctly after applying Japanese locale		Failed	CSCtq61347

Cisco Unified IP Phone

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UCJ86IF.IPPhone.211	Unified IP Phone	Display name in Japanese for 6921,6941 and 6961 Unified IP Phones	Verify that Display name is shown correctly in Japanese when Japanese characters are entered in the display column		Passed	
UCJ86IF.IPPhone.212	Unified IP Phone	Alerting name in Japanese for 6921,6941 and 6961 Unified IP Phones	Verify that Alerting name is shown correctly in Japanese when Japanese characters entered in the Alerting name column		Passed	
UCJ86IF.IPPhone.213	Unified IP Phone	Line Text Lable in Japanese for 6921,6941 and 6961 Unified IP Phones	Verify that Line Text Lable is shown correctly in Japanese when Japanese characters entered in the Line Text Lable column		Passed	
UCJ86IF.IPPhone.214	Unified IP Phone	Soft key of corporate directory display status for 7921 Unified IP Phone	Verify that part of softkey on Corporate Directory is displayed in Japanese for 7921 Unified IP Phone after applying Japanese locale		Passed	
UC861S.IPPhone.T.001	IP Phone Firmware Upgrade	Upgrade 6921 Unified IP Phone SCCP firmware with Japanese Locale	Verify that 6921 Unified IP Phone SCCP firmware with Japanese locale is upgraded from 9.1(1) SR2 to 9.2(1) version successfully		Passed	
UC861S.IPPhone.T.002	IP Phone Firmware Upgrade	Upgrade 6941 Unified IP Phone SCCP firmware with Japanese Locale	Verify that 6941 Unified IP Phone SCCP firmware with Japanese locale is upgraded from 9.1(1) SR2 to 9.2(1) version successfully		Passed	
UC861S.IPPhone.T.003	IP Phone Firmware Upgrade	Upgrade 6961 Unified IP Phone SCCP firmware with Japanese Locale	Verify that 6961 Unified IP Phone SCCP firmware with Japanese locale is upgraded from 9.1(1) SR2 to 9.2(1) version successfully		Passed	
UC861S.IPPhone.T.004	IP Phone Firmware Upgrade	Upgrade 99XX Unified IP Phone SIP firmware with Japanese Locale	Verify that 99XX Unified IP Phone SIP firmware with Japanese locale is upgraded from 9.1(2) to 9.2(1) version successfully		Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UC861S.IPPhone.T.005	IP Phone Firmware Upgrade	Upgrade 8961 Unified IP Phone firmware with Japanese Locale	Verify that 8961 Unified IP Phone SIP firmware with Japanese locale is upgraded from 9.1(2) to 9.2(1) version successfully		Passed	
UC861S.IPPhone.T.010	Unified IP Phone	Extension Mobility in 7937G Unified IP Phone	Verify that Extension Mobility user can login to 7937G Unified IP Phone with Japanese locale.		Failed	CSCtl03266
UC861S.IPPhone.T.011	Unified IP Phone	Initiate a Meet-Me Conference in 7937G Unified IP Phone	Verify that the 7937G Unified IP Phone initiates a meet-me conference successfully		Passed	
UC861S.IPPhone.T.012	Unified IP Phone	Join Meet-Me Conference in 7937G Unified IP Phone	Verify that the 7937G Unified IP Phone joins a meet-me conference successfully.		Passed	
UC861S.IPPhone.T.013	Unified IP Phone	Barge in 7937G Unified IP Phone	Verify that the Barge functionality in 7937G Unified IP Phone works successfully.		Passed	
UC861S.IPPhone.T.014	Unified IP Phone	CBarge in 7937G Unified IP Phone	Verify that the cBarge functionality in 7937G Unified IP Phone works successfully.		Passed	
UC861S.IPPhone.T.015	Unified IP Phone	Join calls in 7937G Unified IP Phone	Verify that Join functionality works successfully in 7937G.		Passed	
UC861S.IPPhone.T.016	Unified IP Phone Firmware Upgrade	Upgrade 6921 Unified IP Phone SIP firmware with Japanese Locale	Verify that the 6921 Unified IP Phone SIP firmware with Japanese locale is upgraded from 9.1(1) to 9.2(1) version successfully.		Passed	
UC861S.IPPhone.T.017	Unified IP Phone Firmware Upgrade	Upgrade 6941 Unified IP Phone SIP firmware with Japanese Locale	Verify that the 6941 Unified IP Phone SIP firmware with Japanese locale is upgraded from 9.1(1) to 9.2(1) version successfully.		Passed	
UC861S.IPPhone.T.018	Unified IP Phone Firmware Upgrade	Upgrade 6961 Unified IP Phone SIP firmware with Japanese Locale	Verify that the 6961 Unified IP Phone SIP firmware with Japanese locale is upgraded from 9.1(1) to 9.2(1) version successfully.		Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UC861S.IPPhone.T.019	Unified IP Phone	Convert 6921 Unified IP Phone SCCP firmware to SIP Firmware	Verify that 6921 Unified IP Phone with SCCP firmware 9.2(1) version is converted to SIP Firmware 9.2(1) successfully		Passed	
UC861S.IPPhone.T.020	Unified IP Phone	Convert 6941 Unified IP Phone SCCP firmware to SIP Firmware	Verify that 6941 Unified IP Phone with SCCP firmware 9.2(1) version is converted to SIP Firmware 9.2(1) successfully		Passed	
UC861S.IPPhone.T.021	Unified IP Phone	Convert 6961 Unified IP Phone SCCP firmware to SIP Firmware	Verify that 6961 Unified IP Phone with SCCP firmware 9.2(1) version converted to SIP Firmware 9.2(1) successfully		Passed	
UC861S.IPPhone.T.001	Unified IP Phone	Call transfer from 79xx series phones to IP phones	Verify that call transfer is successful from 79xx series phone to IP phones		Passed	
UC861S.IPPhone.T.002	Unified IP Phone	Voice quality in CP-6961 phones when speaker on	Verify that during conversation, quality of voice should be good from both the side when the speaker is on		Passed	
UC861S.IPPhone.T.003	Unified IP Phone	Directed Call park operation at 69xx	Verify that Directed Call park is successful in 69xx phones or not		Passed	
UC861S.IPPhone.T.004	Unified IP Phone	Check MOH feature in 7921 series phones	Verify that MOH feature is working fine when call put on hold in IP Phone		Passed	
UC861S.IPPhone.T.005	Unified IP Phone	Extension Mobility in 7921 IP Phone with Japanese locale	Verify that extension mobility login is successful on a 7921 IP Phone with Japanese locale		Passed	
UC861S.IPPhone.T.006	Unified IP Phone	Check all the numbers displaying properly or not in CP- 6961 phone	All the numbers are displaying properly on the screen with proper time		Passed	
UC861S.IPPhone.T.007	Unified IP Phone	Check MOH feature in 79xx phones	Verify that MOH feature is working fine when call put on hold in 79xx phones		Passed	
UC861S.IPPhone.T.008	Unified IP Phone	Conference between 7921, 7925 and other IP phones	Verify that the conference between 79xx phones and other IP phones is successful or not		Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UC861S.IPPhone.T.009	Unified IP Phone	MOH in CP-7937G phones	Verify that MOH should be audible in CP-7937G phones		Passed	
UC861S.IPPhone.T.010	Unified IP Phone	Call between CP-6961 and CP-8961 phones	Verify that the Call between CP-6961 and CP-8961 phones is successfull		Passed	

Cisco Unified Personal Communicator

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UC861S.CUPC.T.001	Cisco Unified Personal Communicator	Presence status when Cisco Unified Personal Communicator is in deskphone mode	Verify that presence status is displayed properly when Cisco Unified Personal Communicator is in deskphone mode		Passed	
UC861S.CUPC.T.002	Cisco Unified Personal Communicator	Presence status in Cisco Unified Personal Communicator (softphone mode)	Verify the Cisco Unified Personal Communicator presence status when Cisco Unified Personal Communicator is registered in secondary call manager		Passed	
UC861S.CUPC.T.003	Cisco Unified Personal Communicator	Video call in Cisco Unified Personal Communicator	Verify that video call Cisco Unified Personal Communicator is successful when you install Cisco Unified Personal Communicator in Japanese Windows 7 OS		Passed	
UC861S.CUPC.T.004	Cisco Unified Personal Communicator	Maximize and minimize the Cisco Unified Personal Communicator window	Verify that there is no effect to the Cisco Unified Personal Communicator call when the Cisco Unified Personal Communicator window is maximized and minimized		Passed	
UCJ86IF.CUPC.001	Cisco Unified Personal Communicator	Closing Active Conversation while another call received.	Verify that the first call is ended gracefully and the Active Conversation window closes correctly while another incoming call is recevied.		Passed	
UCJ86IF.CUPC.002	Cisco Unified Personal Communicator	Digital Signatures for executable files	Verify that Digital Signatures are present for the Cisco Unified Personal Communicator Click to call executable file.		Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
UCJ86IF.CUPC .003	Cisco Unified Personal Communicator	Presence status for Office 2010 Applications	Verify that presence status is shown correctly for Office 2010 applications.		Passed	
UCJ86IF.CUPC .004	Cisco Unified Personal Communicator	Cisco Unified Personal Communicator Call termination	Cisco Unified Personal Communicator Call termination should occur after closing the Unified Personal Communicator		Passed	
UCJ86IF.CUPC .005	Cisco Unified Personal Communicator	Cisco Unified Personal Communicator and DeskPhone	Verify that Cisco Unified Personal Communicator and Desk Phone shares same and different extension number		Passed	
UCJ86IF.CUPC .006	Cisco Unified Personal Communicator	Picking Call Park in Unified Personal Communicator	Verify that picking up a parked call from Cisco Unified Personal Communicator application (shared line) is successful.		Passed	
UCJ86IF.CUPC .007	Cisco Unified Personal Communicator	Cisco Unified Personal Communicator Presence status	Verify that when Cisco Unified Personal Communicator Presence status changes, the appropriate status is displayed.		Passed	
UCJ86IF.CUPC .008	Cisco Unified Personal Communicator	Receiving a call from contacts list	Verify that Calling party info should be displayed to the corresponding locale.		Passed	
UCJ86IF.CUPC .009	Cisco Unified Personal Communicator	Initiating IM from "Recent Communications" module - valid Unified Presence Server user	Verify that the user is able to initiate an IM from the recent communications module		Passed	
UCJ86IF.CUPC .010	Cisco Unified Personal Communicator	Cisco Unified Personal Communicator with video call	Verify that Cisco Unified Personal Communicator is able to do Video Call without any issues.		Passed	

CUCI-LYNC (CUCIMOC)

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ85F.CSF.U.001	CUCI-LYNC	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	CUCI-LYNC	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	CUCI-LYNC	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	CUCI-LYNC	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	CUCI-LYNC	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	CUCI-LYNC	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	CUCI-LYNC	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	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	CUCI-LYNC	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	
UCJ85F.CSF.U.009	CUCI-LYNC	Calls should not begin with volume control is zero	Verify that the call should not proceed when the volume control is zero		Passed	
UCJ85F.CSF.U.010	CUCI-LYNC	Volume should not be set to zero each time user signs out of Lync/CUCI-Lync and back in	Verify that volume should not be set to zero whenever the user logout and login		Passed	
UCJ85F.CSF.U.011	CUCI-LYNC	Multiple phone numbers of user should be listed in drag/drop	Verify that all the phone numbers are listed in drag/drop popup window when the user has multiple phone numbers in contact list,		Passed	
UCJ85F.CSF.U.012	CUCI-LYNC	Successful login to CUCI-lync in Windows 7	Verify that login to CUCI-lync in Windows 7 is successful		Passed	
UCJ85F.CSF.U.013	CUCI-LYNC	Successful drag and drop of contact to the conversation pane	Verify that drag and drop of contacts to the conversation pane is succesful without any time delay		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCJ85F.CSF.U.014	CUCI-LYNC	The photo of the contact that is being called should appear in the active conversation window	Verify that the photo of the called party appears in the active conversation window.		Passed	
UCJ85F.CSF.U.015	CUCI-LYNC	Successful installation of CUCI-lync on Windows 7	Verify that CUCI-lync is installed succesfully in Windows 7 Operating System		Passed	

Cisco Unified Contact Center Express

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UCCXJ861S.CAD.001	UCCX	Inter-cluster call via Gatekeeper ICT to UCCX CAD with IP Phone 9951 & 9971	Verify inter-cluster basic call flow via Gatekeeper ICT to a CAD agent using IP Phone 9951 & 9971	IP Phone A -> CUCM -> GK ICT -> CUCM -> UCCX -> CAD	Passed	
UCCXJ861S.CAD.002	UCCX	Inter-cluster call via Non-Gatekeeper ICT to UCCX CAD with IP Phone 9951 & 9971	Verify inter-cluster basic call flow via Non-Gatekeeper ICT to a CAD agent using IP Phone 9951 & 9971.	IP Phone A -> CUCM -> Non-GK ICT -> CUCM -> UCCX -> CAD	Passed	
UCCXJ861S.CAD.003	UCCX	Inter-cluster call via SIP Trunk to UCCX CAD with IP Phone 9951 & 9971	Verify inter-cluster basic call flow via SIP Trunk to a CAD agent using IP Phone 9951 & 9971.	IP Phone A -> CUCM -> SIP Trunk -> CUCM -> UCCX -> CAD	Passed	
UCCXJ861S.CAD.021	UCCX	CAD Agent chat with CUPC user	Verify that the agent is able to chat with the CUPC user from CAD		Passed	
UCCXJ861S.CAD.022	UCCX	CAD Agent chat with CUPC user in remote site	Verify that the CAD agent is able to chat with the CUPC user in remote site		Passed	
UCCXJ861S. CBE .003	UCCX	Inter-cluster call via SIP Trunk to UCCX CAD BE with IP Phone 9951 & 9971	Verify inter-cluster basic call flow via SIP Trunk to a CAD BE agent using IP Phone 9951& 9971.	IP Phone A -> CUCM -> SIP Trunk -> CUCM -> UCCX -> CAD BE	Passed	
UCCXJ861S. CBE .004	UCCX	Inter-cluster call via Gatekeeper ICT to UCCX CAD BE with 69xx IP Phone	Verify inter-cluster basic call flow via gatekeeper ICT to a CAD BE agent using 69xx IP Phone.	IP Phone A -> CUCM -> GK ICT -> CUCM -> UCCX -> CAD BE	Passed	

Regression Testing

Logical ID	Feature Tested	Title	Description	Call Component Flow	Status	Defect
UCCXJ851S. ACD.101	UCCX	Agent based routing	Verify that an Agent-based routing call is successful.	Cisco Unified IP Phone -> Voice Gateway -> Cisco Unified Communication Manager-> Cisco Unified Contact Center Express-> CAD Agent	Passed	
UCCXJ851S. ACD.103	UCCX	CED resource routing	Verify that call routing based on a caller entered digits works successfully.	Cisco Unified IP Phone -> Voice Gateway -> Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> CAD Agent	Passed	
UCCXJ851S. ACD.105	UCCX	CSQ based agent routing	Verify that the CSQ based agent routing call is successful	Cisco Unified IP Phone -> Voice Gateway-> Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> CAD Agent	Passed	
UCCXJ851S. ACD.107	UCCX	Agent Based routing with multiple agents using same extension	Verify that Agent based routing is successful for multiple agents having the same extension.	PSTN Phone -> Voice Gateway-> Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> Agent	Passed	
UCCXJ851S. ACD.109	UCCX	Call routing using caller entered DN	Verify that call routing based on the caller entered directory number is successful.	PSTN Phone -> Voice Gateway -> Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> Agent	Passed	
UCCXJ851S. CAD.101	UCCX	Team message	Verify that the supervisor is able to post team message from CSD successfully.	CSD -> Cisco Unified Contact Center Express -> CAD	Passed	
UCCXJ851S. CAD.103	UCCX	Agent status	Verify that the supervisor is able to view the agent status from the supervisor desktop successfully.		Passed	
UCCXJ851S. CAD.106	UCCX	Real time display	Verify the real time report when an agent is in an active call.		Passed	
UCCXJ851S. CAD.108	UCCX	Agent chat with Supervisor during active call	Verify that agents can chat with supervisor during an active call successfully.	CAD -> Cisco Unified Contact Center Express -> CSD	Passed	

Logical ID	Feature Tested	Title	Description	Call Component Flow	Status	Defect
UCCXJ851S.CBE.102	UCCX	Conference using CAD-BE	Verify that the conference call feature successfully from CAD BE.	CAD BE -> Cisco Unified Contact Center Express -> CSD	Passed	
UCCXJ851S.CBE.105	UCCX	Call transfer from CAD-BE	Verify ttat the transfer feature using CAD BE is successful.	CAD BE -> Cisco Unified Contact Center Express -> CSD/CAD	Passed	
UCCXJ851S.CBE.107	UCCX	High priority chat	Verify that High priority chat from CAD BE to CSD works successfully.	CAD BE -> Cisco Unified Contact Center Express -> CSD	Passed	
UCCXJ851S.CSD.101	UCCX	Intercept feature	Verify that the supervisor is able to intercept an agent call from CSD.	CSD -> Cisco Unified Contact Center Express -> CAD	Passed	
UCCXJ851S.CSD.103	UCCX	Barge-in feature	Verify that the supervisor can successfully barge-in an active call.		Passed	
UCCXJ851S.CSD.104	UCCX	Forced log off from CSD	Verify that the supervisor can force log out agents from the Cisco supervisor desktop.		Passed	
UCCXJ851S.IPPA.102	UCCX	IPPA agent login with Extension Mobility	Verify that the agents can login with Extension Mobility extension .		Passed	
UCCXJ851S.SBR.101	UCCX	Skill based routing when agents are available	Verify that the skill based routing functionality is successful when agents are available.		Passed	
UCCXJ851S.HA.102	UCCX	CAD with Extension Mobility in high availability	Verify the basic CAD call flow with Extension Mobility user in HA mode.	PSTN Phone -> Voice Gateway ->Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> CAD	Passed	
UCCXJ851S.HA.103	UCCX	Remote CAD agent in high availability	Verify the basic CAD call flow for an agent in remote site in HA mode.	PSTN Phone -> Voice Gateway -> Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> Voice Gateway -> CAD	Passed	

Logical ID	Feature Tested	Title	Description	Call Component Flow	Status	Defect
UCCXJ851S. HA.104	UCCX	IPPA in High availability	Verify the basic IPPA call flow in HA mode .	PSTN Phone -> Voice Gateway ->Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> IPPA	Passed	
UCCXJ851S. HA.106	UCCX	CAD and CSD in High availability	Verify the basic call flow when CAD and CSD are in main location in HA mode.	PSTN Phone -> Voice Gateway -> Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> CAD, CSD	Passed	
UCCXJ851S. HA.108	UCCX	CAD BE in High availability	Verify the basic CAD BE call flow in HA mode.	PSTN Phone -> Voice Gateway -> Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> CAD BE	Passed	
UCCXJ851S. HA.109	UCCX	CAD BE conference call in High availability mode	Verify that a conference call between CAD BE Agents in main site and remote site in HA mode works successfully.	PSTN Phone -> Voice Gateway -> Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> CAD BE -> Voice Gateway -> CAD BE	Passed	
UCCXJ851S. HA.110	UCCX	Remote CAD BE in high availability	Verify that the remote CAD BE agents are able to receive calls when there is a failover.	PSTN Phone -> Voice Gateway ->Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> Voice Gateway -> CAD BE	Passed	
UCCXJ851S. HA.112	UCCX	High availability transfer by CAD BE	Verify that call transfer is possible from CAD BE to supervisor when there is a failover.	PSTN Phone -> Voice Gateway -> Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> CAD BE -> CSD	Passed	
UCCXJ851S. HA.113	UCCX	CAD CSD Conference in High availability	Verify the successful Conference call flow between CAD and CSD in HA mode.	PSTN Phone -> Voice Gateway -> Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> CAD -> CSD	Passed	
UCCXJ851S. HA.117	UCCX	Chat during High availability	Verify that the chat feature works for agents in HA mode.	CAD -> Cisco Unified Contact Center Express -> CAD	Passed	
UCCXJ851S. HA.119	UCCX	High Priority during High availability	Verify that High Priority chat between CAD BE and CSD in HA mode is successful.	CAD BE -> Cisco Unified Contact Center Express -> CSD	Passed	
UCCXJ851S. HA.120	UCCX	High availability Real Time reports	Verify that the administrator can view Real Time reports in HA mode.		Passed	

Logical ID	Feature Tested	Title	Description	Call Component Flow	Status	Defect
UCCXJ851S.HA.121	UCCX	High availability Historical reports	Verify that the supervisor can extract Historical reports in HA mode.		Passed	
UCCXJ851S.HA.123	UCCX	Barge in to CAD BE High availability	Verify that CSD can Barge-in to a CAD BE call during failover.	PSTN Phone -> Voice Gateway -> Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> CAD BE -> CSD	Passed	
UCCXJ851S.HA.124	UCCX	Intercept with CAD BE in High availability mode	Verify that CSD is able to intercept a CAD BE call during failover.	PSTN Phone -> Voice Gateway -> Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> CAD BE -> CSD	Passed	
UCCXJ851S.HA.125	UCCX	Barge in with CAD during High availability	Verify that CSD can Barge-in to a CAD call during failover.	PSTN Phone -> Voice Gateway -> Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> CAD -> CSD	Passed	
UCCXJ851S.HA.126	UCCX	Intercept with CAD in High availability mode	Verify that CSD is able to intercept a CAD call during failover.	PSTN Phone -> Voice Gateway ->Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> CAD -> CSD	Passed	
UCCXJ851S.HA.127	UCCX	Intercept with IPPA in High availability mode	Verify that CSD is able to intercept an IPPA call during failover.	PSTN Phone -> Voice Gateway -> Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> IPPA -> CSD	Passed	
UCCXJ851S.HA.128	UCCX	Barge in to IPPA High availability	Verify that CSD can Barge-in to an IPPA call during failover.	PSTN Phone -> Voice Gateway -> Cisco Unified Communication Manager -> Cisco Unified Contact Center Express -> IPPA -> CSD	Passed	
UCCXJ851S.HR.101	UCCX	Historical Report	Verify that the supervisor can extract Historical Reports on call activity for the last one week successfully.		Passed	
UCCXJ851S.HR.102	UCCX	Scheduled Historical Report	Verify that the Scheduled Historical Reporting working successfully.		Passed	
UCCXJ851S.HR.103	UCCX	Historical Report for user name in Japanese	Verify that the supervisor can extract Historical Reports for a user with Japanese username successfully.		Passed	

Analog TelePhone Adapter

ID	Features Tested	Title	Description	Call component flow	Status	Defects
UCJ861S. Analog Telephone Adapter18 7.U.001	Analog TelePhone Adapter	Check distinctive ringing via Analog Telephone Adapter187	Verify that distinctive ringing via Analog Telephone Adapter187 (different cluster) on both ends works Successfully.	1. Analog phone ->Analog Telephone Adapter187 -> SIP -> Cisco Unified Communications Manager -> Unified IP Phone 2. Unified IP Phone -> Cisco Unified Communications Manager -> Analog Telephone Adapter187Analog phone	Passed	
UCJ861S. Analog Telephone Adapter18 7.U.002	Analog TelePhone Adapter	Long duration call from an IP phone (Cluster A) to Analog phone (Cluster B),hear ring tone	Verify Long Duration call and hear the ring tone cluster wise and internally.	Analog phone -> Analog Telephone Adapter187 -> SIP -> Cisco Unified Communications Manager -> Unified IP Phone.	Passed	
UCJ861S. Analog Telephone Adapter18 7.U.003	Analog TelePhone Adapter	Registration after network connectivity down	Verify the Analog Telephone Adapter187 Registration when connectivity is down		Passed	
UCJ861S. Analog Telephone Adapter18 7.U.004	Analog TelePhone Adapter	Dial tone delays on offhook during randomly unregisters	Verify the Analog Telephone Adapter 187 dial tone delays on when off-hook during random wise call.		Passed	
UCJ861S. Analog Telephone Adapter18 7.U.005	Analog TelePhone Adapter	Blind Transfer in Analog Telephone Adapter187	Verify that Blind Transfer across lines is performed from Analog Phone which is registered to the Analog Telephone Adapter187 to Unified IPPhone.		Passed	
UCJ861S. Analog Telephone Adapter18 7.U.006	Analog TelePhone Adapter	Shared lines in Analog Telephone Adapter187	Verify shared lines across lines is performed in Analog Telephone Adapter187		Passed	

ID	Features Tested	Title	Description	Call component flow	Status	Defects
UCJ861S. Analog Telephone Adapter187.U.007	Analog TelePhone Adapter	Meet-Me Conference in Analog Phone	Verify Meet-me conference works successfully in Analog Telephone Adapter187		Passed	
UCJ861S. Analog Telephone Adapter187.U.008	Analog TelePhone Adapter	Call Hold	Verify Call Hold works successfully in Analog Telephone Adapter187		Passed	
UCJ861S. Analog Telephone Adapter187.U.009	Analog TelePhone Adapter	Semi-Attended transfer	Verify Semi-Attended transfer works successfully in Analog Telephone Adapter 187		Passed	
UCJ861S. Analog Telephone Adapter187.U.010	Analog TelePhone Adapter	Call Forward and Call Waiting	Verify Call Forward and Call Waiting works successfully in Analog Telephone Adapter 187		Passed	
UCJ861S. Analog Telephone Adapter187.U.011	Analog TelePhone Adapter	Speed Dial	Verify Speed Dial works successfully in Analog Telephone Adapter 187		Passed	
UCJ861S. Analog Telephone Adapter187.U.012	Analog TelePhone Adapter	Group Call PickUp	Verify Group Call Pick Up works successfully in Analog Telephone Adapter 187		Passed	
UCJ861S. Analog Telephone Adapter187.U.014	Analog TelePhone Adapter	Fully Unattended Transfer	Verify Fully Unattended Transfer works successfully in Analog Telephone Adapter 187		Passed	
UCJ861S. Analog Telephone Adapter187.U.015	Analog TelePhone Adapter	Redial	Verify Redial works successfully in Analog Telephone Adapter 187		Passed	
UCJ861S. Analog Telephone Adapter187.U.016	Analog TelePhone Adapter	Message Waiting	Verify Message Waiting works successfully in Analog Telephone Adapter 187		Passed	

ID	Features Tested	Title	Description	Call component flow	Status	Defects
UCJ861S. Analog Telephone Adapter18 7.U.017	Hold and Resume	MOH status for 9951 Unified IP Phone when established using ATA 187	Verify that MOH listens on 9951 Unified IP Phone when ATA 187 holds the call and able to listen audio when resumes call	Analog Phone -> ATA 187 (SIP) -> Cisco Unified Communications Maanger -> (SIP) -> 9951 Unified IP phone.	Passed	
UCJ861S. Analog Telephone Adapter18 7.U.018	Hold and Resume	MOH status for 7961 Unified IP Phone when established using ATA 187	Verify that MOH listens on 7961 Unified IP Phone when ATA 187 holds the call and able to listen audio when resumes call	Analog Phone -> ATA 187 (SIP) -> Cisco Unified Communications Maanger -> (SIP) ->7961 Unified IP phone	Passed	
UCJ861S. AnalogTel ephoneAd apter187.U .019	Analog Telephone Adapter	Call Park using an unified IP phone via SME, retrieve with analog Phone connected through ATA 187	Verify that Call Park using an unified IP phone with japanese locale via SME, retrieve with Analog Phone connected through ATA 187	IP Phone A --> CUCM A(8.6(1a)) --> CUCM-SME -> CUCM B (8.6(1a)) ->IP Phone B -> ATA187-> Analog phone A	Failed	CSCtr569 48
UCJ861S. AnalogTel ephoneAd apter187.U .020	Analog Telephone Adapter	Call to a busy Analog Phone connected through ATA 187	Verify that caller ID displays in the Analog Phone which is in busy state.	1.IP Phone A -> CUCM -> ATA187 -> Analog Phone A 2. IP Phone B -> CUCM -> ATA187 -> Analog Phone A (Busy)	Failed	CSCtr403 96

Cisco IP Communicator

ID	Features Tested	Title	Description	Call component flow	Status	Defects
UCJ86IF.CIPC.301	Hold and Resume	MOH status for 9951 Unified IP Phone	Verify that MOH listens on 9951 Unified IP Phone when Cisco IP Communicator holds the call and able to listen audio when resumes call		Passed	
UCJ86IF.CIPC.302	Hold and Resume	MOH status for 6961 Unified IP Phone	Verify that MOH listens on 6961 Unified IP Phone when Cisco IP Communicator holds the call and able to listen audio when resumes call		Passed	
UCJ86IF.CIPC.303	Transfer	Consultative transfer to Cisco IP Communicator	Verify that consultative transfer happens successfully between 6961 Unified IP Phone, 9971 Unified IP Phone and Cisco IP Communicator		Passed	
UCJ86IF.CIPC.304	Conference	Conference call between 6961,9951 Unified IP Phone and Cisco IP Communicator	Verify that conference call is made and communication is successful between 6961 Unified IP Phone, 9971 Unified IP Phone and Cisco IP Communicator		Passed	
UCJ86IF.CIPC.305	Call Forward All	Cfwdall feature for Cisco IP Communicator	Verify that all Calls coming from Unified IP Phone A to Cisco IP Communicator is successfully forwarded to Unified IP Phone B		Passed	
UCJ86IF.CIPC.306	Park	Call Park feature for Cisco IP Communicator	Verify that Call from 9951 Unified IP Phone to Cisco IP Communicator able to attend and park at the assigned number successfully		Passed	

vMotion

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ86IV.CUCM.0011	CUCM	vMotion	Verify that CUCM Publisher installed on UCS Blade Server 1 Successfully migrated to UCS Blade Server 2 using vMotion		Passed	
UCJ86IV.CUCM.0012	CUC	vMotion	Verify that CUC Publisher installed on UCS Blade Server 1 Successfully migrated to UCS Blade Server 2 using vMotion		Passed	
UCJ86IV.CUCM.0013	CUPS	vMotion	Verify that CUP Publisher installed on UCS Blade Server 1 Successfully migrated to UCS Blade Server 2 using vMotion		Passed	
UCJ86IV.CUCM.0014	Audio call	Call between Unified IP Phone A and B when Vmotion in progress	Verify that unified IP Phone A and Unified IP Phone B in connected state when Cisco Unified Communications manager moving from one Blade to another Blade using Vmotion		Passed	
UCJ86IV.CUCM.0015	Hold and Resume	MOH on Unified IP Phone A with Unified IP Phone B hold the call when Vmotion in progress	Verify that MOH listens on Unified IP Phone A successfully when Cisco Unified Communications Manager moving from one Blade to Another Blade using Vmotion		Passed	
UCJ86IV.CUCM.0016	Transfer	MOH on Unified IP Phone A with Unified IP Phone B transfer the call when Vmotion in progress	Verify that MOH listens on Unified IP Phone A successfully when Cisco Unified Communications Manager moving from one Blade to Another Blade using Vmotion		Passed	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ86IV.CUCM.0017	Conference	Conference call between Unified IP Phone A , B and C when Vmotion in progress	Verify that Conference call between Unified IP Phone A ,B and C made successfully and Unified IP Phone A, B and C talk to each other when Cisco Unified Communications Manager moving from one Blade to Another Blade using Vmotion		Passed	
UCJ86IV.CUCM.0018	BLF Dpark	BLF state for the Unified IP Phone A when Vmotion in progress	Verify that BLF status of Unified IP Phone A not having any changes when Phone A park the call coming from Unified IP Phone B using BLF Dpark button in the situation Cisco Unified Communications Manager moving from one Blade to Another Blade using Vmotion		Passed	
UCJ86IV.CUCM.0019	Park	MOH on Unified IP Phone A with Unified IP Phone B park the call when Vmotion in progress	Verify that MOH listens on Unified IP Phone A successfully when Cisco Unified Communications Manager moving from one Blade to Another Blade using Vmotion		Passed	
UCJ86IV.CUCM.0020	Video Call	Video Call between Unified IP Phone A and B when Vmotion in progress	Verify that unified IP Phone A and Unified IP Phone B in connected state with video when Cisco Unified Communications manager moving from one Blade to another Blade using Vmotion		Passed	

Upgrade

This section contains the following sub-sections:

- Upgrade Path to Unified CM 8.6(1a)
- Upgrading to Unified CM 8.6(1a) on a Virtual Server
- Upgrading CUC to 8.6(1a)
- Hardware Requirements for Upgrading from 8.5(1) to 8.6(1a) on a Virtual Server
- License Requirements for Upgrading Unified CM from 4.1(3)
- Upgrade Topology
- UCS Component Matrix for All Release Sets
- Upgrade Matrices and Test Results

Upgrade Path to Unified CM 8.6(1a)

Base Release set	Intermediate Release Set 1	Intermediate Release Set 2	Upgrade Type
4.1.3	7.1.3	8.5	Multi-Stage
5.1.3	7.1.5	8.5	Multi-Stage
6.1.3	7.1.5	8.5	Multi-Stage
6.1.5	8.5	Nil	Multistage
7.1.3	8.5	Nil	Multistage
7.1.5	8.5	Nil	Multistage
8.0.3	Nil	Nil	Single stage

Upgrading to Unified CM 8.6(1a) on a Virtual Server

Before upgrading from existing Cisco Unified Communications Manager to Cisco Unified Communications Manager 8.6(1a) on a virtual server, ensure that the Guest Operating System and RAM of the virtual server meet the requirements for the latest release.

To upgrade Cisco Unified Communications Manager on a virtual server, do the following:

Step 1 Upgrade the virtual machine to the latest release.

Step 2 After you finish the upgrade, shut down the virtual machine.

Step 3 Change the Guest Operating System to Red-Hat Enterprise Linux 5 (32-bit).

Step 4 Check the RAM on the virtual machine and make sure that it meets the minimum RAM requirements for this release.

Step 5 Save the changes.

Step 6 Restart the virtual machine.

For information on supported hardware refer the following URL:
http://www.cisco.com/en/US/prod/collateral/voicesw/ps6790/ps5748/ps378/prod_brochure0900aecd8062a4f9.html

Upgrading CUC to 8.6(1a)

Before upgrading to 8.6(1a), delete the 8.5 locale from CUC. Deletion of CUC locale can be done with the help of command line interface.

Enter the following commands in the CLI:

- Command used to delete cuc locale:

  ```
  delete cuc locale locale-id
  ```

- To get the locale-id use the following command:

  ```
  show cuc locale
  ```

Recommended CUC Build details:

Version	CUC Build Number
8.5	8.5.1.12900-7
8.6	8.6.1.21004-1

Hardware Requirements for Upgrading from 8.5(1) to 8.6(1a) on a Virtual Server

Cisco Unified Communications Manager

S.No	OVA	Error Encountered	Workaround followed
1	cucm_8.0_vmv7_v2.0.ova	Nil	Nil

Cisco Unity Connection

S.No	OVA	Error Encountered	Workaround followed
1	CUC_500_user_v1.0_vmv7.ova	Hardware not supported	Change the memory setting to 6 GB
2	CUC_5000_user_v1.0_vmv7.ova	• Hardware not supported • An Unknown error occurred while accessing the upgrade file	• Change the memory setting to 6 GB • Uninstall the locale of 8.5 server and start with the 8.6(1a) upgrade Reference: CSCtq97369

Cisco Unified Presence

S.No	OVA	Error Encountered	Workaround followed
1	CUP_1000_user_v1.0_vmv7.ova	Nil	Nil
2	CUP_5K_user_v1.0_vmv7.ova	Nil	Nil

License Requirements for Upgrading Unified CM from 4.1(3)

To upgrade Unified CM 4.1(3), follow the steps as given below.

Step 1 After you complete the Cisco Unified Communications Manager upgrade process, navigate to Cisco Unified Communications Manager Administration and choose **System > Licensing > License File**

Upload.The License File Upload window displays.

Step 2 Choose the license file licupgrade_7.1.lic from the Existing Files drop-down list and click View File.The window refreshes and displays the information for the selected license. Copy all the information in this file.

Step 3 Navigate to the License Registration web tool at https://tools.cisco.com/SWIFT/Licensing/PrivateRegistrationServlet?FormId=806.

Step 4 Enter your login credentials.

Step 5 Enter the MAC address of the Ethernet 0 NIC of the first node of the Cisco Unified Communications Manager cluster.

Step 6 In the text box that is provided, paste the license file contents that you copied in Step 2 by using the appropriate keyboard shortcuts, such as Ctrl-V.

Step 7 Enter a valid e-mail address and click Continue. A license file generates.

The system sends the license file to you through e-mail using the e-mail address that you provided.

Note If the web tool in step 3 is not working, use the below procedure to obtain the same license
1. Send MAC and contents of the file licupgrade_7.1.lic to licensing@cisco.com and request for device upgrade license with subject as DMA failed upgrade to ver 7.
2. You will be receiving a mail with a license attached. Upload the same to the server. Once you uploaded the license you can view the DLU of 4.x upgraded to 7.x

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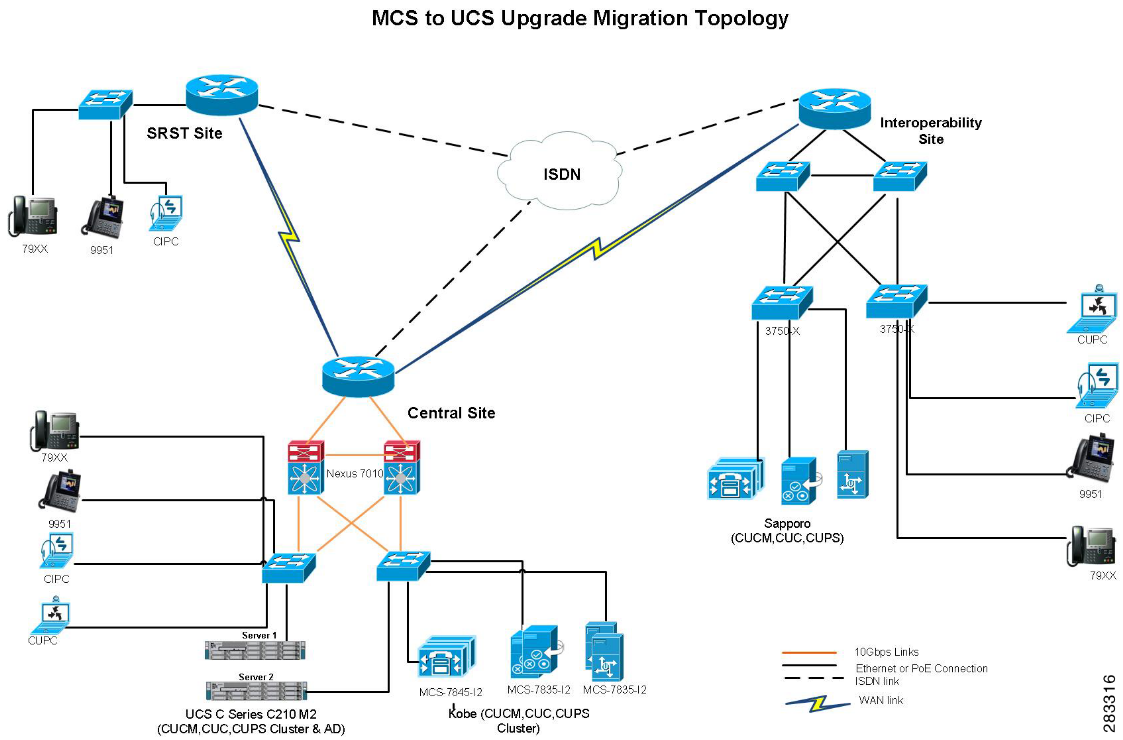
Step 8 You must upload the license file to the server with the matching MAC address.

Step 9 You can obtain licenses for new devices that you are adding to the upgraded system, if your system requires additional device license units. For more information, refer to *Cisco Unified Communications Manager Administration Guide*.

Upgrade Topology

Figure 3-1 Upgrade Topology



UCS Component Matrix for All Release Sets

Category	Component	Hardware	Version
Servers	Cisco C-series	C210 M2	1.4(1m)
Hypervisor	ESXi host on Blade Server		ESXi 4.1
MCS	Cisco Unified Communications Manager	MCS 7845 H2	
MCS	Cisco Unity Connection	MCS 7845 I2	
MCS	Cisco Unified Presence	MCS 7835 I2	
Voice Gateway	IOS	• Voice Gateway 3945 • Voice Gateway 2851	
Switch	Access Switch	Cisco 3750	

Upgrade Matrices and Test Results

This section contains the following sub-sections:

- Upgrade 4.1.3 through 7.1(3) and 8.5(1) to 8.6(1a)
- Upgrade 5.1.3 through 7.1(5) and 8.5(1) to 8.6(1a)
- Upgrade 6.1.3 through 7.1(5) and 8.5(1) to 8.6(1a)
- Upgrade 6.1.5 through 8.5(1) to 8.6(1a)
- Upgrade 7.1.3 through 8.5(1) to 8.6(1a)
- Upgrade 7.1(5) through 8.5(1) to 8.6(1a)
- Standalone Upgrade - Upgrade 8.0.3 to 8.6(1a)

Upgrade 4.1.3 through 7.1(3) and 8.5(1) to 8.6(1a)

Environment matrix of Upgrade 4.1.3

Product/Component	Base Release Set	Intermediate Release set 1	Intermediate Release set 2	Target Release set
Cisco Unified Communications Manager	4.1.3sr8a	7.1.3-10000-11	8.5.1-10000-26	8.6.1.21004-1
DMA(Data Migration Assistant)	7.1(3)	Nil	Nil	Nil
Cisco Unified Communications Manager Locale	cm-locale-ja_JP-4.1.3.4200	cm-locale-ja_JP-7.1.3.2000-1.cop.sgn	cm-locale-ja_JP-8.5.1.1000.1.cop.sgn	cm-locale-ja_JP-8.6.1.1000-1.cop.sgn
Cisco Unity Connection	Nil	7.1.3-10000-11	8.5.1.12900-7	8.6.1.21004-1
Cisco Unity Connection Locale	Nil	uc-locale-ja_JP-7.1.2.0-139.cop.sgn	uc-locale-ja_JP-8.5.1.1-24.cop.sgn	uc-locale-ja_JP-8.6.1.1-2.cop.sgn
Cisco Unified Presence	Nil	7.0(5)-10000-18	8.5.1-10000-35	8.6.1.10000-34
Cisco Unified Presence Locale	Nil	ps-locale-ja_JP-7.0.4.1000-1.cop.sgn	ps-locale-ja_JP-8.5.1.1000-1.cop.sgn	ps-locale-ja_JP-8.6.1.1000-1.cop.sgn
Cisco Unified Survivable Remote Site Telephony	3.3	8	8.5	8.6(1a)
Cisco Unified Survivable Remote Site Telephony IOS	12.3(14) T	15.0(1)M XA	15.1.3 T	15.1(4)M
IOS (Voice gateways 2801)	12.4(13d)	15.0(1)M	15.1.3 T	15.1(4)M
IP Communicator	2.0(1)	7.0(3)	7.0(3)	8-6-1-0
Unified Personal Communicator	Nil	7.0(2)	8.5	8.5.1.18771

Test Results

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UC861S.UPGRADE.U.801	Upgrade	Upgrade Cisco Unified Communications Manager publisher 4.1.3	Verify successful upgrade of Cisco Unified Communications Manager to version 7.1(3)		Passed	
UC861S.UPGRADE.U.806	Upgrade	Upgrade the same DLUs of 4.x to 7.1.3	Verify successful upgrade of licenses		Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UC861S.UPGRADE.U.808	Upgrade	Upgrade Cisco Unified Communications Manager publisher 7.1(3)	Verify successful upgrade of Cisco Unified Communications Manager to 8.5.1		Passed	
UC861S.UPGRADE.U.812	Upgrade	Upgrade Cisco Unified Presence 7.1(3) Primary	Verify successful upgrade of Release set 8.5.1 Cisco Unified Presence		Passed	
UC861S.UPGRADE.U.812	Upgrade	Upgrade of Cisco Unity Connection 7.1.3 primary	Verify successful upgrade of Cisco Unity connection 8.5.1		Passed	
UC861S.UPGRADE.U.1000	Upgrade	8.5(1) CUC Japanese prompt during voicemail	Choose Japanese option from CUCA >> System setting >> General Configuration >> System default language drop down(select Japanese option). Verfiy during voicemail the system prompt is in Japanese		Passed	
UC861S.UPGRADE.U.818	Upgrade	Backup – Cisco Unified Communications Manager using Disaster recovery system in MCS	Verify that 100% backup of Cisco Unified Communications Manager is successful.		Passed	
UC861S.UPGRADE.U.823	Upgrade	Installation of Cisco Unified Presence 8.5.1 on C-series	Verify that Cisco Unified Presence 8.5.1 installation is successful.		Passed	
UC861S.UPGRADE.U.824	Upgrade	Applying License for Cisco Unified Communications Manager 8.5.1 installed on C-series	Verify that license upload is successful.		Passed	
UC861S.UPGRADE.U.828	Upgrade	Restore – Cisco Unified Communications Manager using Disaster recovery system in C-series	Verify that 100% restore of Cisco Unified Communications Manager is successful.		Passed	
UC861S.UPGRADE.U.830	Upgrade	Install the Refresh Upgrade Patch for Unified communications Manager	Verify successful install of refresh upgrade cop file		Passed	
UC861S.UPGRADE.U.832	Upgrade	Upgrade Cisco Unified Communications Manager publisher 8.5.1	Verify successful upgrade of Cisco Unified Communications Manager to 8.6(1a)		Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UC861S.UPGRADE.U.840	Upgrade	Upgrade of Cisco Unity Connection 8.5 primary	Verify successful upgrade for release set 8.6(1a) Cisco Unity connection		Passed	
UC861S.UPGRADE.U.1001	Upgrade	8.6(1a) CUC Japanese prompt during voicemail	Choose Japanese option from CUCA >> System setting >> General Configuration >> System default language drop down(select Japanese option).Verfiy during voicemail the system prompt is in Japanese		Failed	CSCtt21815
UC861S.UPGRADE.U.837	Upgrade	Upgrade Cisco Unified Presence 8.5 Primary	Verify successful upgrade of Release set 8.6(1a) Cisco Unified Presence		Passed	

Upgrade 5.1.3 through 7.1(5) and 8.5(1) to 8.6(1a)

Environment matrix of Upgrade 5.1.3

Product/Component	Base Release Set	Intermediate Release set 1	Intermediate Release set 2	Target Release set
Cisco Unified Communications Manager	5.1.3.1000-12	7.1.5-10000-10	8.5.1-10000-26	8.6.1.21004-1
Cisco Unified Communications Manager Locale	cm-locale-ja_JP-5.1.1.2000-1.cop.sgn	cm-locale-ja_JP-7.1.5.1200-1.cop.sgn	cm-locale-ja_JP-8.5.1.1000-1.cop.sgn	cm-locale-ja_JP-8.6.1.1000-1.cop.sgn
Cisco Unity Connection	Nil	7.1.5-32900-2 with ciscocm.cuc_cluster_715.cop.sgn	8.5.1.12900-7	8.6.1.21004-1
Cisco Unity Connection Locale	Nil	uc-locale-ja_JP-7.1.2.0-139.cop.sgn	uc-locale-ja_JP-8.5.1.1-24.cop.sgn	uc-locale-ja_JP-8.6.1.1-2.cop.sgn
Cisco unified Presence	Nil	7.0.5-10000-18	8.5.1-10000-35	8.6.1.10000-34
Cisco Unified Presence Locale	Nil	ps-locale-ja_JP-7.0.4.1000-1.cop.sgn	ps-locale-ja_JP-8.5.1.1000-1.cop.sgn	ps-locale-ja_JP-8.6.1.1000-1.cop.sgn
Cisco Unified Survivable Remote Site Telephony	4.0(2)	8.0	8.5	8.6(1a)
Cisco Unified Survivable Remote Site Telephony IOS	12.4(11)T3	15.0(1)M XA	15.1.3 T	15.1(4)M
IOS (Voice gateways 2801)	12.4(15)T4	15.0(1)M	15.1.3T	15.1(4)M
IP Communicator	2.0(1)	7.0(3)	7.0(3)	8-6-1-0
Unified Personal Communicator	Nil	7.0(2)	8.5.1	8.5.1.18771

Test Results

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UC861S.UPGRADE.U.852	Upgrade	Upgrade Cisco Unified Communications Manager publisher 5.1.3	Verify successful upgrade of Cisco Unified Communications Manager 7.1(5)		Passed	
UC861S.UPGRADE.U.862	Upgrade	Upgrade Cisco Unified Presence 7.0.5 Primary	Verify successful upgrade of Release set 8.5.1 Cisco Unified Presence		Passed	
UC861S.UPGRADE.U.865	Upgrade	Upgrade of Cisco Unity Connection 7.1.5 primary	Verify successful upgrade for release set 8.5 Cisco Unity connection		Passed	
UC861S.UPGRADE.U.1002	Upgrade	8.5(1) CUC Japanese prompt during voicemail	Choose Japanese option from CUCA >> System setting >> General Configuration >> System default language drop down(select Japanese option). Verfiy during voicemail the system prompt is in Japanese		Passed	
UC861S.UPGRADE.U.867	Upgrade	Backup – Cisco Unified Communications Manager using Disaster recovery system in MCS	Verify that 100% backup of Cisco Unified Communications Manager is successful.		Passed	
UC861S.UPGRADE.U.870	Upgrade	Installation of Cisco Unified Communications Manager 8.5.1 on C-series	Verify that Cisco Unified Communications Manager 8.5.1 installation is successful.		Passed	
UC861S.UPGRADE.U.873	Upgrade	Applying License for Cisco Unified Communications Manager 8.5.1 installed on C-series	Verify that license upload is successful.		Passed	
UC861S.UPGRADE.U.877	Upgrade	Restore – Cisco Unified Communications Manager using Disaster recovery system in UCS	Verify that 100% restore of Cisco Unified Communications Manager is successful.		Passed	
UC861S.UPGRADE.U.879	Upgrade	Install the Refresh Upgrade Patch for Unified communications Manager	Verify successful install of refresh upgrade cop file		Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UC861S.UPGRADE.U.882	Upgrade	Upgrade Cisco Unified Communications Manager publisher 8.5	Verify successful upgrade of Cisco Unified Communications Manager 8.6(1a)		Passed	
UC861S.UPGRADE.U.889	Upgrade	Upgrade of Cisco Unity Connection 8.5 primary	Verify successful upgrade for release set 8.6(1a) Cisco Unity connection		Passed	
UC861S.UPGRADE.U.1003	Upgrade	8.6(1a) CUC Japanese prompt during voicemail	Choose Japanese option from CUCA >> System setting >> General Configuration >> System default language drop down(select Japanese option).Verfiy during voicemail the system prompt is in Japanese		Failed	CSCtt21815
UC861S.UPGRADE.U.886	Upgrade	Upgrade Cisco Unified Presence 8.5 Primary	Verify successful upgrade of Release set 8.6(1a) Cisco Unified Presence		Passed	

Upgrade 6.1.3 through 7.1(5) and 8.5(1) to 8.6(1a)

Environment matrix of Upgrade 6.1.3

Product/Component	Base Release Set	Intermediate Release set 1	Intermediate Release set 2	Target Release set
Cisco Unified Communications Manager	6.1.3-1000-16 to 6.1.3.3102-1	7.1.5-32900-2	8.5.1-10000-26	8.6.1.21004-1
Cisco Unified Communications Manager Locale	cm-locale-ja_JP-6.1.3.3000-1.cop.sgn	cm-locale-ja_JP-7.1.5.1200-1.cop.sgn	cm-locale-ja_JP-8.5.1.1000-1.cop.sgn	cm-locale-ja_JP-8.6.1.1000-1.cop.sgn
Cisco Unity Connection	6.1.3-1000-16 to 6.1.3.3102-1	7.1.5-32900-2 with ciscocm.cuc_cluster_715.cop.sgn	8.5.1.12900-7	8.6.1.21004-1
Cisco Unity Connection Locale	uc-locale-ja_JP-6.1.1.0-362.cop	uc-locale-ja_JP-7.1.2.0-139.cop.sgn	uc-locale-ja_JP-8.5.1.1-24.cop.sgn	uc-locale-ja_JP-8.6.1.1-2.cop.sgn
Cisco unified Presence	6.0.5.1000-13	7.0.9-10000-6	8.5.1-10000-35	8.6.1.10000-34
Cisco Unified Presence Locale	ps-locale-ja_JP-6.0.2.1000-1.cop	ps-locale-ja_JP-7.0.4.1000-1.cop.sgn	ps-locale-ja_JP-8.5.1.1000-1.cop.sgn	ps-locale-ja_JP-8.6.1.1000-1.cop.sgn
Cisco Unified Survivable Remote Site Telephony	4.0(2)	8.0	8.5	8.6(1a)
Cisco Unified Survivable Remote Site Telephony IOS	12.4(11)T3	15.0(1)M XA	15.1.3 T	15.1(4)M
IOS (Voice gateways 2801)	12.4(15)T4	15.0(1)M	15.1.3T	15.1(4)M
IP Communicator	2.0(1)	7.0(3)	7.0(3)	8-6-1-0
Unified Personal Communicator	1.2(4)	7.0(2)	8.5.1	8.5.1.18771

Test Results

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
861S.UPGRADE.U.902	Upgrade	Upgrade Cisco Unified Communications Manager publisher 6.1.3	Verify successful upgrade of Cisco Unified Communications Manager 7.1(5)		Passed	
UC861S.UPGRADE.U.917	Upgrade	Upgrade Cisco Unified Presence 7.0.5 Primary	Verify successful upgrade of Release set 8.5.1 Cisco Unified Presence		Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UC861S.UPGRADE.U.920	Upgrade	Upgrade of Cisco Unity Connection 7.1.5 primary	Verify successful upgrade for release set 8.5 Cisco Unity connection		Passed	
UC861S.UPGRADE.U.1004	Upgrade	8.5(1) CUC Japanese prompt during voicemail	Choose Japanese option from CUCA >> System setting >> General Configuration >> System default language drop down(select Japanese option). Verfiy during voicemail the system prompt is in Japanese		Passed	
UC861S.UPGRADE.U.923	Upgrade	Backup – Cisco Unified Communications Manager using Disaster recovery system in MCS	Verify that 100% backup of Cisco Unified Communications Manager is successful.		Passed	
UC861S.UPGRADE.U.926	Upgrade	Installation of Cisco Unified Communications Manager 8.5.1 on C-series	Verify that Cisco Unified Communications Manager 8.5.1 installation is successful.		Passed	
UC861S.UPGRADE.U.929	Upgrade	Applying License for Cisco Unified Communications Manager 8.5.1 installed on C-series	Verify that license upload is successful.		Passed	
UC861S.UPGRADE.U.933	Upgrade	Restore – Cisco Unified Communications Manager using Disaster recovery system in UCS	Verify that 100% restore of Cisco Unified Communications Manager is successful.		Passed	
UC861S.UPGRADE.U.935	Upgrade	Install the Refresh Upgrade Patch for Unified communications Manager	Verify successful install of refresh upgrade cop file		Passed	
UC861S.UPGRADE.U.938	Upgrade	Upgrade Cisco Unified Communications Manager publisher 8.5	Verify successful upgrade of Cisco Unified Communications Manager 8.6(1a)		Passed	
UC861S.UPGRADE.U.945	Upgrade	Upgrade of Cisco Unity Connection 8.5 primary	Verify successful upgrade for release set 8.6(1a) Cisco Unity connection		Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UC861S.UPGRADE.U.1005	Upgrade	8.6(1a) CUC Japanese prompt during voicemail	Choose Japanese option from CUCA >> System setting >> General Configuration >> System default language drop down(select Japanese option).Verfiy during voicemail the system prompt is in Japanese		Failed	CSCtt21815
UC861S.UPGRADE.U.942	Upgrade	Upgrade Cisco Unified Presence 8.5 Primary	Verify successful upgrade of Release set 8.6(1a) Cisco Unified Presence		Passed	

Upgrade 6.1.5 through 8.5(1) to 8.6(1a)

Environment matrix of Upgrade 6.1.5

Product/Component	Base Release Set	Intermediate Release set	Target Release set
Cisco Unified Communications Manager	6.1.5-10000-10	8.5.1-10000-26	8.6.1.21004-1
Cisco Unified Communications Manager Locale	cm-locale-ja_JP-6.1.3.3000-1.cop	cm-locale-ja_JP-8.5.1.1000-1.cop.sgn	cm-locale-ja_JP-8.6.1.1000-1.cop.sgn
Cisco Unity Connection	6.1.5-10000-10	8.5.1.12900-7	8.6.1.21004-1
Cisco Unity Connection Locale	uc-locale-ja_JP-6.1.1.0-362.cop	uc-locale-ja_JP-8.5.1.1-24.cop.sgn	uc-locale-ja_JP-8.6.1.1-2.cop.sgn
Cisco unified Presence	6.0.7.1000-5	8.5.1-10000-35	8.6.1.10000-34
Cisco Unified Presence Locale	ps-locale-ja_JP-6.0.2.1000-1.cop	ps-locale-ja_JP-8.5.1.1000-1.cop.sgn	ps-locale-ja_JP-8.6.1.1000-1.cop.sgn
Cisco Unified Survivable Remote Site Telephony	4.0(2)	8.5	8.6(1a)
Cisco Unified Survivable Remote Site Telephony IOS	12.4(11)T3	15.1.3 T	15.1(4)M
IOS (Voice gateways 2801)	12.4(15)T4	15.1.3T	15.1(4)M
IP Communicator	2.0(1)	7.0(3)	8-6-1-0
CUPC	1.2(4)	8.5.1	8.5.1.

Test Results

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UC861S.UPGRADE.U.1052	Upgrade	Upgrade Cisco Unified Communications Manager publisher 6.1.5	Verify that upgrade of Cisco Unified Communications Manager 8.5.1 is successful		Passed	
UC861S.UPGRADE.U.1056	Upgrade	Upgrade Cisco Unified Presence 6.0.7 Primary	Verify that upgrade of Release set 7.0 Cisco Unified Presence is successful		Passed	
UC861S.UPGRADE.U.1099	Upgrade	Upgrade Cisco Unified Presence 7.0.9 Primary	Verify that upgrade of Release set 8.5.1 Cisco Unified Presence is successful		Passed	
UC861S.UPGRADE.U.1058	Upgrade	Upgrade of Cisco Unity Connection 6.1.5 primary	Verify that upgrade to Cisco Unity connection 8.5.1 is successful		Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UC861S.UPGRADE .U.1006	Upgrade	8.5(1) CUC Japanese prompt during voicemail	Choose Japanese option from CUCA >> System setting >> General Configuration >> System default language drop down(select Japanese option). Verfiy during voicemail the system prompt is in Japanese		Passed	
UC861S.UPGRADE .U.1061	Upgrade	Backup – Cisco Unified Communications Manager using Disaster recovery system in MCS	To verify that 100% backup of Cisco Unified Communications Manager is successful.		Passed	
UC861S.UPGRADE .U.1064	Upgrade	Installation of Cisco Unified Communications Manager 8.5.1 on C-series	To verify that Cisco Unified Communications Manager 8.5.1 installation is successful.		Passed	
UC861S.UPGRADE .U.1067	Upgrade	Applying License for Cisco Unified Communications Manager 8.5.1 installed on C-series	To verify that license upload is successful.		Passed	
UC861S.UPGRADE .U.1070	Upgrade	Restore – Cisco Unified Communications Manager using Disaster recovery system in C-series	To verify that 100% restore of Cisco Unified Communications Manager is successful.		Passed	
UC861S.UPGRADE .U.1074	Upgrade	Upgrade Cisco Unified Communications Manager publisher 8.5	To verify that upgrade of Cisco Unified Communications Manager 8.6(1a) is successful		Passed	
UC861S.UPGRADE .U.1084	Upgrade	Install the Refresh Upgrade Patch for Unified communications Manager	To verify that installation of refresh upgrade cop file is successful		Passed	
UC861S.UPGRADE .U.1081	Upgrade	Upgrade of Cisco Unity Connection 8.5 primary	To verify that upgrade for release set 8.6(1a) Cisco Unity connection is successful		Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UC861S.UPGRADE.U.1007	Upgrade	8.6(1a) CUC Japanese prompt during voicemail	Choose Japanese option from CUCA >> System setting >> General Configuration >> System default language drop down(select Japanese option).Verfiy during voicemail the system prompt is in Japanese		Failed	CSCtt21815
UC861S.UPGRADE.U.1078	Upgrade	Upgrade Cisco Unified Presence 8.5 Primary	To verify that upgrade of Release set 8.6(1a) Cisco Unified Presence is successful		Passed	

Upgrade 7.1.3 through 8.5(1) to 8.6(1a)

Environment matrix of Upgrade 7.1.3

Product/Component	Base Release Set	Intermediate Release	Target Release set
Cisco Unified Communications Manager	7.1.3.10000-11	8.5.1-10000-26	8.6.1.21004-1
Cisco Unified Communications Manager Locale	cm-locale-ja_JP-7.1.3.1000-1.cop.sgn	cm-locale-ja_JP-8.5.1.1000-1.cop.sgn	cm-locale-ja_JP-8.6.1.1000-1.cop.sgn
Cisco Unity Connection	7.1.3.10000-11	8.5.1.12900-7	8.6.1.21004-1
Cisco Unity Connection Locale	uc-locale-ja_JP-7.1.2.0-139.cop.sgn	uc-locale-ja_JP-8.5.1.1-24.cop.sgn	uc-locale-ja_JP-8.6.1.1-2.cop.sgn
Cisco unified Presence	7.0.9.10000-6	8.5.1-10000-35	8.6.1.10000-34
Cisco Unified Presence Locale	ps-locale-ja_JP-7.0.1.1000-1.cop.sgn	ps-locale-ja_JP-8.5.1.1000-1.cop.sgn	ps-locale-ja_JP-8.6.1.1000-1.cop.sgn
Cisco Unified Survivable Remote Site Telephony	8.0	8.5	8.6(1a)
Cisco Unified Survivable Remote Site Telephony IOS	15.0(1)M XA	15.1.3 T	15.1(4)M
IOS (Voice gateways 2801)	15.0(1)M	15.1.3T	15.1(4)M
IP Communicator	7.0(3)	7.0(3)	8.6(1a)
CUPC	7.0(2)	8.5.1	8.5.1

Test Results

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UC861S.UPGRADE.U.1002	Upgrade	Upgrade Cisco Unified Communications Manager publisher 7.1.3	Verify that upgrade of Cisco Unified Communications Manager 8.5.1 is successful		Passed	
UC861S.UPGRADE.U.1006	Upgrade	Upgrade Cisco Unified Presence 7.1.3 Primary	Verify that upgrade of Release set 8.5.1 Cisco Unified Presence is successful		Passed	
UC861S.UPGRADE.U.1008	Upgrade	Upgrade of Cisco Unity Connection7.1.3 primary	Verify that upgrade to Cisco Unity connection 8.5.1 is successful		Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UC861S.UPGRADE.U.1008	Upgrade	8.5(1) CUC Japanese prompt during voicemail	Choose Japanese option from CUCA >> System setting >> General Configuration >> System default language drop down(select Japanese option). Verfiy during voicemail the system prompt is in Japanese		Passed	
UC861S.UPGRADE.U.1011	Upgrade	Backup – Cisco Unified Communications Manager using Disaster recovery system in MCS	To verify that 100% backup of Cisco Unified Communications Manager is successful.		Passed	
UC861S.UPGRADE.U.1014	Upgrade	Installation of Cisco Unified Communications Manager 8.5.1 on C-series	To verify that Cisco Unified Communications Manager 8.5.1 installation is successful.		Passed	
UC861S.UPGRADE.U.1017	Upgrade	Applying License for Cisco Unified Communications Manager 8.5.1 installed on C-series	To verify that license upload is successful.		Passed	
UC861S.UPGRADE.U.1020	Upgrade	Restore – Cisco Unified Communications Manager using Disaster recovery system in C-series	To verify that 100% restore of Cisco Unified Communications Manager is successful.		Passed	
UC861S.UPGRADE.U.1024	Upgrade	Upgrade Cisco Unified Communications Manager publisher 8.5	To verify that upgrade of Cisco Unified Communications Manager 8.6(1a) is successful.		Passed	
UC861S.UPGRADE.U.1034	Upgrade	Install the Refresh Upgrade Patch for Unified communications Manager	To verify that installation of refresh upgrade cop file is successful.		Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UC861S.UPGRADE .U.1031	Upgrade	Upgrade of Cisco Unity Connection 8.5 primary	To verify that upgrade for release set 8.6(1a) Cisco Unity connection is successful		Passed	
UC861S.UPGRADE .U.1009	Upgrade	8.5(1) CUC Japanese prompt during voicemail	Choose Japanese option from CUCA >> System setting >> General Configuration >> System default language drop down(select Japanese option). Verfiy during voicemail the system prompt is in Japanese		Failed	CSCtt218 15
UC861S.UPGRADE .U.1028	Upgrade	Upgrade Cisco Unified Presence 8.5 Primary	To verify that upgrade of Release set 8.6(1a) Cisco Unified Presence is successful		Passed	

Upgrade 7.1(5) through 8.5(1) to 8.6(1a)

Environment matrix of Upgrade 7.1.5

Product/Component	Base Release Set	Intermediate Release set	Target Release set
Cisco Unified Communications Manager	7.1.5-32900-2	8.5.1-10000-26	8.6.1.21004-1
Cisco Unified Communications Manager Locale	cm-locale-ja_JP-7.1.5.1200-1.cop.sgn	cm-locale-ja_JP-8.5.1.1000-1.cop.sgn	cm-locale-ja_JP-8.6.1.1000-1.cop.sgn
Cisco Unity Connection	7.1.5-32900-2 with ciscocm.cuc_cluster_715.cop.sgn	8.5.1.12900-7	8.6.1.21004-1
Cisco Unity Connection Locale	uc-locale-ja_JP-7.1.2.0-139.cop.sgn	uc-locale-ja_JP-8.5.1.1-24.cop.sgn	uc-locale-ja_JP-8.6.1.1-2.cop.sgn
Cisco unified Presence	7.0.5-10000-18	8.5.1-10000-35	8.6.1.10000-34
Cisco Unified Presence Locale	ps-locale-ja_JP-7.0.4.1000-1.cop.sgn	ps-locale-ja_JP-8.5.1.1000-1.cop.sgn	ps-locale-ja_JP-8.6.1.1000-1.cop.sgn
Cisco Unified Survivable Remote Site Telephony	8.0	8.5	8.6(1a)
Cisco Unified Survivable Remote Site Telephony IOS	15.0(1)M XA	15.1.3 T	15.1(4)M
IOS (Voice gateways 2801)	15.0(1)M	15.1.3T	15.1(4)M
IP Communicator	7.0(3)	7.0(3)	8-6-1-0
Unified Personal Communicator	7.0(2)	8.5.1	8.5.1.18771

Test Results

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UC861S.UPGRADE.U.951	Upgrade	Upgrade Cisco Unified Communications Manager publisher 7.1.5	Verify successful upgrade of Cisco Unified Communications Manager 8.5.1		Passed	
61S.UPGRADE.U.955	Upgrade	Upgrade Cisco Unified Presence 7.1.5 Primary	Verify successful upgrade of Release set 8.5.1 Cisco Unified Presence		Passed	
UC861S.UPGRADE.U.957	Upgrade	Upgrade of Cisco Unity Connection7.1.5 primary	Verify successful upgrade to Cisco Unity connection 8.5.1		Passed	

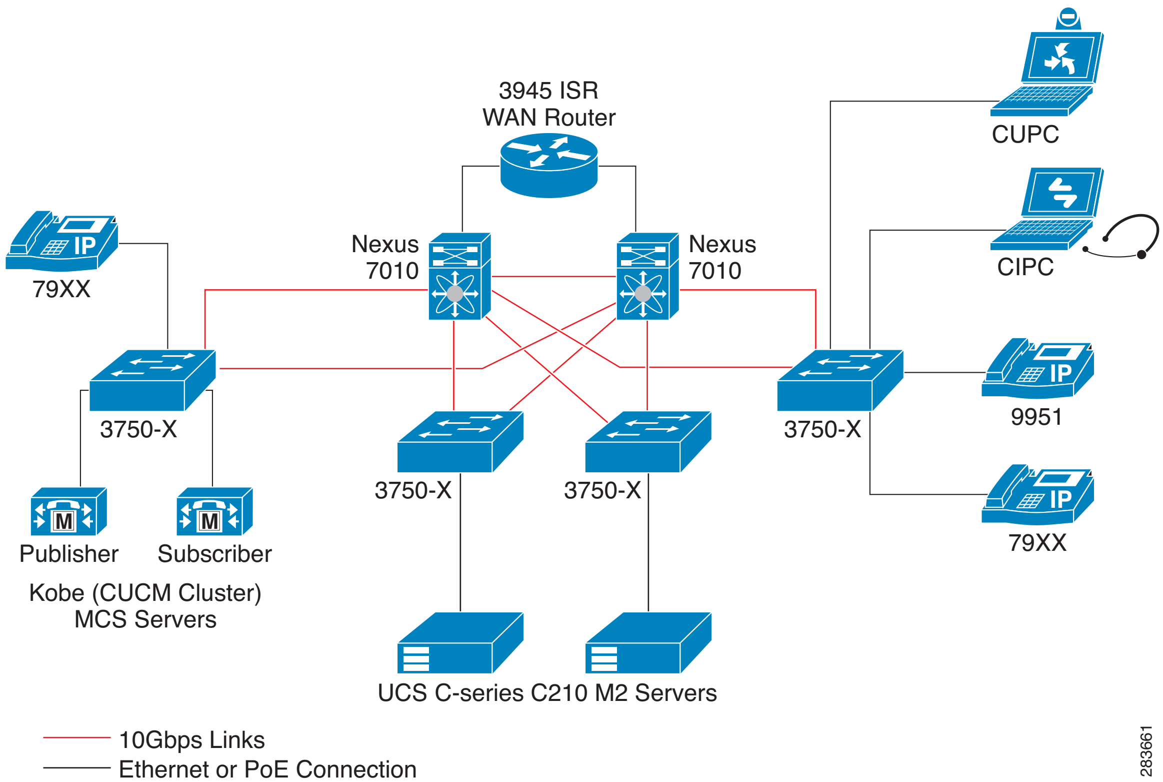
ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UC861S.UPGRADE.U.1010	Upgrade	8.5(1) CUC Japanese prompt during voicemail	Choose Japanese option from CUCA >> System setting >> General Configuration >> System default language drop down(select Japanese option). Verfiy during voicemail the system prompt is in Japanese		Passed	
UC861S.UPGRADE.U.960	Upgrade	Backup – Cisco Unified Communications Manager using Disaster recovery system in MCS	Verify that 100% backup of Cisco Unified Communications Manager is successful.		Passed	
UC861S.UPGRADE.U.963	Upgrade	Installation of Cisco Unified Communications Manager 8.5.1 on C-series	Verify that Cisco Unified Communications Manager 8.5.1 installation is successful.		Passed	
UC861S.UPGRADE.U.966	Upgrade	Applying License for Cisco Unified Communications Manager 8.5.1 installed on C-series	Verify that license upload is successful.		Passed	
UC861S.UPGRADE.U.970	Upgrade	Restore – Cisco Unified Communications Manager using Disaster recovery system in C-series	Verify that 100% restore of Cisco Unified Communications Manager is successful.		Passed	
UC861S.UPGRADE.U.973	Upgrade	Upgrade Cisco Unified Communications Manager publisher 8.5	Verify successful upgrade of Cisco Unified Communications Manager 8.6(1a)		Passed	
UC861S.UPGRADE.U.983	Upgrade	Install the Refresh Upgrade Patch for Unified communications Manager	Verify successful install of refresh upgrade cop file		Passed	

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UC861S.UPGRADE.U.980	Upgrade	Upgrade of Cisco Unity Connection 8.5 primary	Verify successful upgrade for release set 8.6(1a) Cisco Unity connection		Passed	
UC861S.UPGRADE.U.1011	Upgrade	8.6(1a) CUC Japanese prompt during voicemail	Choose Japanese option from CUCA >> System setting >> General Configuration >> System default language drop down(select Japanese option).Verfiy during voicemail the system prompt is in Japanese		Failed	CSCtt21815
UC861S.UPGRADE.U.977	Upgrade	Upgrade Cisco Unified Presence 8.5 Primary	Verify successful upgrade of Release set 8.6(1a) Cisco Unified Presence		Passed	

Standalone Upgrade - Upgrade 8.0.3 to 8.6(1a)

Topology

Figure 3-2 Topology



Environment matrix of Upgrade 8.0.3

Product/Component	Base Release Set	Target Release set
Cisco Unified Communications Manager	8.0.3.10000-8	8.6.1.21004-1
Cisco Unified Communications Manager Locale	cm-locale-ja_JP-8.0.3.2000-1.cop	cm-locale-ja_JP-8.6.1.1000-1.cop.sgn
Cisco Unity Connection	N/A	N/A
Cisco Unity Connection Locale	N/A	N/A
Cisco unified Presence	N/A	N/A
Cisco Unified Survivable Remote Site Telephony	N/A	N/A
Cisco Unified Survivable Remote Site Telephony IOS	N/A	N/A
IOS (Voice gateways 2801)	15.0(1)M	15.0(1)M
Cisco Unified Presence Locale	N/A	N/A
IP Communicator	7.0(3)	8.6.1
CUPC	N/A	N/A

Test Results

ID	Features tested	Case Title	Description	Call Component Flow	Status	Defects
UC861S.UPGRADE.U.1091	Upgrade	Upgrade Cisco Unified Communications Manager publisher 8.0.3	To verify that upgrade of Cisco Unified Communications Manager 8.6(1a) is successful		Passed	
UC861S.UPGRADE.U.1095	Upgrade	Backup – Cisco Unified Communications Manager using Disaster recovery system in MCS	To verify that 100% backup of Cisco Unified Communications Manager is successful.		Passed	
UC861S.UPGRADE.U.1096	Upgrade	Installation of Cisco Unified Communications Manager 8.6(1a) on C-series	To verify that Cisco Unified Communications Manager 8.6(1a) installation is successful.		Passed	
UC861S.UPGRADE.U.1097	Upgrade	Applying License for Cisco Unified Communications Manager 8.6(1a) installed on C-series	To verify that license upload is successful.		Passed	
UC861S.UPGRADE.U.1098	Upgrade	Restore – Cisco Unified Communications Manager using Disaster recovery system in C-series	To verify that 100% restore of Cisco Unified Communications Manager is successful.		Passed	

Regression Testing

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ713F.CUCM.D.001	Cisco Unified Communications Manager	Call routing on a newly-added gateways (members) in a Route Group	Verify that Call routing on a newly-added Route Partition (members) in a Line Group works as expected.		Passed	
UCJ713F.CUCM.D.002	Cisco Unified Communications Manager	Checking CFNA in a Shared Line	Verify that CFNA in a Shared Line works as expected.		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 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 with 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 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 on Shared Lines	Verify that "privacy on hold toggling" setting in the shared line works successfully.		Passed	

ID	Feature Tested	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 shared line	Verify that Auto Pickup feature works successfully with the Privacy Feature and Privacy on hold in Shared line		Passed	
UCJ713F.CUCM.D.009	Cisco Unified Communications Manager	Auto Pick up feature in shared line	Verify that the Auto PickUp feature works successfully on 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 are configured successfully: • Display timer is 0 . • Caller ID display priority enabled is set to True.	IP Phone -> Cisco Unified Communications Manager -> IP Phones in shared DN	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 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 Unified IP Phone.	Cisco Unified IP Phone -> Cisco Unified Communications Manager -> Cisco Unified IP Phone -> Call Park and Blind Transfer	Passed	
UCJ715F.CUCM.D.014	Cisco Unified Communications Manager	Ring back tone in Cisco Unified IP Phones during call transfer	Verify that ring back tone is heard during Call Transfer sucessfully.		Passed	
UCJ715F.CUCM.D.015	Cisco Unified Communications Manager	Blind Transfer across SIP Trunks	Verify Blind transfer across SIP trunks works as successfully		Passed	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ715F.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	
UCJ715F.CUCM.D.017	Cisco Unified Communications Manager	Mutual Hold and Resume Feature	Verify that the Mutual Holdand Resume feature works successfully during a conference.		Passed	
UCJ715F.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	
UCJ715F.CUCM.D.019	Cisco Unified Communications Manager	Ad Hoc 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	
UCJ715F.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	
UCJ715F.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 multiple groups.		Passed	
UCJ715F.CUCM.D.022	Cisco Unified Communications Manager	One Touch Call Pickup	Verify that one touch Call Pickup using a softkey works successfully.		Passed	
UCJ715F.CUCM.D.023	Cisco Unified Communications Manager	Group Pickup	Verify that the one touch Group Pickup using a softkey works successfully.		Passed	
UCJ715F.CUCM.D.024	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 the shared line directory number.		Passed	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ715F.CUCM.D.025	Cisco Unified Communications Manager	Call Park Intra-Cluster Calls between encrypted SIP phones	Verify that the call park feature works successfully for intra-cluster calls between encrypted SIP phones.		Passed	
UCJ715F.CUCM.D.026	Call Forward	Call Forward Chain of 6	Verify that a call can be forwarded 6 times and that each time the call is forwarded to the proper DN and there is voice path		Passed	
UCJ715F.CUCM.D.027	Call Park	Check the settings Call Park Display timer to 0 and Caller ID Display Priority with Enabled set to True	Verify the call park display		Passed	
UCJ715F.CUCM.D.028	Directed Call Pickup	Directed Call Pickup when multiple calls are available for pickup, Check the Call Pick up Group Notification Settings.	Verify that Directed Call Pickup when multiple calls are available for pickup, Check the Call Pick up Group Notification Settings.		Passed	
UCJ715F.CUCM.D.029	CFNA(Roundtable Phone)	Roudtable phone: Verify call forward no answer.	Verify CFNA epAsstnt1 (assistant endpoint 1) calls epFeature (feature applied endpoint). epFeature ringing but no answer. The call is forwarded to epAsstnt2 (assistant endpoint 2)	epAsstnt1 -> epFeature (call forward no answer) -> epAsstnt2	Passed	
UCJ715F.CUCM.D.030	CFNA(IP-to-IP intra-cluster)	IP-to-IP Intra-cluster Call Forward No Answer	Verify that with IP-to-IP intra-cluster call forward no answer the call is forwarded to the proper DN and there is voice path		Passed	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ715F.CUCM.D.031	Forward Hunt No Answer	Roudtable phone: Verify Forward Hunt No Answer	Verify the Forward Hunt No Answer feature. epFeature (feature applied endpoint) is the phone to call hunt pilot number.	epFeature -> hunt pilot number -> epAsstnt1 (no answer) -> (no answer) -> Forward Hunt No Answer -> epAsstnt3	Passed	
UCJ715F.CUCM.D.032	CFUR	Roudtable phone: Verify Call Forward Unregistered CFUR	Verify the Call Forward UnRegistered feature.	1st call: epAsstnt1 -> epFeature; 2nd call:. epAsstnt1 -> epFeature (cfur) -> epAsstnt2	Passed	
UCJ715F.CUCM.D.033	CFB(IP-to-IP Intra-cluster)	IP-to-IP Intra-cluster Call Forward Busy	Verify that with IP-to-IP intra-cluster call forward busy the call is forwarded to the proper DN and there is voice path		Passed	
UCJ715F.CUCM.D.034	CFNA(IP-to-IP Inter-cluster)	IP-to-IP Inter-cluster Call Forward No Answer	Verify that with IP-to-IP inter-cluster call forward no answer the call is forwarded to the proper DN and there is voice path		Passed	
UCJ715F.CUCM.D.035	CF destination Override (Roundtable phone)	Roudtable phone: Verify Call forward destination override	Verify Call forward destination override.		Passed	
UCJ715F.CUCM.D.036	CFNC (Roundtable phone)	Roudtable phone: Verify Call Forward No Coverage (CFNC)	Verify that the CFNC is working fine.	epAsstnt1 -> epFeature (call forward to HuntPilot number) -> epAsstnt3 & epAsstn4 ; Call Back to epFeature ->(CFNC) epAsstn5	Passed	
UCJ715F.CUCM.D.037	Cisco Unified Communications Manager	Hold/Resume During Ad Hoc Conference	Verify the ability to perform a Hold and Resume during an ad hoc conference.	SIP Phone ->SIP Phone -> Softkey	Passed	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ715F.CUCM.D.038	Conference	Consultative Transfer During Ad Hoc Conference	Verify the ability to perform a consultative transfer during an ad hoc conference.	SIP Phone --> adhoc conference-->Transfer -->SIP Phone	Passed	
UCJ715F.CUCM.D.039	Conference	Roudtable phone: Verify Ad hoc conference by using Conf Softkey.	Verify Ad hoc conference by using Conf Softkey		Passed	
UCJ715F.CUCM.D.040	gpickup	iSAC codec: verify <normal> group pick up by GPickUp softkey.	Verify normal group pick up by GPickUp softkey.		Passed	
UCJ715F.CUCM.D.041	oPickup	iSAC codec: Verify <normal> other group pick up by OPickUp softkey.	Verify other group pick up by OPickUp softkey		Passed	
UCJ715F.CUCM.D.042	Cisco Unified Communications Manager	iSAC codec: Verify call forward all	Verify CFA using iSAC codec		Passed	
UCJ715F.CUCM.D.043	DND	DND feature during Callback	Verify the DND feature during Callback		Passed	
UCJ715F.CUCM.D.044	DND	DND Feature and Hold Reversion	Verify the DND Feature and Hold Reversion		Passed	
UCJ715F.CUCM.D.045	Cisco Unified Communications Manager	Hold Reversion for Shared Line SCCP Phone	Verify Hold Reversion for Shared Line SCCP Phone		Passed	
UCJ715F.CUCM.D.046	Cisco Unified Communications Manager	Hold Reversion for Shared Line SIP Phone	Verify Hold Reversion for Shared Line SIP Phone		Passed	
UCJ715F.CUCM.D.047	Barge	Ability of a SIP phone to use the Barge feature on a call involving another SIP phone that then ends the barge-in	Verify the ability of a SIP phone to use the Barge feature on a call involving another SIP phone that then ends the barge-in		Passed	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ715F.CUCM.D.048	Cisco Unified Communications Manager	A call coming from a SIP phone in the "Other Group" list has priority when multiple calls come from different groups in the list.	Verify that a call coming from a SIP phone in the "Other Group" list has priority when multiple calls come from different groups in the list.		Passed	
UCJ715F.CUCM.D.049	Cisco Unified Communications Manager	One touch other group pickup	Verify One touch other group pickup		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.	Cisco Unified IP Phone -> Cisco Unified Communications Manager -> SIP trunk -> SME -> SIP trunk -> Cisco Unified Communications Manager B -> Cisco Unified IP Phone	Passed	
UCJ851S.SME.U.002	SME	Transfer a call hrough 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 -> ICT trunk -> SME -> SIP trunk -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> Transfer -> Cisco Unified IP Phone C.	Passed	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
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 -> ICT trunk -> SME -> ICT trunk -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> Blind Transfer -> ICT trunk -> SME -> ICT trunk -> Cisco Unified Communications Manager -> Cisco Unified IP Phone C	Passed	
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 -> ICT trunk -> SME -> ICT trunk -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> Consult Transfer -> ICT trunk -> SME -> ICT trunk -> Cisco Unified Communications Manager -> Cisco Unified IP Phone C	Passed	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
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 -> SIP trunk -> SME -> ICT trunk -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> Blind Transfer -> ICT trunk -> SME -> SIP trunk -> Cisco Unified Communications Manager -> Cisco Unified IP Phone C.	Passed	
UCJ851S.SME.U.006	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 -> SIP trunk -> SME -> ICT trunk -> Cisco Unified Communications Manager -> Cisco Unified IP Phone B -> Transfer -> Cisco Unified IP Phone C	Passed	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ851S.SME.U.007	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 -> secure SIP trunk -> SME -> secure SIP trunk -> Cisco Unified Communications Manager B -> Cisco Unified IP Phone B -> Call Park(2008) -> Cisco Unified IP Phone C -> Dials 2008.	Passed	
UCJ851S.SME.U.008	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 -> SIPT -> SME -> SIPT -> CUBE -> SIPT -> CME -> Cisco Unified IP Phone B -> Blind Transfer -> SIPT -> CUBE -> SIPT -> SME -> SIPT -> Cisco Unified Communications Manager A -> Cisco Unified IP Phone C	Passed	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ851S.SME.U.009	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 -> SIPT -> SME -> SIPT -> CUBE -> SIPT -> CME -> Cisco Unified IP Phone B -> Consult Transfer -> SIPT -> CUBE -> SIPT -> SME -> SIPT -> Cisco Unified Communications Manager A -> Cisco Unified IP Phone C	Passed	
UCJ851S.SME.U.010	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	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ851S.SME.U.011	SME	Adhoc conference that involves 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 SIP trunk, inter-cluster trunk, and a PSTN phone.	Cisco Unified IP Phone 1 -> Cisco Unified Communications Manager -> SIPT -> SME -> ICT -> Cisco Unified Communications Manager -> Cisco Unified IP Phone 2; Conference -> Cisco Unified IP Phone 1 -> Cisco Unified Communications Manager -> SIPT -> SME -> SIPT -> SIP GW -> PSTN	Passed	
UCJ851S.SME.U.012	SME	Hold and Resume a call that involves ICT trunks to SME	Verify that the Hold and Resume feature for a call works successfully,	Cisco Unified IP Phone -> Cisco Unified Communications Manager -> ICT trunk -> SME -> ICT trunk -> Cisco Unified Communications Manager B -> Cisco Unified IP Phone	Passed	
UCJ851S.SME.U.013	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 -> SIPT -> SME -> SIPT -> Cisco Unified Communications Manager -> Cisco Unified IP Phone 2	Passed	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ713F.CUCM.U.014	Cisco Unified Communications Manager	Call forward all by softkey	Verify if epFeature (feature applied endpoint) is the phone pushing the softkey; epAsstnt1 (assistant SCCP endpoint 1) is the phone making calls; epAsstnt2 (assistant SCCP endpoint 2) is the target of call forward to.		Passed	
UCJ713F.CUCM.U.017	Cisco Unified Communications Manager	Ad hoc Conference between SCCP and SIP phones, using Conf Softkey	Verify by setting up EpAsstnt1 (assistant endpoint 1) makes 1st call to epFeature (feature applied endpoint). EpAsstnt2 (assistant endpoint 2) makes 2nd call to epFeature. EpAsstnt3 (assistant endpoint 3) also places a call to epFeature. epFeature joins these 3 calls by pressing conf softkey and creates a conference call.		Passed	
UCJ713F.Phone.U.019	CFB(IP-to-IP Inter-cluster)	IP-to-IP Inter-cluster Call Forward Busy	Verify that with IP-to-IP inter-cluster call forward busy the call is forwarded to the proper DN and there is voice path		Passed	
UCJ713F.Phone.U.036	Cisco Unified Communications Manager	Multiple Calls on Multiple Lines	Verify EpAsstnt1 (assistant endpoint 1) by making 1st call to epFeature (feature applied endpoint). EpAsstnt2 (assistant endpoint 2) makes 2nd call to epFeature on a different line. EpFeature can switch the 1st call and 2nd call correctly.		Passed	
UCJ713F.Phone.U.046	Cisco Unified Communications Manager	Calls between Two Endpoint	Verify calls between Unified IP Phone A and Unified IP Phone B. Verify the audio path.	Cisco Unified IP Phone 1 -> Cisco Unified Communications Manager -> Cisco Unified IP Phone 2	Passed	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJ713F.CUCM.D.050	SIP Trunk	SIP trunk call using g729 codec	Verify that the calls through SIP Trunk using G729 Codec are successful.	Unified IP Phone -> Unified Communications Manager -> SIP Trunk -> Unified Communications Manager -> Unified IP Phone	Passed	
UCJ713F.CUCM.D.080	Cisco Unified Communications Manager	DND	Verify that DND feature for SCCP phones 7961,7975 Phones is working as expected.		Passed	
UCJ713S.CUCM.T.081	Cisco Unified Communications Manager	CFA	Verify that InterclusterTrunk for Call Forward works as expected between Unified Communications Manager Clusters	Unified IP Phone -> Cisco Unified Communications Manager (Cluster 1) --> ICT -> Cisco Unified Communications Manager (Cluster 2)--> Unified IP Phone	Passed	
UCJ713F.CUCM.T.083	Cisco Unified Communications Manager	Call Back	Verify that the Call Back feature works as expected		Passed	
UCJ713F.CUCM.D.083	Cisco Unified Communications Manager	Meet me conference	Verify that the Meet me conference feature works fine between the SCCP phones		Passed	
UCJGB22.CUCM.D.174	Cisco Unified Communications Manager	IP-to-IP Inter-cluster Call Forward No Answer to Remote Site	Verify that with IP-to-IP inter-cluster call forward no answer to remote site the call is forwarded to the proper DN and there is voice path		Passed	

ID	Feature Tested	Title	Description	Call Component Flow	Status	Defects
UCJGB22.CUCM.D.175	Cisco Unified Communications Manager	IP-to-IP Intra-cluster Multiple, Concurrent Ad-hoc Conferences using Join Softkey and Conference Softkey	Verify that IP-to-IP intra-cluster multiple, concurrent ad-hoc conference calls can be established, one with the Join softkey, the other with the Conference softkey		Passed	
UCJGB22.CUCM.D.253	Cisco Unified Communications Manager	IP-to-PSTN Intra-cluster Call Transfer	Verify IP-to-PSTN intra-cluster call transfer		Passed	

Related Documentation

Cisco Unified Communications Manager Documentation Guide

Software Compatibility Matrix

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/compat/ccmcompmatr.html

IPv6 Deployment Guide

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/ipv6/ipv6srnd.html

Cisco Unified Communications Manager on Virtualized Servers

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/virtual/servers.html

Cisco Unified Communications Manager Administration Guide 8.6(1)

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmcfg/bccm-861-cm.html

Cisco Unified Communications Manager Features and Services Guide 8.6(1)

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fsgd-861-cm.html

CUPS 8.5 Documentation Guide

http://www.cisco.com/en/US/partner/docs/voice_ip_comm/cups/8_0/english/doc_guide/documentation/guide/dgcup.html

CME Design Guide

http://www.cisco.com/en/US/partner/docs/voice_ip_comm/cucme/srnd/design/guide/cmesrnd.html

Supported Hardware

http://www.cisco.com/en/US/prod/collateral/voicesw/ps6790/ps5748/ps378/prod_brochure0900aecd8062a4f9.html

Guideline for upgrading from 8.5 to 8.6

http://www.cisco.com/en/US/docs/voice_ip_comm/connection/8x/upgrade/guide/8xcucrug007.html

Refresh Upgrade

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/rel_notes/8_6_1/cucm-rel_notes-861a.html#wp1996982

OVF template details

http://www.cisco.com/en/US/docs/voice_ip_comm/connection/8x/release/notes/861cucrn.html#wp1086223

Upgrade of 8.5 to 8.6 on UCS Servers (Vmware)

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/rel_notes/8_6_1/cucm-rel_notes-861.html#wp1968761

APPENDIX A

System Test Results for IP Telephony: Cisco Unified Communications System Release 8.6(1a)

This appendix contains test results for the following topics:

- Business Edition
- Cisco Cius
- Cisco Emergency Responder
- Miscellaneous
- Mobile Clients
- Video
- Virtualization Experience Client
- All Tests
- Regression Tests

Business Edition

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.001	PSTN Local Breakout- in the Central and in the Remote Site	Verify the PSTN local break out in the central and remote sites, and check the scenario for both IP and analog type of phones.	Central Phone1 -> Unified CM(E1) -> PSTN -> POTS Phone1; Cisco VG224 Central Phone1 -> Unified CM(E1) -> PSTN -> POTS Phone2; Remote Phone1 -> Unified CM -> Remote1(E1) -> PSTN -> POTS Phone1; Remote1 Analog Ph1 -> Unified CM -> Remote1 (E1) -> PSTN -> POTS Phone2;	Passed	
UC861EF.SMB.002	Multiple Gateway Support in the Central Site	Verify that Cisco Unified Communications Manager supports multiple gateways by making PSTN calls from central site, with first preference being Unified Communications Manager integrated dual E1 PRI link, followed by central gateway (2901) E1 PRI link to the PSTN network in case of a failure. Verify the scenario for both Analog and IP type of phones.	Cen Ph1 -> Unified CM(E1) -> PSTN -> POTS Ph1;Cen Ph2 -> Unified CM -> Cen 2901Gateway(E1) -> PSTN -> POTS Ph3;VG224 Ph1 -> Unified CM(E1) -> PSTN -> POTS Ph3;VG224 Cen Ph2 -> Unified CM -> Cen 2901Gateway(E1) -> PSTN -> POTS Ph3; Cen Ph1 -> Unified CM(E1) -> PSTN -> (E1)Unified CM -> Cen Ph2	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.004	Adhoc Conference using Central Conference Phone	Verify the ability to call Cisco Unified SIP Phone 3905 at remote site from the Unified IP Conference Station 7937 at central site, place the remote site 2 Unified 6900 series IP phone 1 to the conference call, and add the POTS endpoint to the conference call by dialing the PSTN number. Verify the ability to place the remote site 3 Cisco IP Communicator to the same conference call, ensuring each remote site is using different codecs such as iLBC, G279 and G711.	Cen Unified IP Station 7937 Ph1 -> Unified CM -> Rem1 Unified SIP 3905 Ph1;Cen Unified 7937 Ph1->Unified CM->Conf->Rem2 Unified IP 6900 Ph1; Cen Unified 7937 Ph1>Unified CM(E1)>Conf->PSTN->POTS Ph1; Cen Unified 7937 Ph1->Unified CM->Conf->Cisco IP Communicator	Passed	
UC861EF.SMB.005	Adhoc Conference involving Cisco Unified SIP Phone 3905, Unified IP Phone 6900 Series, Cisco IP Communicator and Plain Old Telephone Systems (POTS) Endpoint	Verify the Unified SIP Phone 3905 in central site calls Unified SIP Phone 3905 in remote site 1, and places the Unified SIP Phone 6911 Phone1 in Remote site 2 to the conference call, and the phone in remote site 2 place the central site Cisco IP Communicator Phone1 to the conference call. Verify that from the central site Cisco IP Communicator Phone1 places the POTS endpoint to the conference call by dialing the POTS number.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.006	Voice Mail Deposit and Retrieval Involving Cisco Unified SIP Phone 3905 Endpoint	Verify voicemail deposit and retrieval services are successful when the Plain Old Telephone Systems (POTS) phone calls central site Cisco Unified SIP Phone 3905 (with setting Call Forward No Answer to voice mail), and when the remote site 2 Unified SIP Phone 3905 endpoint calls the central site Unified IP Phone 6900 Series endpoint (with setting Call Forward No Answer to voice mail). Verify the message waiting indicator function in both the scenarios.	POTS Ph1>PSTN>E1Unified CM>Cen 3905 Ph1>CFNA>VM Unified CM;POTS Ph1>Deposit VM; Cen 3905 Ph1>Retrieve VM; Rem1 3905 Ph1>Unified CM1>Cen 69XX Ph2>CFNA>VM Unified CM;Rem 3905 Ph1>Deposit VM; Cen 69XX Ph2>Retrieve VM	Passed	
UC861EF.SMB.007	Extension mobility on Cisco IP Communicator and Unified IP Phone 6900 phone series	Verify the extension mobility on Cisco IP Communicator and Unified IP Phone 6900 phone series.	Rem Cisco IP Communicator->Unified CM->Extension Mobility;(Retrieves the Unified 69XX device profile);Rem Unified 69XX Ph->Unified CM->EM;(retrieves the CUC-RTX device profile);Cen Unified 69XX Ph->Unified CM->EM;(retrieves the CUC-RTX device profile)	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.008	Music on Hold on Cisco Unified SIP Phone 3905 Endpoint	Verify the Music on Hold service by making a call from Unified SIP Phone 3905 Phone1 in central site to Unified IP Phones 69XX Phone2 in remote site, and from Unified SIP Phone 3905, Phone1 places the call on hold. Verify Unified IP Phones 69XX Phone2 receives the music stream from Unified CM, resumes the call on Unified SIP Phone 3905 Phone1, and the Unified IP Phone 69XX Phone2 gets disconnected from the music stream and reconnects to Unified SIP Phone 3905 phone 1 in central site. Verify Music on Hold with Call park and call transfers.	Central Unified SIP Phone 3905 Ph1->Unified CM->Remote Unified IP 69XX Ph1 ;Central Unified SIP 3905 Ph1->hold; Remote Unified IP 69XX Ph1->MoH(Unified CM);Central Ph1->Resume; Central Unfiied SIP 3905 Ph1->Unified CM->Remote Unified IP 69XX Ph1;	Passed	
UC861EF.SMB.009	Emergency Call from Central Site	Verify the Emergency call (911) from central site is routed over the PSTN network by Unified Communications Manager- via its own integrated T1/E1 interface over the PSTN network and reaches the Public Safety Answering Point (PSAP) unit (Plain Old Telephone Systems endpoint).Verify the caller ID on Plain Old Telephone Systems endpoint to ensure it contains the correct translated number(DID) of central site and check the call back from the PSTN endpoint to central site endpoint is successful.	Central Phone1->Unified CM(E1)->PSTN->PSAP(POTS endpoint);PSAP(POTS endpoint)->PSTN->(E1) Unified CM->Central Phone1;-	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.010	Emergency Call from Remote Site	Verify the Emergency call (911) from remote site is routed over the PSTN network by Cisco Unified Communications Manager via remote site and reaches the PSAP unit (POTS endpoint). Verify the caller ID on POTS endpoint and make sure it contains the correct translated number(DID) of remote site and check the call back from the PSTN endpoint to the remote site endpoint is successful.	Remote1 Phone1->Unified CM->Remote1(E1)->PSTN->PSAP(POTS endpoint);PSAP(POTS endpoint)->PSTN->(E1) Remote 1->Unified CM->Remote Phone1;Remote3 Ph1->Unified CM(E1)->PSTN->PSAP(POTS endpoint);PSAP(POTS endpoint)->PSTN->(E1) Unified CM->Remote3 Phone1	Passed	
UC861EF.SMB.011	Plain Old Telephone Systems (POTS) endpoint calls Autoattended number and Transfers to central Unified IP Phone 6900 Series	Verify that the PSTN endpoint dials the AutoAttendant Directory Number in central site over PSTN network, the AutoAttendant requests the user to dial the extension and PSTN endpoint dials and transfers the call to the central 6900 Unified IP Phone Series endpoint. Verify the AutoAttendant handles three simultaneous incoming calls coming to Unified Communications Manager via its own integrated E1 interface, with the type of calls being different, such as the first call being from POTS endpoint in PSTN network and other two calls being VOIP calls coming from different remote sites.	PSTN Ph1->PSTN->(E1) Unified CM->AutoAttendant(Unified CM);AutoAttendant(Unified CM)->request the user to dial the extension; PSTN Ph1->dials Central Unified IP phone Series 6900 (DN);AutoAttendant(Unified CM)->Transfer->Central Unified IP 6900 Series Ph1	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.012	Auto-Attendant with Hunt grouping	Verify that the PSTN endpoint dials the Auto Attendant DN in central site over PSTN network and Auto Attendant requests the user to dial the department number, which needs to be connected (for example 1 for sales, 2 for marketing etc.,). Verify Auto Attendant transfers the call to that department (which is hunt route group of users) based on input key press, and if the first user in the hunt group is busy, then Auto Attendant transfers to second user. Verify if the incoming PSTN call connects to second user in the Hunt route group and ensures media path is established successfully.	PSTN Phone1->PSTN->(E1)Unified CM->Auto Attendant; Auto Attendant->Transfer->Central Unified 69XX Phone1	Failed	CSCto59303 (3|R)
UC861EF.SMB.013	Remote Site Uses Centralized PSTN-Break Out	Verify the PSTN from remote site (which doesn't have local PSTN Gateway) calls the Plain old telephone systems (POTS) number, then it uses central site Unified Communications Manager E1 Internal links to connect to the PSTN network.	Remote2 Phone1->Unified CM(E1)->PSTN->POTS Phone1; POTS Phone2->(E1)Unified CM->Remote2 Phone3;	Passed	
UC861EF.SMB.014	Busy Lamp Field (BLF) Support	Verify the Busy Lamp Field (BLF) indication on central phone 1(On phone1,configure the BLF speed-dial to Remote phone 1),when remote phone 1 is busy on another call with central phone 2.	Central Phone1(Speed-dial Remote Phone1);Remote Phone1->Unified CM->Central Phone2 (Central Phone1 should have the BLF indication)	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.015	Logical Partitioning-Central site	Verify that POTS endpoint from PSTN Network (Geo-Location A) calls the phone in central site (Geo-Location A), then Unified Communications Manager should not be allowed to do the transfer of the call to Phone in remote site1 (in Geo-Location B) over Voice Over Internet Protocol network.	POTS Phone1->PSTN->(E1)Unified CM->Central Phone1->Transfer->Unified CM->Remote Phone1 - Not allowed	Passed	
UC861EF.SMB.016	Logical Partitioning-Remote Site	Verify that POTS endpoint from PSTN Network(Geo-Location A) calls the phone in the remote site(Geo-Location B), then Unified Communications Manager is not allowed to transfer the call to central site (Geo-Location A) over Voice Over Internet Protocol network.	POTS Phone1->PSTN->(E1)Remote->Unified CM->Remote Phone1->Transfer->Unified CM->Central Phone1;Transfer not allowed	Passed	
UC861EF.SMB.017	Toll-by-Pass	Verify that POTS endpoint from PSTN network calls the phone in the central site, and then Unified Communications Manager in central site is allowed to transfer the call to phone in remote site over VOIP network.	POTS Phone1->PSTN->(E1)Unified CM->Central Phone1->Transfer->Unified CM->Remote Phone1; POTS Phone1->PSTN->(E1)Remote->Unified CM->Remote Phone1->Transfer->Unified CM->Central Phone1	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.101	Failover to subscriber on Cisco Media Convergence Server (MCS) when Integrated Services Routers (ISR) hosting CiscoÂ® Service Ready Engine (SRE) with Cisco Unified Communications Manager publisher is down	Verify that the endpoints and call processing failover to subscriber on Cisco Media Convergence Server (MCS) when Integrated Services Routers (ISR) hosting CiscoÂ® Service Ready Engine (SRE) with Cisco Unified Communications Manager publisher is down.	Central SCCP Phone 1->Unified CM Publisher->Central SCCP Phone 2	Passed	
UC861EF.SMB.102	Failover to SRST ISR in central site when both Unified Communications Manager publisher and subscriber is down.	Verify if phones failover to SRST ISR in the central site when Unified Communications Manager goes down.	Central SCCP Phone 1->Unified CM->Rem SCCP Phone 1	Passed	
UC861EF.SMB.103	Conference initialization failover with central to remote RSVP.	Verify status of conference call in its initial states when call manager status goes down.	Central Phone A->Unified CM->Remote 1 Phone B->Unified CM->Conference->Remote 2 Phone C	Passed	
UC861EF.SMB.104	Check for video escalation and de-escalation on call transfer from Unified IP Phones 99xx series to an SCCP phone between central and remote sites	Verify the ability to check for video escalation on call transfer with Central to remote RSVP.	Central Unified IP Phone 99xx series->Unified CM->Remote 1 SCCP phone->Unified CM->Transfer->Remote 1 Unified IP Phone 99xx Series->Unified CM->Transfer->Remote 2 SIP Phone.	Passed	
UC861EF.SMB.105	Hold/Resume on Shared Line with Central - Remote RSVP	Verify the ability to check for hold/resume on shared line between central and remote sites with RSVP. Verify the ability to check for video escalation in supported phones.	Central Unified IP 99xx series phone 1->Unified CM->Remote 1 SCCP->Hold->Remote 1 Unified IP 89xx series phone 2 (Shared line) RESUME.	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.106	Video escalation on call forwarding over SIP Inter Cluster Trunk with End to End RSVP.	Verify video escalation over SIP Inter Cluster Trunk with End to End RSVP when call is forwarded from non-video to video phone.	Unified IP 99xx Phone 1->Unified CM 1->SIP Inter Cluster Trunk->SCCP Phone->Call Forward No Answer->Unified CM2 -> Transfer->Unified IP 99xx Phone2	Passed	
UC861EF.SMB.107	Adhoc conference with Unified IP Phone 89xx , SIP and SCCP phones in two clusters over SIP Inter Cluster Trunk.	Verify audio conference between Unified IP Phone 89xx,SIP and SCCP Phones over two clusters with End to End RSVP over SIP Inter Cluster Trunk.	Cen 89xx->Unified CM1->Remote SCCP Phone->Conference->Unified CM2->Central SIP Phone	Passed	
UC861EF.SMB.109	Early offer / Delayed offer Interworking	Verify interworking between endpoints supporting early offer and those that do not over SIP Inter Cluster Transfer.	Central 7945->Unified CM 1-> SIP Inter Cluster Trunk->Unified CM2->Unified IP phone 99xx Phone	Passed	
UC861EF.SMB.110	Trombone Path replacement	Verify Trombone path replacement in Cisco Unified Communications Manager on Cisco Services Ready Engine	Cluster 1 Phone 1->Unified CM 1 -> SIP Inter Cluster Trunk->Unified CM 2->Cluster 2 SCCP Ph->Transfer->Unified CM 1->Cluster 1 Phone 2	Passed	
UC861EF.SMB.111	Path replacement capability of Cisco Unified Communications Manager on Cisco Services-Ready Engine (SRE)	Verify path replacement on Cisco Unified Communications Manager on Cisco Services Ready Engine.	Cluster 1 Phone->Unified CM 1-> SIP Inter Cluster Trunk->Unified CM 2->Cluster 2 Phone->Transfer->SIP Inter Cluster Trunk->Unified CM 3->Cluster 3 Phone	Passed	
UC861EF.SMB.112	Automated Alternate Routing when Bandwidth Unavailable between Central and Remote Site	Verify call re routing over PSTN when bandwidth is unavailable between central and branch offices.	Remote Phone1->Unified CM->Central Phone1 When bandwidth unavailable Remote Phone1->Remote PSTN Gateway->Unified CM->Central Phone 1	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.113	Using Remote PSTN Capability by Central Phones when Central PRI Link is Down.	Verify the capability to use alternate PSTN gateways when primary PSTN gateway is unavailable.	Central SCCP Phone 1->Remote PSTN Gateway->PSTN Phone.	Passed	
UC861EF.SMB.114	Meet Me Conference Testing	Verify Meet Me Conference in Unified Communications Manager over Cisco Services-Ready Engine.	Central SCCP Phone1->Meet me Remote 1 Unified IP 99xx phone->Meet Me Remote 2 SCCP phone->Meet me	Passed	

Cisco Cius

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.016	Message Actions with Visual Voice Mail on Cisco Cius	Verify the message actions with Visual Voice Mail on Cisco Cius.	Audio Phone->Unified CM1->Cisco Cius Call Forward No Answer->Cisco Unity Connection Voicemail	Passed w/ Exception	CSCtq13847
UC861IF.CUS.017	Cisco Unity Connection in cluster goes down while playing Visual Voice Mail message on Cisco Cius	Verify that the Cisco Unity Connection in a cluster goes down while playing Visual Voice Mail message on Cisco Cius.	Audio Phone->Unified CM1->Cisco Cius Call Forward No Answer->Cisco Unity Connection Voicemail	Passed w/ Exception	Provide a DNS name that resolves to both the server names for Visual Voice Mail to failover to secondary server. Unified CM setting 'Secondary Voicemail Server' setting does not work currently.
UC861IF.CUS.018	Download Voicemails to Cisco Cius from Cisco Unity Connection Server2 When Server1 is Down	Verify the ability to download Voicemails to Cisco Cius from Cisco Unity Connection Server2 when Server1 is down.	Cisco Cius->Cisco Unity Connection Cluster Server2	Passed w/ Exception	Provide a DNS name that resolves to both the server names for Visual Voice Mail to failover to secondary server. Unified CM setting 'Secondary Voicemail Server' setting does not work currently.
UC861IF.CUS.060	Multiway conference from Video Communication Server endpoint to Cisco CIUS and CiscoUnified IP Phone 9971 behind Cisco Unified Communications Manager	Verify whether Cisco CIUS is able to join multiway conference with Cisco Video Communication Server endpoint.	Cisco Telepresence Quickset C20-Cisco Video Communication Server-SIP Trunk ---Unified CM ----Unified IP Phone 9971 ---Cisco Telepresence Quickset C20(multiway)--SIP Trunk-Unified CM-Cisco CIUS	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.062	Verify Ability to Switch Between Shared Lines with Video	Verify a Cisco Cius with two lines (one shared line) is able to switch between two active calls with video.		Passed	
UC861IF.CUS.063	Device Mobility with Cisco Cius	Verify that device mobility works with Cisco Cius.	Cisco Cius SRST location -->Unified CM -->Conference -->IP Phone1 and IP Phone2	Passed	
UC861IF.CUS.066	Cisco Cius Interoperability with Cisco Unified Meeting Place	Verify whether Cisco Cius is able to join Cisco Unified Meeting Place meeting and view the video of all the participants.	Cisco Cius ----Unified CM ---- SIP Trunk ---- -Cisco Unified Meeting Place	Passed	
UC861IF.CUS.098	Cisco Cius can have a voicemail box in Cisco Unity Express	Verify if Cisco Unity Express can provide voicemail service for Cisco Cius.	Ph1-->Unified CM -->SIP Trunk -->Unified CM -->Cisco Cius-->Call Forward No Answer-->Unified CM-->JTAPI -->Cisco Unity Express	Passed	
UC861IF.CUS.201	Point to Point call from Cisco IP Video Phone E20 registered to Video Communication Server to Cisco Cius registered to Cisco Unified Communications Manager	Verify the video call can be placed and put on hold by Cisco Cius registered to Cisco Unified Communications Manager and the call resumes back to video.	Cisco Cius --- Abilene Unified CM --- SIP Trunk ---Cisco IP Video Phone E20->Video Communication Server	Passed	
UC861IF.CUS.202	Call transfer between Cisco Cius and Cisco TelePresence Quick Set C20	Verify video call can be transferred from Cisco Cius registered to Cisco Unified Communications Manager to Cisco TelePresence Quick Set C20 registered to Video Communication Server.	Cisco Cius-> Abilene Unified CM->SIP trunk->Video Communication Server->Cisco TelePresence Quick Set C20->Cisco Cius Transfer ---SIP trunk --- Cisco TelePresence 1700 MXP->Video Communication Server	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.203	Conference Cisco Unified Communications Manager and Cisco Video Communication Server Endpoints using Cisco MeetingPlace Software Bridge	Verify conference can be established between Cisco CIUS , Cisco IP video Phone E20 and Tandberg MXP 1700 series using Cisco MeetingPlace Adhoc bridge.	Cisco CIUS-->MSP Unified CM--H.225 trunk--GateKeeper-> Cisco Video Communication Server--Cisco IP Video Phone E20--conference using Cisco CIUS--Tandberg MXP 1700->H.323->Cisco Video Communication Server	Passed w/ Exception	Bad Video Quality- Known Issue
UC861IF.CUS.204	Conference with Cisco TelePresence Video Communication Server and Cisco Cius Endpoints via Cisco Media Experience Engine and Cisco TelePresence Multipoint Switch	Verify conference between Cisco TelePresence Video Communication Server and Cisco TelePresence System 1000 via Cisco Media Experience Engine and Cisco TelePresence Multipoint Switch.	Cisco IP Video Phone E20->Cisco VCS---SIP trunk----SME ---SIP trunk->Cisco MXE->Unified CM--Cisco TelePresence Multipoint Switch-> Conference ---Cisco Cius->Unified CM---Cisco MXE---SME---SIP Trunk---Cisco TelePresence Multipoint Switch Conference	Passed	
UC861IF.CUS.205	Scheduled Conference using Codian Multipoint Control Unit	Verify whether Cisco IP Video Phone E20, Cisco TelePresence Quick Set C20 registered to Video Communication Server and Cisco Cius registered to Cisco Unified Communications Manager are able to join Scheduled conference using Codian Multipoint Control Unit bridge.	Cisco TelePresence Quick Set C20 Cisco Cius Cisco IP Video Phone E20 ---- H.323 Destination Number --- Codian MCU	Passed w/ Exception	Bad Video Quality- Known Issue

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.206	Adhoc Conference Using Cisco TelePresence MCU and Cisco TelePresence Video Communication Server Multiway Feature	Verify if the endpoints can join the conference bridge using multiway feature in an Adhoc Conference Using Cisco TelePresence MCU and Cisco TelePresence Video Communication Server Multiway Feature.		Passed	
UC861IF.CUS.207	Inter-cluster Cisco Cius Peer-to-Peer call with Cisco TelePresence System through QSIG enabled SIP Trunks With End-to-End RSVP	Verify that Cisco Cius end point can make Peer-to-Peer calls to Cisco TelePresence System with End-to-End RSVP enabled with supplementary services.	Cisco Cius->Cisco Call Manager1->SIP Trunk (QSIG)->Cisco Call Manager2->SIP Trunk (QSIG)->Cisco TelePresence System	Passed	
UC861IF.CUS.208	Inter-cluster Secure Cisco Cius Peer-to-Peer Call with Cisco TelePresence System	Verify if the secure Cisco Cius end point can make Peer-to-Peer calls with Secure Cisco TelePresence System	Cisco Cius->Cisco Call Manager1->SIP Trunk(QSIG)->Cisco Call Manager2->SIP Trunk(QSIG)->Cisco TelePresence System	Passed	
UC861IF.CUS.209	Inter-cluster Cisco Cius native interoperability Peer-to-Peer call with Cisco TelePresence System with Trusted Relay Points Enabled	Verify native interoperability between Cisco Cius and Cisco TelePresence System with Trusted Relay Points enabled.	Cisco Cius->Cisco CallManager1->SIP Trunk->Cisco CallManager2->SIP Trunk->Cisco TelePresence System	Passed	
UC861IF.CUS.210	Cisco CIUS native interoperability with Cisco Telepresence System when in Wireless mode	Verify test video interoperability with Cisco TelePresence System when Cisco CIUS is operating in wireless mode.	Cisco CIUS->Cisco CallManager1->SIP Trunk->Cisco CallManager2->SIP Trunk->Cisco TelePresence System	Passed w/ Exception	Bad Video Quality- Known Issue
UC861IF.CUS.211	Cisco Cius Video Call Preservation	Verify that test video call stays up when the Cisco Unified Communications Manager that Cisco Cius is registered to, goes down.	Cisco Cius->Cisco CallManager1->SIP Trunk->Cisco CallManager2->SIP Trunk->Cisco TelePresence System	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.502	Inter-cluster Instant Messaging interoperability with Cisco Unified Personal Communicator 7 user, multiple Instant Messaging sessions	Verify the ability to begin an Instant Messaging chat with a Cisco Unified Personal Communicator 7 user, and when the chat is ongoing, initiate an Instant Messaging with the Cisco CIUS user from another Cisco Unified Personal Communicator 7 client. Verify messages are properly exchanged between the two clients and that CIUS is able to handle the multiple sessions.	Client Services Framework->Cisco Unified Presence->WAN->Cisco Unified Presence->Cisco Unified Personal Communicator 7	Passed	
UC861IF.CUS.503	Inter-cluster Instant Messaging Interoperability with Cisco Unified Personal Communicator 8 user sending offline messages	Verify the ability to begin an Instant Messaging chat with a Cisco Unified Personal Communicator 8 user that is initially not logged in (presence status is unavailable). Verify when the user logs in offline messages are received, and exchanges Instant Messages between the two clients. Verify interoperability.	Client Service Framework->Cisco Unified Presence->WAN->Cisco Unified Presence->Cisco Unified Personal Communicator 8	Passed	
UC861IF.CUS.504	Cisco CIUS client receiving Instant Messages from Cisco CIUS in a different time zone	Verify the ability to send Instant Messages to a Ciasco CIUS user from a user in another time zone. Verify Instant Messages are displayed with correct time stamps adjusted properly for the current time zone.	Cisco Unified Personal Communicator->Cisco Unified Presence->Wide Area Network->Cisco Unified Presence->Cisco CIUS	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.507	Cisco Unified Presence Service Outage	Verify that Cisco CIUS is able to recover on failing the Cisco Unified Presence services and Cisco Unified Presence network connectivity while the Cisco CIUS client logs in.	Cisco CIUS->LAN->Cisco Unified Presence	Passed	
UC861IF.CUS.508	Auto on the Phone Presence with Shared Line	Verify Cisco CIUS auto switches its self-presence to "on the phone" when a shared line phone goes off-hook.	Cisco CIUS->Unified CM->Phone	Passed	
UC861IF.CUS.509	Add Instant Messaging Session to Active Inter-cluster Phone Call	Verify the ability to add an Instant Messaging session to the phone call during an active call session with a user in another cluster.	Cisco Cius->Unified CM1->SIP trunk->Unified CM2->User2	Passed	
UC861IF.CUS.510	Auto in a Meeting Presence	Verify Cisco Unified Presence is configured to use calendar integration. Verify when a Cisco CIUS user joins a meeting listed on their calendar that the Cisco CIUS client's presence is changed to "In a Meeting".	Cisco CIUS->Cisco Unified Presence -> MicroSoft Exchange	Passed	
UC861IF.CUS.601	Call preservation when primary Cisco Unified Communications Manager goes down, Cisco Cius registers with Clustering over WAN (CoW) Backup Node	Verify that the active call remains preserved and the Cisco Cius is successfully able to register to the secondary node when the primary Cisco Unified Communications Manager that Cisco Cius is registered to during an active call fails, and the secondary node that Cisco Cius registers to is located over the WAN.	Cisco Cius->Unified CM1->SIP Trunk->Unified CM2->Cisco Unified IP Phone 7975; After failover Cisco Cius->WAN->Adaptive Security Appliance->Backup Unified CM->SIP trunk->Unified CM2->Cisco Unified IP Phone 7975	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.602	Cisco Cius Operating in Cisco Survivable Remote Site Telephony Mode, Initiating Adhoc Conference	Verify Cisco Cius registers to the Cisco Survivable Remote Site Telephony router, places a call to the PSTN, to another phone within the branch site and conferences the two calls given that Cisco Cius in a branch site is registered to Cisco Unified Communications Manager, and the WAN link breaks and the site falls back to Cisco Survivable Remote Site Telephony mode.	Cisco Cius->Cisco Survivable Remote Site Telephony router->PSTN; Cisco Cius->Cisco Survivable Remote Site Telephony router->Phone2	Passed	
UC861IF.CUS.603	Cisco Cius calling over End Office SIP Trunks via Session Manager Edition, Consultative Transfer	Verify Cisco Cius registers to the Cisco Survivable Remote Site Telephony Router on placing a call to the PSTN and to another phone within the branch site and conferencing the two calls, given that Cisco Cius in a branch site is registered to Cisco Unified Communications Manager and the WAN link breaks and the site falls back to Cisco Survivable Remote Site Telephony mode.	Cisco Cius->Cisco Survivable Remote Site Telephony router->PSTN; Cisco Cius->Cisco Survivable Remote Site Telephony Router->Phone2	Passed	
UC861IF.CUS.604	Cisco Cius calling over H.323 trunks via Cisco Unified Communications Session Manager Edition (G.722), call transferred to Cisco Survivable Remote Site Telephony site (G.729), Hold/Resume	Verify audio codec renegotiation when Cisco Cius initially calls a phone in another cluster via Cisco Unified Communications Session Management Edition H.323 trunks, given that the audio codec negotiated is G.722 and the far side transfers the call to a branch site using G.729.	Cisco Cius->Unified CM1->H.323 trunk->SME->H.323 trunk->Unified CM2->Unified IP Phones 89XX/99XX; Transfer->TNP phone	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.605	Cisco CIUS call over SIP Gateway, far side rolls over to Unity Connection and leave a voicemaill	Verify the ability to place a call over SIP Gateway to another enterprise. Verify if the call rolls over to Unity Connection on the far side, and Cisco CIUS leaves a voicemail.	Cisco CIUS->Unified CM1->SIP Trunk->Cisco IME inline ASA->Cisco IME offpath ASA->SIP Trunk->Unified CM2->Phone; Transfer->SIP Trunk->Unity Connection	Passed	
UC861IF.CUS.606	Cisco CIUS attending webex conference meeting	Verify that Cisco CIUS is able to attend webex conference meeting and transfers the call to mobile phone, and continues the call from mobile phone.	Cisco CIUS->Unified CM1->SIP Trunk->Cisco IME inline ASA->Cisco IME offpath ASA->SIP Trunk->Unified CM2->Phone; after fallback; Cisco CIUS->Unified CM1->SIP Trunk->PSTN Gateway->PSTN->PSTN Gateway->Unified CM2->Phone	Passed	
UC861IF.CUS.607	Cisco Cius as a local RSVP-enabled endpoint over RSVP enabled SIP Inter Cluster Trunk	Verify local RSVP is invoked and media terminates from Cisco Cius to the RSVP agent on placing a call from CIUS over a SIP trunk requiring RSVP reservations.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.608	Cisco Cius as an End-to-End RSVP-enabled Endpoint, Direct Transfer to Call on Hold	Verify the ability to place a call from Cisco Cius over a SIP trunk with End-to-End RSVP reservations and have another incoming call come to Cisco Cius and to place the call from Cisco Cius on hold and answer the other call. Verify the ability to resume the other call, and then perform a direct transfer to connect call A to call B.	Cisco Cius->Unified CM->SIP trunk->Unified CM->Phone; media flows through RSVP agents	Passed	
UC861IF.CUS.609	Cisco Cius placing 911 call via Cisco Emergency Responder in Wireless Mobility Mode	Verify a 911 call placed using Cisco Cius is routed to the correct Public Safety Answering Point (PSAP), and the PSAP call-back routes the call back to Cisco Cius.	Cisco Cius->Unified CM->Java Telephony Application Programming Interface (JTAPI)->Cisco Emergency Responder; Cisco Cius->Unified CM->PSTN Gateway->PSTN->Public Safety Answering Point	Passed	
UC861IF.CUS.610	Cisco CIUS placing SAF Call (G.711), far side transfers to Unified Survivable RemoteSite Telephony Site (G.729)	Verify the ability to use Cisco CIUS to place a 911 call. Verify the call is routed to the correct PSAP. Verify that PSAP call-back sends the call back to CIUS.	Cisco CIUS->Unified CM->SIP SAF Trunk->Unified CM->Phone; Transfer->Unified SRST phone	Passed	
UC861IF.CUS.611	Cisco Cius Placing SAF Call, IP Call Fails and PSTN Fallback is Used	Verify the ability to place a SAF call with Cisco Cius when the IP call fails and SAF PSTN fallback is invoked. Verify Cisco Cius handles the PSTN fallback properly.	Cisco Cius->Unified CM->Cisco IOS PSTN Gateway->PSTN->PSTN Gateway->Unified CM->Phone	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.612	Call to Analog Phone behind VG.224 shared line device barges in to call	Verify the ability to place a call from Cisco CIUS to an analog phone in another cluster (via SIP trunk) behind a VG224. Cisco CIUS is sharing a line with an Unified IP Phones 89XX/99XX. Verify the ability of the Unified IP Phones 89XX/99XX to barge into the call. Verify a successful 3-way call.	Cisco CIUS->Unified CM->SIP Trunk->Unified CM->VG224> Analog phone;After barge->Cisco CIUS->Unified CM->Unified 99xx IP phone built in bridge	Passed	
UC861IF.CUS.613	Cisco Cius Call Over H.323 Fast Start Inter Cluster Trunk with Trusted Relay Point	Verify that the call sets up and media terminates on the Trusted Relay Point and the call is put on Hold/resume on placing a call from Cisco Cius to a Unified IP Phones 89XX/99XX in another cluster via an H.323 fast start Inter Cluster Trunk, given that the Cisco Cius device has "Use Trusted Relay Point" enabled.	Cisco Cius->Unified CM->H.323 Fast Start->Unified CM->Unified IP Phones 89XX/99XX	Passed	
UC861IF.CUS.614	Cisco Cius call to IPv4 Endpoint, Far Side Transfers to IPv6 Endpoint with Cisco IOS Media Termination Point (MTP) Inserted	Verify that media remains intact between Cisco Cius, the Media Termination Point, and the IPv6 endpoint on placing an audio call from a Cisco Unified Personal Communicator 8 softphone in a different cluster to Cisco Cius via Cisco Unified Communications Session Management Edition, when the Cisco Unified Personal Communicator transfers the call to an IPv6-only device in its same cluster (an MTP should be invoked).	Cisco Cius->Unified CM->SIP Trunk->SME->SIP Trunk->Cisco Unified Personal Communicator8; Transfer->IPv6 phone	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.615	Cisco Cius Conference into an Active Shared Line Call	Verify a conference bridge is invoked and three-way communication is successful on placing a call from a phone over the PSTN to an Unified IP Phones 89XX/99XX sharing a line with Cisco Cius given the conference in the Cisco Cius phone is using its primary line.	Cisco Cius->Unified CM->IOS Conference bridge	Passed	
UC861IF.CUS.617	Call Hold/Resume in Audio Call with Cisco Cius via SIP Delay Offer trunk	Verify the ability to Hold/Resume call in audio call with Cisco Cius via SIP delay offer trunk.	Cisco Cius->Unified CM1->SIP DO trunk->Unified CM2->Unified 79XX IP Phone	Passed	
UC861IF.CUS.618	Call Hold/Resume in Audio Call with Cisco Cius via MGCP Gateway	Verify the ability Hold/Resume call in audio call with Cisco Cius via MGCP Gateway	Cisco Cius->Unified CM1->MGCP Gateway trunk->Unified CM2->Unified 89XX IP Phone	Passed	
UC861IF.CUS.619	Call from Cisco Unified 89XX SIP phone to phone with Call Forward All to Cisco CIUS via SME SIP trunks using Early Offer	Verify that the call from Cisco Unified 89XX SIP phone to phone with Call Forward All to Cisco CIUS via SME SIP trunks uses Early Offer.	Unified 89XX Phone->Unified CM1->Phone CFWD ALL->SIP EO trunk->SME->SIP EO trunk->Unified CM2->Cisco CIUS	Passed	
UC861IF.CUS.620	Call from Cisco Unified 79XX SCCP phone via H.323 trunk to phone with Call Forward All to Cisco Cius	Verify the ability to call from Unified 79XX SCCP phone via H.323 trunk to phone with Call Forward All to Cisco Cius.	Unified 79XX Phone->Unified CM1->H.323 trunk->Unified CM2->Phone Call Forward All->Cisco Cius	Passed	
UC861IF.CUS.623	Call Transfer to Cisco Cius from Unified 89XX SIP Phone via SIP trunk using Delay Offer	Verify the call transfer to Cisco Cius from Unified 89XX SIP phone via SIP trunk using delay offer.	Audio IP Phone->Unified CM1->Unified 89XX IP Phone Call Transfer->SIP DO trunk->Unified CM2->Cisco Cius	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.624	Call to Unified 79XX SCCP Phone via H.323 Gateway Call Transfer to Cisco Cius	Verify the call to Unified 79XX SCCP phone via H.323 Gateway with call transferred to Cisco Cius.	Audio IP Phone->Unified CM1->H.323 Gateway->Unified CM2->Unified 79XX Phone Call Transfer->Cisco Cius	Passed	
UC861IF.CUS.626	Call to Cisco Cius via SIP Gateway with call transferred from Cisco Cius to Unified 69XX SIP Phone	Verify call to Cisco Cius via SIP Gateway with call transferred from Cisco Cius to Unified 69XX SIP phone.	Audio IP Phone->Unified CM1->SIP Gateway->Unified CM2->Cisco Cius Call Transfer->Unified 69XX Phone	Passed	
UC861IF.CUS.627	Cisco CIUS activates Do Not Disturb and calls from PSTN are rejected and CIUS does not ring	verify if CIUS activates Do Not Disturb and calls from PSTN are rejected and Cisco CIUS does not ring.	Unified 79XX Phone->Unified CM1->SIP Gateway->Unified CM2->Cisco CIUS	Passed	
UC861IF.CUS.630	Call to Cisco Unity Connection via PSTN trunk with Cisco Unity Connection Transferring Call to Cisco Cius	Verify call to Cisco Unity Connection via PSTN trunk with Cisco Unity Connection transferring call to Cisco Cius.	Audio Phone->Unified CM1->SIP Gateway->Unified CM2->Cisco Unity Connection Call Transfer->Cisco Cius	Passed	
UC861IF.CUS.631	Cisco Cius placing 911 call via Cisco Emergency Responder in Docked Mode	Verify the call is routed to the correct PSAP and the PSAP call-back sends the call back to Cisco Cius when Cisco Cius is used to place a 911 call.	Cisco Cius->Unified CM->JTAPI->Cisco Emergency Responder; Cisco Cius->Unified CM->PSTN Gateway->PSTN->PSAP	Passed	
UC861IF.CUS.801	Layer 3 Roaming with Cisco Cius	Verify the ability to perform a Layer 3 roaming while Cisco Cius is in a call.	Cisco Cius -->Cisco Lightweight Access Point 1 -->Wireless LAN Control1 -->Wireless LAN Control2 -->Cisco Lightweight Access Point 2 -->Unified CM	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.802	Layer 2 Roaming with Cisco CIUS	Verify the ability to perform layer 2 roaming while Cisco CIUS is in a call.	Cisco CIUS-->LAP1-->WLC1-->LAP2-->Unified CM	Passed	
UC861IF.CUS.803	Call Transfer Over a Secure SIP Trunk via Cisco Unified Session Management Edition	Verify the ability to perform a call transfer over a secure SIP trunk involving Cisco Unified Session Management Edition.	Cisco Cius -->Unified CM1 -->Secure SIP Trunk -->SME -->Secure SIP Trunk -->Unified CM2 -->Cisco IP Phone1; Cisco Cius -->Transfer -->Cisco IP Phone 2	Passed	
UC861IF.CUS.804	Voicemail server in Unity Connection in Cisco Session Manager Edition Cluster	Verify that Cisco CIUS can leave voicemails in Cisco Unity connection in Cisco Session Manager Edition and read the VoiceMails.	Cisco CIUS-->Unified CM1-->Secure SIP Trunk-->Cisco SME-->Secure SIP Trunk-->Unified CM2-->IP Phone1->Secure SIP Trunk-->Unified CM-->SCCP-->Unity Connectionn	Passed	
UC861IF.CUS.805	Handoff Cisco CIUS call to Mobile phone	Verify that call from Cisco CIUS can be handed over to a remote destination across SIP trunk.	Cisco CIUS-->Unified CM1-->SIP Trunk-->Unified CM2-->IP Phone1	Failed	CSCto97665
UC861IF.CUS.955	Cisco Cius device controlled from a Cisco Unified Personal Communicator in deskphone mode accessed through the Virtual Desktop Infrastructure (VDI) application in Cisco Cius	Verify that Cisco Cius can be controlled from a Cisco Unified Personal Communicator in deskphone mode to place, receive a call and to perform other call features.	Cisco Cius->Unified CM1->SIP Trunk -->Unified CM2 -->Cisco IP Phone	Passed	

Cisco Emergency Responder

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CER.001	Make sure Phone in "Power Save Plus" mode is still tracked by Cisco Emergency Responder	Verify that Cisco Emergency Responder continues to track the location of phone in "Power Save Plus" mode.	Phone->Switch->Cisco Emergency Responder	Passed	
UC861IF.CER.002.1	Unified IP Phones 99XX series Out of Power Save Plus Mode Make 911 Calls that is Routed to nearest PSAP	Verify the ability to make sure Unified IP Phone 99XX series coming out of Power Save Plus mode can make 911 calls and the call is routed to nearest PSAP.	Phone->Switch->Unified CM->Cisco Emergency Responder->Unified CM->Gateway->PSAP	Passed	
UC861IF.CER.002.2	Unified 69XX Series IP Phone Out of Power Save Plus Mode Make 911 calls that is routed to nearest PSAP	Verify the ability to ensure that Unified 69XX series IP phone coming out of Power Save Plus mode can make 911 calls and the call is routed to nearest PSAP.	Phone->Switch->Unified CM->Cisco Emergency Responder->Unified CM->Gateway->PSAP	Passed	
UC861IF.CER.002.3	Cisco Unified 79xx series IP Phone Out of Power Save Plus Mode Make 911 calls that is routed to nearest PSAP	Verify that 79XX series phone coming out of Power Save Plus Mode can make 911 calls and the call is routed to nearest PSAP.	Phone->Switch->Unified CM->Cisco Emergency Responder->Unified CM->Gateway->PSAP	Passed	
UC861IF.CER.003.1	Cisco Unified 99XX IP Phone Series in Power Save Plus Mode moved to another switch in the same Unified CM cluster Makes 911 call after waking up from Power Save Plus Mode	Verify that Unified 99XX series IP Phone in power save plus mode moved to another switch in the same Cisco Unified Communications Manager cluster can make 911 call after waking up from power save plus mode.	Phone->Switch->Unified CM->Cisco Emergency Responder->Unified CM->Gateway->PSAP	Passed	
UC861IF.CER.003.2	Cisco Unified 69XX Series IP phone in Power Save Plus Mode moved to another switch in the same Cisco Unified Communications Manager cluster can make 911 call after waking up from Power Save Plus Mode	Verify that Unified 69XX series IP phone in Power Save Plus mode moved to another switch in the same Cisco Unified Communications Manager cluster can make 911 call after waking up from Power Save Plus mode.	Phone->Switch->Unified CM->Cisco Emergency Responder->Unified CM->Gateway->PSAP	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CER.003.3	Cisco Unified 79XX series IP Phone in Power Save Plus Mode Moved to Another Switch in the Same Cisco Unified Communications Manager Cluster Make 911 Call after waking up from Power Save Plus Mode	Verify that Cisco Unified 79XX series IP Phone in Power Save Plus mode moved to another switch in the same Cisco Unified Communications Manager cluster can make 911 call after waking up from Power Save Plus Mode.	Phone->Switch->Unified CM->Cisco Emergency Responder->Unified CM->Gateway->PSAP	Passed	
UC861IF.CER.004.1	Cisco Unified 99XX Series IP Phone in Power Save Plus Mode Moved to Another Switch in the Different Cisco Unified Communications Manager Cluster make 911 Call after Waking up from Power Save Plus Mode.	Verify that Cisco Unified 99XX series IP phone in Power Save Plus mode moved to another switch in a different Cisco Unified Communications Manager cluster can make 911 call after waking up from Power Save Plus mode.	Phone->Switch->Unified CM->Cisco Emergency Responder->Unified CM->Gateway->PSAP	Passed	
UC861IF.CER.004.2	Cisco Unified 69XX series IP Phone in Power Save Plus mode moved to another switch in the different Cisco Unified Communications Manager cluster can make 911 call after waking up from Power Save Plus mode	Verify that Cisco Unified 69XX series IP phone in Power Save Plus mode moved to another switch in the different Cisco Unified Communications Manager cluster can make 911 call after waking up from Power Save Plus mode.	Phone->Switch->Unified CM->Cisco Emergency Responder->Unified CM->Gateway->PSAP	Passed	
UC861IF.CER.004.3	Cisco Unified 79XX series IP Phone in Power Save Plus Mode Moved to Another Switch in the different Cisco Unified Communications Manager cluster make 911 Call after Waking up from Power Save Plus Mode	Verify that Cisco Unified 79XX series IP Phone in Power Save Plus mode moved to another switch in the different Cisco Unified Communications Manager cluster can make 911 call after waking up from Power Save Plus Mode.	Phone->Switch->Unified CM->Cisco Emergency Responder->Unified CM->Gateway->PSAP	Passed	

Miscellaneous

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.OTH.001	Voicemail Deposit and Retrieval for a Connection User whose Directory Number is registered as a E.164 Number	Verify voicemail deposit and retrieval for a Connection user whose directory number is registered as a E.164 number in Cisco Unity Connection 8.6.	PSTN Phone->PSTN Gateway->Unified CM->IP Phone->Call Forward No Answer->Cisco Unity Connection; IP Phone->Unified CM->Cisco Unity Connection	Passed	
UC861EF.OTH.002	Visual Voicemail Feature for a Connection User whose Directory Number is Registered as a E.164 Number	Verify visual voicemail feature for a Connection user whose directory number is registered as a E.164 number in Cisco Unity Connection 8.6.		Passed	
UC861EF.OTH.003	Allow Outside Callers to Mark Messages Private	Verify the ability to allow outside callers to mark messages private in Cisco Unity Connection 8.6.	PSTN Phone->PSTN Gateway->Unified CM->Cisco Unity Connection->IP Phone	Passed	
UC861EF.OTH.004	Allow Users to Strip the Introduction from a Message Prior to Forwarding	Verify the ability to allow users to strip the introduction from a message prior to forwarding in Cisco Unity Connection 8.6.	IP Phone->Unified CM->Cisco Unity Connection	Passed	
UC861EF.OTH.005	Do not Prompt Users to Record an Introduction	Verify users are not prompted to record an introduction in Unity Connection 8.6.	IP Phone->Unified CM->Cisco Unity Connection	Passed	
UC861EF.OTH.006	Warn users on Reply All that their Message is Going to be Greater than X recipients	Verify the ability to warn users on Reply All that their message is going to greater than X recipients in Unity Connection 8.6.		Passed	
UC861EF.OTH.007	Transfer to E.164 Numbers with Call Handlers	Verify transfer to E.164 numbers with call handlers.	Cisco IP Phone->Unified CM->Cisco Unity Connection->Call Handler->Transfer-> Cisco IP Phone with E.164 number	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.OTH.008	Transfer to E.164 numbers using Interview handlers	Verify Transfer to E.164 numbers using Interview handlers.	Cisco IP Phone->Unified CM->Cisco Unity Connection->Interview Handler->Transfer-> Cisco IP Phone with E.164 number	Passed	
UC861EF.OTH.009	Transfer to E.164 numbers with directory handlers	Verify Transfer to E.164 numbers with directory handlers.	Cisco IP Phone->Unified CM->Cisco Unity Connection->Directory Handler->Transfer-> Cisco IP Phone with E.164 number	Passed	
UC861EF.OTH.010	Personal Call Transfer Rules Based Transfers to E.164 Numbers	Verify personal call transfer rules based transfers to E.164 numbers.	Cisco IP Phone->Unified CM->Cisco Unity Connection->Cisco IP Phone->Personal Call Rule Transfer->Cisco IP Phone with E.164 number	Passed	
UC861EF.OTH.011	Alternate Extensions and Restriction Pattern Support for E.164 Numbers	Verify alternate extensions and restriction pattern support for E.164 numbers.	Cisco IP Phone->Unified CM->Cisco Unity Connection	Passed	
UC861EF.OTH.012	Notification devices with E.164 number support	Verify Notification devices with E.164 number support.	Cisco IP Phone->Unified CM->Cisco Unity Connection->Notification->Cisco IP Phone with E.164 number	Passed	
UC861EF.OTH.101	Power Save Mode with EnergyWise Domain Override Disallowed in Cisco Unified 99xx IP Phones and Unified IP Phones 6901/11/21/41/45/61	Verify transition from power save mode to normal mode by user and effect of EnergyWise override that is disallowed in Cisco Unified 99xx IP Phones and Unified IP Phones 6901/11/21/41/45/61.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.OTH.102	Power Save Mode with EnergyWise Domain Overrides Allowed on Unified 99xx IP Phones and Unified IP Phones 6901/11/21/41/45/61 phones	Verify transition from power save mode to normal mode by user and effect of EnergyWise override when they are allowed in Unified 99xx IP Phones and Unified IP Phones 6901/11/21/41/45/61 phones.		Passed	
UC861EF.OTH.103	Powersave Plus Mode with EnergyWise Overrides Disallowed in Unified 99xx IP Phones and Unified IP Phones 6901/11/21/41/45/61 Phones	Verify transition from power save mode to normal mode by user and effect of EnergyWise override when they are not allowed in Unified 99xx IP Phones and Unified IP Phones 6901/11/21/41/45/61 Phones.		Passed	
UC861EF.OTH.104	Powersave plus mode with EnergyWise Overrides Allowed in Unified 99xx IP Phones and Unified IP Phones 6901/11/21/41/45/61 phones	Check transition from power save mode to normal mode by user and effect of EnergyWise override when they are allowed in Unified 99xx IP Phones and Unified IP Phones 6901/11/21/41/45/61 phones.		Passed	
UC861EF.OTH.105	Effect of Cisco EnergyWise when Unified IP Phones 89XX/99XX or Unified IP Phones 6901/11/21/41/45/61 remote is in use	Verify the effect of Cisco EnergyWise on Unified IP Phones 89XX/99XX or Unified IP Phones 6901/11/21/41/45/61 remote that are in use.		Passed	
UC861EF.OTH.106	Alarms, Messages in Unified Communications Manager	Verify alarms and messages in Cisco Unified Communications Manager.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.OTH.107	Effect of EnergyWise on Computer Telephony Integration (CTI) Controlled Phones	Verify the effect of EnergyWise power save and power save plus mode on Computer Telephony Integration controlled phones.		Passed w/ Exception	In an EnergyWise case, there is no visible change to the phone status on the client in the laptop when a phone is not available. Exploring how this could be made more intuitive and why this choice was made.
UC861EF.OTH.108	Cisco 7970 IP Phone in Power Save Plus Mode	Verify the ability to check power save plus mode in Cisco 7970 IP phones.		Passed	
UC861EF.OTH.109	Effect of FirmWare Upgrades on Cisco Energywise Requests	Verify the behavior of phones when firmware upgrade is scheduled when there is a power off request from Cisco Energywise.		Passed	
UC861EF.OTH.110	Phone firmware changed when phone powered off for Cisco Energywise	Verify the phone firmware download when firmware is changed and the phone is powered off by Cisco Energywise.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.OTH.111	Effect of EnergyWise Requests on Virtual Desktop Integration (VDI)/ Virtualization Experience Client (VXC) Backpack and Standalone Independent Computing Architecture/PC Over IP.	Verify the effect of EnergyWise Power Save Plus Mode and EnergyWise Domain Override on Unified IP Phones 89XX/99XX used as Virtual Desktop Integration (VDI)/ Virtualization Experience Client (VXC) on backpack Independent Computing Architecture (ICA)/PC Over IP and effect of EnergyWise on Virtual Desktop Integration (VDI)/ Virtualization Experience Client (VXC) Standalone ICA/PC Over IP.		Passed	
UC861IF.OTH.001	Unified MeetingPlace: Start Multiple Meetings in a Site with Multiple Nodes and Ensure Each Node in the Site gets Used	Verifies each node in a site with multiple nodes gets used when multiple meetings are started in the site.	Endpoint->Unified CM->SME->SIP Trunk->Cisco Unified Meeting Place	Passed	
UC861IF.OTH.002	Unified MeetingPlace: Site Selection Based on Preferred Site Field in User Profile	Verifies site selected via preferred site field in user profile.	SME site - Endpoint->Unified CM->SME ->SIP Trunk->Cisco Unified MeetingPlace	Passed	
UC861IF.OTH.003	Unified MeetingPlace: Site Selection Based on User Profile Time-zone Setting	Verifies site selected via time-zone setting in user profile.	Endpoint->Unified CM->SIP Trunk->Cisco MeetingPlace	Passed	
UC861IF.OTH.004	Unified MeetingPlace: Site Selection based on Default Site when User is not Associated with a Site	Verifies site selected via system default site when user is not associated with a site.	Endpoint->Unified CM->SME->SIP Trunk->Cisco MeetingPlace	Passed	
UC861IF.OTH.005	Unified MeetingPlace: Host Meetings on the Single Active Node of a Multinode Site	Verifies if one node of a two node site hosts meetings when the other node in the site is down.	Endpoint->Unified CM->SME->SIP Trunk->Cisco MeetingPlace	Passed	
UC861IF.OTH.006	Unified MeetingPlace: Host Meetings on an Alternate Site in the same Region with Multiple Sites Available	Verifies meetings start on an alternate site in the same region when all nodes in a site are down.	Endpoint->Unified CM->SIP Trunk->Cisco MeetingPlace	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.OTH.007	Unified MeetingPlace: Host Meetings on an Alternate Site in a Different Region with Multiple Sites Available	Verifies meetings can start on an alternate site in a different region with multiple sites available.	Endpoint->Unified CM->SIP Trunk->Cisco MeetingPlace	Passed	
UC861IF.OTH.008	Unified MeetingPlace: Meeting Restarts on a Two-node Site when Participants Dial Back	Verifies if meeting restarts on a two-node site after one node goes down, when participants dial back into the same site.	Endpoint->Unified CM->SME->SIP Trunk->Cisco MeetingPlace	Passed	
UC861IF.OTH.009	Node goes Down During Meeting, Meeting Restarts on Different Site in the Same Region in Unified MeetingPlace	Verifies that when all nodes go down in a site, participants dial back in and meeting is on node in other site of the same region.		Passed	
UC861IF.OTH.010	Node Goes Down During Meeting and Restarts on Different Site in a Different Region in Unified MeetingPlace	Verifies that when all nodes go down in a site, participants dial back in and meeting is on node in other site of a different region.		Passed	
UC861IF.OTH.011	Unified MeetingPlace: Call into SME MeetingPlace from multiple clusters	Verify that Unified MeetingPlace node in SME site can be accessed via SIP and H.323 Inter Cluster Trunks from multiple Cisco Unified Communications Manager clusters.		Passed	
UC861IF.OTH.012	Unified MeetingPlace: Call into Cisco Unified MeetingPlace Hardware Media Server (HMS) via SME	Verify that Cisco MeetingPlace Hardware Media Server (HMS) node in a site can be accessed via SME tandem Unified Communications Manager cluster.		Passed	
UC861IF.OTH.013	Unified MeetingPlace: Call into MeetingPlace - Enhanced Media Server via SME	Verify MeetingPlace - Enhanced Media Server node in a site can be accessed via SME tandem Unified Communications Manager cluster.		Passed	
UC861IF.OTH.014	Unified MeetingPlace: Outdial from SME MeetingPlace to multiple clusters	Verify outdial from a meeting on a MeetingPlace node in SME site to multiple Unified CM clusters.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.OTH.015	Dial into Cisco Unified MeetingPlace in Session Manager Edition site using dial pattern so call uses Media Termination Point (MTP) resources from one site to the other	Verify that Media Termination Point (MTP) can be used when dialing into a meeting in a Session Manager Edition tandem Cisco Unified Communications Manager with Cisco Unified MeetingPlace.		Passed	
UC861IF.OTH.016	Dial into Cisco Unified MeetingPlace in Session Manager Edition site using dial pattern so call uses Trusted Relay Point (TRP) resources from one site to the other in Unified MeetingPlace	Verify if Trusted Relay Point (TRP) can be used when dialing into a meeting in a Session Manager Edition tandem Cisco Unified Communications Manager with Cisco Unified MeetingPlace.		Passed	
UC861IF.OTH.030	Cisco UC Integration(TM) for Microsoft Lync joining WebEx/Cisco Unified MeetingPlace based meeting	Verify that Cisco UC Integration(TM) for Microsoft Lync can dial into WebEx/Cisco Unified MeetingPlace based meeting and also the reverse way, WebEx/Cisco Unified MeetingPlace calling Cisco UC Integration(TM) for Microsoft Lync.	Cisco UC Integration(TM) for Microsoft Lync -->Unified CM--->Session Manager Edition---Cisco Unified MeetingPlace--->Web Ex	Passed w/ Exception	CSCto50 486
UC861IF.OTH.032	Cisco UC Integration(TM) for Microsoft Lync setting up Video Conference between Unified IP Phone 9971 and Cisco Unified IP Phone 8945 Phones	Verify that Cisco UC Integration(TM) for Microsoft Lync set up a video conference between Unified IP Phone 9971 and Cisco Unified IP Phone 8945 video capable phones.	Cisco UC Integration(TM) for Microsoft Lync-->Unified CM + Cisco Codian---->Unified IP Phone 9971+ Unified IP Phone 8945	Passed	
UC861IF.OTH.033	Cisco UC Integration(TM) for Microsoft Lync in Secure Mode Getting Secure Voicemail	Verify that Cisco UC Integration(TM) for Microsoft Lync in secure mode get Visual VoiceMail indication and can call the VoiceMail server and read the secure VoiceMail.	Cisco UC Integration(TM) for Microsoft Lync-->Unified CM--->Cisco Unity Connection	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.OTH.034	Cisco UC Integration(TM) for Microsoft Lync Coming up in SRST Mode	Verify that Cisco UC Integration(TM) for Microsoft Lync can automatically come up in Cisco Survivable Remote Site Telephony mode when WAN connectivity is broken during the call.	Cisco UC Integration(TM) for Microsoft Lync-->Cisco Survivable Remote Site Telephony--->Unified CM	Passed	
UC861IF.OTH.035	Cisco Unity Express Single Inbox Receiving New Emails and Marking it read through Outlook	Verify that voicemails can be received from Cisco Unity Express to Outlook, and the voicemails can be marked read from Outlook configured to synchronize with Exchange.	Cisco Unity Express -->Exchange 2007 -->Outlook	Passed	
UC861IF.OTH.036	Voicemails Marked Urgent in Cisco Unity Express Single Inbox	Verify that when messages marked urgent in Cisco Unity Express is received with email importance set to high.	Cisco Unity Express -->Exchange 2007 -->Outlook	Passed	
UC861IF.OTH.037	Marking Read Messages Unread in Microsoft Outlook/Microsoft Exchange in Cisco Unity Express Single Inbox	Verify that when read emails/voicemails are marked unread or new in Microsoft outlook, Cisco Unity Express marks that voicemail as new as well and turns on the Message Waiting Indication (MWI).	Cisco Unity Express -->Microsoft Exchange 2007 -->Microsoft Outlook	Passed	
UC861IF.OTH.074	CSF client (Cisco Unified Communications Integration(TM) for Microsoft Lync) can have a voicemail box in Cisco Unity Express	Verify a Cisco Unity Express can provide voicemail service to Cisco Unified Communications Integration(TM) for Microsoft Lync	Phone1 -->Unified CM -->SIP Trunk -->Unified CM -->UC Integration(TM) for Microsoft Lync -->Call Forward No Answer --> Unified CM -->Java Telephony Application Programming Interface -->Cisco Unity Express	Passed	
UC861IF.OTH.101	Dual Tone Multi-frequency (DTMF) Interoperability of Cisco Unified IP Phone 894x with Cisco Unity Connection	Verify that DTMF works fine on Cisco Unified IP Phone 894X with Cisco Unity Connection when the call is placed from a remote cluster over SIP trunk.	Unified IP Phone 894X -->Unified CM -->SIP Trunk -->Unified CM -->Unity Connection	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.OTH.103	Transcoder can be Invoked Dynamically for a Call Involving Cisco Unified IP Phone 894X	Verify that Unified Communications Manager can invoke a transcoder dynamically when there is a Codec mismatch for a call involving Cisco Unified IP Phone 894X.	IP Phone -->Unified CME -->SIP Trunk -->Unified CM -->Transcoder -->Unified IP Phone 894X	Passed	
UC861IF.OTH.104	Cisco IPv6 call from a Dual Stack Unified IP Phone and Call Out on Hold	Verify that a Dual Stack Unified IP phone can be used to place a call with media as Cisco IPv6. Verify that the call can be placed on hold by invoking the Hold key on the Unified IP Phones.	IP Phone Dual Stack -->Unified CM -->SIP Trunk Dual Stack -->Unified CM -->Unified IP Phones; Hold invoked on Unified IP Phones; IP Phone -->Unified CM -->SIP Trunk -->Unified CM -->Music on Hold (MoH)	Passed	
UC861IF.OTH.105	Point-to-Point Call Over a Dual Stack Cisco IPv6 SIP trunk involving Cisco IPv6 only Unified IP Phones.	Verify that Unified IP Phones when configured in Cisco IPv6 only mode can be used in calls over the SIP trunks.	IP Phone Dual Stack -->Unified CM -->SIP Trunk Dual Stack -->Unified CM -->Unified IP Phones only	Passed	
UC861IF.OTH.106	Cisco IPv6 Call from an IPv6 only Unified IP Phones and Call Out on Hold	Verify that a Cisco IPv6 only Unified IP Phone can be used to place a call with media as IPv6. Verify that the call can be placed on hold by invoking the Hold key on the Unified IP Phones.	IP Phone Dual Stack -->Unified CM -->SIP Trunk Dual Stack -->Unified CM -->Unified IP Phones IPv6 only; Hold invoked on Unified IP Phones; IP Phone -->Unified CM -->SIP Trunk -->Unified CM -->Music on Hold	Passed	
UC861IF.OTH.110	Conference Call Using a Cisco Internet Protocol Version 6 (IPv6)-Only 6900 Series Unified IP Phone	Verify that an IPv6 only 6900 Series Unified IP phone can be used to place a conference call, when the IPv6 transcoder will be invoked for the IPv6 only phone.	IP Phone DS -->Unified CM -->SIP Trunk DS -->Unified CM -->Unified IP Phone 6900 Series IPv6 only; Hold invoked on Unified IP Phone 6900 series; IP Phone -->Unified CM -->SIP Trunk -->Unified CM -->Music on Hold	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.OTH.111	Call from IPV4 phone to IPV6 phone over SME connected with Alternative Network Address Types (ANAT) enabled SIP Trunk	Verify that call can be established from an IPv4 phone in one cluster to a dual stack phone in another cluster through SME cluster connected with Alternative Network Address Types (ANAT) enabled SIP trunk.	IP Phone V4 -->Unified CM1 -->SIP Trunk DS -->SME --<SIP Trunk>---Unified CM Unified CM2---->DS IP phone	Passed	
UC861IF.OTH.112	Call from IPV6 Cisco Unified IP Phone 6900 Series to IPV6 phone through IPv6 SIP Gateway connected with Alternative Network Address Types (ANAT) enabled SIP Trunk	Verify that call can be established from an IPv6 Unified IP Phone 6900 Series through an IPv6 SIP Gateway connected with Alternative Network Address Types enabled trunk.	IP Phone V4 -->Unified CM1 -->SIP Trunk DS -->SME --<SIP Trunk>---Unified CM Unified CM2---->DS IP phone	Passed	
UC861IF.OTH.121	SRSV: Provisioning when primary Cisco Unified Communications Manager server is down but secondary is up; Primary Cisco Unity Connection Server is up	Verify that Cisco Unified Messaging Gateway (UMG)-Cisco Survivable Remote Site Voicemail (SRSV) can synchronize with secondary Unified Communications Manager server when the primary server is down and the remote site SRSV-Cisco Unity Express provisioning can continue without any problems.	Unified CM Cisco Unity Connection -->Cisco Survivable Remote Site Voicemail-Cisco Unified Messaging Gateway -->Cisco Survivable Remote Site Voicemail-Cisco Unity Express	Passed	
UC861IF.OTH.122	Cisco Survivable Remote Site Voicemail (SRSV): Provisioning when Primary Unified Communications Manager server is down but secondary is up; Primary Cisco Unity Connection server is down but secondary Cisco Unity Connection server is up	Verify that Cisco Unified Meeting Gateway-Cisco Survivable Remote Site Voicemail (UMG-SRSV) can synchronize with secondary Unified Communications Manager server when the primary server is down. Verify that UMG-SRSV can synchronize with secondary Cisco Unity Connection server when the primary Cisco Unity Connection server is down, and also verify that the provisioning is successful under these conditions.	Unified CM Cisco Unity Connection -->Cisco Survivable Remote Site Voicemail-Cisco Unified Meeting Gateway -->Cisco Survivable Remote Site Voicemail-Cisco Unity Express	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.OTH.123	Cisco Survivable Remote Site Voicemail: Voicemail upload after WAN link restoration with primary Cisco Unity Connection server down and secondary Cisco Unity Conection server active	Verify that Cisco Unified Meeting Gateway-Cisco Survivable Remote Site Voicemail (UMG-SRSV) can upload voicemails from SRSV-Cisco Unity Express to Cisco Unity Connection after the WAN link is restored. Verify that the upload is successful even when the primary Cisco Unity Connection server is down.	Cisco Unity Connection <-->Cisco Survivable Remote Site Voicemail-Cisco Unified Meeting Gateway <-->Cisco Survivable Remote Site Voicemail-Cisco Unity Express	Failed	CSCtq49819
UC861IF.OTH.124	Cisco Survivable Remote Site Voicemail: Primary Cisco Unified Communications Manager server unavailable in the middle of manually initiated provisioning	Verify that provisioning can continue and complete successfully even when the primary Unified Communications Manager server goes offline while a manually initiated provisioning is in progress.	Unified CM Cisco Unity Connection <-->Cisco Survivable Remote Site Voicemail-Cisco Unified Meeting Gateway <-->Cisco Survivable Remote Site Voicemail-Cisco Unity Express	Passed	
UC861IF.OTH.125	Cisco Survivable Remote Site Voicemail: Primary Cisco Unity Connection server unavailable in the middle of manually initiated provisioning	Verify that provisioning can continue and complete successfully even when the primary Cisco Unity Connection server goes offline while a manually initiated provisioning is in progress.	Unified CM Cisco Unity Connection <-->Cisco Survivable Remote Site Voicemail-Cisco Unified Meeting Gateway <-->Cisco Survivable Remote Site Voicemail-Cisco Unity Express	Passed	
UC861IF.OTH.126	Cisco Survivable Remote Site Voicemail: Auto Attendant Dial by Extension when Caller is in a Custom Greeting Linked to Opening Greeting	Verify that Auto Attendant Dial by Extension is provisioned successfully based on the configuration in Cisco Unity Connection, and ensures the functionality works in Cisco Survivable Remote Site Voicemail-Cisco Unity Express.	Unified CM Cisco Unity Connection <-->Cisco Survivable Remote Site Voicemail- Unified Meeting Gateway <-->Survivable Remote Site Voicemail-Cisco Unity Express; Phone -->SRST -->Survivable Remote Site Voicemail-Cisco Unity Express -->Transfer -->Phone	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.OTH.128	Caller Input is Ignored when Additional Key Input is Configured in Cisco Survivable Remote Site Voicemail	Verify caller input is configured to transfer to a call handler which in turn is configured to send the call to a subscribers greeting in Cisco Survivable Remote Site Voicemail.	Unified CM Cisco Unity Connection <-->Cisco SRSV-Unified Messaging Gateway <-->Cisco SRSV-Cisco Unity Express Phone -->Cisco SRST -->Cisco SRSV-Cisco Unity Express -->Transfer -->Phone	Passed	
UC861IF.OTH.130	Cisco Survivable Remote Site Voicemail: Provisioning additional users in Cisco Survivable Remote Site Voicemail-Cisco Unity Express through Unified Messaging Gateway	Verify that Unified Messaging Gateway can automatically provision users in Cisco Survivable Remote Site Voicemail -Cisco Unity Express once users are added in Cisco Unity Connection.	Unified CM Cisco Unity Connection <-->Cisco Survivable Remote Site Voicemail-Unified Messaging Gateway <-->Cisco Survivable Remote Site Variable-Cisco Unity Express	Passed	
UC861IF.OTH.140	Cisco Unity Express: A secure Voice Mail is forwarded to SRST - Unified Express subscriber as Voice Profile for Internet Mail (VPIM) message and the Subscriber Downloads and Plays it.	Verify that a subscriber in Cisco Unity Express controlled by Cisco Unified Communications Manager in a SRST router can download and play a Voice Profile for Internet Mail secure Voicemail forwarded from Unity connection subscriber.	SRST Cisco Unity Express <-->SRST<-->Unified CM<--->Cisco Unity Connection	Passed	
UC861IF.OTH.141	Cisco Unity Express: A secure Voicemail is forwarded to Unity Express-Cisco Unified Communications Manager Express subscriber as Voice Profile for Internet Mail message and the subscriber downloads and plays it.	Verify that a subscriber in Cisco Unity Express controlled by Cisco Unified Communications Manager Express in a SRST router can download and play a Voice Profile for Internet Mail secure Voicemail forwarded from Unity connection subscriber.	Unified CM Cisco Unity Express <-->Cisco Unity Connection Unified CME<-->Cisco Unity Connection	Passed	
UC861IF.OTH.142	Cisco Unity Express: iPhone Mobility Client is Dialing into SRST-Cisco Unity Express and playing a Secure Voicemail	Verify that an iPhone mobility client subscriber in SRST-Cisco Unity Express can dial into Cisco Unity Express and plays secure Voicemails.	SRST Cisco Unity Express <-->SRST<-->Unified CM	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.OTH.143	Cisco Unity Express: iPhone mobility client login to Cisco Unity Express and sends a voicemail to a subscriber in Cisco Unity Connection	Verify that iPhone client can dial into SRST-Cisco Unity Express and sends a secure voicemail to a Cisco Unity Connection subscriber.	SRST Cisco Unity Express <-->SRST<-->Unified CM<--->Cisco Unity Connection	Passed	
UC861IF.OTH.174	Cisco Survivable Remote Site Voicemail-Cisco Unity Express gets updated as and when users are added and deleted in Cisco Unity Connection.	Verify that Cisco Unified Messaging Gateway updates Cisco Survivable Remote Site Voicemail-Cisco Unity Express whenever users are added or removed from Cisco Unity Connection.	Cisco Unity Connection -->Unified Messaging Gateway-->Cisco Survivable Remote Site Voicemail-Cisco Unity Express	Passed	
UC861IF.OTH.175	Supervised Transfer when Ports are Configured to Support Authentication and Encryption	Verify that Unity Connection ports can be configured for authentication and encryption, given that supervised transfer is possible when a call is placed from a secure endpoint.	Cisco IP Phone -->Unified CM -->SIP Trunk -->Unified CM -->SCCP -->Cisco Unity Connection -->Transfer -->Unified CM -->Cisco IP Phone	Passed	

Mobile Clients

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.MOB.001	Nokia Mobility Client Midcall Feature - Hold and Resume over SIP Trunk	Verify Hold and Resume feature of Nokia Mobility Client by holding an incoming call through SIP trunk and resuming it multiple times.	Phone1-Unified CM1----<Cisco IME Trunk>---Unified CM2----<802.11 wireless>--->Nokia Mobility Client	Passed	
UC861IF.MOB.002	Nokia Mobility Client Midcall Feature - An Incoming Call on Call Waiting is Sent to Secure Voice Mail	Verify call waiting and call forward, when a Nokia client in a call receives another call. Verify whether the waiting call is sent to a secure voicemail, the caller deposits a message and later the client retrieves the message.	Phone1-Unified CM1-Nokia Mobility Client -Cisco Unity Connection	Passed	
UC861IF.MOB.003	Nokia Mobility Client Midcall Feature - Conferencing a PSTN phone over SIP Gateway	Verify conferencing feature of Nokia Mobility Client by conferencing an iPhone client and PSTN phone.	Nokia Mobility Client(Dial Via Office call)-- Unified CM1--iPhone Client(dual mode) + SIP Gateway	Passed	
UC861IF.MOB.004	Nokia Mobility Client Midcall Feature - Park and Retrieve Calls from Nokia Client	Verify conference parking feature of Nokia mobility client when an intercluster call is parked at Nokia client and retrieved from Cisco Unified IP phone 894X series, and Unified IP phone 894X phone then parks that call and Nokia client retrieves the call.	Nokia Mobility Client(Unified CM - call park feature)	Passed	
UC861IF.MOB.005	An Instant Message from Client Services Framework (CSF) Client to Nokia Mobility Client is Escalated to a Voice Call	Verify the Presence Status on contacts in Nokia Mobility client and also the presence status update of Nokia client on other clients.	Nokia Mobility Client (Unified CM(-Cisco Unified Presence-Unified Personal Communicator	Passed	
UC861IF.MOB.006	Handoff to Mobile Network using Dial Via Office- Forward (DVO-F) and Dial Via Office- Reverse (DVO-R) methods	Verify handoff call to mobile network using Dial Via Office- Forward and Dial Via Office- Reverse.	Nokia Client->H.323 Gateway--->Unified CM--->iPhone	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.MOB.007	Presence Status of Enterprise Contacts in Call Logs and Directory	Verify that various presence status of enterprise contacts works in call logs and directory list.	Nokia Mobility Client(Cisco Unified Presence+Unified CM	Passed	
UC861IF.MOB.008	Multiple Instant Messaging (IM) Sessions and Escalation of IM to Voice Call	Verify that Nokia Mobility Client can establish multiple IM sessions and it can escalate some of the IM to voice calls.	Nokia Mobility Client-(Unified CM+Cisco Unified Presence)	Passed	
UC861IF.MOB.009	Nokia Mobility Client Attends Webex Meeting, Dials into Meeting and Receives Call Back	Verify that Nokia Mobility Client can attend WebEx meeting by dialing into meeting, entering meeting ID, and by receiving call back from Meeting Place.	Nokia Mobility Client-(Unified CM+Meeting Place)	Passed	
UC861IF.MOB.010	Mobility: Handoff Invoked from Client Having Multiple Calls	Verifies that handoff to mobile works from a client who has multiple calls.	Nokia Mobility Client-(Unified CM+ Gateway)	Passed	
UC861IF.MOB.011	Cisco Android Client receiving a SIP Intercluster Call and Moves the Call to Mobile	Verify that the Cisco Mobile for Android can receive the call while registered to WiFi and then it can send the call to mobile network and continue the call.	Android Mobility Client<----Unified CM1----SIP---Unified CM2---IP Phone	Passed	
UC861IF.MOB.012	Android Mobility Client joining Meeting and Transferring the Call to Cell Number	Verify that Android mobility client can dial into and dial out to WebEx/Meeting Place meeting and then transfer the call to mobile phone.	Nokia Mobility Client->H.323 Gateway->Unified CM-->MeetingPlace	Passed	
UC861IF.MOB.013	Nokia Mobility Client Adapts to the Configuration Changes in Phone Page	Verify that the Nokia Mobility Client can adapt to device pool, Media Resource Group List (MRGL) and Calling Search Space (CSS) changes in the phone configuration page of Unified Communications Manager.		Passed	
UC861IF.MOB.014	Android Mobility Client Establishing a Conference Call between iPhone Client and an Intercluster Destination	Verify that Android client can set up a conference call, when the other parties of conference call are iPhone client and an IP phone across Intercluster SIP trunk.	Nokia Mobility Client	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.MOB.015	Receiving Call through PSTN Carrier when the Client is in Conference	Verify when Android Mobility Client is in conference with enterprise contacts, it receives a call through GSM network.	Soundwave Client----Unified CM----TelePresence and Cisco UC Integration@ for Microsoft Office Communicator	Passed	
UC861IF.MOB.016	Android Mobility Client getting an Incoming Call through Trunk with Early Offer Turned ON and then transfers the call	Verifies if an Android Mobility client receives an incoming call through a Trunk with early offer turned ON, and the soundwave user is able to transfer the call.	Soundwave Client---Unified CM1-----<Cisco IME>----Unified CM2-----iPhone; Soundwave Client---Unified CM1----Cisco IME---Unified CM2	Passed	
UC861IF.MOB.017	Android Mobility Client setting up three way conference and handoff to Extension Mobility logged in deskphone	Verifies that when an Android Mobility Client has Device Mobility turned ON, the user can log into Extension Mobility phone and bring up the client to set up a three way conference, involving one user across SIP trunk, and a third user. Verifies that software conference resource at remote site is used, and the soundwave user can handoff the call to the Extension Mobility deskphone.	Soundwave Client Unified IP Phone1 Unified IP Phone2-----Unified CM1----Extension Mobility Phone	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.MOB.018	Android Mobility Client receiving an incoming call from a Cisco TelePresence endpoint and handoff the call to UC Integration@ for Microsoft Office Communicator	Verify if the Android Mobility Client application is running in background when the user receives an incoming call from a Cisco TelePresence endpoint. Verifies if the Cisco TelePresence user requests the soundwave user for video to handoff the call to UC Integration@ for Microsoft Office Communicator and resume the video call.	Soundwave Client----Unified CM----Cisco TelePresence and UC Integration@ for Microsoft Office Communicator	Passed	
UC861IF.MOB.019	Resilience of Nokia Mobility Client on Registration with Cisco Unified Communications Manager	Verify that the Cisco mobility client can register to standby Unified Communications Manager server when active one fails and continue to work normally.	Nokia Mobility Client	Passed	

Video

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VID.001	Verify BFCP (Binary Floor Control Protocol) reception and Adhoc Conference using Cisco IP Video Phone E20	Verify reception of Binary Floor Control Protocol on Cisco IP Video Phone E20 and Binary Floor Control Protocol initiation on Cisco TelePresence System EX90 which are in a adhoc conference.	Cisco TelePresence System (CTS) 500->Unified CM1->CTS EX90 CTS 500->Unified CM1->ICT->Unified CM2->Cisco IP Video Phone E20; CTS 500->Unified CM1->Conference-> Codian MCU->Presentation Share->CTS EX90 and Cisco IP Video Phone E20	Passed	
UC861EF.VID.002	Cisco IP Video Phone E20 shared line with legacy endpoint	Verify Video escalation/descalation on Cisco IP Video Phone E20.		Passed	
UC861EF.VID.004	Cisco TelePresence ISDN Gateway 3241 Interoperability	Verify adhoc conference involving Expressway and H.320 endpoint.		Failed	CSCtn95798 CSCtq17644
UC861EF.VID.005	Interoperability with Session Management Edition	Verify call transfer with Video Communication Sever endpoints and Cisco Unified Communications Manager endpoints over Session Management Edition with delayed/early offer interworking.	Unified IP Phones 8941/45->Unified CM2->SME1->SME 2->Unified CM1->Cisco IP Video Phone E20->Transfer->Video Communication Server->ISDN Gateway->H.320 Phone	Failed	CSCtn95798 CSCtq17644
UC861EF.VID.101	Point to Point native TelePresence to Unified Communications Interoperability with Video Communication Server Expressway	Verify Point to Point native TelePresence to Unified Communications interoperability with Video Communication Server Expressway.	Cisco IP Video Phone E20->VCS Expressway->Traversal Link->VCS-Control->SIP Trunk->Unified CM->Cisco TelePresence System 500	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VID.102	Point to Point native TelePresence to Unified Communications Interoperability across Session Management Edition Clusters	Verify Point to Point native TelePresence to Unified Communications Interoperability across Session Management Edition clusters.	Cisco IP Phone->Unified CM->SIP Trunk->SME 1->SIP->SME 2->Unified CM->Cisco TelePresence System500	Passed	
UC861EF.VID.103	Tandberg Single Stream High-Definition (HD) and High Definition-Standard Definition Interoperability fixes with Tandberg 550	Verify Tandberg single stream High-Definition (HD) and HD-SD interoperability fixes with Tandberg 550.	Tandberg 550->H.323->Video Communication Server->SIP Trunk->Unified CM->Cisco IP Phone	Passed	
UC861EF.VID.104	Tandberg Single Stream High-Definition (HD) and High Definition-Standard Definition Interoperability Fixes with Cisco TelePresence EX90	Verify Tandberg single stream High-Definition (HD) and High Definition-Standard Definition interoperability fixes with Cisco TelePresence EX90.	Cisco TelePresence EX90->H.323->Video Communication Server->SIP Trunk->Unified CM->Cisco IP Phone	Passed	
UC861EF.VID.105	Presentation share between Cisco TelePresence and Tandberg Endpoints	Verify Presentation share between Cisco TelePresence and Tandberg endpoints.	Cisco TelePresence System 500->Unified CM->SIP Trunk->Video Communication Server->Cisco TelePresence EX90	Passed	
UC861EF.VID.106	Interoperability Testing of Tandberg and Cisco Unified IP Phone 8941 Series Phones	Verify interoperability testing of Tandberg and Cisco Unified IP Phone 8941 series phones.	Cisco IP Phone->Unified CM->SIP Trunk->Video Communication Server-Control->Traversal Link->Video Communication Server Expressway->Cisco IP Video Phone E20	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VID.107	SIP Wideband Audio Codec Support	Verify SIP wideband audio codec support.	Cisco IP Phone->Unified CM->SIP Trunk->Video Communication Server->H.323->Tandberg 550	Passed	
UC861EF.VID.108	Calls between Cisco TelePresence System 500 Phones across SIP Inter Cluster Trunks	Verify calls between Cisco TelePresence System500 phones across SIP Inter cluster trunks.	Cisco TelePresence System 500->Unified CM 1->SIP Inter Cluster Trunk->Unified CM 2->Cisco TelePresence System 500	Passed w/ Exception	One Cisco TelePresence System 500 endpoint was replaced with Cisco TelePresence System 1000 as the codec had issues.
UC861EF.VID.201	Intra Cluster Adhoc Multi-Point Conference among Native Unified Communications endpoints and Cisco TelePresence System 500	Verify Adhoc multipoint conference among Unified IP phones 9971, Unified IP Phones 8941/45, Cisco Unified Communications Integration for RTX, Cisco TelePresence System EX90, H320 PSTN and Cisco TelePresence System 500 endpoints is successful. Verify the resources are released after the conference and repeat the scenario with various native Unified communications endpoints.	Step1)UC integration @for MOC->Unified CM->Cisco TelePresence System 500 Step2) UC integration @for MOC->Unified CM->UC integration @for RTX Step3)UC integration @ for MOC->Unified CM->Conference->Coding MCU->CTS 500 and UC integration @for RTX	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VID.202	Inter Cluster Adhoc Multi-Point Conference among Native Cisco Unified Communications endpoints and Cisco Telepresence Endpoint	Verify Adhoc multipoint conference among CTS 500, CUPC, CUCIMOC, 9971 and 8941 successful. Verify the resources are released after the conference.	Step 1)9971->CUCM1->SIP ICT->CUCM2->CTS 500 Step 2)9971->CUCM1->H225 ICT->CUCM2->8941 ; Step 3)9971->CUCM1->CUPC; Step 4)9971->CUCM1->CUCIMOC ; Step 6)9971->CUCM1->CONF->Codian MCU->9971 & 8941 &CUPC &CUCIMOC&CTS 500	Passed w/ Exception	CSCtq74688
UC861EF.VID.204	Adhoc Multipoint Presentation share among Cisco TelePresence System 500 ,Cisco TelePresence SystemEX90, Cisco IP Video Phone E20	Verify the Presentation share among Cisco TelePresence System 500,Cisco TelePresence System EX90 and Cisco 9971 video phone is successful	Cisco TelePresence System (CTS) 500->Unified CM1->CTS EX90 CTS EX90->Unified CM1->9971; CTS EX90->Unified CM1->Conference->Codian MCU->Presentation Share->CTS 500 and 9971	Passed	
UC861EF.VID.205	Adhoc Multipoint Conferencing using SIP 4501 MCU among Cisco UC Integration(TM) for Microsoft Office Communicator, Unified IP phone 6961,Unified IP Phones 8941/45 and Cisco TelePresence System 500 Endpoint	Verify Adhoc Multipoint conferencing among Unified IP Phone 6961, Cisco UC Integration(TM) for Microsoft Office Communicator, Unified IP Phones 8941/45 and Cisco TelePresence System 500 endpoints.	CUCI-MOC>Unified CM1>Unified IP Phone 8945;CUCI-MOC>UnifiedCM1>ICT->UnifiedCM2>9971;MOC>UnifiedCM1>CUCI-RTX;CUCI-MOC>UnifiedCM1->ICT>UnifiedCM2>7985;CUCIMOC>Unified CM1>H.320 PSTN;CUCIMOC>Unified CM1>Conference>Codian MCU>8945 9971 7985 H.320 PSTN&CUCI-RTX	Failed	CSCtq17644

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VID.206	Secure SIP Video Signaling between Cisco Telepresence System 500 and Tandberg	Verify the call from H.323 endpoint registered to Cisco TelePresence Video Communication Server (VCS) to the Cisco TelePresence System 500 through SME cluster is a non-Secure call.	Step1: Cisco TelePresence 1700 MXP (H.323)->Video Communication Server-Secure SIP->SME1->Non-secure SIP Trunk->SME2->Unified CM2->Cisco TelePresence System 500	Failed	CSCtq17644
UC861EF.VID.301	Tandberg Basic High Definition Video Interoperability: Validate Video Communication Server, Tandberg Codian and Expressway Deployment Scenarios(Reservationless Conference)	Verify that different Unified Communications endpoints are able to participate in video conferencing, and Cisco ISR-G2 is provisioned with PVDM3 (High-Density Packet Voice Digital Signal Processor Module) DSP modules to use with Cisco Unified Communications Manager for video conferencing purposes.	Video Ph1>WAN>Unified CM1>Rem VCB>Meet-me DN Variation:Vid Ph1>Rem Unified IP Ph 9971: 9951:CUPC: CUCIMOC: CUCIRTX: Unified IP Ph 7985: Cisco IP Vid Ph E20(SIP): Tandberg Ph1>Tandberg VCS>SME1>SIP Trunk->SME2>SIP Trunk>Unified CM1>Rem VCB->Meet-me DN	Failed	CSCtq17644
UC861EF.VID.302	Tandberg Basic High Definition Video Interoperability: Validate Video Communication Server, Tandberg Codian and Expressway Deployment Scenarios(Adhoc Conference)	Verify that different Unified Communications endpoints are able to participate in video conferencing, and Cisco ISR-G2 is provisioned with PVDM3 (High-Density Packet Voice Digital Signal Processor Module) DSP modules to use with Cisco Unified Communications Manager for video conferencing purposes.	CentVidPh1->Unified CM1->Cen VCB->Unified CM1->9971->CNF->TandPh1:Variation:CentVidPh1->RT9971: 9951:CUPC:CUCIMOC:CUCIRTX:7985: E20(SIP):EX90(SIP): MXP1700(SIP): Tandberg1000(SCCP):Variation:TandPh1->TandMXP1700(SIP): E20(SIP):EX90(SIP): Tandeberg1000(SCCP)	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VID.303	Cisco ISR-G2 provisioned with PVDM3 (High-Density Packet Voice Digital Signal Processor Module) DSP modules to use with Cisco Unified Communications Manager for video conferencing purposes Connectivity to Cisco Unified Communications Manager for Adhoc Switched Video Conference in Unified Communications	Verify that different UC endpoints are able to participate in video conferencing, and Cisco ISR-G2 is provisioned with PVDM3 (High-Density Packet Voice Digital Signal Processor Module) DSP modules to use with Cisco Unified Communications Manager for video conferencing purposes.	VidPh1>UnifiedCM1 >Rem VCB>UnifiedCM1> Unified IP Phone 9951>CNF>VidPh2: Variation:VidPh1>Rem Unified IP Phone 9971:9951:CUPC:CUCIMOC:CUCIRTX:7985:E20(SIP):Variation:VidPh2>9971:9951:CUPC:CUCIMOC: CUCIRTX:7985:E20(SIP):EX90(SIP):MXP1700(SIP): Tandberg1000 SCCP	Passed	
UC861EF.VID.304	Cisco ISR-G2 provisioned with PVDM3 (High-Density Packet Voice Digital Signal Processor Module) DSP modules to use with Cisco Unified Communications Manager for video conferencing purposes in Tandberg Cisco Unified Communications Interoperability Support(Reservationless Conference)	Verify that different Unified Communications endpoints are able to participate in video conferencing, and Cisco ISR-G2 is provisioned with PVDM3 (High-Density Packet Voice Digital Signal Processor Module) DSP modules to use with Cisco Unified Communications Manager for video conferencing purposes.	VidPh1->Unified CM1->Cent VCB->Meet-me-DN: Variation:Unified IP Phone 9971:9951:CUPC:CUCIMOC: CUCIRTX:7985:EX90(SIP):MXP1700(SIP):Tandberg1000(SCCP):STEP2:Tandberg Ph1->TandVCS->SIP Trunk->Unified CM1->Cent VCB->Meet-me-DN	Passed	
UC861IF.VID.001	Conference Cisco Unified Communications Manager and Cisco TelePresence Video Communication Server (VCS) endpoints using Cisco Codian Software Bridge registered to Unified Communications Manager	Verify conference can be established between Cisco Cius T; Cisco IP Video Phone E20 and Cisco TelePresence System 1700 MXP using Cisco Codian Adhoc bridge registered to Unified Communications Manager as conference resource.	Cisco Cius@ - MSP Unified CM --- H.225 trunk --- GateKeeper - Cisco TelePresence VCS --- Cisco IP Video Phone E20 ---Conference using Cisco Cius@ --- TD Cisco 1700 MXP -H.323 - Cisco TelePresence VCS	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.002	Conference Cisco Unified Communications Manager and Cisco TelePresence Video Communication Server endpoints using Cisco Codian Software Bridge registered to Unified Communications Manager.	Verify conference can be established between Cisco Cius, Cisco Unified Personal Communicator (CUPC) running on the virtual desktop and Cisco IP Communicator/Cisco Unified Video Advantage registered to Unified Communications Manager Express.	Cisco Cius - MSP Unified CM ---SIP Trunk -Abilene Unified CM ---Unified Personal Communicator---- Conference From Unified Personal Communicator --- H.323 Gateway -H.323 Trunk --- Unified CME	Passed	
UC861IF.VID.003	Conference Cisco Unified Communications Manager and Video Communication Server (VCS) endpoints using Tandberg Codian Software Bridge registered to Unified Communications Manager	Verify conference can be established between Cisco TelePresence MoviT registered to VCS; Cisco Unified IP Phone 9971 registered to Unified Communications Manager and Polycom HDX 4000 registered to Unified Communications Manager.	Cisco IP Video Phone E20 - MSP Unified Communications Manager --- SIP Trunk -VCS -MOVi ---- Conference From Cisco IP Video Phone E20 --- SIP Trunk ---Polycom HDX 4000	Passed	
UC861IF.VID.004	Conference Cisco Unified Communications Manager and Cisco TelePresence Video Communication Server endpoints using Cisco Codian Software Bridge registered to Unified Communications Manager.	Verify conference can be established between Cisco CiusT ; Cisco TelePresence System 1000 and Cisco TelePresence 1700 MXP using Cisco Codian Adhoc bridge.	Cisco Cius@ - MSP Unified CM --- SIP Trunk ---Abilene Unified CM ---Cisco TelePresence System ---Conference using Cisco Cius --- Cisco TelePresence MXP 1700 -H.323 - Cisco TelePresence Video Communication Server	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.005	Presentation share between Cisco TelePresence System EX90; Cisco IP Video Phone E20 and Unified IP Phone 89XX/99XX	Verify whether Cisco TelePresence System EX90 user registered to Cisco Unified Communications Manager can share presentation with Cisco IP Video Phone E20 registered to Video Communication Server (VCS) and Unified IP Phone 89XX/99XX registered to Unified Communications Manager.	Unified Personal Communicator - MSP Unified CM --- SIP Trunk ---VCS -Cisco E20 ----Conference from Cisco TelePresence System Ex90 ---- SIP trunk -----Unified CM 9971 IP Phone -- Cisco TelePresence System Ex90 initiate Presentation	Passed	
UC861IF.VID.006	Presentation share from Cisco TelePresence Movi registered to Video Communication Server ; Cisco IP Video Phone E20 and Unified IP Phone 9971 registered to Unified Communications Manager	Verify if Cisco TelePresence MOVi can share presentation with Cisco IP Video Phone E20 registered to Unified Communications Manager and Unified IP Phone 9971 Phone registered to Unified Communications Manager.	Unified Personal Communicator - MSP Unified CM --- SIP Trunk ---VCS -Cisco IP Video Phone E20 ----Conference from Unified Personal Communicator ----- SIP trunk ------Unified CM 9971 IP Phone --Unified Personal Communicator initiate conference	Passed	
UC861IF.VID.007	Cisco TelePresence Quick Set C20 Performs SIP URI based Conference with Cisco Unified Communications Manager Endpoint	Verify Cisco TelePresence Quick Set C20 registered to Cisco Unified Communications Manager as third party SIP endpoint can invoke multiway conference that is registered to Cisco TelePresence VCS.	Cisco TelePresence Quick Set C20 --Unified CM --Unified IP Phone 9971 ---- Multiway ----SIP trunk --Cisco TelePresence VCS ---- Unified CM ---Unified IP Phone 9971	Failed	CSCtl56764

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.008	Presentation share from Cisco TelePresence Quick Set C20 registered to Video Communication Server ; Polycom HDX 4000 and Cisco Unified IP Phone 7985 registered to Unified Communications Manager	Verify whether Cisco TelePresence MOVi can share presentation with Polycom HDX 4000 registered to Unified Communications Manager and Cisco Unified IP Phone 7985 registered to Unified Communications Manager.	Cisco TelePresence Quick Set C20 ---Video Communication Server ----SIP Trunk ---Polycom --- Cisco TelePresence MOVi Conference ---SIP Trunk ---Unified CM ----Unified IP Phone 7985 -Initiate presentation on Cisco TelePresence MOVi	Passed	
UC861IF.VID.009	Presentation share from Cisco TelePresence System MXP 1700 registered to Cisco TelePresence Video Communication Server, Polycom HDX 4000 and Cisco IP Communicator/Cisco Unified Video Advantage phone registered to Unified Communications Manager	Verify whether Cisco TelePresence System MXP 1700 can share presentation with polycom HDX 4000 registered to Cisco Unified Communications Manager and Cisco IP Communicator phone registered to Cisco Unified Communications Manager.	Cisco TelePresence System MXP 1700 ---Cisco TelePresence System VCS ---SIP Trunk --Polycom ---Cisco TelePresence System MXP 1700-- Conference --SIP Trunk --Unified CM ---Cisco IP Communicator -Initiate presentation on Cisco TelePresence System MXP 1700	Passed	
UC861IF.VID.013	Scheduled conference using Cisco TelePresence server and presentation sharing using Cisco Unified Personal Communicator	Verify whether Cisco Unified Personal Communicator, Cisco TelePresence System 1000, Cisco TelePresence System 500 and Cisco TelePresence System EX90 can join Cisco TelePresence server conference and view Cisco Unified Personal Communicator presentation share.	Polycom HDX; Cisco Cius@ ----- SIP Trunk ----Cisco TelePresence VCS ---Cisco TelePresence Server ----- SIP trunk ---- Cisco TelePresence System MXP 1700	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.014	Cisco IP Video Phone E20 registered to Cisco TelePresence Video Communication Server Expressway connected to Demilitarized Zone (DMZ) port	Verify if Cisco IP Video Phone E20 residing in remote location can register to Cisco TelePresence Video Communication Server expressway and is able to join Cisco TelePresence Multipoint Switch conference.	Cisco IP Video Phone E20 - WAN ----DMZ-Switch ---- Cisco TelePresence VCS Expressway ----- Conference ---Abilene -Unified CM ----SIP trunk ---DEN - Session Manager Edition ---Cisco Media Experience Engine ---Cisco TelePresence Multipoint Switch	Passed	
UC861IF.VID.025.1	SIP Wideband Audio Codec Support G722.1; AAC MP4-LATM	Verify SIP Wideband Audio Codec Support G722.1 for Advanced Audio Codec (AAC) MP4-LATM		Passed	
UC861IF.VID.025.2	SIP Wideband Audio Codec Support G722.1; AAC MP4-LATM	Verify SIP Wideband Audio Codec Support G722.1 for AAC MP4-LATM		Passed	
UC861IF.VID.025.3	SIP Wideband Audio Codec Support G722.1; AAC MP4-LATM	Verify SIP Wideband Audio Codec Support G722.1 for AAC MP4-LATM		Passed	
UC861IF.VID.025.4	SIP Wideband Audio Codec Support G722.1; Advanced Audio Codec MP4-LATM	Verify SIP Wideband Audio Codec Support G722.1 for Advanced Audio Codec MP4-LATM		Passed	
UC861IF.VID.025.5	SIP Wideband Audio Codec Support G722.1; Advanced Audio Codec MP4-LATM	Verify SIP Wideband Audio Codec Support G722.1; Advanced Audio Codec MP4-LATM		Passed	
UC861IF.VID.025.6	SIP Wideband Audio Codec Support G722.1; Advanced Audio Codec MP4-LATM	Verify SIP Wideband Audio Codec Support G722.1; Advanced Audio Codec MP4-LATM		Passed	
UC861IF.VID.025.7	SIP Wideband Audio Codec Support G722.1; Advanced Audio Codec MP4-LATM (Testcase 7)	Verify SIP Wideband Audio Codec Support G722.1; Advanced Audio Codec MP4-LATM		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.025.8	SIP Wideband Audio Codec Support G722.1; Advanced Audio Codec MP4-LATM (Testcase 8)	Verify SIP Wideband Audio Codec Support G722.1 for Advanced Audio Codec MP4-LATM		Passed	
UC861IF.VID.033	Conference Cisco Unified IP Phone 9971, Cisco IP Video Phone E20 and Cisco Unified IP Phone 7985 using Tandberg Codian Adhoc Bridge registered to Unified Communications Manager	Verify Video Communication Server (VCS) endpoints can call Cisco Unified IP Phone 8941 registered to Unified Communications Manager.		Passed	
UC861IF.VID.034	Cisco TelePresence System 1000 Joins Adhoc Software Bridge registered to Cisco Unified Communications Manager	Verify Cisco TelePresence System 1000 is able to view other participants video after joining Adhoc Tandberg codian bridge.		Passed	
UC861IF.VID.035	Cisco TelePresence System 1700 MXP, Cisco TelePresence System 1000 and Unified IP Phone 9971 are able to join Adhoc Tandberg Codian Conference	Verify presentation shared on Cisco TelePresence System 1000 can be viewed on other conference endpoints.		Passed	
UC861IF.VID.036	Intercluster Video Conference using Adhoc Bridge	Verify if Unified IP phone 9971 across SIP trunk is able to join Adhoc conference.		Passed	
UC861IF.VID.037	Tandberg 7985 with Trusted Relay Point joins Tandberg Codian conference	Verify that Tandberg 7985 registered to Unified Communications Manager with a Trusted Relay Point can join a Tandberg Codian/Unified Communications Manager conference.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.038	Adhoc Conference with Cisco Unified Communications Manager and Cisco Unified Communications Manager Express endpoints	Verify if Adhoc conference works with Unified Communications Manager and Unified CME endpoints.		Passed	
UC861IF.VID.039	Adhoc Conference with Unified Communications Manager and Conference Share	Verify Adhoc conference with Cisco TelePresence System (CTS), Unified IP Phone 8941 and Cisco VCS endpoint and share the presentation on Cisco TelePresence MOVi.		Passed	
UC861IF.VID.040	Verify Client Services Framework (CSF) clients are able to join Cisco TelePresence MCU Adhoc Conference	Verify Unified Personal Communicator, Cisco UC Integration(TM) for Microsoft Office Communicator and Cisco TelePresence MoviT are able to join Adhoc conference.		Passed	
UC861IF.VID.041	Hold and Resume on Cisco Unified IP Phone 9971 while in Adhoc Conference	Verify whether hold and resume can resume the video on Unified IP Phone 9971 while the endpoint has joined Adhoc conference.		Passed	
UC861IF.VID.049	Verify Intercluster call between Cisco IP Video Phone E20 registered Native to Unified Communications Manager and Cisco Unified IP Phone 7985.	Verify Intercluster SIP call between Cisco IP Video Phone E20 and Cisco Unified IP Phone 7985.		Passed	
UC861IF.VID.050	Unified Communications Manager calls Tandberg Codian conference and joins conference	Verify if Unified Communications Manager can call Tandberg Codian conference and is able to join the conference.		Passed	
UC861IF.VID.051	Hold and Resume with Cisco TelePresence Quick Set C20	Verify if hold and resume works with Cisco TelePresence Quick Set C20 that is registered as third party SIP endpoint.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.052	Video Interoperability with Unified IP Phone 8941	Verify VCS endpoints can call Cisco Unified IP Phone 8941 registered to Unified Communications Manager.		Passed	
UC861IF.VID.053	Cisco Unified IP Phone 9900 Series interoperability with secure Cisco TelePresence System and Cisco Telepresence Multipoint Switch	Verify the inter working of Secure Cisco TelePresence System and Cisco Telepresence Multipoint Switch with unsecure Cisco Unified IP Phone 9900 Series end points joining Cisco Telepresence Multipoint Switch through Media Experience Engine.	Secure Cisco TelePresence System-Unified CM1-Secure SIP Trunk-Unified CM2-Secure SIP Trunk- Cisco Telepresence Multipoint Switch; Unified IP Phone 9900 Series-Unified CM1-SIP Trunk-Unified CM2-SIP Trunk-Cisco Telepresence Multipoint Switch	Passed	
UC861IF.VID.054	Secure Cisco TelePresence System interoperability with Media Experience Engine and Unified 9971 IP Phones	Verify the ability to place a Peer-to-Peer (P2P) call between secure Cisco TelePresence System and unsecure Unified 9971 IP Phones through Media Experience Engine.		Passed	
UC861IF.VID.055	Secure Cisco TelePresence System end point interoperability with SIP Tandberg end points behind Cisco TelePresence Video Communication Server (VCS)	Verify that a secure Cisco TelePresence System end point can make a Peer-to-Peer video call with a non-secure Tandberg video end points behind Cisco TelePresence Video Communication Server.		Passed	
UC861IF.VID.056	Secure Cisco TelePresence System interaction with Cisco TelePresence Movi¿ client behind Cisco TelePresence Video Communication Server (VCS)	Verify the interaction between secure Cisco TelePresence System and Cisco TelePresence Movi¿ client and ensuring that the client can share its desktop.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.057	Cisco TelePresence System Security with Non secure SIP Trunks	Verify that Cisco TelePresence is able to call across non secure SIP Trunks without secure RTP enabled and still have a secure media path with Cisco TelePresence Multipoint Switch using Datagram Transport Layer Security.	Secure Cisco TelePresence System1---Unified CM---SIP Trunk-Unified CM---SIP Trunk---Secure Cisco TelePresence Multipoint Switch; Secure Cisco TelePresence System 2---Unified CM--SIP Trunk-Unified CM---SIP Trunk---Secure Cisco TelePresence Multipoint Switch	Passed	
UC861IF.VID.058	Scheduled conference using Cisco TelePresence server and Presentation Sharing using Cisco TelePresence MOVi	Verify whether Cisco TelePresence MOVi, Cisco TelePresence System 3000, Cisco Unified IP Phone 7945G can join Cisco TelePresence server conference and view Cisco TelePresence MOVi presentation share, given that Cisco Unified IP Phone 7945G should be able to hear audio of all conference participants.	Polycom HDX ; Cisco CIUS ----- SIP Trunk ----Video Communication Server ---Cisco TelePresence Server --- SIP trunk ----Cisco TelePresence 1700 MXP	Passed	
UC861IF.VID.059	Attend Scheduled conference using Cisco TelePresence server	Verify whether Cisco TelePresence System, Cisco Cius, Unified IP Phone 9971, Cisco TelePresence 1700 MXP and Polycom HDX are able to attend scheduled conference on Cisco TelePresence server.	Polycom HDX ; Cisco Cius ;Unified IP Phone 9971--- SIP Trunk ----VCS ---Cisco TelePresence Server ----- SIP trunk ----Cisco TelePresence 1700 MXP	Passed	
UC861IF.VID.060	SIP- SIP call with Cisco TelePresence Video Communication Server via Session Manager Edition Works	rify whether video works fine when the call is placed from Cisco Unifid Communications Manager to Cisco TelePresence Video Communications Server via Session Manager Edition.	Unified IP Phone 89xx/99xx --Unified CM -SIP -SME --SIP -Cisco TelePresence VCS---Cisco IP Video Phone E20	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.061	Call Hold /Resume work with Cisco TelePresence Video Communication Server via Session Manager Edition Works	Verify whether the call placed from Unified IP Phone 9971 to Cisco TelePresence Video Communication Server via Session Manager Edition is able to hold and resume the call.	Unified IP Phone 9971 --Unified CM -SIP -SME --SIP -Cisco TelePresence VCS---Cisco IP Video Phone E20 ---Hold and resume on Unified IP Phone 9971	Passed	
UC861IF.VID.062	Call from Unified Communications Manager to Cisco TelePresence Video Communications Server via Session Manager Edition works	Verify whether Inter Cluster Trunk-SIP interoperability with Session Manager Edition and Cisco TelePresence Video Communications Server results in bi-directional video.	Unified IP Phone 9971 --Unified CM -Inter Cluster Trunk -SME --SIP -Cisco TelePresence VCS---Cisco IP Video Phone E20 ---Hold and resume on Unified IP Phone 9971	Passed	
UC861IF.VID.063	Call from Cisco Unified Communication Manager to Cisco TelePresence Video Communication Server via Session Manager Edition with Early offer trunk	Verify bi-directional video between Unified Communications Manager - Session Manager Edition - Cisco TelePresence Video Communication Server when SIP trunk is set to early offer on both SIP trunks.	Unified IP Phone 9971 --Unified CM -SIP(EO) -SME --SIP(EO) -Cisco TelePresence VCS---Cisco IP Video Phone E20 ---Hold and resume on Unified IP Phone 9971	Passed	
UC861IF.VID.064	Scheduled conference using Cisco TelePresence server and presentation sharing using Cisco TelePresence MOVi / Cisco TelePresence Ex90	Verify whether Cisco TelePresence Ex90, Cisco TelePresence System 1000, Cisco TelePresence System 500 and Cisco TelePresence Ex90 can join Cisco TelePresence server conference and view Cisco TelePresence EX90 and Cisco TelePresence MOVi presentation sharing.	Polycom HDX ; Cisco Cius ----- SIP Trunk ----Video Communication Server --- Cisco TelePresence Server ----- SIP trunk ----Cisco TelePresence 1700 MXP	Passed	

Virtualization Experience Client

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VXC.001	UC Integration@(Deskphone) for RTX Inter-cluster Video Call to Third Party Skinny Endpoint.	Verify if UC Integration@ for RTX(Deskphone) 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) Inter Cluster Trunk->Unified CM2->SCCP video endpoint	Passed	
UC861EF.VXC.002	Escalation of Audio Call to Video Call in Softphone mode	Verify if UC Integration@(Softphone) for RTX from one cluster can make an audio call to another UC Integration@ for RTX client in deskphone mode running on voicemail and then can escalate to video.	UC Integration@ for RTX(softphone)->Unified CM1->UC Integration@ for RTX (deskphone)	Passed	
UC861EF.VXC.003	Escalation of Audio Call to Video Call in Deskphone Mode When Calling Cisco Unified Communications Integration @ for Microsoft Office Communicator in Another Cluster	Verify if Cisco Unified Communications Integration@(deskphone) for RTX can make an inter-cluster call to Cisco Unified Communications Integration @ for Microsoft Office Communicator and can escalate the audio call to video call.	UC Integration@ for RTX->Unified CM1->SIP(QSIG) Inter Cluster Trunk->Unified CM2->UC Integration@ for MOC	Passed	
UC861EF.VXC.004	Inter-cluster Video Conference with Cisco Unified Communications Integration@ for RTX and Unified IP Phone 9900/8900 series	Verify if Cisco Unified Communications Integration@(Softphone) for RTX can make an inter-cluster video call to another UC Integration@ for RTX user in deskphone mode and can join a Unified IP Phones 89XX/99XX phone in another cluster to the conference.	UC Integration@ for RTX->Unified CM1->Annex M1 Inter Cluster Trunk->Unified CM2->Unified IP Phones 89XX/99XX	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VXC.005	Hold/Retrieve from Shared Line with Cisco Unified IP Phone 8900/9900 Series in Another Cluster	Verify if Cisco Unified Communications Integration@(deskphone) for RTX can make an inter-cluster video call to a Cisco Unified IP phone 8900/9900 series and can put the call on hold and retrieve it from a shared line.	UC Integration@ for RTX->Unified CM1->SIP(QSIG) Inter Cluster Trunk->Unified CM2->Unified IP phone 8900/9900	Passed	
UC861EF.VXC.006	Fall Back to Unified Survivable Remote Site Telephony when Unified Communications Manager goes down	Verify if Cisco Unified Communications Integration@ for RTX registers to Unified Survivable Remote Site Telephony 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	
UC861EF.VXC.007	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	
UC861EF.VXC.008	Third party H.323 Endpoint with Gatekeeper Video Call to Cisco Unified Communications Integration@(Deskphone) for RTX	Verify if third party H.323 endpoint with gatekeeper can make a video call to Cisco Unified Communications Integration@(Deskphone) for RTX.	H.323 video endpoint->Unified CM->UC Integration@ for RTX	Passed w/ Exception	Audio calls are working fine. Since Video call needs a video transcoder which was not there in that cluster.
UC861EF.VXC.009	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 UC Integration@ for RTX (DeskPhone) and transfer the call to IP Communicator and Unified Video Advantage.	UC Integration@ for RTX (SoftPhone)->Unified CM1->Annex M1 Inter Cluster Trunk->Unified CM2->UC Integration@ for RTX(Deskphone)->Transfer->Inter Cluster Trunk->Cisco IP Communicator+ Cisco Unified Video Advantage	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VXC.010	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@ (Deskphone) for RTX and then transfer the call to UC Integration@ for MOC.	Remote UC Integration@(Softphone) for RTX->Inter Cluster Trunk-> UC Integration@ (Deskphone) for RTX->Transfer->Central UC Integration@ for MOC	Passed	
UC861EF.VXC.011	Central Site Unified Personal Communicator 8.0 Client Video Call to Cisco Unified Communications Integration@(Softphone) for RTX in Remote Site	Verify if the central site Unified Personal Communicator 8.0 client can make a video call to Cisco Unified Communications Integration@(Softphone) for RTX in remote site and then consult transfer the call to Cisco Unified Communications Integration@ for RTX in deskphone mode.	Central Excession->Unified CM->UC Integration @ for RTX (Remote)->Transfer C->UC Integration @ for RTX (Deskphone)	Passed	
UC861EF.VXC.012	Cisco Unified Communications Integration@(Softphone) for RTX Call to PBX Phone in Interoperability Site	Verify if Cisco Unified Communications Integration@(Softphone) for RTX can make a call to a PBX phone in interoperability site.	UC Integration @ for RTX->Unified CM1->SIP Inter Cluster Trunk(QSIG)->Unified CM2->QSIG Trunk->PBX phone	Passed	
UC861EF.VXC.013	Audio Conference with Q Interface Signalling Protocol (QSIG) Private Branch Exchange (PBX) and PSTN Phones.	Verify if UC Integration@(Deskphone) for RTX can make a conference call with PSTN and QSIG PBX phone.	UC Integration@ for RTX->Unified CM->PSTN Gateway->PSTN->Conference->QSIG Trunk->PBX phone	Passed	
UC861EF.VXC.014	Cisco Unified Communications Integration@ for RTX Failover to PSTN When WAN is Down.	Verify if Cisco Unified Communications Integration@ for RTX calls go through PSTN to remote site when there is insufficient bandwidth.	UC Integration @ for RTX->Unified CM->MGCP PRI Gateway->PSTN->Remote SCCP Phone1	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VXC.015	Escalation of Audio Call to Video When Call is Transferred to a Video Phone	Verify if UC IntegrationT for RTX in remote site can call a central SCCP phone and when the call gets transferred to UC IntegrationT for MOC, then if two way video is established.	UC Integration@ for RTX (remote)->Unified CM->SCCP Phone 1->Transfer->UC Integration@ for MOC	Passed	
UC861EF.VXC.016	PSTN Call to H.320 Endpoint from Cisco Unified Communications Integration@ for RTX	Verify if Cisco Unified Communications 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	Failed	CSCtq17644
UC861EF.VXC.017	Inter-Cluster Call over Cisco IME	Verify if Cisco Unified Communications Integration@ for RTX can make an Intercluster Call to a Cisco IP Phone 7985 over Cisco IME.	UC Integration @ for RTX->Unified CM 1->Adaptive Security Appliances->Cisco IME Trunk->Adaptive Security Appliances->Unified CM 2->Cisco Unified IP Phone 7985	Passed	
UC861EF.VXC.018	Intercluster Video Call over Cisco IME After Transfer from Cisco Unified Communications Integration@ for Microsoft Office Communicator	Verify if Cisco Unified Communications Integration@ for Microsoft Office Communicator can make an inter-cluster call to an SCCP phone which is then transferred to Cisco Unified Communications Integration@ for RTX in remote branch.	UC Integration@ for MOC->Unified CM1->Adaptive Security Appliance->Cisco IME Trunk->Adaptive Security Appliance->Unified CM2->SCCP phone1->Transfer->Cisco IME trunk->Adaptive Security Appliance->Unified CM1->Remote Branch->UC Integration @ for RTX	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VXC.019	Cisco Unified Communications Integration@ for RTX Video Call Between Remote Sites	Verify if Cisco Unified Communications Integration@ for RTX in a remote branch can make a video call to Cisco Unified Communications Integration@ for RTX in another remote site.	UC Integration @ for RTX (Remote1)->Unified CM->UC Integration @ for RTX (Remote 2)	Passed w/ Exception	Made a call from remote Phone to UC Integration @ for RTX running in VoiceMail to cover this scenario.
UC861EF.VXC.020	Inter-Cluster Adhoc Video Conference with Cisco IP Communicator, Cisco Unified Video Advantage and Third Party H.323 Endpoint.	Verify if UC IntegrationT(Deskphone) for RTX can take part in an inter-cluster Ad-hoc video conference with IP Communicator and Unified Video Advantage and third party H.323 endpoints.	Cisco IP Communicator+Unified Video Advantage->Unified CM->UC Integration @ for RTX->Conference->H.323 video endpoint	Passed w/ Exception	Executed the below mentioned callflow: Cisco IP Communicator Cisco Unified Video Advantage->Unified CM->QSIG Inter Cluster Trunk-> UC Integration @ for RTX->Conference->SIP Inter Cluster Trunk -> H.323 video endpoint
UC861IF.VXC.001	Independent Computing Architecture (ICA) Standalone, Mouse, USB KB, and two monitors Powers On and Works via 802.3AT Power Over Ethernet	Verifies that the ICA standalone, the USB mouse, USB KB and two monitors power on and all peripherals work properly via 802.3AT PoE		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VXC.002	PC over IP Standalone, USB Mouse, USB KB, and Two Monitors Powers On and Works via 802.3AT Power over Ethernet	Verify that the PC over IP (PCoIP) standalone, USB mouse, USB KB, and two monitors powers on and all peripherals work properly via 802.3AT Power over Ethernet		Passed w/ Exception	CSCtn12208
UC861IF.VXC.003	Detect the Accessory USB Flash Drive when Device is Operational	Verify to ensure that a user accessing a Voice Mail using VDI/VXC is able to plug in a USB flash drive and access data from it.		Failed	CSCtl74889
UC861IF.VXC.004	UC Integration@ for Microsoft Office Communicator is desk phone mode accessed using a Virtualization Experience Client (VXC 2111)	Verify that UC IntegrationT for Microsoft Office Communicator in deskphone mode works seamlessly when controlled over a Virtual Desktop Interface (VDI) interface, and audio quality of Visual Voice Mail played from the Voice Mail is good.		Passed	
UC861IF.VXC.005	Verify power to all USB ports on Virtual Desktop Interface (VDI)/ Virtualization Experience Client (VXC) standalone	Verify that all USB ports on VDI/VXC Standalone have power when power to VDI/VXC is provided via Power Over Ethernet at switch and power brick.		Passed	
UC861IF.VXC.006	Verify Virtualization Experience Client (VXC) - PC over IP Admin Graphical User Interface Functionality	Verifies if the client can move from Kiosk mode to non-kiosk mode, and whether features under the diagnostics options work for the Admin Graphical User Interface in VMWARE View client on the Virtual Desktop Infrastructure (VDI)/ Virtualization Experience Client (VXC) device. Verifies VMWARE View options for "Auto Launch if only one desktop".		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VXC.007	VXC-ICA Classic Desktop Graphical User Interface Tests	Verify the Independent Computing Architecture (ICA) Classic Desktop Graphical User Interface is user friendly and functions as expected.		Passed	
UC861IF.VXC.008	VXC-ICA Zero Launchpad Graphical User Interface Tests	Verify the ICA Zero Launchpad Graphical User Interface is user friendly and functions as expected.		Passed	
UC861IF.VXC.009	NGPoE switch operational with Cisco Unified IP Phone 9971, camera, PC over IP (PCoIP) Zilch Backpack, four USB peripherals, two monitors, and external speakers	Verify that Cisco Unified IP Phone 9971, camera, PC over IP Zilch BackPack, four USB peripherals, two monitors, and external speakers are powered and operational when plugged into an NGPoE switch.		Passed	
UC861IF.VXC.010	NGPoE switch operational with Cisco Unified IP Phone 9971, camera, Independent Computing Architecture Virtualization Experience Client BackPack, four USB peripherals, two monitors, and external speakers	Verify that Cisco Unified IP Phone 9971, camera, Independent Computing Architecture Virtualization Experience Client backpack, four USB peripherals, two monitors, and external speakers is powered and operational when plugged into an NGPoE switch.		Passed	
UC861IF.VXC.011	Independent Computing Architecture: Camera Disabled on Cisco Unified Communications Manager pages but plugged in	Verify that when camera is disabled via Unified Communications Manager but plugged in, the Cisco Unified IP Phone 9971 Independent Computing Architecture backpack operates with the power specifications of a Unified IP Phone 9971 without camera on 802.3 AT.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VXC.012	Virtualization Experience Client (VXC) verifying peripherals come up after Switch Reset	Verify that phone, VXC backpack and VXC standalone come up with all peripherals powered up after the switch port supplying power is reset.		Passed	
UC861IF.VXC.013	Verify PC Over IP with Secure Socket Layer (SSL) Connection with Backpack and Standalone	Verify that backpack and standalone devices are able to connect to the view connection server using SSL.		Passed	
UC861IF.VXC.014	Use Real-Time Monitoring Tool (RTMT) application on Zilch PC over IP and Independent Computing Architecture	Verify that RTMT application for collecting logs and monitoring Cisco CallManager application works on Zilch backpack and Standalone PC over IP and Independent Computing Architecture.		Passed	
UC861IF.VXC.015	NGPoE and max Key Expansion Module config (Cisco Unified IP Phone 9971 with camera, USB mouse, USB keyboard, one monitor, and three Key Expansion Modules)	Verify a Virtualization Experience Client backpack on NGPoE with Cisco Unified IP Phone 9971 with camera, USB mouse, USB keyboard, one monitor, and three Key Expansion Modules powers on correctly and is operational.		Passed	
UC861IF.VXC.016	NGPoE with Independent Computing Architecture stand-alone, four USB peripherals, two monitors, and external speakers	Verify that an Independent Computing Architecture stand-alone, four USB peripherals, two monitors, and external speakers are powered and operational when plugged into an NGPoE switch.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VXC.017	Not Enough Power for Camera Scenario	Verify that when the backpack is running with a Unified IP Phone 9971, two USB, two monitors on 802.3AT (power is maxed out) and a camera is added, that the phone throws an error indicating there's not enough power for the camera and the backpack still operates normally.		Passed	
UC861IF.VXC.018	Quick Removal and Insertion of Multiple USB Devices	Verify that USB devices can be interchanged quickly on the same port with no adverse affects.		Passed	
UC861IF.VXC.019	Backpack behavior with power negotiation disabled via Unified Communications Manager	Verify that the backpack powers on within the 802.3 AT specifications when power negotiation is disabled via Unified CM.		Passed	
UC861IF.VXC.021	Upgrade Independent Computing Architecture firmware using Virtualization Experience Client (VXC) Manager	Verify the ability to upgrade an Independent Computing Architecture (ICA) backpack and ICA stand-alone by pointing the device to a VXC Manager file server.		Passed	
UC861IF.VXC.022	Bluetooth mouse and USB mouse can be used at same time	Verify Bluetooth USB mouse and wired USB mouse can be used at same time on the PC over IP (PCoIP) and Independent Computing Architecture (ICA) units.		Passed	
UC861IF.VXC.023	Independent Computing Architecture: Swap monitors while logged into VoiceMail	Verify that a monitor of a different size can be swapped and the screen auto-corrects while a stand-alone Virtualization Experience Client unit is logged into an Independent Computing Architecture VoiceMail.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VXC.024	PC over IP: Swap monitors while logged into VoiceMail	Verify that a monitor of a different size can be swapped and the screen auto-corrects while a stand-alone Virtualization Experience Client unit is logged into a PC over IP Voicemail.		Passed	
UC861IF.VXC.025	NG Power over Ethernet with PC over IP stand-alone, Four USB Peripherals, Two Monitors, and External Speakers	Verify that a PC over IP stand-alone, four USB peripherals, two monitors, and external speakers are powered and operational when plugged into an NG Power over Ethernet switch.		Passed	

All Tests

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.OTH.001	Voicemail Deposit and Retrieval for a Connection User whose Directory Number is registered as a E.164 Number	Verify voicemail deposit and retrieval for a Connection user whose directory number is registered as a E.164 number in Cisco Unity Connection 8.6.	PSTN Phone->PSTN Gateway->Unified CM->IP Phone->Call Forward No Answer->Cisco Unity Connection; IP Phone->Unified CM->Cisco Unity Connection	Passed	
UC861EF.OTH.002	Visual Voicemail Feature for a Connection User whose Directory Number is Registered as a E.164 Number	Verify visual voicemail feature for a Connection user whose directory number is registered as a E.164 number in Cisco Unity Connection 8.6.		Passed	
UC861EF.OTH.003	Allow Outside Callers to Mark Messages Private	Verify the ability to allow outside callers to mark messages private in Cisco Unity Connection 8.6.	PSTN Phone->PSTN Gateway->Unified CM->Cisco Unity Connection->IP Phone	Passed	
UC861EF.OTH.004	Allow Users to Strip the Introduction from a Message Prior to Forwarding	Verify the ability to allow users to strip the introduction from a message prior to forwarding in Cisco Unity Connection 8.6.	IP Phone->Unified CM->Cisco Unity Connection	Passed	
UC861EF.OTH.005	Do not Prompt Users to Record an Introduction	Verify users are not prompted to record an introduction in Unity Connection 8.6.	IP Phone->Unified CM->Cisco Unity Connection	Passed	
UC861EF.OTH.006	Warn users on Reply All that their Message is Going to be Greater than X recipients	Verify the ability to warn users on Reply All that their message is going to greater than X recipients in Unity Connection 8.6.		Passed	
UC861EF.OTH.007	Transfer to E.164 Numbers with Call Handlers	Verify transfer to E.164 numbers with call handlers.	Cisco IP Phone->Unified CM->Cisco Unity Connection->Call Handler->Transfer->Cisco IP Phone with E.164 number	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.OTH.008	Transfer to E.164 numbers using Interview handlers	Verify Transfer to E.164 numbers using Interview handlers.	Cisco IP Phone->Unified CM->Cisco Unity Connection->Interview Handler->Transfer->Cisco IP Phone with E.164 number	Passed	
UC861EF.OTH.009	Transfer to E.164 numbers with directory handlers	Verify Transfer to E.164 numbers with directory handlers.	Cisco IP Phone->Unified CM->Cisco Unity Connection->Directory Handler->Transfer->Cisco IP Phone with E.164 number	Passed	
UC861EF.OTH.010	Personal Call Transfer Rules Based Transfers to E.164 Numbers	Verify personal call transfer rules based transfers to E.164 numbers.	Cisco IP Phone->Unified CM->Cisco Unity Connection->Cisco IP Phone->Personal Call Rule Transfer->Cisco IP Phone with E.164 number	Passed	
UC861EF.OTH.011	Alternate Extensions and Restriction Pattern Support for E.164 Numbers	Verify alternate extensions and restriction pattern support for E.164 numbers.	Cisco IP Phone->Unified CM->Cisco Unity Connection	Passed	
UC861EF.OTH.012	Notification devices with E.164 number support	Verify Notification devices with E.164 number support.	Cisco IP Phone->Unified CM->Cisco Unity Connection->Notification->Cisco IP Phone with E.164 number	Passed	
UC861EF.OTH.101	Power Save Mode with EnergyWise Domain Override Disallowed in Cisco Unified 99xx IP Phones and Unified IP Phones 6901/11/21/41/45/61	Verify transition from power save mode to normal mode by user and effect of EnergyWise override that is disallowed in Cisco Unified 99xx IP Phones and Unified IP Phones 6901/11/21/41/45/61.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.OTH.102	Power Save Mode with EnergyWise Domain Overrides Allowed on Unified 99xx IP Phones and Unified IP Phones 6901/11/21/41/45/61 phones	Verify transition from power save mode to normal mode by user and effect of EnergyWise override when they are allowed in Unified 99xx IP Phones and Unified IP Phones 6901/11/21/41/45/61 phones.		Passed	
UC861EF.OTH.103	Powersave Plus Mode with EnergyWise Overrides Disallowed in Unified 99xx IP Phones and Unified IP Phones 6901/11/21/41/45/61 Phones	Verify transition from power save mode to normal mode by user and effect of EnergyWise override when they are not allowed in Unified 99xx IP Phones and Unified IP Phones 6901/11/21/41/45/61 Phones.		Passed	
UC861EF.OTH.104	Powersave plus mode with EnergyWise Overrides Allowed in Unified 99xx IP Phones and Unified IP Phones 6901/11/21/41/45/61 phones	Check transition from power save mode to normal mode by user and effect of EnergyWise override when they are allowed in Unified 99xx IP Phones and Unified IP Phones 6901/11/21/41/45/61 phones.		Passed	
UC861EF.OTH.105	Effect of Cisco EnergyWise when Unified IP Phones 89XX/99XX or Unified IP Phones 6901/11/21/41/45/61 remote is in use	Verify the effect of Cisco EnergyWise on Unified IP Phones 89XX/99XX or Unified IP Phones 6901/11/21/41/45/61 remote that are in use.		Passed	
UC861EF.OTH.106	Alarms, Messages in Unified Communications Manager	Verify alarms and messages in Cisco Unified Communications Manager.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.OTH.107	Effect of EnergyWise on Computer Telephony Integration (CTI) Controlled Phones	Verify the effect of EnergyWise power save and power save plus mode on Computer Telephony Integration controlled phones.		Passed w/ Exception	In an EnergyWise case, there is no visible change to the phone status on the client in the laptop when a phone is not available. Exploring how this could be made more intuitive and why this choice was made.
UC861EF.OTH.108	Cisco 7970 IP Phone in Power Save Plus Mode	Verify the ability to check power save plus mode in Cisco 7970 IP phones.		Passed	
UC861EF.OTH.109	Effect of FirmWare Upgrades on Cisco Energywise Requests	Verify the behavior of phones when firmware upgrade is scheduled when there is a power off request from Cisco Energywise.		Passed	
UC861EF.OTH.110	Phone firmware changed when phone powered off for Cisco Energywise	Verify the phone firmware download when firmware is changed and the phone is powered off by Cisco Energywise.		Passed	
UC861EF.OTH.111	Effect of EnergyWise Requests on Virtual Desktop Integration (VDI)/ Virtualization Experience Client (VXC) Backpack and Standalone Independent Computing Architecture/PC Over IP.	Verify the effect of EnergyWise Power Save Plus Mode and EnergyWise Domain Override on Unified IP Phones 89XX/99XX used as Virtual Desktop Integration (VDI)/ Virtualization Experience Client (VXC) on backpack Independent Computing Architecture (ICA)/PC Over IP and effect of EnergyWise on Virtual Desktop Integration (VDI)/ Virtualization Experience Client (VXC) Standalone ICA/PC Over IP.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.001	PSTN Local Breakout- in the Central and in the Remote Site	Verify the PSTN local break out in the central and remote sites, and check the scenario for both IP and analog type of phones.	Central Phone1->Unified CM(E1)->PSTN->POTS Phone1; Cisco VG224 Central Phone1->Unified CM(E1)->PSTN->POTS Phone2; Remote Phone1->Unified CM->Remote1(E1)->PSTN->POTS Phone1; Remote1 Analog Ph1->Unified CM->Remote1 (E1)->PSTN->POTS Phone2;	Passed	
UC861EF.SMB.002	Multiple Gateway Support in the Central Site	Verify that Cisco Unified Communications Manager supports multiple gateways by making PSTN calls from central site, with first preference being Unified Communications Manager integrated dual E1 PRI link, followed by central gateway (2901) E1 PRI link to the PSTN network in case of a failure. Verify the scenario for both Analog and IP type of phones.	Cen Ph1->Unified CM(E1)->PSTN->POTS Ph1;Cen Ph2->Unified CM->Cen 2901Gateway(E1)->PSTN->POTS Ph3;VG224 Ph1->Unified CM(E1)->PSTN->POTS Ph3;VG224 Cen Ph2->Unified CM->Cen 2901Gateway(E1)->PSTN>POTS Ph3; Cen Ph1->Unified CM(E1)->PSTN->(E1)Unified CM->Cen Ph2	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.004	Adhoc Conference using Central Conference Phone	Verify the ability to call Cisco Unified SIP Phone 3905 at remote site from the Unified IP Conference Station 7937 at central site, place the remote site 2 Unified 6900 series IP phone 1 to the conference call, and add the POTS endpoint to the conference call by dialing the PSTN number. Verify the ability to place the remote site 3 Cisco IP Communicator to the same conference call, ensuring each remote site is using different codecs such as iLBC, G279 and G711.	Cen Unified IP Station 7937 Ph1->Unified CM->Rem1 Unified SIP 3905 Ph1;Cen Unified 7937 Ph1->Unified CM->Conf->Rem2 Unified IP 6900 Ph1; Cen Unified 7937 Ph1>Unified CM(E1)>Conf->PSTN->POTS Ph1; Cen Unified 7937 Ph1->Unified CM->Conf->Cisco IP Communicator	Passed	
UC861EF.SMB.005	Adhoc Conference involving Cisco Unified SIP Phone 3905, Unified IP Phone 6900 Series,Cisco IP Communicator and Plain Old Telephone Systems (POTS) Endpoint	Verify the Unified SIP Phone 3905 in central site calls Unified SIP Phone 3905 in remote site 1, and places the Unified SIP Phone 6911 Phone1 in Remote site 2 to the conference call, and the phone in remote site 2 place the central site Cisco IP Communicator Phone1 to the conference call. Verify that from the central site Cisco IP Communicator Phone1 places the POTS endpoint to the conference call by dialing the POTS number.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.006	Voice Mail Deposit and Retrieval Involving Cisco Unified SIP Phone 3905 Endpoint	Verify voicemail deposit and retrieval services are successful when the Plain Old Telephone Systems (POTS) phone calls central site Cisco Unified SIP Phone 3905 (with setting Call Forward No Answer to voice mail), and when the remote site 2 Unified SIP Phone 3905 endpoint calls the central site Unified IP Phone 6900 Series endpoint (with setting Call Forward No Answer to voice mail). Verify the message waiting indicator function in both the scenarios.	POTS Ph1>PSTN>E1Unified CM>Cen 3905 Ph1>CFNA>VM Unified CM;POTS Ph1>Deposit VM; Cen 3905 Ph1>Retrieve VM; Rem1 3905 Ph1>Unified CM1>Cen 69XX Ph2>CFNA>VM Unified CM;Rem 3905 Ph1>Deposit VM; Cen 69XX Ph2>Retrieve VM	Passed	
UC861EF.SMB.007	Extension mobility on Cisco IP Communicator and Unified IP Phone 6900 phone series	Verify the extension mobility on Cisco IP Communicator and Unified IP Phone 6900 phone series.	Rem Cisco IP Communicator->Unified CM->Extension Mobility;(Retrieves the Unified 69XX device profile);Rem Unified 69XX Ph->Unified CM->EM;(retrieves the CUC-RTX device profile);Cen Unified 69XX Ph->Unified CM->EM;(retrieves the CUC-RTX device profile)	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.008	Music on Hold on Cisco Unified SIP Phone 3905 Endpoint	Verify the Music on Hold service by making a call from Unified SIP Phone 3905 Phone1 in central site to Unified IP Phones 69XX Phone2 in remote site, and from Unified SIP Phone 3905, Phone1 places the call on hold. Verify Unified IP Phones 69XX Phone2 receives the music stream from Unified CM, resumes the call on Unified SIP Phone 3905 Phone1, and the Unified IP Phone 69XX Phone2 gets disconnected from the music stream and reconnects to Unified SIP Phone 3905 phone 1 in central site. Verify Music on Hold with Call park and call transfers.	Central Unified SIP Phone 3905 Ph1->Unified CM->Remote Unified IP 69XX Ph1 ;Central Unified SIP 3905 Ph1->hold; Remote Unified IP 69XX Ph1->MoH(Unified CM);Central Ph1->Resume; Central Unfiied SIP 3905 Ph1->Unified CM->Remote Unified IP 69XX Ph1;	Passed	
UC861EF.SMB.009	Emergency Call from Central Site	Verify the Emergency call (911) from central site is routed over the PSTN network by Unified Communications Manager- via its own integrated T1/E1 interface over the PSTN network and reaches the Public Safety Answering Point (PSAP) unit (Plain Old Telephone Systems endpoint) .Verify the caller ID on Plain Old Telephone Systems endpoint to ensure it contains the correct translated number(DID) of central site and check the call back from the PSTN endpoint to central site endpoint is successful.	Central Phone1->Unified CM(E1)->PSTN->PSAP(POTS endpoint);PSAP(POTS endpoint)->PSTN->(E1)Unified CM->Central Phone1;-	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.010	Emergency Call from Remote Site	Verify the Emergency call (911) from remote site is routed over the PSTN network by Cisco Unified Communications Manager via remote site and reaches the PSAP unit (POTS endpoint) .Verify the caller ID on POTS endpoint and make sure it contains the correct translated number(DID) of remote site and check the call back from the PSTN endpoint to the remote site endpoint is successful.	Remote1 Phone1->Unified CM->Remote1(E1)->PSTN->PSAP(POTS endpoint);PSAP(POTS endpoint)->PSTN->(E1)Remote 1->Unified CM->Remote Phone1;Remote3 Ph1->Unified CM(E1)->PSTN->PSAP(POTS endpoint);PSAP(POTS endpoint)->PSTN->(E1)Unified CM->Remote3 Phone1	Passed	
UC861EF.SMB.011	Plain Old Telephone Systems (POTS) endpoint calls Autoattended number and Transfers to central Unified IP Phone 6900 Series	Verify that the PSTN endpoint dials the AutoAttendant Directory Number in central site over PSTN network, the AutoAttendant requests the user to dial the extension and PSTN endpoint dials and transfers the call to the central 6900 Unified IP Phone Series endpoint. Verify the AutoAttendant handles three simultaneous incoming calls coming to Unified Communications Manager via its own integrated E1 interface, with the type of calls being different, such as the first call being from POTS endpoint in PSTN network and other two calls being VOIP calls coming from different remote sites.	PSTN Ph1->PSTN->(E1) Unified CM->AutoAttendant (Unified CM);AutoAttendant (Unified CM)->request the user to dial the extension; PSTN Ph1->dials Central Unified IP phone Series 6900 (DN);AutoAttendant (Unified CM)->Transfer->Central Unified IP 6900 Series Ph1	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.012	Auto-Attendant with Hunt grouping	Verify that the PSTN endpoint dials the Auto Attendant DN in central site over PSTN network and Auto Attendant requests the user to dial the department number, which needs to be connected (for example 1 for sales, 2 for marketing etc.,). Verify Auto Attendant transfers the call to that department (which is hunt route group of users) based on input key press, and if the first user in the hunt group is busy, then Auto Attendant transfers to second user. Verify if the incoming PSTN call connects to second user in the Hunt route group and ensures media path is established successfully.	PSTN Phone1->PSTN->(E1)Unified CM->Auto Attendant; Auto Attendant->Transfer ->Central Unified 69XX Phone1	Failed	CSCto59303 (3lR)
UC861EF.SMB.013	Remote Site Uses Centralized PSTN-Break Out	Verify the PSTN from remote site (which doesn't have local PSTN Gateway) calls the Plain old telephone systems (POTS) number, then it uses central site Unified Communications Manager E1 Internal links to connect to the PSTN network.	Remote2 Phone1->Unified CM(E1)->PSTN->POTS Phone1; POTS Phone2->(E1)Unified CM->Remote2 Phone3;	Passed	
UC861EF.SMB.014	Busy Lamp Field (BLF) Support	Verify the Busy Lamp Field (BLF) indication on central phone 1(On phone1,configure the BLF speed-dial to Remote phone 1),when remote phone 1 is busy on another call with central phone 2.	Central Phone1(Speed-dial Remote Phone1);Remote Phone1->Unified CM->Central Phone2 (Central Phone1 should have the BLF indication)	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.015	Logical Partitioning-Central site	Verify that POTS endpoint from PSTN Network (Geo-Location A) calls the phone in central site (Geo-Location A), then Unified Communications Manager should not be allowed to do the transfer of the call to Phone in remote site1 (in Geo-Location B) over Voice Over Internet Protocol network.	POTS Phone1->PSTN->(E1)Unified CM->Central Phone1->Transfer-> Unified CM->Remote Phone1 - Not allowed	Passed	
UC861EF.SMB.016	Logical Partitioning-Remote Site	Verify that POTS endpoint from PSTN Network(Geo-Location A) calls the phone in the remote site(Geo-Location B), then Unified Communications Manager is not allowed to transfer the call to central site (Geo-Location A) over Voice Over Internet Protocol network.	POTS Phone1->PSTN->(E1)Remote->Unified CM->Remote Phone1->Transfer-> Unified CM->Central Phone1;Transfer not allowed	Passed	
UC861EF.SMB.017	Toll-by-Pass	Verify that POTS endpoint from PSTN network calls the phone in the central site, and then Unified Communications Manager in central site is allowed to transfer the call to phone in remote site over VOIP network.	POTS Phone1->PSTN->(E1)Unified CM->Central Phone1->Transfer-> Unified CM->Remote Phone1; POTS Phone1->PSTN->(E1)Remote->Unified CM->Remote Phone1->Transfer-> Unified CM->Central Phone1	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.101	Failover to subscriber on Cisco Media Convergence Server (MCS) when Integrated Services Routers (ISR) hosting CiscoÂ® Service Ready Engine (SRE) with Cisco Unified Communications Manager publisher is down	Verify that the endpoints and call processing failover to subscriber on Cisco Media Convergence Server (MCS) when Integrated Services Routers (ISR) hosting CiscoÂ® Service Ready Engine (SRE) with Cisco Unified Communications Manager publisher is down.	Central SCCP Phone 1->Unified CM Publisher->Central SCCP Phone 2	Passed	
UC861EF.SMB.102	Failover to SRST ISR in central site when both Unified Communications Manager publisher and subscriber is down.	Verify if phones failover to SRST ISR in the central site when Unified Communications Manager goes down.	Central SCCP Phone 1->Unified CM->Rem SCCP Phone 1	Passed	
UC861EF.SMB.103	Conference initialization failover with central to remote RSVP.	Verify status of conference call in its initial states when call manager status goes down.	Central Phone A->Unified CM->Remote 1 Phone B->Unified CM->Conference-> Remote 2 Phone C	Passed	
UC861EF.SMB.104	Check for video escalation and de-escalation on call transfer from Unified IP Phones 99xx series to an SCCP phone between central and remote sites	Verify the ability to check for video escalation on call transfer with Central to remote RSVP.	Central Unified IP Phone 99xx series->Unified CM->Remote 1 SCCP phone->Unified CM->Transfer->Remote1 Unified IP Phone 99xx Series->Unified CM->Transfer->Remote 2 SIP Phone.	Passed	
UC861EF.SMB.105	Hold/Resume on Shared Line with Central - Remote RSVP	Verify the ability to check for hold/resume on shared line between central and remote sites with RSVP. Verify the ability to check for video escalation in supported phones.	Central Unified IP 99xx series phone 1->Unified CM->Remote 1 SCCP->Hold->Remote 1 Unified IP 89xx series phone 2 (Shared line) RESUME.	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.106	Video escalation on call forwarding over SIP Inter Cluster Trunk with End to End RSVP.	Verify video escalation over SIP Inter Cluster Trunk with End to End RSVP when call is forwarded from non-video to video phone.	Unified IP 99xx Phone 1->Unified CM 1->SIP Inter Cluster Trunk->SCCP Phone->Call Forward No Answer->Unified CM2 -> Transfer->Unified IP 99xx Phone2	Passed	
UC861EF.SMB.107	Adhoc conference with Unified IP Phone 89xx, SIP and SCCP phones in two clusters over SIP Inter Cluster Trunk.	Verify audio conference between Unified IP Phone 89xx,SIP and SCCP Phones over two clusters with End to End RSVP over SIP Inter Cluster Trunk.	Cen 89xx->Unified CM1->Remote SCCP Phone->Conference->Unified CM2->Central SIP Phone	Passed	
UC861EF.SMB.109	Early offer / Delayed offer Interworking	Verify interworking between endpoints supporting early offer and those that do not over SIP Inter Cluster Transfer.	Central 7945->Unified CM 1-> SIP Inter Cluster Trunk->Unified CM2->Unified IP phone 99xx Phone	Passed	
UC861EF.SMB.110	Trombone Path replacement	Verify Trombone path replacement in Cisco Unified Communications Manager on Cisco Services Ready Engine	Cluster 1 Phone 1->Unified CM 1 -> SIP Inter Cluster Trunk->Unified CM 2->Cluster 2 SCCP Ph->Transfer->Unified CM 1->Cluster 1 Phone 2	Passed	
UC861EF.SMB.111	Path replacement capability of Cisco Unified Communications Manager on Cisco Services-Ready Engine (SRE)	Verify path replacement on Cisco Unified Communications Manager on Cisco Services Ready Engine.	Cluster 1 Phone->Unified CM 1-> SIP Inter Cluster Trunk->Unified CM 2->Cluster 2 Phone->Transfer->SIP Inter Cluster Trunk->Unified CM 3->Cluster 3 Phone	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.SMB.112	Automated Alternate Routing when Bandwidth Unavailable between Central and Remote Site	Verify call re routing over PSTN when bandwidth is unavailable between central and branch offices.	Remote Phone1->Unified CM->Central Phone1 When bandwidth unavailable Remote Phone1->Remote PSTN Gateway->Unified CM->Central Phone 1	Passed	
UC861EF.SMB.113	Using Remote PSTN Capability by Central Phones when Central PRI Link is Down.	Verify the capability to use alternate PSTN gateways when primary PSTN gateway is unavailable.	Central SCCP Phone 1->Remote PSTN Gateway->PSTN Phone.	Passed	
UC861EF.SMB.114	Meet Me Conference Testing	Verify Meet Me Conference in Unified Communications Manager over Cisco Services-Ready Engine.	Central SCCP Phone1->Meet me Remote 1 Unified IP 99xx phone->Meet Me Remote 2 SCCP phone->Meet me	Passed	
UC861EF.VID.001	Verify BFCP (Binary Floor Control Protocol) reception and Adhoc Conference using Cisco IP Video Phone E20	Verify reception of Binary Floor Control Protocol on Cisco IP Video Phone E20 and Binary Floor Control Protocol initiation on Cisco TelePresence System EX90 which are in a adhoc conference.	Cisco TelePresence System (CTS) 500->Unified CM1->CTS EX90 CTS 500->Unified CM1->ICT->Unified CM2->Cisco IP Video Phone E20; CTS 500->Unified CM1->Conference->Codian MCU->Presentation Share->CTS EX90 and Cisco IP Video Phone E20	Passed	
UC861EF.VID.002	Cisco IP Video Phone E20 shared line with legacy endpoint	Verify Video escalation/de-escalation on Cisco IP Video Phone E20.		Passed	
UC861EF.VID.004	Cisco TelePresence ISDN Gateway 3241 Interoperability	Verify adhoc conference involving Expressway and H.320 endpoint.		Failed	CSCtn95798 CSCtq17644

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VID.005	Interoperability with Session Management Edition	Verify call transfer with Video Communication Sever endpoints and Cisco Unified Communications Manager endpoints over Session Management Edition with delayed/early offer interworking.	Unified IP Phones 8941/45->Unified CM2->SME1->SME 2->Unified CM1->Cisco IP Video Phone E20->Transfer->Video Communication Server->ISDN Gateway->H.320 Phone	Failed	CSCtn95798 CSCtq17644
UC861EF.VID.101	Point to Point native TelePresence to Unified Communications Interoperability with Video Communication Server Expressway	Verify Point to Point native TelePresence to Unified Communications interoperability with Video Communication Server Expressway.	Cisco IP Video Phone E20->VCS Expressway->Traversal Link->VCS-Control ->SIP Trunk->Unified CM->Cisco TelePresence System 500	Passed	
UC861EF.VID.102	Point to Point native TelePresence to Unified Communications Interoperability across Session Management Edition Clusters	Verify Point to Point native TelePresence to Unified Communications Interoperability across Session Management Edition clusters.	Cisco IP Phone->Unified CM->SIP Trunk->SME 1->SIP->SME 2->Unified CM->Cisco TelePresence System500	Passed	
UC861EF.VID.103	Tandberg Single Stream High-Definition (HD) and High Definition-Standard Definition Interoperability fixes with Tandberg 550	Verify Tandberg single stream High-Definition (HD) and HD-SD interoperability fixes with Tandberg 550.	Tandberg 550->H.323->Video Communication Server->SIP Trunk->Unified CM->Cisco IP Phone	Passed	
UC861EF.VID.104	Tandberg Single Stream High-Definition (HD) and High Definition-Standard Definition Interoperability Fixes with Cisco TelePresence EX90	Verify Tandberg single stream High-Definition (HD) and High Definition-Standard Definition interoperability fixes with Cisco TelePresence EX90.	Cisco TelePresence EX90->H.323->Video Communication Server->SIP Trunk->Unified CM->Cisco IP Phone	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VID.105	Presentation share between Cisco TelePresence and Tandberg Endpoints	Verify Presentation share between Cisco TelePresence and Tandberg endpoints.	Cisco TelePresence System 500->Unified CM->SIP Trunk->Video Communication Server->Cisco TelePresence EX90	Passed	
UC861EF.VID.106	Interoperability Testing of Tandberg and Cisco Unified IP Phone 8941 Series Phones	Verify interoperability testing of Tandberg and Cisco Unified IP Phone 8941 series phones.	Cisco IP Phone->Unified CM->SIP Trunk->Video Communication Server-Control->Traversal Link->Video Communication Server Expressway->Cisco IP Video Phone E20	Passed	
UC861EF.VID.107	SIP Wideband Audio Codec Support	Verify SIP wideband audio codec support.	Cisco IP Phone->Unified CM->SIP Trunk->Video Communication Server->H.323->Tandberg 550	Passed	
UC861EF.VID.108	Calls between Cisco TelePresence System 500 Phones across SIP Inter Cluster Trunks	Verify calls between Cisco TelePresence System500 phones across SIP Inter cluster trunks.	Cisco TelePresence System 500->Unified CM 1->SIP Inter Cluster Trunk->Unified CM 2->Cisco TelePresence System 500	Passed w/ Exception	One Cisco TelePresence System 500 endpoint was replaced with Cisco TelePresence System 1000 as the codec had issues.

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VID.201	Intra Cluster Adhoc Multi-Point Conference among Native Unified Communications endpoints and Cisco TelePresence System 500	Verify Adhoc multipoint conference among Unified IP phones 9971, Unified IP Phones 8941/45, Cisco Unified Communications Integration for RTX, Cisco TelePresence System EX90, H320 PSTN and Cisco TelePresence System 500 endpoints is successful. Verify the resources are released after the conference and repeat the scenario with various native Unified communications endpoints.	Step1)UC integration @for MOC->Unified CM->Cisco TelePresence System 500 Step2) UC integration @for MOC->Unified CM->UC integration @for RTX Step3)UC integration @ for MOC->Unified CM->Conference-> Codian MCU->CTS 500 and UC integration @for RTX	Passed	
UC861EF.VID.202	Inter Cluster Adhoc Multi-Point Conference among Native Cisco Unified Communications endpoints and Cisco Telepresence Endpoint	Verify Adhoc multipoint conference among CTS 500, CUPC, CUCIMOC, 9971 and 8941 successful. Verify the resources are released after the conference.	Step 1)9971->CUCM1-> SIP ICT->CUCM2->CTS500 Step 2)9971->CUCM1-> H225 ICT->CUCM2->8941; Step 3)9971->CUCM1-> CUPC; Step 4)9971->CUCM1-> CUCIMOC ; Step 6)9971->CUCM1-> CONF->Codian MCU->9971 & 8941 &CUPC &CUCIMOC&CTS 500	Passed w/ Exception	CSCtq74688
UC861EF.VID.204	Adhoc Multipoint Presentation share among Cisco TelePresence System 500 ,Cisco TelePresence SystemEX90 , Cisco IP Video Phone E20	Verify the Presentation share among Cisco TelePresence System 500,Cisco TelePresence System EX90 and Cisco 9971 video phone is successful	Cisco TelePresence System (CTS) 500->Unified CM1->CTS EX90 CTS EX90->Unified CM1->9971; CTS EX90->Unified CM1->Conference->Codian MCU->Presentation Share->CTS 500 and 9971	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VID.205	Adhoc Multipoint Conferencing using SIP 4501 MCU among Cisco UC Integration(TM) for Microsoft Office Communicator, Unified IP phone 6961,Unified IP Phones 8941/45 and Cisco TelePresence System 500 Endpoint	Verify Adhoc Multipoint conferencing among Unified IP Phone 6961, Cisco UC Integration(TM) for Microsoft Office Communicator, Unified IP Phones 8941/45 and Cisco TelePresence System 500 endpoints.	CUCI-MOC>UnifiedCM1>Unified IP Phone 8945;CUCI-MOC> UnifiedCM1>ICT-> UnifiedCM2>9971; MOC>UnifiedCM1 >CUCI-RTX;CUCI-MOC>UnifiedCM1->ICT>UnifiedCM2> 7985;CUCIMOC>Unified CM1>H.320 PSTN;CUCIMOC> Unified CM1>Conference> Codian MCU>8945 9971 7985 H.320 PSTN&CUCI-RTX	Failed	CSCtq17644
UC861EF.VID.206	Secure SIP Video Signaling between Cisco Telepresence System 500 and Tandberg	Verify the call from H.323 endpoint registered to Cisco TelePresence Video Communication Server (VCS) to the Cisco TelePresence System 500 through SME cluster is a non-Secure call.	Step1: Cisco TelePresence 1700 MXP (H.323)->Video Communication Server-Secure SIP->SME1->Non-secure SIP Trunk->SME2->Unified CM2->Cisco TelePresence System 500	Failed	CSCtq17644
UC861EF.VID.301	Tandberg Basic High Definition Video Interoperability: Validate Video Communication Server, Tandberg Codian and Expressway Deployment Scenarios(Reservationless Conference)	Verify that different Unified Communications endpoints are able to participate in video conferencing, and Cisco ISR-G2 is provisioned with PVDM3 (High-Density Packet Voice Digital Signal Processor Module) DSP modules to use with Cisco Unified Communications Manager for video conferencing purposes.	Video Ph1>WAN>Unified CM1>Rem VCB>Meet-me DN Variation:Vid Ph1>Rem Unified IP Ph 9971: 9951:CUPC: CUCIMOC: CUCIRTX: Unified IP Ph 7985: Cisco IP Vid Ph E20(SIP): Tandberg Ph1>Tandberg VCS>SME1>SIP Trunk->SME2>SIP Trunk>Unified CM1>Rem VCB->Meet-me DN	Failed	CSCtq17644

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VID.302	Tandberg Basic High Definition Video Interoperability: Validate Video Communication Server, Tandberg Codian and Expressway Deployment Scenarios(Adhoc Conference)	Verify that different Unified Communications endpoints are able to participate in video conferencing, and Cisco ISR-G2 is provisioned with PVDM3 (High-Density Packet Voice Digital Signal Processor Module) DSP modules to use with Cisco Unified Communications Manager for video conferencing purposes.	CentVidPh1->Unified CM1->Cen VCB->Unified CM1->9971->CNF->TandPh1:Variation: CentVidPh1->RT9971:9951:CUPC:CUCIMOC:CUCIRTX: 7985:E20(SIP):EX90(SIP):MXP1700(SIP): Tandberg1000(SCCP):Variation:TandPh1->TandMXP1700(SIP):E20(SIP):EX90(SIP):Tandeberg1000 (SCCP)	Passed	
UC861EF.VID.303	Cisco ISR-G2 provisioned with PVDM3 (High-Density Packet Voice Digital Signal Processor Module) DSP modules to use with Cisco Unified Communications Manager for video conferencing purposes Connectivity to Cisco Unified Communications Manager for Adhoc Switched Video Conference in Unified Communications	Verify that different UC endpoints are able to participate in video conferencing, and Cisco ISR-G2 is provisioned with PVDM3 (High-Density Packet Voice Digital Signal Processor Module) DSP modules to use with Cisco Unified Communications Manager for video conferencing purposes.	VidPh1>UnifiedCM1>Rem VCB>UnifiedCM1> Unified IP Phone 9951>CNF>VidPh2: Variation:VidPh1>Rem Unified IP Phone 9971:9951:CUPC:CUCIMOC:CUCIRTX:7985:E20(SIP):Variation:VidPh2>9971:9951:CUPC:CUCIMOC:CUCIRTX:7985:E20(SIP):EX90(SIP):MXP1700(SIP): Tandberg1000 SCCP	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VID.304	Cisco ISR-G2 provisioned with PVDM3 (High-Density Packet Voice Digital Signal Processor Module) DSP modules to use with Cisco Unified Communications Manager for video conferencing purposes in Tandberg Cisco Unified Communications Interoperability Support(Reservationless Conference)	Verify that different Unified Communications endpoints are able to participate in video conferencing, and Cisco ISR-G2 is provisioned with PVDM3 (High-Density Packet Voice Digital Signal Processor Module) DSP modules to use with Cisco Unified Communications Manager for video conferencing purposes.	VidPh1->Unified CM1->Cent VCB->Meet-me-DN :Variation:Unified IP Phone 9971:9951:CUPC:CUCIMOC: CUCIRTX:7985:EX90(SIP):MXP1700(SIP):Tandberg1000(SCCP):STEP2:Tandberg Ph1->TandVCS->SIP Trunk->Unified CM1->Cent VCB->Meet-me-DN	Passed	
UC861EF.VXC.001	UC Integration@(Deskphone) for RTX Inter-cluster Video Call to Third Party Skinny Endpoint.	Verify if UC Integration@ for RTX(Deskphone) 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) Inter Cluster Trunk->Unified CM2->SCCP video endpoint	Passed	
UC861EF.VXC.002	Escalation of Audio Call to Video Call in Softphone mode	Verify if UC Integration@(Softphone) for RTX from one cluster can make an audio call to another UC Integration@ for RTX client in deskphone mode running on voicemail and then can escalate to video.	UC Integration@ for RTX(softphone)->Unified CM1->UC Integration@ for RTX (deskphone)	Passed	
UC861EF.VXC.003	Escalation of Audio Call to Video Call in Deskphone Mode When Calling Cisco Unified Communications Integration @ for Microsoft Office Communicator in Another Cluster	Verify if Cisco Unified Communications Integration@(deskphone) for RTX can make an inter-cluster call to Cisco Unified Communications Integration @ for Microsoft Office Communicator and can escalate the audio call to video call.	UC Integration@ for RTX->Unified CM1->SIP(QSIG) Inter Cluster Trunk->Unified CM2->UC Integration@ for MOC	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VXC.004	Inter-cluster Video Conference with Cisco Unified Communications Integration@ for RTX and Unified IP Phone 9900/8900 series	Verify if Cisco Unified Communications Integration@(Softphone) for RTX can make an inter-cluster video call to another UC Integration@ for RTX user in deskphone mode and can join a Unified IP Phones 89XX/99XX phone in another cluster to the conference.	UC Integration@ for RTX->Unified CM1->Annex M1 Inter Cluster Trunk->Unified CM2->Unified IP Phones 89XX/99XX	Passed	
UC861EF.VXC.005	Hold/Retrieve from Shared Line with Cisco Unified IP Phone 8900/9900 Series in Another Cluster	Verify if Cisco Unified Communications Integration@(deskphone) for RTX can make an inter-cluster video call to a Cisco Unified IP phone 8900/9900 series and can put the call on hold and retrieve it from a shared line.	UC Integration@ for RTX->Unified CM1->SIP(QSIG) Inter Cluster Trunk->Unified CM2->Unified IP phone 8900/9900	Passed	
UC861EF.VXC.006	Fall Back to Unified Survivable Remote Site Telephony when Unified Communications Manager goes down	Verify if Cisco Unified Communications Integration@ for RTX registers to Unified Survivable Remote Site Telephony 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	
UC861EF.VXC.007	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	
UC861EF.VXC.008	Third party H.323 Endpoint with Gatekeeper Video Call to Cisco Unified Communications Integration@(Deskphone) for RTX	Verify if third party H.323 endpoint with gatekeeper can make a video call to Cisco Unified Communications Integration@(Deskphone) for RTX.	H.323 video endpoint->Unified CM->UC Integration@ for RTX	Passed w/ Exception	Audio calls are working fine. Since Video call needs a video transcoder which was not there in that cluster.

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VXC.009	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 UC Integration@ for RTX (DeskPhone) and transfer the call to IP Communicator and Unified Video Advantage.	UC Integration@ for RTX (SoftPhone)->Unified CM1->Annex M1 Inter Cluster Trunk->Unified CM2->UC Integration@ for RTX(Deskphone)->Transfer->Inter Cluster Trunk->Cisco IP Communicator+ Cisco Unified Video Advantage	Passed	
UC861EF.VXC.010	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@ (Deskphone) for RTX and then transfer the call to UC Integration@ for MOC.	Remote UC Integration@(Softphone) for RTX->Inter Cluster Trunk-> UC Integration@ (Deskphone) for RTX->Transfer->Central UC Integration@ for MOC	Passed	
UC861EF.VXC.011	Central Site Unified Personal Communicator 8.0 Client Video Call to Cisco Unified Communications Integration@(Softphone) for RTX in Remote Site	Verify if the central site Unified Personal Communicator 8.0 client can make a video call to Cisco Unified Communications Integration@(Softphone) for RTX in remote site and then consult transfer the call to Cisco Unified Communications Integration@ for RTX in deskphone mode.	Central Excession->Unified CM->UC Integration @ for RTX (Remote)->Transfer C->UC Integration @ for RTX (Deskphone)	Passed	
UC861EF.VXC.012	Cisco Unified Communications Integration@(Softphone) for RTX Call to PBX Phone in Interoperability Site	Verify if Cisco Unified Communications Integration@(Softphone) for RTX can make a call to a PBX phone in interoperability site.	UC Integration @ for RTX->Unified CM1->SIP Inter Cluster Trunk(QSIG)->Unified CM2->QSIG Trunk->PBX phone	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VXC.013	Audio Conference with Q Interface Signalling Protocol (QSIG) Private Branch Exchange (PBX) and PSTN Phones.	Verify if UC Integration@(Deskphone) for RTX can make a conference call with PSTN and QSIG PBX phone.	UC Integration@ for RTX->Unified CM->PSTN Gateway->PSTN->Conference->QSIG Trunk->PBX phone	Passed	
UC861EF.VXC.014	Cisco Unified Communications Integration@ for RTX Failover to PSTN When WAN is Down.	Verify if Cisco Unified Communications Integration@ for RTX calls go through PSTN to remote site when there is insufficient bandwidth.	UC Integration @ for RTX->Unified CM->MGCP PRI Gateway->PSTN->Remote SCCP Phone1	Passed	
UC861EF.VXC.015	Escalation of Audio Call to Video When Call is Transferred to a Video Phone	Verify if UC IntegrationT for RTX in remote site can call a central SCCP phone and when the call gets transferred to UC IntegrationT for MOC, then if two way video is established.	UC Integration@ for RTX (remote)->Unified CM->SCCP Phone 1->Transfer->UC Integration@ for MOC	Passed	
UC861EF.VXC.016	PSTN Call to H.320 Endpoint from Cisco Unified Communications Integration@ for RTX	Verify if Cisco Unified Communications 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	Failed	CSCtq17644
UC861EF.VXC.017	Inter-Cluster Call over Cisco IME	Verify if Cisco Unified Communications Integration@ for RTX can make an Intercluster Call to a Cisco IP Phone 7985 over Cisco IME.	UC Integration @ for RTX->Unified CM 1->Adaptive Security Appliances->Cisco IME Trunk->Adaptive Security Appliances->Unified CM 2->Cisco Unified IP Phone 7985	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861EF.VXC.018	Intercluster Video Call over Cisco IME After Transfer from Cisco Unified Communications Integration@ for Microsoft Office Communicator	Verify if Cisco Unified Communications Integration@ for Microsoft Office Communicator can make an inter-cluster call to an SCCP phone which is then transferred to Cisco Unified Communications Integration@ for RTX in remote branch.	UC Integration@ for MOC->Unified CM1->Adaptive Security Appliance->Cisco IME Trunk->Adaptive Security Appliance->Unified CM2->SCCP phone1->Transfer-> Cisco IME trunk->Adaptive Security Appliance->Unified CM1->Remote Branch->UC Integration @ for RTX	Passed	
UC861EF.VXC.019	Cisco Unified Communications Integration@ for RTX Video Call Between Remote Sites	Verify if Cisco Unified Communications Integration@ for RTX in a remote branch can make a video call to Cisco Unified Communications Integration@ for RTX in another remote site.	UC Integration @ for RTX (Remote1)->Unified CM->UC Integration @ for RTX (Remote 2)	Passed w/ Exception	Made a call from remote Phone to UC Integration @ for RTX running in VoiceMail to cover this scenario.
UC861EF.VXC.020	Inter-Cluster Adhoc Video Conference with Cisco IP Communicator, Cisco Unified Video Advantage and Third Party H.323 Endpoint.	Verify if UC IntegrationT(Deskphone) for RTX can take part in an inter-cluster Ad-hoc video conference with IP Communicator and Unified Video Advantage and third party H.323 endpoints.	Cisco IP Communicator+Unified Video Advantage->Unified CM->UC Integration @ for RTX->Conference-> H.323 video endpoint	Passed w/ Exception	Executed the below mentioned callflow: Cisco IP Communicator Cisco Unified Video Advantage->Unified CM->QSIG Inter Cluster Trunk-> UC Integration @ for RTX->Conference->SIP Inter Cluster Trunk -> H.323 video endpoint

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CER.001	Make sure Phone in "Power Save Plus" mode is still tracked by Cisco Emergency Responder	Verify that Cisco Emergency Responder continues to track the location of phone in "Power Save Plus" mode.	Phone->Switch->Cisco Emergency Responder	Passed	
UC861IF.CER.002.1	Unified IP Phones 99XX series Out of Power Save Plus Mode Make 911 Calls that is Routed to nearest PSAP	Verify the ability to make sure Unified IP Phone 99XX series coming out of Power Save Plus mode can make 911 calls and the call is routed to nearest PSAP.	Phone->Switch->Unified CM->Cisco Emergency Responder->Unified CM->Gateway->PSAP	Passed	
UC861IF.CER.002.2	Unified 69XX Series IP Phone Out of Power Save Plus Mode Make 911 calls that is routed to nearest PSAP	Verify the ability to ensure that Unified 69XX series IP phone coming out of Power Save Plus mode can make 911 calls and the call is routed to nearest PSAP.	Phone->Switch->Unified CM->Cisco Emergency Responder->Unified CM->Gateway->PSAP	Passed	
UC861IF.CER.002.3	Cisco Unified 79xx series IP Phone Out of Power Save Plus Mode Make 911 calls that is routed to nearest PSAP	Verify that 79XX series phone coming out of Power Save Plus Mode can make 911 calls and the call is routed to nearest PSAP.	Phone->Switch->Unified CM->Cisco Emergency Responder->Unified CM->Gateway->PSAP	Passed	
UC861IF.CER.003.1	Cisco Unified 99XX IP Phone Series in Power Save Plus Mode moved to another switch in the same Unified CM cluster Makes 911 call after waking up from Power Save Plus Mode	Verify that Unified 99XX series IP Phone in power save plus mode moved to another switch in the same Cisco Unified Communications Manager cluster can make 911 call after waking up from power save plus mode.	Phone->Switch->Unified CM->Cisco Emergency Responder->Unified CM->Gateway->PSAP	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CER.003.2	Cisco Unified 69XX Series IP phone in Power Save Plus Mode moved to another switch in the same Cisco Unified Communications Manager cluster can make 911 call after waking up from Power Save Plus Mode	Verify that Unified 69XX series IP phone in Power Save Plus mode moved to another switch in the same Cisco Unified Communications Manager cluster can make 911 call after waking up from Power Save Plus mode.	Phone->Switch->Unified CM->Cisco Emergency Responder->Unified CM->Gateway->PSAP	Passed	
UC861IF.CER.003.3	Cisco Unified 79XX series IP Phone in Power Save Plus Mode Moved to Another Switch in the Same Cisco Unified Communications Manager Cluster Make 911 Call after waking up from Power Save Plus Mode	Verify that Cisco Unified 79XX series IP Phone in Power Save Plus mode moved to another switch in the same Cisco Unified Communications Manager cluster can make 911 call after waking up from Power Save Plus Mode.	Phone->Switch->Unified CM->Cisco Emergency Responder->Unified CM->Gateway->PSAP	Passed	
UC861IF.CER.004.1	Cisco Unified 99XX Series IP Phone in Power Save Plus Mode Moved to Another Switch in the Different Cisco Unified Communications Manager Cluster make 911 Call after Waking up from Power Save Plus Mode.	Verify that Cisco Unified 99XX series IP phone in Power Save Plus mode moved to another switch in a different Cisco Unified Communications Manager cluster can make 911 call after waking up from Power Save Plus mode.	Phone->Switch->Unified CM->Cisco Emergency Responder->Unified CM->Gateway->PSAP	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CER.004.2	Cisco Unified 69XX series IP Phone in Power Save Plus mode moved to another switch in the different Cisco Unified Communications Manager cluster can make 911 call after waking up from Power Save Plus mode	Verify that Cisco Unified 69XX series IP phone in Power Save Plus mode moved to another switch in the different Cisco Unified Communications Manager cluster can make 911 call after waking up from Power Save Plus mode.	Phone->Switch->Unified CM->Cisco Emergency Responder->Unified CM->Gateway->PSAP	Passed	
UC861IF.CER.004.3	Cisco Unified 79XX series IP Phone in Power Save Plus Mode Moved to Another Switch in the different Cisco Unified Communications Manager cluster make 911 Call after Waking up from Power Save Plus Mode	Verify that Cisco Unified 79XX series IP Phone in Power Save Plus mode moved to another switch in the different Cisco Unified Communications Manager cluster can make 911 call after waking up from Power Save Plus Mode.	Phone->Switch->Unified CM->Cisco Emergency Responder->Unified CM->Gateway->PSAP	Passed	
UC861IF.CUS.016	Message Actions with Visual Voice Mail on Cisco Cius	Verify the message actions with Visual Voice Mail on Cisco Cius.	Audio Phone->Unified CM1->Cisco Cius Call Forward No Answer->Cisco Unity Connection Voicemail	Passed w/ Exception	CSCtq13847

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.017	Cisco Unity Connection in cluster goes down while playing Visual Voice Mail message on Cisco Cius	Verify that the Cisco Unity Connection in a cluster goes down while playing Visual Voice Mail message on Cisco Cius.	Audio Phone->Unified CM1->Cisco Cius Call Forward No Answer->Cisco Unity Connection Voicemail	Passed w/ Exception	Provide a DNS name that resolves to both the server names for Visual Voice Mail to failover to secondary server. Unified CM setting 'Secondary Voicemail Server' setting does not work currently.
UC861IF.CUS.018	Download Voicemails to Cisco Cius from Cisco Unity Connection Server2 When Server1 is Down	Verify the ability to download Voicemails to Cisco Cius from Cisco Unity Connection Server2 when Server1 is down.	Cisco Cius->Cisco Unity Connection Cluster Server2	Passed w/ Exception	Provide a DNS name that resolves to both the server names for Visual Voice Mail to failover to secondary server. Unified CM setting 'Secondary Voicemail Server' setting does not work currently.
UC861IF.CUS.060	Mutiway conference from Video Communication Server endpoint to Cisco CIUS and Cisco Unified IP Phone 9971 behind Cisco Unified Communications Manager	Verify whether Cisco CIUS is able to join multiway conference with Cisco Video Communication Server endpoint.	Cisco Telepresence Quickset C20-Cisco Video Communication Server-SIP Trunk ---Unified CM ----Unified IP Phone 9971 ---Cisco Telepresence Quickset C20(multiway)--SIP Trunk-Unified CM-Cisco CIUS	Passed	
UC861IF.CUS.062	Verify Ability to Switch Between Shared Lines with Video	Verify a Cisco Cius with two lines (one shared line) is able to switch between two active calls with video.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.063	Device Mobility with Cisco Cius	Verify that device mobility works with Cisco Cius.	Cisco Cius SRST location -->Unified CM -->Conference -->IP Phone1 and IP Phone2	Passed	
UC861IF.CUS.066	Cisco Cius Interoperability with Cisco Unified Meeting Place	Verify whether Cisco Cius is able to join Cisco Unified Meeting Place meeting and view the video of all the participants.	Cisco Cius ----Unified CM ---- SIP Trunk ---- -Cisco Unified Meeting Place	Passed	
UC861IF.CUS.098	Cisco Cius can have a voicemail box in Cisco Unity Express	Verify if Cisco Unity Express can provide voicemail service for Cisco Cius.	Ph1-->Unified CM -->SIP Trunk -->Unified CM -->Cisco Cius-->Call Forward No Answer-->Unified CM-->JTAPI -->Cisco Unity Express	Passed	
UC861IF.CUS.201	Point to Point call from Cisco IP Video Phone E20 registered to Video Communication Server to Cisco Cius registered to Cisco Unified Communications Manager	Verify the video call can be placed and put on hold by Cisco Cius registered to Cisco Unified Communications Manager and the call resumes back to video.	Cisco Cius --- Abilene Unified CM --- SIP Trunk ---Cisco IP Video Phone E20->Video Communication Server	Passed	
UC861IF.CUS.202	Call transfer between Cisco Cius and Cisco TelePresence Quick Set C20	Verify video call can be transferred from Cisco Cius registered to Cisco Unified Communications Manager to Cisco TelePresence Quick Set C20 registered to Video Communication Server.	Cisco Cius-> Abilene Unified CM->SIP trunk->Video Communication Server->Cisco TelePresence Quick Set C20->Cisco Cius Transfer ---SIP trunk --- Cisco TelePresence 1700 MXP->Video Communication Server	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.203	Conference Cisco Unified Communications Manager and Cisco Video Communication Server Endpoints using Cisco MeetingPlace Software Bridge	Verify conference can be established between Cisco CIUS, Cisco IP video Phone E20 and Tandberg MXP 1700 series using Cisco MeetingPlace Adhoc bridge.	Cisco CIUS-->MSP Unified CM--H.225 trunk--GateKeeper->Cisco Video Communication Server--Cisco IP Video Phone E20--conference using Cisco CIUS--Tandberg MXP 1700->H.323->Cisco Video Communication Server	Passed w/ Exception	Bad Video Quality-Known Issue
UC861IF.CUS.204	Conference with Cisco TelePresence Video Communication Server and Cisco Cius Endpoints via Cisco Media Experience Engine and Cisco TelePresence Multipoint Switch	Verify conference between Cisco TelePresence Video Communication Server and Cisco TelePresence System 1000 via Cisco Media Experience Engine and Cisco TelePresence Multipoint Switch.	Cisco IP Video Phone E20->Cisco VCS---SIP trunk----SME ---SIP trunk->Cisco MXE->Unified CM--Cisco TelePresence Multipoint Switch-> Conference ---Cisco Cius->Unified CM---Cisco MXE---SME---SIP Trunk---Cisco TelePresence Multipoint Switch Conference	Passed	
UC861IF.CUS.205	Scheduled Conference using Codian Multipoint Control Unit	Verify whether Cisco IP Video Phone E20, Cisco TelePresence Quick Set C20 registered to Video Communication Server and Cisco Cius registered to Cisco Unified Communications Manager are able to join Scheduled conference using Codian Multipoint Control Unit bridge.	Cisco TelePresence Quick Set C20 Cisco Cius Cisco IP Video Phone E20 ---- H.323 Destination Number --- Codian MCU	Passed w/ Exception	Bad Video Quality-Known Issue

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.206	Adhoc Conference Using Cisco TelePresence MCU and Cisco TelePresence Video Communication Server Multiway Feature	Verify if the endpoints can join the conference bridge using multiway feature in an Adhoc Conference Using Cisco TelePresence MCU and Cisco TelePresence Video Communication Server Multiway Feature.		Passed	
UC861IF.CUS.207	Inter-cluster Cisco Cius Peer-to-Peer call with Cisco TelePresence System through QSIG enabled SIP Trunks With End-to-End RSVP	Verify that Cisco Cius end point can make Peer-to-Peer calls to Cisco TelePresence System with End-to-End RSVP enabled with supplementary services.	Cisco Cius->Cisco Call Manager1->SIP Trunk (QSIG)->Cisco Call Manager2->SIP Trunk (QSIG)->Cisco TelePresence System	Passed	
UC861IF.CUS.208	Inter-cluster Secure Cisco Cius Peer-to-Peer Call with Cisco TelePresence System	Verify if the secure Cisco Cius end point can make Peer-to-Peer calls with Secure Cisco TelePresence System	Cisco Cius->Cisco Call Manager1->SIP Trunk(QSIG)->Cisco Call Manager2->SIP Trunk(QSIG)->Cisco TelePresence System	Passed	
UC861IF.CUS.209	Inter-cluster Cisco Cius native interoperability Peer-to-Peer call with Cisco TelePresence System with Trusted Relay Points Enabled	Verify native interoperability between Cisco Cius and Cisco TelePresence System with Trusted Relay Points enabled.	Cisco Cius->Cisco CallManager1->SIP Trunk->Cisco CallManager2->SIP Trunk->Cisco TelePresence System	Passed	
UC861IF.CUS.210	Cisco CIUS native interoperability with Cisco Telepresence System when in Wireless mode	Verify test video interoperability with Cisco TelePresence System when Cisco CIUS is operating in wireless mode.	Cisco CIUS->Cisco CallManager1->SIP Trunk->Cisco CallManager2->SIP Trunk->Cisco TelePresence System	Passed w/ Exception	Bad Video Quality- Known Issue
UC861IF.CUS.211	Cisco Cius Video Call Preservation	Verify that test video call stays up when the Cisco Unified Communications Manager that Cisco Cius is registered to, goes down.	Cisco Cius->Cisco CallManager1->SIP Trunk->Cisco CallManager2->SIP Trunk->Cisco TelePresence System	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.502	Inter-cluster Instant Messaging interoperability with Cisco Unified Personal Communicator 7 user, multiple Instant Messaging sessions	Verify the ability to begin an Instant Messaging chat with a Cisco Unified Personal Communicator 7 user, and when the chat is ongoing, initiate an Instant Messaging with the Cisco CIUS user from another Cisco Unified Personal Communicator 7 client. Verify messages are properly exchanged between the two clients and that CIUS is able to handle the multiple sessions.	Client Services Framework->Cisco Unified Presence->WAN->Cisco Unified Presence->Cisco Unified Personal Communicator 7	Passed	
UC861IF.CUS.503	Inter-cluster Instant Messaging Interoperability with Cisco Unified Personal Communicator 8 user sending offline messages	Verify the ability to begin an Instant Messaging chat with a Cisco Unified Personal Communicator 8 user that is initially not logged in (presence status is unavailable). Verify when the user logs in offline messages are received, and exchanges Instant Messages between the two clients. Verify interoperability.	Client Service Framework->Cisco Unified Presence->WAN->Cisco Unified Presence->Cisco Unified Personal Communicator 8	Passed	
UC861IF.CUS.504	Cisco CIUS client receiving Instant Messages from Cisco CIUS in a different time zone	Verify the ability to send Instant Messages to a Cisco CIUS user from a user in another time zone. Verify Instant Messages are displayed with correct time stamps adjusted properly for the current time zone.	Cisco Unified Personal Communicator->Cisco Unified Presence->Wide Area Network->Cisco Unified Presence->Cisco CIUS	Passed	
UC861IF.CUS.507	Cisco Unified Presence Service Outage	Verify that Cisco CIUS is able to recover on failing the Cisco Unified Presence services and Cisco Unified Presence network connectivity while the Cisco CIUS client logs in.	Cisco CIUS->LAN->Cisco Unified Presence	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.508	Auto on the Phone Presence with Shared Line	Verify Cisco CIUS auto switches its self-presence to "on the phone" when a shared line phone goes off-hook.	Cisco CIUS->Unified CM->Phone	Passed	
UC861IF.CUS.509	Add Instant Messaging Session to Active Inter-cluster Phone Call	Verify the ability to add an Instant Messaging session to the phone call during an active call session with a user in another cluster.	Cisco Cius->Unified CM1->SIP trunk->Unified CM2->User2	Passed	
UC861IF.CUS.510	Auto in a Meeting Presence	Verify Cisco Unified Presence is configured to use calendar integration. Verify when a Cisco CIUS user joins a meeting listed on their calendar that the Cisco CIUS client's presence is changed to "In a Meeting".	Cisco CIUS->Cisco Unified Presence->MicroSoft Exchange	Passed	
UC861IF.CUS.601	Call preservation when primary Cisco Unified Communications Manager goes down, Cisco Cius registers with Clustering over WAN (CoW) Backup Node	Verify that the active call remains preserved and the Cisco Cius is successfully able to register to the secondary node when the primary Cisco Unified Communications Manager that Cisco Cius is registered to during an active call fails, and the secondary node that Cisco Cius registers to is located over the WAN.	Cisco Cius->Unified CM1->SIP Trunk->Unified CM2->Cisco Unified IP Phone 7975; After failover Cisco Cius->WAN->Adaptive Security Appliance->Backup Unified CM->SIP trunk->Unified CM2->Cisco Unified IP Phone 7975	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.602	Cisco Cius Operating in Cisco Survivable Remote Site Telephony Mode, Initiating Adhoc Conference	Verify Cisco Cius registers to the Cisco Survivable Remote Site Telephony router, places a call to the PSTN, to another phone within the branch site and conferences the two calls given that Cisco Cius in a branch site is registered to Cisco Unified Communications Manager, and the WAN link breaks and the site falls back to Cisco Survivable Remote Site Telephony mode.	Cisco Cius->Cisco Survivable Remote Site Telephony router->PSTN; Cisco Cius->Cisco Survivable Remote Site Telephony router->Phone2	Passed	
UC861IF.CUS.603	Cisco Cius calling over End Office SIP Trunks via Session Manager Edition, Consultative Transfer	Verify Cisco Cius registers to the Cisco Survivable Remote Site Telephony Router on placing a call to the PSTN and to another phone within the branch site and conferencing the two calls, given that Cisco Cius in a branch site is registered to Cisco Unified Communications Manager and the WAN link breaks and the site falls back to Cisco Survivable Remote Site Telephony mode.	Cisco Cius->Cisco Survivable Remote Site Telephony router->PSTN; Cisco Cius->Cisco Survivable Remote Site Telephony Router->Phone2	Passed	
UC861IF.CUS.604	Cisco Cius calling over H.323 trunks via Cisco Unified Communications Session Manager Edition (G.722), call transferred to Cisco Survivable Remote Site Telephony site (G.729), Hold/Resume	Verify audio codec renegotiation when Cisco Cius initially calls a phone in another cluster via Cisco Unified Communications Session Management Edition H.323 trunks, given that the audio codec negotiated is G.722 and the far side transfers the call to a branch site using G.729.	Cisco Cius->Unified CM1->H.323 trunk->SME->H.323 trunk->Unified CM2->Unified IP Phones 89XX/99XX; Transfer->TNP phone	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.605	Cisco CIUS call over SIP Gateway, far side rolls over to Unity Connection and leave a Voicemail	Verify the ability to place a call over SIP Gateway to another enterprise. Verify if the call rolls over to Unity Connection on the far side, and Cisco CIUS leaves a voicemail.	Cisco CIUS->Unified CM1->SIP Trunk->Cisco IME inline ASA->Cisco IME offpath ASA->SIP Trunk->Unified CM2->Phone; Transfer->SIP Trunk->Unity Connection	Passed	
UC861IF.CUS.606	Cisco CIUS attending webex conference meeting	Verify that Cisco CIUS is able to attend webex conference meeting and transfers the call to mobile phone, and continues the call from mobile phone.	Cisco CIUS->Unified CM1->SIP Trunk->Cisco IME inline ASA->Cisco IME offpath ASA->SIP Trunk->Unified CM2->Phone; after fallback; Cisco CIUS->Unified CM1->SIP Trunk->PSTN Gateway->PSTN->PSTN Gateway->Unified CM2->Phone	Passed	
UC861IF.CUS.607	Cisco Cius as a local RSVP-enabled endpoint over RSVP enabled SIP Inter Cluster Trunk	Verify local RSVP is invoked and media terminates from Cisco Cius to the RSVP agent on placing a call from CIUS over a SIP trunk requiring RSVP reservations.		Passed	
UC861IF.CUS.608	Cisco Cius as an End-to-End RSVP-enabled Endpoint, Direct Transfer to Call on Hold	Verify the ability to place a call from Cisco Cius over a SIP trunk with End-to-End RSVP reservations and have another incoming call come to Cisco Cius and to place the call from Cisco Cius on hold and answer the other call. Verify the ability to resume the other call, and then perform a direct transfer to connect call A to call B.	Cisco Cius->Unified CM->SIP trunk->Unified CM->Phone; media flows through RSVP agents	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.609	Cisco Cius placing 911 call via Cisco Emergency Responder in Wireless Mobility Mode	Verify a 911 call placed using Cisco Cius is routed to the correct Public Safety Answering Point (PSAP), and the PSAP call-back routes the call back to Cisco Cius.	Cisco Cius->Unified CM->Java Telephony Application Programming Interface (JTAPI)->Cisco Emergency Responder; Cisco Cius->Unified CM->PSTN Gateway->PSTN->Public Safety Answering Point	Passed	
UC861IF.CUS.610	Cisco CIUS placing SAF Call (G.711), far side transfers to Unified Survivable RemoteSite Telephony Site (G.729)	Verify the ability to use Cisco CIUS to place a 911 call. Verify the call is routed to the correct PSAP. Verify that PSAP call-back sends the call back to CIUS.	Cisco CIUS->Unified CM->SIP SAF Trunk->Unified CM->Phone; Transfer->Unified SRST phone	Passed	
UC861IF.CUS.611	Cisco Cius Placing SAF Call, IP Call Fails and PSTN Fallback is Used	Verify the ability to place a SAF call with Cisco Cius when the IP call fails and SAF PSTN fallback is invoked. Verify Cisco Cius handles the PSTN fallback properly.	Cisco Cius->Unified CM->Cisco IOS PSTN Gateway->PSTN->PSTN Gateway->Unified CM->Phone	Passed	
UC861IF.CUS.612	Call to Analog Phone behind VG.224 shared line device barges in to call	Verify the ability to place a call from Cisco CIUS to an analog phone in another cluster (via SIP trunk) behind a VG224. Cisco CIUS is sharing a line with an Unified IP Phones 89XX/99XX. Verify the ability of the Unified IP Phones 89XX/99XX to barge into the call. Verify a successful 3-way call.	Cisco CIUS->Unified CM->SIP Trunk->Unified CM->VG224> Analog phone;After barge->Cisco CIUS->Unified CM->Unified 99xx IP phone built in bridge	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.613	Cisco Cius Call Over H.323 Fast Start Inter Cluster Trunk with Trusted Relay Point	Verify that the call sets up and media terminates on the Trusted Relay Point and the call is put on Hold/resume on placing a call from Cisco Cius to a Unified IP Phones 89XX/99XX in another cluster via an H.323 fast start Inter Cluster Trunk, given that the Cisco Cius device has "Use Trusted Relay Point" enabled.	Cisco Cius->Unified CM->H.323 Fast Start->Unified CM->Unified IP Phones 89XX/99XX	Passed	
UC861IF.CUS.614	Cisco Cius call to IPv4 Endpoint, Far Side Transfers to IPv6 Endpoint with Cisco IOS Media Termination Point (MTP) Inserted	Verify that media remains intact between Cisco Cius, the Media Termination Point, and the IPv6 endpoint on placing an audio call from a Cisco Unified Personal Communicator 8 softphone in a different cluster to Cisco Cius via Cisco Unified Communications Session Management Edition, when the Cisco Unified Personal Communicator transfers the call to an IPv6-only device in its same cluster (an MTP should be invoked).	Cisco Cius->Unified CM->SIP Trunk->SME->SIP Trunk->Cisco Unified Personal Communicator8; Transfer->IPv6 phone	Passed	
UC861IF.CUS.615	Cisco Cius Conference into an Active Shared Line Call	Verify a conference bridge is invoked and three-way communication is successful on placing a call from a phone over the PSTN to an Unified IP Phones 89XX/99XX sharing a line with Cisco Cius given the conference in the Cisco Cius phone is using its primary line.	Cisco Cius->Unified CM->IOS Conference bridge	Passed	
UC861IF.CUS.617	Call Hold/Resume in Audio Call with Cisco Cius via SIP Delay Offer trunk	Verify the ability to Hold/Resume call in audio call with Cisco Cius via SIP delay offer trunk.	Cisco Cius->Unified CM1->SIP DO trunk->Unified CM2->Unified 79XX IP Phone	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.618	Call Hold/Resume in Audio Call with Cisco Cius via MGCP Gateway	Verify the ability Hold/Resume call in audio call with Cisco Cius via MGCP Gateway	Cisco Cius->Unified CM1->MGCP Gateway trunk->Unified CM2->Unified 89XX IP Phone	Passed	
UC861IF.CUS.619	Call from Cisco Unified 89XX SIP phone to phone with Call Forward All to Cisco CIUS via SME SIP trunks using Early Offer	Verify that the call from Cisco Unified 89XX SIP phone to phone with Call Forward All to Cisco CIUS via SME SIP trunks uses Early Offer.	Unified 89XX Phone->Unified CM1->Phone CFWD ALL->SIP EO trunk->SME->SIP EO trunk->Unified CM2->Cisco CIUS	Passed	
UC861IF.CUS.620	Call from Cisco Unified 79XX SCCP phone via H.323 trunk to phone with Call Forward All to Cisco Cius	Verify the ability to call from Unified 79XX SCCP phone via H.323 trunk to phone with Call Forward All to Cisco Cius.	Unified 79XX Phone->Unified CM1->H.323 trunk->Unified CM2->Phone Call Forward All->Cisco Cius	Passed	
UC861IF.CUS.623	Call Transfer to Cisco Cius from Unified 89XX SIP Phone via SIP trunk using Delay Offer	Verify the call transfer to Cisco Cius from Unified 89XX SIP phone via SIP trunk using delay offer.	Audio IP Phone->Unified CM1->Unified 89XX IP Phone Call Transfer->SIP DO trunk->Unified CM2->Cisco Cius	Passed	
UC861IF.CUS.624	Call to Unified 79XX SCCP Phone via H.323 Gateway Call Transfer to Cisco Cius	Verify the call to Unified 79XX SCCP phone via H.323 Gateway with call transferred to Cisco Cius.	Audio IP Phone->Unified CM1->H.323 Gateway->Unified CM2->Unified 79XX Phone Call Transfer->Cisco Cius	Passed	
UC861IF.CUS.626	Call to Cisco Cius via SIP Gateway with call transferred from Cisco Cius to Unified 69XX SIP Phone	Verify call to Cisco Cius via SIP Gateway with call transferred from Cisco Cius to Unified 69XX SIP phone.	Audio IP Phone->Unified CM1->SIP Gateway->Unified CM2->Cisco Cius Call Transfer->Unified 69XX Phone	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.627	Cisco CIUS activates Do Not Disturb and calls from PSTN are rejected and CIUS does not ring	verify if CIUS activates Do Not Disturb and calls from PSTN are rejected and Cisco CIUS does not ring.	Unified 79XX Phone->Unified CM1->SIP Gateway->Unified CM2->Cisco CIUS	Passed	
UC861IF.CUS.630	Call to Cisco Unity Connection via PSTN trunk with Cisco Unity Connection Transferring Call to Cisco Cius	Verify call to Cisco Unity Connection via PSTN trunk with Cisco Unity Connection transferring call to Cisco Cius.	Audio Phone->Unified CM1->SIP Gateway->Unified CM2->Cisco Unity Connection Call Transfer->Cisco Cius	Passed	
UC861IF.CUS.631	Cisco Cius placing 911 call via Cisco Emergency Responder in Docked Mode	Verify the call is routed to the correct PSAP and the PSAP call-back sends the call back to Cisco Cius when Cisco Cius is used to place a 911 call.	Cisco Cius->Unified CM->JTAPI->Cisco Emergency Responder; Cisco Cius->Unified CM->PSTN Gateway->PSTN->PSAP	Passed	
UC861IF.CUS.801	Layer 3 Roaming with Cisco Cius	Verify the ability to perform a Layer 3 roaming while Cisco Cius is in a call.	Cisco Cius -->Cisco Lightweight Access Point 1 -->Wireless LAN Control1 -->Wireless LAN Control2 -->Cisco Lightweight Access Point 2 -->Unified CM	Passed	
UC861IF.CUS.802	Layer 2 Roaming with Cisco CIUS	Verify the ability to perform layer 2 roaming while Cisco CIUS is in a call.	Cisco CIUS-->LAP1-->WLC1-->LAP2-->Unified CM	Passed	
UC861IF.CUS.803	Call Transfer Over a Secure SIP Trunk via Cisco Unified Session Management Edition	Verify the ability to perform a call transfer over a secure SIP trunk involving Cisco Unified Session Management Edition.	Cisco Cius -->Unified CM1 -->Secure SIP Trunk -->SME -->Secure SIP Trunk -->Unified CM2 -->Cisco IP Phone1; Cisco Cius -->Transfer -->Cisco IP Phone 2	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.804	Voicemail server in Unity Connection in Cisco Session Manager Edition Cluster	Verify that Cisco CIUS can leave voicemails in Cisco Unity connection in Cisco Session Manager Edition and read the VoiceMails.	Cisco CIUS-->Unified CM1-->Secure SIP Trunk-->Cisco SME-->Secure SIP Trunk-->Unified CM2-->IP Phone1->Secure SIP Trunk-->Unified CM-->SCCP-->Unity Connection	Passed	
UC861IF.CUS.805	Handoff Cisco CIUS call to Mobile phone	Verify that call from Cisco CIUS can be handed over to a remote destination across SIP trunk.	Cisco CIUS-->Unified CM1-->SIP Trunk-->Unified CM2-->IP Phone1	Failed	CSCto97665
UC861IF.CUS.955	Cisco Cius device controlled from a Cisco Unified Personal Communicator in deskphone mode accessed through the Virtual Desktop Infrastructure (VDI) application in Cisco Cius	Verify that Cisco Cius can be controlled from a Cisco Unified Personal Communicator in deskphone mode to place, receive a call and to perform other call features.	Cisco Cius->Unified CM1->SIP Trunk -->Unified CM2 -->Cisco IP Phone	Passed	
UC861IF.MOB.001	Nokia Mobility Client Midcall Feature - Hold and Resume over SIP Trunk	Verify Hold and Resume feature of Nokia Mobility Client by holding an incoming call through SIP trunk and resuming it multiple times.	Phone1-Unified CM1----<Cisco IME Trunk>---Unified CM2----<802.11 wireless>--->Nokia Mobility Client	Passed	
UC861IF.MOB.002	Nokia Mobility Client Midcall Feature - An Incoming Call on Call Waiting is Sent to Secure Voice Mail	Verify call waiting and call forward, when a Nokia client in a call receives another call. Verify whether the waiting call is sent to a secure voicemail, the caller deposits a message and later the client retrieves the message.	Phone1-Unified CM1-Nokia Mobility Client -Cisco Unity Connection	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.MOB.003	Nokia Mobility Client Midcall Feature - Conferencing a PSTN phone over SIP Gateway	Verify conferencing feature of Nokia Mobility Client by conferencing an iPhone client and PSTN phone.	Nokia Mobility Client(Dial Via Office call)--Unified CM1--iPhone Client(dual mode) + SIP Gateway	Passed	
UC861IF.MOB.004	Nokia Mobility Client Midcall Feature - Park and Retrieve Calls from Nokia Client	Verify conference parking feature of Nokia mobility client when an intercluster call is parked at Nokia client and retrieved from Cisco Unified IP phone 894X series, and Unified IP phone 894X phone then parks that call and Nokia client retrieves the call.	Nokia Mobility Client(Unified CM - call park feature)	Passed	
UC861IF.MOB.005	An Instant Message from Client Services Framework (CSF) Client to Nokia Mobility Client is Escalated to a Voice Call	Verify the Presence Status on contacts in Nokia Mobility client and also the presence status update of Nokia client on other clients.	Nokia Mobility Client (Unified CM(-Cisco Unified Presence-Unified Personal Communicator	Passed	
UC861IF.MOB.006	Handoff to Mobile Network using Dial Via Office- Forward (DVO-F) and Dial Via Office- Reverse (DVO-R) methods	Verify handoff call to mobile network using Dial Via Office- Forward and Dial Via Office- Reverse.	Nokia Client->H.323 Gateway--->Unified CM--->iPhone	Passed	
UC861IF.MOB.007	Presence Status of Enterprise Contacts in Call Logs and Directory	Verify that various presence status of enterprise contacts works in call logs and directory list.	Nokia Mobility Client(Cisco Unified Presence+Unified CM	Passed	
UC861IF.MOB.008	Multiple Instant Messaging (IM) Sessions and Escalation of IM to Voice Call	Verify that Nokia Mobility Client can establish multiple IM sessions and it can escalate some of the IM to voice calls.	Nokia Mobility Client-(Unified CM+Cisco Unified Presence)	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.MOB.009	Nokia Mobility Client Attends Webex Meeting, Dials into Meeting and Receives Call Back	Verify that Nokia Mobility Client can attend WebEx meeting by dialing into meeting, entering meeting ID, and by receiving call back from Meeting Place.	Nokia Mobility Client-(Unified CM+Meeting Place)	Passed	
UC861IF.MOB.010	Mobility: Handoff Invoked from Client Having Multiple Calls	Verifies that handoff to mobile works from a client who has multiple calls.	Nokia Mobility Client-(Unified CM+ Gateway)	Passed	
UC861IF.MOB.011	Cisco Android Client receiving a SIP Intercluster Call and Moves the Call to Mobile	Verify that the Cisco Mobile for Android can receive the call while registered to WiFi and then it can send the call to mobile network and continue the call.	Android Mobility Client<----Unified CM1----SIP---Unified CM2---IP Phone	Passed	
UC861IF.MOB.012	Android Mobility Client joining Meeting and Transferring the Call to Cell Number	Verify that Android mobility client can dial into and dial out to WebEx/Meeting Place meeting and then transfer the call to mobile phone.	Nokia Mobility Client->H.323 Gateway->Unified CM-->MeetingPlace	Passed	
UC861IF.MOB.013	Nokia Mobility Client Adapts to the Configuration Changes in Phone Page	Verify that the Nokia Mobility Client can adapt to device pool, Media Resource Group List (MRGL) and Calling Search Space (CSS) changes in the phone configuration page of Unified Communications Manager.		Passed	
UC861IF.MOB.014	Android Mobility Client Establishing a Conference Call between iPhone Client and an Intercluster Destination	Verify that Android client can set up a conference call, when the other parties of conference call are iPhone client and an IP phone across intercluster SIP trunk.	Nokia Mobility Client	Passed	
UC861IF.MOB.015	Receiving Call through PSTN Carrier when the Client is in Conference	Verify when Android Mobility Client is in conference with enterprise contacts, it receives a call through GSM network.	Soundwave Client----Unified CM----TelePresence and Cisco UC Integration@ for Microsoft Office Communicator	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.MOB.016	Android Mobility Client getting an Incoming Call through Trunk with Early Offer Turned ON and then transfers the call	Verifies if an Android Mobility client receives an incoming call through a Trunk with early offer turned ON, and the soundwave user is able to transfer the call.	Soundwave Client---Unified CM1-----<Cisco IME>----Unified CM2-----iPhone; Soundwave Client---Unified CM1----Cisco IME---Unified CM2	Passed	
UC861IF.MOB.017	Android Mobility Client setting up three way conference and handoff to Extension Mobility logged in deskphone	Verifies that when an Android Mobility Client has Device Mobility turned ON, the user can log into Extension Mobility phone and bring up the client to set up a three way conference, involving one user across SIP trunk, and a third user. Verifies that software conference resource at remote site is used, and the soundwave user can handoff the call to the Extension Mobility deskphone.	Soundwave Client Unified IP Phone1 Unified IP Phone2-----Unified CM1----Extension Mobility Phone	Passed	
UC861IF.MOB.018	Android Mobility Client receiving an incoming call from a Cisco TelePresence endpoint and handoff the call to UC Integration@ for Microsoft Office Communicator	Verify if the Android Mobility Client application is running in background when the user receives an incoming call from a Cisco TelePresence endpoint. Verifies if the Cisco TelePresence user requests the soundwave user for video to handoff the call to UC Integration@ for Microsoft Office Communicator and resume the video call.	Soundwave Client----Unified CM----Cisco TelePresence and UC Integration@ for Microsoft Office Communicator	Passed	
UC861IF.MOB.019	Resilience of Nokia Mobility Client on Registration with Cisco Unified Communications Manager	Verify that the Cisco mobility client can register to standby Unified Communications Manager server when active one fails and continue to work normally.	Nokia Mobility Client	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.OTH.001	Unified MeetingPlace: Start Multiple Meetings in a Site with Multiple Nodes and Ensure Each Node in the Site gets Used	Verifies each node in a site with multiple nodes gets used when multiple meetings are started in the site.	Endpoint->Unified CM->SME->SIP Trunk->Cisco Unified Meeting Place	Passed	
UC861IF.OTH.002	Unified MeetingPlace: Site Selection Based on Preferred Site Field in User Profile	Verifies site selected via preferred site field in user profile.	SME site - Endpoint->Unified CM->SME ->SIP Trunk->Cisco Unified MeetingPlace	Passed	
UC861IF.OTH.003	Unified MeetingPlace: Site Selection Based on User Profile Time-zone Setting	Verifies site selected via time-zone setting in user profile.	Endpoint->Unified CM->SIP Trunk->Cisco MeetingPlace	Passed	
UC861IF.OTH.004	Unified MeetingPlace: Site Selection based on Default Site when User is not Associated with a Site	Verifies site selected via system default site when user is not associated with a site.	Endpoint->Unified CM->SME->SIP Trunk->Cisco MeetingPlace	Passed	
UC861IF.OTH.005	Unified MeetingPlace: Host Meetings on the Single Active Node of a Multinode Site	Verifies if one node of a two node site hosts meetings when the other node in the site is down.	Endpoint->Unified CM->SME->SIP Trunk->Cisco MeetingPlace	Passed	
UC861IF.OTH.006	Unified MeetingPlace: Host Meetings on an Alternate Site in the same Region with Multiple Sites Available	Verifies meetings start on an alternate site in the same region when all nodes in a site are down.	Endpoint->Unified CM->SIP Trunk->Cisco MeetingPlace	Passed	
UC861IF.OTH.007	Unified MeetingPlace: Host Meetings on an Alternate Site in a Different Region with Multiple Sites Available	Verifies meetings can start on an alternate site in a different region with multiple sites available.	Endpoint->Unified CM->SIP Trunk->Cisco MeetingPlace	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.OTH.008	Unified MeetingPlace: Meeting Restarts on a Two-node Site when Participants Dial Back	Verifies if meeting restarts on a two-node site after one node goes down, when participants dial back into the same site.	Endpoint->Unified CM->SME->SIP Trunk->Cisco MeetingPlace	Passed	
UC861IF.OTH.009	Node goes Down During Meeting, Meeting Restarts on Different Site in the Same Region in Unified MeetingPlace	Verifies that when all nodes go down in a site, participants dial back in and meeting is on node in other site of the same region.		Passed	
UC861IF.OTH.010	Node Goes Down During Meeting and Restarts on Different Site in a Different Region in Unified MeetingPlace	Verifies that when all nodes go down in a site, participants dial back in and meeting is on node in other site of a different region.		Passed	
UC861IF.OTH.011	Unified MeetingPlace: Call into SME MeetingPlace from multiple clusters	Verify that Unified MeetingPlace node in SME site can be accessed via SIP and H.323 Inter Cluster Trunks from multiple Cisco Unified Communications Manager clusters.		Passed	
UC861IF.OTH.012	Unified MeetingPlace: Call into Cisco Unified MeetingPlace Hardware Media Server (HMS) via SME	Verify that Cisco MeetingPlace Hardware Media Server (HMS) node in a site can be accessed via SME tandem Unified Communications Manager cluster.		Passed	
UC861IF.OTH.013	Unified MeetingPlace: Call into MeetingPlace - Enhanced Media Server via SME	Verify MeetingPlace - Enhanced Media Server node in a site can be accessed via SME tandem Unified Communications Manager cluster.		Passed	
UC861IF.OTH.014	Unified MeetingPlace: Outdial from SME MeetingPlace to multiple clusters	Verify outdial from a meeting on a MeetingPlace node in SME site to multiple Unified CM clusters.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.OTH.015	Dial into Cisco Unified MeetingPlace in Session Manager Edition site using dial pattern so call uses Media Termination Point (MTP) resources from one site to the other	Verify that Media Termination Point (MTP) can be used when dialing into a meeting in a Session Manager Edition tandem Cisco Unified Communications Manager with Cisco Unified MeetingPlace.		Passed	
UC861IF.OTH.016	Dial into Cisco Unified MeetingPlace in Session Manager Edition site using dial pattern so call uses Trusted Relay Point (TRP) resources from one site to the other in Unified MeetingPlace	Verify if Trusted Relay Point (TRP) can be used when dialing into a meeting in a Session Manager Edition tandem Cisco Unified Communications Manager with Cisco Unified MeetingPlace.		Passed	
UC861IF.OTH.030	Cisco UC Integration(TM) for Microsoft Lync joining WebEx/Cisco Unified MeetingPlace based meeting	Verify that Cisco UC Integration(TM) for Microsoft Lync can dial into WebEx/Cisco Unified MeetingPlace based meeting and also the reverse way, WebEx/Cisco Unified MeetingPlace calling Cisco UC Integration(TM) for Microsoft Lync.	Cisco UC Integration(TM) for Microsoft Lync -->Unified CM--->Session Manager Edition---Cisco Unified MeetingPlace--->WebEx	Passed w/ Exception	CSCto50486
UC861IF.OTH.032	Cisco UC Integration(TM) for Microsoft Lync setting up Video Conference between Unified IP Phone 9971 and Cisco Unified IP Phone 8945 Phones	Verify that Cisco UC Integration(TM) for Microsoft Lync set up a video conference between Unified IP Phone 9971 and Cisco Unified IP Phone 8945 video capable phones.	Cisco UC Integration(TM) for Microsoft Lync-->Unified CM + Cisco Codian---->Unified IP Phone 9971+ Unified IP Phone 8945	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.OTH.033	Cisco UC Integration(TM) for Microsoft Lync in Secure Mode Getting Secure Voicemail	Verify that Cisco UC Integration(TM) for Microsoft Lync in secure mode get Visual VoiceMail indication and can call the VoiceMail server and read the secure VoiceMail.	Cisco UC Integration(TM) for Microsoft Lync-->Unified CM--->Cisco Unity Connection	Passed	
UC861IF.OTH.034	Cisco UC Integration(TM) for Microsoft Lync Coming up in SRST Mode	Verify that Cisco UC Integration(TM) for Microsoft Lync can automatically come up in Cisco Survivable Remote Site Telephony mode when WAN connectivity is broken during the call.	Cisco UC Integration(TM) for Microsoft Lync-->Cisco Survivable Remote Site Telephony--->Unified CM	Passed	
UC861IF.OTH.035	Cisco Unity Express Single Inbox Receiving New Emails and Marking it read through Outlook	Verify that voicemails can be received from Cisco Unity Express to Outlook, and the voicemails can be marked read from Outlook configured to synchronize with Exchange.	Cisco Unity Express -->Exchange 2007 -->Outlook	Passed	
UC861IF.OTH.036	Voicemails Marked Urgent in Cisco Unity Express Single Inbox	Verify that when messages marked urgent in Cisco Unity Express is received with email importance set to high.	Cisco Unity Express -->Exchange 2007 -->Outlook	Passed	
UC861IF.OTH.037	Marking Read Messages Unread in Microsoft Outlook/Microsoft Exchange in Cisco Unity Express Single Inbox	Verify that when read emails/voicemails are marked unread or new in Microsoft outlook, Cisco Unity Express marks that voicemail as new as well and turns on the Message Waiting Indication (MWI).	Cisco Unity Express -->Microsoft Exchange 2007 -->Microsoft Outlook	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.OTH.074	CSF client (Cisco Unified Communications Integration(TM) for Microsoft Lync) can have a voicemail box in Cisco Unity Express	Verify a Cisco Unity Express can provide voicemail service to Cisco Unified Communications Integration(TM) for Microsoft Lync	Phone1 -->Unified CM -->SIP Trunk -->Unified CM -->UC Integration(TM) for Microsoft Lync -->Call Forward No Answer --> Unified CM -->Java Telephony Application Programming Interface -->Cisco Unity Express	Passed	
UC861IF.OTH.101	Dual Tone Multi-frequency (DTMF) Interoperability of Cisco Unified IP Phone 894x with Cisco Unity Connection	Verify that DTMF works fine on Cisco Unified IP Phone 894X with Cisco Unity Connection when the call is placed from a remote cluster over SIP trunk.	Unified IP Phone 894X -->Unified CM -->SIP Trunk -->Unified CM -->Unity Connection	Passed	
UC861IF.OTH.103	Transcoder can be Invoked Dynamically for a Call Involving Cisco Unified IP Phone 894X	Verify that Unified Communications Manager can invoke a transcoder dynamically when there is a Codec mismatch for a call involving Cisco Unified IP Phone 894X.	IP Phone -->Unified CME -->SIP Trunk -->Unified CM -->Transcoder -->Unified IP Phone 894X	Passed	
UC861IF.OTH.104	Cisco IPv6 call from a Dual Stack Unified IP Phone and Call Out on Hold	Verify that a Dual Stack Unified IP phone can be used to place a call with media as Cisco IPv6. Verify that the call can be placed on hold by invoking the Hold key on the Unified IP Phones.	IP Phone Dual Stack -->Unified CM -->SIP Trunk Dual Stack -->Unified CM -->Unified IP Phones; Hold invoked on Unified IP Phones; IP Phone -->Unified CM -->SIP Trunk -->Unified CM -->Music on Hold (MoH)	Passed	
UC861IF.OTH.105	Point-to-Point Call Over a Dual Stack Cisco IPv6 SIP trunk involving Cisco IPv6 only Unified IP Phones.	Verify that Unified IP Phones when configured in Cisco IPv6 only mode can be used in calls over the SIP trunks.	IP Phone Dual Stack -->Unified CM -->SIP Trunk Dual Stack -->Unified CM -->Unified IP Phones only	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.OTH.106	Cisco IPv6 Call from an IPv6 only Unified IP Phones and Call Out on Hold	Verify that a Cisco IPv6 only Unified IP Phone can be used to place a call with media as IPv6. Verify that the call can be placed on hold by invoking the Hold key on the Unified IP Phones.	IP Phone Dual Stack -->Unified CM -->SIP Trunk Dual Stack -->Unified CM -->Unified IP Phones IPv6 only; Hold invoked on Unified IP Phones; IP Phone -->Unified CM -->SIP Trunk -->Unified CM -->Music on Hold	Passed	
UC861IF.OTH.110	Conference Call Using a Cisco Internet Protocol Version 6 (IPv6)-Only 6900 Series Unified IP Phone	Verify that an IPv6 only 6900 Series Unified IP phone can be used to place a conference call, when the IPv6 transcoder will be invoked for the IPv6 only phone.	IP Phone DS -->Unified CM -->SIP Trunk DS -->Unified CM -->Unified IP Phone 6900 Series IPv6 only; Hold invoked on Unified IP Phone 6900 series; IP Phone -->Unified CM -->SIP Trunk -->Unified CM -->Music on Hold	Passed	
UC861IF.OTH.111	Call from IPV4 phone to IPV6 phone over SME connected with Alternative Network Address Types (ANAT) enabled SIP Trunk	Verify that call can be established from an IPv4 phone in one cluster to a dual stack phone in another cluster through SME cluster connected with Alternative Network Address Types (ANAT) enabled SIP trunk.	IP Phone V4 -->Unified CM1 -->SIP Trunk DS -->SME --<SIP Trunk>---Unified CM Unified CM2---->DS IP phone	Passed	
UC861IF.OTH.112	Call from IPV6 Cisco Unified IP Phone 6900 Series to IPV6 phone through IPv6 SIP Gateway connected with Alternative Network Address Types (ANAT) enabled SIP Trunk	Verify that call can be established from an IPv6 Unified IP Phone 6900 Series through an IPv6 SIP Gateway connected with Alternative Network Address Types enabled trunk.	IP Phone V4 -->Unified CM1 -->SIP Trunk DS -->SME --<SIP Trunk>---Unified CM Unified CM2---->DS IP phone	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.OTH.121	SRSV: Provisioning when primary Cisco Unified Communications Manager server is down but secondary is up; Primary Cisco Unity Connection Server is up	Verify that Cisco Unified Messaging Gateway (UMG)-Cisco Survivable Remote Site Voicemail (SRSV) can synchronize with secondary Unified Communications Manager server when the primary server is down and the remote site SRSV-Cisco Unity Express provisioning can continue without any problems.	Unified CM Cisco Unity Connection -->Cisco Survivable Remote Site Voicemail-Cisco Unified Messaging Gateway -->Cisco Survivable Remote Site Voicemail-Cisco Unity Express	Passed	
UC861IF.OTH.122	Cisco Survivable Remote Site Voicemail (SRSV): Provisioning when Primary Unified Communications Manager server is down but secondary is up; Primary Cisco Unity Connection server is down but secondary Cisco Unity Connection server is up	Verify that Cisco Unified Meeting Gateway-Cisco Survivable Remote Site Voicemail (UMG-SRSV) can synchronize with secondary Unified Communications Manager server when the primary server is down. Verify that UMG-SRSV can synchronize with secondary Cisco Unity Connection server when the primary Cisco Unity Connection server is down, and also verify that the provisioning is successful under these conditions.	Unified CM Cisco Unity Connection -->Cisco Survivable Remote Site Voicemail-Cisco Unified Meeting Gateway -->Cisco Survivable Remote Site Voicemail-Cisco Unity Express	Passed	
UC861IF.OTH.123	Cisco Survivable Remote Site Voicemail: Voicemail upload after WAN link restoration with primary Cisco Unity Connection server down and secondary Cisco Unity Connection server active	Verify that Cisco Unified Meeting Gateway-Cisco Survivable Remote Site Voicemail (UMG-SRSV) can upload voicemails from SRSV-Cisco Unity Express to Cisco Unity Connection after the WAN link is restored. Verify that the upload is successful even when the primary Cisco Unity Connection server is down.	Cisco Unity Connection <-->Cisco Survivable Remote Site Voicemail-Cisco Unified Meeting Gateway <-->Cisco Survivable Remote Site Voicemail-Cisco Unity Express	Failed	CSCtq49819

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.OTH.124	Cisco Survivable Remote Site Voicemail: Primary Cisco Unified Communications Manager server unavailable in the middle of manually initiated provisioning	Verify that provisioning can continue and complete successfully even when the primary Unified Communications Manager server goes offline while a manually initiated provisioning is in progress.	Unified CM Cisco Unity Connection <-->Cisco Survivable Remote Site Voicemail-Cisco Unified Meeting Gateway <-->Cisco Survivable Remote Site Voicemail-Cisco Unity Express	Passed	
UC861IF.OTH.125	Cisco Survivable Remote Site Voicemail: Primary Cisco Unity Connection server unavailable in the middle of manually initiated provisioning	Verify that provisioning can continue and complete successfully even when the primary Cisco Unity Connection server goes offline while a manually initiated provisioning is in progress.	Unified CM Cisco Unity Connection <-->Cisco Survivable Remote Site Voicemail-Cisco Unified Meeting Gateway <-->Cisco Survivable Remote Site Voicemail-Cisco Unity Express	Passed	
UC861IF.OTH.126	Cisco Survivable Remote Site Voicemail: Auto Attendant Dial by Extension when Caller is in a Custom Greeting Linked to Opening Greeting	Verify that Auto Attendant Dial by Extension is provisioned successfully based on the configuration in Cisco Unity Connection, and ensures the functionality works in Cisco Survivable Remote Site Voicemail-Cisco Unity Express.	Unified CM Cisco Unity Connection <-->Cisco Survivable Remote Site Voicemail-Unified Meeting Gateway <-->Survivable Remote Site Voicemail-Cisco Unity Express; Phone -->SRST -->Survivable Remote Site Voicemail-Cisco Unity Express -->Transfer -->Phone	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.OTH.128	Caller Input is Ignored when Additional Key Input is Configured in Cisco Survivable Remote Site Voicemail	Verify caller input is configured to transfer to a call handler which in turn is configured to send the call to a subscribers greeting in Cisco Survivable Remote Site Voicemail.	Unified CM Cisco Unity Connection <-->Cisco SRSV-Unified Messaging Gateway <-->Cisco SRSV-Cisco Unity Express Phone -->Cisco SRST -->Cisco SRSV-Cisco Unity Express -->Transfer -->Phone	Passed	
UC861IF.OTH.130	Cisco Survivable Remote Site Voicemail: Provisioning additional users in Cisco Survivable Remote Site Voicemail-Cisco Unity Express through Unified Messaging Gateway	Verify that Unified Messaging Gateway can automatically provision users in Cisco Survivable Remote Site Voicemail -Cisco Unity Express once users are added in Cisco Unity Connection.	Unified CM Cisco Unity Connection <-->Cisco Survivable Remote Site Voicemail-Unified Messaging Gateway <-->Cisco Survivable Remote Site Variable-Cisco Unity Express	Passed	
UC861IF.OTH.140	Cisco Unity Express: A secure VoiceMail is forwarded to SRST - Unified Express subscriber as Voice Profile for Internet Mail (VPIM) message and the Subscriber Downloads and Plays it.	Verify that a subscriber in Cisco Unity Express controlled by Cisco Unified Communications Manager in a SRST router can download and play a Voice Profile for Internet Mail secure Voicemail forwarded from Unity connection subscriber.	SRST Cisco Unity Express <-->SRST<-->Unified CM<--->Cisco Unity Connection	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.OTH.141	Cisco Unity Express: A secure Voicemail is forwarded to Unity Express-Cisco Unified Communications Manager Express subscriber as Voice Profile for Internet Mail message and the subscriber downloads and plays it.	Verify that a subscriber in Cisco Unity Express controlled by Cisco Unified Communications Manager Express in a SRST router can download and play a Voice Profile for Internet Mail secure Voicemail forwarded from Unity connection subscriber.	Unified CM Cisco Unity Express <-->Cisco Unity Connection Unified CME<-->Cisco Unity Connection	Passed	
UC861IF.OTH.142	Cisco Unity Express: iPhone Mobility Client is Dialing into SRST-Cisco Unity Express and playing a Secure Voicemail	Verify that an iPhone mobility client subscriber in SRST-Cisco Unity Express can dial into Cisco Unity Express and plays secure Voicemails.	SRST Cisco Unity Express <-->SRST<-->Unified CM	Passed	
UC861IF.OTH.143	Cisco Unity Express: iPhone mobility client login to Cisco Unity Express and sends a voicemail to a subscriber in Cisco Unity Connection	Verify that iPhone client can dial into SRST-Cisco Unity Express and sends a secure voicemail to a Cisco Unity Connection subscriber.	SRST Cisco Unity Express <-->SRST<-->Unified CM<--->Cisco Unity Connection	Passed	
UC861IF.OTH.174	Cisco Survivable Remote Site Voicemail-Cisco Unity Express gets updated as and when users are added and deleted in Cisco Unity Connection.	Verify that Cisco Unified Messaging Gateway updates Cisco Survivable Remote Site Voicemail-Cisco Unity Express whenever users are added or removed from Cisco Unity Connection.	Cisco Unity Connection -->Unified Messaging Gateway-->Cisco Survivable Remote Site Voicemail-Cisco Unity Express	Passed	
UC861IF.OTH.175	Supervised Transfer when Ports are Configured to Support Authentication and Encryption	Verify that Unity Connection ports can be configured for authentication and encryption, given that supervised transfer is possible when a call is placed from a secure endpoint.	Cisco IP Phone -->Unified CM -->SIP Trunk -->Unified CM -->SCCP -->Cisco Unity Connection -->Transfer -->Unified CM -->Cisco IP Phone	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.001	Conference Cisco Unified Communications Manager and Cisco TelePresence Video Communication Server (VCS) endpoints using Cisco Codian Software Bridge registered to Unified Communications Manager	Verify conference can be established between Cisco Cius T; Cisco IP Video Phone E20 and Cisco TelePresence System 1700 MXP using Cisco Codian Adhoc bridge registered to Unified Communications Manager as conference resource.	Cisco Cius@ - MSP Unified CM --- H.225 trunk --- GateKeeper - Cisco TelePresence VCS --- Cisco IP Video Phone E20 ---Conference using Cisco Cius@ --- TD Cisco 1700 MXP -H.323 - Cisco TelePresence VCS	Passed	
UC861IF.VID.002	Conference Cisco Unified Communications Manager and Cisco TelePresence Video Communication Server endpoints using Cisco Codian Software Bridge registered to Unified Communications Manager.	Verify conference can be established between Cisco Cius, Cisco Unified Personal Communicator (CUPC) running on the virtual desktop and Cisco IP Communicator/Cisco Unified Video Advantage registered to Unified Communications Manager Express.	Cisco Cius - MSP Unified CM ---SIP Trunk -Abilene Unified CM ---Unified Personal Communicator---- Conference From Unified Personal Communicator --- H.323 Gateway -H.323 Trunk --- Unified CME	Passed	
UC861IF.VID.003	Conference Cisco Unified Communications Manager and Video Communication Server (VCS) endpoints using Tandberg Codian Software Bridge registered to Unified Communications Manager	Verify conference can be established between Cisco TelePresence MoviT registered to VCS; Cisco Unified IP Phone 9971 registered to Unified Communications Manager and Polycom HDX 4000 registered to Unified Communications Manager.	Cisco IP Video Phone E20 - MSP Unified Communications Manager --- SIP Trunk -VCS -MOVi ---- Conference From Cisco IP Video Phone E20 --- SIP Trunk ---Polycom HDX 4000	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.004	Conference Cisco Unified Communications Manager and Cisco TelePresence Video Communication Server endpoints using Cisco Codian Software Bridge registered to Unified Communications Manager.	Verify conference can be established between Cisco CiusT ; Cisco TelePresence System 1000 and Cisco TelePresence 1700 MXP using Cisco Codian Adhoc bridge.	Cisco Cius@ - MSP Unified CM --- SIP Trunk ---Abilene Unified CM ---Cisco TelePresence System ---Conference using Cisco Cius --- Cisco TelePresence MXP 1700 -H.323 - Cisco TelePresence Video Communication Server	Passed	
UC861IF.VID.005	Presentation share between Cisco TelePresence System EX90; Cisco IP Video Phone E20 and Unified IP Phone 89XX/99XX	Verify whether Cisco TelePresence System EX90 user registered to Cisco Unified Communications Manager can share presentation with Cisco IP Video Phone E20 registered to Video Communication Server (VCS) and Unified IP Phone 89XX/99XX registered to Unified Communications Manager.	Unified Personal Communicator - MSP Unified CM --- SIP Trunk ---VCS -Cisco E20 ----Conference from Cisco TelePresence System Ex90 ---- SIP trunk -----Unified CM 9971 IP Phone -- Cisco TelePresence System Ex90 initiate Presentation	Passed	
UC861IF.VID.006	Presentation share from Cisco TelePresence Movi registered to Video Communication Server; Cisco IP Video Phone E20 and Unified IP Phone 9971 registered to Unified Communications Manager	Verify if Cisco TelePresence MOVi can share presentation with Cisco IP Video Phone E20 registered to Unified Communications Manager and Unified IP Phone 9971 Phone registered to Unified Communications Manager.	Unified Personal Communicator - MSP Unified CM --- SIP Trunk ---VCS -Cisco IP Video Phone E20 ----Conference from Unified Personal Communicator ----- SIP trunk ------Unified CM 9971 IP Phone --Unified Personal Communicator initiate conference	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.007	Cisco TelePresence Quick Set C20 Performs SIP URI based Conference with Cisco Unified Communications Manager Endpoint	Verify Cisco TelePresence Quick Set C20 registered to Cisco Unified Communications Manager as third party SIP endpoint can invoke multiway conference that is registered to Cisco TelePresence VCS.	Cisco TelePresence Quick Set C20 --Unified CM --Unified IP Phone 9971 ---- Multiway ----SIP trunk --Cisco TelePresence VCS ---- Unified CM ---Unified IP Phone 9971	Failed	CSCtl56764
UC861IF.VID.008	Presentation share from Cisco TelePresence Quick Set C20 registered to Video Communication Server ; Polycom HDX 4000 and Cisco Unified IP Phone 7985 registered to Unified Communications Manager	Verify whether Cisco TelePresence MOVi can share presentation with Polycom HDX 4000 registered to Unified Communications Manager and Cisco Unified IP Phone 7985 registered to Unified Communications Manager.	Cisco TelePresence Quick Set C20 ---Video Communication Server ----SIP Trunk ---Polycom --- Cisco TelePresence MOVi Conference ---SIP Trunk ---Unified CM ----Unified IP Phone 7985 -Initiate presentation on Cisco TelePresence MOVi	Passed	
UC861IF.VID.009	Presentation share from Cisco TelePresence System MXP 1700 registered to Cisco TelePresence Video Communication Server, Polycom HDX 4000 and Cisco IP Communicator/Cisco Unified Video Advantage phone registered to Unified Communications Manager	Verify whether Cisco TelePresence System MXP 1700 can share presentation with polycom HDX 4000 registered to Cisco Unified Communications Manager and Cisco IP Communicator phone registered to Cisco Unified Communications Manager.	Cisco TelePresence System MXP 1700 ---Cisco TelePresence System VCS ---SIP Trunk --Polycom ---Cisco TelePresence System MXP 1700-- Conference --SIP Trunk --Unified CM ---Cisco IP Communicator -Initiate presentation on Cisco TelePresence System MXP 1700	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.013	Scheduled conference using Cisco TelePresence server and presentation sharing using Cisco Unified Personal Communicator	Verify whether Cisco Unified Personal Communicator, Cisco TelePresence System 1000, Cisco TelePresence System 500 and Cisco TelePresence System EX90 can join Cisco TelePresence server conference and view Cisco Unified Personal Communicator presentation share.	Polycom HDX; Cisco Cius@ ----- SIP Trunk ----Cisco TelePresence VCS ---Cisco TelePresence Server ----- SIP trunk ---- Cisco TelePresence System MXP 1700	Passed	
UC861IF.VID.014	Cisco IP Video Phone E20 registered to Cisco TelePresence Video Communication Server Expressway connected to Demilitarized Zone (DMZ) port	Verify if Cisco IP Video Phone E20 residing in remote location can register to Cisco TelePresence Video Communication Server expressway and is able to join Cisco TelePresence Multipoint Switch conference.	Cisco IP Video Phone E20 - WAN ----DMZ-Switch ---- Cisco TelePresence VCS Expressway ----- Conference ---Abilene -Unified CM ----SIP trunk ---DEN - Session Manager Edition ---Cisco Media Experience Engine ---Cisco TelePresence Multipoint Switch	Passed	
UC861IF.VID.025.1	SIP Wideband Audio Codec Support G722.1; AAC MP4-LATM	Verify SIP Wideband Audio Codec Support G722.1 for Advanced Audio Codec (AAC) MP4-LATM		Passed	
UC861IF.VID.025.2	SIP Wideband Audio Codec Support G722.1; AAC MP4-LATM	Verify SIP Wideband Audio Codec Support G722.1 for AAC MP4-LATM		Passed	
UC861IF.VID.025.3	SIP Wideband Audio Codec Support G722.1; AAC MP4-LATM	Verify SIP Wideband Audio Codec Support G722.1 for AAC MP4-LATM		Passed	
UC861IF.VID.025.4	SIP Wideband Audio Codec Support G722.1; Advanced Audio Codec MP4-LATM	Verify SIP Wideband Audio Codec Support G722.1 for Advanced Audio Codec MP4-LATM		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.025.5	SIP Wideband Audio Codec Support G722.1; Advanced Audio Codec MP4-LATM	Verify SIP Wideband Audio Codec Support G722.1; Advanced Audio Codec MP4-LATM		Passed	
UC861IF.VID.025.6	SIP Wideband Audio Codec Support G722.1; Advanced Audio Codec MP4-LATM	Verify SIP Wideband Audio Codec Support G722.1; Advanced Audio Codec MP4-LATM		Passed	
UC861IF.VID.025.7	SIP Wideband Audio Codec Support G722.1; Advanced Audio Codec MP4-LATM (Testcase 7)	Verify SIP Wideband Audio Codec Support G722.1; Advanced Audio Codec MP4-LATM		Passed	
UC861IF.VID.025.8	SIP Wideband Audio Codec Support G722.1; Advanced Audio Codec MP4-LATM (Testcase 8)	Verify SIP Wideband Audio Codec Support G722.1 for Advanced Audio Codec MP4-LATM		Passed	
UC861IF.VID.033	Conference Cisco Unified IP Phone 9971, Cisco IP Video Phone E20 and Cisco Unified IP Phone 7985 using Tandberg Codian Adhoc Bridge registered to Unified Communications Manager	Verify Video Communication Server (VCS) endpoints can call Cisco Unified IP Phone 8941 registered to Unified Communications Manager.		Passed	
UC861IF.VID.034	Cisco TelePresence System 1000 Joins Adhoc Software Bridge registered to Cisco Unified Communications Manager	Verify Cisco TelePresence System 1000 is able to view other participants video after joining Adhoc Tandberg codian bridge.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.035	Cisco TelePresence System 1700 MXP, Cisco TelePresence System 1000 and Unified IP Phone 9971 are able to join Adhoc Tandberg Codian Conference	Verify presentation shared on Cisco TelePresence System 1000 can be viewed on other conference endpoints.		Passed	
UC861IF.VID.036	Intercluster Video Conference using Adhoc Bridge	Verify if Unified IP phone 9971 across SIP trunk is able to join Adhoc conference.		Passed	
UC861IF.VID.037	Tandberg 7985 with Trusted Relay Point joins Tandberg Codian conference	Verify that Tandberg 7985 registered to Unified Communications Manager with a Trusted Relay Point can join a Tandberg Codian/Unified Communications Manager conference.		Passed	
UC861IF.VID.038	Adhoc Conference with Cisco Unified Communications Manager and Cisco Unified Communications Manager Express endpoints	Verify if Adhoc conference works with Unified Communications Manager and Unified CME endpoints.		Passed	
UC861IF.VID.039	Adhoc Conference with Unified Communications Manager and Conference Share	Verify Adhoc conference with Cisco TelePresence System (CTS), Unified IP Phone 8941 and Cisco VCS endpoint and share the presentation on Cisco TelePresence MOVi.		Passed	
UC861IF.VID.040	Verify Client Services Framework (CSF) clients are able to join Cisco TelePresence MCU Adhoc Conference	Verify Unified Personal Communicator, Cisco UC Integration(TM) for Microsoft Office Communicator and Cisco TelePresence MoviT are able to join Adhoc conference.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.041	Hold and Resume on Cisco Unified IP Phone 9971 while in Adhoc Conference	Verify whether hold and resume can resume the video on Unified IP Phone 9971 while the endpoint has joined Adhoc conference.		Passed	
UC861IF.VID.049	Verify Intercluster call between Cisco IP Video Phone E20 registered Native to Unified Communications Manager and Cisco Unified IP Phone 7985.	Verify Intercluster SIP call between Cisco IP Video Phone E20 and Cisco Unified IP Phone 7985.		Passed	
UC861IF.VID.050	Unified Communications Manager calls Tandberg Codian conference and joins conference	Verify if Unified Communications Manager can call Tandberg Codian conference and is able to join the conference.		Passed	
UC861IF.VID.051	Hold and Resume with Cisco TelePresence Quick Set C20	Verify if hold and resume works with Cisco TelePresence Quick Set C20 that is registered as third party SIP endpoint.		Passed	
UC861IF.VID.052	Video Interoperability with Unified IP Phone 8941	Verify VCS endpoints can call Cisco Unified IP Phone 8941 registered to Unified Communications Manager.		Passed	
UC861IF.VID.053	Cisco Unified IP Phone 9900 Series interoperability with secure Cisco TelePresence System and Cisco Telepresence Multipoint Switch	Verify the inter working of Secure Cisco TelePresence System and Cisco Telepresence Multipoint Switch with unsecure Cisco Unified IP Phone 9900 Series end points joining Cisco Telepresence Multipoint Switch through Media Experience Engine.	Secure Cisco TelePresence System- Unified CM1-Secure SIP Trunk-Unified CM2-Secure SIP Trunk- Cisco Telepresence Multipoint Switch; Unified IP Phone 9900 Series-Unified CM1-SIP Trunk-Unified CM2-SIP Trunk-Cisco Telepresence Multipoint Switch	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.054	Secure Cisco TelePresence System interoperability with Media Experience Engine and Unified 9971 IP Phones	Verify the ability to place a Peer-to-Peer (P2P) call between secure Cisco TelePresence System and unsecure Unified 9971 IP Phones through Media Experience Engine.		Passed	
UC861IF.VID.055	Secure Cisco TelePresence System end point interoperability with SIP Tandberg end points behind Cisco TelePresence Video Communication Server (VCS)	Verify that a secure Cisco TelePresence System end point can make a Peer-to-Peer video call with a non-secure Tandberg video end points behind Cisco TelePresence Video Communication Server.		Passed	
UC861IF.VID.056	Secure Cisco TelePresence System interaction with Cisco TelePresence Movi¿ client behind Cisco TelePresence Video Communication Server (VCS)	Verify the interaction between secure Cisco TelePresence System and Cisco TelePresence Movi¿ client and ensuring that the client can share its desktop.		Passed	
UC861IF.VID.057	Cisco TelePresence System Security with Non secure SIP Trunks	Verify that Cisco TelePresence is able to call across non secure SIP Trunks without secure RTP enabled and still have a secure media path with Cisco TelePresence Multipoint Switch using Datagram Transport Layer Security.	Secure Cisco TelePresence System1---Unified CM---SIP Trunk-Unified CM---SIP Trunk---Secure Cisco TelePresence Multipoint Switch; Secure Cisco TelePresence System 2---Unified CM--SIP Trunk-Unified CM---SIP Trunk---Secure Cisco TelePresence Multipoint Switch	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.058	Scheduled conference using Cisco TelePresence server and Presentation Sharing using Cisco TelePresence MOVi	Verify whether Cisco TelePresence MOVi, Cisco TelePresence System 3000, Cisco Unified IP Phone 7945G can join Cisco TelePresence server conference and view Cisco TelePresence MOVi presentation share, given that Cisco Unified IP Phone 7945G should be able to hear audio of all conference participants.	Polycom HDX; Cisco CIUS ----- SIP Trunk ----Video Communication Server ---Cisco TelePresence Server --- SIP trunk ----Cisco TelePresence 1700 MXP	Passed	
UC861IF.VID.059	Attend Scheduled conference using Cisco TelePresence server	Verify whether Cisco TelePresence System, Cisco Cius, Unified IP Phone 9971, Cisco TelePresence 1700 MXP and Polycom HDX are able to attend scheduled conference on Cisco TelePresence server.	Polycom HDX; Cisco Cius;Unified IP Phone 9971--- SIP Trunk ----VCS ---Cisco TelePresence Server ----- SIP trunk ----Cisco TelePresence 1700 MXP	Passed	
UC861IF.VID.060	SIP- SIP call with Cisco TelePresence Video Communication Server via Session Manager Edition Works	Verify whether video works fine when the call is placed from Cisco Unified Communications Manager to Cisco TelePresence Video Communications Server via Session Manager Edition.	Unified IP Phone 89xx/99xx --Unified CM -SIP -SME --SIP -Cisco TelePresence VCS---Cisco IP Video Phone E20	Passed	
UC861IF.VID.061	Call Hold /Resume work with Cisco TelePresence Video Communication Server via Session Manager Edition Works	Verify whether the call placed from Unified IP Phone 9971 to Cisco TelePresence Video Communication Server via Session Manager Edition is able to hold and resume the call.	Unified IP Phone 9971 --Unified CM -SIP -SME --SIP -Cisco TelePresence VCS---Cisco IP Video Phone E20 ---Hold and resume on Unified IP Phone 9971	Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VID.062	Call from Unified Communications Manager to Cisco TelePresence Video Communications Server via Session Manager Edition works	Verify whether Inter Cluster Trunk-SIP interoperability with Session Manager Edition and Cisco TelePresence Video Communications Server results in bi-directional video.	Unified IP Phone 9971 --Unified CM -Inter Cluster Trunk -SME --SIP -Cisco TelePresence VCS---Cisco IP Video Phone E20 ---Hold and resume on Unified IP Phone 9971	Passed	
UC861IF.VID.063	Call from Cisco Unified Communication Manager to Cisco TelePresence Video Communication Server via Session Manager Edition with Early offer trunk	Verify bi-directional video between Unified Communications Manager - Session Manager Edition - Cisco TelePresence Video Communication Server when SIP trunk is set to early offer on both SIP trunks.	Unified IP Phone 9971 --Unified CM -SIP(EO) -SME --SIP(EO) -Cisco TelePresence VCS---Cisco IP Video Phone E20 ---Hold and resume on Unified IP Phone 9971	Passed	
UC861IF.VID.064	Scheduled conference using Cisco TelePresence server and presentation sharing using Cisco TelePresence MOVi / Cisco TelePresence Ex90	Verify whether Cisco TelePresence Ex90, Cisco TelePresence System 1000, Cisco TelePresence System 500 and Cisco TelePresence Ex90 can join Cisco TelePresence server conference and view Cisco TelePresence EX90 and Cisco TelePresence MOVi presentation sharing.	Polycom HDX; Cisco Cius ----- SIP Trunk ----Video Communication Server --- Cisco TelePresence Server ----- SIP trunk ----Cisco TelePresence 1700 MXP	Passed	
UC861IF.VXC.001	Independent Computing Architecture (ICA) Standalone, Mouse, USB KB, and two monitors Powers On and Works via 802.3AT Power Over Ethernet	Verifies that the ICA standalone, the USB mouse, USB KB and two monitors power on and all peripherals work properly via 802.3AT PoE		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VXC.002	PC over IP Standalone, USB Mouse, USB KB, and Two Monitors Powers On and Works via 802.3AT Power over Ethernet	Verify that the PC over IP (PCoIP) standalone, USB mouse, USB KB, and two monitors powers on and all peripherals work properly via 802.3AT Power over Ethernet		Passed w/ Exception	CSCtn12208
UC861IF.VXC.003	Detect the Accessory USB Flash Drive when Device is Operational	Verify to ensure that a user accessing a Voice Mail using VDI/VXC is able to plug in a USB flash drive and access data from it.		Failed	CSCtl74889
UC861IF.VXC.004	UC Integration@ for Microsoft Office Communicator is desk phone mode accessed using a Virtualization Experience Client (VXC 2111)	Verify that UC IntegrationT for Microsoft Office Communicator in deskphone mode works seamlessly when controlled over a Virtual Desktop Interface (VDI) interface, and audio quality of Visual Voice Mail played from the Voice Mail is good.		Passed	
UC861IF.VXC.005	Verify power to all USB ports on Virtual Desktop Interface (VDI)/ Virtualization Experience Client (VXC) standalone	Verify that all USB ports on VDI/VXC Standalone have power when power to VDI/VXC is provided via Power Over Ethernet at switch and power brick.		Passed	
UC861IF.VXC.006	Verify Virtualization Experience Client (VXC) - PC over IP Admin Graphical User Interface Functionality	Verifies if the client can move from Kiosk mode to non-kiosk mode, and whether features under the diagnostics options work for the Admin Graphical User Interface in VMWARE View client on the Virtual Desktop Infrastructure (VDI)/ Virtualization Experience Client (VXC) device. Verifies VMWARE View options for "Auto Launch if only one desktop".		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VXC.007	VXC-ICA Classic Desktop Graphical User Interface Tests	Verify the Independent Computing Architecture (ICA) Classic Desktop Graphical User Interface is user friendly and functions as expected.		Passed	
UC861IF.VXC.008	VXC-ICA Zero Launchpad Graphical User Interface Tests	Verify the ICA Zero Launchpad Graphical User Interface is user friendly and functions as expected.		Passed	
UC861IF.VXC.009	NGPoE switch operational with Cisco Unified IP Phone 9971, camera, PC over IP (PCoIP) Zilch Backpack, four USB peripherals, two monitors, and external speakers	Verify that Cisco Unified IP Phone 9971, camera, PC over IP Zilch BackPack, four USB peripherals, two monitors, and external speakers are powered and operational when plugged into an NGPoE switch.		Passed	
UC861IF.VXC.010	NGPoE switch operational with Cisco Unified IP Phone 9971, camera, Independent Computing Architecture Virtualization Experience Client BackPack, four USB peripherals, two monitors, and external speakers	Verify that Cisco Unified IP Phone 9971, camera, Independent Computing Architecture Virtualization Experience Client backpack, four USB peripherals, two monitors, and external speakers is powered and operational when plugged into an NGPoE switch.		Passed	
UC861IF.VXC.011	Independent Computing Architecture: Camera Disabled on Cisco Unified Communications Manager pages but plugged in	Verify that when camera is disabled via Unified Communications Manager but plugged in, the Cisco Unified IP Phone 9971 Independent Computing Architecture backpack operates with the power specifications of a Unified IP Phone 9971 without camera on 802.3 AT.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VXC.012	Virtualization Experience Client (VXC) verifying peripherals come up after Switch Reset	Verify that phone, VXC backpack and VXC standalone come up with all peripherals powered up after the switch port supplying power is reset.		Passed	
UC861IF.VXC.013	Verify PC Over IP with Secure Socket Layer (SSL) Connection with Backpack and Standalone	Verify that backpack and standalone devices are able to connect to the view connection server using SSL.		Passed	
UC861IF.VXC.014	Use Real-Time Monitoring Tool (RTMT) application on Zilch PC over IP and Independent Computing Architecture	Verify that RTMT application for collecting logs and monitoring Cisco CallManager application works on Zilch backpack and Standalone PC over IP and Independent Computing Architecture.		Passed	
UC861IF.VXC.015	NGPoE and max Key Expansion Module config (Cisco Unified IP Phone 9971 with camera, USB mouse, USB keyboard, one monitor, and three Key Expansion Modules)	Verify a Virtualization Experience Client backpack on NGPoE with Cisco Unified IP Phone 9971 with camera, USB mouse, USB keyboard, one monitor, and three Key Expansion Modules powers on correctly and is operational.		Passed	
UC861IF.VXC.016	NGPoE with Independent Computing Architecture stand-alone, four USB peripherals, two monitors, and external speakers	Verify that an Independent Computing Architecture stand-alone, four USB peripherals, two monitors, and external speakers are powered and operational when plugged into an NGPoE switch.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VXC.017	Not Enough Power for Camera Scenario	Verify that when the backpack is running with a Unified IP Phone 9971, two USB, two monitors on 802.3AT (power is maxed out) and a camera is added, that the phone throws an error indicating there's not enough power for the camera and the backpack still operates normally.		Passed	
UC861IF.VXC.018	Quick Removal and Insertion of Multiple USB Devices	Verify that USB devices can be interchanged quickly on the same port with no adverse affects.		Passed	
UC861IF.VXC.019	Backpack behavior with power negotiation disabled via Unified Communications Manager	Verify that the backpack powers on within the 802.3 AT specifications when power negotiation is disabled via Unified CM.		Passed	
UC861IF.VXC.021	Upgrade Independent Computing Architecture firmware using Virtualization Experience Client (VXC) Manager	Verify the ability to upgrade an Independent Computing Architecture (ICA) backpack and ICA stand-alone by pointing the device to a VXC Manager file server.		Passed	
UC861IF.VXC.022	Bluetooth mouse and USB mouse can be used at same time	Verify Bluetooth USB mouse and wired USB mouse can be used at same time on the PC over IP (PCoIP) and Independent Computing Architecture (ICA) units.		Passed	
UC861IF.VXC.023	Independent Computing Architecture: Swap monitors while logged into VoiceMail	Verify that a monitor of a different size can be swapped and the screen auto-corrects while a stand-alone Virtualization Experience Client unit is logged into an Independent Computing Architecture VoiceMail.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.VXC.024	PC over IP: Swap monitors while logged into VoiceMail	Verify that a monitor of a different size can be swapped and the screen auto-corrects while a stand-alone Virtualization Experience Client unit is logged into a PC over IP Voicemail.		Passed	
UC861IF.VXC.025	NG Power over Ethernet with PC over IP stand-alone, Four USB Peripherals, Two Monitors, and External Speakers	Verify that a PC over IP stand-alone, four USB peripherals, two monitors, and external speakers are powered and operational when plugged into an NG Power over Ethernet switch.		Passed	

Regression Tests

Project Features Tested	Total Test Cases	% of Total	Passed	% Pass	Pass w/ X	% Pass w/ X	Failed	% Failed
FARE	1421		1,389	97.70%	2	0.20%	30	2.10%
Auto Express	88	6.19%	88	100.00%	0	0.00%	0	0.00%
CCM-BASIC	228	16.05%	228	100.00%	0	0.00%	0	0.00%
CCM-CFWD	33	2.32%	33	100.00%	0	0.00%	0	0.00%
CCM-CONF	60	4.22%	60	100.00%	0	0.00%	0	0.00%
CCM-EMOB	7	0.49%	7	100.00%	0	0.00%	0	0.00%
CCM-INTER	18	1.27%	18	100.00%	0	0.00%	0	0.00%
CCM-MISC	86	6.05%	86	100.00%	0	0.00%	0	0.00%
CCM-SHARED	32	2.25%	32	100.00%	0	0.00%	0	0.00%
CCM-XFER	41	2.89%	40	97.50%	0	0.00%	1	2.50%
CME-BASIC	14	0.99%	14	100.00%	0	0.00%	0	0.00%
CME-CFWD	21	1.48%	21	100.00%	0	0.00%	0	0.00%
CME-CONF	38	2.67%	38	100.00%	0	0.00%	0	0.00%
CME-MISC	13	0.91%	13	100.00%	0	0.00%	0	0.00%
CME-XFER	28	1.97%	27	96.40%	0	0.00%	1	3.60%
CUE	14	0.99%	14	100.00%	0	0.00%	0	0.00%
ENDPOINTS	2	0.14%	2	100.00%	0	0.00%	0	0.00%
FAILOVER	18	1.27%	18	100.00%	0	0.00%	0	0.00%
FAXMOD	40	2.81%	40	100.00%	0	0.00%	0	0.00%
GW-SIP	8	0.56%	8	0.00%	0	0.00%	0	0.00%
ICT	25	1.76%	24	96.00%	0	0.00%	1	4.00%
INTEROP	39	2.74%	39	100.00%	0	0.00%	0	0.00%
IPCCX	79	5.56%	78	98.70%	0	0.00%	1	1.30%
IPMA	0	0.00%	0	100.00%	0	0.00%	0	0.00%
MP	8	0.56%	8	100.00%	0	0.00%	0	0.00%
MPE	31	2.18%	31	100.00%	0	0.00%	0	0.00%
New for Toledo	11	0.77%	11	100.00%	0	0.00%	0	0.00%
QOS	96	6.76%	71	74.00%	0	0.00%	25	26.00%
SECURITY	53	3.73%	53	100.00%	0	0.00%	0	0.00%
SRST	47	3.31%	47	100.00%	0	0.00%	0	0.00%
UNC	45	3.17%	45	100.00%	0	0.00%	0	0.00%
UNITY	90	6.33%	88	97.80%	2	2.20%	0	0.00%
VIDEO	41	2.89%	41	100.00%	0	0.00%	0	0.00%
WAN	6	0.42%	6	100.00%	0	0.00%	0	0.00%

Project Features Tested	Total Test Cases	% of Total	Passed	% Pass	Pass w/ X	% Pass w/ X	Failed	% Failed
WIRELESS	3	0.21%	3	100.00%	0	0.00%	0	0.00%
Manual Regression	58	4.08%	57	98.30%	0	0.00%	1	1.70%