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 Retrievel 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 botht he 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	Veirfy 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 Comunications 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 Comunications 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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	

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.

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.060	Mutiway 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	
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	Failed	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	Failed	
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 Messenging interoperability with Cisco Unified Personal Communicator 7 user, multiple Instant Messenging sessions	Verify the ability to begin an Instant Messenging chat with a Cisco Unified Personal Communicator 7 user, and when the chat is ongoing, initiate an Instant Messenging 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 Messenging Interoperability with Cisco Unified Personal Communicator 8 user sending offline messages	Verify the ability to begin an Instant Messenging 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	
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 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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.630	Call to Cisco Unity Connection via PSTN trunk with Cisco Unity Connection Transfering Call to Cisco Cius	Verify call to Cisco Unity Connection via PSTN trunk with Cisco Unity Connection transfering 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	
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

ID	Case Title	Description	Call Component Flow	Status	Defects
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	

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 Commujications 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	

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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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 -->Musci 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	Failed	

ID	Case Title	Description	Call Component Flow	Status	Defects
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 Unfied 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 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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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.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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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 Transfering 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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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 Interlcuster Destination	Verify that Andorid client can set up a conference call, when the other parties of conference call are iPhone client and an IP phone across interlcuster 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	
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	

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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->SME2->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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	Verfiy 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	Verfiy 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

ID	Case Title	Description	Call Component Flow	Status	Defects
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 successfull	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>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):MXP 1700(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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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 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	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	
UC861IF.VID.061	Call Hold /Resume work with Cisco TelePresence Video Communication Server via Session Manager Edition Works	erify 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	

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	Intercluser 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 Unfiied 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.

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	

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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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 Retrievel 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 botht he 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	Veirfy 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 Comunications 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 Comunications 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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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/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->SME2->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 Manangement 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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	Verfiy 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	Verfiy 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

ID	Case Title	Description	Call Component Flow	Status	Defects
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 successfull	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>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->CenVCB->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>RemVCB>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):MXP 1700(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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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

ID	Case Title	Description	Call Component Flow	Status	Defects
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	Intercluser 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 Unfiied 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.

ID	Case Title	Description	Call Component Flow	Status	Defects
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.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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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 Commujications 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 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	Failed	
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
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	Failed	
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 Messenging interoperability with Cisco Unified Personal Communicator 7 user, multiple Instant Messenging sessions	Verify the ability to begin an Instant Messenging chat with a Cisco Unified Personal Communicator 7 user, and when the chat is ongoing, initiate an Instant Messenging 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 Messenging Interoperability with Cisco Unified Personal Communicator 8 user sending offline messages	Verify the ability to begin an Instant Messenging 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	
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 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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.CUS.630	Call to Cisco Unity Connection via PSTN trunk with Cisco Unity Connection Transfering Call to Cisco Cius	Verify call to Cisco Unity Connection via PSTN trunk with Cisco Unity Connection transfering 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	
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

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
UC861IF.MOB.012	Android Mobility Client joining Meeting and Transfering 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 Interlcuster Destination	Verify that Andorid client can set up a conference call, when the other parties of conference call are iPhone client and an IP phone across interlcuster 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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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 -->Musci 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	Failed	
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 Unfied 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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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 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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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 between secure Cisco TelePresence System and unsecure Unified 9971 IP Phones through Media Experience Engine.		Passed	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
UC861IF.VID.061	Call Hold /Resume work with Cisco TelePresence Video Communication Server via Session Manager Edition Works	erify 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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	
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	

ID	Case Title	Description	Call Component Flow	Status	Defects
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	

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%
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%
		100.00%						