ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC702EF.ARC.029	Unified Attendant Server	Unified Attendant Server IP phone Loses Network Connectivity	Verification of Unified Attendant Server attendant IP phone losing network connectivity.		Passed	
UC702EF.ARC.031	Unified Attendant Server	Unified Attendant Server Client Loses Network Connectivity When Active on a Call and Re-establishes Connection After Some Time	Verify the following: Make a call from PSTN to operator. Operator answers the call and is initiating a transfer to a SCCP phone in remote site. Disconnect the operator from the network when the transfer is going on and examine the behavior. Bring back the system to normal after few minutes and check if the operator is back to normal.		Passed	
UC712EF.ARC.012	Unified Attendant Server	Server Application is Restarted with Different CTI Manager as Primary	Verify the following: Unified Attendant Server is registered to Unified CM1 as primary & Unified CM2 as secondary. Purposefully interchange the settings on Unified Attendant Server so as to make Unified CM2 as primary and Unified CM1 as secondary. Restart the Unified Attendant Server .		Passed	
UC712EF.ARC.024	Unified Attendant Server	Retrieval of Call Parked by Operator		Unified IP Phones 6921/6941/6961 PH1->Unified CM->OP Console->Call Parking;Unified IP Phones 6921/6941/6961 PH 2->Park Retrieval.	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713EF.ARC.056	Unified Attendant Server	Multichannel Conferencing by Operator Console		Phone 1->Unified CM->Op Console;Opconsole->Unified CM->Conference->Extension A;Opconsole->Unified CM->Conference->Extension B	Passed	
UC802EF.ARC.005	Unified Attendant Server	Unified IP Phone 69x1 as Operator Endpoint	Verify if IP Phone 69x1 acting as Operator end point can Blind transfer (XFER_B) the incoming call to UC Integration™ for Microsoft Office Communicator clients.	Stage 1: SIP Ph 1->Unified CM->Op Cons 1 (IP Phone 69xx); Stage 2: Op Cons 1->XFER_B->UC Integration™ for Microsoft Office Communicator	Passed w/ Exception	The UC Integration™ for Microsoft Office Communicator end point is replaced with SIP/SCCP endpoint.
UC802EF.ARC.006	Unified Attendant Server	BLF Monitoring in Unified Attendant Server	Verify the following: A video call made to Unified Enterprise Attendant Console (CUEAC) where IP Phone 6961 with Camera acts as the Operator. This video call is transfer to another IP Phone 9971 with Camera in the same network .The BLF status on the IP Phone 9971 with Camera changes to busy.	Stage 1: Video Ph 1->Unified CM->OP Cons 1 (6961+CAM); Stage2: Op Cons 1(6961+CAM)->XFER_C->Video Ph 2	Passed	
UC802EF.ARC.007	Unified Attendant Server	Cisco Unity as Queue Overflow Designations	Verifying the overflow of Unified Enterprise Attendant Console (CUEAC) Queue calls transfer to Voice mail.	Stage 1 : SCCP Ph 1->Unified CM->OP Cons 1(UC Integration™ for Microsoft Office Communicator);Stage 2: OP Cons 1->Voice mail.	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.ARC.008	Unified Attendant Server	E164 Support in Unified Attendant Server	Verify if an international call from Phone 1 can call Operator of Unified Enterprise Attendant Console (CUEAC) .The CUEAC Consult Transfer (XFER_C) the call to PSTN Phone. The CUEAC operator must show the E164 numbering of the calling party.	Stage 1 : SCCP Ph 1->Unified CM->OP Cons 1(SIP Ph 1);Stage 2 : OP Cons 1->XFER_C->Unified CM->MGCP PRI GW->PSTN->MGCP PRI GW->Unified CM->Rem SIP Ph 1	Passed	
UC802EF.ARC.010	Unified Attendant Server	Enhanced Night Service in Unified Attendant Server	Verify if all calls to the Operator lands on a phone designated on the Admin page to attend Night service calls, when Unified Enterprise Attendant Console operator IP Phone 9971 is in Night service mode.	Stage 1 : SIP Ph 1->Unified CM->OP Cons Ph 1(9971 Phone)->Night service;Stage 2: SIP Ph1->Unified CM->MGCP BRI GW->PSTN->MGCP BRI GW->Unified CM->Rem SCCP Ph 1	Passed w/ Exception	Only basic night service is available in Unified Attendant Console build for UC 8.0 release.
UC802EF.ARC.011	Unified Attendant Server	Enhanced Emergency Mode in Unified Attendant Server	Verify is all calls to the Operator lands on the Emergency number designated on Admin page, when Unified Enterprise Attendant Console operator is set to Emergency mode.	Stage 1 : SIP Ph 1->Unified CM->OP Cons Ph (9971 Phone)->Emergency Mode;Stage 2: SIP Ph1->Unified CM->MGCP BRI GW->PSTN->MGCP BRI GW->Unified CM->Rem SCCP Ph 1	Passed w/ Exception	Only basic emergency service is available in Unified Attendant Console build for UC 8.0 release.
UC802EF.ARC.014	Unified Attendant Server	Unified IP Phone 7931 as Operator Endpoint	Verify if Unified IP Phone 7931 Phone acting as Operator end point can consult transfer (XFER_C)s the incoming call to SIP phone across clusters.	SIP Ph 1->Unified CM->Op Cons 1 (7931 Ph); Stage 2: Op Cons 1->XFER_C->Unified CM1->ICT->Unified CM2->SIP Ph.	Passed w/ Exception	Replaced 7931 Phones with 6961 phones since 7931 phones are not supported as Operator Console.

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.ARC.018	Unified Attendant Console	Unified IP Phone 6901/6911 as Operator End Points	Verify if Unified IP Phone 6901/6911 can act as operator end point and can blind transfers (XFER_B) the incoming call to UC Integration for Microsoft Office Communicator Clients.	Stage 1: SIP Ph 1->Unified CM->Op Cons 1 (IP Phone 69xx); Stage 2: Op Cons 1->XFER_B->UC Integration™ for Microsoft Office Communicator 1	Passed	
UC802EF.ARC.019	Unified Attendant Server	BLF Monitoring in Unified Attendant Server	Verify if a video call made to Unified Business Assistant Console where SCCP Phone with camera is the Operator. This video call is transferred to another SCCP Phone 2 with Camera in the same network. The BLF status on the SCCP Phone 2 with Camera changes to busy.	Stage 1: Video Ph 1->Unified CM->OP Cons 1 (SCCP Ph+CAM); Stage2: Op Cons 1(SCCP Ph+CAM)->XFER_C->Video Ph 2	Passed	
UC802EF.ARC.020	Unified Attendant Server	Cisco Unity as Queue Overflow Designations	Verify the overflow of Unified Business Attendant Console Queue calls transfer to Voice mail.	Stage 1 : SCCP Ph 1->Unified CM->OP Cons 1(UC Integration™ for Microsoft Office Communicator);Stage 2: OP Cons 1->Voice mail.	Passed	
UC802EF.ARC.021	nified Attendant Server	E164 Support in Unified Attendant Server	Verify if an international call from Phone 1 can call Operator of Unified Business Attendant Console. The Unified Business Attendant Console Consult transfers (XFER_C) the call to PSTN Phone. The Unified Business Attendant Console operator must display the E164 numbering of the calling party.	Stage 1 : SCCP Ph 1->Unified CM->OP Cons 1(SIP Ph 1);Stage 2 : OP Cons 1->XFER_C->Unified CM->MGCP PRI GW->PSTN->MGCP PRI GW->Unified CM->Rem SIP Ph 1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.ARC.023	Unified Attendant Server	Enhanced Night Service in Unified Attendant Server	Verify if all calls to the Operator lands on a phone designated on the Admin page to attend Night service calls, when Unified Business Attendant Console operator IP Phone 9971 is in Night service mode.	Stage 1 : SIP Ph 1->Unified CM->OP Cons Ph 1(9971 Phone)->Night service;Stage 2: SIP Ph1->Unified CM->MGCP BRI GW->PSTN->MGCP BRI GW->Unified CM->Rem SCCP Ph 1	Passed w/ Exception	Only basic night service is available in Unified Attendant Console build for UC 8.0
UC802EF.ARC.024	Unified Attendant Server	Enhanced Emergency Mode in Unified Attendant Server	Verify is all calls to the Operator lands on the Emergency number designated on Admin page, when Unified Business Attendant Console operator is set to Emergency mode.	Stage 1 : SIP Ph 1->Unified CM->OP Cons Ph (9971 Phone)->Emergency Mode;Stage 2: SIP Ph1->Unified CM->MGCP BRI GW->PSTN->MGCP BRI GW->Unified CM->Rem SCCP Ph 1	Passed w/ Exception	Only basic emergency service is available in Unified Attendant Console build for UC 8.0
UC802EF.ARC.026	Unified Attendant Server	Unified IP Phone 7931 as Operator End Point	Verify if Unified IP Phone 7931 as an Operator end point can Consult transfers (XFER_C)s the incoming calls to SCCP phone.	SIP Ph 1->Unified CM->Op Cons 1 (7931 Ph); Stage 2: Op Cons 1->XFER_C->Unified CM1->ICT->Unified CM2->SCCP Ph.	Passed	
UC802EF.ARC.027	Unified Attendant Server	Unified IP Phone 7931 as Operator End Point	Verify if Unified IP Phone 7931 as Operator end point can blind transfers (XFER_B)s the incoming call to another SCCP phone.	SIP Ph 1->Unified CM->Op Cons 1 (7931 Ph); Stage 2: Op Cons 1->XFER_B->Unified CM1->ICT->Unified CM2->SCCP Ph.	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.ARC.028	Unified Attendant Server	Server Application Restarted with Different CTI Manager as Primary	Verify the following when: Unified Attendant Server is registered to Unified CM1 as primary & Unified CM2 as secondary, purposefully interchange the settings on Unified Attendant Server so as to make Unified CM2 as primary and Unified CM1 as secondary. Restart the Unified Attendant server.		Passed w/ Exception	This TC was executed without having a secondary CTI manager in the Testbed.
UC802EF.ARC.029	Unified Attendant Server	Unified Attendant Console Client Loses Network Connectivity When Active	Verify the following: Make a call from PSTN to operator. When operator answers the call and initiates a transfer to a SCCP phone in remote site, disconnect the operator from the network and examine the behavior. Bring back the system to normal after few minutes and check if the operator is back to normal.		Passed	
UC802EF.ARC.030	Unified Attendant Server	Unified Attendant Console IP Phone Loses Network Connectivity	Verify if the Unified Attendant Console IP Phone loses network connectivity.		Passed	
UC802EF.ARC.031	Unified Attendant Server	Multichannel Conferencing by Operator Console	Verify if multichannel Conferencing by operator console is possible.	Phone 1->Unified CM->Op Console;Opconsole->Unified CM->Conference->Extension A;Opconsole->Unified CM->Conference->Extension B	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.ARC.032	Unified Attendant Server	Retrieval of Call Parked by Operator	Verify if retrieval of call parked by operator is possible.	Unified IP Phones 6921/6941/6961 PH1->Unified CM->OP Console->Call Parking;Unified IP Phones 6921/6941/6961 PH 2->Park Retrieval.	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IA.ISAC.501	Codec Protocols	iSAC Codec: Verify Calls Between Two Endpoints	Verify if phone A can call Phone B and also if Phone B can call Phone A.		Passed	
UC802IA.ISAC.502	Codec Protocols	iSAC Codec: Verify Hold and Resume	Verify if during a call between Phone A and B, can hold/resume either Phone A or B three times.		Passed	
UC802IA.ISAC.503	Codec Protocols	iSAC Codec: Verify Calls Between Two Endpoints	Verify if phone A can call Phone B and also if Phone B can call Phone A.		Passed	
UC802IA.ISAC.601	Codec Protocols	iSAC Codec: Verify Multiple Calls on a single Line	Verify if EpAsstnt1 (assistant endpoint 1) makes first call to epFeature (feature applied endpoint) and EpAsstnt2 (assistant endpoint 2) makes second call to epFeature on the same line. EpFeature can switch the first and second call correctly.		Passed	
UC802IA.ISAC.602	Codec Protocols	iSAC Codec: Verify Multiple Calls on Multiple Lines on Same Phone	Verify if EpAsstnt1 (assistant endpoint 1) makes first call to epFeature (feature applied endpoint) and EpAsstnt2 (assistant endpoint 2) makes second call to epFeature on a different line. EpFeature can switch the first and second calls correctly.		Passed	
UC802IA.ISAC.603	Codec Protocols	iSAC Codec: Verify Call Forward All	Verify if epAsstnt1 (assistant endpoint 1) calls epFeature (feature applied endpoint) and epFeature forwards all call to epAsstnt2 (assistant endpoint 2).		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IA.ISAC.604	Codec Protocols	iSAC codec: Verify Call Forward Busy	Verify if epAsstnt1 (assistant endpoint 1) calls epFeature (feature applied endpoint). when epFeature is busy the call is forwarded to epAsstnt2 (assistant endpoint 2).		Passed	
UC802IA.ISAC.605	Codec Protocols	iSAC Codec: Verify Call Forward No Answer	Verify if epAsstnt1 (assistant endpoint 1) calls epFeature (feature applied endpoint), epFeature is ringing but there is no answer, then the call is forwarded to epAsstnt2 (assistant endpoint 2).		Passed	
UC802IA.ISAC.606	Codec Protocols	iSAC Codec: Verify Same Group <Normal> Pickup by Pickup Softkey	Verify if a user can setup epFeature (feature applied endpoint) and epAsstnt2 (assistant endpoint 2) in the same pickup group and Unified CM service parameter <Auto Call Pickup Enabled> as false. EpAsstnt1 (assistant endpoint 1) calls epAsstnt2. During epAsstnt2 ringing EpFeature push PickUp softkey. After the ringing EpFeature goes off-hook to connect the call.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IA.ISAC.607	Codec Protocols	iSAC Codec: Verify <Normal> Group Pick up by GPickUp softkey	Verify if a user can setup epFeature (feature applied endpoint) and epAsstnt2 (assistant endpoint 2) in the different pickup group and <Auto Call Pickup Enabled> as false in Unified CM. EpAsstnt1 (assistant endpoint 1) calls epAsstnt2. During epAsstnt2 ringing. EpFeature push GPickUp softkey and dials the pick code. After ringing epFeature goes off hook and connect the call.		Passed	
UC802IA.ISAC.608	Codec Protocols	iSAC Codec: Verify <Normal> Other Group Pick Up by OPickUp softkey	Verify is a user can setup epAsstnt2 (assistant endpoint 2) in an associate pickup group with epFeature (feature applied endpoint) and Unified CM service parameter <Auto Call Pickup Enabled> as false. EpAsstnt1 (assistant endpoint 1) calls epAsstnt2. During epAsstnt2 ringing EpFeature pushes OPickUp softkey. After the ringing EpFeature goes off-hook to connect the call.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.IME.001	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Audio call from Cluster with Inline ASA to Another Cluster with Inline ASA	Verify a Cisco IME audio call from cluster with inline ASA to another cluster with inline ASA.	Originating Phone->Cisco IME->CCM->Inline ASA->SIP Trunk->Inline ASA->CCM->Cisco IME->Terminating Phone	Passed	
UC802EF.IME.002	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Audio call from Cluster with Inline ASA to Another Cluster with Off-Path ASA	Verify a Cisco IME audio call from cluster with inline ASA to another cluster with off-path ASA.	Originating Phone->Cisco IME->CCM->Off-path ASA->SIP Trunk->Off-path ASA->CCM->Cisco IME->Terminating Phone	Passed	
UC802EF.IME.003	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Audio Call from Cluster with Off-Path ASA to Another Cluster with Inline ASA	Verify a Cisco IME audio call from cluster with off-path ASA to another cluster with inline ASA.	Originating Phone->Cisco IME->CCM->Inline ASA->SIP Trunk->Off-path ASA->CCM->Cisco IME->Terminating Phone	Passed	
UC802EF.IME.004	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Video Call from Cluster with Inline ASA to Another Cluster with Inline ASA	Verify a Cisco IME video call from cluster with inline ASA to another cluster with inline ASA.	Originating Video Phone->Cisco IME->CCM->Inline ASA->SIP Trunk->Inline ASA->CCM->Cisco IME->Terminating Video Phone	Passed	
UC802EF.IME.005	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Video Call from Cluster with Inline ASA to Another Cluster with Off-Path ASA	Verify a Cisco IME video call from cluster with inline ASA to another cluster with off-path ASA.	Originating Video Phone->Cisco IME->CCM->Inline ASA->SIP Trunk->Off-path ASA->CCM->Cisco IME->Terminating Video Phone	Passed	
UC802EF.IME.006	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Video Call from Cluster with Off-Path ASA to Another Cluster with Inline ASA	Verify a Cisco IME video call from cluster with off-path ASA to another cluster with inline ASA.	Originating Video Phone->Cisco IME->CCM->Off-path ASA->SIP Trunk->Inline ASA->CCM->Cisco IME->Terminating Video Phone	Passed	
UC802EF.IME.007	ASA, Unified B2B Link/ Cisco Internet Media Engine	PSTN Fallback on Cluster with Inline ASA	Verify PSTN fallback on cluster with inline ASA when call is degraded.	Originating Phone->Cisco IME->CCM->Inline ASA->SIP Trunk->Inline ASA->CCM->Cisco IME->Terminating Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.IME.008	ASA, Unified B2B Link/ Cisco Internet Media Engine	PSTN Fallback on Cluster with Off-Path ASA	Verify PSTN fallback on cluster with off-path ASA when call is degraded.	Originating Phone->Cisco IME->CCM->Off-path ASA->SIP Trunk->Off-path ASA->CCM->Cisco IME->Terminating Phone	Passed	
UC802EF.IME.009	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call with Audio Escalation to Video	Verify Cisco IME call with audio escalation to video.	Originating Phone->Cisco IME->CCM->SIP Trunk->CCM->Cisco IME->Terminating Phone	Passed	
UC802EF.IME.010	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call with Hold/Resume	Verify Cisco IME call interaction with Hold/Resume.	Originating Phone->Cisco IME->CCM->SIP Trunk->CCM->Cisco IME->Terminating Phone (Hold/Resume)	Passed	
UC802EF.IME.011	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call Transfer	Verify Cisco IME call interaction with call transfers.	Originating Phone->Cisco IME->CCM->SIP Trunk->CCM->Cisco IME->Terminating Phone->Transfer->Destination Phone	Passed	
UC802EF.IME.012	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call Conference	Verify Cisco IME call interaction with conference calls.	Originating Phone->Cisco IME->CCM->SIP Trunk->CCM->Cisco IME->Terminating Phone->Conf->Destination Phone	Passed	
UC802EF.IME.013	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call with Shared Line	Verify Cisco IME call interaction with shared line phones.	Originating Phone->Cisco IME->CCM->SIP Trunk->CCM->Cisco IME->Shared Line Terminating Phone	Passed	
UC802EF.IME.014	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call can be Transferred to Another Cluster Over Inter-Cluster Trunks	Verify if a Cisco IME call can be transferred to another cluster over inter-cluster trunks.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->ICT->Unified CM->Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.IME.015	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call Gets Transferred to Originating Cluster over Inter-cluster Trunk	Verify if a Cisco IME call can be transferred to the originating cluster over inter-cluster trunks.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->ICT->Unified CM->Phone	Passed	
UC802EF.IME.016	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call gets Transferred to Originating Cluster as Cisco IME Call	Verify if a Cisco IME call can be transferred to the originating cluster as a Cisco IME call.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->SIP Trunk->Unified CM->Phone	Passed	
UC802EF.IME.017	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call gets Transferred to Unified CME site over H.323 Network	Verify if a Cisco IME call can be transferred to a Unified CME site over H.323 network.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->H.323 GK->H.323 CUBE->H.323 GK->CME->Phone	Passed	
UC802EF.IME.018	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call gets Transferred to Unified CME Site Aggregated by Unified SIP Proxy	Verify if a Cisco IME call can be transferred to a Unified CME site aggregated by Unified SIP Proxy.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->SIP Trunk->Unified SIP Proxy->CME->Phone	Passed	
UC802EF.IME.019	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call get Transferred to Unified CM-Interop Site	Verify if a Cisco IME call can be transferred to a phone in interop-site with Unified CM 7.x version.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->QSIG ICT->Unified CM 7.x->Phone	Passed	
UC802EF.IME.020	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call Transferred to PSTN Phone	Verify if a Cisco IME call can be transferred to a PSTN phone.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->Unified CM->PSTN GW->PSTN Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.IME.021	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call Transferred to PBX Phone	Verify if a Cisco IME call can be transferred to a PBX phone.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->Unified CM->QSIG PBX->PBX Phone	Passed	
UC802EF.IME.022	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call Transferred to a Dual-Stack Phone in a Dual-Stack IPv6 Aware Cluster	Verify if a Cisco IME call can be transferred to a dual-stack phone in a IPv6 aware cluster.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->QSIG ICT->DS Unified CM->DS Phone	Passed	
UC802EF.IME.023	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call Transferred to a Remote Branch Phone	Verify if a Cisco IME call can be transferred to a phone in remote branch.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->Unified CM->Remote Branch->Phone	Passed	
UC802EF.IME.024	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Negative Scenarios	Verify negative test scenarios with Cisco IME.		Passed	
UC802IF.IME.201	Cisco IME	Secure Conferencing Over Unified B2B Trunks With SRTP Enabled Inside the Firewall	Verify that secure conferencing can be placed over Cisco IME trunks. Also verify that if SRTP is enabled on the inside, ASA supports pass through and inserts as media intermediary.	Phone1->Unified CM1->Secure ConfBridge; Phone2->Unified CM2->Cisco IME ASA->SIPT ->Off Path Cisco IME ASA->Unified CM->Conf Bridge; Phone3->Unified CM->PSTN->GW->Unified CM->Conf Bridge	Passed	
UC802IF.IME.202	Cisco IME	Early Offer Over Unified B2B Trunks	Verify that Cisco IME trunks and Cisco IME aware ASA supports early offer.	Phone1->Unified CM1->Off Path Cisco IME ASA->MTP->SIPT (WAN)->Cisco IME ASA->Unified CM2->Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.IME.203	Cisco IME	Video Call Over Unified B2B Trunk From TRP Enabled Endpoint	Verify that video calls are supported over Cisco IME trunks for a TRP enabled endpoint.	Phone1->Unified CM1->Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM2->TRP->Phone2	Passed	
UC802IF.IME.204	Cisco IME	Hold/Transfer and MoH Playback Over Unified B2B Trunks	Verify that Unified B2B calls can be placed on hold and transferred. Also verify that MoH is available when the call is placed on hold.	Phone1->Unified CM1->Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM2->Phone2;	Passed	
UC802IF.IME.205	Cisco IME	Password Mismatch Between Unified B2B and ASA	Verify that an incoming call fails after the expiry of a ticket and the call is also denied when there is a mismatch in the password between Cisco IME server and ASA.	Phone1->Unified CM1->ASA->Cisco IME ASA->SIPT (WAN)->Cisco IME ASA->Unified CM2->Phone2;	Passed	
UC802IF.IME.206	Cisco IME	Expired and Revoked Tickets for Calls Over Unified B2B Trunks	Verify that Cisco IME calls get rejected if the tickets are expired and also Cisco IME calls when the ticket is revoked.	Phone1->Unified CM1->Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM2->IP Phone2;	Passed	
UC802IF.IME.207	Cisco IME	PSTN Fallback With Unified B2B	Verify the fallback to PSTN feature of Unified B2B when the quality of the voice call deteriorates. Also verify the fallback under various sensitivity settings for different codec's using different call quality condition.	Phone1->Unified CM1->Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM2->Phone2; PSTN Fallback: IP Phone1->Unified CM->GW->PSTN->GW->	Passed	
UC802IF.IME.212	Cisco IME	Unified CM Overlap Dialing Support and MTP Insertion for PSTN Fallback	Verify that Unified CM has support for overlap dialing with Unified B2B. Also verify that PSTN MTP insertion occurs when dynamically if there is a need during PSTN fallback.	Phone1->ASA->Unified CM1->Off-path Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM2->Phone2	Failed	CSCtf09331

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.IME.215	Cisco IME	PSTN Fallback for Conference Call Over a Unified B2B Link	Verify that a video conference call successfully falls back to an audio conference.	Phone1->Unified CM1->ConfBridge; Phone2->Unified CM2->Cisco IME ASA->SIPT->Off Path Cisco IME ASA->Unified CM->Conf Bridge; Phone2->Unified CM2->Cisco IME ASA->SIPT->Off Path Cisco IME ASA->Unified CM->Conf Bridge	Passed	
UC802IF.IME.216	Cisco IME	PSTN Fallback with Unified B2B Using SIP Gateway	Verify the fallback to PSTN feature of Unified B2B when the quality of the voice call deteriorates. Also to verify the fallback under various sensitivity settings for different codec's using different call quality condition.	Phone1->Unified CM1->Off path Cisco IME ASA->SIPT->Cisco IME ASA (MSP)->ASA->Unified CM (MSP)->Phone2; Phone3->Unified CM1->Off path Cisco IME ASA->SIPT->Cisco IME ASA (SEA)->Unified CM (SEA)->Phone4	Passed	
UC802IF.IME.217	Cisco IME	Unified B2B for a Phone in Unified SRST Site Phone	To verify that Unified B2B calls can be placed to a phone in Unified SRST mode.	Phone1->Unified CM1->Cisco IME ASA->Cisco IME SIPT->Off path Cisco IME ASA->Unified CM2->Call Fwd Unregister->GW->PSTN->SRST->Phone2	Passed	
UC802IF.IME.218	Cisco IME	Scheduled Conferencing With WebEx and Unified MeetingPlace Using Software Conference Resource	To verify that when a scheduled conference is held over Unified B2B link, rich media experience is available.	Phone1->Unified CM1->Off Path Cisco IME ASA->Cisco IME SIPT->Cisco IME ASA->Unified CM2->SIPT->Unified MeetingPlace; Phone2->Unified CM1->Off Path Cisco IME ASA->Cisco IME SIPT->Cisco IME ASA->Unified CM2->SIPT->Unified MeetingPlace; Phone3->Unified CM->SIPT->Unified MeetingPlace	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.IME.219	Cisco IME	Scheduled Conferencing With WebEx and Unified MeetingPlace Using Hardware Conference Resource	To verify that when a scheduled conference is held over Unified B2B link, rich media experience is available.	Phn1->Unified CM1->Off Path Cisco IME ASA->Cisco IME SIPT->Cisco IME ASA->Unified CM2->SIPT->Unified MeetingPlace; Phn2->Unified CM1->ASA->Off Path Cisco IME ASA->Cisco IME SIPT->Cisco IME ASA->Unified CM2->SIPT->Unified MeetingPlace; Phn3->Unified CM->SIPT->Unified MeetingPlace	Failed	CSCtf02177
UC802IF.IME.220	Cisco IME	Unified B2B Calls to Phone With Mobile Connect Enabled	To verify that Mobile Connect feature works with Unified B2B.	Pn1->ASA->Unified CM1->Cisco IME ASA->SIPT->ASA->Off Path Cisco IME ASA->Unified CM2->GW->PSTN	Passed	
UC802IF.IME.221	Cisco IME	Call Forward All and iDivert to Cisco Unity Over a Unified B2B link	To verify that callers can leave a voicemail in Cisco Unity when the call is over a Unified B2B link.	Phn1->ASA->Unified CM1->Off Path Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM->Phn2->CFA->Phn3->iDivert->Unity	Passed	
UC802IF.IME.222	Cisco IME	Supervised Transfer With Cisco Unity Connection for a Call From Unified B2B Link	To verify that callers are successfully supervise transferred by Cisco Unity Connection when the call is from a Unified B2B link.	Phn1->Unified CM1->Cisco IME ASA->SIPT->Off Path Cisco IME ASA->Unified CM2->SIPT->Connection->Supervise Xfer->Unified CM2->Off Path Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM1->Phn2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.IME.223	Cisco IME	Dual Stack Trunks and Phones With Unified B2B Link	To verify that dual stack endpoints can place calls over the Unified B2B link where they communicate using IPv4 and if fallback, to communicate over the PSTN they use IPv6.	DS Phn1->Unified CM1->Off Path Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM2->DS Phn2; PSTN Fallback->DS Phn1->ASA->Unified CM1->DS SIPT->DS SIP GW->PSTN->DS SIP GW->Unified CM2->DS Phn2	Passed	
UC802IF.IME.224	Cisco IME	GPickUp Using Analog Phones and Dual stack VG224 GateWay With Unified B2B Link	To verify that GPickUp can be performed from an analog endpoint connected to dual stack VG224 for a call over the Unified B2B.	DS Phn1->Unified CM1->Cisco IME ASA->SIPT->Off Path Cisco IME ASA->Unified CM2->DS Phn2->GPickUp->DS VG224->Analog Phone	Passed	
UC802IF.IME.225	Cisco IME	Monitoring Unified B2B Link Using Unified Service Monitor	Verify the Unified B2B link and calls over Unified B2B link, which can be monitored using Unified Service Monitor.		Failed	CSCte76074
UC802IF.IME.226	Cisco IME	Locations Based Call Admission Control With Unified B2B link	Verify that calls through Unified B2B link follows the CAC settings in Locations. Also, confirm that the mid-call PSTN fallback adheres to Locations based CAC setting.	IP Phone1->ASA->Unified CM1->ASA->Off Path Cisco IME ASA->ASA->SIPT (WAN)->Cisco IME ASA->ASA->Unified CM2->IP Phone2	Passed	
UC802IF.IME.227	Cisco IME	Voicemail Deposit and Retrieval From Cisco Unity Express Over Unified B2B Link	Verify that callers can deposit a voicemail in Cisco Unity Express and call Cisco Unity Express over the Cisco IME trunk to retrieve a voicemail.	Phn1->Unified CM1->Cisco IME ASA->SIPT->Off Path Cisco IME ASA->Unified CM2->IP Phone2->CFNA->Unified CM2->ASA->JTAPI->Cisco Unity Express	Passed	
UC802IF.IME.228	Cisco IME	Call Routing With Unified B2B Link When Primary Unified CM is Down	Verify that the calls can be connected successfully even when the primary Unified CM server on which Unified B2B link is down.	IP Phone1->ASA->Unified CM1->ASA->Cisco IME ASA->SIPT (WAN)->ASA->Off Path Cisco IME ASA->ASA->Unified CM2->IP Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.IME.229	Cisco IME	Call Routing When Internet Connectivity on Calling Cluster is Down	Verify that calls can be routed successfully when the internet connectivity is down on the calling cluster.	IP Phone1->ASA->Unified CM1->ASA->Off Path Cisco IME ASA->ASA->SIPT (WAN)->Cisco IME ASA->ASA->Unified CM2->IP Phone2	Passed	
UC802IF.IME.230	Cisco IME	Call Routing Wth Unified B2B Link When the Primary Off Path ASA is Down	Verify that the calls can be connected successfully through the secondary ASA when the primary off path ASA on the called enterprise is down.	IP Phone1->ASA->Unified CM1->ASA->Cisco IME ASA->SIPT (WAN)->ASA->Off Path Cisco IME ASA (Secondary)->ASA->Unified CM2->IP Phone2	Passed	
UC802IF.IME.231	Cisco IME	Link Between Unified B2B Server and Unified CM Server Down	Verify the behavior when link between Unified CM and B2B server is down. Test if any changes to the advertised DN are updated after the link is restored.		Passed	
UC802IF.IME.232	Cisco IME	Unified B2B Link Availability When Primary Cisco IME Server is Unavailable	Verify that Cisco IME functionality is available and the secondary Unified B2B server provides the services when primary server is down.		Passed	
UC802IF.IME.233	Cisco IME	Unified B2B Link Interoperability With EMCC	Verify the Cisco IME functionality with EMCC.	IP Phone1->ASA->Unified CM1->ASA->Cisco IME ASA->SIPT (WAN)->ASA->Off Path Cisco IME ASA->ASA->Unified CM2->EMCC->IP Phone2	Passed	
UC802IF.IME.235	Cisco IME	Inter-Company Video Call With Various Supported Endpoints	Verify and validate inter-company video call with various supported endpoints.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.IME.250	Cisco IME	Scheduled Conferencing With WebEx and Unified MeetingPlace Using Hardware Conference Resource-Out Dial	To verify that when Unified Meeting Place out dials over a Unified B2B link, rich media experience is available.	Unified MeetingPlace->SIPT->Unified CM1->Cisco IME ASA->SIPT WAN->Off Path ASA->Unified CM2->Phn1; Unified MeetingPlace->SIPT->Unified CM1->Phn2; Unified MeetingPlace->SIPT->Unified CM1->Phn2	Failed	CSCtf02177
UC802IF.IME.996	Cisco IME	IME Number Normalization on Call Diversion	Verify for a phone called that has "iDivert" enabled over Cisco IME that the calling party sees the forwarded party's normalized number with "Connected Party" transformation configurations.	Phone1->Unified CM1->Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM2->TRP->Phone2	Passed	
UC802IF.IME.997	Cisco IME	Cisco IME Number Normalization on Call Transfer	Verify for a transferred call over Cisco IME that the calling party sees the newly connected called party's normalized number with "Connected Party" transformation configurations.	Phone1->Unified CM1->Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM2->TRP->Phone2	Passed	
UC802IF.IME.998	Cisco IME	Cisco IME Number Normalization on Call Connect	Verify when a call is placed over Cisco IME that the calling party sees the called party's normalized number using "Connected Party" transformation configurations.	IP Phone1->ASA->Unified CM1->ASA->Cisco IME ASA->SIPT (WAN)->ASA->Off Path Cisco IME ASA->ASA->Unified CM2->IP Phone2	Passed	
UC802IF.IME.999	Cisco IME	Cisco IME Call When There is Congestion on Internet During Call Setup	To verify the behavior of Cisco IME when the packets are dropped or when there is network congestion on the Internet during call setup.	Phone1->Unified CM1->Off Path Cisco IME ASA->MTP->SIPT (WAN)->Cisco IME ASA->Unified CM2->Phone2	Failed	CSCtf14161

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802.DPN.001	DPNSS VG30D Converter	Call Back in RDX Extension	Verify if RDX extension can invoke Call Back when free against IP-Phone in Legacy Unified CM Cluster.	PBX Ph1->PBX->Westell->Unified CM->ICT(QSIG)->Unified CM->SCCP/SIP Ph1	Passed	
UC802.DPN.002	DPNSS VG30D Converter	Call Forward in RDX Extension	Verify if RDX extension can initiate a call which is forwarded by SIP Phone in CDG Central, Video endpoint and VG248 endpoint back to the originating RDX extension.	PBX Ph1->PBX->Westell->Unified CM->CDG Central SIP Ph1->transfer->VG248 Pots Ph1->Unified CM->Westell->PBX->PBX Ph1	Passed	
UC802.DPN.003	DPNSS VG30D Converter	Call Forward with DPNSS Phone	Verify if RDX extension can initiate a call which is forwarded by SCCP Phone, and transfers to Unified CME and IP Phone on CME back to the originating RDX extension.	PBX Ph1->PBX->Westell->Unified CM->CDG Central SCCP Ph1->Transfer-> CME->SCCP Ph1->CFA->Unified CM->Westell-> PBX->PBX Ph1	Passed	
UC802.DPN.004	DPNSS VG30D Converter	iSDX Extension with Cisco Unity	Verify if iSDX extension can call into Unity Connection VoiceMail and retrieve a previously deposited message.	PBX Ph1->PBX->Westell->Unified CM->ICT (QSIG)->Unified CM->Unity	Passed	
UC802.DPN.005	DPNSS VG30D Converter	Call Forward in iSDX Extension	Verify if iSDX extension can initiate a call which is forwarded by SIP Phone in CDG Remote, SIP Phone on CSPS and M1 extension back to the originating iSDX extension.	PBX Ph1->PBX->Westell->Unified CM->CDG Remote SIP Ph1->iDivert->SIP Trunk->CSPS->SIP Ph1->Westell->PBX->PBX Ph2->Unified CM->Westell->PBX->PBX Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.CER.100.2	Cisco Emergency Responder	Track Current Location of IP Phone and E911 Call Routed to Nearest PSAP-Unified IP Phone 7971	verify to ensure that Cisco Emergency Responder can track current location of IP Phones and E911 calls by users getting routed to nearest PSAP.		Passed	
SR60.CER.100.4	Cisco Emergency Responder	Track Current Location of IP Phone and E911 Call Routed to Nearest PSAP-Unified Personal Communicator	Verify to ensure that Cisco Emergency Responder can track current location of IP Phones and E911 calls by users get routed to nearest PSAP.		Passed	
SR60.CER.101	Cisco Emergency Responder	Phone Calls From Unlocated Phone	Verify to ensure if unlocated phones can make 911 calls and their call is routed to default PSAP location configured in Cisco Emergency Responder. A phone is considered unlocated if it is registered with Unified CM, but Emergency Responder has not located it yet. This situation can happen if the phone is behind a switch not defined in Emergency Responder.		Passed	
SR60.CER.102.1	Cisco Emergency Responder	PSAP Callback E911 Caller Through SIP VoIP Protocol	Verify to ensure that PSAP can call back the E911 caller.		Passed	
SR60.CER.103.3	Cisco Emergency Responder	System Reliability When Single Emergency Responder Server Within the Server Group Fails - Fallback	Verify to ensure there is no single point of failure in the Emergency Responder Server Group.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.CER.103.4	Cisco Emergency Responder	System Reliability When Single Emergency Responder Server Within the Server Group Fails - LossOfHeartbeat	Verify to ensure there is no single point of failure in the Emergency Responder Server Group.		Passed	
SR61.CER.100.1	Cisco Emergency Responder	LLDP-MED and CDP Enabled in Both TNP Phone and Access Switch	Verify that Cisco Emergency Responder (CER) can track the location of TNP Phones enabled with both Link Layer Discovery Protocol - Media Endpoint Discovery (LLDP-MED) and Cisco Discovery Protocol (CDP) and 911 calls from these Phones are routed to the local Public Safety Answering Point (PSAP).		Passed	
SR61.CER.102	Cisco Emergency Responder	E911 Call Handling When Both Cisco Emergency Responder and Unified SRST are Capable of Routing E911 Calls to PSAP	Verify that Cisco Emergency Responder can route E911 calls from a branch Phone when both Emergency Responder and Unified SRST routers are configured to route the E911 call to a local Public Safety Answering Point (PSAP).		Passed	
UC802IF.CER.101	Cisco Emergency Responder	JTAPI Over WAN	Verify the validation of Cisco Emergency Responder in CoW deployment such as the Cisco Emergency Responder node registering CTI ports and route points to Unified CM over WAN.	Phone->Emergency Responder->ASA->WAN->ASA->Unified CM->Gateway->PSAP	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CER.102	Cisco Emergency Responder	Cisco Emergency Responder Database Replication	Verify to ensure Cisco Emergency Responder database replication reliability during network instability or Subscriber node going offline for few hours.	Phone->Emergency Responder->ASA->WAN->ASA->Unified CM->Gateway->PSAP	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802L.GTW.003		IP to PSTN to IP Calls	Verify that calls from SFO-ORD IP phone to SFO-ORD IP phone over PSTN are successful.		Passed	
UC802L.GTW.105		IP to PSTN to IP Calls	Verify that calls from AZO IP phone to AZO IP phone over PSTN are successful.		Passed	
UC802L.GTW.303		IP to PSTN to IP Calls	Verify that calls from SFO-ORD IP phone to SFO-ORD IP phone over PSTN are successful.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.IPP.107.3	Unified IP Phone	DTMF Transport to Unified MeetingPlace	Verify to ensure IP Communicator can interwork with Unified Meeting Place, supports RFC2833, inter-cluster call and both wired and wireless LAN: VoiceProto:SIP and LANProto:Wireless.		Passed	
UC712IF.IPC.001	IP Communicator	Encrypted SIP IP Communicator Joining Secure Adhoc Conference Using iLBC	To verify if a user can place an encrypted SIP IP Communicator call and join a secure adhoc conference using the iLBC codec.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EL.PER.001	Reliability	Basic IP to IP Intra Cluster Calls for One Day Load Run	Verify that IP to IP basic calls and supplementary services calls in Cisco Unified Communications Manager Business Edition (Unified CMBE) with 1530 BHCA are successful for one day load run.	SCCP Phone1->Unified CMBE->SCCP Phone2	Passed	
UC802EL.PER.002	Reliability	Basic IP to IP Intra Cluster Calls for Three Day Load Run	Verify that IP to IP basic calls and supplementary services in Unified CMBE with 1530 BHCA successful for three day load run.	SCCP Phone1->Unified CMBE->SCCP Phone2	Passed	
UC802EL.PER.003	Reliability	IP to IP Calls Over PSTN in Unified CMBE for One Day Load Run	Verify that IP to PSTN-to IP basic calls and supplementary services in Unified CMBE with 486 BHCA successful for one day load run.	SCCP Phone1->Unified CMBE->MGCP g/w->PSTN->MGCP g/w->Unified CMBE->SCCP Phone2	Passed	
UC802EL.PER.0038	Reliability	Basic IP to IP Intra cluster Calls	Verify that IP to IP calls in a large size cluster of 1692 BHCA are successful for 24 hours.	SCCP Phone1->Unified CM->SCCP Phone2	Passed	
UC802EL.PER.0039	Reliability	Basic IP to IP Intra cluster Calls	Verify that IP to IP calls in a large size cluster of 1692 BHCA are successful for 72 hours.	SCCP Phone1->Unified CM->SCCP Phone2	Passed	
UC802EL.PER.004	Reliability	IP to IP Calls Over PSTN in Unified CMBE for Three Day Load Run	Verify that IP to PSTN-to IP basic calls and supplementary services in Unified CMBE with 486 BHCA are successful for three day load run.	SCCP Phone1->Unified CMBE->MGCP g/w->PSTN->MGCP g/w->Unified CMBE->SCCP Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EL.PER.0040	Reliability	IP to IP Calls Over PSTN	Verify that IP to IP calls over PSTN in a large size cluster of 8460 BHCA are successful for 24 hours.	SCCP Phone 1->Unified CM->MGCP g/w ->PSTN->MGCP g/w ->Unified CM->SCCP Phone2;SIP Phone1->Unified CM->MGCP g/w->PSTN->MGCP g/w->Unified CM->SIP Phone2	Passed	
UC802EL.PER.0041	Reliability	IP to IP Calls Over PSTN	Verify that IP to IP calls over PSTN in a large size cluster of 8460 BHCA are successful for 72 hours.	SCCP Phone 1->Unified CM->MGCP g/w ->PSTN->MGCP g/w ->Unified CM->SCCP Phone2;SIP Phone1->Unified CM->MGCP g/w->PSTN->MGCP g/w->Unified CM->SIP Phone2	Passed	
UC802EL.PER.0042	Reliability	IP to IP Calls Over Inter Cluster Trunk	Verify that IP to IP Inter Cluster Trunk (ICT) calls with supplementary services (hold, bxfer, cxfer) in a large size cluster 2520 BHCA are successful.	SCCP Phone1->Unified CM->ICT Trunk1 (QSIG ICT)->Unified CM->SCCP Phone2	Passed	
UC802EL.PER.0044	Reliability	IP to IP Calls Over Inter Cluster Trunk with Unified CM 7.x	Verify that IP to IP Inter Cluster Trunk (ICT) calls with supplementary services (hold, bxfer, cxfer) in a large size cluster 2520 BHCA are successful for 24 hours.	SCCP Phone1->Unified CM->ICT Trunk1 (QSIG ICT)->Unified CM(Release 7.x)->SCCP Phone2	Passed	
UC802EL.PER.0045	Reliability	IP to IP Calls Over Inter Cluster Trunk with Unified CM 7.x	Verify that IP to IP Inter Cluster Trunk (ICT) calls with supplementary services (hold, bxfer, cxfer) in a large size cluster 2520 BHCA are successful for 72 hours.	SCCP Phone1->Unified CM->ICT Trunk1 (QSIG ICT)->Unified CM(Release 7.x)->SCCP Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EL.PER.0046	Reliability	IP to IP Calls Between Unified CM Express and Unified CM Over H.323 Gatekeeper Controller Trunk	Verify that IP to IP calls between Unified CM Express to Unified CM of 3960 BHCA are successful for 24 hrs.	SCCP Phone1->Unified CM->GateKeeper 1 ->IPIPGW 1->GateKeeper 1-> GK 2-> CME ->SCCP Phone2	Passed	
UC802EL.PER.0047	Reliability	IP to IP Calls Between Unified CM Express and Unified CM Over H.323 Gatekeeper Controller Trunk	Verify that IP to IP calls between Unified CM Express to Unified CM of 3960 BHCA are successful for 72 hrs.	SCCP Phone1->Unified CM->GateKeeper 1 ->IPIPGW 1->GateKeeper 1-> GK 2-> CME ->SCCP Phone2	Passed	
UC802EL.PER.0048	Reliability	Calls to Unity Connection	Verify that Cisco Unity Connection calls of 3600 BHCA are successful for 24 hrs.	SCCP Phone1->Unified CM->SCCPPhone2->CFNA->UNITY; SCCPPhone2->UNITY->RETRIEVAL	Passed	
UC802EL.PER.0049	Reliability	Calls to Unity Connection	Verify that Cisco Unity Connection calls of 3600 BHCA are successful for 72 hrs.	SCCP Phone1->Unified CM->SCCPPhone2->CFNA->UNITY ;SCCPPhone2->UNITY->RETRIEVAL	Passed	
UC802EL.PER.005	Reliability	IP to IP Calls Over Remote PSTN in Unified CMBE for One Day Load Run	Verify that IP to PSTN-to IP basic calls in Unified CMBE with 324 BHCA are successful for one day load run.	SCCP Phone1->Unified CMBE->MGCP g/w->PSTN->MGCP g/w-> Unified CMBE->SCCP Phone2	Passed	
UC802EL.PER.0050	Reliability	IP to IP RSVP WAN calls	Verify that RSVP WAN calls of 4788 BHCA are successful for 24 hrs.	REM SCCP Phone1->Unified CM->RSVP Agent 1 (Remote) -> RSVP Agent 2(Central site)->Unified CM-> SCCP Phone2;SCCP Phone1->Unified CM->RSVP Agent 2(Central site)-> RSVP Agent 1(Remote)->Unified CM-> REM SCCP Phoneone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EL.PER.0051	Reliability	IP to IP RSVP WAN calls		REM SCCP Phone1->Unified CM->RSVP Agent 1 (Remote) -> RSVP Agent 2(Central site)->Unified CM-> SCCP Phone2;SCCP Phone1->Unified CM->RSVP Agent 2(Central site)-> RSVP Agent 1(Remote)->Unified CM-> REM SCCP Phone2	Passed	
UC802EL.PER.0052	Reliability	Remote IP to Remote IP Over PSTN Calls	Verify that remote IP to remote IP over PSTN calls of 8460 BHCA are successful for 24 hrs.	REM SCCP Phone 1->Unified CM->MGCP/H.323/SIP g/w ->PSTN->MGCP/H.323/SIP g/w->Unified CM->REM SCCP Phone2;REM SIP Phone1->Unified CM->MGCP/H.323/SIP g/w->PSTN->MGCP/H.323/SIP g/w->Unified CM->REM SIP Phone2	Passed	
UC802EL.PER.0053	Reliability	Remote IP to Remote IP Over PSTN Calls	Verify that remote IP to remote IP over PSTN calls of 8460 BHCA are successful for 72 hrs.	REM SCCP Phone 1->Unified CM->MGCP/H.323/SIP g/w ->PSTN->MGCP/H.323/SIP g/w->Unified CM->REM SCCP Phone2;REM SIP Phone1->Unified CM->MGCP/H.323/SIP g/w->PSTN->MGCP/H.323/SIP g/w->Unified CM->REM SIP Phone2	Passed	
UC802EL.PER.006	Reliability	IP to IP Calls Over Remote PSTN in Unified CMBE for Three Day Load Run	Verify that IP to PSTN-to IP basic calls in Unified CMBE with 324 BHCA are successful for three day load run.	SCCP Phone1->Unified CMBE->MGCP g/w->PSTN->MGCP g/w-> Unified CMBE->SCCP Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EL.PER.007	Reliability	RSVP Calls in Unified CMBE for One Day Load Run	Verify that IP to IP calls in Unified CMBE (RSVP configured between Unified CMBE & Remotes) with 360 BHCA are successful for one day load. run	SCCP Phone1->Unified CMBE->SCCP Phone2	Passed	
UC802EL.PER.008	Reliability	RSVP Calls in Unified CMBE for Three Day Load Run	Verify that IP to IP calls in Unified CMBE (RSVP configured between Unified CMBE & Remotes) with 360 BHCA are successful for one day load run.	SCCP Phone1->Unified CMBE->SCCP Phoneone2	Passed	
UC802EL.PER.009	Reliability	Calls to Unity Connection Voicemail for One Day Load Run	Verify that Unity Connection calls in Unified CMBE of 300 BHCA are successful for one day load run.	SCCP Phone1->Unified CM1->SCCPPhone2->CFNA->VOICE MAIL; SCCPPhone2->VOICE MAIL->RETRIEVAL	Passed	
UC802EL.PER.010	Reliability	Calls to Unity Connection Voicemail for Three Day Load Run	Verify that Unity Connection calls in Unified CMBE of 300 BHCA are successful for three day load run.	SCCP Phone1->Unified CM1->SCCPPhone2->CFNA->VOICE MAIL; SCCPPhone2->VOICE MAIL->RETRIEVAL	Passed	
UC802EL.PER.023	Reliability	Basic IP to IP Intra Cluster Calls for One Day Load Run	Verify that IP to IP calls in a medium size cluster of 1740 BHCA are successful.	SCCPPhone1->Unified CM->SCCP Phone2	Passed	
UC802EL.PER.024	Reliability	IP to IP Calls Over PSTN for One Day Load Run	Verify that IP to PSTN-to IP calls in a medium size cluster of 5700 BHCA are successful.	SCCP Phone1->Unified CM->MGCP g/w->Unified CM->SCCP Phone2; SIP Phone1->Unified CM->MGCP g/w->Unified CM->SIP Phone2	Passed	
UC802EL.PER.025	Reliability	IP to IP Calls Over Inter-Cluster Trunk for One Day	Verify that IP to IP inter cluster calls with supplementary services (hold, bxfer, cxfer) in a medium size cluster 720 BHCA are successful.	SCCP Phone1->Unified CM->ICT->Unified CM->SCCP Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EL.PER.026	Reliability	IP to IP Calls Between Unified CME and Unified CM Over Inter-Cluster Trunk for One Day	Verify that IP to IP calls between Unified CME to Unified CM of 1800 BHCA are successful.	SCCP Phone1->Unified CM->GateKeeper1->IPIPGW->GateKeeper1->GateKeeper2->CME->SCCP Phone2	Passed	
UC802EL.PER.027	Reliability	Calls to Unity Connection Voicemail for One Day	Verify that Unity Connection calls in a medium size cluster of 600 BHCA are successful.	Stage1: SCCP Phone1->Unified CM1->SCCPPhone2->CFNA->VOICE MAIL Stage2: SCCPPhone2->VOICE MAIL->RETRIEVAL	Passed	
UC802EL.PER.028	Reliability	IP to IP ICT Call Going Through Tandem Cluster for One Day	Verify that IP to IP inter cluster calls with supplementary services (hold, bxfer, cxfer) involving third cluster of 720 BHCA are successful.	SCCP Phone1->Unified CM1->ICT 1->Unified CM2->ICT 2->Unified CM3->SCCP Phone2	Passed	
UC802EL.PER.029	Reliability	IP to IP ICT Call Checking Interoperability with Previous Version of Unified CM (6.1.3) for One Day	Verify that IP to IP inter cluster calls with supplementary services (hold, bxfer, cxfer) in a medium size cluster 720 BHCA are successful.	SCCP Phone1->Unified CM->ICT 1->Unified CM->SCCP Phone2	Passed	
UC802EL.PER.030	Reliability	Basic IP to IP Intra Cluster Calls for Three Days Load Run	Verify that IP to IP calls in a medium size cluster of 1740 BHCA are successful.	SCCPPhone1->Unified CM->SCCP Phone2	Passed	
UC802EL.PER.031	Reliability	IP to IP Calls Over PSTN for Three Days	Verify that IP to PSTN-to IP calls in a medium size cluster of 5700 BHCA are successful.	SCCP Phone1->Unified CM->MGCP g/w->Unified CM->SCCP Phone2; SIP Phone1->Unified CM->MGCP g/w->Unified CM->SIP Phone2	Passed	
UC802EL.PER.032	Reliability	IP to IP Calls Over Inter-Cluster Trunk for Three Days	Verify that IP to IP inter cluster calls with supplementary services (hold, bxfer, cxfer) in a medium size cluster 720 BHCA are successful.	SCCP Phone1->Unified CM->ICT->Unified CM->SCCP Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EL.PER.033	Reliability	IP to IP calls between Unified CME and Unified CM over Inter-Cluster Trunk for three day	Verify that IP to IP calls between Unified CME to Unified CM of 1800 BHCA are successful.	SCCP Phone1->Unified CM->GateKeeper1->IPIPGW->GateKeeper1->GateKeeper2->CME->SCCP Phone2	Passed	
UC802EL.PER.034	Reliability	Calls to Unity Connection Voicemail for Three Days	Verify that Unity Connection calls in a medium size cluster of 600 BHCA are successful.	Stage1: SCCP Phone1->Unified CM1->SCCPPhone2->CFNA->VOICE MAIL Stage2: SCCPPhone2->VOICE MAIL->RETRIEVAL	Passed	
UC802EL.PER.035	Reliability	IP to IP ICT Call Going Through Tandem Cluster for Three Day	Verify that IP to IP inter cluster calls with supplementary services (hold, bxfer, cxfer) involving third cluster of 720 BHCA are successful.	SCCP Phone1->Unified CM1->ICT 1->Unified CM2->ICT 2->Unified CM3->SCCP Phone2	Passed	
UC802EL.PER.036	Reliability	IP to IP ICT Call Checking Interoperability with Previous Version of Unified CM (6.1.3) for Three Days	Verify that IP to IP inter cluster calls with supplementary services (hold, bxfer, cxfer) in a medium size cluster 720 BHCA are successful.	SCCP Phone1->Unified CM->ICT 1->Unified CM->SCCP Phone2	Passed	
UC802EL.PER.043	Reliability	IP to IP Calls Over Inter Cluster Trunk	Verify that IP to IP Inter Cluster Trunk (ICT) calls with supplementary services (hold, bxfer, cxfer) in a large size cluster 2520 BHCA are successful.	SCCP Phone1->Unified CM->ICT Trunk1 (QSIG ICT)->Unified CM->SCCP Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.UOM.001	Unified Operations Manager	Unified Operations Manager in Unified CM Cluster Monitor UC Components	Verify if Unified Operations Manager installed in Unified Communications Cluster can monitor UC components.	CUOM->SNMP-> UC Components	Passed	
UC802IF.UOM.002	Unified Operations Manager	Unified Service Monitor MOS Score Reporting Capability Within Unified Communications 8.0 Deployment	Validate Unified Service Monitor MOS score reporting capability within Unified Communications deployment model.	Unified CM->Download CDR->CUSM	Passed	
UC802L.NME.001	Network Management	Cisco Unified Operations Manager	Verifies functionality of Cisco Unified Operations Manager during use with a large scale testbed supporting up to 30,000 devices and users.		Passed	
UC802L.NME.002	Network Management	Cisco Unified Service Monitor	Verify comprehensive voice quality measurements through the combination of Cisco 1040 Sensors and Cisco VTQ (Voice Transmission Quality) and alert generation sent to an upstream application (Cisco Unified Operations Manager) when an MOS threshold is violated. Verification is accomplished using testbeds supporting up to 30,000 devices.		Passed	
UC802L.NME.003	Network Management	Cisco Unified Service Statistics Manager	Verify the ability to provide visibility into and usability of key metrics such as call volume, service availability, call quality, resource utilization, and capacity for a large enterprise testbed of up to 30,000 devices.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701EF.QSG.003	Call Forward	Call Forwarding Scenario Involving Calls with MGCP GW and ICT to a H323 GW	Verify if a call from a Unified CM MGCP Gateway phone via an Inter Cluster Trunk to a Unified CM controlled phone can be call forwarded on busy or all to a PSTN phone via H.323 Gateway.	VG224 Ph 1-> MGCP GW-> Unified CM->ICT(QSIG)->Unified CM->SCCP/SIP Ph1->CFB->Unified CM->H.323 BRI/PRI -> PSTN Ph1	Passed	
UC701EF.QSG.004	Call Forward	Call Forward Interaction Involving MGCP Gateway, QSIG PBX , Unified CM ICT, CME Phone and Unity	Verify if a call From a MGCP Gateway to a QSIG PBX phone can be call forwarded on no answer to Cisco Call Manager Express phone and then to Unity.	PSTN ph1-> MGCP (BRI/PRI) ->Unified CM-> QSIG Trunk->QSIG PBX->PBX Ph1->CFNA->QSIG PBX->QSIG Trunk->Unified CM->ICT (QSIG)->Unified CM->IPIPGW->GateKeeper (H323)->CME->CME SCCP Ph1->Unity	Passed	
UC701EF.QSG.006	Call Forward and Call Transfer	H.323 GW Call Transfer by QSIG PBX to SIP Proxy	Verify if a call from a H.323 Gateway to a Unified CM SCCP phone can be transferred to a QSIG PBX phone and call forwarded on no answer to a SIP proxy.	PSTN Ph 1-> Unified CM H323 (BRI/PRI)-> Unified CM->SCCP Ph1->XFER_C->Unified CM->QSIG Trunk->PBX->PBX Ph1->CFNA->PBX->QSIG Trunk->Unified CM->SIP Trunk->CSPS->SIP Ph1	Passed	
UC701EF.QSG.010	Call Transfer	Blind Transfer of MGCP GW Call by SCCP Phone via ICT to a Third party Operator console	Verify if a call from a CMM MGCP phone via ICT to a Unified CM SCCP phone can be blind transferred via ICT to a Third party Operator console.	Pots CMM Ph1-> MGCP GW->Unified CM->ICT (QSIG)->Unified CM->SIP Ph1->XFER_B->ICT (QSIG)->OP Cons1	Passed	
UC701EF.QSG.011	Call Transfer and Call Forward	Interaction of Call Transfer with MGCP GW, ICT, SIP phone, QSIG PBX and IPMA.	Verify if a call From a Unified CM MGCP Gateway phone via ICT to a Unified CM SIP phone can be transferred to a QSIG PBX phone and call forwarded on busy or all to an IPMA Manager phone.	Pots Ph1->Unified CM->ICT (QSIG)->Unified CM->SCCP Ph1->XFER_B->QSIG Trunk->PBX->PBX ph1->CFA->IPMA Manager Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701EF.QSG.015	Call Forward and Call Transfer	Interaction of Call forward and Transfer with Unified CM,SIP phone, QSIG PBX,IP communicator	Verify if a call from a Unified CM SIP phone to a QSIG PBX phone can be transferred to another QSIG PBX phone and call forwarded on no answer to a Unified CM IP communicator.	SIP Ph1-> Unified CM->QSIG Trunk->QSIG PBX->PBX Ph1->XFER_C ->PBX Ph2-> CFNA->QSIG Trunk->Unified CM ->IPC1	Passed	
UC701EF.QSG.017	Call Transfer and Call Forward	CFNA and Transfer Interaction with Unified CME, QSIG PBX ,MGCP GW ,ICT and IPMA manager	Verify if a call from a Unified CME phone to a QSIG PBX phone is transferred to a MGCP Gateway CMM phone and the call is forwarded on no answer via ICT to an IPMA Manager phone.	SCCP Ph1->CME->IPIPGW (H323)->GateKeeper->Unified CM->QSIG Trunk->PBX->PBX Ph1->XFER_C->QSIG Trunk->Unified CM ->MGCP GW->CMM pots Ph1->CFNA->Unified CM->ICT (QSIG)->Unified CM ->IPMA Mgr Ph1	Passed	
UC701EF.QSG.020	Call Forward	Multiple Call forwarding over SIP proxy ,Unified CM and ICT creating a loop	Verify the call From a SIP phone under a SIP proxy to a Unified CM phone is forwarded via ICT to a CCM controlled phone and forwarded again via ICT to a Unified CM phone which forwards back to a Unified CM controlled phone, creating a loop.	SIP Ph1->CSPS->SIP Trunk-> Unified CM->SIP Ph2 ->CFA->ICT (QSIG)-> Unified CM ->SCCP Ph1->CFA->ICT (QSIG)->Unified CM->SCCP Ph2-> CFA->ICT(QSIG)->Unified CM->SIP Ph2	Passed	
UC701EF.QSG.024	CLIP/CLIR	Calling Line ID Restriction from a QSIG PBX phone via ICT to a CCM SCCP phone	Verify the calling and called line and name Restriction on a call from a QSIG trunk to Unified CM and an Inter Cluster Trunk to a Unified CM SCCP phone.	PBX Ph1->PBX ->QSIG Trunk->Unified CM->ICT(QSIG)->Unified CM->SCCP Ph1	Passed	
UC702EF.QSG.040	Callback	Interaction of Callback Request and Call Forwarding with QSIG PBX and Unified CM	Verify the callback request from a PBX phone on forwarded busy call on Unified CM.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF.QSG.003	QSIG per trunk	Inter PBX Call via Unified CM clusters		PBX ph1->QSIG trunk->Unified CM->QSIG ICT ->Unified CM->QSIG Trunk->PBX ph1->CFNA->QSIG Trunk->Unified CM->Unity	Passed	
UC712EF.QSG.005	QSIG per trunk	Interaction with Callback Feature		SCCP Ph1->Unified CM->QSIG ICT->Unified CM->QSIG ICT->Unified CM ->QSIG trunk->Westell->PBX Ph1	Passed	
UC712EF.QSG.016	Call Diversion by Reroute	Call diversion by reroute ,EM and VM		EMA1->Unified CM 1->ICT->Unified CM 2->Ph1->CFNA->ICT->Unified CM 3 >VM	Passed	
UC712EF.QSG.020	Path replacement in Trombone call	Path Replacement in Trombone Call Involving DPNSS PBX and Two Line IP Phone		Ph1->Unified CM 1->ICT->Unified CM 2->Ph1_L1->CFB->Ph1_L2->XFER->QSIG Trunk->Westell->PBX Ph1->XFER->Westell ->QSIG Trunk->Unified CM 1->Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.001	E2E RSVP, SIP Preconditions	E2E RSVP Call Between Phones in two Unified CM Clusters	Verify an E2E RSVP call between phones in two Unified CM clusters.	Originating Phone->RSVP Agent->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2->RSVP Agent->Terminating Phone	Passed	
UC802EF.QOS.002	E2E RSVP, SIP Preconditions	E2E RSVP Call Between a Phone in One Cluster to a Phone in Remote Branch of Another Cluster	Verify an E2E RSVP call between a phone in one cluster to a phone in a remote branch of another cluster.	Originating Phone->Remote RSVPAgent->Remote Branch->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2->RSVPAgent->Terminating Phone	Passed	
UC802EF.QOS.003	E2E RSVP, SIP Preconditions	E2E RSVP Call from a Remote Phone in one Cluster to a Remote Phone in Another Cluster	Verify an E2E RSVP call from a remote phone in one cluster to a remote phone in another cluster.	Originating Phone->Remote RSVPAgent->Remote Branch->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2->Remote RSVPAgent->Terminating Phone	Passed	
UC802EF.QOS.004	E2E RSVP, SIP Preconditions	E2E RSVP Call Between a Phone in a Remote Branch of one Cluster to a Phone in Another Cluster Where Location is Set as Hub-None	Verify an E2E RSVP call between a phone in a remote branch of one cluster to a phone in another cluster where location is set as hub-none.	Originating Phone->Remote RSVPAgent->Remote Branch->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2 (hub_none)->RSVPAgent->Terminating Phone	Passed	
UC802EF.QOS.005	E2E RSVP, SIP Preconditions	E2E RSVP Call from a Phone in Cluster A to a Phone in Cluster B and Location is Set as Hub-None for Both Endpoints	Verify an E2E RSVP call from a phone in cluster A to a phone in cluster B and location is set as hub-none for both the end-points.	Originating Phone->Remote RSVPAgent->Remote Branch->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2 (hub_none)->RSVPAgent->Terminating Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.006	E2E RSVP, SIP Preconditions	E2E RSVP Call from an IP Phone in one Cluster to a Remote FXS Phone in Another Cluster Registered to a Remote Branch (SIP GW) which has Pre-conditions Enabled	Verify an E2E RSVP call from an IP phone in one cluster to a remote FXS phone in another cluster that is registered to a Remote Branch (SIP GW) which has pre-conditions enabled.	IP Phone->RSVPAgent->Unified CM 1->SIP ICT->Unified CM 2->SIP Trunk->Remote SIP GW->FXS Phone	Passed	
UC802EF.QOS.007	E2E RSVP, SIP Preconditions	E2E RSVP Call from a Remote IP Phone in one Cluster to a Remote FXS Phone in Different Cluster Registered to a SIP Gateway with Pre-conditions Enabled	Verify an E2E RSVP call from a remote IP phone one cluster to a remote FXS phone in a different cluster that is registered to a SIP Gateway with pre-conditions enabled.	IP Phone->RSVPAgent->Unified CM 1->SIP ICT->Unified CM 2->SIP Trunk->Remote SIP GW ->FXS Phone	Passed	
UC802EF.QOS.008	E2E RSVP, SIP Preconditions	Unified Enterprise Attendant Console RSVP Interoperability	Verify an E2E RSVP call transferred by Unified Enterprise Attendant Console to a remote FXS phone.	IP Phone->CUEAC->IP Phone->RSVP Agent->CCM->SIP Trunk->Remote SIP GW->FXS Phone	Passed	
UC802EF.QOS.009	E2E RSVP, SIP Preconditions	E2E RSVP Call Between IP Phone to FXS Phone on Same Remote Site	Verify an E2E RSVP call between IP phone to FXS phone on the same remote site.	Remote IP Phone->CCM->SIP Trunk->Remote FXS Phone	Passed	
UC802EF.QOS.010	E2E RSVP, SIP Preconditions	E2E RSVP Audio Call Between UC Integration™ for Microsoft Office Communicator Endpoints in two Clusters	Verify E2E RSVP audio call between UC Integration for Microsoft Office Communicator endpoints in two clusters.	Exploratory	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.011	E2E RSVP, SIP Preconditions	E2E RSVP Call from a Phone in Central Site to Remote FXS Phone Registered to a Remote Branch (SIP GW) with Pre-conditions enabled	Verify an E2E RSVP call from a phone in central site to a remote FXS phone which is registered to a Remote Branch (SIP GW) with pre-conditions enabled.	Phone->RSVPAgent->Unified CM->SIP Trunk->Remote FXS Phone	Passed	
UC802EF.QOS.012	E2E RSVP, SIP Preconditions	E2E RSVP Call from a Remote Phone to Another Remote FXS Phone Registered to a Remote Branch (SIP GW) with Pre-conditions Enabled	Verify an E2E RSVP call from a remote phone to another remote FXS phone registered to a Remote Branch (SIP GW) with pre-conditions enabled.	Remote Phone->RSVP Agent->Unified CM->SIP Trunk->Remote FXS Phone	Passed	
UC802EF.QOS.013	E2E RSVP, SIP Preconditions	E2E RSVP Call from Remote FXS Phone to Another Remote FXS Phone	Verify an E2E RSVP call from a remote FXS phone to another remote FXS phone where both the remote branches are SIP GWs with pre-conditions enabled.	Remote FXS Phone 1->SIP Trunk->Unified CM->SIP Trunk->Remote FXS Phone 2	Passed	
UC802EF.QOS.014	E2E RSVP, SIP Preconditions	E2E RSVP Call from Phone in Unified CM Cluster to Unified CME Phone Aggregated by Unified SIP Proxy	Verify an E2E RSVP call from a phone in Unified CM cluster to a Unified CME phone aggregated by Unified SIP Proxy.	Originating Phone->RSVPAgent->Unified CM Cluster 1->SIP Trunk->Unified SIP Proxy->CME->RSVPAgent->Terminating Phone	Passed	
UC802EF.QOS.015	E2E RSVP, SIP Preconditions	E2E RSVP Call from Remote Phone in Unified CM Cluster to Unified CME Phone Aggregated by Unified SIP Proxy	Verify an E2E RSVP call from a remote phone in Unified CM cluster to a Unified CME phone aggregated by Unified SIP Proxy .	Originating Phone->Remote Branch->Rem RSVPAgent->Unified CM Cluster->SIP Trunk->Unified SIP Proxy->CME->RSVPAgent->Terminating Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.016	E2E RSVP, SIP Preconditions	E2E RSVP Call from FXS Phone Registered to Remote Branch (SIP GW) that has Pre-conditions Enabled to Unified CME Phone Aggregated by Unified SIP Proxy	Verify an E2E RSVP call from a FXS phone registered to a Remote Branch (SIP GW) that has pre-conditions enabled to a CME phone aggregated by Unified SIP Proxy.	FXS Phone->Remote Branch SIP GW (With Preconditions)->Unified CM->SIP Trunk->Unified SIP Proxy->CME->IP Phone	Passed	
UC802EF.QOS.017	E2E RSVP, SIP Preconditions	Supplementary Services in an E2E RSVP Call Between Central Site Phones in two Unified CM Clusters	Verify supplementary services in an E2E RSVP call between central site phones in two Unified CM clusters. Check for Hold/Resume, Transfer and conference.	Originating Phone->RSVP Agent->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2->RSVP Agent->Terminating Phone	Passed	
UC802EF.QOS.018	E2E RSVP, SIP Preconditions	Supplementary Services in an E2E RSVP Call Between a Central Site Phone in one Cluster to Remote Phone in Another Cluster	Verify supplementary services in an E2E RSVP call between a central site phone in one cluster to a remote phone in another cluster. Check for Hold/Resume, Transfer and Conference.	Originating Phone->RSVP Agent->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2->RSVP Agent->Terminating Phone	Passed	
UC802EF.QOS.019	E2E RSVP, SIP Preconditions	Supplementary Services in an E2E RSVP Call Between Central Site Phones in two Unified CM Clusters Whose Location is Set to Hub-none	Verify supplementary services in an E2E RSVP call between central site phones in two Unified CM clusters whose location is set to hub-none. Check for Hold/resume, transfer and conference.	Originating Phone->RSVP Agent->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2->RSVP Agent->Terminating Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.020	E2E RSVP, SIP Preconditions	Supplementary Services in an E2E RSVP Call Between a Central Site Phone in one Unified CM Cluster with Location is Set to Hub-none to a Remote Phone in Another Cluster	Verify supplementary services in an E2E RSVP call between a central site phone in one Unified CM cluster with location set to hub-none to a remote phone in another cluster. Check for Hold/resume, transfer and conference.	Originating Phone->RSVP Agent->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2->RSVP Agent->Terminating Phone	Passed	
UC802EF.QOS.021	E2E RSVP, SIP Preconditions	Supplementary Services for an E2E RSVP Call Between two Remote Phones Across Separate Clusters	Verify supplementary services for an E2E RSVP call between two remote phones across separate clusters.	Remote Phone 1->RSVP Agent->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2->RSVP Agent->Remote Phone	Passed	
UC802EF.QOS.022	E2E RSVP, SIP Preconditions	Supplementary Services for an E2E RSVP Call from a Central Phone to Remote FXS Phone	Verify supplementary services for an E2E RSVP call from a central phone to Remote FXS phone where the gateway is enabled with SIP preconditions.	Originating Phone->RSVP Agent->Unified CM->SIP Trunk->Remote FXS Phone	Passed	
UC802EF.QOS.023	E2E RSVP, SIP Preconditions	Supplementary Services for an E2E RSVP Call from a Remote Phone to a Remote FXS Phone	Verify supplementary services for an E2E RSVP call from a remote phone to a remote FXS phone.	Remote IP Phone->RSVP Agent->Unified CM->SIP Trunk->Remote SIP GW->FXS Phone	Passed	
UC802EF.QOS.024	E2E RSVP, SIP Preconditions	E2E RSVP Call from Unified Personal Communicator in Unified CM Cluster to Unified CME Phone via Unified SIP Proxy	Verify E2E RSVP call from Unified Personal Communicator in Unified CM cluster to CME phone via Unified SIP Proxy.	CUPC->CUP->CCM->SIP trunk->Unified SIP Proxy->CME->IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.025	E2E RSVP, SIP Preconditions	E2E RSVP Call from IP Communicator in Central Site to Unified CME phone via Unified SIP Proxy	E2E RSVP call from IP Communicator in central site to Unified CME phone via Unified SIP Proxy.	CIPC->CCM->SIP Trunk->Unified SIP Proxy->CME->IP Phone	Passed	
UC802EF.QOS.026	E2E RSVP, SIP Preconditions	E2E RSVP Call from Unified Personal Communicator in one Cluster to another Unified Personal Communicator in a Different Cluster	Verify the E2E RSVP call from Unified Personal Communicator in one cluster to another Unified Personal Communicator in a different cluster.	CUPC->CUP->CCM->SIP Trunk->CCM->CUP->CUPC	Passed	
UC802EF.QOS.027	E2E RSVP, SIP Preconditions	E2E RSVP Call from Remote IP Phone to PSTN Phone Over SIP PRI	Verify E2E RSVP call from remote IP phone to PSTN phone over SIP PRI.	Remote IP Phone->CCM 0> SIP Trunk->SIP PRI GW->PSTN Phone	Passed	
UC802EF.QOS.028	E2E RSVP, SIP Preconditions	E2E RSVP Call from Remote IP Phone to PSTN Phone over SIP BRI	Verify the E2E RSVP call from remote IP phone to PSTN phone over SIP BRI.	Remote IP Phone->CCM 0> SIP Trunk->SIP PRI GW->PSTN Phone	Passed	
UC802EF.QOS.029	E2E RSVP, SIP Preconditions	E2E RSVP and Unified Enterprise Attendant Console Interoperability	Verify an E2E RSVP call when a call is transferred by Unified Enterprise Attendant Console to a phone in another cluster.	IP Phone->CUEAC->IP Phone->RSVP Agent->CCM 1->SIP ICT->CCM 2->RSVP Agent->IP Phone	Passed	
UC802EF.QOS.030	E2E RSVP, SIP Preconditions	E2E RSVP and Unified Enterprise Attendant Console interoperability	Verify an E2E RSVP call for a call transferred by Unified Enterprise Attendant Console to a remote phone of another cluster.	IP Phone->CUEAC->IP Phone->RSVP Agent->CCM 1->SIP ICT->CCM 2->RSVP Agent->Remote IP Phone	Passed	
UC802EF.QOS.201.1	RSVP Video	Basic E2E RSVP Video Intercluster Call	Verify that: Video EPs can call from one cluster to other cluster. Both the clusters invoke only one central RSVP agent per cluster.	IP Video Phone 99xx->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->UC Integration™ for Microsoft Office Communicator	Passed w/ Exception	CSCtf44669

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.201.2	RSVP Video	Basic E2E RSVP Video Intercluster Call	Verify that: Video EPs can call from one cluster to other cluster. Both the clusters invoke only one central RSVP agent per cluster.	IP Video Phone 99xx->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->IP Video Phone 99xx	Passed	
UC802EF.QOS.201.3	RSVP Video	Basic E2E RSVP Video Intercluster Call	Verify that: Video EPs can call from one cluster to another cluster. Both the clusters invoke only one central RSVP agent per cluster.	CUVA->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G	Passed	
UC802EF.QOS.202.1	RSVP Video	E2E RSVP Video Call Between Remote and Central Phones Between Two Different Clusters	Verify that: Remote video EP can call from one cluster to other cluster's central video EP. Cluster 1 must invoke its remote RSVP agent. Cluster 2 must invoke its central RSVP agent.	Remote IP Video Phone 99xx->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->CUVA	Passed	
UC802EF.QOS.202.2	RSVP Video	E2E RSVP Video Call Between Remote and Central Phones Between Two Different Clusters	Verify that: Remote video EP can call from one cluster to other cluster's central video EP. Cluster 1 must invoke its remote RSVP agent. Cluster 2 must invoke its central RSVP agent.	Remote IP Video Phone 99xx->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G	Passed	
UC802EF.QOS.202.3	RSVP Video	E2E RSVP Video Call Between Remote and Central Phones Between Two Different Clusters	Verify that: Remote video EP can call from one cluster to other cluster's central video EP. Cluster 1 must invoke its remote RSVP agent. Cluster 2 must invoke its central RSVP agent.	Remote UC Integration™ for Microsoft Office Communicator->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G	Passed w/ Exception	CSCtf44669

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.203.1	RSVP Video	E2E RSVP Video Call Between Two Remote Phones Between Two Different Clusters	Verify that: Remote video EPs can call from one cluster to other cluster. Both the clusters must invoke only one remote RSVP agent per cluster.	Remote CUVA->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->remote RSVP cluster 2->Unified CM cluster 2->remote IP Video Phone 99xx	Passed	
UC802EF.QOS.203.2	RSVP Video	E2E RSVP Video Call Between Two Remote Phones Between Two Different Clusters	Verify that: Remote video EPs can call from one cluster to other cluster. Both the clusters must invoke only one remote RSVP agent per cluster.	Remote 7985G->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->remote RSVP cluster 2->Unified CM cluster 2->remote IP Video Phone 99xx	Passed	
UC802EF.QOS.204.1	RSVP Video	E2E RSVP Video Intercluster Call and Transfer to Remote Site of Called Cluster	Verify that: Video EP can call from one cluster to other cluster. Both the clusters invoke only one central RSVP agent per cluster. Transfer the call to the remote Video EP of the second cluster.	IP Video Phone 99xx->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G->XFER->remote 7985G	Passed	
UC802EF.QOS.204.2	RSVP Video	E2E RSVP Video Intercluster Call and transfer to the Remote Site of Called Cluster	Verify that: Video EP can call from one cluster to other cluster. Both the clusters invoke only one central RSVP agent per cluster. Transfer the call to the remote Video EP of the second cluster.	IP Video Phone 99xx->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G->XFER->remote UC Integration™ for Microsoft Office Communicator	Passed w/ Exception	CSCtf44669

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.204.3	RSVP Video	E2E RSVP Video Intercluster Call and Transfer to Remote Site of Called Cluster	Verify that: Video EP can call from one cluster to other cluster. Both the clusters invoke only one central RSVP agent per cluster. Transfer the call to the remote Video EP of the second cluster	UC Integration™ for Microsoft Office Communicator->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G->XFER->remote CUVA	Passed w/ Exception	CSCtf44669
UC802EF.QOS.205.1	RSVP Video	E2E RSVP Video Intercluster Call and Conference with Remote Site of Calling Cluster	Verify that: Video EP can call from one cluster to other cluster. Both the clusters invoke only one central RSVP agent per cluster. The second phone initiates a conference with a Remote Video EP in first cluster.	IP Video Phone 99xx->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G->CNF->Unified CM cluster 1->remote IP Video Phone 99xx	Passed	
UC802EF.QOS.205.2	RSVP Video	E2E RSVP Video Intercluster Call and Conference with Remote Site of Calling Cluster	Verify that: Video EP can call from one cluster to other cluster. Both the clusters invoke only one central RSVP agent per cluster. The second phone initiates a conference with a Remote Video EP in first cluster.	7985G->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G->CNF->Unified CM cluster 1->remote 7985G	Passed	
UC802EF.QOS.206.1	RSVP Video	E2E RSVP Video Call Between Two Remote Phones Between Two Different Clusters and Transfer to Central Site of Called Cluster	Verify that: Remote video EPs can call from one cluster to other cluster. Both the clusters must invoke only one remote RSVP agent per cluster. Transfer the call to the central Video EP of the called cluster.	Remote IP Video Phone 99xx->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->remote RSVP cluster 2->Unified CM cluster 2->remote 7985G->XFER->7985G	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.206.2	RSVP Video	E2E RSVP Video Call Between Two Remote Phones Between Two Different Clusters and transfer to Central Site of Called Cluster	Verify that: Remote video EPs can call from one cluster to other cluster. Both the clusters must invoke only one remote RSVP agent per cluster. Transfer the call to the central Video EP of the called cluster.	Remote UC Integration™ for Microsoft Office Communicator->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->remote RSVP cluster 2->Unified CM cluster 2->remote CUVA->XFER->7985G	Passed w/ Exception	CSCtf44669
UC802EF.QOS.207.1	RSVP Video	E2E RSVP Video Call Between Two Remote Phones Between Two Different Clusters and Conference with Central Site of Calling Cluster	Verify that: Remote video EPs can call from one cluster to other cluster. Both the clusters must invoke only one remote RSVP agent per cluster. The second phone initiates a conference with a central Video EP in the calling cluster.	Remote IP Video Phone 99xx->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->remote RSVP cluster 2->Unified CM cluster 2->remote UC Integration™ for Microsoft Office Communicator->CNF->Unified CM cluster 1->7985G	Passed w/ Exception	CSCtf44669
UC802EF.QOS.207.2	RSVP Video	E2E RSVP Video Call Between Two Remote Phones Between Two Different Clusters and Conference with Central Site of Calling Cluster	Verify that: Remote video EPs can call from one cluster to other cluster. Both the clusters must invoke only one remote RSVP agent per cluster. The second phone initiates a conference with a central Video EP in the calling cluster.	Remote CUVA->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->remote RSVP cluster 2->Unified CM cluster 2->remote UC Integration™ for Microsoft Office Communicator->CNF->Unified CM cluster 1->7985G	Passed w/ Exception	CSCtf44669
UC802EF.QOS.208.1	RSVP Video	E2E RSVP Video Call Between a Remote Phone of One Cluster to a Central Phone in Another Cluster where Location is Set as Hub-None	Verify that: Remote video EP from one cluster can call to other cluster central video EP. The phone in another cluster where location is set as hub-none.	Remote IP Video Phone 99xx->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->central CUVA	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.208.2	RSVP Video	E2E RSVP Video Call Between a Remote Phone of One Cluster to a Central Phone in Another Cluster where Location is Set as Hub-None	Verify that: Remote video EP from one cluster can call to other cluster central video EP. The phone in another cluster where location is set as hub-none.	Remote UC Integration™ for Microsoft Office Communicator->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->central 7985G	Passed w/ Exception	CSCtf44669
UC802EF.QOS.209	RSVP Video	E2E RSVP Video Call Between Central Phones of Two Different Clusters and Location is set as Hub-None for Both the End-Points	Verify that: Video EP can call from one cluster to other cluster. Both the clusters invoke only one central RSVP agent per cluster. The location is set as Hub-None for both the Eps	IP Video Phone 99xx->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->UC Integration™ for Microsoft Office Communicator	Passed w/ Exception	CSCtf44669

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

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.002	Service Advertisement Framework	Unified CM Advertises Modified PSTN Prefix Information to SRST Sites in SAF Network	Verify that SAF CCD on Unified CM advertises the SRST1 DN ranges and its corresponding modified 'To PSTN' prefix to the SRST sites via SAF enabled SIP trunk in the network, when the SRST1 'To PSTN' prefix is modified. Verify by making a PSTN call from SRST2 to SRST1, when the WAN connectivity from SRST sites to Unified CM cluster in main site is down (All branch routers in SRST mode). Verify by making a PSTN call from SRST1 to SRST2 . Additionally, verify the scenarios by dialing VOIP number instead of direct PSTN Numbers.	Stage 1: Ph 1->SRST 2->PSTN->SRST 1->Ph 2;Stage 2: Ph 1->SRST 1->PSTN->SRST 2->Ph 2	Passed w/ Exception	CSCtd16671
UC802EF.SAF.003	Service Advertisement Framework	SAF Advertises Non-Reachable Information to SRST Sites in SAF Network	Verify if a PSTN call from SRST2 to SRST1 is unsuccessful and does not result in any call loop, assuming that on SRST2 no static route exists about the SRST1, after Unified CM removes the advertised SRST1 DN ranges and its reachability in the network.	Unified CM removes SAF route to SRST1;Unified CM->SAF enabled SIP Trunk(udates the SRST2 via SAF Advertisements that SRST1 no longer reachable via SAF check the SRST1 route removed on the SRST2 router)	Passed	

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

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.005	Service Advertisement Framework	Unified CM Advertises SRST PSTN Prefix Information to SRST and Remote sites in SAF Network	Verify if a PSTN call can be made from SRST1 site to Remote site (in Gateway mode) when WAN connectivity between SRST1 and Unified UCM cluster is down. (Assuming there is no WAN connectivity issue between Remote Site to Unified CM and as well SAF CCD on Unified CM advertised the SRST sites DN pattern and its corresponding PSTN prefix information to the SRST sites in the SAF network).	Ph 1->SRST 1 (using the PSTN route learned via SAF)->PSTN->Remote Site (Gateway mode)->Unified CM1->Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.006	Service Advertisement Framework	Unified CM1 Advertises DN Information to other SAF Clients in the SAF Network	Verify the SAF CCD on Unified CM1 advertises its own site DN pattern and its reachability information to Unified CM2 cluster sites via SAF enabled SIP trunk between Unified CM1 and Unified CM2. Similarly verify the SAF CCD on Unified CM2 advertises its own site DN pattern and its reachability information to Unified CM1 cluster sites via SAF enabled SIP trunk between Unified CM2 and Unified CM1. Verify by making a VOIP call from Unified CM2 to Unified CM1 and second VOIP call from Unified CM1 to Unified CM2. Also check the End-to End RSVP reservation in the above scenarios.	Stage 1:Ph 1->Unified CM 2 -> SIP Precondition enabled SAF Trunk->Unified CM 1 ->Ph 2 Stage 2:Ph 1 ->Unified CM 1->SIP Precondition enabled SAF Trunk->Unified CM 2->Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.007	Service Advertisement Framework	Unified CM1 Advertises DN Information to Other Clients in the SAF Network in Load-Balancing mode	Verify the following: 1. The SAF CCD on Unified CM1 on SAF network advertises its own site DN pattern and its reachability information to Unified CM2 cluster sites via SAF enabled SIP Trunk and H.225 enabled SAF between the Unified CM1 and Unified CM2 . 2. The SAF CCD on Unified CM2 on SAF network advertises its own DN pattern and its reachability information to Unified CM1 cluster via SAF enabled SIP trunk. 3. The Unified CM1 SAF trunks are used in load balancing mode by making first VOIP call from Unified CM2 to Unified CM1, by making second VOIP call from Unified CM2 to Unified CM 1, by making third VOIP call from the Unified CM2 to Unified CM1, and all calls are successful. 4. Also verify the End-to-End RSVP reservation in the above scenarios.	Stage 1:Ph 1->Unified CM2->SIP precondition enabled SAF Trunk ->Unified CM1 ->Ph 2 Stage 2:Ph 1->Unified CM2 ->H.225 Trunk (SAF enabled Trunk)->Unified CM1 ->Phone2 Stage 3:Ph 1->Unified CM2->SIP precondition enabled SAF Trunk->Unified CM1 ->Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.008	Service Advertisement Framework	Interworking of SAF Learnt Route and Static Route	Verify the following: 1. The SAF CCD on Unified CM2 advertises the DN pattern and its reachability information to Unified CM1 via SAF enabled SIP trunk between the Unified CM2 and Unified CM1 and also configure static SIP trunk on Unified CM1 to route the calls from Unified CM1 to Unified CM2 . 2. Verify Inter working of SAF learnt route and Static route on the Unified CM1 by making a VOIP call from Phone 1(its corresponding CSS first preference set to Static partition) in Unified CM1 call to Phone 2 in Unified CM2 and also verify by making a second call from Phone 2 (its corresponding CSS first preference set to SAF learned partition) in Unified CM1 call to Phone 3 in Unified CM2, both the calls are successful.	Stage 1:Phone 1(CSS to static partition) ->Unified CM1-> SIP ICT trunk -> Unified CM 2->Phone 2 Stage 2: Phone 2 (CSS to SAF learned partition)->Unified CM1-> SAF enabled SIP trunk ->Unified CM2->Phone 3	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.009	Service Advertisement Framework	Unified CM Client Service Advertisements in Redundancy Mode	Verify the following: 1. The SAF CCD on the Unified CM1 advertising its own DN pattern and reachability information twice in the network by associating the same SAF enabled SIP trunk created on the Unified CM1 to the Node1 and Node2. Bring the Unified CM1 Node 2 down, modify the DN pattern on the Unified CM1 and advertise the modified DN pattern by the SAF CCD service in the network. 2. Verify if Unified CM node 1 can advertise the service on the network even if node 2 is down by making a VOIP call (modified DN) from Unified CM2 to Unified CM1 (node2).(Assuming the SAF CCD on the Unified CM2 advertising its own DN Pattern and reachability information to Unified CM1).	Stage 1:Ph 1->Unified CM2->SAF -SIP trunk->Unified CM1(node 1)->Ph 2;Stage 2:Ph 2->Unified CM2->SAF-SIP trunk->Unified CM1(node 2)->Ph 2;Stage 3:Ph 1->Unified CM2->SAF-SIP Trunk->Unified CM 1(node 1)->Ph 2;Stage 4:Ph 1->Unified CM2->SAF-SIP Trunk->Unified CM 1 (node 1)->Ph 2	Passed	
UC802EF.SAF.010	Service Advertisement Framework	Unified CM Client Service Advertisements Passing through Non-SAF to SAF Network	Verify the SAF CCD on Unified CM1 advertisement reaches to SAF client Unified CM2 via SAF unaware network by making a VOIP call from the Unified CM1 to Unified CM2.	Stage 1: Ph 1->Unified CM1->SIP Trunk (SAF enabled)->SAF unaware Network->Unified CM 2->Ph 2Stage 2: Ph 1->Unified CM1->SIP Trunk (SAF enabled)->SAF unaware Network->Unified CM 2->Ph 2	Passed	

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

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

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.013	Service Advertisement Framework	Unified CM1 Advertises Information to Other Clients in the SAF Network and Checks RSVP Application ID	Verify the following: The SAF CCD on Unified CM1 advertises its own site DN pattern and its reachability information to Unified CM2 via SAF enabled SIP trunk between Unified CM1 and Unified CM2. 2. Verify the RSVP Application ID functionality on Unified CM1 by making a Video call from Phone 1 in Unified CM1 to Phone 2 in Unified CM2 and then by making an audio call from Phone 2 in Unified CM1 to Phone 3 in Unified CM2. Now, de-escalate the video call to audio call by pressing MUTE on Phone 1 in Unified CM1.	Stage 1:Video Ph 1->Unified CM1->SIP Trunk (RSVP-SAF)->Unified CM2->Video Ph 2 Stage 2:Ph 1->Unified CM1->SIP Trunk (RSVP-SAF)->Unified CM 2->Ph 2 Stage 3:Video ph1(mute ON)->Unified CM1->SIP Trunk(RSVP-SAF)->Unified CM 2->Video ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.014	Service Advertisement Framework	Unified CM1 Advertises DN Information to Other Clients in SAF Network	Verify the following: 1. The SAF CCD on Unified CM1 advertises its own site DN pattern and "To PSTN" prefix information to Unified CM2 via SAF enabled SIP trunk between Unified CM1 and Unified CM2. 2. Verify by making a VOIP call from the Unified CM2 to Unified CM1, and then bring down the WAN connection between the Unified CM1 and Unified CM2 down. Verify by making a PSTN call from central Phone 1 in Unified CM2 to Phone 1 in SRST Site. Also verify by making a PSTN call from the SRST (in Unified CM1 site) to Unified CM2 .	Stage 1:Ph 1->Unified CM 1->SIP Trunk (RSVP enabled SAF)->Unified CM2->Ph 2 Stage 2:Ph 1->Unified CM 2->MGCPGW->PSTN->MGCP GW->Unified CM 1->Ph 1Stage 3: Ph 1->Unified CM1->MGCP Gateway->PSTN->MGCP Gateway->Unified CM2->Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.015	Service Advertisement Framework	Unified CM1 SAF Traffic Load-Balanced Across Call-Manager Nodes	1. On Unified CM2, create two SAF enabled SIP Trunks, associate first trunk to CCD Advertising service and Unified CM1 group consisting of Node1, Node2 and Node3. Similarly associate the second trunk to CCD requesting service and Unified CM1 group consisting of Node4, Node5, Node 6 on Unified CM1 Cluster. By CCD advertising service on Unified CM1 advertise its DN pattern in the Network. 2. By SAF CCD on Unified CM2 advertise the DN Pattern it serves in the Network via SAF enabled SIP Trunk created between Unified CM2 and Unified CM1. 3. Verify that on Unified CM1,the calls from Unified CM2 is load-balanced across Unified CM1 nodes by making four consecutive VOIP calls from Unified CM2 to Unified CM1. Make sure the first call from Unified CM2 is served by Unified CM1 node1, second call is served by Unified CM1 Node 2, third Call by Unified CM1 Node 3, and fourth call by Unified CM1	Stage 1:Ph 1to4->Unified CM2->SAF enabled SIP Trunk->Unified CM1 Node1-3 1->Ph 5-6 Stage 2:Ph 1to4->Unified CM1Node 4-6 4->SAF enabled SIP trunk->Unified CM2->Ph 5-6	Passed	CSCtd71187

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.016	Service Advertisement Framework	SAF - Forwarder Failure Results in Alternate PSTN Routing	Verify the SAF Forwarder failure results in PSTN routing: 1. On Unified CM1 by SAF CCD service advertise its DN pattern and "To PSTN" reachability information via SAF enabled SIP Trunk created between the Unified CM1 and Unified CM2. 2. Assuming SAFF1 is acting for the Service Advertisement for Unified CM1, and SAFF2 is acting for the Service Advertisement for the Unified CM2, Verify by making first VOIP call from Unified CM2 to Unified CM1, then bring down SAFF1 in the network. Ensure that on Unified CM2 it marks the DN pattern it learned as down and starts the age-out timer. After the Age-out Timer of DN pattern is learnt, make second call from Unified CM2 to Unified CM1, and it should be routed via the PSTN network. Then Bring back SAFF1 in Unified CM1 site and modify the DN Pattern on Unified CM1 and advertise it in the SAF network. Verify by making a VOIP call from Unified CM2 to Unified CM1.	Stage 1:Ph 1->Unified CM2->SAF enabled SIP Trunk->Unified CM 1->Ph 2 Stage 2:Ph 1 - Unified CM2->MGCP GW->PSTN->MGCP GW->Unified CM 1->Ph 2 Stage 3:Ph 1->Unified CM2->SAF enabled SIP Trunk->Unified CM 1->Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.017	Service Advertisement Framework	Join Across Lines over SAF Enabled SIP Trunk	Verify if the SAF CCD on Unified CM1 advertises its own site DN pattern and its reachability information to Unified CM2 cluster sites via SAF enabled SIP trunk between Unified CM1 and Unified CM2. Similarly verify if the SAF CCD on Unified CM2 advertises its own site DN pattern and its reachability information to Unified CM1 cluster sites via SAF enabled SIP trunk between Unified CM2 and Unified CM1. Also verify if the Join Across Lines over SAF enabled SIP Trunk works fine.	Stage 1:Phone 1(first line) ->Unified CM2->SAF enabled SIP trunk->Unified CM1 (Phone 2: Stage 2:Phone 1->Unified CM2->SAF enabled SIP Trunk (Unified CM1->Phone 3	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.018	Service Advertisement Framework	Location Based CAC on Calling Cluster Results in Alternate Routing	Verify the following: 1. On the Unified CM1 restrict location based CAC bandwidth setting for one audio call, then by SAF CCD service on Unified CM1 and Unified CM2 advertise DN pattern and its "To PSTN" information via SAF enabled SIP trunks, to inform each Unified CM the other Unified CM's DN pattern and reachability information. 2. Verify the location based CAC on the Calling Cluster (Unified CM1) by making VOIP call from the Unified CM1 to Unified CM2; the calls should be connected successfully. Then make the consult transfer from Unified CM1 to Unified CM2. Locations based CAC bandwidth settings on the Unified CM1, will reject the transfer, and should get routed over the PSTN and get connected successfully to Unified CM2.	Stage 1:Phone 1->Unified CM1->SAF enabled SIP Trunk->Unified CM2->Phone 2 Stage 2: Phone 1->Unified CM1->MGCP->PSTN->MGCP (Unified CM2->Phone 3	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.019	Service Advertisement Framework	Location Based CAC on Calling Cluster Results in Alternate Routing	Verify the following: 1. On Unified CM2, restrict the location based CAC bandwidth setting for one audio call, then by SAF CCD service on Unified CM1 and Unified CM2 advertise DN pattern and its "To PSTN" information via SAF enabled SIP trunks, to publicize each Unified CM about the other's DN pattern and reachability information. 2. Verify the location based CAC on the called cluster (Unified CM2) by making VOIP call from Unified CM1 to Unified CM2; the calls should be connected successfully. Make a consult transfer from Unified CM1 to Unified CM2. Locations based CAC bandwidth settings on the Unified CM2 will reject the transfer, and verify if it gets routed over the PSTN successfully to Unified CM2.	Stage 1:Phone 1->Unified CM1->SAF enabled SIP Trunk->Unified CM2->Phone 2 Stage 2:Phone 1->Unified CM1->MGCP->PSTN->MGCP (Unified CM2->Phone 3	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.020	Service Advertisement Framework	SAF Working on SRST During Connection Monitor Period	Verify SAF working on SRST during connection monitor period, assuming that SAF CCD on the Unified CM advertises the SRST sites DN ranges and its corresponding 'To PSTN' prefix to the SRST sites via SAF enabled SIP trunk in the network. Verify by making a PSTN call from SRST 1 to Central Phone 1 in Unified CM1 when WAN connectivity from the SRST 1 to Unified CM is down. Bring back the WAN connectivity, Verify by making a call in SRST1 site phone to Central Phone 1 in Unified CM1 and before the expiry of connection monitor, the call should be routed via PSTN.	Stage 1:Phone 1->SRST 1->PSTN->Unified CM ->Central Phone 2;Stage 2:Phone 1->SRST 1->PSTN->Unified CM (Central Phone 2	Passed	
UC802EF.SAF.021	Service Advertisement Framework	Age-Out Timer Expiry and PSTN Flush Out Timer Expiry	Verify if the learned DN Patterns are marked down when connectivity to SAFF is lost. DN or patterns are flushed out after age-out timer expiry and PSTN routes are also deleted after PSTN age-out expiry.	Stage 1:Ph 1->Unified CM2->SAF enabled SIP Trunk->Unified CM1->Ph 2;Stage 2: Ph 1->Unified CM2->SAF enabled SIP trunk->Unified CM1->Ph 1 Stage 3: Ph 1->Unified CM1->MGCP->PSTN->MGCP->Unified CM2->Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.022	Service Advertisement Framework	Unified CM Client Advertises its Own DN Pattern to Unified CME Client in SAF Network	Verify if Unified CM on Service Advertisement Framework (SAF) network advertises its own site DN pattern and its reachability information to other Unified CME sites through SIP preconditions enabled SAF Trunk by making a VOIP call from the Unified CME to Unified CM.	Variation 1 :Phone 1->Unified CME->SIP Precondition enabled SAF Trunk->Unified CM->Phone 2;Variation 2: Phone 1->Unified CM->SIP Precondition enabled SAF Trunk->Unified CME->Phone 2	Passed	
UC802EF.SAF.023	Service Advertisement Framework	Unified CM Client Advertises Modified DN Pattern to Unified CME Client in SAF Network	Verify if Unified CM on SAF network advertises its own site modified DN pattern and its reachability information to other Unified CME cluster sites through SIP preconditions enabled SAF Trunk by making a VOIP call from the Unified CME to Unified CM.	Variation 1 :Phone 1->Unified CME->SIP Precondition enabled SAF Trunk->Unified CM->Phone 2;Variation 2: Phone 1->Unified CM->SIP Precondition enabled SAF Trunk->Unified CME->Phone 2	Passed	
UC802EF.SAF.024	Service Advertisement Framework	Interworking of SAF Learnt Route and Static Route	Verify if static route interworks with the route learnt from the Service Advertisement Framework network for the same destined service on a Unified CM by making VOIP call from the Unified CM to Unified CME by varying the learnt partition and CSS on the Unified CM. Also check for the interworking of SAF learnt route and static route on the Unified CME to Unified CM (vice versa call flow) as well.	Variation1 : Phone 1->Unified CM->SIP Precondition enabled SAF Trunk->Unified CME->Phone 2;Variation2: Phone 1->Unified CME->SIP Precondition enabled SAF Trunk->Unified CM->Phone 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.025	Service Advertisement Framework	Unified CME Client Advertises Modified DN Pattern to Unified CM Cluster2 Client in the SAF Network	Verify that the newly added DN Pattern in Unified CME gets advertised to Cluster2 through SAF network directly.	Variation 1: Phone 1->Unified CM Cluster2->SIP precondition enabled SAF Trunk ->Unified CME->Phone 2;Variation 2: Phone 1->Unified CME->SIP precondition enabled SAF Trunk ->Unified CM Cluster2->Phone 2	Passed	
UC802EF.SAF.026	Service Advertisement Framework	Unified CME Client Advertises PBX DN Pattern to other Unified CM Clients in the SAF Network	Verify if Unified CM Cluster can learn the PBX route pattern from the Unified CME Client and route the call when the already associated static CME PBX route pattern in the Unified CM cluster has been deleted.	Variation1:PBXPh1->QSIG PRI->Unified CM->SIP Precond enabled SAFTrunk->CME->QSIG PRI->PBXPh2;Variation2: IPPh1->Unified CM->SIP Precond enabled SAFTrunk->CME->QSIG PRI->PBX Ph2	Passed	
UC802EF.SAF.027	Service Advertisement Framework	Unified CME Client Advertises DN Pattern to Unified CME Client in the SAF Network Through SIP and H.225	Verify the following: 1. A call can be made from a Unified CME registered to Unified SIP Proxy to a Unified CME client that is associated with H.225 Trunk in the Unified CM cluster. 2. When Unified CME1 is registered to Unified SIP Proxy and Unified CME2 to a CME gatekeeper, a call made fails initially since there is no route pattern in Unified CME1 to reach Unified CME2. 3. After Unified CME1 advertises its modified DN range to Unified CME2 Cluster, a call made from Unified CME1 to Unified CME2 should work.	Variation1 :Ph1->Unified CME1->SAF trunk->Unified CME2->Ph2 (advertisement);Variation 2: Ph1->Unified CME2->H225 trunk->Unified CME1->Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.028	Service Advertisement Framework	Unified CM Client Advertises its Remote DN Pattern and "TO PSTN" Prefix Information to Other Unified CME Clients in SAF Network	Verify if Unified CM on Service Advertisement Framework (SAF) network advertises its remote site DN pattern and "TO PSTN" prefix information to other Unified CME client through SIP preconditions enabled SAF Trunk by making a VOIP call from the Unified CME to remote DN.	Variation 1 : Phone 1->Unified CME->SIP Precondition enabled SAF Trunk->Unified CM->Rem Phone 2;Variation 2: Rem Phone 1->Unified CM->SIP Precondition enabled SAF Trunk->Unified CME->Phone 2	Passed	
UC802EF.SAF.029	Service Advertisement Framework	Unified CM Client Advertises its Modified Remote DN Pattern and "TO PSTN" prefix information to Other Unified CME Clients in the SAF Network	Verify if Unified CM on Service Advertisement Framework network advertises its remote site modified DN pattern and "TO PSTN" prefix information to other Unified CME client through SIP preconditions enabled SAF Trunk by making a VOIP call from the Unified CME to Unified CM.	Variation 1 : Ph1->Unified CME->SIP Precondition enabled SAF Trunk->Unified CM1->Rem Ph2;Variation 2 : Rem Ph1->Unified CM1->SIP Precondition enabled SAF Trunk->Unified CME->Ph2	Passed	
UC802EF.SAF.030	Service Advertisement Framework	Unified CME Client Advertises its own DN pattern and "To PSTN" prefix to SRST Client in the SAF Network	Verify if Unified CME on Service Advertisement Framework (SAF) network advertises its own site DN pattern and "To PSTN" prefix and its reachability information to SRST client in the SAF network.	Unified CM is down: Variation 1:Phone 1->Unified CME->PSTN->SRST->Phone 2; Variation 2: Phone1->SRST->PSTN->Unified CME->Phone2	Passed	
UC802EF.SAF.031	Service Advertisement Framework	Unified CME Client Advertises Modified DN Pattern and "To PSTN" prefix to SRST client in the SAF Network	Verify if Unified CME on Service Advertisement Framework network advertises its own site modified DN pattern and "To PSTN" prefix and its reachability information to SRST client in the SAF network.	Unified CM is down: Variation 1:Phone 1->Unified CME->PSTN->SRST->Phone 2;Variation 2: Phone1->SRST->PSTN->Unified CME->Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.032	Service Advertisement Framework	Load balancing of Calls to Remote Unified CM Cluster Advertising Same HostedDN from Unified CME	Verify when a trunk in the advertising cluster is assigned to two Unified CM, Unified CME receives the same pattern twice, once for each Unified CM node. Calls from CME to the advertised DN should alternate between the two Unified CM nodes.	Variation 1 :Phone 1->Unified CME->SIP Precondition enabled SAF Trunk->Unified CM->Phone 2;Variation 2: Phone 1->Unified CM->SIP Precondition enabled SAF Trunk->Unified CME->Phone 2	Passed	
UC802EF.SAF.033	Service Advertisement Framework	Calls to Cisco Unity Express Registered to Unified CM over SAF Trunks	Verify that route points can be advertised and calls can be placed successfully over SAF trunk.	Phone 1->Unified CM->SIP Precondition enabled SAF Trunk->Unified CM->Phone 2->JTAPI->CUE	Passed	
UC802EF.SAF.034	Service Advertisement Framework	Calls to Cisco Unity Express Registered to Unified CME over SAF Trunks	Verify that route points can be advertised and calls can be placed successfully over SAF trunk.	Phone 1->Unified CM->SIP Precondition enabled SAF Trunk->Unified CME->Phone 2->JTAPI->CUE	Passed	
UC802EF.SAF.035	Service Advertisement Framework	End to End RSVP over SIP- SAF Trunks and check for RSVP Application IDs	Verify that the SAF CCD on Unified CM1 advertisement reaches Unified CME SAF enabled trunk between Unified CM1 and Unified CME. Check the End-to-End RSVP reservation in this scenario.	Stage 1:Ph 1->Unified CM1->SIP Trunk (SIP precond enabled SAF Trunk)->Unified CME->Ph 2 Stage 2: Ph 3->Unified CM1->SIP Trunk(SIP precond enabled SAF Trunk)->Unified CME-> Ph 4	Passed	
UC802EF.SAF.036	Service Advertisement Framework	BLF Indication Available for Unified CM1 Phone after Learning Unified CM2's DN Pattern and Busy Information	Verify that SAF CCD on Unified CM1 advertises its own site DN pattern and its reachability information to Unified CM2 via SAF enabled SIP trunk between Unified CM1 and Unified CM2. Also verify that the BLF indication is available for Unified CM1 Phone when Unified CM2's Phone is busy on another call.	Stage1 : Ph 2->Unified CM2->Ph3 ; Stage 2: Ph 1->Unified CM1->SIP Trunk (RSVP- SAF)->Unified CM2->Ph 2 (Ph1 Should get the BLF indication)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.SAF.101	Service Advertisement Framework	Load Balancing of Calls to Remote Unified CM Cluster Advertising Same HostedDN From Unified CME	Verify when a trunk in the advertising cluster is assigned to two Unified CM, Unified CME receives the same pattern twice, one for each Unified CM node. Calls from Unified CME to the advertised DN should alternate between the two Unified CM nodes.	IP Phone->Unified CME->H.225 Trunk->ASA->Unified CM->ASA->IP Phone	Passed	
UC802IF.SAF.102	Service Advertisement Framework	Load Balancing Over SIP and H225 Trunks for Learned Patterns and Alternate Routing Between Unified CM Nodes	Verify when a requesting service has both SIP and H.255 trunk attached to it, call to the learned pattern alternates over both SIP and H.225 trunk.	IP Phone->Unified CM->H.225/SIP Trunk->ASA->Unified CM->ASA->IP Phone	Passed	
UC802IF.SAF.103	Service Advertisement Framework	Co-Existing With Static Routes on Unified CME	Verify when Service Advertisement Framework is advertising a pattern that matches a statically configured dialpeer, then routes from Unified CME to the advertised DN should be prioritized appropriately.	IP Phone->Unified CME->H.225 Trunk->Unified CME->IP Phone	Passed	
UC802IF.SAF.104	Service Advertisement Framework	Join Across Lines Over Service Advertisement Framework Trunks	Verify that join across lines works successfully over Service Advertisement Framework trunks.	IP Phone->ASA->Unified CM->ASA->H.225 Trunk->ASA->Unified CM->ASA->IP Phone	Passed	
UC802IF.SAF.105	Service Advertisement Framework	Video Conference Over H225 Trunks	Verify that patterns can be blocked from being learnt based on the remote call control field.	IP Phone1->Unified CM->H.225 Trunk->Unified CM->IP Phone2; IP Phone1->Unified CM->H.225->Unified CME->IP Phone3; IP Phone1->Conference->IP Phone2 and IP Phone3	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.SAF.106	Service Advertisement Framework	Geophysical Location Based Advertisement and Service Advertisement Framework Redundancy	Verify that patterns are advertised based on the HostedDN configuration in CoW. And also verify that redundancy is available for Service Advertisement Framework.	IP Phone->Unified CME->H.225 Trunk->Unified CM->IP Phone	Passed	
UC802IF.SAF.107	Service Advertisement Framework	PSTN Failover Due to Bandwidth Over Subscription on Calling Cluster	Verify the bandwidth control for calls over Service Advertisement Framework trunks are controlled using Locations based CAC. Verify that calls failover to PSTN when sufficient bandwidth is not available.	IP Phone1->Unified CM->H.225 Trunk->Unified CM->IP Phone2; IP Phone1->Xfer->Unified CM->PSTN->IP Phone3	Passed	
UC802IF.SAF.108	Service Advertisement Framework	PSTN Failover Due to Bandwidth Over Subscription on Called Cluster	Verify the bandwidth control for calls over Service Advertisement Framework trunks are controlled using Locations based CAC. Verify that calls failover to PSTN when sufficient bandwidth is not available.	IP Phone1->Unified CM->H.225 Trunk->Unified CM->IP Phone2; IP Phone3->Unified CM->PSTN->IP Phone4	Passed	
UC802IF.SAF.109	Service Advertisement Framework	Service Advertisement Framework Aware H323 Gateway Providing Unified SRST Services	Verify that Unified SRST router when acting as a gateway can learn advertised DN's and route incoming calls based on the learned pattern. The same router should also route calls to PSTN when if goes into Unified SRST mode.	IP Phone1->Unified CME->PSTN->H.323 GW->Unified CM->IP Phone2	Passed	

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

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.SAF.115	Service Advertisement Framework	Calls From Unified Mobile Communicator to SAF Learned Pattern	Verify that incoming calls from a mobility agent can be routed over SAF trunks. In case of failure the call also failover PSTN.	Unified Mobile Communicator->Unified CM->H.225 Trunk->Unified CM->IP Phone; Unified Mobile Communicator->Unified CM->GW->PSTN->GW->Unified CM->IP Phone	Passed	
UC802IF.SAF.116	Service Advertisement Framework	Calls to Cisco Unity Connection Over SAF Trunk	Verify that voicemail pilot can be advertised and calls can be placed successfully over SAF trunk.	IP Phone->Unified CM->H.225 Trunk->Unified CM->Unity Connection->Xfer->IP Phone	Passed	
UC802IF.SAF.117	Service Advertisement Framework	Scheduled Video Conference With Unified MeetingPlace Over SAF Trunk	Verify that invitees of Unified Meeting Place conference can call into Meeting Place over SAF trunk and video is enabled for those participants with video capable endpoints.	IP Phone->Unified CM->H.225 Trunk->Unified CM->SIPT->Unified MeetingPlace	Passed	
UC802IF.SAF.118	Service Advertisement Framework	Meeting Place Dial Out Over SAF Trunk	Verify that calls from Unified MeetingPlace to users in remote cluster are successful over SAF trunk.	Unified MeetingPlace->SIPT->Unified CM->H.225 (SAF)->Unified CM->IP Phone	Passed	
UC802IF.SAF.119	Service Advertisement Framework	Calls to Unified CCX Over SAF trunks	Verify that route points can be advertised and calls can be placed successfully over SAF trunk.	IP Phone->Unified CM->H.225 Trunk (SAF)->Unified CM->JTAPI->Unified CCX->Xfer->IP Phone	Passed	
UC802IF.SAF.120	Service Advertisement Framework	Calls to Cisco Unity Express Registered to Unified CM Over SAF Trunks	Verify that route points can be advertised and calls can be placed successfully over SAF trunk.	IP Phone->Unified CM->H.225 Trunk (SAF)->Unified CM->JTAPI->Cisco Unity Express	Passed	
UC802IF.SAF.121	Service Advertisement Framework	Advertising DN Ranges Associated With Voice Gateway	Verify that calls are routed to the voice gateway based on SAF advertisements by the gateway.	IP Phone->Unified CM->H.225 Trunk (SAF)->Gateway->PSTN	Passed	
UC802IF.SAF.122	Service Advertisement Framework	Call Over SAF Trunk to a Busy Phone	Verify that when a call over SAF trunk is placed to a busy phone then the call does get rerouted over PSTN.	IP Phone1->Unified CM->H.225 Trunk->Unified CM->IP Phone2;	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.SAF.123	Service Advertisement Framework	Incoming SAF Call Terminated on Remote Destination Phone and Dusted to Desk Phone	To Verify that an incoming call through SAF trunk can be answered on a mobile phone, it's remote destination. Move the call to Desk phone by disconnecting the call at mobile phone and resuming the call at Desk phone. Move call to and fro between desk phone and Mobile phone.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.SAF.124	Service Advertisement Framework	Incoming SAF Call is Call Forwarded to PSTN Through SIP Gateway	Verify if an incoming SAF call can be Call Forwarded All to PSTN destination through a SIP gateway. Verify if call is established successfully.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.SAF.125	Service Advertisement Framework	Incoming SAF Trunk Call Forwarded Through SAF Trunk to a Third Cluster	Verify if cluster 2 phone is unregistered and set for Call Forwarded Unregistered to a number (phone2) in cluster 3 or 1. Verify that incoming call through SAF trunk is established through another SAF trunk to phone 2.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.SAF.126	Service Advertisement Framework	Failure of Active Unified CM When SAF Calls are Active	Verify that when SAF calls are active, fail the active Unified CM by shutting down the port. Verify that the active calls stay and subsequent SAF calls are successful.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.SAF.127	Service Advertisement Framework	Reload Unified SRST Router Which is also the SAF Forwarder	Verify that Unified SRST router re-learns all the patterns after reloading.	Unified CM->SAFF->SAFF->SAFF/SRST	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF.CSF.003	Client Services Framework	Call from PSTN Phone to UC Integration™ for Microsoft Office Communicator	Verify the call from PSTN Phone to UC Integration for Microsoft Office Communicator.	PSTN Ph->MGCP PRI GW->Unified CM->SCCP Ph->Xfer->MOC + CSF	Passed	
UC712EF.CSF.006	Client Services Framework	Call from an Interop Site SCCP Phone to UC Integration™ for Microsoft Office Communicator	Verify the call from an Interop site SCCP phone to UC Integration for Microsoft Office Communicator.	SCCP ph1->Unified CM(Interop)->QSIG ICT->Unified CM->MOC + CSF	Passed	
UC712EF.CSF.007	Client Services Framework	Call from Unified CME Site Phone to UC Integration™ for Microsoft Office Communicator	Verify the call from Unified CME site phone to UC Integration for Microsoft Office Communicator.	SCCP ph->CME->H.323 GK->H.323 IP-IP-GW->Unified CM->MOC + CSF	Passed	
UC712EF.CSF.012	Client Services Framework	UC Integration™ for Microsoft Office Communicator interaction with Unified CCX	Verify the UC Integration for Microsoft Office Communicator interaction with Unified CCX as a caller phone.	MOC + CSF->Unified CM->UCCX->CAD Agent	Passed	
UC712EF.CSF.015	Client Services Framework	UC Integration™ for Microsoft Office Communicator interaction with Unified Attendant Server Attendant Console	Verify the UC Integration for Microsoft Office Communicator interaction with Unified Attendant Server Attendant Console.	SCCP ph->Unified CM->Arc Console->Xfer->Unified CM->MOC + CSF	Passed	
UC712EF.CSF.016	Client Services Framework	Microsoft Office Communicator Behavior When Connection to OCS is Lost	Verify the Microsoft Office Communicator behavior when connection to OCS is lost.		Passed	
UC712EF.CSF.023	Client Services Framework	Interop Site QSIG PBX Phone to UC Integration™ for Microsoft Office Communicator	Verify the Interop site QSIG PBX phone to UC Integration for Microsoft Office Communicator.	PBX ph->QSIG PBX->QSIG Trunk->Unified CM (Interop)->QSIG ICT->Unified CM->CSF + MOC	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713EF.CSF.051	Client Services Framework	UC Integration™ for Microsoft Office Communicator Click-to-Call Feature	Verify the UC Integration for Microsoft Office Communicator click-to-call feature.	UC Integration™ for Microsoft Office Communicator 1->Unified CM->UC Integration™ for Microsoft Office Communicator 2	Passed	
UC713EF.CSF.052	Client Services Framework	UC Integration™ for Microsoft Office Communicator Multi-Party Conference	Verify the UC Integration for Microsoft Office Communicator multi-party conference.	UC Integration™ for Microsoft Office Communicator 1->Unified CM->UC Integration™ for Microsoft Office Communicator 2->CONF->UC Integration™ for Microsoft Office Communicator 3	Passed	
UC713EF.CSF.055	Client Services Framework	UC Integration™ for Microsoft Office Communicator Multi-Party Conference with PBX and PSTN Phones	Verify the UC Integration for Microsoft Office Communicator multi-party conference with PBX and PSTN phones.	UC Integration™ for Microsoft Office Communicator->Unified CM->QSIG Trunk->PBX->PBX Ph1->CONF->MGCP PRI GW->PSTN Ph1	Passed	
UC802EF.CSF.001	CSF, UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator with Unified IP Phones 9971/9951/8961 (deskphone) makes a video call to IP Phone 7985 in another cluster over SIP ICT	Verify if a UC Integration for Microsoft Office Communicator in deskphone mode can make a video call to a IP Phone 7985 over SIP ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->SIP ICT->Unified CM->7985	Passed	
UC802EF.CSF.002	CSF, UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator with IP Phones 9971/9951/8961 (deskphone) makes Video Call to IP Phone 7985 in Another Cluster over QSIG ICT	Verify if a UC Integration for Microsoft Office Communicator in deskphone mode can make a video call to a IP Phone 7985 over QSIG ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->QSIG ICT->Unified CM->7985	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CSF.003	CSF, UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator with IP Phones 9971/9951/8961 (deskphone) makes a Video Call to IP Phones 9971/9951/8961 in Another Cluster over SIP ICT	Verify if a UC Integration for Microsoft Office Communicator in deskphone mode can make a video call to a IP Phones 9971/9951/8961 over SIP ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->SIP ICT->Unified CM->IP Phone 99xx/89xx	Passed	
UC802EF.CSF.004	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator with IP Phones 9971/9951 (deskphone) makes a Video call to IP Phones 9971/9951 in another cluster over QSIG ICT	Verify if a UC Integration for Microsoft Office Communicator in deskphone mode can make a video call to a IP Phones 9971/9951 over QSIG ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->QSIG ICT->Unified CM->IP Phone 99xx/89xx	Passed	
UC802EF.CSF.005	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator Video Calls Between Two Clusters over QSIG ICT	Verify UC Integration for Microsoft Office Communicator (Softphone) video calls between two clusters over QSIG ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->QSIG ICT->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	
UC802EF.CSF.006	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator Video Calls Between Two Clusters over SIP ICT	Verify UC Integration for Microsoft Office Communicator (Softphone) video calls between two clusters over SIP ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->SIP ICT->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CSF.009	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator (softphone) on Remote Site in SRST Mode with Audio	Verify the behavior of UC Integration for Microsoft Office Communicator (softphone) on a remote site in SRST Mode with audio.	IP Phone 99xx/89xx->Unified CM->Remote Branch->UC Integration™ for Microsoft Office Communicator (SRST)	Passed	
UC802EF.CSF.011	CSF, UC Integration™ for Microsoft Office CommunicatorTM	Visual Voicemail with UC Integration™ for Microsoft Office Communicator in Deskphone Mode	Verify Visual voicemail with UC Integration for Microsoft Office Communicator in deskphone mode.	UC Integration™ for Microsoft Office Communicator->Unified CM->Unity Connection	Passed	
UC802EF.CSF.012	CSF, UC Integration™ for Microsoft Office CommunicatorTM	Visual Voicemail with UC Integration™ for Microsoft Office Communicator in Softphone Mode	Verify Visual voicemail with UC Integration for Microsoft Office Communicator in softphone mode.	UC Integration™ for Microsoft Office Communicator->Unified CM->Unity Connection	Passed	
UC802EF.CSF.013	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator Video Call from Remote Phone to Central Site UC Integration™ for Microsoft Office Communicator	Verify UC Integration for Microsoft Office Communicator Video call from remote phone to central site UC Integration for Microsoft Office Communicator.	UC Integration™ for Microsoft Office Communicator->Remote Branch->Unified CM->Central UC Integration™ for Microsoft Office Communicator	Passed	
UC802EF.CSF.014	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator Video call from Remote Phone to Central Site IP Phone 7985	Verify UC Integration for Microsoft Office Communicator Video call from remote phone to central site IP Phone 7985.	UC Integration™ for Microsoft Office Communicator->Remote Branch->Unified CM->Central 7985	Passed	
UC802EF.CSF.015	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator Video Call from Remote Phone to Central Site H.323 Endpoint	Verify UC Integration for Microsoft Office Communicator video call from remote phone to central site H.323 endpoint.	UC Integration™ for Microsoft Office Communicator->Remote Branch->Unified CM->H.323 Video endpoint	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CSF.016	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator (deskphone) Video Call to H.323 Video Endpoint in Same Cluster	Verify UC Integration for Microsoft Office Communicator (deskphone) video call to H.323 video endpoint in same cluster.	UC Integration™ for Microsoft Office Communicator->Unified CM->H.323 Video endpoint	Passed	
UC802EF.CSF.017	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator (deskphone) Video Call to H.323 Video Endpoint in Different Cluster Over QSIG ICT	Verify UC Integration for Microsoft Office Communicator (deskphone) video call to H.323 video endpoint in different cluster over QSIG ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->QSIG ICT->Unified CM->H.323 Video endpoint	Passed	
UC802EF.CSF.018	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator (deskphone) Video Call to H.323 Video Endpoint in Different Cluster over SIP ICT	Verify UC Integration for Microsoft Office Communicator (deskphone) video call to H.323 video endpoint in different cluster over SIP ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->SIP ICT->Unified CM->H.323 Video endpoint	Passed	
UC802EF.CSF.019	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator (softphone) Video Call to H.323 Video Endpoint in Same Cluster	Verify UC Integration for Microsoft Office Communicator (softphone) video call to H.323 video endpoint in same cluster.	UC Integration™ for Microsoft Office Communicator (Softphone)->Unified CM->H.323 Video endpoint	Passed	
UC802EF.CSF.020	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator (softphone) Video Call to H.323 MCU in Different Cluster over QSIG ICT	Verify UC Integration for Microsoft Office Communicator (softphone) video call to a H.323 MCU in a different cluster over QSIG ICT.	UC Integration™ for Microsoft Office Communicator (Softphone)->Unified CM->QSIG ICT->Unified CM->H.323-MCU	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CSF.021	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator (softphone) Video Call to H.323 Video Endpoint in Different Cluster over SIP ICT	Verify UC Integration for Microsoft Office Communicator (softphone) video call to H.323 video endpoint in different cluster over SIP ICT.	UC Integration™ for Microsoft Office Communicator (Softphone)->Unified CM->SIP ICT->Unified CM->H.323 Video endpoint	Passed	
UC802EF.CSF.022	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator Video Conference Call Across Different Clusters over QSIG ICT	Verify UC Integration for Microsoft Office Communicator Video conference call across different clusters over QSIG ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->QSIG ICT->Unified CM->UC Integration™ for Microsoft Office Communicator->CONF->UC Integration™ for Microsoft Office Communicator	Passed	
UC802EF.CSF.023	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator Video Conference Call Across Different Clusters over SIP ICT	Verify UC Integration for Microsoft Office Communicator Video Conference Call Across Different Clusters over SIP ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->SIP ICT->Unified CM->UC Integration™ for Microsoft Office Communicator->CONF->UC Integration™ for Microsoft Office Communicator	Passed	
UC802EF.CSF.024	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator Video Call to Unified Video Advantage and IP Communicator in Different Cluster over QSIG ICT	Verify UC Integration for Microsoft Office Communicator video call to Unified Video Advantage and IP Communicator in different cluster over QSIG ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->QSIG ICT->Unified CM->CIPC + CUVA	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CSF.025	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator Video Call to Unified Video Advantage and IP Communicator in Different Cluster over SIP ICT	Verify UC Integration for Microsoft Office Communicator video call to Unified Video Advantage and IP Communicator in different cluster over SIP ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->SIP ICT->Unified CM->CIPC + CUVA	Passed	
UC802IF.CSF.001	Cisco UC Integration™ for Microsoft Office Communicator	SD Video call between UC Integration™ for Microsoft Office Communicator and H323 and SIP 3rd party video end points	To verify standard definition video call can be established between UC Integration™ for Microsoft Office Communicator and H323 and SIP 3rd party end points.	UC Integration™ for Microsoft Office Communicator->Firewall ASA->Unified CM->Firewall ASA->Sony/Tandberg/ Video end points	Failed	CSCtf33615
UC802IF.CSF.002	Cisco UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator in Soft Phone Mode Setting Up Adhoc Video Conference With Cisco Endpoints	To verify if adhoc video conference can be established using Unified MeetingPlace video conference bridge resource from UC Integration for MOC inviting Unified IP Phone models (9971, 9951, and 8961), Unified IP Phone 7985, Unified Personal Communicator, and SIP and H.323 3rd party video end points.	UC Integration™ for Microsoft Office Communicator->Firewall ASA->Unified CM->Firewall ASA->Unified MeetingPlace Video conference resouce+Sony+Tandberg Video end points+RT+7985	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CSF.003	Cisco UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator in Desk Phone Mode Setting Up Adhoc HD Video Conference	To verify if adhoc HD video conference can be established using Unified MeetingPlace Video conference bridge resource from Cisco UC Integration for Microsoft Office Communicator in desk phone mode associated to a multi-line Unified IP Phone (9971, 9951, and 8961) model inviting SIP intercluster Unified IP Phone (9971, 9951, and 8961) model and UC Integration for Microsoft Office Communicator clients.	UC Integration™ for Microsoft Office Communicator->Firewall ASA->Unified CM->Firewall ASA->Unified MeetingPlace Video conference resouce+Sony+Tandberg Video end points+RT+7985	Failed	CSCtf65671
UC802IF.CSF.004	Cisco UC Integration™ for Microsoft Office Communicator	Cisco UC Integration™ for Microsoft Office Communicator in Desk Phone Mode	To verify Cisco UC Integration for Microsoft Office Communicator in remote Clustering over WAN (CoW) site can receive an intercluster SIP HD video call.	UC Integration™ for Microsoft Office Communicator1->Firewall ASA->Unified CM1->firewall ASA---<SIP>-(Firewall ASA->Unified CM2--(UC Integration™ for Microsoft Office Communicator2	Passed	
UC802IF.CSF.005	Cisco UC Integration™ for Microsoft Office Communicator	Cisco UC Integration™ for Microsoft Office Communicator in Unified SRST Mode	To verify if Cisco UC Integration for Microsoft Office Communicator in a remote site registered to a Unified SRST Gateway can retrieve voicemails from the Mainsite.	UC Integration™ for Microsoft Office Communicator->SRST	Passed w/ Exception	CSCte64265
UC802IF.CSF.009	Client Services Framework	Client Services Framework Joining Unified MeetingPlace Scheduled Video Conference	To verify Client Services Framework client can join a scheduled Unified MeetingPlace video conference.	CSF->Firewall ASA->Unified CM->FIREWALL ASA->Cisco IME	Passed w/ Exception	CSCte70278

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CSF.010	Cisco UC Integration™ for Microsoft Office Communicator	Cisco UC Integration™ for Microsoft Office Communicator in Desk Phone Mode	To verify Cisco UC Integration for Microsoft Office Communicator can associate to an EM enabled Unified IP phone .	UC Integration™ for Microsoft Office Communicator->Firewall ASA->Unified CM	Passed	
UC802IF.CSF.012	Cisco UC Integration™ for Microsoft Office Communicator	Incoming Cisco IME Video Call Escalation	To verify Cisco UC Integration for Microsoft Office Communicator or CSF can carry out mid call escalation of incoming Cisco IME video calls to HD video and establish a HD Video conference inviting parties across Cisco IME trunk.	UC Integration™ for Microsoft Office Communicator1->Firewall ASA->Unified CM1->Firewall ASA->Cisco IME->Unified CM2-Firewall ASA->UC Integration™ for Microsoft Office Communicator2	Failed	CSCtf65671
UC802IF.CSF.013	Cisco UC Integration™ for Microsoft Office Communicator	CSF Client Receiving And Transferring DVO Calls Through Cisco IME Trunk	To verify CSF client can receive DVO call from Unified Mobile Communicator client and transfer the call to another enterprise destination through Cisco IME trunk.	Mobile Cllient->Unified CM of Enterprise 1->CSF Client--XFER operation--Cisco IME Trunk->Unified CM of Enterprise 2-> CSF client2	Passed	
UC802IF.CSF.014	Cisco UC Integration™ for Microsoft Office Communicator	HD/SD Video Call Between CSF and SCCP Video Endpoint	To verify CSF client can successfully establish a SD/HD video call to a SCCP video end point.	CSF client->Unified CM of Enterprise1->Cisco IME Trunk->Unified CM of Enterprise2-->SCCP Video endpoint	Passed	
UC802IF.CSF.015	Cisco UC Integration™ for Microsoft Office Communicator	Cisco UC Integration™ for Microsoft Office Communicator/CSF Call Processing Nodes Unavailable	To verify that Cisco UC Integration for Microsoft Office Communicator or CSF can perform normally when the active Unified CM server become unavailable, or the AD server become unavailable, or the OCS server become unavailable.	UC Integration™ for Microsoft Office Communicator->Firewall ASA->Unified CM	Failed	CSCtf57532

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CSF.016	Cisco UC Integration™ for Microsoft Office Communicator	Cisco UC Integration™ for Microsoft Office Communicator/CSF in Desk Phone Receiving Cisco IME Call	To verify that Cisco UC Integration for Microsoft Office Communicator in desk phone mode to a Unified IP Phone 9971 can successfully terminate a HD video call from a video end point in a different enterprise through Cisco IME trunk.	UC Integration™ for Microsoft Office Communicator->SRST(Firewall ASA->Unified CM	Passed	
UC802IF.CSF.020	Unified Contact Center Express	Connection Monitor Timer Interaction With Cisco UC Integration™ for Microsoft Office Communicator Registered in Unified SRST	To verify that Cisco UC Integration for Microsoft Office Communicator in Unified SRST with flapping WAN link remains registered in Unified SRST mode.	UC Integration™ for Microsoft Office Communicator--→SRST----<WAN Link>---Unified CM	Passed	
UCS712IF.CSF.010	UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator Client in Phone Associated Mode	Verify that the UC Integration for Microsoft Office Communicator Client is in phone associated mode to a secure IP Phone 7916 and can set up a secure ad-hoc conference.	MOC+CSF1 --->CUCM-->conference <---- MOC+CSF2	Passed	
UCS712IF.CSF.015	UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator Client on Desk Phone	Verify if the UC Integration for Microsoft Office Communicator client is associated to an Extension Mobility Unified IP 9900 or 8900 series desk phone. This client is part of a group pick up and group pick up an incoming DVO call set up by Unified Mobile Communicator.	MOC+CSF1->CUCM1(Extension Mobility+Group Pickup)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.DP.100	Unified Analysis Manager	Unified Analysis Manager With Unified CM Subscribers Installed in Unified Computing System Environment	To verify if Unified Analysis Manager can be used with components running on Unified Computing System platform.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC700IF.CUB.010	Video Fallback to Audio via IP-to-IP GateWay	Video Call Falls Back to Audio When Endpoint is Not Video Capable	Verify de-escalation of video call to audio call when endpoint is not video capable.		Passed	
UC701IF.CUB.007	Unified Border Element	Secure Transcoding and Conference Between G722/iLBC and G711	Verify that calls between Unified CM and Unified CME phone via IP-to-IP Gateway with secure transcoding and secure conference between Unified CM and Unified CME participants.	Unified CM Phone->SIP Trunk->IP-to-IP Gateway->Unified CME->Unified IP Phone->Conference->IP-to-IP Gateway->Unified CM->Unified IP Phone	Passed	
UC713IF.CUB.001	Unified Border Element	iSAC Transcoding on Unified Border Element for SIP / H323 Trunk	Verifies Transcoding support between iSAC ((g711. g729 , g722 , iLBC.		Passed	
UC713IF.CUB.003	Unified Border Element	sRTP to RTP Interworking	Verify Interworking between sRTP and RTP.	Unified Communications Manager->sRTP->Unified Border Element->RTP->Unified CME	Passed	
UC802IF.CUB.807	Unified Border Element	SIP to H323 Interoperability Through Unified Border Element on ASR	To verify if a user can call a Unified CME Phone from Headquarter Phone through Unified Border element running on ASR platform and can also conference another Unified CM Phone through SIP Trunk.	IP Phone ---Unified CM (1)--- SIP T -Unified SIP Proxy --- SIPT- Unified Border Element --- H323 ---- IP Phone ---conference -SIP T--- IP Phone 3 --- Unified CM (1)	Passed	
UC802IF.CUBE.808	SIP Supplementary Services	SIP to SIP Audio Supplementary Services Through Unified Border Element on ASR	Verify if an audio call can be transferred through Unified Border Element to another Unified CM video phone and if the call can be established.	IP Phone --- SIP T -Unified SIP Proxy --- SIPT- Unified Border Element -SIP T---- IP Phone--- Unified CM 2 -Xfer --- IP Phone 3 --- Unified CM 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CUBE.809	Unified SRST	Failover Between Cisco ISR Unified Border Element and ASR 1006 Unified Border Element	Verifies the fail over between ISR Unified Border Element and ASR 1006 . Unified CM SRV record is configured with ISR and ASR Unified Border Element IP addresses.	Unified CM--SIP Trunk --- (ISR/ASR Unified Border Element) --- SIP Trunk --- CME	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
GB42.CRS.015.80	Regular Agent	Call to Unified CCX from MGCP Gateway to CAD Desktop Agent IP Phone 7970/71; Agent Action=Touch Tones - Transfer/Conference	Verify communication between various devices of a Unified CCX system across a secured infrastructure in an effort to identify any incompatibilities between the configuration of the security infrastructure and the normal message flow of the Unified CCX system. Various agent and supervisor actions in different locations and with different equipments are used.		Passed	
GB42.CRS.020.13	Regular Agent	Originate call to Unified CCX from GK-ICT Trunk to IPPA agent phone 7970/71, agent action Transfer - Blind	The objective of these tests is to generate communication between various devices of a Unified CCX system across a secured infrastructure in an effort to identify any incompatibilities between the configuration of the security infrastructure and the normal message flow of the Unified CCX system. Various agent and supervisor actions in different locations and with different equipment are used.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
GB42.CRS.030.1	Supervisor Agent	Supervisor for Agent, Barge-In	Verify communication between various devices of Unified CCX system across a secured infrastructure in an effort to identify any incompatibilities between the configuration of the security infrastructure and the normal message flow of the Unified CCX system. Various agent and supervisor actions in different locations and with different equipment are used.		Passed	
SR60.CRS.097	Unified Contact Center Express	Utilizing Blended CSQ in An Outbound Campaign	Verify to ensure that blended CSQ does not use more agents than allowed for outbound campaign.		Passed	
SR60.CRS.098	Supervisor Agent	Supervisor for Agent, Silent Monitoring	Verify Supervisor can perform silent monitoring of agents enabled for VoIP Monitoring as well as Desktop Monitoring.		Passed	
SR60.CRS.110	Unified Contact Center Express	High Availability	To verify that for the outbound Preview dialer capability: 1. If one node goes down, the Outbound Campaigns are stopped till either the failed node is restored or removed from the Unified CCX cluster. 2. Data integrity for Dialing List (customer records) maintained for all customer records except for records that are being presented or have been accepted by agents at the time of failover.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC700IF.CCX.201	Unified CCX	Unified IP Phone 7921G Agent: IP Phone Agent	Verify to ensure IP Phone 7921G can be used as IP Phone Agents, receive inbound call and transfer the call to another 7971 CAD Agent.		Passed	
UC700IF.CCX.202	Unified CCX	Unified IP Phone 7921G Agent: CAD Agent	Verify to ensure that IP Phone 7921G can be used as CAD Agents, receive inbound call and conferences the call to Supervisor.		Passed	
UC700IF.CCX.400	Unified Contact Center Express	Unified CCX Backup And Restore	To verify if Unified CCX backup and restore is successful.		Passed	
UC701IF.CRS.093	Agents in SRST Location	Unified CCX Reroutes Call to Next Available Agent in Different Location	Verify to ensure Unified CCX reroutes the call to next available agent in different location if the call to current agent is rejected due to CAC.		Passed	
UC701IF.CRS.094	Agents in SRST Location	Unified SRST Loses WAN Connectivity While Remote Agent Handles Inbound Call	Verify to ensure that Agent becomes Not Ready when Unified SRST location loses WAN link and then becomes Ready again when WAN connectivity is restored.		Passed	
UC701IF.CRS.100.3	Unified Contact Center Express	Monitor Presence Status And Establish Chat Session With SME	Verify to ensure Cisco Agent Desktop can monitor presence status and establish chat session with non-agent SME using Unified Personal Communicator/IPPM.		Passed	
UC701IF.CRS.101.2	Unified CCX	Click to Dial, Transfer, Conference	Verify to ensure CAD Desktop can click to dial, transfer or conference the Subject Matter Expert (SME).		Failed	CSCtc91625

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701IF.CRS.103.2	Unified CCX	Personal Contacts	Verify to ensure CAD Desktop can show Unified Personal Communicator Agent's buddies as Personal Contacts in CAD Chat Selection Window.		Failed	CSCtc91625
UC701IF.CRS.105.2	Unified CCX	SME Initiating Chat Session with CAD	Verify to ensure SME can initiate Chat Session with CAD.		Passed	
UC701IF.CRS.106.1	Unified CCX	CAD Attempting to Register to Incorrect Unified Presence Server	Verify to ensure CAD is redirected to register to correct Unified Presence Server in the Cluster if it attempts to register to a Unified Presence Server where the CAD user is not defined.		Failed	CSCtc91625
UC701IF.CRS.107	Unified CCX	Real Time Update of Presence Status of CAD and SME in Chat Selection Window	Verify to ensure Real Time Update of presence status of CAD and SME in the Chat Selection Window.		Failed	CSCtc91625
UC701IF.CRS.109.4	Unified Contact Center Express	Distribute Support New Entry Level Email offering	Verify to ensure that incoming emails to a CSQ are distributed to idle agents and voice has higher priority than email.	Email->Email CSQ->CAD	Passed	
UC701IF.CRS.120	Unified Contact Center Express	Voice Mail Left in Cisco Unity is Sent as Attachment to the Inbound Email in Unified CCX	To verify if the voice mail is sent as attachment to email-CSQ and if the email is delivered to the agent, when an end-user leaves a voice mail to the support number associated with email CSQ.	Customer->Unified CCX Pilot number->Voice Mail->Unified CCX Email CSQ->Email ready Agent	Passed	
UC701IF.CRS.121	Unified Contact Center Express	Customer Replies Voice Mail to Unified CCX Inbound Email Mailbox	Verify to ensure that Unified CCX inbound email can handle replies to voice mails.	Customer->Voice Mail->Reply the voice mail->Unified CCX Email CSQ->Email ready Agent	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701IF.CRS.155	Unified CCX	CAD Generates Both Audio and Visual Notification When Subject Matter Expert (SME) Sends IM to CAD	Verify to ensure CAD desktop plays audio tone and chat button in windows taskbar flashes to indicate arrival of new IM.		Passed	
UC701IF.CRS.300	Unified Contact Center Express	Importing Agent User from Unified CM to Cisco Unity Connection Using AXL	Verify to ensure that Cisco Unity Connection can import Agent users from Unified CM using AXL.		Passed	
UC701IF.CRS.301	Unified Contact Center Express	Call to Agent's Voice Mail is Re-Queued to Unified CCX-CSQ by Call Transfer Option in Cisco Unity	Verify to ensure Cisco Unity can provide option in agent's voice mail to re-queue the call to CSQ so that it is serviced by another available agent.		Passed	
UC701IF.CRS.302	Unified Contact Center Express	Call to Agent's Voice Mail is Re-Queued to Unified CCX-CSQ by Call Transfer Option in Cisco Unity Connection	Verify to ensure that Cisco Unity Connection provides an option in agent's voice mail to re-queue the call to CSQ so that it is serviced by another available agent.		Passed	
UC701IF.CRS.303	Unified Contact Center Express	Cisco Unity Auto Attendant Call Transferred to Agent's Voice Mail for Re-Queue to Unified CCX	Verify to ensure that Cisco Unity auto attendant interaction with Unified CCX agent is possible and calls get transferred to CTI route points.		Passed	
UC802EF.CRS.001	Unified CCX Inbound Call	Unified Contact Center Express with High-Availability	Verify Unified CCX with high-availability: Inbound call from a PSTN phone to a Cisco Agent Desktop agent with SCCP phone in the central site when one of the Unified CCX server is down.	PSTN Ph1->MGCP GW->Unified CM->UCCX->SCCP Ph1 (CAD)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CRS.002	Unified CCX Outbound Call	Outbound Call from Cisco Agent Desktop SCCP Remote Phone over H.323 FXO	Verify if outbound call from a Cisco Agent Desktop SCCP phone in a remote site to a PSTN phone over H.323 FXO remote gateway is successful.	UCCX->Unified CM->SCCP Ph1 (CAD)->H.323 FXO->PSTN Ph1	Passed	
UC802EF.CRS.003	Unified CCX	Outbound Call from Cisco Agent Desktop SCCP Remote Phone over MGCP FXO	Verify if outbound call from a Cisco Agent Desktop SCCP phone in a remote site to a PSTN phone over MGCP FXO remote gateway is successful.	UCCX->Unified CM->SCCP Ph1 (CAD)->MGCP FXO->PSTN Ph1	Passed	
UC802EF.CRS.004	Unified CCX Outbound call	Outbound Call from Cisco Agent Desktop IP Phone 6941	Verify if an outbound call from a Cisco Agent Desktop IP Phone 6941 to a PSTN phone is successful.	UCCX->Unified CM->6941 Phone(CAD)->MGCP PRI->Rem PSTN Ph	Passed	
UC802EF.CRS.005	Unified CCX Outbound Call	Outbound Call from a Cisco Agent Desktop IP Phone 9971 to a PSTN phone	Verify if an outbound call from a Cisco Agent Desktop IP Phone 9971 to a PSTN phone is successful.	UCCX->Unified CM->9971 Phone(CAD)->MGCP BRI->Rem PSTN Ph	Passed	
UC802EF.CRS.006	Unified CCX Outbound Call	Outbound Call from a Cisco Agent Desktop SCCP Phone	Verify if an outbound call from a Cisco Agent Desktop SCCP phone in a remote site to a PSTN phone over MGCP BRI remote gateway is successful.	UCCX->Unified CM->SCCP Ph1 (CAD)->MGCP BRI->PSTN Ph1	Passed	
UC802EF.CRS.007	Unified CCX Outbound Call	Outbound Call from a Cisco Agent Desktop SIP phone Gateway	Verify if an outbound call from a Cisco Agent Desktop SIP phone in the central site to a remote SCCP phone is consult transferred to a PSTN phone over MGCP PRI remote gateway.	UCCX->Unified CM->SIP Ph1 (CAD)->Unified CM->SCCP Ph1->XFR_C->MGCP PRI->PSTN Ph1	Passed	
UC802EF.CRS.008	Unified CCX Outbound call	Outbound Call from a Cisco Agent Desktop SCCP Phone	Verify if an outbound call from a Cisco Agent Desktop SCCP phone in a remote site to a PSTN phone over SIP BRI remote gateway is successful.	UCCX->Unified CM->SCCP Ph1 (CAD)->SIP BRI->PSTN Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CRS.009	Unified CCX Outbound Call	Outbound Call from a Cisco Agent Desktop SCCP Phone	Verify if an outbound call from a Cisco Agent Desktop SCCP phone in a remote site to a PSTN phone over SIP PRI remote gateway is successful.	UCCX->Unified CM->SCCP Ph1 (CAD)->SIP PRI->PSTN Ph1	Passed	
UC802EF.CRS.010	Unified CCX Outbound call	Unified CCX with Unity connection as Front-End	Verify if Unified CCX with Unity connection can function as front-end.	PSTN Ph1->MGCP PRI->Unified CM->Unity connection->XFR->UCCX->9971 Phone (CAD)	Passed	
UC802EF.CRS.011	Unified CCX and Cisco Unity Interoperability	Unified CCX with Cisco Unity as Back-End	Verify if Unified CCX with Cisco Unity can function as back-end.	PSTN Ph1->MGCP PRI->Unified CM->UCCX->SCCP Ph1->CFNA->Unity	Passed	
UC802EF.CRS.012	Unified CCX and Cisco Unity Interoperability	QSIG PBX Phone Call to Unified CCX Agent via ICT	Verify if the call from a QSIG PBX phone to Unified CCX via ICT is transferred to a Cisco Agent Desktop with a SCCP phone in the central site.	PBX Ph1->QSIG Trunk->Unified CM->ICT (QSIG)->Unified CM->UCCX->Unified CM->6941 Phone (CAD)	Passed	
UC802EF.CRS.013	Unified CCX Call Transfer	Negative Testing of Unified CCX Redundancy	Verify the negative testing of Unified CCX redundancy.	SCCP Ph1->Unified CM->UCCX->6941 Ph (CAD)	Passed	
UC802EF.CRS.014	Unified CCX Redundancy	Blind Transfers Between Agents Cause Agents to be put in Reserved State	Verify if blind transfers between agents is causing agents to be put in Reserved State.	PSTN Ph1->MGCP PRI->Unified CM->UCCX->SCCP Ph1 (CAD)->XFR_B->SCCP Ph2 (CAD)	Passed	
UC802EF.CRS.015	Unified CCX	Unified CCX and Unity Connection Interaction	Verify the transfer scenario of Unified CCX and Unity Connection Interaction.	SCCP Ph1->Unified CM->Unity connection->UCCX->9971 (CAD)->XFR_C->SCCP Ph2->XFR_C->UCCX->SCCP Ph3 (CAD)	Passed	
UC802EF.CRS.016	Unified CCX	Unified CCX and Cisco Unity Auto Attendant Call Handler Interaction	Verify the call transfer from Cisco Unity Auto Attendant call handler to SCCP Cisco Agent Desktop Phone.	SCCP Ph1->Unified CM->Unity AA->UCCX->SCCP Ph1 (CAD)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CRS.017	Unified CCX	Agent Gets Stuck in Reserved When Call is Blind-Transferred from Cisco Unity to Unified CCX	Verify if the agent gets stuck in Reserved when a call is blind-transferred from Cisco Unity to Unified CCX.	SCCP Ph1->Unified CM->Unity->XFR_B->UCCX->SCCP Ph2 (CAD)	Passed	
UC802EF.CRS.018	Unified CCX	Agent Cannot Hang Up Call After Transfer to an Extension	Verify if an agent is not able to hang up a call after transferring to an extension.	PSTN Ph1->MGCP PRI->Unified CM->UCCX->SCCP Ph1 (CAD)->XFR_C->SCCP Ph2	Passed	
UC802EF.CRS.019	Unified CCX	Call to Cisco Unity Call Handler Redirected to Unified CCX Route Point	Verify if a call to Cisco Unity call handler gets redirected to Unified CCX route point.	SCCP Ph1->Unified CM->Unity->UCCX	Passed	
UC802EF.CRS.020	Unified CCX	Non-ICD Transferred Call to Agent Disconnected from Source	Verify if a non-ICD transferred call to an agent gets disconnected from source side.	SCCP Ph1->Unified CM->ICT (QSIG)->Unified CM->SCCP Ph1 (CAD)	Passed	
UC802EF.CRS.021	Unified CCX	H323 Video Phone Call to Unified CCX Agent	Verify if a call from a H323 video phone to Unified CCX is transferred to SCCP Cisco Agent Desktop agent in a remote site. The remote Cisco Agent Desktop agent consult transfers the call to a CSD supervisor in the central site.		Passed	
UC802EF.CRS.022	Extension Mobility	Extension Mobility of IP Phone A in IP Phone Agent B	Verify if IP Phone Agent A can perform an Extension Mobility in IP agent B to accept the queued call.	Stage 1:9971 Ph (CAD)->EM->SCCP Ph1(CAD);Stage 2:SIP Ph->Unified CM->UCCX->9971 Ph(CAD)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CRS.023	Unified CCX	CSD Supervisor's Silent Monitoring and Intercepting of Calls from a PSTN Phone	Verify if a call from a PSTN phone to Unified CCX via an MGCP gateway over BRI is transferred to a Cisco Agent Desktop with an SCCP phone in central site. A CSD supervisor in the central site silently monitors and intercepts the call.	Stage 1: PSTN Ph1->MGCP BRI->Unified CM->UCCX->Unified CM->SCCP Ph1(CAD) Stage 2: SCCP Ph2 (CSD)->Unified CM->UCCX->Unified CM->SCCP Ph1 (CAD)	Passed	
UC802IF.CRS.100	Unified Contact Center Express	High Availability over WAN Installation With Each Unified CCX Node Located in Different Time Zones	To Verify that Unified CCX can be installed successfully in high availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.CRS.100	Unified CCX	Unified Personal Communicator and VTIII	To Verify that Unified CCX can be installed successfully in High Availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.CRS.100	Unified CCX	Unified IP Phone 6911 and VTIII	To Verify that Unified CCX can be installed successfully in High Availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.CRS.100	Unified CCX	Unified IP Phone 6900 series and VTIII	To Verify that Unified CCX can be installed successfully in High Availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CRS.100	Unified CCX	Unified IP Phone 7916 and VTIII	To Verify that Unified CCX can be installed successfully in High Availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.CRS.100	Unified CCX	IP Communicator and VTIII	To Verify that Unified CCX can be installed successfully in High Availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.CRS.100	Unified CCX	Unified CME, Unified Video Advantage, and VTIII	To Verify that Unified CCX can be installed successfully in High Availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.CRS.100	Unified CCX	UC Integration™ for Microsoft Office Communicator and VTIII	To Verify that Unified CCX can be installed successfully in High Availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.CRS.100	Unified CCX	Unified Personal Communicator 8.0 and VTIII	To Verify that Unified CCX can be installed successfully in High Availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.CRS.100	Unified CCX	Unified CCX Installed Successfully in High Availability Over WAN	To Verify that Unified CCX can be installed successfully in High Availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CRS.101	Unified Contact Center Express	Remote Agents Functionality Where an Agent and Primary Unified CCX Node are Separated by WAN	To verify that agents located in one campus location get service from Unified CCX node located in different campus locations separated by WAN.	Agent->ASA->Unified CM->ASA->WAN->ASA->Unified CM->Unified CCX	Passed	
UC802IF.CRS.102	Unified Contact Center Express	PSTN Call Gets Anchored on Voice Gateway in One Data Center With Primary Unified CCX in Another Data Center	To verify that Unified CM invokes transcoder to allow Unified CCX to play Unified IP IVR when incoming call uses non G.711 codec.	PSTN Caller->Voice Gateway->Unified CM->ASA->WAN (G.729)->ASA->Unified CM->Unified CCX->ASA->Agent	Passed	
UC802IF.CRS.103	Unified Contact Center Express	Unified CCX Active/Master Node and Primary/Active CTI Manager Separated by WAN	Verify that Unified CCX can be integrated with Unified CM separated by WAN.	PSTN Caller->Voice Gateway->Unified CM->ASA->WAN (G.729)->ASA->Unified CM->Unified CCX->ASA->Agent	Passed	
UC802IF.CRS.104	Unified Contact Center Express	System Behavior During Unified CCX Node Failover and CTI Manager Failover	To validate the failover behavior of Unified CCX.	1: PSTN Caller->Voice Gateway->Unified CM->ASA->WAN(G.729)->ASA->Unified CM->Unified CCX->ASA->Agent. 2: PSTN Caller->Voice Gateway->Unified CM->Unified CCX->ASA->Agent	Passed	
UC802IF.CRS.105.1	Unified Contact Center Express	Validate Island Mode Operation of Unified CCX	To verify and validate the island mode moderation of Unified CCX deployed in High Availability over WAN when Unified CCX node is active due to WAN link failure.	PSTN Caller->Voice Gateway->Unified CM->Unified CCX->ASA->Agent	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CRS.105.2	Unified Contact Center Express	Validate Island Mode Operation of Unified CCX	To validate island mode moderation of Unified CCX deployed in High Availability over WAN when Unified CCX node is active due to WAN link failure. (test case: When WAN link is flapping, to make sure Unified CCX Primary Unified CCX node remains in stable operating condition and there is no loss of call processing and the system remains in stable island mode).	PSTN Caller->Voice Gateway->Unified CM->Unified CCX->ASA->Agent	Passed	
UC802IF.CRS.105.3	Unified Contact Center Express	Validate Island Mode Operation of Unified CCX After WAN Link Recovery	To validate island mode moderation of Unified CCX deployed in High Availability over WAN and to ensure that the Unified CCX nodes and database is setup properly after a WAN link recovery.	PSTN Caller->Voice Gateway->Unified CM->Unified CCX->ASA->Agent	Passed	
UC802IF.CRS.105.4	Unified Contact Center Express	Island Mode Operation of Unified CCX When CAD and Agent Phones are Initially Register to Different Sides on Datacenters	To validate island mode moderation of Unified CCX deployed in High Availability over WAN when Unified CCX node is active due to WAN link failure.	PSTN Caller->Voice Gateway->Unified CM->Unified CCX->ASA->Agent	Passed	
UC802IF.CRS.106.1	Unified Contact Center Express	Validate Agents with Multi-line Unified IP Phones and Single Line Unified IP Phones	To verify that Unified IP Phone series (9971, 9951, and 8961) and Unified IP Phone (6900 series) models can be used as agents.	Caller->ASA->Unified CM->>Unified CCX->ASA->Agent	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CRS.106.2	Unified Contact Center Express	Validate Agents with Multi-line Unified IP Phones and Single Line Unified IP Phones	To verify that Unified IP Phone series (9971, 9951, and 8961) and Unified IP Phone (6900 series) models can be used as agents.	Caller->ASA->Unified CM->Unified CCX->ASA->Agent	Passed	
UC802IF.CRS.106.3	Unified Contact Center Express	Validate Agents with Multi-line Unified IP Phones and Single Line Unified IP Phones [Test case 3]	To verify that Unified IP Phone series (9971, 9951, and 8961) and Unified IP Phone (6900 series) models can be used as agents.	Caller->ASA->Unified CM->Unified CCX->ASA->Agent	Passed	
UC802IF.CRS.107.1	Unified Contact Center Express	JTAL And DTAL Feature Support on Unified IP Phones With 4 Lines	To verify that Unified CCX can monitor multiple lines of Unified IP Phone models (9971, 9951, and 8961).	Caller->ASA->Unified CM->Unified CCX->ASA->Agent -1 (JAL/DTAL)->Agent-2	Passed	
UC802IF.CRS.107.2	Unified Contact Center Express	JTAL And DTAL Feature Support on Unified IP Phone with 4 Lines	To verify if Unified CCX can monitor multiple lines of Unified IP Phone (9971, 9951, and 8961) models.	Caller->ASA->Unified CM->Unified CCX->ASA->Agent -1 (JAL/DTAL)->Agent-2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.CCM.080.25	Basic Call Flow	Do Not Disturb	Verifies basic functionality of the Do Not Disturb (DND) feature.		Passed	
SR60.CCM.410.5	Conference	Shared Line Secure MeetMe Conference	To verify if Secure Conferencing properly works with MeetMe Conference when a shared line has a different security setting on each endpoint.		Passed	
SR60.CCM.603.12	Basic Call Flow	CODEC Support: Phone to Application	To verify calls between phones and applications using different CODECs.		Passed	
UC700IF.CCM.050	Unified CM	BLF For Speed Dial for SIP IP Phone7916	To verify if a user can configure BLF-Speed dial for first 3 lines, middle 3 lines and last 3 lines. There are 36 lines with the two Unified IP Phone 7916 expansion modules. Monitor the presence status of the numbers configured.		Passed	
UC700IF.CCM.051	Unified CM	TNP SCCP Phone With Extension Module Configured	Verify that a user logged in a TNP SCCP phone with extension module configured for BLF speed dial for can see the presence status.		Passed	
UC700IF.CCM.052	Unified CM	Unified CM Presence Status on Secure Guiness Phone	Verify that Unified CM presence status works on a Secure Guiness phone for call history lists, BLF speed dials, and SIP URI.		Passed	
UC700IF.CCM.052	Unified CM Presence	Unified CM Presence Works on Secure IP Phone 7900 series for Call History lists, BLF Speed dials, and SIP URI	Verify that Unified CM presence works on a Secure IP Phone 7900 series for Call History lists, BLF Speed dials, and SIP URI.		Failed	CSCtc32278

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC700IF.CCM.060	Basic Call Flow	Unified CM Gatekeeper Based CAC	To verify that Unified CM Gatekeeper based CAC works as defined in UC 6.X SRND.	Phone->Unified CM (ZoneA)->GateKeeper Unified CM (ZoneB)->Phone	Passed	
UC700IF.CCM.061	Basic Call Flow	Unified CM Gatekeeper Based CAC With Video	To verify that Unified CM Gatekeeper based CAC works as defined in UC 6.X SRND.	Phone->Unified CM (ZoneA)->GateKeeper Unified CM (ZoneB)->Phone	Passed	
UC700IF.CCM.062	Basic Call Flow	Unified CM Gatekeeper Based CAC With Session Bandwidth Restrictions	To verify if Unified CM Gatekeeper based CAC works as defined in UC 6.X SRND.	Phone->Unified CM (ZoneA)->GateKeeper Unified CM (ZoneB)->Phone	Passed	
UC700IF.CCM.063	Basic Call Flow	Unified CM Gatekeeper Based CAC With Remote Zones	To verify that Unified CM Gatekeeper based CAC works as defined in UC 6.X SRND.	Phone->Unified CM (ZoneA)->GateKeeper1->GateKeeper2->Unified CM (ZoneB)->Phone	Passed	
UC700IF.CCM.098	Basic Call Flow	Unified CM Static Location based CAC Video Enabled Endpoints	To verify if Unified CM detects audio bandwidth from the location when video enabled phone makes an audio call.	Video Phone->(Region 1)->Unified CM->(Region 2)->Audio Phone	Passed	
UC700IF.CCM.099	Basic Call Flow	Unified CM Static Location based CAC Adjusts Bandwidth Usage	To verify if Unified CM Static Location based CAC can adjust the bandwidth usage when video call is downgraded to audio only call.	Video Phone->(Region 1; Location 1)->Unified CM->(Region 2; Location 2)->Audio Phone	Passed	
UC700IF.CCM.140.3	Unified CM	Local Route Group And Transformation	Verify to ensure that Unified CM correctly routes calls using 'Virtual Local Route Group' provisioning.	siteA endpoint->Unified CM->siteA PSTN; siteA endpoint->Unified CM->siteA PSTN (no bandwidth to go to toll free site); Called number transformed; siteA endpoint->Unified CM->siteA PSTN (route to toll free site); Called number transformed	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC700IF.CCM.604	Unified CM	Conference Chaining of Secure and NonSecure Conferences	To verify the conference chaining of secure and non-secure Conferences.	Phone->Unified CM->ConfBridge	Passed	
UC712EF.CCM.007	IPV6	Dual-stack Phone Operation Based on Media and Signaling Preference Settings	Verify a call from the NON DS cluster to the DS Phone on the DS cluster which CFWDALL is set to DS remote Phone on the DS cluster over SIP GW(DS).	Stage1: SCCP Ph1->Unified CM->QSIG ICT->Unified CM (DS)->SCCP Ph2 (DS); Stage2: SCCP Ph2(DS)->CFWDALL->PSTN GW (SIP GW DS)-> PSTN->PSTN GW (SIP GW DS)->Unified CM(DS)-> Rem SCCP Ph3(DS)	Passed	
UC712EF.CCM.009	IPV6	Transferring the Call from IPv4 Phone on DS Cluster to VG224 GW POTS Phone	Verify a call from the NON DS cluster to the IPv4 Phone on the DS cluster and then transfer it to VG224 GW POTS Phone.	SIP Ph1->Unified CM ->QSIG ICT->Unified CM(DS)->SCCP Ph1->XFER->Unified CM(DS)->VG224 GW(DS)->POTS Ph1	Passed	
UC712EF.CCM.021	IPV6	Voicemail Deposit and Retrieval for Central Site DS Phone	Verify if a call from a PSTN phone to central site DS phone is forwarded to voicemail in Unity connection.	Stage1: PSTN Ph1->MGCP GW->Unified CM DS->DS Ph1->CFNA->Unity Connection	Passed	
UC712EF.CCM.022	IPV6	Voicemail Deposit and Retrieval for Remote Site DS Phone	Verify the call from a PSTN phone via SIP Gateway to remote site DS phone is forwarded to voicemail in Unity connection.	Stage1: PSTN Ph1->SIP GW(DS)->Unified CM DS->Rem DS Ph1->CFNA->Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF.CCM.023	CAC	Location Based CAC with IPv4/IPv6 Interworking	Verify if a user can place a call from a SCCP phone A to SIP Phone B which are in different locations and later initiate consultative transfer from SIP Phone B to SCCP Phone C resulting speech path established between Phone A & Phone B and consultative transfer fails with prompt " Not enough bandwidth".	Rem SCCP Ph A (DS) > Unified CM (DS) > SIP Ph B > Xfer_C > Unified CM > CAC > Rem SCCP Ph C	Passed	
UC712EF.CCM.024	IPv6	Intercom Call from SCCP (IPv4) to SCCP (DS) Phone	Verify is a user can make an intercom call from SCCP phone to another remote SCCP phone in different site. SCCP phone which is target gets the tone and goes on speaker mode with audio muted, after "talk" softkey is pressed two ways audio is established by the target phone.	Rem SCCP Ph 1 (v4) > Intercom > Unified CM (DS) > Rem SCCP Ph 2 (DS)	Passed	
UC712EF.CCM.025	IPv6	Unified CME Call to SCCP Phone in DS Cluster Having CFNA to IPMA Manager phone in Non-DS cluster	Verify the call from Unified CME SCCP Phone to a QSIG PBX phone in interop site which is transferred to SCCP Ph (DS) in central DS site which has CFNA to IPMA phone in Non-DS site.	SCCP Ph 1 > CME > IPIPGW(H323) > GK > Unified CM (interop) > QSIG Trunk > PBX > PBX Ph 1 > Xfer > ICT > Unified CM(DS) > SCCP Ph (DS) > CFNA > ICT > Unified CM (Non-DS) > IPMA Manager Ph 1 (v4)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF.CCM.026	IPv6	IPv4/IPv6 Interworking When DS Phone Interacts with SIP IPv4 Phone in Interop Site which has CFwdAll set to UCCX IPPA Phone	Verify the call from SCCP (DS) phone from DS cluster to SIP IPv4 phone in Interop site via ICT which has CFwdAll set to UCCX which transfers the call to IPPA agent in the remote site.	SCCP Ph (ds) > Unified CM (DS) > ICT > SIP Ph > Unified CM (Interop) > CFwdAll > ICT (QSIG) > Unified CM > UCCX > Unified CM > Rem SCCP Ph 1 (IPPA)	Passed	
UC712EF.CCM.027	CAC	Unity Connection Call Failing Due to Unavailable Bandwidth	Verify that Voice mail deposit can not be done when policy is mandatory and when there is not enough bandwidth available for unity connection system from target phone.	Rem SCCP Ph A (ds) > Unified CM (DS) > Rem SCCP Ph B > CFNA > Unified CM(DS) > CAC > Unity Connection VM#	Passed	
UC712EF.CCM.028	CAC	Call Back Notification Under Limited Bandwidth Condition and IPv4/IPv6 Interworking	Verify if a user can place a call from SCCP Phone 1 in central DS site to another Busy Rem SCCP Phone 2, activate the call back feature, free SCCP Phone 2 where there is not enough bandwidth situation available between Phone 1 and Phone 2, with call back notification sent and ultimately call should fail with prompt "not enough bandwidth" on SCCP phone in central DS phone.	SCCP Ph 1 (ds) > Unified CM(DS) > Rem v4 SCCP Ph 2(Busy)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF.CCM.029	IPv6	IPv4/IPv6 Interactions for IPMA in Shared Line Mode and Using Extension Mobility for Manager	Verify IPMA in shared line mode in central DS cluster. Configure IPMA to use EM phone for manager, login as EM user to manager phone such that after EM login, manager has shared line with assistant, make a call to manager phone(SCCP Ph) from SCCP ph in Unified CM(DS).	Rem SCCP Ph (DS) > Unified CM (DS) > SCCP IPMA manager phone	Passed	
UC712EF.CCM.030	IPv6	IPv4/IPv6 Interworking and MTP Interactions During Blind Transfer & CFNA to Unity Connection VM	Verify if a PSTN call made to IPv4 SIP IP phone in central site can do blind transfer to DS SCCP phone in Remote 3 site which has CFNA to Unity Connection VM.	PSTN Ph > MGCP GW > Unified CM (ds) > IPv4 SIP phone > Blind Transfer > Rem SCCP phone (DS) > CFNA > Unity Connection	Passed	
UC712EF.CCM.031	IPv6	Interaction of BRI PSTN Call from Interop Site to Dual Stack Phone	Verify if a user can make a PSTN call from to SIP phone in interop site, answer the call and do a consult transfer to SCCP DS phone in remote site which is in active-active or active-stand mode.	Rem SCCP Ph 1 > PSTN GW (FXO interface) > Unified CM (Interop site) > SIP ph > XFER_C > Unified CM (ds) > Rem SCCP ph (DS)	Passed	
UC712EF.CCM.032	MTP	PSTN Caller is Consult Transferred to VG224 Analog Phone	Verify if a PSTN call from central DS cluster to Rem SCCP (ds) Phone can consult transfer to VG224 Analog phone.	PSTN > H323 GW > Unified CM (DS) > Rem SCCP Phone (ds) > Consults > VG224 Ph > Answers > Completes Transfer	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF.CCM.033	IPv6	Conferencing During Load Balancing Condition with IPv4/IPv6 Interworking	Verify the following: SIP IPv4 phone calls SCCP (ds) phone in one remote, SIP IPv4 phone calls SCCP (ds) ph in another remote site which is in active-active / active-standby mode. When remote 3a goes down, connected call can be handled by remote 3b with out any interruption. Having established the call between SIP IPv4 and SCCP (ds), SIP IPv4 phone completes conference by pressing conf softkey.	SIP ipv4 > Unified CM > Rem SCCP ph	Passed	
UC712EF.CCM.034	IPv6	IPv4/IPv6 Interworking and MTP Interactions During Call back and DND scenario	Verify the following: Call from a SCCP phone (ds) in central DS cluster which has Do Not Disturb (DND) feature activated to a remote SCCP phone that is busy. The SCCP phone in the central site should have Callback activated. Make a call from a DS Central SCCP phone to Remote SCCP phone to get a busy tone. Central SCCP phone should press Callback softkey and go on hook. Originating SCCP phone should be alerted to say called remote SCCP phone is free.	SCCP Ph (DS) > Unified CM (DS) > Rem SCCP Ph (dnd activated) Call gets connected	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF.CCM.035	IPv6	Multiple Line Appearance & Join Operation with IPv4/IPv6 Interworking	Verify the following: SCCP phone (ds) in central site has 2 lines. Make a call from Rem SCCP ph to SCCP phone in central site on line 1. Again make another call from SIP ph in central site to SCCP ph on line 2, now select both the calls on both lines and press Join Key.	Rem SCCP Ph > Unified CM (DS) > SCCP Ph Line 1 (DS	Passed	
UC712EF.CCM.036	IPv6	IPv4/IPv6 Interworking in SRST Mode during Unified CM Failure	Verify if a user can make a call from SCCP (DS) phone from central DS site to remote SIPv4 phone in SRST mode which does blind transfer to PSTN phone.	SCCP Ph (ds) > Central SRST > Rem SRST > Rem SIP Ph	Passed	
UC712EF.CCM.037	IPv6	IPv4/IPv6 Interworking for a Unified SRST Remote Call to Central Site User Which goes to Unity Connection VM	Verify the following: Make a PSTN call from remote SIP IPv4 phone to central DS SCCP Phone, remote site has WAN down so the call goes over PSTN to central site which has CFNA to Unity Connection VM.	Rem Ph > Rem SRST > PSTN > Central SCCP Ph	Passed	
UC712EF.CCM.038	IPv6	Legacy and Dual Stack Phone Interop with Dual-Stack SCCP Analog Line Gateway	Verify if a PBX phone from a non-dual stack site can call to an Dual Stack phone in a dual stack site and then can be transferred to a FXS phone over SCCP gateway.	PBX ph1->QSIG trunk->Unified CM (Non DS)->QSIG ICT->Unified CM (DS)->SCCP ph1 (DS)->XFER->SCCP GW(DS)->FXS Ph1	Passed	
UC712EF.CCM.052	Blind Conference call	Blind Conference of PSTN phone ~ SIP Phone Call to a CME Phone	Verify the call from a PSTN Phone to SIP Phone, and the SIP Phone blind conferences the call to CME phone.	PSTN 1->MGCP PRI->CCM->SIP PHONE->Blind conference->IP-IP GW (H.323)->CME phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF.CCM.054	Consult Conference Call	Consult Conference of PBX Phone, SCCP Phone and CME Phone	Verify if a Consult Conference call among PBX phone,SCCP Phone and CME Phone under same cluster, where the SCCP Phone and CME phone are in same geo-location is possible.	Phone->QSIG PBX->CCM->SCCP Ph ->Consult conference->H.323 IP-IP GW->CME phone	Passed	
UC712EF.CCM.057	Call Forwarding	Call Forwarding Call from IP call to CME phone	Verify if a user can make a Call forwarding in IP call to a CME phone call.	SIP ph->CCM->SCCP Ph->CALL FORWARD->H.323(IP-IP)GW->CME ph	Passed	
UC712EF.CCM.058	Call Forwarding Call	Call Forwarding Call from a IP call to PSTN Phone	Verify if a user can make a Call forwarding in IP call to a PSTN phone call in another geo-location.	Phone->Qsig PBX->CCM->SCCP Ph->CALL FORWARD->>MGCP PRI->PSTN	Passed	
UC712EF.CCM.059	Park Retrieval	Park Retrieval from Different Cluster	Verify if SIP Phone From Cluster 1 calls SIP phone in Cluster2 and the SIP Phone in Cluster 2 parks the call, and Call parking retrieval is done from PSTN phone.	SIP Phone->CCM->ICT->CCM->SIP ph;SIP Ph Parks the call;	Passed	
UC712EF.CCM.060	EM and JAL	Extension Mobility Call in Same Cluster	Verify the following scenario: SCCP Phone A EM login to another SCCP Phone B in same Cluster, where both phones are in different geo-locations. SIP Phone A makes a PSTN call to SIP phone B in geo-location X and SIP phone B transfer the call to SCCP ph A which had done EM and now in Geo-location Y. The call transfer should be denied.	SCCP Ph 1->CCM->EM->SCCP Ph 2 ;SIP Ph 1->MGCP PRI->PSTN->CCM->SIP Ph 2;SIP Ph 2->Blind transfers->CCM->SCCP Ph 2	Passed	
UC713EF.CCM.152	Tandem Cluster	Interaction with SIP and Annex M1 trunks with call Transfer remote			Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713EF.CCM.153	Tandem Cluster	Interaction with SIP and Annex M1 trunks with call Transfer Local			Passed	
UC713EF.CCM.156	Tandem Cluster	Interaction with SIP and Annex M1 trunks with QSIG PBX			Passed	
UC713EF.CCM.160	Tandem Cluster	Interaction of Tandem SIP trunks with iLBC			Passed	
UC713EF.CCM.164	Tandem Cluster	Interaction of Tandem Unified CM with Round Robin trunks			Passed	
UC713EF.CCM.201	Direct Call Park	Direct Call Park to a Different Geo-location	Verify the following: SIP Phone 1 uses it PSTN line to call another SIP Phone 2 in same geo-location. SIP Phone 2 directly park the call to a VOIP Phone in Different Geo-location.	SIP Ph A->Unified CM->MGCP PRI->PSTN Gateway->MGCP PRI->SIP Ph B;SIP Ph B-->Direct Call Park->Unified CM->SIP Ph C.	Passed	
UC713EF.CCM.202	Call Pick up	Call Pick up from Different Geo-location	Verify when a SCCP Phone A uses its PSTN line to make PSTN call to another SCCP Phone B in same geo-location X. A SIP Phone in Geo-location Y of same Pick group of SCCP Phone B tries to pick the call.	SCCP Ph A->Unified CM->MGCP PRI->PSTN->MGCP PRI->SCCP Ph B;SIP Ph A->call pick up->SCCP Ph B	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713EF.CCM.203	Meet Me conference	Meet Me Conference among Different Geo-location Phone	Verify the following scenario: SCCP Phone A in Geo-location X attends meet conference through VOIP line. SCCP Phone B in Geo-location Y attends meet conference through VOIP Line. SIP Phone A of Geo-location X attends Meet me conference through PSTN Line.	SCCP Ph A->CCM->Meet me Conference; SCCP PhB->CCM->meet me conference;SIP PhA->MGCP PRI->PSTN->Meet me conference;	Passed	
UC713EF.CCM.204	Shared Line	Shared Lines in Different Geo-location	Verify the following scenario: SCCP Phone A and SCCP Phone B are shared line Phones and SCCP Ph A is in Geo-location X and SCCP Phone B in Geo-location Y. When PSTN call in Geo-location X is transferred as VOIP call to Shared lines, only SCCP Phone A should be able to accept the call.	SCCP Ph C->Unified CM->SIP Ph A;SIP Ph A->Blind transfer->Unified CM->SCCP Ph A(Shared line with SCCP B);	Passed	
UC802.CCM.201	Session Management Edition	UC Integration™ for Microsoft Office Communicator Calls Over Unified CM Session Management Edition	Verify that UC Integration for Microsoft Office Communicator clients in leaf clusters can call each other over Unified CM Session Management Edition (audio calls).	UC Integration™ for Microsoft Office Communicator->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->UC Integration™ for Microsoft Office Communicator	Passed	
UC802.CCM.201	Session Management Edition	UC Integration™ for Microsoft Office Communicator Calls Over Unified CM Session Management Edition	Verify that UC Integration for Microsoft Office Communicator clients in leaf clusters can call each other over Unified CM Session Management Edition(audio calls).	UC Integration™ for Microsoft Office Communicator->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->UC Integration™ for Microsoft Office Communicator	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802.CCM.201	Session Management Edition	UC Integration™ for Microsoft Office Communicator Calls Over Unified CM Session Management Edition	Verify that UC Integration for Microsoft Office Communicator clients in leaf clusters can call each other over Unified CM Session Management Edition(audio calls).	UC Integration™ for Microsoft Office Communicator->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->UC Integration™ for Microsoft Office Communicator	Passed	
UC802.CCM.202	Session Management Edition	VG224 Calls Over Unified CM Session Management Edition	Verify that VG224 clients in leaf clusters can call each other over Unified CM Session Management Edition.	Vg224->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->Vg224	Passed	
UC802.CCM.202	Session Management Edition	VG224 Calls Over Unified CM Session Management Edition	Verify that VG224 clients in leaf clusters can call each other over Unified CM Session Management Edition.	Vg224->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->Vg224	Passed	
UC802.CCM.202	Session Management Edition	VG224 Calls Over Unified CM Session Management Edition	Verify that VG224 clients in leaf clusters can call each other over Unified CM Session Management Edition.	Vg224->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->Vg224	Passed	
UC802.CCM.203	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using UC Integration™ for Microsoft Office Communicator Clients	Verify that UC Integration for Microsoft Office Communicator clients in leaf clusters can make a video call over Unified CM Session Management Edition (video calls).	UC Integration™ for Microsoft Office Communicator->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->UC Integration™ for Microsoft Office Communicator	Passed	
UC802.CCM.204	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using UC Integration™ for Microsoft Office Communicator Clients and Unified Video Advantage	Verify that UC Integration for Microsoft Office Communicator clients and Unified Video Advantage in leaf clusters can make a video call over Unified CM Session Management Edition	UC Integration™ for Microsoft Office Communicator->Unified CM1->QSIG ICT->Unified CM-Tand->QSIG ICT->Unified CM2->CUVA	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802.CCM.204	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using UC Integration™ for Microsoft Office Communicator Clients and Unified Video Advantage	Verify that UC Integration for Microsoft Office Communicator clients and Unified Video Advantage in leaf clusters can make a video call over Unified CM Session Management Edition.	UC Integration™ for Microsoft Office Communicator->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->CUVA	Passed	
UC802.CCM.204	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using UC Integration™ for Microsoft Office Communicator Clients and Unified Video Advantage	Verify that UC Integration for Microsoft Office Communicator clients and Unified Video Advantage in leaf clusters can make a video call over Unified CM Session Management Edition.	UC Integration™ for Microsoft Office Communicator->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->CUVA	Passed	
UC802.CCM.205	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using UC Integration™ for Microsoft Office Communicator Clients and 69xx series IP Phones	Verify that UC Integration for Microsoft Office Communicator clients and 69xx series IP phones in leaf clusters can make a video call over Unified CM Session Management Edition.	UC Integration™ for Microsoft Office Communicator->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->RT	Passed	
UC802.CCM.205	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using UC Integration™ for Microsoft Office Communicator Clients and 69xx series IP phones	Verify that UC Integration for Microsoft Office Communicator clients and 69xx series IP phones in leaf clusters can make a video call over Unified CM Session Management Edition.	UC Integration™ for Microsoft Office Communicator->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->RT	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802.CCM.205	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using UC Integration™ for Microsoft Office Communicator Clients and 69xx series IP phones	Verify that UC Integration for Microsoft Office Communicator clients and 69xx series IP phones in leaf clusters can make a video call over Unified CM Session Management Edition.	UC Integration™ for Microsoft Office Communicator->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->RT	Passed	
UC802.CCM.206	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using Unified Video Advantage and 69xx series IP Phones	Verify that Unified Video Advantage clients and 69xx series IP phones in leaf clusters can make a video call over Unified CM Session Management Edition.	CUVA->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->RT	Passed	
UC802.CCM.206	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using Unified Video Advantage and 69xx series IP phones	Verify that Unified Video Advantage clients and 69xx series IP phones in leaf clusters can make a video call over Unified CM Session Management Edition.	CUVA->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->RT	Passed	
UC802.CCM.206	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using Unified Video Advantage and 69xx series IP phones	Verify that Unified Video Advantage clients and 69xx series IP phones in leaf clusters can make a video call over Unified CM Session Management Edition.	CUVA->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->RT	Passed	
UC802.CCM.207	Session Management Edition	Service Advertisement Framework (SAF) Over Unified CM Session Management Edition		Route Pattern->Unified CM1->SAF->Unified CM-Tand->SAF->Unified CM2	Passed	
UC802.CCM.208	Session Management Edition	RSVP Over Unified CM Session Management Edition		Unified CM1(RSVP)->SIP->Unified CM-Tand->SIP->(RSVP)Unified CM2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CCM.001	Unified CCX	PSTN Call to a Central Site	Verify the call from a PSTN phone to a central site SCCP phone over MGCP gateway which can transfer the call to a Cisco Agent Desktop (CAD) with a SIP phone in central site.	PSTN Ph1->MGCP PRI->PSTN->MGCP PRI->Unified CMBE->SCCP Ph1->XFER_B->Unified CMBE-> UCCX->Unified CMBE->99xx/89xxSIP Ph1 (CAD)	Passed	
UC802EF.CCM.002	Unified CCX	PSTN Call from a Remote SCCP Phone to a Central Site	Verify the call from a remote SCCP phone to a central site SIP Phone acting as a CAD agent over H.323 gateway (FXO/BRI/PRI) and then transfer the call to remote agents.	Rem SCCP Ph1->Unified CMBE->H323 GW->PSTN->H323 GW->Unified CMBE->UCCX->Unified CMBE->SIP Ph1 (CAD)->XFER_C->Unified CMBE->UCCX->Unified CMBE->Rem SCCP Ph2 (CAD)	Passed	
UC802EF.CCM.003	Unified CCX	PSTN Call from a Remote SCCP Phone to a Central Site	Verify the call from a remote SCCP phone to a central site SIP Phone acting as CAD agent over SIP gateway (BRI/PRI) and conference the call to remote agents.	Rem SCCP Ph1->Unified CMBE->SIP GW->PSTN->SIP GW->Unified CMBE->UCCX->Unified CMBE->SIP Ph1 (CAD)->CNF->Unified CMBE->UCCX->Unified CMBE->Rem SCCP Ph2(CAD)	Passed	
UC802EF.CCM.004	Unified CCX	IM session on IP phones, Remote CAD agent, IP Phones 9971/9951/8961 and Unified Personal Communicator	Verify if IM from IP Phones 9971/9951/8961 and SIP Phone acting as CAD agent in remote sites to Unified Personal Communicator in central site is possible and also check the presence status of all buddies. Also verify if IM from SIP/SCCP (IPPM clients) to IP Phones 9971/9951/8961 and remote site SIP Phone acting as CAD agent is possible and check the presence status of all buddies.	V 1 :Rem 99xx/89xxSIP Ph1->CUP->SCCP Ph/SIP Ph/ CUPC; V 2: Rem SIP Ph1 (CAD)->Unified CMBE->UCCX->Unified CMBE->CUP-> SCCP Ph/SIP Ph/ CUPC	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CCM.005	Unified CCX	IM session on IP phones, Central CAD agent, IP Phones 9971/9951/8961 and Unified Personal Communicator	Verify if IM from IP Phones 9971/9951/8961 and SIP Phone acting as CAD agent in central sites to Unified Personal Communicator in central site is possible and also check the presence status of all buddies. Also verify if IM from SIP/SCCP (IPPM clients) to IP Phones 9971/9951/8961 and central site SIP Phone acting as CAD agent is possible and also check the presence status of all buddies.	Variation 1 :99xx/89xxSIP Ph1->CUP->SCCP Ph/SIP Ph/ CUPC Variation 2: SIP Ph1 (CAD)->Unified CMBE->UCCX->Unified CMBE->CUP-> SCCP Ph/SIP Ph/ CUPC	Passed	
UC802EF.CCM.006	Unified CCX	IM session on IP phones, Central CAD agent, IP Phones 9971/9951/8961 and Unified Personal Communicator	Verify if IM from IP Phones 9971/9951/8961 and SIP/SCCP (IPPM clients) to Unified Personal Communicator/CAD agent in central site is possible and can also check the presence status of all buddies when both CAD agent and Unified Personal Communicator applications are installed on the same laptop.	99xx/89xxSIP Ph1/SCCP Ph1/SIP Ph1 ->CUP->CAD agent/ CUPC	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CCM.007	Unified Personal Communicator	Video Calls with RSVP on Video Phones, IP Phones 9971/9951 and Unified Personal Communicator	Verify the following: 1. Video calls can be made between video phones, IP phones, IP Phones 9971/9951/8961 and Unified Personal Communicator with RSVP enabled. 2. IM from IP Phones 9971/9951 and video phone in remote site to Unified Personal Communicator in central site is possible and can also check the presence status of all buddies. 3. IM from SIP/SCCP (IPPM clients) to IP Phones 9971/9951 and Video phone in remote site is possible and can also check the presence status of all buddies. 4. Can exhaust the available video bandwidth on a remote site. 5. Can make a video call which goes as audio only due to lack of video bandwidth.	RT/Video Ph->REM->WAN->CUP->Unified CMBE->SCCP Ph/SIP Ph/ CUPC	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CCM.008	Unified Personal Communicator	Unified Personal Communicator Inter-Working with Extension Mobility	Verify if Unified Personal Communicator in desk phone mode can control a user logged in using extension mobility. A central site user logs into a remote site phone using EM feature. Unified Personal Communicator which was earlier configured for desk phone control of the above user in central site should still work. Check the presence status of this user on Unified Personal Communicator and check IM between Unified Personal Communicator and the EM user.	CUPC->CUP->Unified CMBE->WAN->Rem->EM Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CCM.009	Unified Personal Communicator	Inter-Working with Unified Attendant Server Console	Verify the interworking of Unified Attendant Server and Unified Personal Communicator. The IM session and call transfer from Unified Attendant Server to Unified Personal Communicator in desk phone can control a PSTN call made to a remote site via a FXO. The call lands on the Unified Attendant Server and the Unified Attendant Server console checks the presence status of the user to which the call has to be transferred. An IM is sent to the user to take the call and the call is transferred, Unified Personal Communicator in desk phone mode picks up the call.	PSTN->FXO->REM->Unified CMBE->ARC ARC->Unified CMBE->CUP->CUPC	Passed	
UC802EF.CCM.010	Unified IP Phones 6921/6941/6961	Unified IP Phones 6921/6941/6961 in SRST mode	Verify if a PSTN call can be made from a remote Unified IP Phones 6921/6941/6961 in Unified SRST mode to another remote phone in SRST mode which has CFWDALL set to a PSTN Phone.	Rem RT-Lite SCCP Ph1->SRST1->PSTN GW->PSTN->PSTN GW->SRST 2->Rem SCCP Ph2;Rem SCCP Ph2->CFWDALL->PSTN GW->PSTN Ph1	Passed	
UC802EF.CCM.011	Unified IP Phones 9971/9951/8961	Unified IP Phones 9971/9951/8961 Interworking with IP Communicator	Verify if a call can be made from the SIP IP Communicator to the remote SCCP Phone which has CFNA set to a remote Unified IP Phones 9971/9951/8961.	SIP CIPC1->Unified CMBE->Rem SCCP Ph1; Rem SCCP Ph1->CFNA->Rem 99xx/89xxSIP Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CCM.012	Unified IP Phones 9971/9951/8961	Unified IP Phones 9971/9951/8961 Interworking with Unified CCX	Verify if a call made from a PSTN phone to a SCCP phone is transferred to Unified CCX CAD agent with a Unified IP Phones 9971/9951/8961.	PSTN Ph1->MGCP PRI->Unified CMBE->SCCP Ph1->XFER_B->Unified CMBE-> UCCX->Unified CMBE->99xx/89xxSIP Ph1(CAD)	Passed	
UC802EF.CCM.013	Unified IP Phones 9971/9951/8961	Call Park Using IP Phones 9971/9951/8961	Verify if a call from a PSTN Phone to the IP Phones 9971/9951/8961 (SIP Phone 1 and Remote IP Phones 6921/6941/6961 SCCP Phone 2 are shared lines) over PSTN Gateway is successful. IP Phones 9971/9951/8961 parks the call and remote IP Phones 6921/6941/6961 retrieves the call.	Stage 1: PSTN Ph 1->Unified CMBE->PSTN GW->PSTN->PSTN GW->Unified CMBE->99xx/89xxSIP Ph1; Stage 2: 99xx/89xxSIP Ph1->answers and parks the call; Stage 3: Rem 69xx SCCP Ph2->retrieves the call	Passed	
UC802EF.CCM.014	Unified IP Phones 9971/9951/8961	Hunt List using Unified IP Phones 9971/9951/8961	Verify if a call made from a Central SCCP Phone to a Central IP Phone 9971/9951/8961 is not answered, then the call should go to all members in the hunt group until it is answered depending upon the algorithm configured. Hunt group should have a combination of Unified IP Phones 9971/9951/8961, Unified IP Phones 6921/6941/6961s and Unified Personal Communicator.	SCCP Ph1->Unified CMBE->99xx/89xxSIP Ph1; 99xx/89xxSIP Ph1->Don't answer; The call Should go to all the members in the hunt group until it is answered depending upon the algorithm configured.	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CCM.015	Unified IP Phones 9971/9951/8961	Depositing VM from Unified IP Phones 6921/6941/6961 using Unity Connection to Unified IP Phones 9971/9951/8961	Verify if a PSTN call can be made from the Unified IP Phones 6921/6941/6961 to a remote IP Phone 9971/9951/8961 over H.323/SIP gateways.	Stage 1: 69xx SCCP Ph1->Unified CMBE->PSTN GW->PSTN->PSTN GW->Unified CMBE->Rem 99xx/89xxSIP Ph1->CFNA->UNC; Stage 2: 69xxSCCP Ph1->Deposits a VM; Stage 3: Rem 99xx/89xxSIP Ph1->Retrieves a VM	Passed	
UC802EF.CCM.016	Unified IP Phones 9971/9951/8961	Hold Reversion with Extension Mobility Using Unified IP Phones 6921/6941/6961 and Unified IP Phones 9971/9951/8961	Verify the hold reversion feature with Extension Mobility by creating an Extension Mobility profile that is similar to one of the SIP Phones in the central site. Use this profile to login from a remote Unified IP Phones 6921/6941/6961 and IP Phones 9971/9951/8961.	Stage 1:PSTN Ph 1->PSTN GW->Unified CMBE->SIP Ph 1->Hold Stage 2:SIP Ph 1 ->Resume	Passed	
UC802EF.CCM.201	Extension Mobility Cross Cluster	Depositing a Voicemail from HC Phone to the VC Logged in Phone	Verify that Extension Mobility Cross Cluster works and is able to retrieve successfully the voice mails from the VC logged in phones.	IP Phone 99xx/89xx/69xx->EMCC->TNP Phone;	Passed	
UC802EF.CCM.202	Extension Mobility Cross Cluster	Extension Mobility Cross Cluster from Remote Phones to Another Cluster	Verify that Extension Mobility Cross Cluster works from IP Phones 8961/9951/9971 and gets the profile of the other cluster remote TNP Phones.	IP Phone 99xx/89xx/69xx->Unified CM1->SIP ICT->Unified CM2->TNP Phone	Passed	
UC802EF.CCM.203	Extension Mobility Cross Cluster	Extension Mobility Cross Cluster from the IP Phones 8961/9951/9971 in V4 cluster to TNP Phone in DS cluster	Verify that Extension Mobility Cross Cluster works from the IP Phones 8961/9951/9971 in V4 cluster and gets the profile of the dual stack cluster TNP Phones.	IP Phone 99xx/89xx/69xx->Unified CM1(V4)->SIP ICT->Unified CM2(DS)->TNP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CCM.021	Unified CM	iPhone Client Joining Unified MeetingPlace Internal Meeting by Receiving VOIP Call and Handoff Call to Deskphone	To verify that iPhone client can join a Unified MeetingPlace hosted meeting by answering call back. When meeting in progress iPhone handoff the call to deskphone.	iPhone end point (Unified CM(Unified MeetingPlace	Passed	
UC802IF.CCM.109	SCCP Adhoc Conference	Adhoc Conference Using Unified MeetingPlace via H.225 Trunk	Verify if video conference can be established between 3 parties with H.225 trunk between Unified CM and Unified CME.	7985 -CME--- H225 ---7985 -- Unified CM --- Conference 7985 ---h225 --- Unified CM	Passed	
UC802IF.CCM.170	Unified CM	E911 Call Handling When Users Logs in Across Cluster Using EMCC Feature	To verify if the E911 call from users logged in using EMCC feature in visiting Unified CM cluster is routed to local PSAP.	IP Phone->ASA->Unified CM->Emergency Responder->PSAP	Passed	
UC802IF.CCM.200	Unified CM	iPhone Client Joins MeetMe Conference with Handoff Call to GSM	Verify if iPhone client is joining a conference by dialing meetme number and while in conference going out of Wifi coverage, continue the call in GSM.	iPhone end point (Unified CM	Passed	
UC802IF.CCM.203	Unified CM	iPhone Client Receiving a Cisco IME Call and Setting Up Adhoc Conference	To verify that iPhone client can answer an incoming IME call and then setup an adhoc conference.	iPhone end point (Unified CM	Passed	
UC802IF.CCM.204	Unified CM	iPhone Client Setting Up an IME Call and Transferring the Call to PSTN	To verify that iPhone client can set up an inter enterprise IME call and then transfer the call to a PSTN destination through H323 gateway.	iPhone end point (Unified CM(H323GW	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CCM.205	Unified CM	Visual VM Indication at iPhone Client, Downloading and Playing VM Using Unity Connection	To verify that iPhone client can receive Visual VM indication and can dial into Unity Connection VM server and check VM, and the DTMF is set for kpml.	iPhone end point (Unified CM(Unity Connection	Passed	
UC802IF.CCM.206	Unified CM	Visual VM Indication at iPhone client and Downloading and Playing VM Messages When Unity Connection VM Servers Active-Active	To verify iPhone clients get VM alerts and see visual VM indication and it can download and play the VM messages.	iPhone end point (Unified CM(Unity Connection	Passed	
UC802IF.CCM.207	Unified CM	iPhone Client Downloading and Playing VPIM Forwarded and Replied VM Messages When VM Server is Cisco Unity Connection	To verify iPhone clients get VM alerts for VPIM forwarded and replied voicemails and see visual VM indication and it can download and play these VM messages.	iPhone end point (Unified CM(Unity Connection	Passed	
UC802IF.CCM.208	Unified CM	iPhone Client Receiving Visual VM Notification and Downloading and Playing of Forwarded VM Messages	To verify iPhone client gets voicemail alert and can see visual VM indication and it can download and play VM messages from Unity VM servers in active-standby configuration and when active is down.	iPhone client end point(Unified CM(Unity	Passed	
UC802IF.CCM.209	Unified CM	Layer 2/3 Roaming When iPhone Client is in a Call	To verify iPhone client in a call can roam from one AccessPoint to another seamlessly and maintain the call.	iPhone client endpoint (Unified CM(Wireless	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CCM.211	Unified CM	Call Pick Up for iPhone Client	To verify if iPhone client and Unified IP Phones 9971, 9951, and 8961 are in same call pickup group. Verify that Unified IP Phones 9971, 9951, and 8961 can pick the call for iPhone client and vice versa (iPhone Client can pick up the call for Unified IP Phones 9971, 9951, and 8961 if supported).	iPhone Client endpoint (Unified CM	Passed	
UC802IF.CCM.300	Unified CM	BLF Speed Dial for Unified IP Phone 9971 With Three CKEM Expansion Modules	To verify and monitor the presence status of 108 BLF-Speed dials for three Unified IP Phone 9971 CKEM expansion modules.		Passed	
UC802IF.CCM.301	Cisco Unity Connection	eMWI on Shared Lines With Unity Connection-Unified CM SIP Integration	Verify that MWI count is seen on Unified IP Phone (8900 and 9900) series when a new voicemail is left for the subscriber in Unity Connection integrated to Unified CM using SIP.	IP Phone->Unified CM->SIP->Unity Connection	Passed	
UC802IF.CCM.302	Cisco Unity Connection	eMWI on Shared Lines With Unity Connection-Unified CM SCCP Integration	Verify that MWI count is seen on Unified IP Phone (8900 and 9900) series when a new voicemail is left for the subscriber in Unity Connection integrated to Unified CM using SIP.	IP Phone->Unified CM->SIP->Unity Connection	Passed	
UC802IF.CCM.303	Cisco Unity Connection	eMWI for Extension Mobility Across Cluster	Verify that eMWI works for Extension Mobility across cluster.	Unity Connection->Unified CM->SIPT->Unified CM->IP Phone 9971/9951/8961 (Extension Mobility)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CCM.304	Cisco Unity Connection	Support of eMWI Over Inter Cluster SIP Trunks	Verify that eMWI works over inter cluster SIP trunks.	Unity Connection (MWI)->Unified CM->SIPT->Unified CM->IP Phone 9971/9951/8961	Passed	
UC802IF.CCM.305	Cisco Unity Connection	Support of eMWI During Failover and Bulk Re-Synchronization in Unity Connection	Verify that eMWI works when primary VM server is down and the secondary server is the acting primary.		Passed	
UC802IF.CCM.500.1	Unified CM	Unified CM CoW Deployment Model	To verify and validate the CoW Deployment Model.	Unified CM->ASA->WAN->ASA->Unified CM	Passed	
UC802IF.CCM.501	Unified CM	Validate ASA TLS Proxy Feature	To verify and validate non-secure, encrypted and authenticate phone registration to Unified CM via ASA TLS Proxy.	Phone->ASA TLS Proxy->Unified CM	Passed w/ Exception	CSCtc06130
UC802IF.CCM.520	Unified IP Phone	Unified IP Phone 6911 Joining Unified MeetingPlace Meeting	To verify that Unified IP Phone 6911 can join a Unified MeetingPlace meeting.	Phone->Unified CM -->Unified MeetingPlace	Passed	
UC802IF.CCM.521	Unified IP Phone	Unified Personal Communicator Working in Phone Associated Mode to IP Phone 6911	To verify that Unified Personal Communicator can work in phone associated mode to IP Phone 6911.	Phone->Unified CM -->Unified MeetingPlace	Passed	
UC802IF.CCM.522	Unified IP Phone	IP Phone 6911 Dials Emergency Number	To verify that Unified IP Phone 6911 can dial emergency number.	Phone->Unified CM -->Unified MeetingPlace	Passed	
UC802IF.CCM.523	Unified IP Phone	Support for Different Codecs Including ILBC Codec by IP Phone 6911	To verify that Unified IP Phone 6911 can support codecs, g711u, g729r8, g729br8, ILBC codecs.	Phone->Unified CM -->Unified MeetingPlace	Passed	
UC802IF.CCM.524	Unified IP Phone	IP Phone 6911 Working as Agent Phone in Unified CCX Network	To verify that Unified IP Phone 6911 can be used as an agent phone in Unified CCX.	Phone->Unified CM -->Unified CCX	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CCM.525	Unified IP Phone	IP Phone 6911 and Unified Video Advantage Interworking	To verify that Unified Ip Phone 6911 can interwork with Unified Video Advantage and use as a video phone.	Phone->Unified CM -->Unified Video Advantage	Passed	
UC802IF.CCM.526	Unified IP Phone	IP Phone 6901 Joining Unified MeetingPlace Meeting	To verify that Unified IP Phone 6901 can join Unified MeetingPlace scheduled meetings.	Phone->Unified CM -->Unified MeetingPlace	Passed	
UC802IF.CCM.527	Unified IP Phone	Support for Different Codecs Including ILBC Codec by IP Phone 6901	To verify that Unified IP Phone 6901 can support codecs, g711u, g729r8, g729br8, ILBC codecs.	Phone->Unified CM -->Unified MeetingPlace	Passed	
UC802IF.CCM.600	Unified CM Extension Mobility Cross Cluster	Extension Mobility Cross Cluster Attend Home Cluster MeetingPlace Conference, Primary Home Unified CM Failure	Verify user is able to log into a visiting cluster using Extension Mobility Cross Cluster and attend a meeting place conference hosted on the home cluster and the home cluster's primary Unified CM fails.	EMCC IP Phone->ASA->Unified CM->MeetingPlace	Passed	
UC802IF.CCM.601	Unified CM Extension Mobility Cross Cluster	Extension Mobility Cross Cluster Calls MGCP PSTN Phone Using Local Route Group, Signaling WAN Link Failure	Verify if user is able to log into a visiting cluster using Extension Mobility Cross Cluster and call an external number through an MGCP PSTN gateway configured using Local Route Group and there is a Signaling WAN link failure during the call.	EMCC IP Phone->ASA->Unified CM->MGCP PSTN->PSTN Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CCM.602	Unified CM Extension Mobility Cross Cluster	Extension Mobility Cross Cluster Ad-Hoc Video Conference With Second Incoming Call, Local WAN Failover to Unified SRST	Verify if user is able to log into a visiting cluster using Extension Mobility Cross Cluster and initiate a video ad-hoc conference on a Unified IP Phone 9900 series in the Home cluster, a Unified IP 7985 phone in a remote Unified CME site, and a Cisco UC Integration for Microsoft Office Communicator client in the visiting cluster. A second incoming call also occurs during the conference. Local WAN failure causes endpoints to failover to Unified SRST.	EMCC IP Phone->ASA->Unified CM->ConfBridge->ICT->CME->IP Phone7985; EMCC IP Phone->ASA->Unified CM->ConfBridge ->ICT->ASA->Unified CM->UC Integration™ for Microsoft Office Communicator;	Passed	
UC802IF.CCM.603	Unified CM Extension Mobility Cross Cluster	Extension Mobility Cross Cluster Video ICT Call, Gatekeeper Based CAC Reverts the Call to Audio-Only	Verify if user is able to log into a visiting cluster using Extension Mobility Cross Cluster and attempt to Pickup an incoming video call in the user's Pickup Group. Call is reverted to audio-only due to Gatekeeper Based CAC bandwidth restrictions.	EMCC IP Phone->ASA->Unified CM->GateKeeper ICT->CME->IP Phone7985;	Passed	
UC802IF.CCM.604	Unified Computing System	Unified Computing System Blade Failover and Recover	Verify the applications running on a Unified Computing System Blade are able to properly failover and recover after the blade has failed.	NA	Passed	
UC802IF.CCM.605	Unified CM	TCP Connection Reuse With Unified CM SIP Trunk	Verify that TCP connection reuse behavior does not cause any adverse effect and calls succeed over SIP trunks.	Phn1 -->Unified CM1 -->SIPT -->Unified CM2 -->Unified CM2 -->Phn2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CCM.606	Unified CM	SIP Trunks With CODEC G729/G729a and MTP Required	Verify that calls over SIP trunks with MTP required are successful when the codec preference on the SIP trunk is G.729/G.729a.	Phn1 -->Unified CM1 -->SIPT (MTP G729 G729a) -->Unified CM2 -->Unified CM2 -->Phn2	Passed	
UC802IF.CCM.607	Unified Computing System	Unified Computing System Blade Reboot	Verify the applications running on a Unified Computing System Blade are able to properly recover after the Unified Computing System blade is rebooted.	NA	Passed	
UC802IF.CCM.651	IPv6	Dual Stack SIP Trunk With Trunk Codec Set to G729	Verify that MTP is invoked when the trunk has been configured for early offer. And to verify that calls works over DS SIP Trunks with codec preference set to G729/G729a.	SCCP (v6/v4) -->Unified CM -->SIPT (v6/v4) (v6 media/v4 sig pref) (ANAT on) (MTP) -->Unified CM -->IP Phone	Passed	
UC802IF.CCM.700	Unified CM	Clock in Various Components Fallback to Standard Time in Fall 2009	Verify that the clocks in various components fallback to Standard time in Fall 2009.		Passed	
UC802IF.CCM.701	Unified CM	Clock in Various Components Spring Forward to North American Daylight Savings Time	Verify if the Clocks in various components spring forward to North American Daylight Savings Time.		Passed	
UC802IF.CCM.702	Unified CM	Daylight Savings Time Regulation Using COP File	Verify support for any update in Daylight Savings Time Regulation using COP file.		Passed	
UC802IF.CCM.809	Video	Call Transfer Through Cisco IME Trunk	To verify if an audio call can escalate to video when call is transferred from audio capable endpoint to video capable endpoint.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CUCM.801	Voice Activated Conference	Voice Activated Conference Using Adhoc Software	Verify if a user can conference IP Phones listed under different Unified CM clusters, through Unified MeetingPlace software conference bridge.	IP Phone 1->Unified CM 1->SIP Trunk->Unified CM2 -IP Phone 2 ----Conference->Unified CM 1---IP Phone 3	Passed	
UC802IF.CUCM.802	Video	Group Pick Up Video Call	To verify if a user can make a call from a <variable > to group pick number, take the call from video phone, and can establish two way video or audio.		Passed	
UC802IF.CUCM.807	Video	Video Over Cisco IME Trunk	To verify if a video call can be placed over PSTN and subsequent calls can go over Cisco IME trunk.		Passed	
UC802IF.CUCM.808	Video	Video Call Failover to PSTN When WAN Link is Clogged	To verify whether Cisco IME call failover to PSTN when WAN is clogged between Unified CM Clusters and if the video call falls back to audio.		Passed	
UC802L.CCM.001		Table Out of Sync Detection Service Parameter ON	Verify that no adverse affect of call processing occurs during DB table sync operations when Table Out of Sync Detection service parameter is enabled.		Passed	
UC802L.CCM.005		IP to IP Calls	Verify that calls from SFO-ORD IP phone to SFO-ORD IP phone are successful.		Passed	
UC802L.CCM.020		SJC/RFD Cluster Upgrade Times	Verify that when the tests upgrades with IO throttling is enabled, the system is under moderate load, no phones unregister, no code yellows occur, and the upgrade completes in a reasonable amount of time.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802L.CCM.108		IP to IP Calls	Verify that calls from AZO IP phone to AZO IP phone are successful.		Passed	
UC802L.CCM.120		AZO Cluster Upgrade Times	Verify that when the tests upgrades with io throttling is enabled, the system is under moderate load, no phones unregister, no code yellows occur, and the upgrade completes in a reasonable amount of time.		Passed	
UC802L.CCM.301		Table Out of Sync Detection Service Parameter ON	Verify that no adverse affect of call processing occurs during DB table sync operations when Table Out of Sync Detection service parameter is enabled.		Passed	
UC802L.CCM.305		IP to IP Calls	Verify that calls from SFO-ORD IP phone to SFO-ORD IP phone are successful.		Passed	
UC802L.CCM.320		DFW Cluster Upgrade Times	Verify that when the tests upgrades with io throttling is enabled, the system is under moderate load, no phones unregister, no code yellows occur, and the upgrade completes in a reasonable amount of time.		Passed	
UCS712IF.CCM.101	IPv6	Blind Transfer of IPv6 Call to Unified CCX Phone Agent	Verify the call from a remote cluster across a dual stack SIP trunk to dual stack SCCP phone. From the SCCP phone the call is blind transferred to Unified CCX agent.	SCCP (v6/v4) -->Unified CM -->SIPT (ANAT-on) (addressing mode v6/v4) (v6 sig pref/media pref) -->Unified CM -->SCCP (v6/v4) -->blind transfer -->IPCCX -->SCCP Phone (v6/v4) IP Phone Agent	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF.CCM.103	IPv6	Video Call Over SIP Trunk with ANAT Enabled	Verify the call from Unified IP Phone 7985 over a SIP trunk with ANAT enabled to another IP Phone 7985. The call finally being transferred to a Unified Personal Communicator with Unified Video Advantage.	7985 (v4) -->Unified CM -->SIPT (ANAT-on) (addressing mode v6/v4) (v6 sig pref) (v6 media pref) -->Unified CM -->7985 (v4) -->Xfer -->Unified Personal Communicator	Passed	
UCS712IF.CCM.106	IPv6	PSTN Call to Dual Stack VG224 and Call Answered by Group PickUp.	Verify if a call from PSTN to dual stack VG224 is answered by another phone using Group Pick Up.	PSTN Phone -->SIP GW (v6/v4) -->SIPT (v6 sig/media pref)-->Unified CM -->VG224 GW (v6/v4) -->FXS phone -->GPickup -->FXSPhone.	Passed	
UCS712IF.CCM.107	IPv6	Call Over a Dual Stack SIP Trunk Forwarded Over ICT to Unified CME	Verify when a call is placed from a dual stack phone to another dual stack phone where the SIP trunk has also been configured to support both IPv4 and IPv6, but the media and signaling preference has been set to IPv6. The call is forwarded on busy to Unified CME over ICT.	SCCP Phone (v4/v6) -->Unified CM -->SIPT (v4/v6) -->Unified CM -->SCCP Phone (v4/v6) -->CFB -->ICT -->CME -->SIP Phone	Passed	
UCS712IF.CCM.109	IPv6	Call Park Over SIP Trunk With Media And Signaling Preference Set to IPV6 And IPv4	To verify if a call can be placed from a dual stack phone to another dual stack phone where the SIP trunk is also dual stack and is configured to prefer IPv6 for signaling and media. The call is then parked and retrieved by a Unified IP 7925 phone.	SCCP (v6/v4) -->Unified CM -->SIPT (v4/v6) (v6 sig/media pref) -->SCCP (v6/v4) -->Call Park; SCCP (v6/v4) -->Unified CM -->SIPT (v4/v6) (v6 sig/media pref) -->7925	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF.CCM.109	IPv6	Call Park Over a SIP Trunk With Media and Signaling Preference Set to IPv6	Verify if a user can place a call from a dual stack phone to a another dual stack phone where the SIPT is also dual stack and is configured to prefer IPv6 for signaling and media. The call is then parked and retrieved by a IP 7925 phone.	SCCP (v6/v4) -->CUCM -->SIPT (v4/v6) (v6 sig/media pref) -->SCCP (v6/v4) -->Call Park; SCCP (v6/v4) -->CUCM -->SIPT (v4/v6) (v6 sig/media pref) -->7925	Passed	
UCS712IF.CCM.111	IPv6	Join Across Lines With IPv6 And IPv4 Media	To verify if a user can join calls across lines where the media on line 1 is IPv6 and media on line 2 is IPv4.	SCCP Phn1 (v6/v4) (secure) -->Unified CM -->SCCP Phn2 Line 1 (v6/v4) (secure) (v6 sig/media pref); SIP Phn3 (secure) -->Unified CM -->SIPT -->SCCP Phn2 Line 2 (v4/v6) (secure); SCCP Phn2 Line 2 -->Join across Lines -->Conf Bridge (secure)	Passed	
UCS712IF.CCM.114	IPv6	Dual Stack Phone to Cisco Unity Express AA Transferred to Another Dual Stack Phone	To verify that calls can be placed from a dual stack phone to CTI devices such as Cisco Unity Express.	SCCP (v6/v4) -->Unified CM -->CTI -->Cisco Unity Express -->AA -->Unified CM -->SCCP (v6/v4)	Passed	
UCS712IF.CCM.118	IPv6	Dual Stack SIP Trunk to TRP Enabled Endpoint	Verify that MTP is invoked when the endpoint has been configured with TRP. Also to verify that when the call is IPv6 then only one MTP is invoked for the call.	SCCP (v6/v4) -->Unified CM -->SIPT (v6/v4) (v6 media/sig pref) (ANAT on) (MTP) -->Unified CM -->MTP/TRP -->Unified Personal Communicator	Passed w/ Exception	CSCsa60566

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701EF.SMB.005	Unity Connection	Voice Mail Deposit of Another User in Unity Connection on Unified Communications Manager Business Edition Using Speech Recognition	Verify the ability to use speech recognition to deposit a voice message for another user in Unity Connection on Unified Communications Manager Business Edition.	Rem SCCP Ph 1->Unified CMBE->CFNA->Unified CMBE->Unity Connection	Passed	
UC701EF.SMB.007	Unity Connection	Using Unity Personal Communications Assistant in Unity Connection on Unified Communications Manager Business Edition to call users in a Hunt List	Verify the ability to use Unity Personal Communications Assistant in Unity Connection on Unified Communications Manager Business Edition to call other users specified in a hunt list.	Stage1:RemSCCP Ph1->Unified CMBE->SCCP Ph2->CFNA-Unified CMBE->Unity Connection Stage2:Unity Connection->Unified CMBE->SCCP Ph1/RemSCCP Ph2/7985G	Passed	
UC701EF.SMB.012	RSVP	Calling From Unified IP Video Phone When Video Reservation Is Mandatory	Verify the ability of making a call from a central Unified IP Video Phone to a remote Unified IP Video Phone 7985 when the RSVP reservation for video is mandatory.	Stage1:Vid Ph1->Unified CMBE->RSVP->RemVid Ph2 Stage2:Vid Ph1->Unified CMBE->RSVP->RemVid Ph2 Stage3:Vid Ph1->Unified CMBE->RSVP->RemVid Ph2	Passed	
UC701EF.SMB.014	RSVP	Video Call when Audio Reservation is Optional and Video if Reservation Succeeds	Verify the ability of making a call from a central Unified IP Video Phone to a SCCP Phone with Unified Video Advantage in a remote site where audio reservation is optional and video if reservation succeeds over SCCP.	Stage 1:Vid Ph 1->Unified CMBE->RSVP->RemVid Ph 2 Stage 2:Vid Ph 1->Unified CMBE->RSVP->RemVid Ph2 Stage 3:VidPh1->Unified CMBE->RSVP->Vid Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701EF.SMB.017	RSVP	Calling from Video SCCP Phone to Cisco Unified Video Telephony Advantage (CUVTA) When Audio Reservation Is Mandatory and Video If Reservation Succeeds	Verify the ability to call from a central Video SCCP Phone with Unified Video Advantage when audio reservation is mandatory and video if reservation succeeds over SCCP.	Stage 1: SCCP Ph 1(CUVTA)->Unified CMBE->RSVP->Rem SCCP Ph 2(CUVTA) Stage 2: SCCP Ph 1(CUVTA)->UCMBE->RSVP->Rem SCCP Ph 2(CUVTA)	Passed	
UC701EF.SMB.030	Unified CM Business Edition Failure	Call Transfer from Central Survivable Remote Site Telephony Phone to PSTN Phone	Verify if a user can place a call from a SCCP phone to a SIP Phone in the central site during Unified CM Business Edition Failure. Transfer the call to a PSTN phone.	SCCP Ph 1->SRST 1->SIP Ph 1->XFER->PSTN GW->PSTN Ph 1	Passed	
UC701EF.SMB.032	Unified CM Business Edition Failure	Call from Remote Site to Central Site Using WAN during Co-Res Recovery	Verify by making a call from a remote site to the Central Site during Unified CM Business Edition Failure.	Rem SCCP Ph 1->SRST 1 ->SRST 2->SCCP Ph 2	Passed	
UC701EF.SMB.033	Unified CM Business Edition Failure	PSTN Forwarded Call During Unified CM Business Edition Recovery	Verify if a call from a SCCP phone to a SIP Phone whose CFNA is set to a PSTN Phone in the central site during Unified CM Business Edition fails.	Stage 1: SIP Ph 1->CFNA->PSTN Ph 1 Stage 2: Rem SCCP Ph 1->SRST 1->SRST 2->SIP Ph 1->CFNA->PSTN GW->PSTN Ph 1	Passed	
UC701EF.SMB.039	Unified CM Feature IPMA	IP Manager Assistant (IPMA), SIP, Proxy Mode Manager in Central Site and Assistant in Remote Site	Verify if a user can configure a SIP phone on remote site for Assistant on phone proxy mode and have Manager at central site. Place a call from PSTN to the Manager Phone . The call goes to Assistant, the Assistant uses transfer to VM feature and sends the call to Manager VM box.	Stage 1:PSTN Ph 1->PSTN GW->Unified CMBE->IPMA SIP Ph 2 (Asst) Stage 2:SIP Ph 2 (Asst)->transfer VM->IPMA->Unified CMBE->Unity Connection Stage 3:SIP Ph 1->Unified CMBE->Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC702EF.SMB.004	Unity Connection	Voice Mail Retrieval from Unity Connection on Unified Communications Manager Business Edition from SIP Phone	Verify the ability to successfully retrieve a voice message in Unity Connection on Unified Communications Manager Business Edition from a SIP phone.	Stage 1: PSTN Ph 1->PSTN GW->Unified CMBE->SCCP Ph 1->XFER ->SIP Ph 1->Unified CMBE->Unity Connection Stage 2: SIP Ph 1->Unified CMBE->Unity Connection	Passed	
UC702EF.SMB.010	Unity Connection	Voice Mail Deposit in Unity Connection Using G.729 Codec From Remote Site	Verify the ability of depositing a voice message in Unity Connection using G.729 codec from a remote site.	Stage 1: Rem SCCP Ph 1->Unified CMBE->Rem SCCP Ph 2 ->CFNA ->Unified CMBE->Unity Connection Stage 2: Rem SCCP Ph 1->Unified CMBE->Unity Connection	Passed	
UC702EF.SMB.021	DND	DND Feature During Callback	Verify by making a call from a central SCCP phone which has Do Not Disturb (DND) feature activated to a remote SIP phone that is busy. The SCCP phone in the central site should have Callback activated	Stage 1: Rem SIP Ph 1->Unified CMBE->PSTN GW->PSTN Ph 1 Stage 2: SCCP Ph 1->Unified CMBE->Rem SIP Ph 1->CALLBACK	Passed	
UC702EF.SMB.025	Unified CM Feature Intercom	Hold Reversion with Extension Mobility	Verify Hold reversion with Extension Mobility. Create Extension Mobility profile that is similar to one of the SIP Phones in the central site. Use this profile to login to a remote phone.	Stage 1:PSTN Ph 1->PSTN GW->Unified CMBE->SIP Ph 1->Hold Stage 2:SIP Ph 1 ->Resume	Passed	
UC702EF.SMB.034	Unified CM Commercial Feature Intercom	Intercom Between SIP and SCCP Phone on Different Remote Sites	Verify if a user can place an intercom call between SIP phone on a remote site to a SCCP phone on another remote site while the SCCP phone on the first remote site is active on call.	Rem SIP Ph 1->Intercom->Unified CMBE->Rem SCCP Ph 1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
CO10.CME.002	Feature termination, Tandem	Hold and Resume in Unified CME SIP IP phone for Call Connected to Unified CM SCCP IP phone over H.225 Trunk	To verify hold and resume features in Unified CME IP Phones over H.225 trunks.		Passed	
CO10.CME.007	Feature termination, Tandem	IOS Hardware Transcoding for Calls from Unified CME to Unified CM Sent to Cisco Unity Express Voice Mail	Verify the ability to invoke IOS hardware transcoding for calls from Unified CME to Unified CM, and verify that calls requiring dissimilar codecs are sent to voice mail in Cisco Unity Express.		Passed	
CO10.CME.016	Feature termination, Tandem working	Call Between SIP and H.323 Site via IP-to-IP Gateway Forwarded to Unified CM via H.323/SIP ICT	Verify the ability to make a successful call from a SIP site to a H.323 site over an IP-to-IP Gateway, which is then forwarded to Cisco Unified CM via either a SIP or H.323 inter-cluster trunk (ICT).		Passed	
SR60.CME.008.6	Basic Call Flow	MeetMe Conference	To verify a Meet-Me conference using H/W conference resources within Unified CME.		Passed	
SR60.CME.107.1	Basic Call Flow	Unified CME Support for Out of Dialog	To verify the use of OOD-R from one Unified CME site to remote Unified CME sites.		Passed	
SR60.CME.108.13	Basic Call Flow	Call From a PSTN Phone Through a H.323 Gateway Forwarded to Shared Line	Verify if a call from a PSTN phone to an IP Phone registered to Unified CME can be forwarded to a remote Unified CME phone configured for shared line.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.CME.108.5	Basic Call Flow	Hold And Resume the Call Placed on a SIP Network	Verify if a call from a Gatekeeper controlled Unified CME SCCP IP Phone through a IP-to-IP gateway to a Unified CME SIP IP Phone via SIP trunk can be placed on hold and then resumed.		Passed	
UC702EF.CME.005	CME H.450	Call Forward All from QSIG PBX on Unified CME to IPMA Manager	Verify if a call from a SIP proxy controlled SIP phone via Unified CM to QSIG PBX phone on Unified CM is call forwarded all to a QSIG PBX phone connected to Unified CME which in turn has call forward busy to IPMA Manager.	SIP Ph1->CSPS->SIP Trunk->Unified CM->QSIG Trunk->PBX Ph1->CFB->QSIG Trunk->Unified CM->IPIP GW (H323)->CME->QSIG Trunk->PBX Ph1->CFA->QSIG Trunk-> CME->IPIP GW (H323)->Unified CM->IPMA	Passed	
UC702EF.CME.015	Conference	Ad-Hoc Conference on Unified CME with DPNSS PBX Phone and IPMA Manager Phone	Verify the Ad-Hoc Conference Setup by Local Unified CME Phone with DPNSS PBX Phone and IPMA Manager Phone.	1: SCCP Ph1 -->CME -->IPIP GW (H323)->Unified CM->QSIG Trunk->Westell GW->DPNSS PBX Ph1 2: SCCP Ph1 --> CME->CNF->CME->IPIP GW (H323) -->Unified CM -->IPMA Manager 3: SCCP Ph1->CME->CNF->DPNSS PBX Ph1 & IPMA Manager phone	Passed	
UC702EF.CME.028	Unified CME	Meet Me Conference between QSIG PBX Phone on Unified CME and DPNSS PBX Phone	Verify the Meet Me conference feature on Unified CME between 2 Unified CME phones (SCCP and QSIG PBX phone connected to Unified CME) and a DPNSS PBX phone.	1 : SCCP Ph1->CME->CNF_MM 2: PBX ph1->QSIG Trunk->CME->CNF_MM 3: DPNSS PBX Ph1->Westell->QSIG Trunk->Unified CM->IPIPGW (H323)->CME->CNF_MM	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC702EF.CME.078	Transfer to Voice Mail	Transferring Calls from PSTN Phone to Unified CME Phone Voicemail	Verify when a call from PSTN phone (Phone A) to Unified CME SCCP phone (Phone B) is forwarded on no answer to another Unified CME SCCP phone (Phone C), and if the Unified CME SCCP phone (Phone C) can transfer the call to voice mail box of Unified CME phone (Phone B) using TrnsfVM soft key.	1: PSTN Ph1->MGCP BRI->Unified CM->GateKeeper->IP-IP GW->GateKeeper->CME->SCCP Ph1->CFNA->CME->SCCP Ph2 2: SCCP Ph2->TrnsfVM->SCCP Ph1 DN# ->CME->NM CUE 3: SCCP Ph1->CME->NM CUE	Passed	
UC702EF.CME.156	Shared Line cBarge and Privacy	Shared line cBarge and Privacy support on Unified CME with Consult Transfer to Unified CM Video Phone	Verify if a call from QSIG PBX phone (Phone C) to SCCP Unified CME shared line phone (Phone A) is barged from another Unified CME shared line SCCP phone (Phone B) and if the call is consult transferred to a Unified CM video phone (Phone D) from Unified CME phone (Phone A).	1: PBX Ph1->QSIG Trunk->Unified CM->GateKeeper->IP-IP GW->GateKeeper->CME->SCCP Phone1(SL) 2: SCCP Ph2 (SL)->CME->cBrg->SCCP Ph1 (SL)->XFER_C->CME->GateKeeper->IP-IP GW->GateKeeper->Unified CM->SCCP Video Ph1	Passed	
UC713IF.CME.001	Unified CME	Parallel/Serial Hunt with Transcoder Invoked on Unified CME for Codec Mismatch	Verifies whether Transcoder is invoked on Unified CME for codec mismatch on IP Phone 6921/6941/6961	IP Phone 6921/6941/6961->Unified Communications Manager->Unified Border Element->Unified CME->Hunt Pilot ---Parallel Hunt group -- (xcoder Invoked)->IP Phone 6921/6941/6961	Passed	
UC802IF.CME.103	Cisco VTIII Support	Unified CME Interop With Cisco VTIII Camera	Verifies if Cisco VTIII camera behind Ip Communicator registered to Unified CME calls Cisco VTIII endpoint.	IP Communicator/Unified Video Advantage---- CME --- H323 --- Unified CM -- IP Communicator/Unified Video Advantage	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CME.901	Unified CME	Unified CME Stream Multicast MOH	Verify whether Unified CME can stream multicast MOH when SNR phone is put on hold.	IP Phone 6921/6941/6961 --- Unified CM ---- SIP T -----Unified Border Element ----- SIPT ---- CME --- SNR -Hold---- MMOH --- Unified CM IP Phone	Passed	
UC802IF.CME.902	Video Over SIP Trunk	Video Over SIP Trunk Using Endpoints	Verify the video over SIP trunk using Unified IP Phones 8900, 9900, and 6900 series.	IP Phone 6921/6941/6961 -CME 1--- SIP T - CME 2 ---IP Phone 9971/9951/8961 ----Conference ---Unified CM 1---IP Phone 3	Passed	
UC802IF.CME.903	Video	Video Call Resumed During Hold/Resume	Verify whether video call is resumed during hold/resume and transfer over SIP trunk.	IP Phone 6921/6941/6961 -CME 1--- SIP T - CME 2 ---IP Phone 9971/9951/8961 ----Transfer->Unified CM 1---IP Phone 3	Passed	
UC802L.CME.002		IP to ICT to Communications Manager Express to IP Calls	Verify that calls from SFO-ORD IP phone to Unified CME IP phone over Inter-Cluster Trunk are successful.		Passed	
UC802L.CME.302		IP to ICT to Communications Manager Express to IP Calls	Verify that calls from SFO-ORD IP phone to Unified CME IP phone over Inter-Cluster Trunk are successful.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CIP.001	Unified IP Phones 9951/9971	Unified IP Phones 9951/9971 with Video Support Interworking with Unified PC and IP Phones 7985	Verify the call from Unified IP Phones 9951/9971 with Secace USB Video Camera attachment to Unified Personal Communicator (video enabled) over QSIG ICT and then transfer the call to UC Integration for Microsoft Office Communicator/IP Phone 7985 attached with Unified Video Advantage in the same cluster. Verify the video-on-hold by pressing the hold button on Unified IP 99xx/89xx Phones.	Unified IP 99xx/89xx SIP Ph1 with USB Video camera->Unified CM1->QSIG ICT->Unified CM2->CUPC(video enabled)->XFER_C->Unified CM2->UC Integration™ for Microsoft Office Communicator/7985	Passed	
UC802EF.CIP.002	Unified IP Phones 9951/9971	Unified IP Phones 9951/9971 Interworking with IP Communicator and Unified Video Advantage	Verify the call from the SIP IP Communicator with Unified Video Advantage attached to the remote SCCP Phone which has CFNA set to a remote Unified IP Phones 9951/9971.	SIP CIPC1(with CUVA) ->Unified CM1->QSIG ICT->Unified CM2 -> Rem SCCP Ph1; Rem SCCP Ph1->CFNA->Rem RT Std+ SIP Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CIP.003	Unified IP Phones 9951/9971	Unified IP Phones 9951/9971 with Video Support Interworking with UC Integration™ for Microsoft Office Communicator	Verify the call from the Unified IP Phones 9951/9971 with Secace USB Video Camera attachment to UC Integration for Microsoft Office Communicator which is in deskphone mode to IP Phones 9951/9971 with Secace USB Video Camera attachment over SIP ICT and then conference the call to a IP Communicator with Unified Video Advantage attached. Then drop the call from UC Integration for Microsoft Office Communicator.	Unified IP 99xx/89xx SIP Ph1 with USB Video camera->Unified CM1->SIP ICT->Unified CM2->UC Integration™ for Microsoft Office Communicator (in deskphone mode to Unified IP 99xx/89xx SIP Ph1 with USB Video camera)->Conf->SCCP CIPC1 (with CUVA)	Passed	
UC802EF.CIP.004	Unified IP Phones 9951/9971	Unified IP Phones 9951/9971 with Video Support over Tandem Cluster	Verify the call from the UC Integration for Microsoft Office Communicator client and IP Phones 9951/9971 in leaf clusters over tandem and then transfer the call to the Video enabled IP Phones 9951/9971 over QSIG ICT.	UC Integration™ for Microsoft Office Communicator(in deskphone mode to Video phone) ->Unified CM1->QSIG->Unified CM-Tand->QSIG->Unified CM2->Unified IP 99xx/89xx SIP Ph1 with USB Video camera->XFER_C->Unified CM1->QSIG ICT->Unified CM2->IP Video Phone 99xx enabled	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CIP.005	Unified IP Phones 9951/9971	Unified IP Phones 9951/9971 with Video support Interworking with TNP Phones with Unified Video Advantage Attached	Verify the call from the TNP Phone attached with the Unified Video Advantage to the Unified Personal Communicator client(Video enabled) over SIP ICT and then conference the call to the IP Phones 9951/9971 with Secace USB Video Camera attachment over QSIG ICT.	TNP SCCP Ph1(CUVA) ->Unified CM1->SIP ICT->Unified CM2->CUPC(video enabled)->CFB->Unified CM3->QSIG ICT->99xx/89xx SIP Ph1 with USB Video camera	Passed	
UC802EF.CIP.006	Unified IP Phones 9951/9971	Unified IP Phones 9971/9951/8961 in Unified SRST	Verify the PSTN call from the remote Unified IP Phones 9971/9951/8961 in Unified SRST mode to another remote Unified IP Phones 9971/9951/8961 in Unified SRST mode which has CFWDALL set to a PSTN Phone.	Rem 99xx/89xx SIP Ph1->SRST1->PSTN GW->PSTN->PSTN GW->SRST 2->Rem 99xx/89xx SIP Ph2;Rem 99xx/89xx SIP Ph2->CFWDALL->PSTN GW->PSTN Ph1	Passed	
UC802EF.CIP.007	Unified IP Phones 9951/9971	Plus Dialing in Unified IP Phones 9971/9951/8961 in Unified SRST	To verify the following: Registered Unified IP Phones 9971/9951/8961 with + sign along with the Directory number before bringing the WAN interface down. Make a PSTN call from the remote IP Phones 9971/9951/8961 in SRST mode to another IP Phones 9971/9951/8961 with + sign in another remote which then transfers the call to PSTN Phone through the PSTN gateway.	Rem 99xx/89xx SIP Ph1->SRST1/PSTN GW->PSTN->SRST2 /PSTN GW->PSTN->Rem 99xx/89xx SIP Ph2->XFER_C->PSTN GW->PSTN Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CIP.008	Unified IP Phones 9951/9971	Reply to Voice mail with Call or IM or Voicemail	Verify if Unified Personal Communicator can send a voice mail to IP Phones 9971/9951/8961 and the phone replies through calling or through voicemail.	Stage 1: CUPC1->Unified CM->99xx/89xx SIP Ph 1->CFNA->Voice Mail; Stage 2:99xx/89xx SIP Ph 1->Unified CM-> CUPC ;	Passed	
UC802EF.CIP.009	Unified IP Phones 9951/9971	Inline Playback of Visual Voice Mail	Unified Personal Communicator deposits a voicemail to SIP IP Phones 9971/9951/8961. Toast pop up of Visual voice mail indication is received and Playback of Visual voice mail is done with fast forward and rewind options.	CUPC->Unified CM-> Unified IP 99xx/89xx SIP Ph1 ->Voice mail	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701IF.MP.104	Video Endpoints	Out-dial Using Web to SCCP and SIP with Third Party Video Endpoints	To verify Out-dial using Web to SCCP and SIP with Third party video endpoints within and across cluster using SIP Trunk DTMF preference with RFC2833, OOB & RFC2833 with transport type TCP on security profile.		Passed	
UC701IF.MP.110	Audio and Video Codec Using Different Endpoints	MeetingPlace Endpoints Joining G.729 and iLBC Codec	To verify Unified MeetingPlace endpoints joining with G.729 and iLBC codec from within and across cluster environment and using H.264 with High quality video preference within cluster and across cluster ICT trunk.		Passed	
UC701IF.MP.113	Unified MeetingPlace	Schedule Continuous Meeting for Web, Audio and Video	To verify if a user can schedule continuous meeting for web, audio and video and initiate out-dial but CM Sub1 server experience network outage after 1st out-dial succeeded.		Passed	
UC701IF.MP.122	Unified MeetingPlace	Conference With 22 or More participants For Audio, Web And Video	To verify if a conference with 22 or more participants for audio, web and video is successful.	Video endpoint->Unified CM->SIP trunk->Unified MeetingPlace	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.MP.100.1	Unified MeetingPlace	Establish And Test Various Web Conference Features	Verify if a user can establish a Unified MeetingPlace conference with WebEx as the web conference provider and test the various meeting features such as joining the conference, network based recording, window/desktop sharing, changing roles and ending the meeting from various browser types.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	
UC802IF.MP.100.2	Unified MeetingPlace	Establish And Utilize Web Conference With Internal And External Participants	Verify if a user can establish a meeting place conference with WebEx as the web conference provider. Test various meeting features such as joining the conference, network based recording, window/desktop sharing, changing roles and ending the meeting from various browser types.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	
UC802IF.MP.101.1	Unified MeetingPlace	Establish And Test Web And Audio Conferences Through Different DTMF Modes	Verify is a user can establish meeting place conferences with WebEx as the web conference provider and dial in utilizing different DTMF modes.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	
UC802IF.MP.102.1	Unified MeetingPlace	Test Video Endpoints With Hardware Media Server	Verify if a user can establish MeetingPlace conferences with WebEx as the web conference provider utilizing a hardware media server. Join the conferences using multiple endpoint models configured to either a standard or high video rate and different video codecs.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.MP.103.1	Unified MeetingPlace	Test Video Endpoints With Software Media Server	Verify if a user can establish MeetingPlace conferences with WebEx as the web conference provider utilizing a software media server. Join the conference using multiple endpoint models configured to the available video modes and different video codecs.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	
UC802IF.MP.104	Unified MeetingPlace	Unified MeetingPlace Upgrade Support Existing Recordings	Verify if a user can establish and record multiple MeetingPlace meetings where Unified MeetingPlace is the web conference provider. Upgrade Unified MeetingPlace 7.0MR1 to Unified MeetingPlace 8.0 and ensure that existing recordings are still accessible and playable.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	
UC802IF.MP.105	Unified MeetingPlace	AppServer Failover With WebEx Node Recovery on Hardware Media Server System	Verify is a user can perform Unified MeetingPlace Application Server failover while internal meetings are in progress on a Hardware Media Server based system. The WebEx node will try to connect to the active WebEx TSP connection on the backup Application Server. Once the backup server is up, users will be prompted to enter the phone number for dialout. The same meeting ID will be utilized.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.MP.106.1	Unified MeetingPlace	Test Endpoints Audio CODECS With Software Media Server	Verify if a user can establish meeting place conferences with WebEx as the web conference provider utilizing a software media server. Join the conference using multiple endpoint models configured to different supported audio CODECS.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	
UC802IF.MP.107.1	Unified MeetingPlace	Test Endpoints Audio CODECS With Hardware Media Server	Verify is a user can establish a meeting place conference with WebEx as the web conference provider utilizing a hardware media server. Join the conference using multiple endpoint models configured to the different supported audio codecs.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	
UC802IF.MP.108.1	Unified MeetingPlace	Create And Utilize a Web Conference Through Microsoft Outlook	To verify if a user can create and establish a meeting Place conference with WebEx as the web conference provider. The meeting should be created with Mcrosoft Outlook and attended by clicking the meeting link within the corresponding meeting email. Send the email to users whose email format is set to SMTP and Exchange, HTML and plain text.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.MP.109	Unified MeetingPlace	Application Server Failover With WebEx Node Recovery on SMS System	To verify if Unified MeetingPlace Application Server can failover while internal meetings are in progress on a Software Media Server based system. The WebEx node will try to connect to the active WebEx TSP connection on the backup Application Server. Once the backup server is up, users will be prompted to enter the phone number for dialout. The same meeting ID will be used.		Passed	
UC802IF.MP.803	Video Conference	Video Conference Through Unified MeetingPlace	Verify is a user can establish an adhoc video conference, activate voice call, and schedule conference.	IP Phone 1 / IP Phone 2 / IP Phone 3->Unified CM 1->SIP Trunk->Unified MeetingPlace	Passed	
UC802IF.MP.804	Video	Transcoding Using Unified MeetingPlace Hardware Conference	Verify if transcoding between two Unified MeetingPlace participants having different video definitions 1.1 and 1.2 respectively is successful.		Passed	
UC802IF.MP.810	Video	Schedule Adhoc Conference With Various Video Definition Endpoints	To verify the conference with H.263, H.264 QCIF, CIF, 4CIF endpoints using Software Media Server.		Passed	
UC802IF.MP.811	Video	Adhoc Video Conference With Third Party Video Endpoints	To verify whether Third Party video endpoints (Sony, Tandberg , Polycom VSX) can attend a meeting place meeting hosted by Unified MeetingPlace application.		Passed	
UC802L.MP.101		IP to MeetingCalls	Verify that calls from IP phone to Unified MeetingPlace are successful.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR61.UMG.100.1	Unified Messaging Gateway	Unified Messaging Gateway Redundancy and Resiliency	Verify the ability to demonstrate Cisco Unified Messaging Gateway resilience to failures.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713IF.CUM.001	Unified Mobility	Secure VM Download When VM Servers and Unity Connection in Active-Active Mode	To verify if an iPhone Unified Mobile Communicator client can download the Voicemail from paired Unity Connection server when the Voicemail servers and Unity Connection is in Active-Active mode.	iPhone-(Unified Mobility--(Unity Connection	Passed	
UC713IF.CUM.001	Unified Mobility	Secure VM Download When VM Servers are Unity Connection in Active-Active Mode	Verify iPhone Unified Mobile Communicator client can download the Voicemail from paired Unity Connection Server when the Voicemail servers are Unity Connection in Active-Active mode.	iPhone->Unified Mobility Advantage->Unity Connection	Failed	CSCtd35094
UC713IF.CUM.002	Unified Mobility	Secure VM Download When VM Servers and Unity Connection in Digital Networking Mode	To verify if an iPhone Unified Mobile Communicator client can download Voicemail when the Voicemail servers are Unity Connection in Digital networking mode.	iPhone-(Unified Mobility--(Unity Connection	Passed	
UC713IF.CUM.002	Unified Mobility	Secure VM download when VM servers are Unity Connection in Digital Networking mode	Verify iPhone Unified Mobile Communicator client can download the Voicemail when the Voicemail servers are Unity Connection in Digital networking mode.	iPhone->Unified Mobility Advantage->Unity Connection	Failed	CSCtd35094
UC713IF.CUM.003	Unified Mobility	Voicemail Download Onto Unified Mobile Communicator in Unified Messaging Environment	To verify if an iPhone Unified Mobile Communicator client can download the Voicemail from Cisco Unity where Cisco Unity is in Unified Messaging environment.	iPhone-(Unified Mobility--(Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713IF.CUM.003	Unified Mobility	Voicemail Download onto Unified Mobile Communicator in a Unified Messaging Environment	Verify iPhone Unified Mobile Communicator client can download the Voicemail from Unity where Unity is in Unified Messaging environment.	iPhone->Unified Mobility Advantage->Unity Connection	Passed	
UC713IF.CUM.004	Unified Mobility	Secure VM Download When VM Servers are Cisco Unity in Digital Networking Mode	To verify if an iPhone Unified Mobile Communicator client can download the Voicemail from both Cisco Unity servers when 2 Voicemail servers are Cisco Unity in Digital Networking mode.	iPhone-(Unified Mobility-(Unity	Passed	
UC713IF.CUM.007	Unified Mobility	Unity Connection in Active-Active Configuration	To verify if an iPhone Unified Mobile Communicator client can download a reply Voicemail from Unity Connection in active-active mode with one of them down.	iPhone-(Unified Mobility--(CUV	Passed	
UC713IF.CUM.007	Unified Mobility	Unity Connection in Active-Active Configuration Downloading of Replied VM	Verify iPhone Unified Mobile Communicator client can download a reply Voicemail from Unity Connection in active-active mode with one of them down.	iPhone->Unified Mobility Advantage->Unity Connection	Failed	CSCtd35094
UC713IF.CUM.008	Unified Mobility	Directory Search and DVO-F to Intercluster Destination	To verify if an iPhone Unified Mobile Communicator client can reach an enterprise DN across a SIP trunk, by performing a directory search.	iPhone-(Unified Mobility---Unified CM1---<sip secure>---Unified CM2---IPPhone	Passed	
UC713IF.CUM.008	Unified Mobility	Directory Search and DVO-F to Intercluster Destination	Verify iPhone Unified Mobile Communicator client can reach an enterprise DN which is across a SIP trunk. The user is located by doing a directory search.	iPhone->Unified Mobility Advantage->Unified Communications Manager1-><SIP Secure>->Unified Communications Manager2->Unified IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713IF.CUM.010	Unified Mobility	Voicemail Access When Number is Set For CFNA to VM	To verify if an iPhone Unified Mobile Communicator client can reach the voicemail of a DN when the called number is set for CFNA to VM. The desk phone of the iPhone client is also set for CFA to another destination and the user's DN is found by performing a directory search on AD2008.	iPhone-(Unified Mobility--(Unified CM-(Unity	Passed	
UC713IF.CUM.012	Unified Mobility	DVO-F, Unified Mobile Communicator Clients Joining Conference	To verify that if an iPhone user can join meetme conference by dialing the meetme conference number.	iPhone-(Unified Mobility--(Unified CM-(IPPhone	Passed	
UC713IF.CUM.015	Unified Mobility	Callback on Unified MeetingPlace Only Meeting	To verify that Unified Mobile Communicator gets the meeting list for all types of meetings and call back works for Unified MeetingPlace only meetings.	iPhone-(Unified Mobility--(Unified CM-(Unified MeetingPlace+Exchange	Passed	
UC713IF.CUM.015	Unified Mobility	Unified MeetingPlace only Meeting, Meeting List and Unified MeetingPlace Call Back	Verify that Unified Mobile Communicator gets the meeting list for all types of meetings and call back work for a Unified MeetingPlace only meeting.	iPhone->Unified Mobility Advantage->Unified Communications Manager->Unified MeetingPlace+Exchange	Passed	
UC713IF.CUM.017	Unified Mobility	Unified MeetingPlace and WebEx Hybrid Meeting	To verify that Unified Mobile Communicator gets the meeting list for all types of meetings and WebEx client can be launched, where the WebEx client assists in joining the meeting.	iPhone-(Unified Mobility--(Unified CM-(Unified MeetingPlace+Exchange	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713IF.CUM.017	Unified Mobility	Unified MeetingPlace and WebEx Hybrid Meeting, Meeting List and Call Back with Launch of WebEx Meeting Client	Verify that Unified Mobile Communicator gets the meeting list for all types of meetings and WebEx client can be launched. The WebEx client assists in joining the meeting.	iPhone->Unified Mobility Advantage->Unified Communications Manager->Unified MeetingPlace+Exchange	Failed	CSCtd61787
UC713IF.CUM.021	Unified Mobility	Fail and Failback of Active Unified CM Server	To verify that iPhone Unified Mobile Communicator client gets call logs and DVO works when active Unified CM is down. Also after failback of Unified CM, Unified Mobility reinstate the communication to active Unified CM.	iPhone-(Unified Mobility--(Unified CM-(	Passed	
UC713IF.CUM.021	Unified Mobility	Fail and Failback Active Unified CM server	Verify that iPhone Unified Mobile Communicator client keeps getting call logs and DVO works when the active Unified CM is down. Also after the failed Unified CM is back, Unified Mobility Advantage reinstate the communication to active Unified CM.	iPhone->Unified Mobility Advantage->Unified Communications Manager	Failed	CSCtd26931
UC713IF.CUM.022	Unified Mobility	VM Download When Voicemail is Deposited Using VPIM	To verify if an iPhone Unified Mobile Communicator client can download the Voicemail from both Unity Connection servers when 2 Voicemail servers are in Digital Networking mode.	iPhone-(Unified Mobility--(Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713IF.CUM.022	Unified Mobility	VM Download When Voicemail is Deposited Using VPIM	Verify iPhone Unified Mobile Communicator client can download the Voicemail from both Unity Connection servers when 2 Voicemail servers are in Digital Networking mode.	iPhone->Unified Mobility Advantage->Unity Connection	Failed	CSCtd35094
UC802IF.CUM.001	Unified Mobility	Transfer of Calls Between Proxy Deskphone, iPhone And Deskphone	To verify if a user can start a meeting from his phone proxy IP Phone, transfer the call to his iPhone using Cisco Unified Mobility feature while on commute and then again transfer the call to his desk Unified IP Phone by dialing *74 Move feature.	IPPhone->ASA Phone Proxy->FireWall ASA->Unified CM->Unified Mobility->DMZ ASA->iPhone	Passed	
UC802IF.CUM.002	Unified Mobility	Mobile Call From Proxy IP Phone Gets transferred to Another Desk IP Phone	To verify if user A can call user B on his mobile phone and during conversation User A can put the call on hold while user B reaches office and transfers the call to his desk Unified IP Phone by dialing *74 Move feature and then resume the call.	IPPhone->ASA Phone Proxy->Firewall ASA->Unified CM->Unified Mobility->DMZ ASA->Mobile Phone	Passed	
UC802IF.CUM.003	Unified Mobility	An iPhone User in Meeting Transfers Call to Desk IP Phone	Verify if a user can receive and attend a Unified MeetingPlace meeting on his iPhone and later transfer the call to his desk IP Phone by dialing *74.	iPhone->DMZ ASA->Firewall ASA->Unified Mobility->Unified CM->Firewall ASA->IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CUM.008	Unified Mobility	Working of VoiceMail When One of the Active-Active Unity Connection Server is Down	Verify that Unified Mobile communicator clients can continue to get Voicemails when one of the Unity Connection servers is not reachable when they are in active-active configuration mode.	iPhone1->DMZ ASA->Firewall ASA->Unified Mobility->UnityCxn	Passed	
UC802IF.CUM.010	Unified Mobility	Directory Search and DVO to Intercluster Destination From Symbian Client	To verify that Symbian Unified Mobile Communicator client can reach an enterprise DN across a SIP trunk through a directory search.	iPhone-(Unified Mobility---Unified CM1---<sip secure>---Unified CM2---IPPhone	Passed	
UC802IF.CUM.011	Unified Mobility	Call Logs For Missed, Placed and Received Calls From Symbian Client	To verify that Symbian Unified Mobile Communicator client can see the call logs for Missed, Placed and Received calls and are updated correctly.	Symbian->Unified Mobility-->Unified CM	Passed	
UC802IF.CUM.012	Unified Mobility	Deactivate And Activate Users, Delete And Add Phones, Swap SIM Cards	To verify if Unified Mobility can demonstrate resiliency when working users are deactivated and activated, phones are deleted and added, and swapping of SIMs between different phones like WM to Symbian and vice versa.	Unified Mobility	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
CO10.UCA.042	Unified Personal Communicator	Unified Personal Communicator in Softphone Mode, Merge an Incoming Video Call	Verify the following: Set up a video call from video IP Phone 7985 to Unified Personal Communicator in softphone mode. Set up an out going call from same Unified Personal Communicator to an IP communicator. Check for bi-directional voice path. Merge both the calls.		Passed	
CO10.UCA.043	Unified Personal Communicator	Unified Personal Communicator in Softphone Mode Merge an Outgoing Video Call to SCCP Video Endpoint	Verify the following: Bring up Unified Personal Communicator in softphone mode. Establish an outgoing video call to SCCP end point. Check for audio and video of the call. Set up an incoming video call from a H.323 video end point through ICT, and answer the call. Verify that the first call is fed with Music On Hold and no Video. Click on the Merge button and establish a 3 way conference call. Check for audio and video of this conference call.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
OXN51.IPP.006	Unified Personal Communicator	Verify Five Way Web Collaboration from Unified Personal Communicator on Macintosh	To verify Unified Personal Communicator running on a MAC initiates a 5 way conference call involving Unified Personal Communicator running on a windows platform and another MAC-Unified Personal Communicator end point, all in same cluster. Web Collaboration is initiated afterwards.		Passed	
UC700IF.UCA.010.1	Unified Personal Communicator	Listen to Voicemail Under Secure Connection to Voicemail Servers	To verify that Unified Personal Communicator (Unified PC) can talk to Cisco Unity and Unity Connection Servers which are configured in the test bed in different ways using IMAP and download the Voicemail and listen to it.		Passed	
UC700IF.UCA.016.1	Unified Mobility Advantage, Unified Personal Communicator, Unified Presence	Inter Cluster DND	Verify that Unified Personal Communicator client set DND status and all presence enabled endpoints should able to see it's DND status.		Passed	
UC802IF.EXC.001	Unified PC 7.x, Unified PC 8.0, Unified presence 8.0	Basic Presence/IM With Other Unified Presence Clients	Verify if the Unified Personal Communicator 8.0 client can search an LDAP database for contacts, add these contacts to their roster, and send P2P IMs with Unified Personal Communicator 7.x and third party XMPP clients on the same cluster and on different clusters.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.EXC.002	Unified PC 8.0	Unified Presence Adhoc Chatroom	Verify if the Unified Personal Communicator 8.0 client can initiate an adhoc text conferencing session with other Unified Personal Communicator 8.0/XMPP clients on an inter-cluster setup. Verify that the option to initiate a chat session with a Unified Personal Communicator 7.x user is not available.		Passed	
UC802IF.EXC.003	Unified PC 8.0	Unified Presence Persistent Chat Room	Verify if the Unified Personal Communicator 8.0 client can create a persistent chatroom, and verify that the client properly re-displays the persistent instant messages when it leaves and rejoins the room.		Passed w/ Exception	CSCtg66391
UC802IF.EXC.004	Unified PC 8.0	Unity: Visual Voicemail Capabilities for Nonsecure Messages	Verify for a Cisco Unity 8.0 subscriber that message notifications are received when a new nonsecure voicemail arrives, that the user can retrieve and listen to the voicemail, perform operations on the message, and then call the user back who left the voicemail.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.EXC.005	Unified PC 8.0	Unity: Visual Voicemail Capabilities for Secure Messages	Verify for a Cisco Unity 8.0 subscriber that message notifications are received when a new secure voicemail arrives, that the user can retrieve and listen to the voicemail, perform operations on the message, and then call the user back who left the voicemail.		Passed	
UC802IF.EXC.006	Unified PC 8.0	Unity Connection: Visual Voicemail Capabilities for Nonsecure Messages	Verify for a Unity Connection 8.0 subscriber that message notifications are received when a new nonsecure voicemail arrives, that the user can retrieve and listen to the voicemail, perform operations on the message, and then call the user back who left the voicemail.		Passed	
UC802IF.EXC.007	Unified PC 8.0	Unity Connection: Visual Voicemail Capabilities for Secure Messages	Verify for a Unity Connection 8.0 subscriber that message notifications are received when a new secure voicemail arrives, that the user can retrieve and listen to the voicemail, perform operations on the message, and then call the user back who left the voicemail.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.EXC.008	MeetingPlace, Unified CM, Unified PC 8.0, Unified Presence 8.0	Soft Phone Escalate Incoming Call to Video, Merge with Outgoing Video Call	Verify if a user can set up an audio call from a Tandberg H.323 video endpoint in another cluster via SIP trunk to Unified Personal Communicator 8.0 in soft phone mode. Escalate the call to video. Set up an outgoing video call from the same Unified Personal Communicator 8.0 client to SIP IP Phone 7985. Check for bi-directional voice path. Merge both the calls.		Failed	CSCtg71142
UC802IF.EXC.009	MeetingPlace, Unified CM, Unified PC 8.0, Unified Presence 8.0	Soft Phone Escalate Incoming Call to Video, Merge with Outgoing Video Call	Verify if a user can set up an audio call from a SIP IP Phone 9900 series video endpoint in another cluster via SIP trunk to Unified Personal Communicator 8.0 in soft phone mode. Escalate the call to video. Set up an outgoing video call from the same Unified Personal Communicator 8.0 client to SCCP (Unified Video Advantage) endpoint. Check for bi-directional voice path. Merge both the calls.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.EXC.010	MeetingPlace, Unified CM, Unified PC8.0, Unified Presence 8.0	Soft Phone Escalate Incoming Call to Video, Merge with Incoming Video Call	Verify if a user can set up an audio call from a SIP IP Phone 9900 series video endpoint in another cluster via H.323 trunk to Unified Personal Communicator 8.0 in soft phone mode. Escalate the call to video. Set up an incoming video call from another SIP endpoint (IP Phone 7985) to the same Unified Personal Communicator 8.0 client. Check for bi-directional voice path. Merge both the calls.		Passed	
UC802IF.EXC.011	MeetingPlace, Unified CM, Unified PC 8.0, Unified Presence 8.0	Soft Phone Escalate Incoming Call to Video, Merge with Incoming Video Call	Verify if a user can set up an audio call from a SCCP (Sony) video endpoint in another cluster via SIP trunk to Unified Personal Communicator 8.0 in soft phone mode. Escalate the call to video. Set up an incoming video call from a SIP/CSF (Unified Personal Communicator 8.0) endpoint to the same Unified Personal Communicator 8.0 client. Check for bi-directional voice path. Merge both the calls.		Passed	
UC802IF.EXC.012	Unified CM, Unified PC 8.0, Unified Presence 8.0	Desk Phone Mode Device Selection with Unified IP Phones (9900, 7900, and 6900 series)	Verify if a user is configured with 3 phones: (9900, 7900, and 6900 series) . Verify Unified Personal Communicator 8.0 is able to select between the 3 devices in desk phone mode.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.EXC.013	Unified CM, Unified PC 8.0, Unified Presence 8.0	Desk Phone Mode with Unified PC 8.0 Extension Mobility in Unified SRST Site	Verify a can user log in to an extension mobility enabled phone and have Unified Personal Communicator 8.0 control this phone in desk phone mode in a Unified SRST site.		Passed	
UC802IF.EXC.014	Unified CM, Unified Mobility, Unified PC 8.0, Unified presence 8.0	Dusting with Soft Phone Mode, Hand-Off to Mobility Device	Verify a call can be handed off to a mobility device from Unified Personal Communicator 8.0 soft phone.		Passed	
UC802IF.EXC.015	MeetingPlace, Unified CM, Unified PC 8.0, Unified Presence 8.0	Escalate Inter-Cluster Group IM Session to Audio Conference, and Then Video	Verify if a user can begin an adhoc IM group chat with users from multiple clusters. Escalate the group chat to an meeting place audio conference. Escalate the audio conference to video, then de-escalate the conference.		Passed	
UC802IF.EXC.016	MeetingPlace, Unified CM, Unified PC 8.0, Unified Presence 8.0	Escalate Inter-Cluster Group IM Session to MeetingPlace Web Share	Verify if a user can begin an adhoc IM group chat with users from multiple clusters. Escalate the group chat to a Meeting Place web share.		Passed w/ Exception	CSCtf84197
UC802IF.EXC.017	MeetingPlace, Unified CM, Unified PC 8.0, Unified Presence 8.0	Ad-hoc Audio Conference (Audio-Only Bridge) Escalate to Video Attempt	verify if a user can begin an adhoc audio conference with audio-only endpoints. Have 1 video-capable endpoint join later. If the video endpoint (Unified Personal Communicator 8.0) attempts to escalate to video, verify the request is rejected and the conference continues.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.EXC.018	MeetingPlace, Unified CM, Unified PC 8.0, Unified Presence 8.0	P2P Audio Escalate to Chat, then Web Share	Verify if a user can place a call to another endpoint. Escalate this call to an IM session, and from the IM session escalate to a web share session.		Passed w/ Exception	CSCtf84197
UC802IF.EXC.019	Unified CM, Unified PC 8.0	Unified PC 8.0 Through IPSEC VPN in Soft Phone Mode with Video Call	Verify if a user can bring up Unified PC 8.0 through an IPSEC VPN connection. Originate a call from Unified PC 8.0 to a Unified IP Phone through SIP ICT. Bring up video during the call. Hold and Resume the call. Turn off the video and turn on during the call. Check the call again for audio and video connection.		Passed	
UC802IF.EXC.020	Cisco IME, Unified CM, Unified PC 8.0	Softphone Mode B2B Audio Call flow Sanity	Verify if a user can use Unified PC 8.0 soft phone and verify various basic P2P phone flows in Cisco IME deployment.		Passed	
UC802IF.EXC.021	Cisco IME, Unified CM, Unified PC 8.0	Softphone Mode B2B Video Call flow Sanity	Verify if a user can use Unified PC 8.0 soft phone and verify various basic P2P video phone flows in Cisco IME deployment.		Passed w/ Exception	CSCte52965

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.EXC.022	Unified CM, Unified PC 8.0, Unified Presence 8.0	Soft Phone Escalate Incoming Call to Video, Merge with Incoming Video Call	Verify if a user can set up an audio call from a SIP video endpoint in another cluster via SIP trunk to Unified PC 8.0 in soft phone mode. Escalate the call to video. Set up an incoming video call from another SIP endpoint (7985) to the same Unified PC 8.0 client. Check for bi-directional voice path. Merge both the calls.		Passed	
UC802IF.EXC.023	Cisco IME, Unified CM, Unified PC 8.0, Unified Presence 8.0	Softphone Mode B2B Video Call to UC Integration™ for Microsoft Office Communicator	Verify operation of a video call over a Cisco IME link to a UC Integration for Microsoft Office Communicator client.		Passed	
UC802IF.EXC.024	MCU, Unified CM, Unified PC 8.0, Unified Presence 8.0	Ad-hoc Video Conference Using Software Media Bridge	Verify operation of an ad hoc inter-cluster video conference using a software media bridge with Unified PC 8.0 clients and Unified IP Phones 8900 and 9900 series.		Passed	
UC802IF.EXC.025	Unified CM, Unified PC 8.0, Unified Presence 8.0	Intercluster Scheduled Video Conference with Multiple Parties	Verify operation of an inter-cluster video conference using a Meeting Place hardware media conferencing bridge with Unified PC 8.0 clients and Unified IP Phones 8900 and 9900 series.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.EXC.027	Cisco IME, Unified CM, Unified PC 8.0, Unified Presence 8.0	PSTN Fallback with Unified PC 8.0 Client	Verify if a user can place a call over Cisco IME to another phone. Degrade the quality of the link such that PSTN fallback is initiated. Verify the Unified PC 8.0 client handles the fall back properly.		Passed	
UC802IF.EXC.028	Unified CM, Unified Mobility, Unified PC 8.0, Unified Presence 8.0	Dusting with Soft Phone Mode, Inbound From Mobile	Verify a call can be transferred from a mobility device to the Unified PC 8.0 soft phone.		Passed	
UC802IF.EXC.029	Unified CM, Unified PC 8.0, Unified Presence 8.0	LDAP Telephone Number Update	Verify if a user can change a contact's telephone number in LDAP. Verify Unified PC 8.0 receives new contact info for this user and click-to-call reaches the new number.		Failed	CSCtf17373
UC802IF.EXC.030	Unified CM, Unified PC 8.0	Unified PC 8.0 Making a Secure Call, Transfer to Nonsecure Phone	verify if a user can place a call using Unified PC 8.0 as a secure endpoint to another secure phone across a SIP trunk. Transfer the call to a non-secure endpoint and verify two-way audio remains.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.PPR.020.7	PhoneProxy	Hold and Resume, Music on Hold	Verifies Cisco Unified PhoneProxy based phones to hold and resume calls. Calls include other phones in cluster, gateway calls, and other PhoneProxy phones.		Passed	
SR60.PPR.050.2	PhoneProxy	Standard (Ad-Hoc) Conference	Verifies Cisco Unified PhoneProxy based phones are able to create a standard (ad-hoc) conference using the confirm and join softkeys.		Passed	
UC71.PPR.001	PhoneProxy	Encrypt End-to-End Conversation With TLS/sRTP	Verify the PhoneProxy functionality. Verify the ability to encrypt end-to-end conversation with TLS/sRTP between phone connected via internet/PhoneProxy and campus phone.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.CUP.007	Interoperability	Instant Messaging Between IP Phone Messenger on Extension Mobility Phone Controlled by Subscriber Unified Presence Server and Unified Personal Communicator Clients	Verify that a traveling user using IP Phone Messenger on an Extension Mobility SIP TNP Cisco Unified IP phone (IP Phone Messenger is controlled by a Subscriber Unified Presence Server) can exchange Instant Messages with a Unified Personal Communicator user.		Passed	
UC700IF.CUP.026	Unified Presence 8.0	Intercluster IM Between IPPM and Unified Personal Communicator Clients	Verify that IPPM and Unified Personal Communicator clients in separate clusters can exchange IM successfully.		Passed	
UC700IF.CUP.027	Unified Presence 8.0	Intercluster IM Between Unified Personal Communicator Clients Distributed in Two Clusters	To verify if a user can setup a IM session involving more than two Unified Personal Communicator clients and IPPM client distributed in multiple clusters.		Passed	
UC802EF.CUP.002	IM, Presence	Call from Unified Personal Communicator in Desk Phone to UC Integration™ for Microsoft Office Communicator		CUPC->Unified Presence->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	
UC802EF.CUP.004	Unified Presence	IM and Presence in Unified Presence 8.0	Verify if Office Communicator Server goes down.	CUPC->Unified Presence->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CUP.005	RSVP, Unified Presence 8.0	Video phones, Unified IP Phones 9971/9951/8961, Unified Personal Communicator Make Video Calls with RSVP		RT/Video Ph->REM->WAN->Unified Presence->Unified CM->SCCP Ph/SIP Ph/ CUPC	Passed	
UC802EF.CUP.006	IM, Unified Presence	Unified Personal Communicator, IM session, Unified Presence	Verify the inter working of Unified Personal Communicator with Extension mobility.	CUPC->Unified Presence->Unified CM->WAN->Rem->EM Ph1	Passed	
UC802EF.CUP.008	IM session, Presence in Unified Attendant Server	Interworking with Unified Attendant Server		Stage1: PSTN->FXO->REM->Unified CM->ARC;Stage2: ARC->Unified CM->Unified Presence->CUPC/UC Integration™ for Microsoft Office Communicator	Passed	
UC802EF.CUP.009	IM session, Presence in Unified Attendant Server	Interworking with Unified Attendant Server		Stage1: PSTN->FXO->REM->Unified CM->ARC;Stage2: ARC->Unified CM->Unified Presence->CUPC/UC Integration™ for Microsoft Office Communicator	Passed	
UC802EF.CUP.010	IM Session and RSVP Interworking	IM session on IP Phones, Video phones, Unified IP Phones 9971/9951/8961, and Unified Personal Communicator		RT/Video Ph->REM->WAN->Unified Presence->Unified CM->SCCP Ph/SIP Ph/ CUPC	Passed	
UC802EF.CUP.401	IM session, Unified PC 8.0, Unified Presence	Unified Personal Communicator 8.0 Interworking with Extension Mobility			Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CUP.402	IM session, Presence in Unified Attendant Server	Interworking with Unified Attendant Server			Passed	
UC802EF.CUP.403	RSVP Interworking, Video calls	Video phones, Unified IP Phones 9971/9951/8961, Unified Personal Communicator 8.0 Make Video Calls with RSVP			Passed	
UC802EF.CUP.403	Video Calls and RSVP	Video phones, Unified IP Phones 9971/9951/8961, Unified Personal Communicator 8.0 Make Video Calls with RSVP			Passed	
UC802EF.CUP.405	Unified Personal Communicator 8.0	Unified Personal Communicator 8.0 and Unity Connection			Passed	
UC802EF.CUP.405	Unified Personal Communicator 8.0 and Voicemail	Unified Personal Communicator 8.0 and Unity Connection			Passed	
UC802EF.CUP.406	Unified Personal Communicator 8.0 and CGPN	Unified Personal Communicator 8.0 and Calling Party Normalization			Passed	
UC802EF.CUP.406	Unified Personal Communicator 8.0 and CGPN	Unified Personal Communicator 8.0 and Calling Party Normalization			Passed	
UC802EF.CUP.407	Video Interoperability	Third Party Video Interoperability			Passed w/ Exception	CSCtg21657

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CUP.409	IM session, Presence, Unified Personal Communicator 8.0	Unified Personal Communicator 8.0 IM with Moment-IM, MOC and Unified Personal Communicator 7.x			Passed	
UC802EF.CUP.804	Video Conference Calls	Video phones, Unified IP Phones 9971/9951/8961, Unified Personal Communicator 8.0 are in Video Conference			Passed	
UC802EF.CUP.804	Video Conference Calls	Video Phones, Unified IP Phones 9971/9951/8961, Unified Personal Communicator 8.0 are in Video Conference			Passed	
UC802IF.CUP.001	Unified Presence 8.0	Unified Presence Upgrade	To verify the upgrade of Unified Presence 7.x high availability capable to Unified Presence 8.0 without high availability.		Passed	
UC802IF.CUP.004	Unified Presence 8.0	Unified Personal Communicator Intracluster to XMPP Client (IM and Presence Status)	Verify flows and compatibility when multiple client types are connected to one multinode Unified Presence cluster.	Client->ASA->Unified Presence 8.0->ASA->Client	Passed	CSCtc68619
UC802IF.CUP.005	Unified Presence 8.0	Unified Personal Communicator Intercluster to XMPP Client (IM and Presence Status)	Verify mixed client intercluster communications using Cisco and third party clients.	Unified Personal Communicator->ASA->Unified Presence->Unified Presence->ASA->Client	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CUP.006	Unified Presence 8.0	Unified SIP IP Phone 9971, Deskphone Mode with Unified Personal Communicator	To verify if a user can define Unified Personal Communicator and Unified IP Phone 9971 in phone associated mode. Ensure deskphone mode works correctly.	Unified Personal Communicator/RT Pro->ASA _> Unified CM->ASA->IP phone. Unified Personal Communicator/RT Pro->ASA->Unified Presence 8.0	Passed	
UC802IF.CUP.007	Unified Presence 8.0	Unified Presence 8.0 Server Fails and Restarts	Verify after the Unified Presence server failure, can clients reconnect successfully as high availability is not supported.	Client->ASA->Unified Presence8.0->ASA->client (Clients reside on separate Unified Presence nodes)	Passed	
UC802IF.CUP.008	Unified Presence 8.0	Client Fails and Restarts	To verify that a client (Unified Personal Communicator, Unified Mobility Advantage/Unified Mobile Communicator, XMPP) can restart and connect after failure. Ensure that failed clients are removed and allowed to reconnect to the Unified Presence server.	Unified Personal Communicator->ASA->Unified Presence->Unified Mobility->ASA->client	Passed	
UC802IF.CUP.009	Unified Presence 8.0	Intercluster Communications Failure	Verify the communications breakdown between clusters.	Client->ASA->Unified Presence->SDNS->Unified Presence->ASA->client	Passed	
UC802IF.CUP.010	Unified Presence 8.0	Client Redirect	Verify when a client logins to a wrong server, if login gets redirected.	Client->ASA->Unified Presence node->Unified Presence node	Passed	
UC802IF.CUP.012	Unified Presence 8.0	Remote Clients Across WAN	Verify that clients located in remote sites are able to log in to centralized Unified Presence server and communicate with users in central and remote site across WAN with 80 ms delay.	Unified Personal Communicator->ASA->WAN->ASA->Unified Presence->Unified Personal Communicator	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CUP.013	Unified Presence 8.0	Adhoc Text Conferencing	Verify the operation of XMPP ad-hoc text conferencing intra-cluster and inter-cluster.	XMPP->Unified Presence->XMPP	Passed	
UC802IF.CUP.014	Unified Presence 8.0	Off-Line Storage	Verify that offline IMs are received by users once they log in across multiple clusters.		Passed	
UC802IF.CUP.015	Unified Presence 8.0	User Rebalance	Verify the user rebalance.		Passed	
UC802IF.CUP.016	Unified Presence 8.0	Basic Compliancy Using Message Archiver	Verify the operation/integration of basic compliancy using an external PostgreSQL database for message archiving.		Passed	
UC802IF.CUP.017	Unified Presence 8.0	Inter-Cluster Persistent Chat	Verify the inter-cluster persistent chat functionality.		Passed	
UC802IF.CUP015	Unified Presence 8.0	Unified Presence Replication Intact During Extended Period of NTP Server Outage	To verify that the Unified Presence replication remains intact during extended period of NTP server outage.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.SRST.201.1	Basic Call Flow	Unified SRST Support for Out of Dialog	To verify the use of OOD-R from one Unified SRST site to another Unified SRST site.		Passed	
SR60.SRST.202.2	Basic Call Flow	Consultative Transfer from Unified SRST SIP Network to Unified SRST H.323 Network Involving IP-to-IP Gateway	Verify if a user can call from a Unified SRST SCCP IP Phone through a IP-to-IP gateway to a Cisco Unified SRST SIP IP Phone via SIP trunk. The call transferred (blind) to a SIP IP Phone registered to another Unified SRST Site and transferred back (blind) to the originating SRST site (the same cluster as called party) SCCP IP Phone.		Passed	
SR60.SRST.202.5	Basic Call Flow	Hold and Resume Where Call is Placed on Hold on Unified SRST SIP Site	Verify if a call from a Unified SRST SCCP IP Phone through a IP-to-IP gateway to a Unified SRST SIP IP Phone via SIP trunk can be placed on hold and resumed.		Passed	
SR60.SRST.202.7	Basic Call Flow	Call Forward Across IP-to-IP Gateway Involving Two Unified SRST Sites	Verify if a call from a Unified SRST SIP IP Phone through an IP-to-IP gateway to a Unified SRST SIP IP Phone via SIP trunk with 'Call Forward All' set on the called party to a SCCP Phone registered to Unified SRST as the called party is successful.		Passed	
SR61.SRST.102	Unified SRST	Secured SCCP Phones on Unified SRST Gateway When WAN Link Goes Down	To verify the behavior of secured SCCP Phones on Unified SRST gateway when WAN link goes down.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713IF.SRS.001	Unified SRST	Failover and Fallback Features in IP Phone 6921/6941/6961 in SRST mode	Verifies whether IP Phone 6921/6941/6961 can Failover to SRST Mode when WAN Link or Unified CM is down and FallBack to HQ Site when WAN Link recovers.	6921/6941/6961->Unified Communications Manager->WAN->SRST Gateway->Fail WAN Link->SRST Mode Ephone->PSTN->HQ Site->Phone	Passed	
UC713IF.SRS.002	Unified SRST	Dial International Number Preceded by + from IP Phone 6921/6941/6961 Registered to Unified SRST Site	Verify whether IP Phone 6921/6941/6961 in Unified SRST Site can dial International number preceded by +.	6921/6941/6961->Unified Communications Manager->WAN->SRST Gateway->Fail WAN Link->SRST Mode Ephone->Phone (dial +1 followed digits by International number)	Passed	
UC713IF.SRS.005	Unified SRST	IP Phone 6921/6941/6961 Access to Voicemail After FallBack	Verify whether IP Phone 6921/6941/6961 can Failover to SRST Mode when WAN Link is Down and Fallback to HQ Site when WAN Link is recovered or if Unified CM is down and is able to access voicemail located in main site.	6921/6941/6961->Unified Communications Manager->WAN->SRST Gateway->Fail WAN Link->SRST Mode Ephone->PSTN->Unified IP Phone(main site)	Passed w/ Exception	CSCtc66818
UC802IA.SRST.501	Unified SRST	Unified SRST: Verify Calls Between Two Endpoints	Verify if Phone A can call Phone B and also if Phone B can call Phone A when either or both are in Unified SRST mode.		Passed	
UC802IA.SRST.502	Unified SRST	Unified SRST: Verify Hold and Resume When Phone is in SRST Mode	Verify if during a call between Phone A and Phone B (either or both are in SRST mode), hold/resume either Phone A or B three times.		Passed	
UC802IA.SRST.503	Unified SRST	Unified SRST: Verify Hold and Resume When Phone is in SRST Mode	Verify if during a call between Phone A and Phone B (either or both are in SRST mode), hold/resume either Phone A or B three times.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IA.SRST.601	Unified SRST	Unified SRST: Verify Multiple Calls on Multiple Lines of a Phone in SRST mode	Verify if EpAsstnt1 (assistant endpoint 1) makes first call to epFeature (feature applied endpoint). EpAsstnt2 (assistant endpoint 2) makes second call to epFeature on a different line. EpFeature can switch the first and second calls correctly.		Passed	
UC802IA.SRST.602	Unified SRST	Unified SRST: Verify Multiple Calls on Multiple Lines of a phone in SRST mode	Verify if EpAsstnt1 (assistant endpoint 1) makes first call to epFeature (feature applied endpoint), EpAsstnt2 (assistant endpoint 2) makes second call to epFeature on a different line and if EpFeature can switch the first and second calls correctly.		Passed	
UC802IA.SRST.603	Unified SRST	Unified SRST: Verify Call Forward All by Softkey When Phone is SRST mode	Verify when epFeature (feature applied endpoint) is the phone pushing the softkey; epAsstnt1 (assistant endpoint 1) is the phone making calls; epAsstnt2 (assistant endpoint 2) is the target of call forward to.		Passed	
UC802IA.SRST.604	Unified SRST	Unified SRST: Verify Call Forward Busy When Phone is SRST Mode	Verify when epAsstnt1 (assistant endpoint 1) calls epFeature (feature applied endpoint) and when epFeature is busy the call is forwarded to epAsstnt2 (assistant endpoint 2).		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IA.SRST.605	Unified SRST	Unified SRST: Verify Call Forward No Answer When Phone is SRST mode	Verify when epAsstnt1 (assistant endpoint 1) calls epFeature (feature applied endpoint), and if epFeature is ringing but no answer, then the call is forwarded to epAsstnt2 (assistant endpoint 2).		Passed	
UC802IA.SRST.606	Unified SRST	Unified SRST: Verify Call Forward All by Softkey When Phone is SRST mode	Verify when epFeature (feature applied endpoint) is the phone pushing the softkey; epAsstnt1 (assistant endpoint 1) is the phone making calls; epAsstnt2 (assistant endpoint 2) is the target of call forward to.		Passed	
UC802IA.SRST.607	Unified SRST	Unified SRST: Verify Call Forward Busy When Phone is SRST mode	Verify when epAsstnt1 (assistant endpoint 1) calls epFeature (feature applied endpoint), when epFeature is busy the call is forwarded to epAsstnt2 (assistant endpoint 2).		Passed	
UC802IA.SRST.608	Unified SRST	Unified SRST: Verify Call Forward No Answer When Phone is SRST mode	Verify when epAsstnt1 (assistant endpoint 1) calls epFeature (feature applied endpoint), epFeature is ringing but no answer, then the call is forwarded to epAsstnt2 (assistant endpoint 2).		Passed	
UC802IF.SRST.905	Unified CME and SRST	Supplementary Services Through SIP Phones With Unified CME in Unified SRST Mode	To verify whether the user can perform Supplementary services through Unified SIP IP Phones in Unified SRST Mode.	RT SIP Phone --- WAN Link --- SRST -----Transfer/Hold Resume /Conference ----- PSTN ---HQ Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.SRST.906	Unified SRST	Unified SRST SCCP Phone Hears Multicast MOH	Verifies whether Unified SRST SCCP phone can hear multicast MOH when Central Site Unified CM phone puts the call on hold.	IP Phone 6921/6941/6961 --- Unified CM ---- WAN Link --- SRST Phone ----Unified CM-on Hold ---- MMOH --- SRST Phone	Passed	
UC802IF.SRST.907	Unified SRST	Unified SRST Phone Joins MeetingPlace Meeting Hosted by Central Site Unified CM User	Verify whether SIP Secured Unified SRST can join meeting via PSTN phone.	Unified CM--WAN Link---SRST -- Secure ---PSTN --- Meetingplace	Passed	
UC802IF.SRST.908	Unified SRST	Call From Unified SRST to H323 Gateway	To verify whether Unified SRST Phone is able to place a PSTN Call to Unified CCX agent when the router is in fall back mode.	Unified CM -- WAN Link---SRST -- PSTN --- Unified CCX Agent	Passed	
UC802IF.SRST.909	Unified SRST	Call From Unified SRST:SIP Gateway to Unified CM Phone	Verifies whether a phone can talk to Unity Connection through PSTN and is also able to deposit mail to Unified CM subscriber.	Unified CM -- WAN Link --- SIP --- SRST ---Fail over ----PSTN -- Unified CM Phone --Unity connection	Passed	
UC802IF.SRST.910	Unified SRST	MGCP Fallback to PSTN	Verifies whether Unified SRST phones can call PSTN when Unified SRST:MGCP gateway is in Fall back mode.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712IF.CUSP.002	Cisco Unified SIP Proxy	Call From SIP Unified CME Site to Unified CM Site Via Cisco Unified SIP Proxy	Verify Unified CME SIP Phone registered with ISR can place a call through Cisco Unified SIP Proxy 1 and Cisco Unified SIP Proxy 2 to Unified CM Phone.		Passed	
UC712IF.CUSP.004	Unified SIP Proxy	Unified CME Phone Calling PSTN Phone via Unified SIP Proxy	To verify calls from Unified CM Phone to PSTN Phone via Unified SIP Proxy.		Passed	
UC712IF.CUSP.006	Unified SIP Proxy	Verify Supplementary Services via Unified SIP Proxy	Verify call from Unified CME Phone to Unified CM Phone and perform the following: a. Call Transfer b. Hold and resume c. Call Forward		Passed	
UC712IF.CUSP.009	Cisco Unified SIP Proxy	Call From Unified CME Phone to Unified CCX via Cisco Unified SIP proxy	Verify the calls from Unified CME Phone to Unified CCX through Cisco Unified SIP proxy.		Passed	
UC712IF.CUSP.010	Cisco Unified SIP Proxy	Unified IP Phone 9971 Video Call Between Unified CM via Cisco Unified SIP Proxy	To verify the inter cluster calls between Unified IP Phones 9971 and Unified CM Phone through Cisco Unified SIP Proxy.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.UNI.001.1	Voice Mail	Dropped Call Recovery-Caller Hangs Up	To verify that the subscriber can resume a previously dropped call into Cisco Unity back to the place where the subscriber drops the call by hanging up the phone.		Passed	
SR60.UNI.001.13	Voice Mail	Dropped Call Recovery-Caller Hangs Up	To verify that the subscriber can resume a previously dropped call into Cisco Unity back to the place where the subscriber drops the call by hanging up the phone.		Passed	
SR60.UNI.001.14	Voice Mail	Dropped Call Recovery-Caller Hangs Up	To verify that the subscriber can resume a previously dropped call into Cisco Unity back to the place where the subscriber drops the call by hanging up the phone.		Passed	
SR60.UNI.002.19	Voice Mail	Dropped Call Recovery-Network Failure	To verify that the subscriber is not negatively impacted when calls are dropped due to network failures. Dropped call recovery is not supported with network failures, but the solution components should not encounter any negative impacts.		Passed	
SR60.UNI.002.2	Voice Mail	Dropped Call Recovery-Network Failure	To verify that the subscriber is not negatively impacted when calls are dropped due to network failures. Dropped call recovery is not supported with network failures, but the solution components should not encounter any negative impacts.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.UNI.002.7	Voice Mail	Dropped Call Recovery-Network Failure	To verify that the subscriber is not negatively impacted when calls are dropped due to network failures. Dropped call recovery is not supported with network failures, but the solution components should not encounter any negative impacts.		Passed	
SR60.UNI.003.14	Voice Mail	Message Monitor-Gateway Call	To verify the message monitor feature from gateway callers.		Passed	
SR60.UNI.003.58	Voice Mail	Message Monitor-Gateway Call	To verify the message monitor feature from gateway callers.		Passed	
SR60.UNI.005.1	Voice Mail	Secure Messaging Enhancements-Secure Messages From VPIM Subscribers	To verify that the messages will be properly decrypted and sent to Trusted Internet Subscribers through VPIM.		Passed	
SR60.UNI.006.3	Voice Mail	Secure Messaging Enhancements-Secure Messages From VPIM Subscribers	To verify that the messages will be properly encrypted when received from Trusted Internet Subscribers through VPIM.		Passed	
SR60.UNI.007.4	Voice Mail	Call Forward Configuration-Gateway	To verify that Cisco Unity will forward calls from a gateway to the appropriate mailbox depending on the call forward configuration.		Passed	
SR60.UNI.007.8	Voice Mail	Call Forward Configuration-Gateway	To verify that Cisco Unity will forward calls from a gateway to the appropriate mailbox depending on the call forward configuration.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.UNI.008.2	Voice Mail	Call Forward Configuration-Phone	To verify that Cisco Unity will forward calls from a phone to the appropriate mailbox depending on the call forward configuration.		Passed	
UC700IF.UNI.103	Cisco Unity	Live Reply to External VPIM Subscriber Homed on Another Unity Server	To verify live replies to external subscriber.		Passed	
UC802IF.UNI.101	Cisco Unity	eMWI on Shared Lines With Cisco Unity-Unified CM SCCP Integration	Verify that MWI count is seen on Unified IP Phone 8900 and 9900 series when a new voicemail is left for the subscriber in Cisco Unity integrated to Unified CM using SCCP.	IP Phone->Unified CM->SCCP->Unity	Passed	
UC802IF.UNI.102	Cisco Unity	eMWI on Shared Lines With Unity-Unified CM SIP Integration	Verify that MWI count is seen on Unified IP Phone 8900 and 9900 series when a new voicemail is left for the subscriber in Cisco Unity integrated to Unified CM using SIP.	IP Phone->Unified CM->SIP->Unity	Passed	
UC802IF.UNI.103	Cisco Unity	eMWI for Extension Mobility Across Cluster	Verify that eMWI works for Extension Mobility across cluster.	Unity (MWI)->Unified CM->SIPT->Unified CM->IP Phone 9971/9951/8961 (Extension Mobility)	Passed	
UC802IF.UNI.104	Cisco Unity	Support of eMWI Over Inter Cluster SIP Trunks	Verify that eMWI works over inter cluster CIP trunks.	Unity (MWI)->Unified CM->SIPT->Unified CM->IP Phone 9971/9951/8961	Passed	
UC802IF.UNI.105	Cisco Unity	Support of eMWI During Failover and Bulk Re-Synchronization in Cisco Unity	Verify that eMWI works when primary VM server is down and the standby server is providing VM service.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.UNI.110	Cisco Unity	Cisco Unity in UMR Mode - Partner Exchange Server Unavailable for Short Duration	Verify that Cisco Unity can handle calls when it looses connectivity to the partner exchange server.	IP Phone->Unified CM->SCCP->Unity	Passed	
UC802IF.UNI.111	Cisco Unity	Operation of Cisco Unity When Non-Partner Exchange Server is Offline	Verify that Cisco Unity can handle calls when it looses connectivity to the partner exchange server.	IP Phone->Unified CM->SCCP->Unity	Passed	
UC802IF.UNI.112	Cisco Unity	Unity in UMR mode - Partner Exchange Server Unavailable for Long Duration	Verify that Cisco Unity can handle calls when it looses connectivity to the partner exchange server for more than 24 hours.	IP Phone->Unified CM->SCCP->Unity	Passed	
UC802IF.UNI.113	Cisco Unity	Primary Unified CM Server Unavailable to Cisco Unity Server	Verify that Cisco Unity can handle calls when it looses connectivity to the primary Unified CM server.	IP Phone->Unified CM->SCCP->Unity	Passed	
UC802IF.UNI.114	Cisco Unity	Primary Global Catalog Server Unavailable	Verify that Cisco Unity can handle calls when it looses connectivity to the primary Global Catalog server.	IP Phone->Unified CM->SCCP->Unity	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.UNC.100.2	Basic Call Flow, Voice Mail	Unity Connection VPIMv2 Standalone, Reply, Private	Verification of Unity Connection VPIM message delivery to multiple Cisco VPIM networked systems. Once a message is delivered from Unity Connection to the VPIM networked system it will be replied to. This will verify that Unity Connection is able to send and receive a VPIM message between supported Cisco VPIM voice messaging systems and process correctly.		Passed	
SR60.UNC.100.4	Basic Call Flow, Voice Mail	Unity Connection VPIMv2 Standalone, Dist List, Not Private	Verification of Unity Connection VPIM message delivery to multiple Cisco VPIM networked systems. Once a message is delivered from Unity Connection to the VPIM networked system it will be replied to. This will verify that Unity Connection is able to send and receive a VPIM message between supported Cisco VPIM voice messaging systems and process correctly.		Passed	
SR60.UNC.110	Basic Call Flow, Voice Mail	Unity Connection Standalone VPIMv2 NDR	Verify if non-delivery receipts (NDRs) work when addressing two multiple Cisco VPIM networked systems using Cisco Unity Connection Standalone.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR61.UNC.101	Unity Connection	Encrypted Call Involving Unity Connection Blind Transferred to a Non-Secure IP Phone	Verify that an encrypted call between an IP Phone and Unity Connection can be successfully transferred to a non-secure IP Phone.		Passed	
SR61.UNC.102		Call From Secure Gateway Supervised Transfer to Encrypted IP Phone	Verify that a call from a secure gateway can be successfully transferred to an IP Phone that supports media encryption.		Passed	
UC700IF.UNC.502	Unity Connection	DTMF Negotiation Between a RFC 2833 Capable Device and KPML Capable Unity Connection	To verify that MTP is invoked for a call between SIP gateway configured to support only RFC 2833 and Unity Connection configured to support only KPML.		Passed	
UC700IF.UNC.503	Unity Connection	Support for RFC 2833 DTMF in a Secure SIP Integration Between Unified CM and Unity Connection	To verify that Unity Connection can successfully decrypt RFC2833 digits received from the endpoint for a encrypted call.		Passed	
UC701IF.UNC.100.1	Unity Connection	Unity Connection Redundancy And Resiliency	To verify the Unity Connection resilience to failures and capability to load share in a Active-Active deployment.		Passed	
UC701IF.UNC.100.3	Unity Connection	Unity Connection Redundancy And Resiliency	To verify the Unity Connection resilience to failures and capability to load share in a Active-Active deployment.		Passed	
UC701IF.UNC.100.5	Unity Connection	Unity Connection Redundancy And Resiliency	To verify the Unity Connection resilience to failures and capability to load share in a Active-Active deployment.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701IF.UNC.500	Unity Connection	SIP Gateway and Unity Connection Configured to Support Both KPML and RFC 2833	To verify that when RFC 2833 is negotiated for the call then Connection ignores KPML and sends/receives RFC 2833 only.		Passed	
UC701IF.UNC.700.2	Unity Connection	Unity Connection Codec Support And Transcoding Capability	To verify the Unity Connection capabalility to support a suite of codecs for RTP streams and transcode the RTP stream to the recording format.		Passed	
UC701IF.UNC.700.4	Unity Connection	Unity Connection Codec Support and Transcoding Capability	To verify the Unity Connection capabalility to support a suite of codecs for RTP streams and transcode the RTP stream to the recording format.		Passed	
UC701IF.UNC.700.5	Unity Connection	Unity Connection Codec Support and Transcoding Capability	To verify the Unity Connection capabalility to support a suite of codecs for RTP streams and transcode the RTP stream to the recording format.		Passed	
UC701IF.UNC.900	Unity Connection	Chaining Message Notification Limited by Sender When Secondary Becomes Acting Primary	Verify that message notification is triggered based on the subscriber configured setting when a message arrives.		Passed	
UC701IF.UNC.904	Unity Connection	Unity Connection Integration With Unified CME Over SIP With DTMF Support for RFC2833	To verify that RFC 2833 is negotiated successfully between Unified CME and Unity Connection.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701IF.UNC.908	Unity Connection	Blind Transfer with Unity Connection Over a Secure SIP Trunk	To verify that a secure call to Unity Connection can be transferred to another secure phone and the call remains secure. The transfer is being done by the phone and not Unity Connection.		Passed	
UC701IF.UNC.912	Unity Connection	Last Redirecting Number With Multiple Call Forwarding in Unified CME	To verify that Unified CME provides the last redirecting number to Unity Connection rather than first.		Passed	
UC701IF.UNC.915	Unity Connection	NDR for Message Sent by Unity Connection to Unified Messaging Gateway	Verify if a NDR sent by Unified Messaging Gateway reaches Unity Connection and is delivered to the senders mailbox.		Failed	
UC802EF.UNC.001	Unity Connection	Migration of Cisco Unity 7.x to Unity Connection 8.0	Verify the migration of Unity 7.x high-availability setup to Unity Connection 8.0.		Passed	
UC802EF.UNC.002	Unity Connection	Visual Voicemail with Unity Connection	Verify visual voicemail capability on Unified IP Phones (8961/9951/9971) with Unity Connection	IP Phones (8961/9951/9971)->CCM->Unity Connection	Passed	
UC802EF.UNC.003	Unity Connection	Enhanced MWI with Unity Connection	Verify enhanced MWI capability on Unified IP Phones (8961/9951/9971) with Unity Connection.	IP Phone 99xx/89xx->CCM->Unity Connection	Passed	
UC802EF.UNC.004	Unity Connection	Voicemail for SIP IP Phones (8961/9951/9971) in Unified CM 8.0 Cluster	Verify deposit and retrieval of a message for a Unified IP Phones (8961/9951/9971) in Unified CM 8.0 cluster.	Phone->PSTN GW->CCM->IP Phone 99xx/89xx->CFNA->Unity Connection	Passed	
UC802EF.UNC.005	Unity Connection	Voicemail for SCCP Unified IP Phones 6921/6941/6961 in Unified CM 8.0 cluster	Verify deposit and retrieval of a message for Unified IP Phones 6921/6941/6961 in Unified CM 8.0 cluster.	Phone->PSTN GW->CCM->IP Phone 69xx->CFNA->Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.UNC.006	Unity Connection	Voicemail for QSIG PBX Phone in Unified CM 8.0 Cluster	Verify deposit and retrieval of a message for a QSIG PBX phone in Unified CM 8.0 cluster.	Phone->CCM->QSIG PBX->PBX Phone->CFNA->Unity Connection	Passed	
UC802EF.UNC.007	Unity Connection	Voicemail for IP Phone 99xx/89xx (SIP) in CCM 7.x Cluster	Verify deposit and retrieval of a message for a RT phone in CCM 7.x cluster	Phone->PSTN GW->CCM 7.x->IP Phone 99xx/89xx->CFNA->Unity Connection	Passed	
UC802EF.UNC.008	Unity Connection	Voicemail for SCCP Unified IP Phones 6921/6941/6961 in Unified CM 7.x cluster	Verify deposit and retrieval of a message for Unified IP Phones 6921/6941/6961 in Unified CM 7.x cluster.	Phone->PSTN GW->CCM 7.x->IP Phone 99xx/89xx->CFNA->Unity Connection	Passed	
UC802EF.UNC.009	Unity Connection	Voicemail for QSIG PBX Phone in Unified CM 7.x Cluster	Verify deposit and retrieval of a message for a QSIG PBX in Unified CM 7.x cluster.	Phone ->CCM 7.x->QSIG PBX->PBX Phone->CFNA->Unity Connection	Passed	
UC802EF.UNC.010	Unity Connection	Voicemail for Unified IP Phones in Unified CME site	Verify deposit and retrieval of a message for Unified IP Phones in Unified CME site over H.323 network.	Phone->PSTN GW->CME->Phone->CFNA->Unity Connection	Passed w/ Exception	This testcase was executed with TNP phones and not with IP Phones 99xx/89xx series as some of these phones are not supported with Unified CME.
UC802EF.UNC.011	Unity Connection	Voicemail for QSIG PBX Phone in Unified CME Site	Verify deposit and retrieval of a message for a QSIG PBX phone in Unified CME site over H.323 network.	Phone->CME->QSIG PBX->PBX Phone->CFNA->Unity Connection	Passed	
UC802EF.UNC.012	Unity Connection	Voicemail Server Redundancy/Negative testing	Verify Unity Connection voicemail server redundancy.		Passed	
UC802EF.UNC.013	Unity Connection	Supervised Transfer in Unity Connection	Verify operation of supervised transfer in Unity connection.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.UNC.014	Unity Connection	Directory Handlers in Unity Connection	Verify operation of directory handlers in Unity connection.		Passed	
UC802EF.UNC.015	Unity Connection	Interview Handlers in Unity connection	Verify operation of interview handlers in Unity connection.		Passed	
UC802EF.UNC.016	Unity Connection	Voicemail for Remote Phones Over H.323/FXO PSTN Gateway	Verify voicemail deposit and retrieval for remote phones over H.323/FXO PSTN gateway.		Passed	
UC802EF.UNC.016	Unity Connection	Voicemail for Remote Phones Over MGCP/FXO PSTN Gateway	Verify voicemail deposit and retrieval for remote phones over MGCP/FXO PSTN gateway.		Passed	
UC802EF.UNC.016	Unity Connection	Voicemail for Remote Phones Over MGCP/PRI PSTN Gateway	Verify voicemail deposit and retrieval for remote phones over MGCP/PRI PSTN gateway.		Passed	
UC802EF.UNC.016	Unity Connection	Voicemail for Remote Phones Over MGCP BRI PSTN Gateway	Verify voicemail deposit and retrieval for remote phones over MGCP BRI PSTN gateway.		Passed	
UC802EF.UNC.016	Unity Connection	Voicemail for Remote Phones Over SIP/BRI PSTN Gateway	Verify voicemail deposit and retrieval for remote phones over SIP/BRI PSTN gateway.		Passed	
UC802EF.UNC.016	Unity Connection	Voicemail for Remote Phones Over Various PSTN Gateways	Verify voicemail deposit and retrieval for remote phones over various PSTN gateways.		Passed	
UC802EF.UNC.016	Unity Connection	Voicemail for Remote Phones Over H.323/PRI PSTN Gateway	Verify voicemail deposit and retrieval for remote phones over H.323/PRI PSTN gateway.		Passed	
UC802EF.UNC.016	Unity Connection	Voicemail for Remote Phones Over SIP/PRI PSTN Gateway	Verify voicemail deposit and retrieval for remote phones over SIP/PRI PSTN gateway.		Passed	
UC802EF.UNC.017	Unity Connection	Voicemail for DPNSS PBX Phone with Unity Connection	Verify deposit and retrieval of messages for a DPNSS PBX phone registered to a Unified CM 8.0 cluster.	IP Phone->Unified CM->VG 30D->DPNSS PBX->Unified CM->DPNSS PBX Phone->CFNA->Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.UNC.018	Unity Connection	Voicemail for a Remote Unified IP Phones 8961/9951/9971 Registered to Unified CM 7.x Cluster	Verify deposit and retrieval of messages for a Remote Unified IP Phones 8961/9951/9971 registered to Unified CM 7.x cluster.	PSTN Phone->PSTN GW->Rem IP Phone 99xx/89xx->CCM 7.x->Unity Connection	Passed	
UC802IF.UNC.201	Cisco Unity Connection	Download Direct and Forwarded Voicemails From Unity Connection Server2 When Server1 is Down	To verify that Cisco UC Integration for Microsoft Office Communicator can successfully download direct and forwarded voicemail from currently active Unity Connection server when one of the Unity Connection servers in active-active cluster is down where the VM is originally deposited.	UC Integration™ for Microsoft Office Communicator->Unified CM->Unity Connection	Passed	
UC802IF.UNC.202	Cisco Unity Connection	Updating list, Message With attachment, and Sender's Presence Status With Visual Voicemail on Cisco UC Integration™ for Microsoft Office Communicator	To verify that visual voicemail list is updated on arrival of new message while list is displayed, ability to display messages with attachment along with the presence status of the sender.	Unity Connection->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	
UC802IF.UNC.203	Cisco Unity Connection	Message Actions With Visual Voicemail on Cisco UC Integration™ for Microsoft Office Communicator	Verify that message actions such as play, pause, rewind, mark as new, delete, reply, reply to IM can be performed.	Unity Connection->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.UNC.204	Cisco Unity Connection	Retrieve, Reply And Send a Secure Message Using Cisco UC Integration™ for Microsoft Office Communicator	Verify that messages that have been marked secure can be retrieved and replied to. Also if it is possible to send a voicemail and mark it secure.	Unity Connection->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	
UC802IF.UNC.205	Cisco Unity Connection	Securing Visual Voicemail With Cisco UC Integration™ for Microsoft Office Communicator	Verify that security (HTTPS) can be enabled for visual voicemail.	Unity Connection->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	
UC802IF.UNC.206	Cisco Unity Connection	Download Direct and Forwarded Voicemails From Unity Connection Server2 When Server1 is Down	To verify that Unified IP Phone 8900 and 9900 series can successfully download direct and forwarded voicemails from currently active Unity Connection server when the primary Unity Connection server in active-active cluster is down where the VM is originally deposited.	RT phone->Unified CM->Unity Connection	Passed	
UC802IF.UNC.207	Cisco Unity Connection	Updating list, Message With attachment, and Sender's Presence Status With Visual Voicemail on Unified IP Phone 8900 and 9900 series	To verify that visual voicemail list is updated on arrival of new message while list is displayed, ability to display messages with attachment along with the presence status of the sender.	Unity Connection->Unified CM->RT phone	Passed	
UC802IF.UNC.208	Cisco Unity Connection	Message Actions With Visual Voicemail on Unified IP Phone 8900 and 9900 series	Verify that message actions such as play, pause, rewind, mark as new, delete, reply, reply to IM can be performed.	Unity Connection->Unified CM->RT phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.UNC.209	Cisco Unity Connection	Retrieve, Reply And Send a Secure Message Using Unified IP Phone 8900 and 9900 series	Verify that messages that have been marked secure can be retrieved and replied to. Also if it is possible to send a voicemail and mark it secure.	Unity Connection->Unified CM->RT phone	Passed	
UC802IF.UNC.210	Cisco Unity Connection	Securing Visual Voicemail With Unified IP Phone 8900 and 9900 series	Verify that security (HTTPS) can be enabled for visual voicemail.	Unity Connection->Unified CM->RT phone	Passed	
UC802IF.UNC.211	Unity Connection	Short Duration Failure of Unity Connection Publisher During Light Traffic	To verify that Unity Connection Publisher comes back up and running after short duration failure and under light traffic.	UNC1(active(UNC2--<skinny>---Unified CM	Passed	
UC802IF.UNC.212	Unity Connection	Short Duration Failure of Unity Connection Subscriber During Light Traffic	To verify that Unity Connection Subscriber comes back up and running after short duration failure and under light traffic.	UNC1(active(UNC2--<skinny>---Unified CM	Passed	
UC802IF.UNC.213	Unity Connection	Long Duration Failure of Unity Connection Subscriber During Light Traffic	To verify that Unity Connection Subscriber comes back up and running after Long duration failure(>12 hours) and under light traffic.	UNC1(active(UNC2--<skinny>---Unified CM	Passed	
UC802IF.UNC.214	Unity Connection	Unity Connection Failover in Active-Active Setup When Critical Service is Stopped	To verify that Unity Connection subscriber server picks up the functionality of the other server in the cluster when it fails.	UNC1(active(UNC2--<skinny>---Unified CM	Passed	
UC802IF.UNC.251	Unity Connection	Replication of Objects From Remote Site to Unity Connection Digitally Networked	Verify that objects such as users, SDL and CSS on a remote connection (inter-site link) are replicated to a Unity Connection server that is digitally networked (intra-site link).	Node A <-IntraSite->Node B<-InterSite->NodeC	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.UNC.252	Unity Connection	Sending and Receiving Messages Across Network of Unity Connection Servers	Verify that messages can be sent to any user on inter-site or intra-site link.	Node A <-IntraSite->Node B<-InterSite->NodeC	Passed	
UC802IF.UNC.253	Unity Connection	Sending Messages to SDL	Verify that messages can be sent to a SDL that has members from all the networked servers (intra-site and inter-site links).	Node A <-IntraSite->Node B<-InterSite->NodeC	Passed	
UC802IF.UNC.254	Unity Connection	Non-Delivery Receipt for Messages Sent to Non-Existent Mailbox	Verify that Non-Delivery Receipts (NDR's) are received when messages are sent to a non-existent subscriber.	Node A <-IntraSite->Node B<-InterSite->NodeC	Passed	
UC802IF.UNC.255	Unity Connection	Cross Box Handoff to User in Remote Site	Verify that the cross box handoff is successful with CCI.	Node A <-IntraSite->Node B<-InterSite->NodeC	Passed	
UC802IF.UNC.256	Unity Connection	Deleting SDL Used by Directory Handler in Remote Site	Verify that deleting SDL used by Directory Handler does not cause any error condition.	Node A <-IntraSite->Node B<-InterSite->NodeC	Passed	
UC802L.UNC.001	Cisco Unity Calls	IP to Unity Calls	Verify that calls to SCCP integrated Unity Voicemail are able to deposit and retrieve voicemail successfully.		Passed	
UC802L.UNC.301		IP to Unity Calls, SIP Integrated	Verify that calls to SIP integrated Unity Voicemail are able to deposit and retrieve voicemail successfully.		Passed	
UCS712IF.UNC.103	Unity Connection	Message Handling for Non-Existent Subscriber with Unified Messaging Gateway	To verify that a fax received by Cisco Gateway can be relayed to Unity Connection via Cisco Fax Server. And to verify that NDR is received by Connection from Unified Messaging Gateway when the message is forwarded to non-existent subscriber in Cisco Unity Express.	Fax -->PSTN -->IOS Gateway -->CFS -->Unity Connection <-->VPIM <-->UMG <-->VPIM <-->Cisco Unity Express	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF.UNC.103	Messaging Gateway, Unity Connection, Unity Express	Message Handling for Non-Existent Subscriber with Unified Messaging Gateway	To verify that a fax received by Cisco Gateway can be relayed to Unity Connection via Cisco Fax Server. And to verify that NDR is received by Unity Connection from Unified Messaging Gateway when the message is forwarded to non-existent subscriber in Unity Express.	Fax -->PSTN -->IOS Gateway -->CFS -->Unity Connection <-->VPIM <-->UMG <-->VPIM <-->CUE	Passed	
UCS712IF.UNC.201	Unity Connection	Forwarding a Fax Message From Unity Connection to Cisco Unity Express Over VPIM	To verify that a fax received by Cisco Gateway can be relayed to Unity Connection via Cisco Fax Server. And to verify that the Connection subscriber can forward the message to Cisco Unity Express over VPIM and Cisco Unity Express subscriber can in turn forward back the message to Connection subscriber over a VPIM network.	Fax -->PSTN -->IOS Gateway -->CFS -->Unity Connection <-->VPIM <-->Cisco Unity Express	Passed	
UCS712IF.UNC.201	Cisco Unity Express, Unity Connection	Forwarding a Fax message From Unity Connection to Cisco Unity Express over VPIM and back	To verify that a fax received by Cisco Gateway can be relayed to Unity Connection via Cisco Fax Server. And to verify that the Connection subscriber can forward the message to Unity Express over VPIM and Unity Express subscriber can in turn forward back the message to Unity Connection subscriber over a VPIM network.	Fax -->PSTN -->IOS Gateway -->CFS -->Unity Connection <-->VPIM <-->CUE	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF.UNC.206	Unity Connection	Incoming Fax Message to Remote Contact	To verify that Unity Connection can receive a fax for Cisco Unity Express subscriber from CFS and send it to the remote contact.	Fax -->PSTN -->IOS Gateway -->CFS -->Unity Connection <-->VPIM <-->Cisco Unity Express	Passed	
UCS712IF.UNC.206	Messaging Gateway, Unity Connection, Unity Express	Incoming Fax Message to a Remote Contact	To verify that Unity Connection can receive a fax for Unity Express subscriber from CFS and send it to the remote contact.	Fax -->PSTN -->IOS Gateway -->CFS -->Unity Connection <-->VPIM <-->CUE	Passed	
UCS712IF.UNC.301	Cisco Unity Express, Unified Messaging Gateway, Unity Connection	Forwarding Fax Messages from Cisco Unity Express to Cisco Unity Connection Over Unified Messaging Gateway	To verify that a fax received by Cisco Gateway can be relayed to Cisco Unity Express through Cisco Fax Server. And to verify that the Cisco Unity Express subscriber can forward the message to Unity Connection over Unified Messaging Gateway (VPIM) and Unity Connection subscriber can in turn forward the message back to Cisco Unity Express subscriber over a VPIM network with Unified Messaging Gateway in between.	Fax -->PSTN -->IOS Gateway -->CFS -->Cisco Unity Express <-->VPIM <-->UMG <-->VPIM <-->Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF.UNC.302	Cisco Unity Express, Unified Messaging Gateway, Unity Connection	Connection Subscriber Replying to a Fax Message From Cisco Unity Express to Unity Connection Over Unified Messaging Gateway	To verify that a fax received by Unified Messaging Gateway can be relayed to Cisco Unity Express through Cisco Fax Server. And to verify that the Cisco Unity Express subscriber can forward the message to Unity Connection over Unified messaging Gateway (VPIM) and Unity Connection subscriber can in turn reply back to that message.	Fax -->PSTN -->IOS Gateway -->CFS -->Cisco Unity Express <-->VPIM <-->UMG <-->VPIM <-->Unity Connection	Passed	
UCS712IF.UNC.304	Unity Connection	Forwarding a Fax to a System Distribution List in Unified Messaging Gateway	To verify that fax message forwarded to SDL is received by each member in the SDL.	Fax -->PSTN -->IOS Gateway -->CFS -->Cisco Unity Express <-->VPIM <-->UMG <-->VPIM <-->Unity Connection	Passed	
UCS712IF.UNC.304	Messaging Gateway, Unity Connection, Unity Express	Forwarding a Fax to a System Distribution List in Unified Messaging Gateway	To verify that fax message forwarded to SDL is received by each member in the SDL.	Fax -->PSTN -->IOS Gateway -->CFS -->CUE <-->VPIM <-->UMG <-->VPIM <-->Unity Connection	Passed	
UCS712IF.UNC.501	Unity Connection	Fax Call Forwarded to Unity Connection in Connect First Mode-Single Number Fax	To verify that fax message is successfully delivered to the subscriber when the call is forwarded to Unity Connection. The gateway is configured in Connection-first fax detection mode.	Fax -->PSTN -->IOS Gateway -->SMTP -->Unity Connection	Passed	
UCS712IF.UNC.501	Unity Connection	Fax Call Forwarded to Unity Connection in Connect First Mode - Single Number Fax	To verify that fax message is successfully delivered to the subscriber when the call is forwarded to Unity Connection. The gateway is configured in Connection-first fax detection mode.	Fax -->PSTN -->IOS Gateway -->SMTP -->Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF.UNC.504	Unity Connection	Fax Call Disconnect by User-Single Number Fax Connect First Mode	To verify if the fax message transmission to the subscriber is canceled when the call is answered by the phone and is disconnected.	Fax -->PSTN -->IOS Gateway -->SMTP -->Unity Connection	Passed	
UCS712IF.UNC.504	Unity Connection	Fax Call Disconnect by User- Single Number Fax Connect First Mode	To verify if the fax message is successfully delivered to the subscriber when the call is answered by the phone and is disconnected.	Fax -->PSTN -->IOS Gateway -->SMTP -->Unity Connection	Passed	
UCS712IF.UNC.505	Unity Connection	Single Number Fax-Listen First Mode Fax call	To verify if the fax message is successfully delivered to the subscriber when the call is answered by the phone and is disconnected.	Fax -->PSTN -->IOS Gateway -->SMTP -->Unity Connection	Passed	
UCS712IF.UNC.505	Unity Connection	Single Number Fax - Listen First Mode - Fax Call	To verify if the fax message is successfully delivered to the subscriber when the call is answered by the phone and is disconnected.	Fax -->PSTN -->IOS Gateway -->SMTP -->Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
OXN51.CUE.002	Voice Mail	Calls Through MGCP Gateway Forwarded to Cisco Unity Express Integrated to Unified CM	Verify that forwarded calls to Cisco Unity Express from PSTN through a MGCP gateway is successful.		Passed	
OXN51.CUE.003	Voice Mail	Calls Through a H.323 Gateway Forwarded to Cisco Unity Express Integrated to Unified CM	Verify that forwarded calls to Cisco Unity Express from PSTN through a H.323 gateway is successful.		Passed	
OXN51.CUE.008	Voice Mail	Attended Call Transfer Using REFER With Cisco Unity Express	Verify that Cisco Unity Express can successfully transfer (attended) a call using REFER registered to a Unified SRST router. Codec type of the call is sip-notify.		Passed	
UC802EF.CUE.001	Cisco Unity Express	Depositing a Voicemail from DPNSS PBX Phone	Verify the ability of depositing a VM from a DPNSS PBX Phone over QSIG ICT to the remote phone which has been integrated with NME-Cisco Unity Express.	DPNSS PBX Ph1->Unified CM1->QSIG ICT->Unified CM2->Rem SCCP Ph1->CFNA->NME-CUE	Passed	
UC802EF.CUE.002	Cisco Unity Express	Configure Multiple Greetings Through CLI in Cisco Unity Express	Verify the multiple greetings like busy, Closed, Internal, Vacation, Meeting and Extended absence greetings through CLI in Cisco Unity Express can be configured and then make a call from QSIG PBX Phone to Unified CME SIP Phone which has CFNA to Cisco Unity Express voicemail.	PBX Ph1->QSIG PBX->Unified CM->SIP Trunk->Unified SIP Proxy->CME->SIP Ph1-> CFNA->CUE	Failed	CSCtf47467

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CUE.003	Cisco Unity Express	Depositing Voicemail to the Unified CME Phones Over Unified SIP Proxy from a Unified CM IP Phones 8961/9951/9971	Verify that the Unified IP Phones 8961/9951/9971 are able to successfully deposit a voice mail to the Unified CME phone over Unified SIP Proxy.	RT Pro/Biz/Std + Ph1 ->Unified CM->SIP Trunk->Unified SIP Proxy->CME->Ph2 ->CFNA->CUE	Failed	CSCtf47467
UC802EF.CUE.004	Cisco Unity Express	Depositing Voicemail to Unified CME SIP Phones over Unified SIP Proxy from a PBX Phone	Verify that PBX phones are able to successfully deposit a voice mail to the Unified CME SIP Phones over Unified SIP Proxy.	PBX Ph1->PBX->VG30D->MGCP GW->Unified CM->SIP Trunk->Unified SIP Proxy->CME->SIP Ph1 (First Line)->CFNA->SIP Ph1 (Second Line)->CFNA->CUE	Failed	CSCtf47467
UC802EF.CUE.005	Cisco Unity Express	SRSV Functionality	Verify that when the subscribers in remote sites are able to deposit a voicemail using the local Cisco Unity Express module attached to the remote router in Unified SRST mode when the Unified CM goes down. When the Unified CM is up, the remote subscribers should be able to deposit voicemail using the central site's Cisco Unity and in Unified SRST mode the remote subscribers should be able to use the local Cisco Unity Express module in SRST router.	Rem Unified IP Phones 6921/6941/6961/Pro/Biz/Std + Ph 1->SRST1/PSTN gateway->PSTN->SRST1/PSTN gateway->Rem Unified IP Phones 6921/6941/6961/Pro/Biz/Std + Ph 2->CFNA->CUE	Failed	CSCtf47467
UC802IF.CUE.100	Cisco Unity Express Visual Voicemail	Cisco Unity Express Subscriber Listens to Message via VoiceView Express	Verifies whether the user can listen to voicemail messages via VoiceView xpress feature of Cisco Unity Express.	CUE Subscriber --->VoiceView Express --->Message	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CUE.101	Cisco Unity Express Visual Voicemail	Cisco Unity Express Subscriber Sends Message via VoiceView Express	Verify whether Cisco Unity Express subscriber can send message to another subscriber via voice view express.	CUE Subscriber --->VoiceView Express --->Message	Passed	
UC802IF.CUE.102	Cisco Unity Express Visual Voicemail	Cisco Unity Express Subscriber Sends Broadcast Message	Verify whether Cisco Unity Express subscriber can send broadcast message via VoiceView Express.	CUE Subscriber --->VoiceView Express --->Broad cast Message	Passed	
UC802IF.CUE.103	Cisco Unity Express Visual Voicemail	Forward Fax Message to Blind Address Over VPIM With Recorded Greeting	Verifies whether voicemail express view shows fax attachment and play recorded greeting.	CUE Subscriber --->VoiceView Express --->Fax Message	Passed	
UC802IF.CUE.751	Cisco Unity Express-SRSV	SRSV-Unified Messaging Gateway Interaction With Active-Active Cisco Unity Connection	Verify that Survivable Remote Site Voicemail feature supports Cisco Unity Connection in Active-Active deployment.		Failed	CSCtf54209
UC802IF.CUE.752	Cisco Unity Express-SRSV	Using Multiple SRSV Unified Messaging Gateway to Support Multiple SRSV-Cisco Unity Express	Verify that multiple SRSV-Unified Messaging Gateway can be used with one Unified CM cluster to provision multiple SRSV-Cisco Unity Express locations where each SRSV-UMG is assigned with a subset of available SRSV-Cisco Unity Express.	IP Phone->SRST->SIP->Cisco Unity Express; IP Phone->Unified CM->SCCP/SIP->Connection	Passed	
UC802IF.CUE.753	Cisco Unity Express-SRSV	SRSV Extension List for Users in Different Partition and CSS	To verify and validate that CSS and partition settings does not impact SRSV functionality and the configured restrictions in Cisco Unity Connection do not get ported in SRSV.	IP Phone->SRST->SIP->Cisco Unity Express; IP Phone->Unified CM->SCCP/SIP->Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CUE.754	Cisco Unity Express-SRSV	Synchronization When Secondary Cisco Unity Connection Server Acts as Primary After WAN Link Restoration	Verify when Cisco Unity Connection is deployed in Active-Active mode, SRSV-Unified Messaging Gateway is able to interact with voicemail servers when they are in both primary and secondary mode.	IP Phone->SRST->SIP->Cisco Unity Express; IP Phone->Unified CM->SCCP/SIP->Connection1	Passed	
UC802IF.CUE.755	Cisco Unity Express-SRSV	Voicemail Upload Order After WAN Link Restoration From Cisco Unity Express to Cisco Unity Connection	Verify that voicemail order and state is maintained when voicemails are uploaded to Cisco Unity Connection.		Passed	
UC802IF.CUE.756	Cisco Unity Express-SRSV	Provisioning and Functioning for Unified SRST with SIP Phones	Verify that SRSV feature can operate seamlessly in SIP Unified SRST deployment.	SIP Phone->SRST->SIP->Cisco Unity Express; SIP Phone->Unified CM->Connection	Passed w/ Exception	CSCta76151, CSCta78369
UC802IF.CUE.757	Cisco Unity Express-SRSV	SRSV-Unified Messaging Gateway and SRSV-Cisco Unity Express Monitoring Using Unified Operations Manager	Verify that SRSV-UMG and SRSV-CUE can be monitored using Unified Operations Manager		Passed	
UC802IF.CUE.758	Cisco Unity Express-SRSV	Voicemail Deposit, Retrieval, Forwarding and MWI Status When in Fallback Mode	Verify that voicemail features are available when in fallback mode and MWI status is updated reflecting the current state.	IP Phone->SRST->SIP->Cisco Unity Express; IP Phone->Unified CM->SCCP/SIP->Connection1	Passed	
UC802IF.CUE.759	Cisco Unity Express-SRSV	SRSV-Unified Messaging Gateway Support for Multiple Unified SRST Locations	Verify that one SRSV-Unified Messaging Gateway can be used to support two Unified SRST-Cisco Unity Express deployment.	IP Phone->SRST->SIP->Cisco Unity Express	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CUE.761	Cisco Unity Express-SRSV	Alternate Greeting With End Date Reverting Back to Standard Greeting	Verify that alternate greeting is in effect after the fallback and SRSV-Cisco Unity Express switches automatically to standard greeting at the configured end date and time (as configured in Cisco Unity Connection).	IP Phone->SRST->SIP->Cisco Unity Express	Passed	
UC802IF.CUE.762	Cisco Unity Express-SRSV	Custom Holiday Schedule and Holiday Greeting	Verify that Holiday Greeting is played by SRSV-Cisco Unity Express when in fallback mode.	IP Phone->SRST->SIP->Cisco Unity Express	Passed	
UC802IF.CUE.763	Cisco Unity Express-SRSV	Cisco Unity Connection Call Action Set to "Hang Up"	Verify if SRSV-Cisco Unity Express hangs up after the greeting is played.	IP Phone->SRST->SIP->Cisco Unity Express	Passed	
UC802IF.CUE.765	Cisco Unity Express-SRSV	PSTN Caller Leaving a Voicemail in SRSV-Cisco Unity Express	Verify that PSTN callers can leave a voicemail in SRSV-Cisco Unity Express without any additional configuration.	PSTN Phone->SRST->SIP->SRSV-Cisco Unity Express	Passed	
UC802IF.CUE.766	Cisco Unity Express-SRSV	Addressing Messages to Distribution List Containing Members from Digitally Networked Cisco Unity Connection	Verify that messages can be addressed to distribution lists from SRSV-Cisco Unity Express and they are successfully delivered on WAN link restoration.	IP Phone->SRST->SIP->Cisco Unity Express; IP Phone->Unified CM->SCCP/SIP->Connection	Passed	CSCtf34050
UC802IF.CUE.767	Cisco Unity Express-SRSV	Addressing Messages to Subscriber From Auto Attendant	Verify that calls can be routed to Auto Attendant in SRSV-Cisco Unity Express and caller can lookup a subscriber by dialing the extension.	PSTN Phone->SRST->SIP->SRSV-CUE	Failed	CSCtf01694
UC802IF.CUE.806	Unified SRST	Accessing DN Voicemail on Cisco Unity Express	To verify if a phone configured with + prefix can access voicemail.	IP Phone --- WAN Link --- Unified CM --- Fail WAN Link ---- IP Phone --- SRST ----Cisco Unity Express	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
CO10.VID.005	Video	SIP Non-Video to SCCP Video Call Transfer to SCCP Video Endpoint Inter-Cluster SIP Trunk	To verify the call transfer from SIP (TNP) non-video endpoint, transferring a SCCP Video endpoint to another SCCP Video capable endpoint across inter-cluster SIP Trunk.		Passed	
CO10.VID.025	Video	IP Communicator and CVTA With Mid-Call Video Inter-Cluster SIP Trunk	To verify if IP Communicator with CVTA endpoint can call another video endpoint while CVTA application is down. After call connects CVTA application is started and mid-call video is established through inter-cluster SIP Trunk.		Passed	
OXN51.VID.003	Video	Unified Video Advantage and IP Communicator Locations Based CAC and Retry Video Call as Audio and Consultative Transfer Intracluster	To verify Unified Video Advantage and IP Communicator on the same laptop or PC calls to other Video endpoint when there is not enough bandwidth available. After call connects audio only call is transferred consultatively to another intracluster Video endpoint where there is enough Video bandwidth. During the transfer MOH is available and the first video call is retried audio only.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
OXN51.VID.005	Video	Unified Video Advantage and IP Communicator Mid-Call Video Intracluster with Audio Conference	To verify that audio only endpoint calls to Unified Video Advantage and IP Communicator on the same laptop or PC creates an audio conference by inviting another intracluster Video endpoint. After the conference call is connected audio only conference initiator hangs up and conference ends. Call is then connected between 2 video endpoints with Video.		Passed	
OXN51.VID.010	Conference, Dial Plan, ICT Call flow, Video	Unified Video Advantage Endpoint Intercluster SIP Trunk	To verify if Unified Video Advantage and IP Phone 7971 endpoint can create an adhoc video conference on MCU 5.X SCCP resources by inviting two Windows Unified Personal Communicator endpoints in to a meeting across intercluster SIP Trunks.		Passed	
OXN51.VID.021	Video	Unified Video Advantage Endpoint Call Transfer From Non-Video to Another Unified Video Advantage Endpoint	To verify if a Unified Video Advantage endpoint behind IP Phone 7960 fails over to Unified SRST mode and gets transferred from audio call to another Unified Video Advantage endpoint. Audio call is upgraded to a Video call after the transfer is complete.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701EF.VID.005	Adhoc Conference	Adhoc Conference Using Continuous Presence and Transrating on MCU 3545	Verify if a user can make an adhoc conference call on MCU 3545 using SCCP and enable continuous presence.	Stage1:SCCP Video Ph1->Unified CM->Rem SCCP Video ph2 Stage2:Rem SCCP Video Ph2->Unified CM->3545->SCCP Video Ph3 Stage3:Rem SCCP Video Ph2->Unified CM->3545->SCCP Video Ph4	Passed	
UC701EF.VID.006	Video	Video Activation During an Audio Call in Progress Between H.320 Device and IP Phone with Unified Video Advantage	Verify if video can be activated on Unified Video Advantage while audio call is already setup between the IP Phone and H.320 device.		Passed	
UC701EF.VID.007	Video	Video Deactivation During an Audio Call in Progress Between H.320 Device and IP Phone with Unified Video Advantage	Verify if video can be deactivated on Unified Video Advantage while Video call is already setup between the IP Phone and H.320 device.	Stage 1:H.320->3545(GW)->Unified CM->SCCP Ph1 (CUVA)	Passed	
UC701EF.VID.008	Reservational Conference	Reservationless Videoconference on 3545 through H.323 Interface	Verify if a user can make reservationless videoconference using an IPVC 3545 through H.323 interface and calling users are H.320 terminal, Video endpoints supporting H.323 and SCCP.	Stage1: H.320->3526 (GW)->Unified CM->3545 Stage2: H.323 Video Ph1->Unified CM->3545 Stage3: Rem SCCP Video Ph2->Unified CM->3545	Passed	
UC701EF.VID.009	Reservationless Conference	Reservationless Videoconference Through SIP Interface	Verify if a user can make reservationless videoconference using an MCU 3545 through SIP interface and calling user are H.320 terminal, Video endpoints supporting H.323 and SCCP.	Stage1: H.320->3527(GW)->Unified CM->3545 Stage2: H.323 Ph1->Unified CM->3545 Stage3: Rem SCCP Ph1->Unified CM->3545	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701EF.VID.010	Basic Video call	Video Call from SCCP Sony Video Phone to H.320 Video Terminal Through 3545	Verify if a user can make a video call from SCCP Sony video phone to H.320 video terminal which is connected through an IPVC 3545.	SCCP Sony Video Ph1->Unified CM->3545 (GW)->H.320 terminal	Passed	
UC701EF.VID.013	IPVC	H.323 Endpoints Calling H.320 Endpoint	Verify fi a user can make a video call from H.323 endpoint registered to the gatekeeper to H.320 endpoint through IPVC 3545.	H.323 Video Ph1->GateKeeper->Unified CM->IPVC 3545->PSTN->H.320 terminal	Passed	
UC701EF.VID.016	Adhoc Video Conference	SCCP Adhoc Video Conference through IPVC 3545 Using H.263	Verify if a user can make adhoc Video conference through IPVC 3545 using H.263.	Stage1: SCCP Video Ph1->Unified CM->SCCP Video Ph2 Stage2: SCCP Video Ph2->Unified CM->3545->CNF_AD->Unified CM->SCCP Video Ph3	Passed	
UC701EF.VID.018	Adhoc video Conference	SCCP Adhoc Video Conference using Transcoding on 3545	Verify if a user can make an adhoc conference call through 3545 between users using different codec.	Stage1: SCCP Video Ph1->Unified CM->SCCP Video Ph2; Stage2: SCCP Video Ph2->Unified CM->3545->CNF_AD->Unified CM->SCCP Video Ph3	Passed	
UC701IF.VID.101	Video	Video Call to H.323 Third Party Video Endpoint Using RAS Aggregator Trunk	To verify if a user can place call from a SCCP video endpoint to third party H.323 video endpoint using RAS aggregator trunk. Third party endpoint is defined on Unified CM by name, not IP address, which allows the endpoint to use DHCP for IP addressing.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC702EF.VID.030	Video	Reservationless Conference Using Cisco 3540 Involving H.320 Endpoints and Remote 3rd Party Video SCCP Endpoint	Verify the reservationless video conference call using Cisco 3540 involving H.320 endpoint and Remote 3rd party SCCP video phone.	Stage1: H.320 Video Ph1->PSTN->Unified CM->3540; Stage 2: 3rd Party Video SCCP Ph2->Unified CM->PSTN->Unified CM->3540; Stage 3: SCCP Video Ph3->Unified CM->PSTN->Unified CM->3540	Passed	CSCtd03018
UC702EF.VID.031	Reservationless Video Conference	Reservationless Videoconference through H.323 using IPVC 3515 with Unified Video Advantage Phone	Verify if a user can make a reservationless videoconference through H.323 using IPVC 3515. The calling users are Unified Video Advantage phones and SCCP Video Phones.	Stage1: SCCP Ph1 (CUVA)->Unified CM->3515 Stage2: Rem SCCP Video Ph2->Unified CM->3515 Stage3: Rem SCCP Video Ph3 (CUVA)->Unified CM->3515	Passed	
UC702EF.VID.032	Basic Video Call	Video Call from H.320 to SCCP with Unified Video Advantage through IPVC 3522	Verify if a user can make a video call using IPVC 3522 from a remote H.320 video endpoint to another remote site SCCP Phone with Unified Video Advantage enabled.	H.320 Video Ph1->IPVC 3522->PSTN->Unified CM->SCCP Ph2 (CUVA)	Passed	
UC702EF.VID.033	Adhoc Video Conference	Adhoc Video Conference Using IPVC MCU 3511 with H.323 and Unified Video Advantage Endpoints	Verify if a user can make adhoc Video conference using an IPVC MCU 3511 involving H.323 endpoint and Unified Video Advantage.	Stage1: SCCP Video ph1->Unified CM->H.323 Video ph2 Stage2: H.323 Video ph2->Unified CM->3511(MCU)->Unified CM->SCCP ph3 (CUVA)	Passed	
UC702EF.VID.035	Video	H.320 Endpoints Calling IP communicator with Unified Video Advantage Endpoint through IPVC 3521 and Transfer to 3rd party SCCP Video Phone	Verify if a user can make a video call from H.320 endpoint to IP communicator with Unified Video Advantage through IPVC 3521 and transfer the call to 3rd Party SCCP Video Phone.	Stage1:H.320 Video Ph1->IPVC 3521->Unified CM->IP Communicator(CUVA)->XFER->3rd Party SCCP Ph3	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC702EF.VID.036	Video	SCCP Video Endpoints Calling SCCP Phone and Transfer to H.320 Terminal Through IPVC 3521	Verify if a user can make an audio call from SCCP Video phone to SCCP Phone and transfer the call to H.320 terminal through IPVC 3521.	Stage1:SCCP Video Ph1->Unified CM->SCCP Ph2->Unified CM->IPVC 3521->H.320 terminal	Passed	
UC702EF.VID.037	IPVC	Video Call Between H.323 Endpoints	Verify if a user can make a video call from a Remote H.323 endpoint registered to the gatekeeper to another H.323 endpoint in Central site.	H.323 Video Ph1->GateKeeper1->Unified CM->GateKeeper1->Rem H.323 Video Ph2	Passed	
UC702EF.VID.038	IPVC	PSTN Video Call Between Unified Video Advantage Phones in Central and Remote Sites Through IPVC 3527	Verify if a user can make a PSTN video call from a Remote SCCP phone enabled with Unified Video Advantage to another SCCP Phone in central enabled with Unified Video Advantage through IPVC 3527.	SCCP Ph1 (CUVA)->IPVC 3527->Unified CM->SCCP Ph2 (CUVA)	Passed	
UC702EF.VID.039	IPVC	Video Call Between H.320 Endpoint to IP Communicator with Unified Video Advantage Through IPVC 3526	Verify if a user can make a video call from a H.320 endpoint to IP Communicator with Unified Video Advantage enabled in central site through IPVC 3526.	H.320 endpoint->IPVC 3526->Unified CM->IP Communicator (CUVA)	Passed	
UC702EF.VID.040	Adhoc Video Conference	Adhoc Video Conference Using IPVC 3515 with SCCP and IP Communicator Endpoints	Verify if a user can make adhoc Video conference using an IPVC 3515 involving SCCP and IP Communicator endpoints.	Stage1: SCCP Video ph1->Unified CM->H.323 Video ph2 Stage2: H.323 Video ph2->Unified CM->3515(MCU)->Unified CM->IP Communicator (CUVA)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC702EF.VID.041	Transrating	Continuous Presence and Transrating on MCU 3545 using H.323 Interface Involving 3rd party Video Phone	Verify if a user can make reservationless conference using MCU 3545 and enable continuous presence involving 3rd Party Video Phone.	Stage1: SCCP Video Ph1->Unified CM->3545; Stage2: Rem SCCP Video Ph2->Unified CM->3545; Stage3: H.323 Video Ph3->Unified CM->3545; Stage4: 3rd Party SCCP Video Ph4->Unified CM->3545	Passed	
UC702EF.VID.042	Transrating	Continuous Presence and Transrating on MCU 3545 Using SIP Interface	Verify if a user can make reservationless conference using 3545 and enable continuous presence.	Stage1: SCCP Video Ph1 & Rem SCCP Video Ph2->Unified CM->3545; Stage2: H.323 Video Ph3 & 3rd Party SCCP Video Ph4->Unified CM->3545	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.UUE.001	Voicemail	Reply to Voicemail with Call or IM or Voicemail	Verify if Unified IP Phone 9971 receives a voice mail from SIP phone and can reply through voicemail, or IM or call.	stage 1: SIP Ph 1->Unified CM->9971 Ph->Voice Mail;Stage 2:9971 Ph->Unified CM->IM->SIP Ph1 ; Stage 3 :9971 Ph->Unified CM->SIP ph1;Stage 4:9971 Ph->Unified CM->SIP Ph 1->Voice mail.	Passed	
UC802EF.UUE.002	Voicemail	Display of Text Transcript of Voicemail in Context of Visual Voicemail	Verify if Unified Personal Communicator receives a voice mail from the SCCP Phone and if the text transcript of voice mail is available in the context of Visual voice mail.	SCCP Ph 1->Unified CM->CUPC 1->Voice mail	Passed w/ Exception	Text Transcript feature is not supported in VVM for UC 8.0.
UC802EF.UUE.003	Visual Voicemail	Inline Playback of Visual Voicemail	Verify when Unified Personal Communicator receives a voice mail from the SIP Phone, toast pop up of visual voice mail indication is received and playback of visual voice mail is done with fast forward and rewind options.	SIP Ph 1->Unified CM-> CUPC 1->Voice mail	Passed w/ Exception	Text Transcript feature is not supported in VVM for UC 8.0.
UC802EF.UUE.004	Presence	Reply to Voicemail with Call or IM or Voicemail	Verify if Unified Personal Communicator receives a voice mail from SIP phone and can reply through Instant messaging or calling or through voicemail.	stage 1: SIP Ph 1->Unified CM->CUPC 1->Voice Mail;Stage 2:CUPC 1->Unified CM->IM->SIP Ph1 ; Stage 3 :CUPC 1->Unified CM->SIP ph1;Stage 4:CUPC 1->Unified CM->SIP Ph 1->Voice mail.	Passed w/ Exception	CSCtd33303
UC802EF.UUE.005	Presence	Real-time Presence Status in Unified Personal Communicator	Verify when Unified Personal communicator is idle and when an incoming call from SIP phone is answered in, the presence status changes from idle to Busy.	SIP Ph->Unified CM-> CUPC	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.UUE.006	Buddy Lists and Contact	Realtime Presence Status in Unified IP Phone 9971	Verify when a Unified IP Phone 9971 presence status is idle and when an incoming call from SIP phone is answered, the presence status changes from idle to Busy.	SIP Ph->Unified CM->9971 Ph	Passed	
UC802EF.UUE.007	Buddy Lists and Contact	Click to Add Add Participant to Voice Session in Unified Personal Communicator	Verify if Unified Personal Communicator can add a participant from contact list to conference with active calls.	Stage 1: SIP Ph 1->Unified CM->CUPC ;Stage 2 : CUPC->Unified CM->Conference->SIP Ph 2.	Passed w/ Exception	"Click to Add" participants to IM sessions or call is not available for UC 8.0. Only Click to add into contacts is available.
UC802EF.UUE.008	Buddy Lists and Contact	Click to Add Add Participant to Voice Session in IP Phone 9971	Verify if Unified IP Phone 9971 can add a participant from contact list to conference with active calls.	Stage 1: SIP Ph 1->Unified CM->9971 Ph ;Stage 2 : 9971 Ph->Unified CM->Conference->SIP Ph 2.	Passed w/ Exception	"Click to Add" participants to IM sessions or call is not available for UC 8.0. Only Click to add into contacts is available.
UC802IF.MID.001	Unified IP Phone	Visual Voicemail Installation	To verify the end-user installation guide and verify initial Visual Voicemail installation.	IP Phone->Unified CM; IP Phone->web server	Passed	
UC802IF.MID.002	Unified IP Phone	Version Upgrade	Verify process of upgrading a Visual Voicemail IP Phone Service.	IP Phone->Unified CM; IP Phone->web server	Passed	
UC802IF.MID.003	Unified IP Phone	Version Downgrade	Verify process of downgrading a Visual Voicemail IP Phone Service.	IP Phone->Unified CM; IP Phone->web server	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.MID.004	Unified IP Phone	Incoming Call with Visual Voicemail Running	Verify the ability to receive an incoming call while a Visual Voicemail is running.	IP Phone->Unified CM; IP Phone->web server	Passed	
UC802IF.MID.005	Unified IP Phone	General Service Provisioning	Verify the ability to subscribe/unsubscribe the Visual Voicemail and enable/disable the Visual Voicemail service.	Unified CM->IP Phone	Passed	
UC802IF.MID.006	Unified IP Phone	Weather Application Ease-of-Use/Basic Function	Verify general ease-of-use and functionality of Weather Forecast Visual Voicemail.	IP Phone->Web Proxy->WAN	Passed	
UC802IF.MID.007	Unified IP Phone	Stock Application Ease-of-Use/Basic Function	Verify general ease-of-use and functionality of the Stocks Update Visual Voicemail.	IP Phone->Web Proxy->WAN	Passed	
UC802IF.MID.008	Unified IP Phone	Calculator Ease-of-Use/Basic Function	Verify general ease-of-use and functionality of the Quick Calculator Visual Voicemail.	IP Phone	Passed	
UC802IF.MID.009	Unified IP Phone	Visual Voicemail Download Behavior with IP Phone Service Version	Verify the download behavior of a Visual Voicemail depending on IP phone service version.	IP Phone->Unified CM; IP Phone->web server	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.WIR.102		LAG Link Failure	verify physical link failure with atleast two links configured in LAG mode.		Passed	
SR60.WIR.105	Wireless	Unified IP Phone 7925 L3 Roaming	Verify the Layer 3 Unified IP Phone 7925 roaming between two WLAN controllers in different subnets.		Passed	
SR60.WIR.111		IP Communicator/Unified Video Advantage/Data Across Wireless Connection	Verify and test wireless data PC with voice and video applications.		Passed	
UC802IF.WIR.001	Wireless	Wireless IP Phone 9971 WMM SIP Snooping	Verify SIP snooping support on Unified IP Phone 9971 WiFi.	9971->AP->WLAN controller->Unified CM->IP phone	Passed	
UC802IF.WIR.010	Wireless	H-REAP/Unified SRST remote site, Flapping WAN link	Verify to establish wireless connections from devices at remote sites and force the remote sites into Unified SRST mode.	7925->AP->GW->Unified CM->IP phone	Passed	
UC802IF.WIR.011	Wireless	Redundant WLAN Controllers in Active-Active Mode	Verify the active/active WLAN controller redundancy.	Wireless phone->AP->WLAN controller->Unified CM->ip phone	Passed	
UCS713-WIR-002	Wireless Unified IP Phone 9971	WiFi connected Unified IP Phone 9971 Remote Site (WAN Failure)	Verify the Wan failure on Unified SRST/H-REAP.	Phone->AP->LWAPP/WAN->WLAN controller->Unified CM	Passed	
UCS713-WIR-002	Wireless	WiFi Connected Unified IP Phones in Remote Site	Verify H-REAP Access Point failure in remote Unified SRST site.	IP Phone->Access Point->LWAPP/WAN->Wireless LAN Controller->Unified Communications Manager	Passed	
UCS713-WIR-003	Wireless Unified IP Phone 9971	WiFi Unified IP Phone 9971 Central Site	Verify forced roam due to Access Point failure.	Phone->AP->WLAN controller->Unified CM	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS713-WIR-008	Unified IP Phone 9971 (WiFi Connected)	WiFi Connected Unified IP Phone 9971 Central Site (Authentication)	Verify that Unified IP Phone 9971 in central site can perform TKIP/AES authentication with CCKM.	Phone->AP->LWAPP/WAN->WLAN controller->Unified CM	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
CO10.CME.002	Feature termination, Tandem	Hold and Resume in Unified CME SIP IP phone for Call Connected to Unified CM SCCP IP phone over H.225 Trunk	To verify hold and resume features in Unified CME IP Phones over H.225 trunks.		Passed	
CO10.CME.007	Feature termination, Tandem	IOS Hardware Transcoding for Calls from Unified CME to Unified CM Sent to Cisco Unity Express Voice Mail	Verify the ability to invoke IOS hardware transcoding for calls from Unified CME to Unified CM, and verify that calls requiring dissimilar codecs are sent to voice mail in Cisco Unity Express.		Passed	
CO10.CME.016	Feature termination, Tandem working	Call Between SIP and H.323 Site via IP-to-IP Gateway Forwarded to Unified CM via H.323/SIP ICT	Verify the ability to make a successful call from a SIP site to a H.323 site over an IP-to-IP Gateway, which is then forwarded to Cisco Unified CM via either a SIP or H.323 inter-cluster trunk (ICT).		Passed	
CO10.UCA.042	Unified Personal Communicator	Unified Personal Communicator in Softphone Mode, Merge an Incoming Video Call	Verify the following: Set up a video call from video IP Phone 7985 to Unified Personal Communicator in softphone mode. Set up an out going call from same Unified Personal Communicator to an IP communicator. Check for bi-directional voice path. Merge both the calls.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
CO10.UCA.043	Unified Personal Communicator	Unified Personal Communicator in Softphone Mode Merge an Outgoing Video Call to SCCP Video Endpoint	Verify the following: Bring up Unified Personal Communicator in softphone mode. Establish an outgoing video call to SCCP end point. Check for audio and video of the call. Set up an incoming video call from a H.323 video end point through ICT, and answer the call. Verify that the first call is fed with Music On Hold and no Video. Click on the Merge button and establish a 3 way conference call. Check for audio and video of this conference call.		Passed	
CO10.VID.005	Video	SIP Non-Video to SCCP Video Call Transfer to SCCP Video Endpoint Inter-Cluster SIP Trunk	To verify the call transfer from SIP (TNP) non-video endpoint, transferring a SCCP Video endpoint to another SCCP Video capable endpoint across inter-cluster SIP Trunk.		Passed	
CO10.VID.025	Video	IP Communicator and CVTA With Mid-Call Video Inter-Cluster SIP Trunk	To verify if IP Communicator with CVTA endpoint can call another video endpoint while CVTA application is down. After call connects CVTA application is started and mid-call video is established through inter-cluster SIP Trunk.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
GB42.CRS.015.80	Regular Agent	Call to Unified CCX from MGCP Gateway to CAD Desktop Agent IP Phone 7970/71; Agent Action=Touch Tones - Transfer/Conference	Verify communication between various devices of a Unified CCX system across a secured infrastructure in an effort to identify any incompatibilities between the configuration of the security infrastructure and the normal message flow of the Unified CCX system. Various agent and supervisor actions in different locations and with different equipments are used.		Passed	
GB42.CRS.020.13	Regular Agent	Originate call to Unified CCX from GK-ICT Trunk to IPPA agent phone 7970/71, agent action Transfer - Blind	The objective of these tests is to generate communication between various devices of a Unified CCX system across a secured infrastructure in an effort to identify any incompatibilities between the configuration of the security infrastructure and the normal message flow of the Unified CCX system. Various agent and supervisor actions in different locations and with different equipment are used.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
GB42.CRS.030.1	Supervisor Agent	Supervisor for Agent, Barge-In	Verify communication between various devices of Unified CCX system across a secured infrastructure in an effort to identify any incompatibilities between the configuration of the security infrastructure and the normal message flow of the Unified CCX system. Various agent and supervisor actions in different locations and with different equipment are used.		Passed	
OXN51.CUE.002	Voice Mail	Calls Through MGCP Gateway Forwarded to Cisco Unity Express Integrated to Unified CM	Verify that forwarded calls to Cisco Unity Express from PSTN through a MGCP gateway is successful.		Passed	
OXN51.CUE.003	Voice Mail	Calls Through a H.323 Gateway Forwarded to Cisco Unity Express Integrated to Unified CM	Verify that forwarded calls to Cisco Unity Express from PSTN through a H.323 gateway is successful.		Passed	
OXN51.CUE.008	Voice Mail	Attended Call Transfer Using REFER With Cisco Unity Express	Verify that Cisco Unity Express can successfully transfer (attended) a call using REFER registered to a Unified SRST router. Codec type of the call is sip-notify.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
OXN51.IPP.006	Unified Personal Communicator	Verify Five Way Web Collaboration from Unified Personal Communicator on Macintosh	To verify Unified Personal Communicator running on a MAC initiates a 5 way conference call involving Unified Personal Communicator running on a windows platform and another MAC-Unified Personal Communicator end point, all in same cluster. Web Collaboration is initiated afterwards.		Passed	
OXN51.VID.003	Video	Unified Video Advantage and IP Communicator Locations Based CAC and Retry Video Call as Audio and Consultative Transfer Intracluster	To verify Unified Video Advantage and IP Communicator on the same laptop or PC calls to other Video endpoint when there is not enough bandwidth available. After call connects audio only call is transferred consultatively to another intracluster Video endpoint where there is enough Video bandwidth. During the transfer MOH is available and the first video call is retried audio only.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
OXN51.VID.005	Video	Unified Video Advantage and IP Communicator Mid-Call Video Intracluster with Audio Conference	To verify that audio only endpoint calls to Unified Video Advantage and IP Communicator on the same laptop or PC creates an audio conference by inviting another intracluster Video endpoint. After the conference call is connected audio only conference initiator hangs up and conference ends. Call is then connected between 2 video endpoints with Video.		Passed	
OXN51.VID.010	Conference, Dial Plan, ICT Call flow, Video	Unified Video Advantage Endpoint Intercluster SIP Trunk	To verify if Unified Video Advantage and IP Phone 7971 endpoint can create an adhoc video conference on MCU 5.X SCCP resources by inviting two Windows Unified Personal Communicator endpoints in to a meeting across intercluster SIP Trunks.		Passed	
OXN51.VID.021	Video	Unified Video Advantage Endpoint Call Transfer From Non-Video to Another Unified Video Advantage Endpoint	To verify if a Unified Video Advantage endpoint behind IP Phone 7960 fails over to Unified SRST mode and gets transferred from audio call to another Unified Video Advantage endpoint. Audio call is upgraded to a Video call after the transfer is complete.		Passed	
SR60.CCM.080.25	Basic Call Flow	Do Not Disturb	Verifies basic functionality of the Do Not Disturb (DND) feature.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.CCM.410.5	Conference	Shared Line Secure MeetMe Conference	To verify if Secure Conferencing properly works with MeetMe Conference when a shared line has a different security setting on each endpoint.		Passed	
SR60.CCM.603.12	Basic Call Flow	CODEC Support: Phone to Application	To verify calls between phones and applications using different CODECs.		Passed	
SR60.CER.100.2	Cisco Emergency Responder	Track Current Location of IP Phone and E911 Call Routed to Nearest PSAP-Unified IP Phone 7971	verify to ensure that Cisco Emergency Responder can track current location of IP Phones and E911 calls by users getting routed to nearest PSAP.		Passed	
SR60.CER.100.4	Cisco Emergency Responder	Track Current Location of IP Phone and E911 Call Routed to Nearest PSAP-Unified Personal Communicator	Verify to ensure that Cisco Emergency Responder can track current location of IP Phones and E911 calls by users get routed to nearest PSAP.		Passed	
SR60.CER.101	Cisco Emergency Responder	Phone Calls From Unlocated Phone	Verify to ensure if unlocated phones can make 911 calls and their call is routed to default PSAP location configured in Cisco Emergency Responder. A phone is considered unlocated if it is registered with Unified CM, but Emergency Responder has not located it yet. This situation can happen if the phone is behind a switch not defined in Emergency Responder.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.CER.102.1	Cisco Emergency Responder	PSAP Callback E911 Caller Through SIP VoIP Protocol	Verify to ensure that PSAP can call back the E911 caller.		Passed	
SR60.CER.103.3	Cisco Emergency Responder	System Reliability When Single Emergency Responder Server Within the Server Group Fails - Fallback	Verify to ensure there is no single point of failure in the Emergency Responder Server Group.		Passed	
SR60.CER.103.4	Cisco Emergency Responder	System Reliability When Single Emergency Responder Server Within the Server Group Fails - LossOfHeartbeat	Verify to ensure there is no single point of failure in the Emergency Responder Server Group.		Passed	
SR60.CME.008.6	Basic Call Flow	MeetMe Conference	To verify a Meet-Me conference using H/W conference resources within Unified CME.		Passed	
SR60.CME.107.1	Basic Call Flow	Unified CME Support for Out of Dialog	To verify the use of OOD-R from one Unified CME site to remote Unified CME sites.		Passed	
SR60.CME.108.13	Basic Call Flow	Call From a PSTN Phone Through a H.323 Gateway Forwarded to Shared Line	Verify if a call from a PSTN phone to an IP Phone registered to Unified CME can be forwarded to a remote Unified CME phone configured for shared line.		Passed	
SR60.CME.108.5	Basic Call Flow	Hold And Resume the Call Placed on a SIP Network	Verify if a call from a Gatekeeper controlled Unified CME SCCP IP Phone through a IP-to-IP gateway to a Unified CME SIP IP Phone via SIP trunk can be placed on hold and then resumed.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.CRS.097	Unified Contact Center Express	Utilizing Blended CSQ in An Outbound Campaign	Verify to ensure that blended CSQ does not use more agents than allowed for outbound campaign.		Passed	
SR60.CRS.098	Supervisor Agent	Supervisor for Agent, Silent Monitoring	Verify Supervisor can perform silent monitoring of agents enabled for VoIP Monitoring as well as Desktop Monitoring.		Passed	
SR60.CRS.110	Unified Contact Center Express	High Availability	To verify that for the outbound Preview dialer capability: 1. If one node goes down, the Outbound Campaigns are stopped till either the failed node is restored or removed from the Unified CCX cluster. 2. Data integrity for Dialing List (customer records) maintained for all customer records except for records that are being presented or have been accepted by agents at the time of failover.		Passed	
SR60.CUP.007	Interoperability	Instant Messaging Between IP Phone Messenger on Extension Mobility Phone Controlled by Subscriber Unified Presence Server and Unified Personal Communicator Clients	Verify that a traveling user using IP Phone Messenger on an Extension Mobility SIP TNP Cisco Unified IP phone (IP Phone Messenger is controlled by a Subscriber Unified Presence Server) can exchange Instant Messages with a Unified Personal Communicator user.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.IPP.107.3	Unified IP Phone	DTMF Transport to Unified MeetingPlace	Verify to ensure IP Communicator can interwork with Unified Meeting Place, supports RFC2833, inter-cluster call and both wired and wireless LAN: VoiceProto:SIP and LANProto:Wireless.		Passed	
SR60.PPR.020.7	PhoneProxy	Hold and Resume, Music on Hold	Verifies Cisco Unified PhoneProxy based phones to hold and resume calls. Calls include other phones in cluster, gateway calls, and other PhoneProxy phones.		Passed	
SR60.PPR.050.2	PhoneProxy	Standard (Ad-Hoc) Conference	Verifies Cisco Unified PhoneProxy based phones are able to create a standard (ad-hoc) conference using the confirm and join softkeys.		Passed	
SR60.SRST.201.1	Basic Call Flow	Unified SRST Support for Out of Dialog	To verify the use of OOD-R from one Unified SRST site to another Unified SRST site.		Passed	
SR60.SRST.202.2	Basic Call Flow	Consultative Transfer from Unified SRST SIP Network to Unified SRST H.323 Network Involving IP-to-IP Gateway	Verify if a user can call from a Unified SRST SCCP IP Phone through a IP-to-IP gateway to a Cisco Unified SRST SIP IP Phone via SIP trunk. The call transferred (blind) to a SIP IP Phone registered to another Unified SRST Site and transferred back (blind) to the originating SRST site (the same cluster as called party) SCCP IP Phone.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.SRST.202.5	Basic Call Flow	Hold and Resume Where Call is Placed on Hold on Unified SRST SIP Site	Verify if a call from a Unified SRST SCCP IP Phone through a IP-to-IP gateway to a Unified SRST SIP IP Phone via SIP trunk can be placed on hold and resumed.		Passed	
SR60.SRST.202.7	Basic Call Flow	Call Forward Across IP-to-IP Gateway Involving Two Unified SRST Sites	Verify if a call from a Unified SRST SIP IP Phone through an IP-to-IP gateway to a Unified SRST SIP IP Phone via SIP trunk with 'Call Forward All' set on the called party to a SCCP Phone registered to Unified SRST as the called party is successful.		Passed	
SR60.UNC.100.2	Basic Call Flow, Voice Mail	Unity Connection VPIMv2 Standalone, Reply, Private	Verification of Unity Connection VPIM message delivery to multiple Cisco VPIM networked systems. Once a message is delivered from Unity Connection to the VPIM networked system it will be replied to. This will verify that Unity Connection is able to send and receive a VPIM message between supported Cisco VPIM voice messaging systems and process correctly.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.UNC.100.4	Basic Call Flow, Voice Mail	Unity Connection VPIMv2 Standalone, Dist List, Not Private	Verification of Unity Connection VPIM message delivery to multiple Cisco VPIM networked systems. Once a message is delivered from Unity Connection to the VPIM networked system it will be replied to. This will verify that Unity Connection is able to send and receive a VPIM message between supported Cisco VPIM voice messaging systems and process correctly.		Passed	
SR60.UNC.110	Basic Call Flow, Voice Mail	Unity Connection Standalone VPIMv2 NDR	Verify if non-delivery receipts (NDRs) work when addressing two multiple Cisco VPIM networked systems using Cisco Unity Connection Standalone.		Passed	
SR60.UNI.001.1	Voice Mail	Dropped Call Recovery-Caller Hangs Up	To verify that the subscriber can resume a previously dropped call into Cisco Unity back to the place where the subscriber drops the call by hanging up the phone.		Passed	
SR60.UNI.001.13	Voice Mail	Dropped Call Recovery-Caller Hangs Up	To verify that the subscriber can resume a previously dropped call into Cisco Unity back to the place where the subscriber drops the call by hanging up the phone.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.UNI.001.14	Voice Mail	Dropped Call Recovery-Caller Hangs Up	To verify that the subscriber can resume a previously dropped call into Cisco Unity back to the place where the subscriber drops the call by hanging up the phone.		Passed	
SR60.UNI.002.19	Voice Mail	Dropped Call Recovery-Network Failure	To verify that the subscriber is not negatively impacted when calls are dropped due to network failures. Dropped call recovery is not supported with network failures, but the solution components should not encounter any negative impacts.		Passed	
SR60.UNI.002.2	Voice Mail	Dropped Call Recovery-Network Failure	To verify that the subscriber is not negatively impacted when calls are dropped due to network failures. Dropped call recovery is not supported with network failures, but the solution components should not encounter any negative impacts.		Passed	
SR60.UNI.002.7	Voice Mail	Dropped Call Recovery-Network Failure	To verify that the subscriber is not negatively impacted when calls are dropped due to network failures. Dropped call recovery is not supported with network failures, but the solution components should not encounter any negative impacts.		Passed	
SR60.UNI.003.14	Voice Mail	Message Monitor-Gateway Call	To verify the message monitor feature from gateway callers.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.UNI.003.58	Voice Mail	Message Monitor-Gateway Call	To verify the message monitor feature from gateway callers.		Passed	
SR60.UNI.005.1	Voice Mail	Secure Messaging Enhancements-Secure Messages From VPIM Subscribers	To verify that the messages will be properly decrypted and sent to Trusted Internet Subscribers through VPIM.		Passed	
SR60.UNI.006.3	Voice Mail	Secure Messaging Enhancements-Secure Messages From VPIM Subscribers	To verify that the messages will be properly encrypted when received from Trusted Internet Subscribers through VPIM.		Passed	
SR60.UNI.007.4	Voice Mail	Call Forward Configuration-Gateway	To verify that Cisco Unity will forward calls from a gateway to the appropriate mailbox depending on the call forward configuration.		Passed	
SR60.UNI.007.8	Voice Mail	Call Forward Configuration-Gateway	To verify that Cisco Unity will forward calls from a gateway to the appropriate mailbox depending on the call forward configuration.		Passed	
SR60.UNI.008.2	Voice Mail	Call Forward Configuration-Phone	To verify that Cisco Unity will forward calls from a phone to the appropriate mailbox depending on the call forward configuration.		Passed	
SR60.WIR.102		LAG Link Failure	verify physical link failure with atleast two links configured in LAG mode.		Passed	
SR60.WIR.105	Wireless	Unified IP Phone 7925 L3 Roaming	Verify the Layer 3 Unified IP Phone 7925 roaming between two WLAN controllers in different subnets.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR60.WIR.111		IP Communicator/Unified Video Advantage/Data Across Wireless Connection	Verify and test wireless data PC with voice and video applications.		Passed	
SR61.CER.100.1	Cisco Emergency Responder	LLDP-MED and CDP Enabled in Both TNP Phone and Access Switch	Verify that Cisco Emergency Responder (CER) can track the location of TNP Phones enabled with both Link Layer Discovery Protocol - Media Endpoint Discovery (LLDP-MED) and Cisco Discovery Protocol (CDP) and 911 calls from these Phones are routed to the local Public Safety Answering Point (PSAP).		Passed	
SR61.CER.102	Cisco Emergency Responder	E911 Call Handling When Both Cisco Emergency Responder and Unified SRST are Capable of Routing E911 Calls to PSAP	Verify that Cisco Emergency Responder can route E911 calls from a branch Phone when both Emergency Responder and Unified SRST routers are configured to route the E911 call to a local Public Safety Answering Point (PSAP).		Passed	
SR61.SRST.102	Unified SRST	Secured SCCP Phones on Unified SRST Gateway When WAN Link Goes Down	To verify the behavior of secured SCCP Phones on Unified SRST gateway when WAN link goes down.		Passed	
SR61.UMG.100.1	Unified Messaging Gateway	Unified Messaging Gateway Redundancy and Resiliency	Verify the ability to demonstrate Cisco Unified Messaging Gateway resilience to failures.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
SR61.UNC.101	Unity Connection	Encrypted Call Involving Unity Connection Blind Transferred to a Non-Secure IP Phone	Verify that an encrypted call between an IP Phone and Unity Connection can be successfully transferred to a non-secure IP Phone.		Passed	
SR61.UNC.102		Call From Secure Gateway Supervised Transfer to Encrypted IP Phone	Verify that a call from a secure gateway can be successfully transferred to an IP Phone that supports media encryption.		Passed	
UC700IF.CCM.050	Unified CM	BLF For Speed Dial for SIP IP Phone7916	To verify if a user can configure BLF-Speed dial for first 3 lines, middle 3 lines and last 3 lines. There are 36 lines with the two Unified IP Phone 7916 expansion modules. Monitor the presence status of the numbers configured.		Passed	
UC700IF.CCM.051	Unified CM	TNP SCCP Phone With Extension Module Configured	Verify that a user logged in a TNP SCCP phone with extension module configured for BLF speed dial for can see the presence status.		Passed	
UC700IF.CCM.052	Unified CM	Unified CM Presence Status on Secure Guiness Phone	Verify that Unified CM presence status works on a Secure Guiness phone for call history lists, BLF speed dials, and SIP URI.		Passed	
UC700IF.CCM.052	Unified CM Presence	Unified CM Presence Works on Secure IP Phone 7900 series for Call History lists, BLF Speed dials, and SIP URI	Verify that Unified CM presence works on a Secure IP Phone 7900 series for Call History lists, BLF Speed dials, and SIP URI.		Failed	CSCtc32278
UC700IF.CCM.060	Basic Call Flow	Unified CM Gatekeeper Based CAC	To verify that Unified CM Gatekeeper based CAC works as defined in UC 6.X SRND.	Phone->Unified CM (ZoneA)->GateKeeper Unified CM (ZoneB)->Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC700IF.CCM.061	Basic Call Flow	Unified CM Gatekeeper Based CAC With Video	To verify that Unified CM Gatekeeper based CAC works as defined in UC 6.X SRND.	Phone->Unified CM (ZoneA)->GateKeeper Unified CM (ZoneB)->Phone	Passed	
UC700IF.CCM.062	Basic Call Flow	Unified CM Gatekeeper Based CAC With Session Bandwidth Restrictions	To verify if Unified CM Gatekeeper based CAC works as defined in UC 6.X SRND.	Phone->Unified CM (ZoneA)->GateKeeper Unified CM (ZoneB)->Phone	Passed	
UC700IF.CCM.063	Basic Call Flow	Unified CM Gatekeeper Based CAC With Remote Zones	To verify that Unified CM Gatekeeper based CAC works as defined in UC 6.X SRND.	Phone->Unified CM (ZoneA)->GateKeeper1->GateKeeper2->Unified CM (ZoneB)->Phone	Passed	
UC700IF.CCM.098	Basic Call Flow	Unified CM Static Location based CAC Video Enabled Endpoints	To verify if Unified CM detects audio bandwidth from the location when video enabled phone makes an audio call.	Video Phone->(Region 1)->Unified CM->(Region 2)->Audio Phone	Passed	
UC700IF.CCM.099	Basic Call Flow	Unified CM Static Location based CAC Adjusts Bandwidth Usage	To verify if Unified CM Static Location based CAC can adjust the bandwidth usage when video call is downgraded to audio only call.	Video Phone->(Region 1; Location 1)->Unified CM->(Region 2; Location 2)->Audio Phone	Passed	
UC700IF.CCM.140.3	Unified CM	Local Route Group And Transformation	Verify to ensure that Unified CM correctly routes calls using 'Virtual Local Route Group' provisioning.	siteA endpoint->Unified CM->siteA PSTN; siteA endpoint->Unified CM->siteA PSTN (no bandwidth to go to toll free site); Called number transformed; siteA endpoint->Unified CM->siteA PSTN (route to toll free site); Called number transformed	Passed	
UC700IF.CCM.604	Unified CM	Conference Chaining of Secure and NonSecure Conferences	To verify the conference chaining of secure and non-secure Conferences.	Phone->Unified CM->ConfBridge	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC700IF.CCX.201	Unified CCX	Unified IP Phone 7921G Agent: IP Phone Agent	Verify to ensure IP Phone 7921G can be used as IP Phone Agents, receive inbound call and transfer the call to another 7971 CAD Agent.		Passed	
UC700IF.CCX.202	Unified CCX	Unified IP Phone 7921G Agent: CAD Agent	Verify to ensure that IP Phone 7921G can be used as CAD Agents, receive inbound call and conferences the call to Supervisor.		Passed	
UC700IF.CCX.400	Unified Contact Center Express	Unified CCX Backup And Restore	To verify if Unified CCX backup and restore is successful.		Passed	
UC700IF.CUB.010	Video Fallback to Audio via IP-to-IP GateWay	Video Call Falls Back to Audio When Endpoint is Not Video Capable	Verify de-escalation of video call to audio call when endpoint is not video capable.		Passed	
UC700IF.CUP.026	Unified Presence 8.0	Intercluster IM Between IPPM and Unified Personal Communicator Clients	Verify that IPPM and Unified Personal Communicator clients in separate clusters can exchange IM successfully.		Passed	
UC700IF.CUP.027	Unified Presence 8.0	Intercluster IM Between Unified Personal Communicator Clients Distributed in Two Clusters	To verify if a user can setup a IM session involving more than two Unified Personal Communicator clients and IPPM client distributed in multiple clusters.		Passed	
UC700IF.UCA.010.1	Unified Personal Communicator	Listen to Voicemail Under Secure Connection to Voicemail Servers	To verify that Unified Personal Communicator (Unified PC) can talk to Cisco Unity and Unity Connection Servers which are configured in the test bed in different ways using IMAP and download the Voicemail and listen to it.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC700IF.UCA.016.1	Unified Mobility Advantage, Unified Personal Communicator, Unified Presence	Inter Cluster DND	Verify that Unified Personal Communicator client set DND status and all presence enabled endpoints should able to see it's DND status.		Passed	
UC700IF.UNC.502	Unity Connection	DTMF Negotiation Between a RFC 2833 Capable Device and KPML Capable Unity Connection	To verify that MTP is invoked for a call between SIP gateway configured to support only RFC 2833 and Unity Connection configured to support only KPML.		Passed	
UC700IF.UNC.503	Unity Connection	Support for RFC 2833 DTMF in a Secure SIP Integration Between Unified CM and Unity Connection	To verify that Unity Connection can successfully decrypt RFC2833 digits received from the endpoint for a encrypted call.		Passed	
UC700IF.UNI.103	Cisco Unity	Live Reply to External VPIM Subscriber Homed on Another Unity Server	To verify live replies to external subscriber.		Passed	
UC701EF.QSG.003	Call Forward	Call Forwarding Scenario Involving Calls with MGCP GW and ICT to a H323 GW	Verify if a call from a Unified CM MGCP Gateway phone via an Inter Cluster Trunk to a Unified CM controlled phone can be call forwarded on busy or all to a PSTN phone via H.323 Gateway.	VG224 Ph 1-> MGCP GW-> Unified CM->ICT(QSIG)->Unified CM->SCCP/SIP Ph1->CFB->Unified CM->H.323 BRI/PRI -> PSTN Ph1	Passed	
UC701EF.QSG.004	Call Forward	Call Forward Interaction Involving MGCP Gateway, QSIG PBX , Unified CM ICT, CME Phone and Unity	Verify if a call From a MGCP Gateway to a QSIG PBX phone can be call forwarded on no answer to Cisco Call Manager Express phone and then to Unity.	PSTN ph1-> MGCP (BRI/PRI) ->Unified CM-> QSIG Trunk->QSIG PBX->PBX Ph1->CFNA->QSIG PBX->QSIG Trunk->Unified CM->ICT (QSIG)->Unified CM->IPIPGW->GateKeeper (H323)->CME->CME SCCP Ph1->Unity	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701EF.QSG.006	Call Forward and Call Transfer	H.323 GW Call Transfer by QSIG PBX to SIP Proxy	Verify if a call from a H.323 Gateway to a Unified CM SCCP phone can be transferred to a QSIG PBX phone and call forwarded on no answer to a SIP proxy.	PSTN Ph 1-> Unified CM H323 (BRI/PRI)-> Unified CM->SCCP Ph1->XFER_C->Unified CM->QSIG Trunk->PBX->PBX Ph1->CFNA->PBX->QSIG Trunk->Unified CM->SIP Trunk->CSPS->SIP Ph1	Passed	
UC701EF.QSG.010	Call Transfer	Blind Transfer of MGCP GW Call by SCCP Phone via ICT to a Third party Operator console	Verify if a call from a CMM MGCP phone via ICT to a Unified CM SCCP phone can be blind transferred via ICT to a Third party Operator console.	Pots CMM Ph1-> MGCP GW->Unified CM->ICT (QSIG)->Unified CM->SIP Ph1->XFER_B->ICT (QSIG)->OP Cons1	Passed	
UC701EF.QSG.011	Call Transfer and Call Forward	Interaction of Call Transfer with MGCP GW, ICT, SIP phone, QSIG PBX and IPMA.	Verify if a call From a Unified CM MGCP Gateway phone via ICT to a Unified CM SIP phone can be transferred to a QSIG PBX phone and call forwarded on busy or all to an IPMA Manager phone.	Pots Ph1->Unified CM->ICT (QSIG)->Unified CM->SCCP Ph1->XFER_B->QSIG Trunk->PBX->PBX ph1->CFA->IPMA Manager Phone	Passed	
UC701EF.QSG.015	Call Forward and Call Transfer	Interaction of Call forward and Transfer with Unified CM,SIP phone, QSIG PBX,IP communicator	Verify if a call from a Unified CM SIP phone to a QSIG PBX phone can be transferred to another QSIG PBX phone and call forwarded on no answer to a Unified CM IP communicator.	SIP Ph1-> Unified CM->QSIG Trunk->QSIG PBX->PBX Ph1->XFER_C ->PBX Ph2-> CFNA->QSIG Trunk->Unified CM ->IPC1	Passed	
UC701EF.QSG.017	Call Transfer and Call Forward	CFNA and Transfer Interaction with Unified CME, QSIG PBX ,MGCP GW ,ICT and IPMA manager	Verify if a call from a Unified CME phone to a QSIG PBX phone is transferred to a MGCP Gateway CMM phone and the call is forwarded on no answer via ICT to an IPMA Manager phone.	SCCP Ph1->CME->IPIPGW (H323)->GateKeeper->Unified CM->QSIG Trunk->PBX->PBX Ph1->XFER_C->QSIG Trunk->Unified CM ->MGCP GW->CMM pots Ph1->CFNA->Unified CM->ICT (QSIG)->Unified CM ->IPMA Mgr Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701EF.QSG.020	Call Forward	Multiple Call forwarding over SIP proxy ,Unified CM and ICT creating a loop	Verify the call From a SIP phone under a SIP proxy to a Unified CM phone is forwarded via ICT to a CCM controlled phone and forwarded again via ICT to a Unified CM phone which forwards back to a Unified CM controlled phone, creating a loop.	SIP Ph1->CSPS->SIP Trunk->Unified CM->SIP Ph2 ->CFA->ICT (QSIG)-> Unified CM ->SCCP Ph1->CFA->ICT (QSIG)->Unified CM->SCCP Ph2-> CFA->ICT(QSIG)->Unified CM->SIP Ph2	Passed	
UC701EF.QSG.024	CLIP/CLIR	Calling Line ID Restriction from a QSIG PBX phone via ICT to a CCM SCCP phone	Verify the calling and called line and name Restriction on a call from a QSIG trunk to Unified CM and an Inter Cluster Trunk to a Unified CM SCCP phone.	PBX Ph1->PBX ->QSIG Trunk->Unified CM->ICT(QSIG)->Unified CM->SCCP Ph1	Passed	
UC701EF.SMB.005	Unity Connection	Voice Mail Deposit of Another User in Unity Connection on Unified Communications Manager Business Edition Using Speech Recognition	Verify the ability to use speech recognition to deposit a voice message for another user in Unity Connection on Unified Communications Manager Business Edition.	Rem SCCP Ph 1->Unified CMBE->CFNA->Unified CMBE->Unity Connection	Passed	
UC701EF.SMB.007	Unity Connection	Using Unity Personal Communications Assistant in Unity Connection on Unified Communications Manager Business Edition to call users in a Hunt List	Verify the ability to use Unity Personal Communications Assistant in Unity Connection on Unified Communications Manager Business Edition to call other users specified in a hunt list.	Stage1:RemSCCP Ph1->Unified CMBE->SCCP Ph2->CFNA-Unified CMBE->Unity Connection Stage2:Unity Connection->Unified CMBE->SCCP Ph1/RemSCCP Ph2/7985G	Passed	
UC701EF.SMB.012	RSVP	Calling From Unified IP Video Phone When Video Reservation Is Mandatory	Verify the ability of making a call from a central Unified IP Video Phone to a remote Unified IP Video Phone 7985 when the RSVP reservation for video is mandatory.	Stage1:Vid Ph1->Unified CMBE->RSVP->RemVid Ph2 Stage2:Vid Ph1->Unified CMBE->RSVP->RemVid Ph2 Stage3:Vid Ph1->Unified CMBE->RSVP->RemVid Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701EF.SMB.014	RSVP	Video Call when Audio Reservation is Optional and Video if Reservation Succeeds	Verify the ability of making a call from a central Unified IP Video Phone to a SCCP Phone with Unified Video Advantage in a remote site where audio reservation is optional and video if reservation succeeds over SCCP.	Stage 1:Vid Ph 1->Unified CMBE->RSVP->RemVid Ph 2 Stage 2:Vid Ph 1->Unified CMBE->RSVP->RemVid Ph2 Stage 3:VidPh1->Unified CMBE->RSVP->Vid Ph 2	Passed	
UC701EF.SMB.017	RSVP	Calling from Video SCCP Phone to Cisco Unified Video Telephony Advantage (CUVTA) When Audio Reservation Is Mandatory and Video If Reservation Succeeds	Verify the ability to call from a central Video SCCP Phone with Unified Video Advantage when audio reservation is mandatory and video if reservation succeeds over SCCP.	Stage 1: SCCP Ph 1(CUVTA)->Unified CMBE->RSVP->Rem SCCP Ph 2(CUVTA) Stage 2: SCCP Ph 1(CUVTA)->UCMBE->RSVP->Rem SCCP Ph 2(CUVTA)	Passed	
UC701EF.SMB.030	Unified CM Business Edition Failure	Call Transfer from Central Survivable Remote Site Telephony Phone to PSTN Phone	Verify if a user can place a call from a SCCP phone to a SIP Phone in the central site during Unified CM Business Edition Failure. Transfer the call to a PSTN phone.	SCCP Ph 1->SRST 1->SIP Ph 1->XFER->PSTN GW->PSTN Ph 1	Passed	
UC701EF.SMB.032	Unified CM Business Edition Failure	Call from Remote Site to Central Site Using WAN during Co-Res Recovery	Verify by making a call from a remote site to the Central Site during Unified CM Business Edition Failure.	Rem SCCP Ph 1->SRST 1 ->SRST 2->SCCP Ph 2	Passed	
UC701EF.SMB.033	Unified CM Business Edition Failure	PSTN Forwarded Call During Unified CM Business Edition Recovery	Verify if a call from a SCCP phone to a SIP Phone whose CFNA is set to a PSTN Phone in the central site during Unified CM Business Edition fails.	Stage 1: SIP Ph 1->CFNA->PSTN Ph 1 Stage 2: Rem SCCP Ph 1->SRST 1->SRST 2->SIP Ph 1->CFNA->PSTN GW->PSTN Ph 1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701EF.SMB.039	Unified CM Feature IPMA	IP Manager Assistant (IPMA), SIP, Proxy Mode Manager in Central Site and Assistant in Remote Site	Verify if a user can configure a SIP phone on remote site for Assistant on phone proxy mode and have Manager at central site. Place a call from PSTN to the Manager Phone . The call goes to Assistant, the Assistant uses transfer to VM feature and sends the call to Manager VM box.	Stage 1:PSTN Ph 1->PSTN GW->Unified CMBE->IPMA SIP Ph 2 (Asst) Stage 2:SIP Ph 2 (Asst)->transfer VM->IPMA->Unified CMBE->Unity Connection Stage 3:SIP Ph 1->Unified CMBE->Unity Connection	Passed	
UC701EF.VID.005	Adhoc Conference	Adhoc Conference Using Continuous Presence and Transrating on MCU 3545	Verify if a user can make an adhoc conference call on MCU 3545 using SCCP and enable continuous presence.	Stage1:SCCP Video Ph1->Unified CM->Rem SCCP Video ph2 Stage2:Rem SCCP Video Ph2->Unified CM->3545->SCCP Video Ph3 Stage3:Rem SCCP Video Ph2->Unified CM->3545->SCCP Video Ph4	Passed	
UC701EF.VID.006	Video	Video Activation During an Audio Call in Progress Between H.320 Device and IP Phone with Unified Video Advantage	Verify if video can be activated on Unified Video Advantage while audio call is already setup between the IP Phone and H.320 device.		Passed	
UC701EF.VID.007	Video	Video Deactivation During an Audio Call in Progress Between H.320 Device and IP Phone with Unified Video Advantage	Verify if video can be deactivated on Unified Video Advantage while Video call is already setup between the IP Phone and H.320 device.	Stage 1:H.320->3545(GW)->Unified CM->SCCP Ph1 (CUVA)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701EF.VID.008	Reservational Conference	Reservationless Videoconference on 3545 through H.323 Interface	Verify if a user can make reservationless videoconference using an IPVC 3545 through H.323 interface and calling users are H.320 terminal, Video endpoints supporting H.323 and SCCP.	Stage1: H.320->3526 (GW)->Unified CM->3545 Stage2: H.323 Video Ph1->Unified CM->3545 Stage3: Rem SCCP Video Ph2->Unified CM->3545	Passed	
UC701EF.VID.009	Reservationless Conference	Reservationless Videoconference Through SIP Interface	Verify if a user can make reservationless videoconference using an MCU 3545 through SIP interface and calling user are H.320 terminal, Video endpoints supporting H.323 and SCCP.	Stage1: H.320->3527(GW)->Unified CM->3545 Stage2: H.323 Ph1->Unified CM->3545 Stage3: Rem SCCP Ph1->Unified CM->3545	Passed	
UC701EF.VID.010	Basic Video call	Video Call from SCCP Sony Video Phone to H.320 Video Terminal Through 3545	Verify if a user can make a video call from SCCP Sony video phone to H.320 video terminal which is connected through an IPVC 3545.	SCCP Sony Video Ph1->Unified CM->3545 (GW)->H.320 terminal	Passed	
UC701EF.VID.013	IPVC	H.323 Endpoints Calling H.320 Endpoint	Verify fi a user can make a video call from H.323 endpoint registered to the gatekeeper to H.320 endpoint through IPVC 3545.	H.323 Video Ph1->GateKeeper->Unified CM->IPVC 3545->PSTN->H.320 terminal	Passed	
UC701EF.VID.016	Adhoc Video Conference	SCCP Adhoc Video Conference through IPVC 3545 Using H.263	Verify if a user can make adhoc Video conference through IPVC 3545 using H.263.	Stage1: SCCP Video Ph1->Unified CM->SCCP Video Ph2 Stage2: SCCP Video Ph2->Unified CM->3545->CNF_AD->Unified CM->SCCP Video Ph3	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701EF.VID.018	Adhoc video Conference	SCCP Adhoc Video Conference using Transcoding on 3545	Verify if a user can make an adhoc conference call through 3545 between users using different codec.	Stage1: SCCP Video Ph1->Unified CM->SCCP Video Ph2; Stage2: SCCP Video Ph2->Unified CM->3545->CNF_AD->Unified CM->SCCP Video Ph3	Passed	
UC701IF.CRS.093	Agents in SRST Location	Unified CCX Reroutes Call to Next Available Agent in Different Location	Verify to ensure Unified CCX reroutes the call to next available agent in different location if the call to current agent is rejected due to CAC.		Passed	
UC701IF.CRS.094	Agents in SRST Location	Unified SRST Loses WAN Connectivity While Remote Agent Handles Inbound Call	Verify to ensure that Agent becomes Not Ready when Unified SRST location loses WAN link and then becomes Ready again when WAN connectivity is restored.		Passed	
UC701IF.CRS.100.3	Unified Contact Center Express	Monitor Presence Status And Establish Chat Session With SME	Verify to ensure Cisco Agent Desktop can monitor presence status and establish chat session with non-agent SME using Unified Personal Communicator/IPPM.		Passed	
UC701IF.CRS.101.2	Unified CCX	Click to Dial, Transfer, Conference	Verify to ensure CAD Desktop can click to dial, transfer or conference the Subject Matter Expert (SME).		Failed	CSCtc91625
UC701IF.CRS.103.2	Unified CCX	Personal Contacts	Verify to ensure CAD Desktop can show Unified Personal Communicator Agent's buddies as Personal Contacts in CAD Chat Selection Window.		Failed	CSCtc91625
UC701IF.CRS.105.2	Unified CCX	SME Initiating Chat Session with CAD	Verify to ensure SME can initiate Chat Session with CAD.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701IF.CRS.106.1	Unified CCX	CAD Attempting to Register to Incorrect Unified Presence Server	Verify to ensure CAD is redirected to register to correct Unified Presence Server in the Cluster if it attempts to register to a Unified Presence Server where the CAD user is not defined.		Failed	CSCtc91625
UC701IF.CRS.107	Unified CCX	Real Time Update of Presence Status of CAD and SME in Chat Selection Window	Verify to ensure Real Time Update of presence status of CAD and SME in the Chat Selection Window.		Failed	CSCtc91625
UC701IF.CRS.109.4	Unified Contact Center Express	Distribute Support New Entry Level Email offering	Verify to ensure that incoming emails to a CSQ are distributed to idle agents and voice has higher priority than email.	Email->Email CSQ->CAD	Passed	
UC701IF.CRS.120	Unified Contact Center Express	Voice Mail Left in Cisco Unity is Sent as Attachment to the Inbound Email in Unified CCX	To verify if the voice mail is sent as attachment to email-CSQ and if the email is delivered to the agent, when an end-user leaves a voice mail to the support number associated with email CSQ.	Customer->Unified CCX Pilot number->Voice Mail->Unified CCX Email CSQ->Email ready Agent	Passed	
UC701IF.CRS.121	Unified Contact Center Express	Customer Replies Voice Mail to Unified CCX Inbound Email Mailbox	Verify to ensure that Unified CCX inbound email can handle replies to voice mails.	Customer->Voice Mail->Reply the voice mail->Unified CCX Email CSQ->Email ready Agent	Passed	
UC701IF.CRS.155	Unified CCX	CAD Generates Both Audio and Visual Notification When Subject Matter Expert (SME) Sends IM to CAD	Verify to ensure CAD desktop plays audio tone and chat button in windows taskbar flashes to indicate arrival of new IM.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701IF.CRS.300	Unified Contact Center Express	Importing Agent User from Unified CM to Cisco Unity Connection Using AXL	Verify to ensure that Cisco Unity Connection can import Agent users from Unified CM using AXL.		Passed	
UC701IF.CRS.301	Unified Contact Center Express	Call to Agent's Voice Mail is Re-Queued to Unified CCX-CSQ by Call Transfer Option in Cisco Unity	Verify to ensure Cisco Unity can provide option in agent's voice mail to re-queue the call to CSQ so that it is serviced by another available agent.		Passed	
UC701IF.CRS.302	Unified Contact Center Express	Call to Agent's Voice Mail is Re-Queued to Unified CCX-CSQ by Call Transfer Option in Cisco Unity Connection	Verify to ensure that Cisco Unity Connection provides an option in agent's voice mail to re-queue the call to CSQ so that it is serviced by another available agent.		Passed	
UC701IF.CRS.303	Unified Contact Center Express	Cisco Unity Auto Attendant Call Transferred to Agent's Voice Mail for Re-Queue to Unified CCX	Verify to ensure that Cisco Unity auto attendant interaction with Unified CCX agent is possible and calls get transferred to CTI route points.		Passed	
UC701IF.CUB.007	Unified Border Element	Secure Transcoding and Conference Between G722/iLBC and G711	Verify that calls between Unified CM and Unified CME phone via IP-to-IP Gateway with secure transcoding and secure conference between Unified CM and Unified CME participants.	Unified CM Phone->SIP Trunk->IP-to-IP Gateway->Unified CME->Unified IP Phone->Conference->IP-to-IP Gateway->Unified CM->Unified IP Phone	Passed	
UC701IF.MP.104	Video Endpoints	Out-dial Using Web to SCCP and SIP with Third Party Video Endpoints	To verify Out-dial using Web to SCCP and SIP with Third party video endpoints within and across cluster using SIP Trunk DTMF preference with RFC2833, OOB & RFC2833 with transport type TCP on security profile.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701IF.MP.110	Audio and Video Codec Using Different Endpoints	MeetingPlace Endpoints Joining G.729 and iLBC Codec	To verify Unified MeetingPlace endpoints joining with G.729 and iLBC codec from within and across cluster environment and using H.264 with High quality video preference within cluster and across cluster ICT trunk.		Passed	
UC701IF.MP.113	Unified MeetingPlace	Schedule Continuous Meeting for Web, Audio and Video	To verify if a user can schedule continuous meeting for web, audio and video and initiate out-dial but CM Sub1 server experience network outage after 1st out-dial succeeded.		Passed	
UC701IF.MP.122	Unified MeetingPlace	Conference With 22 or More participants For Audio, Web And Video	To verify if a conference with 22 or more participants for audio, web and video is successful.	Video endpoint->Unified CM->SIP trunk->Unified MeetingPlace	Passed	
UC701IF.UNC.100.1	Unity Connection	Unity Connection Redundancy And Resiliency	To verify the Unity Connection resilience to failures and capability to load share in a Active-Active deployment.		Passed	
UC701IF.UNC.100.3	Unity Connection	Unity Connection Redundancy And Resiliency	To verify the Unity Connection resilience to failures and capability to load share in a Active-Active deployment.		Passed	
UC701IF.UNC.100.5	Unity Connection	Unity Connection Redundancy And Resiliency	To verify the Unity Connection resilience to failures and capability to load share in a Active-Active deployment.		Passed	
UC701IF.UNC.500	Unity Connection	SIP Gateway and Unity Connection Configured to Support Both KPML and RFC 2833	To verify that when RFC 2833 is negotiated for the call then Connection ignores KPML and sends/receives RFC 2833 only.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701IF.UNC.700.2	Unity Connection	Unity Connection Codec Support And Transcoding Capability	To verify the Unity Connection capabalility to support a suite of codecs for RTP streams and transcode the RTP stream to the recording format.		Passed	
UC701IF.UNC.700.4	Unity Connection	Unity Connection Codec Support and Transcoding Capability	To verify the Unity Connection capabalility to support a suite of codecs for RTP streams and transcode the RTP stream to the recording format.		Passed	
UC701IF.UNC.700.5	Unity Connection	Unity Connection Codec Support and Transcoding Capability	To verify the Unity Connection capabalility to support a suite of codecs for RTP streams and transcode the RTP stream to the recording format.		Passed	
UC701IF.UNC.900	Unity Connection	Chaining Message Notification Limited by Sender When Secondary Becomes Acting Primary	Verify that message notification is triggered based on the subscriber configured setting when a message arrives.		Passed	
UC701IF.UNC.904	Unity Connection	Unity Connection Integration With Unified CME Over SIP With DTMF Support for RFC2833	To verify that RFC 2833 is negotiated successfully between Unified CME and Unity Connection.		Passed	
UC701IF.UNC.908	Unity Connection	Blind Transfer with Unity Connection Over a Secure SIP Trunk	To verify that a secure call to Unity Connection can be transferred to another secure phone and the call remains secure. The transfer is being done by the phone and not Unity Connection.		Passed	
UC701IF.UNC.912	Unity Connection	Last Redirecting Number With Multiple Call Forwarding in Unified CME	To verify that Unified CME provides the last redirecting number to Unity Connection rather than first.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC701IF.UNC.915	Unity Connection	NDR for Message Sent by Unity Connection to Unified Messaging Gateway	Verify if a NDR sent by Unified Messaging Gateway reaches Unity Connection and is delivered to the senders mailbox.		Failed	
UC701IF.VID.101	Video	Video Call to H.323 Third Party Video Endpoint Using RAS Aggregator Trunk	To verify if a user can place call from a SCCP video endpoint to third party H.323 video endpoint using RAS aggregator trunk. Third party endpoint is defined on Unified CM by name, not IP address, which allows the endpoint to use DHCP for IP addressing.		Passed	
UC702EF.ARC.029	Unified Attendant Server	Unified Attendant Server IP phone Loses Network Connectivity	Verification of Unified Attendant Server attendant IP phone losing network connectivity.		Passed	
UC702EF.ARC.031	Unified Attendant Server	Unified Attendant Server Client Loses Network Connectivity When Active on a Call and Re-establishes Connection After Some Time	Verify the following: Make a call from PSTN to operator. Operator answers the call and is initiating a transfer to a SCCP phone in remote site. Disconnect the operator from the network when the transfer is going on and examine the behavior. Bring back the system to normal after few minutes and check if the operator is back to normal.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC702EF.CME.005	CME H.450	Call Forward All from QSIG PBX on Unified CME to IPMA Manager	Verify if a call from a SIP proxy controlled SIP phone via Unified CM to QSIG PBX phone on Unified CM is call forwarded all to a QSIG PBX phone connected to Unified CME which in turn has call forward busy to IPMA Manager.	SIP Ph1->CSPS->SIP Trunk->Unified CM->QSIG Trunk->PBX Ph1->CFB->QSIG Trunk->Unified CM->IPIP GW (H323)->CME->QSIG Trunk->PBX Ph1->CFA->QSIG Trunk-> CME->IPIP GW (H323)->Unified CM->IPMA	Passed	
UC702EF.CME.015	Conference	Ad-Hoc Conference on Unified CME with DPNSS PBX Phone and IPMA Manager Phone	Verify the Ad-Hoc Conference Setup by Local Unified CME Phone with DPNSS PBX Phone and IPMA Manager Phone.	1: SCCP Ph1 -->CME -->IPIP GW (H323)->Unified CM->QSIG Trunk->Westell GW->DPNSS PBX Ph1 2: SCCP Ph1 --> CME->CNF->CME->IPIP GW (H323) -->Unified CM -->IPMA Manager 3: SCCP Ph1->CME->CNF->DPNSS PBX Ph1 & IPMA Manager phone	Passed	
UC702EF.CME.028	Unified CME	Meet Me Conference between QSIG PBX Phone on Unified CME and DPNSS PBX Phone	Verify the Meet Me conference feature on Unified CME between 2 Unified CME phones (SCCP and QSIG PBX phone connected to Unified CME) and a DPNSS PBX phone.	1 : SCCP Ph1->CME->CNF_MM 2: PBX ph1->QSIG Trunk->CME->CNF_MM 3: DPNSS PBX Ph1->Westell->QSIG Trunk->Unified CM->IPIPGW (H323)->CME->CNF_MM	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC702EF.CME.078	Transfer to Voice Mail	Transferring Calls from PSTN Phone to Unified CME Phone Voicemail	Verify when a call from PSTN phone (Phone A) to Unified CME SCCP phone (Phone B) is forwarded on no answer to another Unified CME SCCP phone (Phone C), and if the Unified CME SCCP phone (Phone C) can transfer the call to voice mail box of Unified CME phone (Phone B) using TrnsfVM soft key.	1: PSTN Ph1->MGCP BRI->Unified CM->GateKeeper->IP-IP GW->GateKeeper->CME->SCCP Ph1->CFNA->CME->SCCP Ph2 2: SCCP Ph2->TrnsfVM->SCCP Ph1 DN# ->CME->NM CUE 3: SCCP Ph1->CME->NM CUE	Passed	
UC702EF.CME.156	Shared Line cBarge and Privacy	Shared line cBarge and Privacy support on Unified CME with Consult Transfer to Unified CM Video Phone	Verify if a call from QSIG PBX phone (Phone C) to SCCP Unified CME shared line phone (Phone A) is barged from another Unified CME shared line SCCP phone (Phone B) and if the call is consult transferred to a Unified CM video phone (Phone D) from Unified CME phone (Phone A).	1: PBX Ph1->QSIG Trunk->Unified CM->GateKeeper->IP-IP GW->GateKeeper->CME->SCCP Phone1(SL) 2: SCCP Ph2 (SL)->CME->cBrg->SCCP Ph1 (SL)->XFER_C->CME->GateKeeper->IP-IP GW->GateKeeper->Unified CM->SCCP Video Ph1	Passed	
UC702EF.QSG.040	Callback	Interaction of Callback Request and Call Forwarding with QSIG PBX and Unified CM	Verify the callback request from a PBX phone on forwarded busy call on Unified CM.		Passed	
UC702EF.SMB.004	Unity Connection	Voice Mail Retrieval from Unity Connection on Unified Communications Manager Business Edition from SIP Phone	Verify the ability to successfully retrieve a voice message in Unity Connection on Unified Communications Manager Business Edition from a SIP phone.	Stage 1: PSTN Ph 1->PSTN GW->Unified CMBE->SCCP Ph 1->XFER ->SIP Ph 1->Unified CMBE->Unity Connection Stage 2: SIP Ph 1->Unified CMBE->Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC702EF.SMB.010	Unity Connection	Voice Mail Deposit in Unity Connection Using G.729 Codec From Remote Site	Verify the ability of depositing a voice message in Unity Connection using G.729 codec from a remote site.	Stage 1: Rem SCCP Ph 1->Unified CMBE->Rem SCCP Ph 2 ->CFNA ->Unified CMBE->Unity Connection Stage 2: Rem SCCP Ph 1->Unified CMBE->Unity Connection	Passed	
UC702EF.SMB.021	DND	DND Feature During Callback	Verify by making a call from a central SCCP phone which has Do Not Disturb (DND) feature activated to a remote SIP phone that is busy. The SCCP phone in the central site should have Callback activated	Stage 1: Rem SIP Ph 1->Unified CMBE->PSTN GW->PSTN Ph 1 Stage 2: SCCP Ph 1->Unified CMBE->Rem SIP Ph 1->CALLBACK	Passed	
UC702EF.SMB.025	Unified CM Feature Intercom	Hold Reversion with Extension Mobility	Verify Hold reversion with Extension Mobility. Create Extension Mobility profile that is similar to one of the SIP Phones in the central site. Use this profile to login to a remote phone.	Stage 1:PSTN Ph 1->PSTN GW->Unified CMBE->SIP Ph 1->Hold Stage 2:SIP Ph 1 ->Resume	Passed	
UC702EF.SMB.034	Unified CM Commercial Feature Intercom	Intercom Between SIP and SCCP Phone on Different Remote Sites	Verify if a user can place an intercom call between SIP phone on a remote site to a SCCP phone on another remote site while the SCCP phone on the first remote site is active on call.	Rem SIP Ph 1->Intercom->Unified CMBE->Rem SCCP Ph 1	Passed	
UC702EF.VID.030	Video	Reservationless Conference Using Cisco 3540 Involving H.320 Endpoints and Remote 3rd Party Video SCCP Endpoint	Verify the reservationless video conference call using Cisco 3540 involving H.320 endpoint and Remote 3rd party SCCP video phone.	Stage1: H.320 Video Ph1->PSTN->Unified CM->3540; Stage 2: 3rd Party Video SCCP Ph2->Unified CM->PSTN->Unified CM->3540; Stage 3: SCCP Video Ph3->Unified CM->PSTN->Unified CM->3540	Passed	CSCtd03018

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC702EF.VID.031	Reservationless Video Conference	Reservationless Videoconference through H.323 using IPVC 3515 with Unified Video Advantage Phone	Verify if a user can make a reservationless videoconference through H.323 using IPVC 3515. The calling users are Unified Video Advantage phones and SCCP Video Phones.	Stage1: SCCP Ph1 (CUVA)->Unified CM->3515 Stage2: Rem SCCP Video Ph2->Unified CM->3515 Stage3: Rem SCCP Video Ph3 (CUVA)->Unified CM->3515	Passed	
UC702EF.VID.032	Basic Video Call	Video Call from H.320 to SCCP with Unified Video Advantage through IPVC 3522	Verify if a user can make a video call using IPVC 3522 from a remote H.320 video endpoint to another remote site SCCP Phone with Unified Video Advantage enabled.	H.320 Video Ph1->IPVC 3522->PSTN->Unified CM->SCCP Ph2 (CUVA)	Passed	
UC702EF.VID.033	Adhoc Video Conference	Adhoc Video Conference Using IPVC MCU 3511 with H.323 and Unified Video Advantage Endpoints	Verify if a user can make adhoc Video conference using an IPVC MCU 3511 involving H.323 endpoint and Unified Video Advantage.	Stage1: SCCP Video ph1->Unified CM->H.323 Video ph2 Stage2: H.323 Video ph2->Unified CM->3511(MCU)->Unified CM->SCCP ph3 (CUVA)	Passed	
UC702EF.VID.035	Video	H.320 Endpoints Calling IP communicator with Unified Video Advantage Endpoint through IPVC 3521 and Transfer to 3rd party SCCP Video Phone	Verify if a user can make a video call from H.320 endpoint to IP communicator with Unified Video Advantage through IPVC 3521 and transfer the call to 3rd Party SCCP Video Phone.	Stage1:H.320 Video Ph1->IPVC 3521->Unified CM->IP Communicator(CUVA)->XFER->3rd Party SCCP Ph3	Passed	
UC702EF.VID.036	Video	SCCP Video Endpoints Calling SCCP Phone and Transfer to H.320 Terminal Through IPVC 3521	Verify if a user can make an audio call from SCCP Video phone to SCCP Phone and transfer the call to H.320 terminal through IPVC 3521.	Stage1:SCCP Video Ph1->Unified CM->SCCP Ph2->Unified CM->IPVC 3521->H.320 terminal	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC702EF.VID.037	IPVC	Video Call Between H.323 Endpoints	Verify if a user can make a video call from a Remote H.323 endpoint registered to the gatekeeper to another H.323 endpoint in Central site.	H.323 Video Ph1->GateKeeper1->Unified CM->GateKeeper1->Rem H.323 Video Ph2	Passed	
UC702EF.VID.038	IPVC	PSTN Video Call Between Unified Video Advantage Phones in Central and Remote Sites Through IPVC 3527	Verify if a user can make a PSTN video call from a Remote SCCP phone enabled with Unified Video Advantage to another SCCP Phone in central enabled with Unified Video Advantage through IPVC 3527.	SCCP Ph1 (CUVA)->IPVC 3527->Unified CM->SCCP Ph2 (CUVA)	Passed	
UC702EF.VID.039	IPVC	Video Call Between H.320 Endpoint to IP Communicator with Unified Video Advantage Through IPVC 3526	Verify if a user can make a video call from a H.320 endpoint to IP Communicator with Unified Video Advantage enabled in central site through IPVC 3526.	H.320 endpoint->IPVC 3526->Unified CM->IP Communicator (CUVA)	Passed	
UC702EF.VID.040	Adhoc Video Conference	Adhoc Video Conference Using IPVC 3515 with SCCP and IP Communicator Endpoints	Verify if a user can make adhoc Video conference using an IPVC 3515 involving SCCP and IP Communicator endpoints.	Stage1: SCCP Video ph1->Unified CM->H.323 Video ph2 Stage2: H.323 Video ph2->Unified CM->3515(MCU)->Unified CM->IP Communicator (CUVA)	Passed	
UC702EF.VID.041	Transrating	Continuous Presence and Transrating on MCU 3545 using H.323 Interface Involving 3rd party Video Phone	Verify if a user can make reservationless conference using MCU 3545 and enable continuous presence involving 3rd Party Video Phone.	Stage1: SCCP Video Ph1->Unified CM->3545; Stage2: Rem SCCP Video Ph2->Unified CM->3545; Stage3: H.323 Video Ph3->Unified CM->3545; Stage4: 3rd Party SCCP Video Ph4->Unified CM->3545	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC702EF.VID.042	Transrating	Continuous Presence and Transrating on MCU 3545 Using SIP Interface	Verify if a user can make reservationless conference using 3545 and enable continuous presence.	Stage1: SCCP Video Ph1 & Rem SCCP Video Ph2->Unified CM->3545; Stage2: H.323 Video Ph3 & 3rd Party SCCP Video Ph4->Unified CM->3545	Passed	
UC71.PPR.001	PhoneProxy	Encrypt End-to-End Conversation With TLS/sRTP	Verify the PhoneProxy functionality. Verify the ability to encrypt end-to-end conversation with TLS/sRTP between phone connected via internet/PhoneProxy and campus phone.		Passed w/ Exception	CSCtc06130
UC712EF.ARC.012	Unified Attendant Server	Server Application is Restarted with Different CTI Manager as Primary	Verify the following: Unified Attendant Server is registered to Unified CM1 as primary & Unified CM2 as secondary. Purposefully interchange the settings on Unified Attendant Server so as to make Unified CM2 as primary and Unified CM1 as secondary. Restart the Unified Attendant Server .		Passed	
UC712EF.ARC.024	Unified Attendant Server	Retrieval of Call Parked by Operator		Unified IP Phones 6921/6941/6961 PH1->Unified CM->OP Console->Call Parking;Unified IP Phones 6921/6941/6961 PH 2->Park Retrieval.	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF.CCM.007	IPV6	Dual-stack Phone Operation Based on Media and Signaling Preference Settings	Verify a call from the NON DS cluster to the DS Phone on the DS cluster which CFWDALL is set to DS remote Phone on the DS cluster over SIP GW(DS).	Stage1: SCCP Ph1->Unified CM->QSIG ICT->Unified CM (DS)->SCCP Ph2 (DS); Stage2: SCCP Ph2(DS)->CFWDALL->PSTN GW (SIP GW DS)->PSTN->PSTN GW (SIP GW DS)->Unified CM(DS)-> Rem SCCP Ph3(DS)	Passed	
UC712EF.CCM.009	IPV6	Transferring the Call from IPv4 Phone on DS Cluster to VG224 GW POTS Phone	Verify a call from the NON DS cluster to the IPv4 Phone on the DS cluster and then transfer it to VG224 GW POTS Phone.	SIP Ph1->Unified CM ->QSIG ICT->Unified CM(DS)->SCCP Ph1->XFER->Unified CM(DS)->VG224 GW(DS)->POTS Ph1	Passed	
UC712EF.CCM.021	IPV6	Voicemail Deposit and Retrieval for Central Site DS Phone	Verify if a call from a PSTN phone to central site DS phone is forwarded to voicemail in Unity connection.	Stage1: PSTN Ph1->MGCP GW->Unified CM DS->DS Ph1->CFNA->Unity Connection	Passed	
UC712EF.CCM.022	IPV6	Voicemail Deposit and Retrieval for Remote Site DS Phone	Verify the call from a PSTN phone via SIP Gateway to remote site DS phone is forwarded to voicemail in Unity connection.	Stage1: PSTN Ph1->SIP GW(DS)->Unified CM DS->Rem DS Ph1->CFNA->Unity Connection	Passed	
UC712EF.CCM.023	CAC	Location Based CAC with IPv4/IPv6 Interworking	Verify if a user can place a call from a SCCP phone A to SIP Phone B which are in different locations and later initiate consultative transfer from SIP Phone B to SCCP Phone C resulting speech path established between Phone A & Phone B and consultative transfer fails with prompt " Not enough bandwidth".	Rem SCCP Ph A (DS) > Unified CM (DS) > SIP Ph B > Xfer_C > Unified CM > CAC > Rem SCCP Ph C	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF.CCM.024	IPv6	Intercom Call from SCCP (IPv4) to SCCP (DS) Phone	Verify is a user can make an intercom call from SCCP phone to another remote SCCP phone in different site. SCCP phone which is target gets the tone and goes on speaker mode with audio muted, after "talk" softkey is pressed two ways audio is established by the target phone.	Rem SCCP Ph 1 (v4) > Intercom > Unified CM (DS) > Rem SCCP Ph 2 (DS)	Passed	
UC712EF.CCM.025	IPv6	Unified CME Call to SCCP Phone in DS Cluster Having CFNA to IPMA Manager phone in Non-DS cluster	Verify the call from Unified CME SCCP Phone to a QSIG PBX phone in interop site which is transferred to SCCP Ph (DS) in central DS site which has CFNA to IPMA phone in Non-DS site.	SCCP Ph 1 > CME > IPIPGW(H323) > GK > Unified CM (interop) > QSIG Trunk > PBX > PBX Ph 1 > Xfer > ICT > Unified CM(DS) > SCCP Ph (DS) > CFNA > ICT > Unified CM (Non-DS) > IPMA Manager Ph 1 (v4)	Passed	
UC712EF.CCM.026	IPv6	IPv4/IPv6 Interworking When DS Phone Interacts with SIP IPv4 Phone in Interop Site which has CFwdAll set to UCCX IPPA Phone	Verify the call from SCCP (DS) phone from DS cluster to SIP IPv4 phone in Interop site via ICT which has CFwdAll set to UCCX which transfers the call to IPPA agent in the remote site.	SCCP Ph (ds) > Unified CM (DS) > ICT > SIP Ph > Unified CM (Interop) > CFwdAll > ICT (QSIG) > Unified CM > UCCX > Unified CM > Rem SCCP Ph 1 (IPPA)	Passed	
UC712EF.CCM.027	CAC	Unity Connection Call Failing Due to Unavailable Bandwidth	Verify that Voice mail deposit can not be done when policy is mandatory and when there is not enough bandwidth available for unity connection system from target phone.	Rem SCCP Ph A (ds) > Unified CM (DS) > Rem SCCP Ph B > CFNA > Unified CM(DS) > CAC > Unity Connection VM#	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF.CCM.028	CAC	Call Back Notification Under Limited Bandwidth Condition and IPv4/IPv6 Interworking	Verify if a user can place a call from SCCP Phone 1 in central DS site to another Busy Rem SCCP Phone 2, activate the call back feature, free SCCP Phone 2 where there is not enough bandwidth situation available between Phone 1 and Phone 2, with call back notification sent and ultimately call should fail with prompt "not enough bandwidth" on SCCP phone in central DS phone.	SCCP Ph 1 (ds) > Unified CM(DS) > Rem v4 SCCP Ph 2(Busy)	Passed	
UC712EF.CCM.029	IPv6	IPv4/IPv6 Interactions for IPMA in Shared Line Mode and Using Extension Mobility for Manager	Verify IPMA in shared line mode in central DS cluster. Configure IPMA to use EM phone for manager, login as EM user to manager phone such that after EM login, manager has shared line with assistant, make a call to manager phone(SCCP Ph) from SCCP ph in Unified CM(DS).	Rem SCCP Ph (DS) > Unified CM (DS) > SCCP IPMA manager phone	Passed	
UC712EF.CCM.030	IPv6	IPv4/IPv6 Interworking and MTP Interactions During Blind Transfer & CFNA to Unity Connection VM	Verify if a PSTN call made to IPv4 SIP IP phone in central site can do blind transfer to DS SCCP phone in Remote 3 site which has CFNA to Unity Connection VM.	PSTN Ph > MGCP GW > Unified CM (ds) > IPv4 SIP phone > Blind Transfer > Rem SCCP phone (DS) > CFNA > Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF.CCM.031	IPv6	Interaction of BRI PSTN Call from Interop Site to Dual Stack Phone	Verify if a user can make a PSTN call from to SIP phone in interop site, answer the call and do a consult transfer to SCCP DS phone in remote site which is in active-active or active-stand mode.	Rem SCCP Ph 1 > PSTN GW (FXO interface) > Unified CM (Interop site) > SIP ph > XFER_C > Unified CM (ds) > Rem SCCP ph (DS)	Passed	
UC712EF.CCM.032	MTP	PSTN Caller is Consult Transferred to VG224 Analog Phone	Verify if a PSTN call from central DS cluster to Rem SCCP (ds) Phone can consult transfer to VG224 Analog phone.	PSTN > H323 GW > Unified CM (DS) > Rem SCCP Phone (ds) > Consults > VG224 Ph > Answers > Completes Transfer	Passed	
UC712EF.CCM.033	IPv6	Conferencing During Load Balancing Condition with IPv4/IPv6 Interworking	Verify the following: SIP IPv4 phone calls SCCP (ds) phone in one remote, SIP IPv4 phone calls SCCP (ds) ph in another remote site which is in active-active / active-standby mode. When remote 3a goes down, connected call can be handled by remote 3b with out any interruption. Having established the call between SIP IPv4 and SCCP (ds), SIP IPv4 phone completes conference by pressing conf softkey.	SIP ipv4 > Unified CM > Rem SCCP ph	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF.CCM.034	IPv6	IPv4/IPv6 Interworking and MTP Interactions During Call back and DND scenario	Verify the following: Call from a SCCP phone (ds) in central DS cluster which has Do Not Disturb (DND) feature activated to a remote SCCP phone that is busy. The SCCP phone in the central site should have Callback activated. Make a call from a DS Central SCCP phone to Remote SCCP phone to get a busy tone. Central SCCP phone should press Callback softkey and go on hook. Originating SCCP phone should be alerted to say called remote SCCP phone is free.	SCCP Ph (DS) > Unified CM (DS) > Rem SCCP Ph (dnd activated) Call gets connected	Passed	
UC712EF.CCM.035	IPv6	Multiple Line Appearance & Join Operation with IPv4/IPv6 Interworking	Verify the following: SCCP phone (ds) in central site has 2 lines. Make a call from Rem SCCP ph to SCCP phone in central site on line 1. Again make another call from SIP ph in central site to SCCP ph on line 2, now select both the calls on both lines and press Join Key.	Rem SCCP Ph > Unified CM (DS) > SCCP Ph Line 1 (DS	Passed	
UC712EF.CCM.036	IPv6	IPv4/IPv6 Interworking in SRST Mode during Unified CM Failure	Verify if a user can make a call from SCCP (DS) phone from central DS site to remote SIPv4 phone in SRST mode which does blind transfer to PSTN phone.	SCCP Ph (ds) > Central SRST > Rem SRST > Rem SIP Ph	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF.CCM.037	IPv6	IPv4/IPv6 Interworking for a Unified SRST Remote Call to Central Site User Which goes to Unity Connection VM	Verify the following: Make a PSTN call from remote SIP IPv4 phone to central DS SCCP Phone, remote site has WAN down so the call goes over PSTN to central site which has CFNA to Unity Connection VM.	Rem Ph > Rem SRST > PSTN > Central SCCP Ph	Passed	
UC712EF.CCM.038	IPv6	Legacy and Dual Stack Phone Interop with Dual-Stack SCCP Analog Line Gateway	Verify if a PBX phone from a non-dual stack site can call to an Dual Stack phone in a dual stack site and then can be transferred to a FXS phone over SCCP gateway.	PBX ph1->QSIG trunk->Unified CM (Non DS)->QSIG ICT->Unified CM (DS)->SCCP ph1 (DS)->XFER->SCCP GW(DS)->FXS Ph1	Passed	
UC712EF.CCM.052	Blind Conference call	Blind Conference of PSTN phone ~ SIP Phone Call to a CME Phone	Verify the call from a PSTN Phone to SIP Phone, and the SIP Phone blind conferences the call to CME phone.	PSTN 1->MGCP PRI->CCM->SIP PHONE->Blind conference->IP-IP GW (H.323)->CME phone	Passed	
UC712EF.CCM.054	Consult Conference Call	Consult Conference of PBX Phone, SCCP Phone and CME Phone	Verify if a Consult Conference call among PBX phone,SCCP Phone and CME Phone under same cluster, where the SCCP Phone and CME phone are in same geo-location is possible.	Phone->QSIG PBX->CCM->SCCP Ph ->Consult conference->H.323 IP-IP GW->CME phone	Passed	
UC712EF.CCM.057	Call Forwarding	Call Forwarding Call from IP call to CME phone	Verify if a user can make a Call forwarding in IP call to a CME phone call.	SIP ph->CCM->SCCP Ph->CALL FORWARD->H.323(IP-IP)GW->CME ph	Passed	
UC712EF.CCM.058	Call Forwarding Call	Call Forwarding Call from a IP call to PSTN Phone	Verify if a user can make a Call forwarding in IP call to a PSTN phone call in another geo-location.	Phone->Qsig PBX->CCM->SCCP Ph->CALL FORWARD->>MGCP PRI->PSTN	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF.CCM.059	Park Retrieval	Park Retrieval from Different Cluster	Verify if SIP Phone From Cluster 1 calls SIP phone in Cluster2 and the SIP Phone in Cluster 2 parks the call, and Call parking retrieval is done from PSTN phone.	SIP Phone->CCM->ICT->CCM->SIP ph;SIP Ph Parks the call;	Passed	
UC712EF.CCM.060	EM and JAL	Extension Mobility Call in Same Cluster	Verify the following scenario: SCCP Phone A EM login to another SCCP Phone B in same Cluster, where both phones are in different geo-locations. SIP Phone A makes a PSTN call to SIP phone B in geo-location X and SIP phone B transfer the call to SCCP ph A which had done EM and now in Geo-location Y. The call transfer should be denied.	SCCP Ph 1->CCM->EM->SCCP Ph 2 ;SIP Ph 1->MGCP PRI->PSTN->CCM->SIP Ph 2;SIP Ph 2->Blind transfers->CCM->SCCP Ph 2	Passed	
UC712EF.CSF.003	Client Services Framework	Call from PSTN Phone to UC Integration™ for Microsoft Office Communicator	Verify the call from PSTN Phone to UC Integration for Microsoft Office Communicator.	PSTN Ph->MGCP PRI GW->Unified CM->SCCP Ph->Xfer->MOC + CSF	Passed	
UC712EF.CSF.006	Client Services Framework	Call from an Interop Site SCCP Phone to UC Integration™ for Microsoft Office Communicator	Verify the call from an Interop site SCCP phone to UC Integration for Microsoft Office Communicator.	SCCP ph1->Unified CM(Interop)->QSIG ICT->Unified CM->MOC + CSF	Passed	
UC712EF.CSF.007	Client Services Framework	Call from Unified CME Site Phone to UC Integration™ for Microsoft Office Communicator	Verify the call from Unified CME site phone to UC Integration for Microsoft Office Communicator.	SCCP ph->CME->H.323 GK->H.323 IP-IP-GW->Unified CM->MOC + CSF	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF.CSF.012	Client Services Framework	UC Integration™ for Microsoft Office Communicator interaction with Unified CCX	Verify the UC Integration for Microsoft Office Communicator interaction with Unified CCX as a caller phone.	MOC + CSF->Unified CM->UCCX->CAD Agent	Passed	
UC712EF.CSF.015	Client Services Framework	UC Integration™ for Microsoft Office Communicator interaction with Unified Attendant Server Attendant Console	Verify the UC Integration for Microsoft Office Communicator interaction with Unified Attendant Server Attendant Console.	SCCP ph->Unified CM->Arc Console->Xfer->Unified CM->MOC + CSF	Passed	
UC712EF.CSF.016	Client Services Framework	Microsoft Office Communicator Behavior When Connection to OCS is Lost	Verify the Microsoft Office Communicator behavior when connection to OCS is lost.		Passed	
UC712EF.CSF.023	Client Services Framework	Interop Site QSIG PBX Phone to UC Integration™ for Microsoft Office Communicator	Verify the Interop site QSIG PBX phone to UC Integration for Microsoft Office Communicator.	PBX ph->QSIG PBX->QSIG Trunk->Unified CM (Interop)->QSIG ICT->Unified CM->CSF + MOC	Passed	
UC712EF.QSG.003	QSIG per trunk	Inter PBX Call via Unified CM clusters		PBX ph1->QSIG trunk->Unified CM->QSIG ICT ->Unified CM->QSIG Trunk->PBX ph1->CFNA->QSIG Trunk->Unified CM->Unity	Passed	
UC712EF.QSG.005	QSIG per trunk	Interaction with Callback Feature		SCCP Ph1->Unified CM->QSIG ICT->Unified CM->QSIG ICT->Unified CM ->QSIG trunk->Westell->PBX Ph1	Passed	
UC712EF.QSG.016	Call Diversion by Reroute	Call diversion by reroute ,EM and VM		EMA1->Unified CM 1->ICT->Unified CM 2->Ph1->CFNA->ICT->Unified CM 3 >VM	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC712EF.QSG.020	Path replacement in Trombone call	Path Replacement in Trombone Call Involving DPNSS PBX and Two Line IP Phone		Ph1->Unified CM 1->ICT->Unified CM 2->Ph1_L1->CFB->Ph1_L2->XFER->QSIG Trunk->Westell->PBX Ph1->XFER->Westell ->QSIG Trunk->Unified CM 1->Ph2	Passed	
UC712IF.CUSP.002	Cisco Unified SIP Proxy	Call From SIP Unified CME Site to Unified CM Site Via Cisco Unified SIP Proxy	Verify Unified CME SIP Phone registered with ISR can place a call through Cisco Unified SIP Proxy 1 and Cisco Unified SIP Proxy 2 to Unified CM Phone.		Passed	
UC712IF.CUSP.004	Unified SIP Proxy	Unified CME Phone Calling PSTN Phone via Unified SIP Proxy	To verify calls from Unified CM Phone to PSTN Phone via Unified SIP Proxy.		Passed	
UC712IF.CUSP.006	Unified SIP Proxy	Verify Supplementary Services via Unified SIP Proxy	Verify call from Unified CME Phone to Unified CM Phone and perform the following: a. Call Transfer b. Hold and resume c. Call Forward		Passed	
UC712IF.CUSP.009	Cisco Unified SIP Proxy	Call From Unified CME Phone to Unified CCX via Cisco Unified SIP proxy	Verify the calls from Unified CME Phone to Unified CCX through Cisco Unified SIP proxy.		Passed	
UC712IF.CUSP.010	Cisco Unified SIP Proxy	Unified IP Phone 9971 Video Call Between Unified CM via Cisco Unified SIP Proxy	To verify the inter cluster calls between Unified IP Phones 9971 and Unified CM Phone through Cisco Unified SIP Proxy.		Passed	
UC712IF.IPC.001	IP Communicator	Encrypted SIP IP Communicator Joining Secure Adhoc Conference Using iLBC	To verify if a user can place an encrypted SIP IP Communicator call and join a secure adhoc conference using the iLBC codec.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713EF.ARC.056	Unified Attendant Server	Multichannel Conferencing by Operator Console		Phone 1->Unified CM->Op Console;Opconsole->Unified CM->Conference->Extension A;Opconsole->Unified CM->Conference->Extension B	Passed	
UC713EF.CCM.152	Tandem Cluster	Interaction with SIP and Annex M1 trunks with call Transfer remote			Passed	
UC713EF.CCM.153	Tandem Cluster	Interaction with SIP and Annex M1 trunks with call Transfer Local			Passed	
UC713EF.CCM.156	Tandem Cluster	Interaction with SIP and Annex M1 trunks with QSIG PBX			Passed	
UC713EF.CCM.160	Tandem Cluster	Interaction of Tandem SIP trunks with iLBC			Passed	
UC713EF.CCM.164	Tandem Cluster	Interaction of Tandem Unified CM with Round Robin trunks			Passed	
UC713EF.CCM.201	Direct Call Park	Direct Call Park to a Different Geo-location	Verify the following: SIP Phone 1 uses it PSTN line to call another SIP Phone 2 in same geo-location. SIP Phone 2 directly park the call to a VOIP Phone in Different Geo-location.	SIP Ph A->Unified CM->MGCP PRI->PSTN Gateway->MGCP PRI->SIP Ph B;SIP Ph B-->Direct Call Park->Unified CM->SIP Ph C.	Passed	
UC713EF.CCM.202	Call Pick up	Call Pick up from Different Geo-location	Verify when a SCCP Phone A uses its PSTN line to make PSTN call to another SCCP Phone B in same geo-location X. A SIP Phone in Geo-location Y of same Pick group of SCCP Phone B tries to pick the call.	SCCP Ph A->Unified CM->MGCP PRI->PSTN->MGCP PRI->SCCP Ph B;SIP Ph A->call pick up->SCCP Ph B	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713EF.CCM.203	Meet Me conference	Meet Me Conference among Different Geo-location Phone	Verify the following scenario: SCCP Phone A in Geo-location X attends meet conference through VOIP line. SCCP Phone B in Geo-location Y attends meet conference through VOIP Line. SIP Phone A of Geo-location X attends Meet me conference through PSTN Line.	SCCP Ph A->CCM->Meet me Conference; SCCP PhB->CCM->meet me conference;SIP PhA->MGCP PRI->PSTN->Meet me conference;	Passed	
UC713EF.CCM.204	Shared Line	Shared Lines in Different Geo-location	Verify the following scenario: SCCP Phone A and SCCP Phone B are shared line Phones and SCCP Ph A is in Geo-location X and SCCP Phone B in Geo-location Y. When PSTN call in Geo-location X is transferred as VOIP call to Shared lines, only SCCP Phone A should be able to accept the call.	SCCP Ph C->Unified CM->SIP Ph A;SIP Ph A->Blind transfer->Unified CM->SCCP Ph A(Shared line with SCCP B);	Passed	
UC713EF.CSF.051	Client Services Framework	UC Integration™ for Microsoft Office Communicator Click-to-Call Feature	Verify the UC Integration for Microsoft Office Communicator click-to-call feature.	UC Integration™ for Microsoft Office Communicator 1->Unified CM->UC Integration™ for Microsoft Office Communicator 2	Passed	
UC713EF.CSF.052	Client Services Framework	UC Integration™ for Microsoft Office Communicator Multi-Party Conference	Verify the UC Integration for Microsoft Office Communicator multi-party conference.	UC Integration™ for Microsoft Office Communicator 1->Unified CM->UC Integration™ for Microsoft Office Communicator 2->CONF->UC Integration™ for Microsoft Office Communicator 3	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713EF.CSF.055	Client Services Framework	UC Integration™ for Microsoft Office Communicator Multi-Party Conference with PBX and PSTN Phones	Verify the UC Integration for Microsoft Office Communicator multi-party conference with PBX and PSTN phones.	UC Integration™ for Microsoft Office Communicator->Unified CM->QSIG Trunk->PBX->PBX Ph1->CONF->MGCP PRI GW->PSTN Ph1	Passed	
UC713IF.CME.001	Unified CME	Parallel/Serial Hunt with Transcoder Invoked on Unified CME for Codec Mismatch	Verifies whether Transcoder is invoked on Unified CME for codec mismatch on IP Phone 6921/6941/6961	IP Phone 6921/6941/6961->Unified Communications Manager->Unified Border Element->Unified CME->Hunt Pilot ---Parallel Hunt group -- (xcoder Invoked)->IP Phone 6921/6941/6961	Passed	
UC713IF.CUB.001	Unified Border Element	iSAC Transcoding on Unified Border Element for SIP / H323 Trunk	Verifies Transcoding support between iSAC ((g711. g729 , g722 , iLBC.		Passed	
UC713IF.CUB.003	Unified Border Element	sRTP to RTP Interworking	Verify Interworking between sRTP and RTP.	Unified Communications Manager->sRTP->Unified Border Element->RTP->Unified CME	Passed	
UC713IF.CUM.001	Unified Mobility	Secure VM Download When VM Servers and Unity Connection in Active-Active Mode	To verify if an iPhone Unified Mobile Communicator client can download the Voicemail from paired Unity Connection server when the Voicemail servers and Unity Connection is in Active-Active mode.	iPhone-(Unified Mobility--(Unity Connection	Passed	
UC713IF.CUM.001	Unified Mobility	Secure VM Download When VM Servers are Unity Connection in Active-Active Mode	Verify iPhone Unified Mobile Communicator client can download the Voicemail from paired Unity Connection Server when the Voicemail servers are Unity Connection in Active-Active mode.	iPhone->Unified Mobility Advantage->Unity Connection	Failed	CSCtd35094

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713IF.CUM.002	Unified Mobility	Secure VM Download When VM Servers and Unity Connection in Digital Networking Mode	To verify if an iPhone Unified Mobile Communicator client can download Voicemail when the Voicemail servers are Unity Connection in Digital networking mode.	iPhone-(Unified Mobility--(Unity Connection	Passed	
UC713IF.CUM.002	Unified Mobility	Secure VM download when VM servers are Unity Connection in Digital Networking mode	Verify iPhone Unified Mobile Communicator client can download the Voicemail when the Voicemail servers are Unity Connection in Digital networking mode.	iPhone->Unified Mobility Advantage->Unity Connection	Failed	CSCtd35094
UC713IF.CUM.003	Unified Mobility	Voicemail Download Onto Unified Mobile Communicator in Unified Messaging Environment	To verify if an iPhone Unified Mobile Communicator client can download the Voicemail from Cisco Unity where Cisco Unity is in Unified Messaging environment.	iPhone-(Unified Mobility--(Unity Connection	Passed	
UC713IF.CUM.003	Unified Mobility	Voicemail Download onto Unified Mobile Communicator in a Unified Messaging Environment	Verify iPhone Unified Mobile Communicator client can download the Voicemail from Unity where Unity is in Unified Messaging environment.	iPhone->Unified Mobility Advantage->Unity Connection	Passed	
UC713IF.CUM.004	Unified Mobility	Secure VM Download When VM Servers are Cisco Unity in Digital Networking Mode	To verify if an iPhone Unified Mobile Communicator client can download the Voicemail from both Cisco Unity servers when 2 Voicemail servers are Cisco Unity in Digital Networking mode.	iPhone-(Unified Mobility-(Unity	Passed	
UC713IF.CUM.007	Unified Mobility	Unity Connection in Active-Active Configuration	To verify if an iPhone Unified Mobile Communicator client can download a reply Voicemail from Unity Connection in active-active mode with one of them down.	iPhone-(Unified Mobility--(CUV	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713IF.CUM.007	Unified Mobility	Unity Connection in Active-Active Configuration Downloading of Replied VM	Verify iPhone Unified Mobile Communicator client can download a reply Voicemail from Unity Connection in active-active mode with one of them down.	iPhone->Unified Mobility Advantage->Unity Connection	Failed	CSCtd35094
UC713IF.CUM.008	Unified Mobility	Directory Search and DVO-F to Intercluster Destination	To verify if an iPhone Unified Mobile Communicator client can reach an enterprise DN across a SIP trunk, by performing a directory search.	iPhone-(Unified Mobility---Unified CM1---<sip secure>---Unified CM2---IPPhone	Passed	
UC713IF.CUM.008	Unified Mobility	Directory Search and DVO-F to Intercluster Destination	Verify iPhone Unified Mobile Communicator client can reach an enterprise DN which is across a SIP trunk. The user is located by doing a directory search.	iPhone->Unified Mobility Advantage->Unified Communications Manager1-><SIP Secure>->Unified Communications Manager2->Unified IP Phone	Passed	
UC713IF.CUM.010	Unified Mobility	Voicemail Access When Number is Set For CFNA to VM	To verify if an iPhone Unified Mobile Communicator client can reach the voicemail of a DN when the called number is set for CFNA to VM. The desk phone of the iPhone client is also set for CFA to another destination and the user's DN is found by performing a directory search on AD2008.	iPhone-(Unified Mobility--(Unified CM-(Unity	Passed	
UC713IF.CUM.012	Unified Mobility	DVO-F, Unified Mobile Communicator Clients Joining Conference	To verify that if an iPhone user can join meetme conference by dialing the meetme conference number.	iPhone-(Unified Mobility--(Unified CM-(IPPhone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713IF.CUM.015	Unified Mobility	Callback on Unified MeetingPlace Only Meeting	To verify that Unified Mobile Communicator gets the meeting list for all types of meetings and call back works for Unified MeetingPlace only meetings.	iPhone-(Unified Mobility--(Unified CM-(Unified MeetingPlace+Exchange	Passed	
UC713IF.CUM.015	Unified Mobility	Unified MeetingPlace only Meeting, Meeting List and Unified MeetingPlace Call Back	Verify that Unified Mobile Communicator gets the meeting list for all types of meetings and call back work for a Unified MeetingPlace only meeting.	iPhone->Unified Mobility Advantage->Unified Communications Manager->Unified MeetingPlace+Exchange	Passed	
UC713IF.CUM.017	Unified Mobility	Unified MeetingPlace and WebEx Hybrid Meeting	To verify that Unified Mobile Communicator gets the meeting list for all types of meetings and WebEx client can be launched, where the WebEx client assists in joining the meeting.	iPhone-(Unified Mobility--(Unified CM-(Unified MeetingPlace+Exchange	Passed	
UC713IF.CUM.017	Unified Mobility	Unified MeetingPlace and WebEx Hybrid Meeting, Meeting List and Call Back with Launch of WebEx Meeting Client	Verify that Unified Mobile Communicator gets the meeting list for all types of meetings and WebEx client can be launched. The WebEx client assists in joining the meeting.	iPhone->Unified Mobility Advantage->Unified Communications Manager->Unified MeetingPlace+Exchange	Failed	CSCtd61787
UC713IF.CUM.021	Unified Mobility	Fail and Failback of Active Unified CM Server	To verify that iPhone Unified Mobile Communicator client gets call logs and DVO works when active Unified CM is down. Also after failback of Unified CM, Unified Mobility reinstate the communication to active Unified CM.	iPhone-(Unified Mobility--(Unified CM-(	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713IF.CUM.021	Unified Mobility	Fail and Failback Active Unified CM server	Verify that iPhone Unified Mobile Communicator client keeps getting call logs and DVO works when the active Unified CM is down. Also after the failed Unified CM is back, Unified Mobility Advantage reinstate the communication to active Unified CM.	iPhone->Unified Mobility Advantage->Unified Communications Manager	Failed	CSCtd26931
UC713IF.CUM.022	Unified Mobility	VM Download When Voicemail is Deposited Using VPIM	To verify if an iPhone Unified Mobile Communicator client can download the Voicemail from both Unity Connection servers when 2 Voicemail servers are in Digital Networking mode.	iPhone-(Unified Mobility--(Unity Connection	Passed	
UC713IF.CUM.022	Unified Mobility	VM Download When Voicemail is Deposited Using VPIM	Verify iPhone Unified Mobile Communicator client can download the Voicemail from both Unity Connection servers when 2 Voicemail servers are in Digital Networking mode.	iPhone->Unified Mobility Advantage->Unity Connection	Failed	CSCtd35094
UC713IF.SRS.001	Unified SRST	Failover and Fallback Features in IP Phone 6921/6941/6961 in SRST mode	Verifies whether IP Phone 6921/6941/6961 can Failover to SRST Mode when WAN Link or Unified CM is down and FallBack to HQ Site when WAN Link recovers.	6921/6941/6961->Unified Communications Manager->WAN->SRST Gateway->Fail WAN Link->SRST Mode Ephone->PSTN->HQ Site->Phone	Passed	
UC713IF.SRS.002	Unified SRST	Dial International Number Preceded by + from IP Phone 6921/6941/6961 Registered to Unified SRST Site	Verify whether IP Phone 6921/6941/6961 in Unified SRST Site can dial International number preceded by +.	6921/6941/6961->Unified Communications Manager->WAN->SRST Gateway->Fail WAN Link->SRST Mode Ephone->Phone (dial +1 followed digits by International number)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC713IF.SRS.005	Unified SRST	IP Phone 6921/6941/6961 Access to Voicemail After FallBack	Verify whether IP Phone 6921/6941/6961 can Failover to SRST Mode when WAN Link is Down and Fallback to HQ Site when WAN Link is recovered or if Unified CM is down and is able to access voicemail located in main site.	6921/6941/6961->Unified Communications Manager->WAN->SRST Gateway->Fail WAN Link->SRST Mode Ephone->PSTN->Unified IP Phone(main site)	Passed w/ Exception	CSCtc66818
UC802.CCM.201	Session Management Edition	UC Integration™ for Microsoft Office Communicator Calls Over Unified CM Session Management Edition	Verify that UC Integration for Microsoft Office Communicator clients in leaf clusters can call each other over Unified CM Session Management Edition (audio calls).	UC Integration™ for Microsoft Office Communicator->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->UC Integration™ for Microsoft Office Communicator	Passed	
UC802.CCM.201	Session Management Edition	UC Integration™ for Microsoft Office Communicator Calls Over Unified CM Session Management Edition	Verify that UC Integration for Microsoft Office Communicator clients in leaf clusters can call each other over Unified CM Session Management Edition(audio calls).	UC Integration™ for Microsoft Office Communicator->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->UC Integration™ for Microsoft Office Communicator	Passed	
UC802.CCM.201	Session Management Edition	UC Integration™ for Microsoft Office Communicator Calls Over Unified CM Session Management Edition	Verify that UC Integration for Microsoft Office Communicator clients in leaf clusters can call each other over Unified CM Session Management Edition(audio calls).	UC Integration™ for Microsoft Office Communicator->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->UC Integration™ for Microsoft Office Communicator	Passed	
UC802.CCM.202	Session Management Edition	VG224 Calls Over Unified CM Session Management Edition	Verify that VG224 clients in leaf clusters can call each other over Unified CM Session Management Edition.	Vg224->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->Vg224	Passed	
UC802.CCM.202	Session Management Edition	VG224 Calls Over Unified CM Session Management Edition	Verify that VG224 clients in leaf clusters can call each other over Unified CM Session Management Edition.	Vg224->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->Vg224	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802.CCM.202	Session Management Edition	VG224 Calls Over Unified CM Session Management Edition	Verify that VG224 clients in leaf clusters can call each other over Unified CM Session Management Edition.	Vg224->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->Vg224	Passed	
UC802.CCM.203	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using UC Integration™ for Microsoft Office Communicator Clients	Verify that UC Integration for Microsoft Office Communicator clients in leaf clusters can make a video call over Unified CM Session Management Edition (video calls).	UC Integration™ for Microsoft Office Communicator->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->UC Integration™ for Microsoft Office Communicator	Passed	
UC802.CCM.204	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using UC Integration™ for Microsoft Office Communicator Clients and Unified Video Advantage	Verify that UC Integration for Microsoft Office Communicator clients and Unified Video Advantage in leaf clusters can make a video call over Unified CM Session Management Edition	UC Integration™ for Microsoft Office Communicator->Unified CM1->QSIG ICT->Unified CM-Tand->QSIG ICT->Unified CM2->CUVA	Passed	
UC802.CCM.204	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using UC Integration™ for Microsoft Office Communicator Clients and Unified Video Advantage	Verify that UC Integration for Microsoft Office Communicator clients and Unified Video Advantage in leaf clusters can make a video call over Unified CM Session Management Edition.	UC Integration™ for Microsoft Office Communicator->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->CUVA	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802.CCM.204	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using UC Integration™ for Microsoft Office Communicator Clients and Unified Video Advantage	Verify that UC Integration for Microsoft Office Communicator clients and Unified Video Advantage in leaf clusters can make a video call over Unified CM Session Management Edition.	UC Integration™ for Microsoft Office Communicator->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->CUVA	Passed	
UC802.CCM.205	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using UC Integration™ for Microsoft Office Communicator Clients and 69xx series IP Phones	Verify that UC Integration for Microsoft Office Communicator clients and 69xx series IP phones in leaf clusters can make a video call over Unified CM Session Management Edition.	UC Integration™ for Microsoft Office Communicator->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->RT	Passed	
UC802.CCM.205	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using UC Integration™ for Microsoft Office Communicator Clients and 69xx series IP phones	Verify that UC Integration for Microsoft Office Communicator clients and 69xx series IP phones in leaf clusters can make a video call over Unified CM Session Management Edition.	UC Integration™ for Microsoft Office Communicator->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->RT	Passed	
UC802.CCM.205	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using UC Integration™ for Microsoft Office Communicator Clients and 69xx series IP phones	Verify that UC Integration for Microsoft Office Communicator clients and 69xx series IP phones in leaf clusters can make a video call over Unified CM Session Management Edition.	UC Integration™ for Microsoft Office Communicator->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->RT	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802.CCM.206	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using Unified Video Advantage and 69xx series IP Phones	Verify that Unified Video Advantage clients and 69xx series IP phones in leaf clusters can make a video call over Unified CM Session Management Edition.	CUVA->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->RT	Passed	
UC802.CCM.206	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using Unified Video Advantage and 69xx series IP phones	Verify that Unified Video Advantage clients and 69xx series IP phones in leaf clusters can make a video call over Unified CM Session Management Edition.	CUVA->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->RT	Passed	
UC802.CCM.206	Session Management Edition	Video Calls Over Unified CM Session Management Edition Using Unified Video Advantage and 69xx series IP phones	Verify that Unified Video Advantage clients and 69xx series IP phones in leaf clusters can make a video call over Unified CM Session Management Edition.	CUVA->Unified CM1->SIP->Unified CM-Tand->SIP->Unified CM2->RT	Passed	
UC802.CCM.207	Session Management Edition	Service Advertisement Framework (SAF) Over Unified CM Session Management Edition		Route Pattern->Unified CM1->SAF->Unified CM-Tand->SAF->Unified CM2	Passed	
UC802.CCM.208	Session Management Edition	RSVP Over Unified CM Session Management Edition		Unified CM1(RSVP)->SIP->Unified CM-Tand->SIP->(RSVP)Unified CM2	Passed	
UC802.DPN.001	DPNSS VG30D Converter	Call Back in RDX Extension	Verify if RDX extension can invoke Call Back when free against IP-Phone in Legacy Unified CM Cluster.	PBX Ph1->PBX->Westell->Unified CM->ICT(QSIG)->Unified CM->SCCP/SIP Ph1	Passed	
UC802.DPN.002	DPNSS VG30D Converter	Call Forward in RDX Extension	Verify if RDX extension can initiate a call which is forwarded by SIP Phone in CDG Central, Video endpoint and VG248 endpoint back to the originating RDX extension.	PBX Ph1->PBX->Westell->Unified CM->CDG Central SIP Ph1->transfer->VG248 Pots Ph1->Unified CM->Westell->PBX->PBX Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802.DPN.003	DPNSS VG30D Converter	Call Forward with DPNSS Phone	Verify if RDX extension can initiate a call which is forwarded by SCCP Phone, and transfers to Unified CME and IP Phone on CME back to the originating RDX extension.	PBX Ph1->PBX->Westell->Unified CM->CDG Central SCCP Ph1->Transfer-> CME->SCCP Ph1->CFA->Unified CM->Westell-> PBX->PBX Ph1	Passed	
UC802.DPN.004	DPNSS VG30D Converter	iSDX Extension with Cisco Unity	Verify if iSDX extension can call into Unity Connection VoiceMail and retrieve a previously deposited message.	PBX Ph1->PBX->Westell->Unified CM->ICT (QSIG)->Unified CM->Unity	Passed	
UC802.DPN.005	DPNSS VG30D Converter	Call Forward in iSDX Extension	Verify if iSDX extension can initiate a call which is forwarded by SIP Phone in CDG Remote, SIP Phone on CSPS and M1 extension back to the originating iSDX extension.	PBX Ph1->PBX->Westell->Unified CM->CDG Remote SIP Ph1->iDivert->SIP Trunk->CSPS->SIP Ph1->Westell->PBX->PBX Ph2->Unified CM->Westell->PBX->PBX Ph1	Passed	
UC802EF.ARC.005	Unified Attendant Server	Unified IP Phone 69x1 as Operator Endpoint	Verify if IP Phone 69x1 acting as Operator end point can Blind transfer (XFER_B) the incoming call to UC Integration™ for Microsoft Office Communicator clients.	Stage 1: SIP Ph 1->Unified CM->Op Cons 1 (IP Phone 69xx); Stage 2: Op Cons 1->XFER_B->UC Integration™ for Microsoft Office Communicator	Passed w/ Exception	The UC Integration™ for Microsoft Office Communicator end point is replaced with SIP/SCCP
UC802EF.ARC.006	Unified Attendant Server	BLF Monitoring in Unified Attendant Server	Verify the following: A video call made to Unified Enterprise Attendant Console (CUEAC) where IP Phone 6961 with Camera acts as the Operator. This video call is transfer to another IP Phone 9971 with Camera in the same network .The BLF status on the IP Phone 9971 with Camera changes to busy.	Stage 1: Video Ph 1->Unified CM->OP Cons 1 (6961+CAM); Stage2: Op Cons 1(6961+CAM)->XFER_C->Video Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.ARC.007	Unified Attendant Server	Cisco Unity as Queue Overflow Designations	Verifying the overflow of Unified Enterprise Attendant Console (CUEAC) Queue calls transfer to Voice mail.	Stage 1 : SCCP Ph 1->Unified CM->OP Cons 1(UC Integration™ for Microsoft Office Communicator);Stage 2: OP Cons 1->Voice mail.	Passed	
UC802EF.ARC.008	Unified Attendant Server	E164 Support in Unified Attendant Server	Verify if an international call from Phone 1 can call Operator of Unified Enterprise Attendant Console (CUEAC) .The CUEAC Consult Transfer (XFER_C) the call to PSTN Phone. The CUEAC operator must show the E164 numbering of the calling party.	Stage 1 : SCCP Ph 1->Unified CM->OP Cons 1(SIP Ph 1);Stage 2 : OP Cons 1->XFER_C->Unified CM->MGCP PRI GW->PSTN->MGCP PRI GW->Unified CM->Rem SIP Ph 1	Passed	
UC802EF.ARC.010	Unified Attendant Server	Enhanced Night Service in Unified Attendant Server	Verify if all calls to the Operator lands on a phone designated on the Admin page to attend Night service calls, when Unified Enterprise Attendant Console operator IP Phone 9971 is in Night service mode.	Stage 1 : SIP Ph 1->Unified CM->OP Cons Ph 1(9971 Phone)->Night service;Stage 2: SIP Ph1->Unified CM->MGCP BRI GW->PSTN->MGCP BRI GW->Unified CM->Rem SCCP Ph 1	Passed w/ Exception	Only basic night service is available in Unified Attendant Console build for UC 8.0 release.
UC802EF.ARC.011	Unified Attendant Server	Enhanced Emergency Mode in Unified Attendant Server	Verify is all calls to the Operator lands on the Emergency number designated on Admin page, when Unified Enterprise Attendant Console operator is set to Emergency mode.	Stage 1 : SIP Ph 1->Unified CM->OP Cons Ph (9971 Phone)->Emergency Mode;Stage 2: SIP Ph1->Unified CM->MGCP BRI GW->PSTN->MGCP BRI GW->Unified CM->Rem SCCP Ph 1	Passed w/ Exception	Only emergency service is available in Unified Attendant Console build for UC 8.0 release.

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.ARC.014	Unified Attendant Server	Unified IP Phone 7931 as Operator Endpoint	Verify if Unified IP Phone 7931 Phone acting as Operator end point can consult transfer (XFER_C)s the incoming call to SIP phone across clusters.	SIP Ph 1->Unified CM->Op Cons 1 (7931 Ph); Stage 2: Op Cons 1->XFER_C->Unified CM1->ICT->Unified CM2->SIP Ph.	Passed w/ Exception	Replaced 7931 Phones with 6961 phones since 7931 phones are not supported as Operator Console.
UC802EF.ARC.018	Unified Attendant Console	Unified IP Phone 6901/6911 as Operator End Points	Verify if Unified IP Phone 6901/6911 can act as operator end point and can blind transfers (XFER_B) the incoming call to UC Integration for Microsoft Office Communicator Clients.	Stage 1: SIP Ph 1->Unified CM->Op Cons 1 (IP Phone 69xx); Stage 2: Op Cons 1->XFER_B->UC Integration™ for Microsoft Office Communicator 1	Passed	
UC802EF.ARC.019	Unified Attendant Server	BLF Monitoring in Unified Attendant Server	Verify if a video call made to Unified Business Assistant Console where SCCP Phone with camera is the Operator. This video call is transferred to another SCCP Phone 2 with Camera in the same network. The BLF status on the SCCP Phone 2 with Camera changes to busy.	Stage 1: Video Ph 1->Unified CM->OP Cons 1 (SCCP Ph+CAM); Stage2: Op Cons 1(SCCP Ph+CAM)->XFER_C->Video Ph 2	Passed	
UC802EF.ARC.020	Unified Attendant Server	Cisco Unity as Queue Overflow Designations	Verify the overflow of Unified Business Attendant Console Queue calls transfer to Voice mail.	Stage 1 : SCCP Ph 1->Unified CM->OP Cons 1(UC Integration™ for Microsoft Office Communicator);Stage 2: OP Cons 1->Voice mail.	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.ARC.021	nified Attendant Server	E164 Support in Unified Attendant Server	Verify if an international call from Phone 1 can call Operator of Unified Business Attendant Console. The Unified Business Attendant Console Consult transfers (XFER_C) the call to PSTN Phone. The Unified Business Attendant Console operator must display the E164 numbering of the calling party.	Stage 1 : SCCP Ph 1->Unified CM->OP Cons 1(SIP Ph 1);Stage 2 : OP Cons 1->XFER_C->Unified CM->MGCP PRI GW->PSTN->MGCP PRI GW->Unified CM->Rem SIP Ph 1	Passed	
UC802EF.ARC.023	Unified Attendant Server	Enhanced Night Service in Unified Attendant Server	Verify if all calls to the Operator lands on a phone designated on the Admin page to attend Night service calls, when Unified Business Attendant Console operator IP Phone 9971 is in Night service mode.	Stage 1 : SIP Ph 1->Unified CM->OP Cons Ph 1(9971 Phone)->Night service;Stage 2: SIP Ph1->Unified CM->MGCP BRI GW->PSTN->MGCP BRI GW->Unified CM->Rem SCCP Ph 1	Passed w/ Exception	Only basic night service is available in Unified Attendant Console build for UC 8.0
UC802EF.ARC.024	Unified Attendant Server	Enhanced Emergency Mode in Unified Attendant Server	Verify is all calls to the Operator lands on the Emergency number designated on Admin page, when Unified Business Attendant Console operator is set to Emergency mode.	Stage 1 : SIP Ph 1->Unified CM->OP Cons Ph (9971 Phone)->Emergency Mode;Stage 2: SIP Ph1->Unified CM->MGCP BRI GW->PSTN->MGCP BRI GW->Unified CM->Rem SCCP Ph 1	Passed w/ Exception	Only basic emergency service is available in Unified Attendant Console build for UC 8.0
UC802EF.ARC.026	Unified Attendant Server	Unified IP Phone 7931 as Operator End Point	Verify if Unified IP Phone 7931 as an Operator end point can Consult transfers (XFER_C)s the incoming calls to SCCP phone.	SIP Ph 1->Unified CM->Op Cons 1 (7931 Ph); Stage 2: Op Cons 1->XFER_C->Unified CM1->ICT->Unified CM2->SCCP Ph.	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.ARC.027	Unified Attendant Server	Unified IP Phone 7931 as Operator End Point	Verify if Unified IP Phone 7931 as Operator end point can blind transfers (XFER_B)s the incoming call to another SCCP phone.	SIP Ph 1->Unified CM->Op Cons 1 (7931 Ph); Stage 2: Op Cons 1->XFER_B->Unified CM1->ICT->Unified CM2->SCCP Ph.	Passed	
UC802EF.ARC.028	Unified Attendant Server	Server Application Restarted with Different CTI Manager as Primary	Verify the following when: Unified Attendant Server is registered to Unified CM1 as primary & Unified CM2 as secondary, purposefully interchange the settings on Unified Attendant Server so as to make Unified CM2 as primary and Unified CM1 as secondary. Restart the Unified Attendant server.		Passed w/ Exception	This TC was executed without having a secondary CTI manager in the Testbed.
UC802EF.ARC.029	Unified Attendant Server	Unified Attendant Console Client Loses Network Connectivity When Active	Verify the following: Make a call from PSTN to operator. When operator answers the call and initiates a transfer to a SCCP phone in remote site, disconnect the operator from the network and examine the behavior. Bring back the system to normal after few minutes and check if the operator is back to normal.		Passed	
UC802EF.ARC.030	Unified Attendant Server	Unified Attendant Console IP Phone Loses Network Connectivity	Verify if the Unified Attendant Console IP Phone loses network connectivity.		Passed	
UC802EF.ARC.031	Unified Attendant Server	Multichannel Conferencing by Operator Console	Verify if multichannel Conferencing by operator console is possible.	Phone 1->Unified CM->Op Console;Opconsole->Unified CM->Conference->Extension A;Opconsole->Unified CM->Conference->Extension B	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.ARC.032	Unified Attendant Server	Retrieval of Call Parked by Operator	Verify if retrieval of call parked by operator is possible.	Unified IP Phones 6921/6941/6961 PH1->Unified CM->OP Console->Call Parking;Unified IP Phones 6921/6941/6961 PH 2->Park Retrieval.	Passed	
UC802EF.CCM.001	Unified CCX	PSTN Call to a Central Site	Verify the call from a PSTN phone to a central site SCCP phone over MGCP gateway which can transfer the call to a Cisco Agent Desktop (CAD) with a SIP phone in central site.	PSTN Ph1->MGCP PRI->PSTN->MGCP PRI->Unified CMBE->SCCP Ph1->XFER_B->Unified CMBE-> UCCX->Unified CMBE->RT SIP Ph1 (CAD)	Passed	
UC802EF.CCM.002	Unified CCX	PSTN Call from a Remote SCCP Phone to a Central Site	Verify the call from a remote SCCP phone to a central site SIP Phone acting as a CAD agent over H.323 gateway (FXO/BRI/PRI) and then transfer the call to remote agents.	Rem SCCP Ph1->Unified CMBE->H323 GW->PSTN->H323 GW->Unified CMBE->UCCX->Unified CMBE->SIP Ph1 (CAD)->XFER_C->Unified CMBE->UCCX->Unified CMBE->Rem SCCP Ph2 (CAD)	Passed	
UC802EF.CCM.003	Unified CCX	PSTN Call from a Remote SCCP Phone to a Central Site	Verify the call from a remote SCCP phone to a central site SIP Phone acting as CAD agent over SIP gateway (BRI/PRI) and conference the call to remote agents.	Rem SCCP Ph1->Unified CMBE->SIP GW->PSTN->SIP GW->Unified CMBE->UCCX->Unified CMBE->SIP Ph1 (CAD)->CNF->Unified CMBE->UCCX->Unified CMBE->Rem SCCP Ph2(CAD)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CCM.004	Unified CCX	IM session on IP phones, Remote CAD agent, IP Phones 9971/9951/8961 and Unified Personal Communicator	Verify if IM from IP Phones 9971/9951/8961 and SIP Phone acting as CAD agent in remote sites to Unified Personal Communicator in central site is possible and also check the presence status of all buddies. Also verify if IM from SIP/SCCP (IPPM clients) to IP Phones 9971/9951/8961 and remote site SIP Phone acting as CAD agent is possible and check the presence status of all buddies.	V 1 :Rem RT SIP Ph1->CUP->SCCP Ph/SIP Ph/ CUPC; V 2: Rem SIP Ph1 (CAD)->Unified CMBE->UCCX->Unified CMBE->CUP-> SCCP Ph/SIP Ph/ CUPC	Passed	
UC802EF.CCM.005	Unified CCX	IM session on IP phones, Central CAD agent, IP Phones 9971/9951/8961 and Unified Personal Communicator	Verify if IM from IP Phones 9971/9951/8961 and SIP Phone acting as CAD agent in central sites to Unified Personal Communicator in central site is possible and also check the presence status of all buddies. Also verify if IM from SIP/SCCP (IPPM clients) to IP Phones 9971/9951/8961 and central site SIP Phone acting as CAD agent is possible and also check the presence status of all buddies.	Variation 1 :RT SIP Ph1->CUP->SCCP Ph/SIP Ph/ CUPC Variation 2: SIP Ph1 (CAD)->Unified CMBE->UCCX->Unified CMBE->CUP-> SCCP Ph/SIP Ph/ CUPC	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CCM.006	Unified CCX	IM session on IP phones, Central CAD agent, IP Phones 9971/9951/8961 and Unified Personal Communicator	Verify if IM from IP Phones 9971/9951/8961 and SIP/SCCP (IPPM clients) to Unified Personal Communicator/CAD agent in central site is possible and can also check the presence status of all buddies when both CAD agent and Unified Personal Communicator applications are installed on the same laptop.	RT SIP Ph1/SCCP Ph1/SIP Ph1 ->CUP->CAD agent/ CUPC	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CCM.007	Unified Personal Communicator	Video Calls with RSVP on Video Phones, IP Phones 9971/9951 and Unified Personal Communicator	Verify the following: 1. Video calls can be made between video phones, IP phones, IP Phones 9971/9951/8961 and Unified Personal Communicator with RSVP enabled. 2. IM from IP Phones 9971/9951 and video phone in remote site to Unified Personal Communicator in central site is possible and can also check the presence status of all buddies. 3. IM from SIP/SCCP (IPPM clients) to IP Phones 9971/9951 and Video phone in remote site is possible and can also check the presence status of all buddies. 4. Can exhaust the available video bandwidth on a remote site. 5. Can make a video call which goes as audio only due to lack of video bandwidth.	RT/Video Ph->REM->WAN->CUP->Unified CMBE->SCCP Ph/SIP Ph/ CUPC	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CCM.008	Unified Personal Communicator	Unified Personal Communicator Inter-Working with Extension Mobility	Verify if Unified Personal Communicator in desk phone mode can control a user logged in using extension mobility. A central site user logs into a remote site phone using EM feature. Unified Personal Communicator which was earlier configured for desk phone control of the above user in central site should still work. Check the presence status of this user on Unified Personal Communicator and check IM between Unified Personal Communicator and the EM user.	CUPC->CUP->Unified CMBE->WAN->Rem->EM Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CCM.009	Unified Personal Communicator	Inter-Working with Unified Attendant Server Console	Verify the interworking of Unified Attendant Server and Unified Personal Communicator. The IM session and call transfer from Unified Attendant Server to Unified Personal Communicator in desk phone can control a PSTN call made to a remote site via a FXO. The call lands on the Unified Attendant Server and the Unified Attendant Server console checks the presence status of the user to which the call has to be transferred. An IM is sent to the user to take the call and the call is transferred, Unified Personal Communicator in desk phone mode picks up the call.	PSTN->FXO->REM->Unified CMBE->ARC ARC->Unified CMBE->CUP->CUPC	Passed	
UC802EF.CCM.010	Unified IP Phones 6921/6941/6961	Unified IP Phones 6921/6941/6961 in SRST mode	Verify if a PSTN call can be made from a remote Unified IP Phones 6921/6941/6961 in Unified SRST mode to another remote phone in SRST mode which has CFWDALL set to a PSTN Phone.	Rem RT-Lite SCCP Ph1->SRST1->PSTN GW->PSTN->PSTN GW->SRST 2->Rem SCCP Ph2;Rem SCCP Ph2->CFWDALL->PSTN GW->PSTN Ph1	Passed	
UC802EF.CCM.011	Unified IP Phones 9971/9951/8961	Unified IP Phones 9971/9951/8961 Interworking with IP Communicator	Verify if a call can be made from the SIP IP Communicator to the remote SCCP Phone which has CFNA set to a remote Unified IP Phones 9971/9951/8961.	SIP CIPC1->Unified CMBE-> Rem SCCP Ph1; Rem SCCP Ph1->CFNA->Rem RT SIP Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CCM.012	Unified IP Phones 9971/9951/8961	Unified IP Phones 9971/9951/8961 Interworking with Unified CCX	Verify if a call made from a PSTN phone to a SCCP phone is transferred to Unified CCX CAD agent with a Unified IP Phones 9971/9951/8961.	PSTN Ph1->MGCP PRI->Unified CMBE->SCCP Ph1->XFER_B->Unified CMBE-> UCCX->Unified CMBE->RT SIP Ph1(CAD)	Passed	
UC802EF.CCM.013	Unified IP Phones 9971/9951/8961	Call Park Using IP Phones 9971/9951/8961	Verify if a call from a PSTN Phone to the IP Phones 9971/9951/8961 (SIP Phone 1 and Remote IP Phones 6921/6941/6961 SCCP Phone 2 are shared lines) over PSTN Gateway is successful. IP Phones 9971/9951/8961 parks the call and remote IP Phones 6921/6941/6961 retrieves the call.	Stage 1: PSTN Ph 1->Unified CMBE->PSTN GW->PSTN->PSTN GW->Unified CMBE->RT SIP Ph1; Stage 2: RT SIP Ph1->answers and parks the call; Stage 3: Rem RT-Lite SCCP Ph2->retrieves the call	Passed	
UC802EF.CCM.014	Unified IP Phones 9971/9951/8961	Hunt List using Unified IP Phones 9971/9951/8961	Verify if a call made from a Central SCCP Phone to a Central IP Phone 9971/9951/8961 is not answered, then the call should go to all members in the hunt group until it is answered depending upon the algorithm configured. Hunt group should have a combination of Unified IP Phones 9971/9951/8961, Unified IP Phones 6921/6941/6961s and Unified Personal Communicator.	SCCP Ph1->Unified CMBE->RT SIP Ph1; RT SIP Ph1->Don't answer; The call Should go to all the members in the hunt group until it is answered depending upon the algorithm configured.	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CCM.015	Unified IP Phones 9971/9951/8961	Depositing VM from Unified IP Phones 6921/6941/6961 using Unity Connection to Unified IP Phones 9971/9951/8961	Verify if a PSTN call can be made from the Unified IP Phones 6921/6941/6961 to a remote IP Phone 9971/9951/8961 over H.323/SIP gateways.	Stage 1: RT-Lite SCCP Ph1->Unified CMBE->PSTN GW->PSTN->PSTN GW->Unified CMBE->Rem RT SIP Ph1->CFNA->UNC; Stage 2: RT-Lite SCCP Ph1->Deposits a VM; Stage 3: Rem RT SIP Ph1->Retrieves a VM	Passed	
UC802EF.CCM.016	Unified IP Phones 9971/9951/8961	Hold Reversion with Extension Mobility Using Unified IP Phones 6921/6941/6961 and Unified IP Phones 9971/9951/8961	Verify the hold reversion feature with Extension Mobility by creating an Extension Mobility profile that is similar to one of the SIP Phones in the central site. Use this profile to login from a remote Unified IP Phones 6921/6941/6961 and IP Phones 9971/9951/8961.	Stage 1:PSTN Ph 1->PSTN GW->Unified CMBE->SIP Ph 1->Hold Stage 2:SIP Ph 1 ->Resume	Passed	
UC802EF.CCM.201	Extension Mobility Cross Cluster	Depositing a Voicemail from HC Phone to the VC Logged in Phone	Verify that Extension Mobility Cross Cluster works and is able to retrieve successfully the voice mails from the VC logged in phones.	IP Phone 99xx/89xx/69xx->EMCC->TNP Phone;	Passed	
UC802EF.CCM.202	Extension Mobility Cross Cluster	Extension Mobility Cross Cluster from Remote Phones to Another Cluster	Verify that Extension Mobility Cross Cluster works from IP Phones 8961/9951/9971 and gets the profile of the other cluster remote TNP Phones.	IP Phone 99xx/89xx/69xx->Unified CM1->SIP ICT->Unified CM2->TNP Phone	Passed	
UC802EF.CCM.203	Extension Mobility Cross Cluster	Extension Mobility Cross Cluster from the IP Phones 8961/9951/9971 in V4 cluster to TNP Phone in DS cluster	Verify that Extension Mobility Cross Cluster works from the IP Phones 8961/9951/9971 in V4 cluster and gets the profile of the dual stack cluster TNP Phones.	IP Phone 99xx/89xx/69xx->Unified CM1(V4)->SIP ICT->Unified CM2(DS)->TNP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CIP.001	Unified IP Phones 9951/9971	Unified IP Phones 9951/9971 with Video Support Interworking with Unified PC and IP Phones 7985	Verify the call from Unified IP Phones 9951/9971 with Secace USB Video Camera attachment to Unified Personal Communicator (video enabled) over QSIG ICT and then transfer the call to UC Integration for Microsoft Office Communicator/IP Phone 7985 attached with Unified Video Advantage in the same cluster. Verify the video-on-hold by pressing the hold button on Unified IP 99xx/89xx Phones.	Unified IP 99xx/89xx SIP Ph1 with USB Video camera->Unified CM1->QSIG ICT->Unified CM2->CUPC(video enabled)->XFER_C->Unified CM2->UC Integration™ for Microsoft Office Communicator/7985	Passed	
UC802EF.CIP.002	Unified IP Phones 9951/9971	Unified IP Phones 9951/9971 Interworking with IP Communicator and Unified Video Advantage	Verify the call from the SIP IP Communicator with Unified Video Advantage attached to the remote SCCP Phone which has CFNA set to a remote Unified IP Phones 9951/9971.	SIP CIPC1(with CUVA) ->Unified CM1->QSIG ICT->Unified CM2 -> Rem SCCP Ph1; Rem SCCP Ph1->CFNA->Rem RT Std+ SIP Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CIP.003	Unified IP Phones 9951/9971	Unified IP Phones 9951/9971 with Video Support Interworking with UC Integration™ for Microsoft Office Communicator	Verify the call from the Unified IP Phones 9951/9971 with Secace USB Video Camera attachment to UC Integration for Microsoft Office Communicator which is in deskphone mode to IP Phones 9951/9971 with Secace USB Video Camera attachment over SIP ICT and then conference the call to a IP Communicator with Unified Video Advantage attached. Then drop the call from UC Integration for Microsoft Office Communicator.	Unified IP 99xx/89xx SIP Ph1 with USB Video camera->Unified CM1->SIP ICT->Unified CM2->UC Integration™ for Microsoft Office Communicator (in deskphone mode to Unified IP 99xx/89xx SIP Ph1 with USB Video camera)->Conf->SCCP CIPC1 (with CUVA)	Passed	
UC802EF.CIP.004	Unified IP Phones 9951/9971	Unified IP Phones 9951/9971 with Video Support over Tandem Cluster	Verify the call from the UC Integration for Microsoft Office Communicator client and IP Phones 9951/9971 in leaf clusters over tandem and then transfer the call to the Video enabled IP Phones 9951/9971 over QSIG ICT.	UC Integration™ for Microsoft Office Communicator(in deskphone mode to Video phone) ->Unified CM1->QSIG->Unified CM-Tand->QSIG->Unified CM2->Unified IP 99xx/89xx SIP Ph1 with USB Video camera->XFER_C->Unified CM1->QSIG ICT->Unified CM2->IP Video Phone 99xx enabled	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CIP.005	Unified IP Phones 9951/9971	Unified IP Phones 9951/9971 with Video support Interworking with TNP Phones with Unified Video Advantage Attached	Verify the call from the TNP Phone attached with the Unified Video Advantage to the Unified Personal Communicator client(Video enabled) over SIP ICT and then conference the call to the IP Phones 9951/9971 with Secace USB Video Camera attachment over QSIG ICT.	TNP SCCP Ph1(CUVA) ->Unified CM1->SIP ICT->Unified CM2->CUPC(video enabled)->CFB->Unified CM3->QSIG ICT->Unified IP 99xx/89xx SIP Ph1 with USB Video camera	Passed	
UC802EF.CIP.006	Unified IP Phones 9951/9971	Unified IP Phones 9971/9951/8961 in Unified SRST	Verify the PSTN call from the remote Unified IP Phones 9971/9951/8961 in Unified SRST mode to another remote Unified IP Phones 9971/9951/8961 in Unified SRST mode which has CFWDALL set to a PSTN Phone.	Rem 99xx/89xx SIP Ph1->SRST1->PSTN GW->PSTN->PSTN GW->SRST 2->Rem 99xx/89xx SIP Ph2;Rem 99xx/89xx SIP Ph2->CFWDALL->PSTN GW->PSTN Ph1	Passed	
UC802EF.CIP.007	Unified IP Phones 9951/9971	Plus Dialing in Unified IP Phones 9971/9951/8961 in Unified SRST	To verify the following: Registered Unified IP Phones 9971/9951/8961 with + sign along with the Directory number before bringing the WAN interface down. Make a PSTN call from the remote IP Phones 9971/9951/8961 in SRST mode to another IP Phones 9971/9951/8961 with + sign in another remote which then transfers the call to PSTN Phone through the PSTN gateway.	Rem 99xx/89xx SIP Ph1->SRST1/PSTN GW->PSTN->SRST2 /PSTN GW->PSTN->Rem 99xx/89xx SIP Ph2->XFER_C->PSTN GW->PSTN Ph1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CIP.008	Unified IP Phones 9951/9971	Reply to Voice mail with Call or IM or Voicemail	Verify if Unified Personal Communicator can send a voice mail to IP Phones 9971/9951/8961 and the phone replies through calling or through voicemail.	Stage 1: CUPC1->Unified CM->99xx/89xx SIP Ph 1->CFNA->Voice Mail; Stage 2:99xx/89xx SIP Ph 1->Unified CM-> CUPC ;	Passed	
UC802EF.CIP.009	Unified IP Phones 9951/9971	Inline Playback of Visual Voice Mail	Unified Personal Communicator deposits a voicemail to SIP IP Phones 9971/9951/8961. Toast pop up of Visual voice mail indication is received and Playback of Visual voice mail is done with fast forward and rewind options.	CUPC->Unified CM-> Unified IP 99xx/89xx/STD + SIP Ph1 ->Voice mail	Passed	
UC802EF.CRS.001	Unified CCX Inbound Call	Unified Contact Center Express with High-Availability	Verify Unified CCX with high-availability: Inbound call from a PSTN phone to a Cisco Agent Desktop agent with SCCP phone in the central site when one of the Unified CCX server is down.	PSTN Ph1->MGCP GW->Unified CM->UCCX->SCCP Ph1 (CAD)	Passed	
UC802EF.CRS.002	Unified CCX Outbound Call	Outbound Call from Cisco Agent Desktop SCCP Remote Phone over H.323 FXO	Verify if outbound call from a Cisco Agent Desktop SCCP phone in a remote site to a PSTN phone over H.323 FXO remote gateway is successful.	UCCX->Unified CM->SCCP Ph1 (CAD)->H.323 FXO->PSTN Ph1	Passed	
UC802EF.CRS.003	Unified CCX	Outbound Call from Cisco Agent Desktop SCCP Remote Phone over MGCP FXO	Verify if outbound call from a Cisco Agent Desktop SCCP phone in a remote site to a PSTN phone over MGCP FXO remote gateway is successful.	UCCX->Unified CM->SCCP Ph1 (CAD)->MGCP FXO->PSTN Ph1	Passed	
UC802EF.CRS.004	Unified CCX Outbound call	Outbound Call from Cisco Agent Desktop IP Phone 6941	Verify if an outbound call from a Cisco Agent Desktop IP Phone 6941 to a PSTN phone is successful.	UCCX->Unified CM->6941 Phone(CAD)->MGCP PRI->Rem PSTN Ph	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CRS.005	Unified CCX Outbound Call	Outbound Call from a Cisco Agent Desktop IP Phone 9971 to a PSTN phone	Verify if an outbound call from a Cisco Agent Desktop IP Phone 9971 to a PSTN phone is successful.	UCCX->Unified CM->9971 Phone(CAD)->MGCP BRI->Rem PSTN Ph	Passed	
UC802EF.CRS.006	Unified CCX Outbound Call	Outbound Call from a Cisco Agent Desktop SCCP Phone	Verify if an outbound call from a Cisco Agent Desktop SCCP phone in a remote site to a PSTN phone over MGCP BRI remote gateway is successful.	UCCX->Unified CM->SCCP Ph1 (CAD)->MGCP BRI->PSTN Ph1	Passed	
UC802EF.CRS.007	Unified CCX Outbound Call	Outbound Call from a Cisco Agent Desktop SIP phone Gateway	Verify if an outbound call from a Cisco Agent Desktop SIP phone in the central site to a remote SCCP phone is consult transferred to a PSTN phone over MGCP PRI remote gateway.	UCCX->Unified CM->SIP Ph1 (CAD)->Unified CM->SCCP Ph1->XFR_C->MGCP PRI->PSTN Ph1	Passed	
UC802EF.CRS.008	Unified CCX Outbound call	Outbound Call from a Cisco Agent Desktop SCCP Phone	Verify if an outbound call from a Cisco Agent Desktop SCCP phone in a remote site to a PSTN phone over SIP BRI remote gateway is successful.	UCCX->Unified CM->SCCP Ph1 (CAD)->SIP BRI->PSTN Ph1	Passed	
UC802EF.CRS.009	Unified CCX Outbound Call	Outbound Call from a Cisco Agent Desktop SCCP Phone	Verify if an outbound call from a Cisco Agent Desktop SCCP phone in a remote site to a PSTN phone over SIP PRI remote gateway is successful.	UCCX->Unified CM->SCCP Ph1 (CAD)->SIP PRI->PSTN Ph1	Passed	
UC802EF.CRS.010	Unified CCX Outbound call	Unified CCX with Unity connection as Front-End	Verify if Unified CCX with Unity connection can function as front-end.	PSTN Ph1->MGCP PRI->Unified CM->Unity connection->XFR->UCCX->9971 Phone (CAD)	Passed	
UC802EF.CRS.011	Unified CCX and Cisco Unity Interoperability	Unified CCX with Cisco Unity as Back-End	Verify if Unified CCX with Cisco Unity can function as back-end.	PSTN Ph1->MGCP PRI->Unified CM->UCCX->SCCP Ph1->CFNA->Unity	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CRS.012	Unified CCX and Cisco Unity Interoperability	QSIG PBX Phone Call to Unified CCX Agent via ICT	Verify if the call from a QSIG PBX phone to Unified CCX via ICT is transferred to a Cisco Agent Desktop with a SCCP phone in the central site.	PBX Ph1->QSIG Trunk->Unified CM->ICT (QSIG)->Unified CM->UCCX->Unified CM->6941 Phone (CAD)	Passed	
UC802EF.CRS.013	Unified CCX Call Transfer	Negative Testing of Unified CCX Redundancy	Verify the negative testing of Unified CCX redundancy.	SCCP Ph1->Unified CM->UCCX->6941 Ph (CAD)	Passed	
UC802EF.CRS.014	Unified CCX Redundancy	Blind Transfers Between Agents Cause Agents to be put in Reserved State	Verify if blind transfers between agents is causing agents to be put in Reserved State.	PSTN Ph1->MGCP PRI->Unified CM->UCCX->SCCP Ph1 (CAD)->XFR_B->SCCP Ph2 (CAD)	Passed	
UC802EF.CRS.015	Unified CCX	Unified CCX and Unity Connection Interaction	Verify the transfer scenario of Unified CCX and Unity Connection Interaction.	SCCP Ph1->Unified CM->Unity connection->UCCX->9971 (CAD)->XFR_C->SCCP Ph2->XFR_C->UCCX->SCCP Ph3 (CAD)	Passed	
UC802EF.CRS.016	Unified CCX	Unified CCX and Cisco Unity Auto Attendant Call Handler Interaction	Verify the call transfer from Cisco Unity Auto Attendant call handler to SCCP Cisco Agent Desktop Phone.	SCCP Ph1->Unified CM->Unity AA->UCCX->SCCP Ph1 (CAD)	Passed	
UC802EF.CRS.017	Unified CCX	Agent Gets Stuck in Reserved When Call is Blind-Transferred from Cisco Unity to Unified CCX	Verify if the agent gets stuck in Reserved when a call is blind-transferred from Cisco Unity to Unified CCX.	SCCP Ph1->Unified CM->Unity->XFR_B->UCCX->SCCP Ph2 (CAD)	Passed	
UC802EF.CRS.018	Unified CCX	Agent Cannot Hang Up Call After Transfer to an Extension	Verify if an agent is not able to hang up a call after transferring to an extension.	PSTN Ph1->MGCP PRI->Unified CM->UCCX->SCCP Ph1 (CAD)->XFR_C->SCCP Ph2	Passed	
UC802EF.CRS.019	Unified CCX	Call to Cisco Unity Call Handler Redirected to Unified CCX Route Point	Verify if a call to Cisco Unity call handler gets redirected to Unified CCX route point.	SCCP Ph1->Unified CM->Unity->UCCX	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CRS.020	Unified CCX	Non-ICD Transferred Call to Agent Disconnected from Source	Verify if a non-ICD transferred call to an agent gets disconnected from source side.	SCCP Ph1->Unified CM->ICT (QSIG)->Unified CM->SCCP Ph1 (CAD)	Passed	
UC802EF.CRS.021	Unified CCX	H323 Video Phone Call to Unified CCX Agent	Verify if a call from a H323 video phone to Unified CCX is transferred to SCCP Cisco Agent Desktop agent in a remote site. The remote Cisco Agent Desktop agent consult transfers the call to a CSD supervisor in the central site.		Passed	
UC802EF.CRS.022	Extension Mobility	Extension Mobility of IP Phone A in IP Phone Agent B	Verify if IP Phone Agent A can perform an Extension Mobility in IP agent B to accept the queued call.	Stage 1:9971 Ph (CAD)->EM->SCCP Ph1(CAD);Stage 2:SIP Ph->Unified CM->UCCX->9971 Ph(CAD)	Passed	
UC802EF.CRS.023	Unified CCX	CSD Supervisor's Silent Monitoring and Intercepting of Calls from a PSTN Phone	Verify if a call from a PSTN phone to Unified CCX via an MGCP gateway over BRI is transferred to a Cisco Agent Desktop with an SCCP phone in central site. A CSD supervisor in the central site silently monitors and intercepts the call.	Stage 1: PSTN Ph1->MGCP BRI->Unified CM->UCCX->Unified CM->SCCP Ph1(CAD) Stage 2: SCCP Ph2 (CSD)->Unified CM->UCCX->Unified CM->SCCP Ph1 (CAD)	Passed	
UC802EF.CSF.001	CSF, UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator with Unified IP Phones 9971/9951/8961 (deskphone) makes a video call to IP Phone 7985 in another cluster over SIP ICT	Verify if a UC Integration for Microsoft Office Communicator in deskphone mode can make a video call to a IP Phone 7985 over SIP ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->SIP ICT->Unified CM->7985	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CSF.002	CSF, UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator with IP Phones 9971/9951/8961 (deskphone) makes Video Call to IP Phone 7985 in Another Cluster over QSIG ICT	Verify if a UC Integration for Microsoft Office Communicator in deskphone mode can make a video call to a IP Phone 7985 over QSIG ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->QSIG ICT->Unified CM->7985	Passed	
UC802EF.CSF.003	CSF, UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator with IP Phones 9971/9951/8961 (deskphone) makes a Video Call to IP Phones 9971/9951/8961 in Another Cluster over SIP ICT	Verify if a UC Integration for Microsoft Office Communicator in deskphone mode can make a video call to a IP Phones 9971/9951/8961 over SIP ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->SIP ICT->Unified CM->IP Phone 99xx/89xx	Passed	
UC802EF.CSF.004	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator with IP Phones 9971/9951 (deskphone) makes a Video call to IP Phones 9971/9951 in another cluster over QSIG ICT	Verify if a UC Integration for Microsoft Office Communicator in deskphone mode can make a video call to a IP Phones 9971/9951 over QSIG ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->QSIG ICT->Unified CM->IP Phone 99xx/89xx	Passed	
UC802EF.CSF.005	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator Video Calls Between Two Clusters over QSIG ICT	Verify UC Integration for Microsoft Office Communicator (Softphone) video calls between two clusters over QSIG ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->QSIG ICT->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CSF.006	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator Video Calls Between Two Clusters over SIP ICT	Verify UC Integration for Microsoft Office Communicator (Softphone) video calls between two clusters over SIP ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->SIP ICT->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	
UC802EF.CSF.009	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator (softphone) on Remote Site in SRST Mode with Audio	Verify the behavior of UC Integration for Microsoft Office Communicator (softphone) on a remote site in SRST Mode with audio.	IP Phone 99xx/89xx->Unified CM->Remote Branch->UC Integration™ for Microsoft Office Communicator (SRST)	Passed	
UC802EF.CSF.011	CSF, UC Integration™ for Microsoft Office CommunicatorTM	Visual Voicemail with UC Integration™ for Microsoft Office Communicator in Deskphone Mode	Verify Visual voicemail with UC Integration for Microsoft Office Communicator in deskphone mode.	UC Integration™ for Microsoft Office Communicator->Unified CM->Unity Connection	Passed	
UC802EF.CSF.012	CSF, UC Integration™ for Microsoft Office CommunicatorTM	Visual Voicemail with UC Integration™ for Microsoft Office Communicator in Softphone Mode	Verify Visual voicemail with UC Integration for Microsoft Office Communicator in softphone mode.	UC Integration™ for Microsoft Office Communicator->Unified CM->Unity Connection	Passed	
UC802EF.CSF.013	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator Video Call from Remote Phone to Central Site UC Integration™ for Microsoft Office Communicator	Verify UC Integration for Microsoft Office Communicator Video call from remote phone to central site UC Integration for Microsoft Office Communicator.	UC Integration™ for Microsoft Office Communicator->Remote Branch->Unified CM->Central UC Integration™ for Microsoft Office Communicator	Passed	
UC802EF.CSF.014	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator Video call from Remote Phone to Central Site IP Phone 7985	Verify UC Integration for Microsoft Office Communicator Video call from remote phone to central site IP Phone 7985.	UC Integration™ for Microsoft Office Communicator->Remote Branch->Unified CM->Central 7985	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CSF.015	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator Video Call from Remote Phone to Central Site H.323 Endpoint	Verify UC Integration for Microsoft Office Communicator video call from remote phone to central site H.323 endpoint.	UC Integration™ for Microsoft Office Communicator->Remote Branch->Unified CM->H.323 Video endpoint	Passed	
UC802EF.CSF.016	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator (deskphone) Video Call to H.323 Video Endpoint in Same Cluster	Verify UC Integration for Microsoft Office Communicator (deskphone) video call to H.323 video endpoint in same cluster.	UC Integration™ for Microsoft Office Communicator->Unified CM->H.323 Video endpoint	Passed	
UC802EF.CSF.017	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator (deskphone) Video Call to H.323 Video Endpoint in Different Cluster Over QSIG ICT	Verify UC Integration for Microsoft Office Communicator (deskphone) video call to H.323 video endpoint in different cluster over QSIG ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->QSIG ICT->Unified CM->H.323 Video endpoint	Passed	
UC802EF.CSF.018	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator (deskphone) Video Call to H.323 Video Endpoint in Different Cluster over SIP ICT	Verify UC Integration for Microsoft Office Communicator (deskphone) video call to H.323 video endpoint in different cluster over SIP ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->SIP ICT->Unified CM->H.323 Video endpoint	Passed	
UC802EF.CSF.019	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator (softphone) Video Call to H.323 Video Endpoint in Same Cluster	Verify UC Integration for Microsoft Office Communicator (softphone) video call to H.323 video endpoint in same cluster.	UC Integration™ for Microsoft Office Communicator (Softphone)->Unified CM->H.323 Video endpoint	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CSF.020	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator (softphone) Video Call to H.323 MCU in Different Cluster over QSIG ICT	Verify UC Integration for Microsoft Office Communicator (softphone) video call to a H.323 MCU in a different cluster over QSIG ICT.	UC Integration™ for Microsoft Office Communicator (Softphone)->Unified CM->QSIG ICT->Unified CM->H.323-MCU	Passed	
UC802EF.CSF.021	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator (softphone) Video Call to H.323 Video Endpoint in Different Cluster over SIP ICT	Verify UC Integration for Microsoft Office Communicator (softphone) video call to H.323 video endpoint in different cluster over SIP ICT.	UC Integration™ for Microsoft Office Communicator (Softphone)->Unified CM->SIP ICT->Unified CM->H.323 Video endpoint	Passed	
UC802EF.CSF.022	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator Video Conference Call Across Different Clusters over QSIG ICT	Verify UC Integration for Microsoft Office Communicator Video conference call across different clusters over QSIG ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->QSIG ICT->Unified CM->UC Integration™ for Microsoft Office Communicator->CONF->UC Integration™ for Microsoft Office Communicator	Passed	
UC802EF.CSF.023	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator Video Conference Call Across Different Clusters over SIP ICT	Verify UC Integration for Microsoft Office Communicator Video Conference Call Across Different Clusters over SIP ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->SIP ICT->Unified CM->UC Integration™ for Microsoft Office Communicator->CONF->UC Integration™ for Microsoft Office Communicator	Passed	
UC802EF.CSF.024	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator Video Call to Unified Video Advantage and IP Communicator in Different Cluster over QSIG ICT	Verify UC Integration for Microsoft Office Communicator video call to Unified Video Advantage and IP Communicator in different cluster over QSIG ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->QSIG ICT->Unified CM->CIPC + CUVA	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CSF.025	CSF, UC Integration™ for Microsoft Office CommunicatorTM	UC Integration™ for Microsoft Office Communicator Video Call to Unified Video Advantage and IP Communicator in Different Cluster over SIP ICT	Verify UC Integration for Microsoft Office Communicator video call to Unified Video Advantage and IP Communicator in different cluster over SIP ICT.	UC Integration™ for Microsoft Office Communicator->Unified CM->SIP ICT->Unified CM->CIPC + CUVA	Passed	
UC802EF.CUE.001	Cisco Unity Express	Depositing a Voicemail from DPNSS PBX Phone	Verify the ability of depositing a VM from a DPNSS PBX Phone over QSIG ICT to the remote phone which has been integrated with NME-Cisco Unity Express.	DPNSS PBX Ph1->Unified CM1->QSIG ICT->Unified CM2->Rem SCCP Ph1->CFNA->NME-CUE	Passed	
UC802EF.CUE.002	Cisco Unity Express	Configure Multiple Greetings Through CLI in Cisco Unity Express	Verify the multiple greetings like busy, Closed, Internal, Vacation, Meeting and Extended absence greetings through CLI in Cisco Unity Express can be configured and then make a call from QSIG PBX Phone to Unified CME SIP Phone which has CFNA to Cisco Unity Express voicemail.	PBX Ph1->QSIG PBX->Unified CM->SIP Trunk->Unified SIP Proxy->CME->SIP Ph1-> CFNA->CUE	Failed	CSCtf47467
UC802EF.CUE.003	Cisco Unity Express	Depositing Voicemail to the Unified CME Phones Over Unified SIP Proxy from a Unified CM IP Phones 8961/9951/9971	Verify that the Unified IP Phones 8961/9951/9971 are able to successfully deposit a voice mail to the Unified CME phone over Unified SIP Proxy.	RT Pro/Biz/Std + Ph1 ->Unified CM->SIP Trunk->Unified SIP Proxy->CME->Ph2 ->CFNA->CUE	Failed	CSCtf47467

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CUE.004	Cisco Unity Express	Depositing Voicemail to Unified CME SIP Phones over Unified SIP Proxy from a PBX Phone	Verify that PBX phones are able to successfully deposit a voice mail to the Unified CME SIP Phones over Unified SIP Proxy.	PBX Ph1->PBX->VG30D->MGCP GW->Unified CM->SIP Trunk->Unified SIP Proxy->CME->SIP Ph1 (First Line)->CFNA->SIP Ph1 (Second Line)->CFNA->CUE	Failed	CSCtf47467
UC802EF.CUE.005	Cisco Unity Express	SRSV Functionality	Verify that when the subscribers in remote sites are able to deposit a voicemail using the local Cisco Unity Express module attached to the remote router in Unified SRST mode when the Unified CM goes down. When the Unified CM is up, the remote subscribers should be able to deposit voicemail using the central site's Cisco Unity and in Unified SRST mode the remote subscribers should be able to use the local Cisco Unity Express module in SRST router.	Rem Unified IP Phones 6921/6941/6961/Pro/Biz/Std + Ph 1->SRST1/PSTN gateway->PSTN->SRST1/PSTN gateway->Rem Unified IP Phones 6921/6941/6961/Pro/Biz/Std + Ph 2->CFNA->CUE	Failed	CSCtf47467
UC802EF.CUP.002	IM, Presence	Call from Unified Personal Communicator in Desk Phone to UC Integration™ for Microsoft Office Communicator		CUPC->Unified Presence->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	
UC802EF.CUP.004	Unified Presence	IM and Presence in Unified Presence 8.0	Verify if Office Communicator Server goes down.	CUPC->Unified Presence->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CUP.005	RSVP, Unified Presence 8.0	Video phones, Unified IP Phones 9971/9951/8961, Unified Personal Communicator Make Video Calls with RSVP		RT/Video Ph->REM->WAN->Unified Presence->Unified CM->SCCP Ph/SIP Ph/ CUPC	Passed	
UC802EF.CUP.006	IM, Unified Presence	Unified Personal Communicator, IM session, Unified Presence	Verify the inter working of Unified Personal Communicator with Extension mobility.	CUPC->Unified Presence->Unified CM->WAN->Rem->EM Ph1	Passed	
UC802EF.CUP.008	IM session, Presence in Unified Attendant Server	Interworking with Unified Attendant Server		Stage1: PSTN->FXO->REM->Unified CM->ARC;Stage2: ARC->Unified CM->Unified Presence->CUPC/UC Integration™ for Microsoft Office Communicator	Passed	
UC802EF.CUP.009	IM session, Presence in Unified Attendant Server	Interworking with Unified Attendant Server		Stage1: PSTN->FXO->REM->Unified CM->ARC;Stage2: ARC->Unified CM->Unified Presence->CUPC/UC Integration™ for Microsoft Office Communicator	Passed	
UC802EF.CUP.010	IM Session and RSVP Interworking	IM session on IP Phones, Video phones, Unified IP Phones 9971/9951/8961, and Unified Personal Communicator		RT/Video Ph->REM->WAN->Unified Presence->Unified CM->SCCP Ph/SIP Ph/ CUPC	Passed	
UC802EF.CUP.401	IM session, Unified PC 8.0, Unified Presence	Unified Personal Communicator 8.0 Interworking with Extension Mobility			Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CUP.402	IM session, Presence in Unified Attendant Server	Interworking with Unified Attendant Server			Passed	
UC802EF.CUP.403	RSVP Interworking, Video calls	Video phones, Unified IP Phones 9971/9951/8961, Unified Personal Communicator 8.0 Make Video Calls with RSVP			Passed	
UC802EF.CUP.403	Video Calls and RSVP	Video phones, Unified IP Phones 9971/9951/8961, Unified Personal Communicator 8.0 Make Video Calls with RSVP			Passed	
UC802EF.CUP.405	Unified Personal Communicator 8.0	Unified Personal Communicator 8.0 and Unity Connection			Passed	
UC802EF.CUP.405	Unified Personal Communicator 8.0 and Voicemail	Unified Personal Communicator 8.0 and Unity Connection			Passed	
UC802EF.CUP.406	Unified Personal Communicator 8.0 and CGPN	Unified Personal Communicator 8.0 and Calling Party Normalization			Passed	
UC802EF.CUP.406	Unified Personal Communicator 8.0 and CGPN	Unified Personal Communicator 8.0 and Calling Party Normalization			Passed	
UC802EF.CUP.407	Video Interoperability	Third Party Video Interoperability			Passed w/ Exception	CSCtg21657

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.CUP.409	IM session, Presence, Unified Personal Communicator 8.0	Unified Personal Communicator 8.0 IM with Moment-IM, MOC and Unified Personal Communicator 7.x			Passed	
UC802EF.CUP.804	Video Conference Calls	Video phones, Unified IP Phones 9971/9951/8961, Unified Personal Communicator 8.0 are in Video Conference			Passed	
UC802EF.CUP.804	Video Conference Calls	Video Phones, Unified IP Phones 9971/9951/8961, Unified Personal Communicator 8.0 are in Video Conference			Passed	
UC802EF.IME.001	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Audio call from Cluster with Inline ASA to Another Cluster with Inline ASA	Verify a Cisco IME audio call from cluster with inline ASA to another cluster with inline ASA.	Originating Phone->Cisco IME->CCM->Inline ASA->SIP Trunk->Inline ASA->CCM->Cisco IME->Terminating Phone	Passed	
UC802EF.IME.002	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Audio call from Cluster with Inline ASA to Another Cluster with Off-Path ASA	Verify a Cisco IME audio call from cluster with inline ASA to another cluster with off-path ASA.	Originating Phone->Cisco IME->CCM->Off-path ASA->SIP Trunk->Off-path ASA->CCM->Cisco IME->Terminating Phone	Passed	
UC802EF.IME.003	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Audio Call from Cluster with Off-Path ASA to Another Cluster with Inline ASA	Verify a Cisco IME audio call from cluster with off-path ASA to another cluster with inline ASA.	Originating Phone->Cisco IME->CCM->Inline ASA->SIP Trunk->Off-path ASA->CCM->Cisco IME->Terminating Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.IME.004	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Video Call from Cluster with Inline ASA to Another Cluster with Inline ASA	Verify a Cisco IME video call from cluster with inline ASA to another cluster with inline ASA.	Originating Video Phone->Cisco IME->CCM->Inline ASA->SIP Trunk->Inline ASA->CCM->Cisco IME->Terminating Video Phone	Passed	
UC802EF.IME.005	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Video Call from Cluster with Inline ASA to Another Cluster with Off-Path ASA	Verify a Cisco IME video call from cluster with inline ASA to another cluster with off-path ASA.	Originating Video Phone->Cisco IME->CCM->Inline ASA->SIP Trunk->Off-path ASA->CCM->Cisco IME->Terminating Video Phone	Passed	
UC802EF.IME.006	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Video Call from Cluster with Off-Path ASA to Another Cluster with Inline ASA	Verify a Cisco IME video call from cluster with off-path ASA to another cluster with inline ASA.	Originating Video Phone->Cisco IME->CCM->Off-path ASA->SIP Trunk->Inline ASA->CCM->Cisco IME->Terminating Video Phone	Passed	
UC802EF.IME.007	ASA, Unified B2B Link/ Cisco Internet Media Engine	PSTN Fallback on Cluster with Inline ASA	Verify PSTN fallback on cluster with inline ASA when call is degraded.	Originating Phone->Cisco IME->CCM->Inline ASA->SIP Trunk->Inline ASA->CCM->Cisco IME->Terminating Phone	Passed	
UC802EF.IME.008	ASA, Unified B2B Link/ Cisco Internet Media Engine	PSTN Fallback on Cluster with Off-Path ASA	Verify PSTN fallback on cluster with off-path ASA when call is degraded.	Originating Phone->Cisco IME->CCM->Off-path ASA->SIP Trunk->Off-path ASA->CCM->Cisco IME->Terminating Phone	Passed	
UC802EF.IME.009	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call with Audio Escalation to Video	Verify Cisco IME call with audio escalation to video.	Originating Phone->Cisco IME->CCM->SIP Trunk->CCM->Cisco IME->Terminating Phone	Passed	
UC802EF.IME.010	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call with Hold/Resume	Verify Cisco IME call interaction with Hold/Resume.	Originating Phone->Cisco IME->CCM->SIP Trunk->CCM->Cisco IME->Terminating Phone (Hold/Resume)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.IME.011	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call Transfer	Verify Cisco IME call interaction with call transfers.	Originating Phone->Cisco IME->CCM->SIP Trunk->CCM->Cisco IME->Terminating Phone->Transfer->Destination Phone	Passed	
UC802EF.IME.012	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call Conference	Verify Cisco IME call interaction with conference calls.	Originating Phone->Cisco IME->CCM->SIP Trunk->CCM->Cisco IME->Terminating Phone->Conf->Destination Phone	Passed	
UC802EF.IME.013	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call with Shared Line	Verify Cisco IME call interaction with shared line phones.	Originating Phone->Cisco IME->CCM->SIP Trunk->CCM->Cisco IME->Shared Line Terminating Phone	Passed	
UC802EF.IME.014	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call can be Transferred to Another Cluster Over Inter-Cluster Trunks	Verify if a Cisco IME call can be transferred to another cluster over inter-cluster trunks.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->ICT->Unified CM->Phone	Passed	
UC802EF.IME.015	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call Gets Transferred to Originating Cluster over Inter-cluster Trunk	Verify if a Cisco IME call can be transferred to the originating cluster over inter-cluster trunks.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->ICT->Unified CM->Phone	Passed	
UC802EF.IME.016	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call gets Transferred to Originating Cluster as Cisco IME Call	Verify if a Cisco IME call can be transferred to the originating cluster as a Cisco IME call.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->SIP Trunk->Unified CM->Phone	Passed	
UC802EF.IME.017	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call gets Transferred to Unified CME site over H.323 Network	Verify if a Cisco IME call can be transferred to a Unified CME site over H.323 network.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->H.323 GK->H.323 CUBE->H.323 GK->CME->Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.IME.018	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call gets Transferred to Unified CME Site Aggregated by Unified SIP Proxy	Verify if a Cisco IME call can be transferred to a Unified CME site aggregated by Unified SIP Proxy.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->SIP Trunk->Unified SIP Proxy->CME->Phone	Passed	
UC802EF.IME.019	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call get Transferred to Unified CM-Interop Site	Verify if a Cisco IME call can be transferred to a phone in interop-site with Unified CM 7.x version.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->QSIG ICT->Unified CM 7.x->Phone	Passed	
UC802EF.IME.020	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call Transferred to PSTN Phone	Verify if a Cisco IME call can be transferred to a PSTN phone.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->Unified CM->PSTN GW->PSTN Phone	Passed	
UC802EF.IME.021	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call Transferred to PBX Phone	Verify if a Cisco IME call can be transferred to a PBX phone.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->Unified CM->QSIG PBX->PBX Phone	Passed	
UC802EF.IME.022	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call Transferred to a Dual-Stack Phone in a Dual-Stack IPv6 Aware Cluster	Verify if a Cisco IME call can be transferred to a dual-stack phone in a IPv6 aware cluster.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->QSIG ICT->DS Unified CM->DS Phone	Passed	
UC802EF.IME.023	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Call Transferred to a Remote Branch Phone	Verify if a Cisco IME call can be transferred to a phone in remote branch.	Originating Phone->Cisco IME->Unified CM->SIP Trunk->Unified CM->Cisco IME->Terminating Phone->Xfer->Unified CM->Remote Branch->Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.IME.024	ASA, Unified B2B Link/ Cisco Internet Media Engine	Cisco IME Negative Scenarios	Verify negative test scenarios with Cisco IME.		Passed	
UC802EF.QOS.001	E2E RSVP, SIP Preconditions	E2E RSVP Call Between Phones in two Unified CM Clusters	Verify an E2E RSVP call between phones in two Unified CM clusters.	Originating Phone->RSVP Agent->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2->RSVP Agent->Terminating Phone	Passed	
UC802EF.QOS.002	E2E RSVP, SIP Preconditions	E2E RSVP Call Between a Phone in One Cluster to a Phone in Remote Branch of Another Cluster	Verify an E2E RSVP call between a phone in one cluster to a phone in a remote branch of another cluster.	Originating Phone->Remote RSVPAgent->Remote Branch->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2->RSVPAgent->Terminating Phone	Passed	
UC802EF.QOS.003	E2E RSVP, SIP Preconditions	E2E RSVP Call from a Remote Phone in one Cluster to a Remote Phone in Another Cluster	Verify an E2E RSVP call from a remote phone in one cluster to a remote phone in another cluster.	Originating Phone->Remote RSVPAgent->Remote Branch->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2->Remote RSVPAgent->Terminating Phone	Passed	
UC802EF.QOS.004	E2E RSVP, SIP Preconditions	E2E RSVP Call Between a Phone in a Remote Branch of one Cluster to a Phone in Another Cluster Where Location is Set as Hub-None	Verify an E2E RSVP call between a phone in a remote branch of one cluster to a phone in another cluster where location is set as hub-none.	Originating Phone->Remote RSVPAgent->Remote Branch->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2 (hub_none)->RSVPAgent->Terminating Phone	Passed	
UC802EF.QOS.005	E2E RSVP, SIP Preconditions	E2E RSVP Call from a Phone in Cluster A to a Phone in Cluster B and Location is Set as Hub-None for Both Endpoints	Verify an E2E RSVP call from a phone in cluster A to a phone in cluster B and location is set as hub-none for both the end-points.	Originating Phone->Remote RSVPAgent->Remote Branch->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2 (hub_none)->RSVPAgent->Terminating Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.006	E2E RSVP, SIP Preconditions	E2E RSVP Call from an IP Phone in one Cluster to a Remote FXS Phone in Another Cluster Registered to a Remote Branch (SIP GW) which has Pre-conditions Enabled	Verify an E2E RSVP call from an IP phone in one cluster to a remote FXS phone in another cluster that is registered to a Remote Branch (SIP GW) which has pre-conditions enabled.	IP Phone->RSVPAgent->Unified CM 1->SIP ICT->Unified CM 2->SIP Trunk->Remote SIP GW->FXS Phone	Passed	
UC802EF.QOS.007	E2E RSVP, SIP Preconditions	E2E RSVP Call from a Remote IP Phone in one Cluster to a Remote FXS Phone in Different Cluster Registered to a SIP Gateway with Pre-conditions Enabled	Verify an E2E RSVP call from a remote IP phone one cluster to a remote FXS phone in a different cluster that is registered to a SIP Gateway with pre-conditions enabled.	IP Phone->RSVPAgent->Unified CM 1->SIP ICT->Unified CM 2->SIP Trunk->Remote SIP GW ->FXS Phone	Passed	
UC802EF.QOS.008	E2E RSVP, SIP Preconditions	Unified Enterprise Attendant Console RSVP Interoperability	Verify an E2E RSVP call transferred by Unified Enterprise Attendant Console to a remote FXS phone.	IP Phone->CUEAC->IP Phone->RSVP Agent->CCM->SIP Trunk->Remote SIP GW->FXS Phone	Passed	
UC802EF.QOS.009	E2E RSVP, SIP Preconditions	E2E RSVP Call Between IP Phone to FXS Phone on Same Remote Site	Verify an E2E RSVP call between IP phone to FXS phone on the same remote site.	Remote IP Phone->CCM->SIP Trunk->Remote FXS Phone	Passed	
UC802EF.QOS.010	E2E RSVP, SIP Preconditions	E2E RSVP Audio Call Between UC Integration™ for Microsoft Office Communicator Endpoints in two Clusters	Verify E2E RSVP audio call between UC Integration for Microsoft Office Communicator endpoints in two clusters.	Exploratory	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.011	E2E RSVP, SIP Preconditions	E2E RSVP Call from a Phone in Central Site to Remote FXS Phone Registered to a Remote Branch (SIP GW) with Pre-conditions enabled	Verify an E2E RSVP call from a phone in central site to a remote FXS phone which is registered to a Remote Branch (SIP GW) with pre-conditions enabled.	Phone->RSVPAgent->Unified CM->SIP Trunk->Remote FXS Phone	Passed	
UC802EF.QOS.012	E2E RSVP, SIP Preconditions	E2E RSVP Call from a Remote Phone to Another Remote FXS Phone Registered to a Remote Branch (SIP GW) with Pre-conditions Enabled	Verify an E2E RSVP call from a remote phone to another remote FXS phone registered to a Remote Branch (SIP GW) with pre-conditions enabled.	Remote Phone->RSVP Agent->Unified CM->SIP Trunk->Remote FXS Phone	Passed	
UC802EF.QOS.013	E2E RSVP, SIP Preconditions	E2E RSVP Call from Remote FXS Phone to Another Remote FXS Phone	Verify an E2E RSVP call from a remote FXS phone to another remote FXS phone where both the remote branches are SIP GWs with pre-conditions enabled.	Remote FXS Phone 1->SIP Trunk->Unified CM->SIP Trunk->Remote FXS Phone 2	Passed	
UC802EF.QOS.014	E2E RSVP, SIP Preconditions	E2E RSVP Call from Phone in Unified CM Cluster to Unified CME Phone Aggregated by Unified SIP Proxy	Verify an E2E RSVP call from a phone in Unified CM cluster to a Unified CME phone aggregated by Unified SIP Proxy.	Originating Phone->RSVPAgent->Unified CM Cluster 1->SIP Trunk->Unified SIP Proxy->CME->RSVPAgent->Terminating Phone	Passed	
UC802EF.QOS.015	E2E RSVP, SIP Preconditions	E2E RSVP Call from Remote Phone in Unified CM Cluster to Unified CME Phone Aggregated by Unified SIP Proxy	Verify an E2E RSVP call from a remote phone in Unified CM cluster to a Unified CME phone aggregated by Unified SIP Proxy .	Originating Phone->Remote Branch->Rem RSVPAgent->Unified CM Cluster->SIP Trunk->Unified SIP Proxy->CME->RSVPAgent->Terminating Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.016	E2E RSVP, SIP Preconditions	E2E RSVP Call from FXS Phone Registered to Remote Branch (SIP GW) that has Pre-conditions Enabled to Unified CME Phone Aggregated by Unified SIP Proxy	Verify an E2E RSVP call from a FXS phone registered to a Remote Branch (SIP GW) that has pre-conditions enabled to a CME phone aggregated by Unified SIP Proxy.	FXS Phone->Remote Branch SIP GW (With Preconditions)->Unified CM->SIP Trunk->Unified SIP Proxy->CME->IP Phone	Passed	
UC802EF.QOS.017	E2E RSVP, SIP Preconditions	Supplementary Services in an E2E RSVP Call Between Central Site Phones in two Unified CM Clusters	Verify supplementary services in an E2E RSVP call between central site phones in two Unified CM clusters. Check for Hold/Resume, Transfer and conference.	Originating Phone->RSVP Agent->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2->RSVP Agent->Terminating Phone	Passed	
UC802EF.QOS.018	E2E RSVP, SIP Preconditions	Supplementary Services in an E2E RSVP Call Between a Central Site Phone in one Cluster to Remote Phone in Another Cluster	Verify supplementary services in an E2E RSVP call between a central site phone in one cluster to a remote phone in another cluster. Check for Hold/Resume, Transfer and Conference.	Originating Phone->RSVP Agent->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2->RSVP Agent->Terminating Phone	Passed	
UC802EF.QOS.019	E2E RSVP, SIP Preconditions	Supplementary Services in an E2E RSVP Call Between Central Site Phones in two Unified CM Clusters Whose Location is Set to Hub-none	Verify supplementary services in an E2E RSVP call between central site phones in two Unified CM clusters whose location is set to hub-none. Check for Hold/resume, transfer and conference.	Originating Phone->RSVP Agent->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2->RSVP Agent->Terminating Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.020	E2E RSVP, SIP Preconditions	Supplementary Services in an E2E RSVP Call Between a Central Site Phone in one Unified CM Cluster with Location is Set to Hub-none to a Remote Phone in Another Cluster	Verify supplementary services in an E2E RSVP call between a central site phone in one Unified CM cluster with location set to hub-none to a remote phone in another cluster. Check for Hold/resume, transfer and conference.	Originating Phone->RSVP Agent->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2->RSVP Agent->Terminating Phone	Passed	
UC802EF.QOS.021	E2E RSVP, SIP Preconditions	Supplementary Services for an E2E RSVP Call Between two Remote Phones Across Separate Clusters	Verify supplementary services for an E2E RSVP call between two remote phones across separate clusters.	Remote Phone 1->RSVP Agent->Unified CM Cluster 1->SIP ICT->Unified CM Cluster 2->RSVP Agent->Remote Phone	Passed	
UC802EF.QOS.022	E2E RSVP, SIP Preconditions	Supplementary Services for an E2E RSVP Call from a Central Phone to Remote FXS Phone	Verify supplementary services for an E2E RSVP call from a central phone to Remote FXS phone where the gateway is enabled with SIP preconditions.	Originating Phone->RSVP Agent->Unified CM->SIP Trunk->Remote FXS Phone	Passed	
UC802EF.QOS.023	E2E RSVP, SIP Preconditions	Supplementary Services for an E2E RSVP Call from a Remote Phone to a Remote FXS Phone	Verify supplementary services for an E2E RSVP call from a remote phone to a remote FXS phone.	Remote IP Phone->RSVP Agent->Unified CM->SIP Trunk->Remote SIP GW->FXS Phone	Passed	
UC802EF.QOS.024	E2E RSVP, SIP Preconditions	E2E RSVP Call from Unified Personal Communicator in Unified CM Cluster to Unified CME Phone via Unified SIP Proxy	Verify E2E RSVP call from Unified Personal Communicator in Unified CM cluster to CME phone via Unified SIP Proxy.	CUPC->CUP->CCM->SIP trunk->Unified SIP Proxy->CME->IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.025	E2E RSVP, SIP Preconditions	E2E RSVP Call from IP Communicator in Central Site to Unified CME phone via Unified SIP Proxy	E2E RSVP call from IP Communicator in central site to Unified CME phone via Unified SIP Proxy.	CIPC->CCM->SIP Trunk->Unified SIP Proxy->CME->IP Phone	Passed	
UC802EF.QOS.026	E2E RSVP, SIP Preconditions	E2E RSVP Call from Unified Personal Communicator in one Cluster to another Unified Personal Communicator in a Different Cluster	Verify the E2E RSVP call from Unified Personal Communicator in one cluster to another Unified Personal Communicator in a different cluster.	CUPC->CUP->CCM->SIP Trunk->CCM->CUP->CUPC	Passed	
UC802EF.QOS.027	E2E RSVP, SIP Preconditions	E2E RSVP Call from Remote IP Phone to PSTN Phone Over SIP PRI	Verify E2E RSVP call from remote IP phone to PSTN phone over SIP PRI.	Remote IP Phone->CCM 0> SIP Trunk->SIP PRI GW->PSTN Phone	Passed	
UC802EF.QOS.028	E2E RSVP, SIP Preconditions	E2E RSVP Call from Remote IP Phone to PSTN Phone over SIP BRI	Verify the E2E RSVP call from remote IP phone to PSTN phone over SIP BRI.	Remote IP Phone->CCM 0> SIP Trunk->SIP PRI GW->PSTN Phone	Passed	
UC802EF.QOS.029	E2E RSVP, SIP Preconditions	E2E RSVP and Unified Enterprise Attendant Console Interoperability	Verify an E2E RSVP call when a call is transferred by Unified Enterprise Attendant Console to a phone in another cluster.	IP Phone->CUEAC->IP Phone->RSVP Agent->CCM 1->SIP ICT->CCM 2->RSVP Agent->IP Phone	Passed	
UC802EF.QOS.030	E2E RSVP, SIP Preconditions	E2E RSVP and Unified Enterprise Attendant Console interoperability	Verify an E2E RSVP call for a call transferred by Unified Enterprise Attendant Console to a remote phone of another cluster.	IP Phone->CUEAC->IP Phone->RSVP Agent->CCM 1->SIP ICT->CCM 2->RSVP Agent->Remote IP Phone	Passed	
UC802EF.QOS.201.1	RSVP Video	Basic E2E RSVP Video Intercluster Call	Verify that: Video EPs can call from one cluster to other cluster. Both the clusters invoke only one central RSVP agent per cluster.	IP Video Phone 99xx->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->UC Integration™ for Microsoft Office Communicator	Passed w/ Exception	CSCtf44669

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.201.2	RSVP Video	Basic E2E RSVP Video Intercluster Call	Verify that: Video EPs can call from one cluster to other cluster. Both the clusters invoke only one central RSVP agent per cluster.	IP Video Phone 99xx->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->IP Video Phone 99xx	Passed	
UC802EF.QOS.201.3	RSVP Video	Basic E2E RSVP Video Intercluster Call	Verify that: Video EPs can call from one cluster to another cluster. Both the clusters invoke only one central RSVP agent per cluster.	CUVA->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G	Passed	
UC802EF.QOS.202.1	RSVP Video	E2E RSVP Video Call Between Remote and Central Phones Between Two Different Clusters	Verify that: Remote video EP can call from one cluster to other cluster's central video EP. Cluster 1 must invoke its remote RSVP agent. Cluster 2 must invoke its central RSVP agent.	Remote IP Video Phone 99xx->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->CUVA	Passed	
UC802EF.QOS.202.2	RSVP Video	E2E RSVP Video Call Between Remote and Central Phones Between Two Different Clusters	Verify that: Remote video EP can call from one cluster to other cluster's central video EP. Cluster 1 must invoke its remote RSVP agent. Cluster 2 must invoke its central RSVP agent.	Remote IP Video Phone 99xx->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G	Passed	
UC802EF.QOS.202.3	RSVP Video	E2E RSVP Video Call Between Remote and Central Phones Between Two Different Clusters	Verify that: Remote video EP can call from one cluster to other cluster's central video EP. Cluster 1 must invoke its remote RSVP agent. Cluster 2 must invoke its central RSVP agent.	Remote UC Integration™ for Microsoft Office Communicator->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G	Passed w/ Exception	CSCtf44669

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.203.1	RSVP Video	E2E RSVP Video Call Between Two Remote Phones Between Two Different Clusters	Verify that: Remote video EPs can call from one cluster to other cluster. Both the clusters must invoke only one remote RSVP agent per cluster.	Remote CUVA->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->remote RSVP cluster 2->Unified CM cluster 2->remote IP Video Phone 99xx	Passed	
UC802EF.QOS.203.2	RSVP Video	E2E RSVP Video Call Between Two Remote Phones Between Two Different Clusters	Verify that: Remote video EPs can call from one cluster to other cluster. Both the clusters must invoke only one remote RSVP agent per cluster.	Remote 7985G->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->remote RSVP cluster 2->Unified CM cluster 2->remote IP Video Phone 99xx	Passed	
UC802EF.QOS.204.1	RSVP Video	E2E RSVP Video Intercluster Call and Transfer to Remote Site of Called Cluster	Verify that: Video EP can call from one cluster to other cluster. Both the clusters invoke only one central RSVP agent per cluster. Transfer the call to the remote Video EP of the second cluster.	IP Video Phone 99xx->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G->XFER->remote 7985G	Passed	
UC802EF.QOS.204.2	RSVP Video	E2E RSVP Video Intercluster Call and transfer to the Remote Site of Called Cluster	Verify that: Video EP can call from one cluster to other cluster. Both the clusters invoke only one central RSVP agent per cluster. Transfer the call to the remote Video EP of the second cluster.	IP Video Phone 99xx->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G->XFER->remote UC Integration™ for Microsoft Office Communicator	Passed w/ Exception	CSCtf44669

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.204.3	RSVP Video	E2E RSVP Video Intercluster Call and Transfer to Remote Site of Called Cluster	Verify that: Video EP can call from one cluster to other cluster. Both the clusters invoke only one central RSVP agent per cluster. Transfer the call to the remote Video EP of the second cluster	UC Integration™ for Microsoft Office Communicator->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G->XFER->remote CUVA	Passed w/ Exception	CSCtf44669
UC802EF.QOS.205.1	RSVP Video	E2E RSVP Video Intercluster Call and Conference with Remote Site of Calling Cluster	Verify that: Video EP can call from one cluster to other cluster. Both the clusters invoke only one central RSVP agent per cluster. The second phone initiates a conference with a Remote Video EP in first cluster.	IP Video Phone 99xx->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G->CNF->Unified CM cluster 1->remote IP Video Phone 99xx	Passed	
UC802EF.QOS.205.2	RSVP Video	E2E RSVP Video Intercluster Call and Conference with Remote Site of Calling Cluster	Verify that: Video EP can call from one cluster to other cluster. Both the clusters invoke only one central RSVP agent per cluster. The second phone initiates a conference with a Remote Video EP in first cluster.	7985G->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->7985G->CNF->Unified CM cluster 1->remote 7985G	Passed	
UC802EF.QOS.206.1	RSVP Video	E2E RSVP Video Call Between Two Remote Phones Between Two Different Clusters and Transfer to Central Site of Called Cluster	Verify that: Remote video EPs can call from one cluster to other cluster. Both the clusters must invoke only one remote RSVP agent per cluster. Transfer the call to the central Video EP of the called cluster.	Remote IP Video Phone 99xx->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->remote RSVP cluster 2->Unified CM cluster 2->remote 7985G->XFER->7985G	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.206.2	RSVP Video	E2E RSVP Video Call Between Two Remote Phones Between Two Different Clusters and transfer to Central Site of Called Cluster	Verify that: Remote video EPs can call from one cluster to other cluster. Both the clusters must invoke only one remote RSVP agent per cluster. Transfer the call to the central Video EP of the called cluster.	Remote UC Integration™ for Microsoft Office Communicator->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->remote RSVP cluster 2->Unified CM cluster 2->remote CUVA->XFER->7985G	Passed w/ Exception	CSCtf44669
UC802EF.QOS.207.1	RSVP Video	E2E RSVP Video Call Between Two Remote Phones Between Two Different Clusters and Conference with Central Site of Calling Cluster	Verify that: Remote video EPs can call from one cluster to other cluster. Both the clusters must invoke only one remote RSVP agent per cluster. The second phone initiates a conference with a central Video EP in the calling cluster.	Remote IP Video Phone 99xx->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->remote RSVP cluster 2->Unified CM cluster 2->remote UC Integration™ for Microsoft Office Communicator->CNF->Unified CM cluster 1->7985G	Passed w/ Exception	CSCtf44669
UC802EF.QOS.207.2	RSVP Video	E2E RSVP Video Call Between Two Remote Phones Between Two Different Clusters and Conference with Central Site of Calling Cluster	Verify that: Remote video EPs can call from one cluster to other cluster. Both the clusters must invoke only one remote RSVP agent per cluster. The second phone initiates a conference with a central Video EP in the calling cluster.	Remote CUVA->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->remote RSVP cluster 2->Unified CM cluster 2->remote UC Integration™ for Microsoft Office Communicator->CNF->Unified CM cluster 1->7985G	Passed w/ Exception	CSCtf44669
UC802EF.QOS.208.1	RSVP Video	E2E RSVP Video Call Between a Remote Phone of One Cluster to a Central Phone in Another Cluster where Location is Set as Hub-None	Verify that: Remote video EP from one cluster can call to other cluster central video EP. The phone in another cluster where location is set as hub-none.	Remote IP Video Phone 99xx->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->central CUVA	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.QOS.208.2	RSVP Video	E2E RSVP Video Call Between a Remote Phone of One Cluster to a Central Phone in Another Cluster where Location is Set as Hub-None	Verify that: Remote video EP from one cluster can call to other cluster central video EP. The phone in another cluster where location is set as hub-none.	Remote UC Integration™ for Microsoft Office Communicator->Unified CM cluster 1->remote RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->central 7985G	Passed w/ Exception	CSCtf44669
UC802EF.QOS.209	RSVP Video	E2E RSVP Video Call Between Central Phones of Two Different Clusters and Location is set as Hub-None for Both the End-Points	Verify that: Video EP can call from one cluster to other cluster. Both the clusters invoke only one central RSVP agent per cluster. The location is set as Hub-None for both the Eps	IP Video Phone 99xx->Unified CM cluster 1->central RSVP cluster 1->SIP ICT->central RSVP cluster 2->Unified CM cluster 2->UC Integration™ for Microsoft Office Communicator	Passed w/ Exception	CSCtf44669
UC802EF.SAF.001	Service Advertisement Framework	Unified CM Advertises SRST PSTN Prefix Information to SRST Sites in SAF Network	Verify that SAF CCD on Unified CM advertises the SRST sites DN ranges and its corresponding 'To PSTN' prefix to the SRST sites via SAF enabled SIP trunk in the network. Verify by making a PSTN call from SRST 1 to SRST 2 when WAN connectivity from SRST sites to Unified CM are down (All branch routers in SRST mode). Also verify by making a PSTN call from SRST 2 to SRST 1. Additionally, verify the scenarios by dialing VOIP number instead of direct PSTN Number from phone 1 in SRST1 to Phone 2 in SRST2, and from Phone 1 in SRST2 to Phone 2 in the SRST1 site.	Stage 1:Ph 1->SRST 1->PSTN->SRST 2-> Ph 2; Stage 2: Ph 1->SRST 2->PSTN->SRST 1-> Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.002	Service Advertisement Framework	Unified CM Advertises Modified PSTN Prefix Information to SRST Sites in SAF Network	Verify that SAF CCD on Unified CM advertises the SRST1 DN ranges and its corresponding modified 'To PSTN' prefix to the SRST sites via SAF enabled SIP trunk in the network, when the SRST1 'To PSTN' prefix is modified. Verify by making a PSTN call from SRST2 to SRST1, when the WAN connectivity from SRST sites to Unified CM cluster in main site is down (All branch routers in SRST mode). Verify by making a PSTN call from SRST1 to SRST2 . Additionally, verify the scenarios by dialing VOIP number instead of direct PSTN Numbers.	Stage 1: Ph 1->SRST 2->PSTN->SRST 1->Ph 2;Stage 2: Ph 1->SRST 1->PSTN->SRST 2->Ph 2	Passed w/ Exception	CSCtd16671
UC802EF.SAF.003	Service Advertisement Framework	SAF Advertises Non-Reachable Information to SRST Sites in SAF Network	Verify if a PSTN call from SRST2 to SRST1 is unsuccessful and does not result in any call loop, assuming that on SRST2 no static route exists about the SRST1, after Unified CM removes the advertised SRST1 DN ranges and its reachability in the network.	Unified CM removes SAF route to SRST1;Unified CM->SAF enabled SIP Trunk(udates the SRST2 via SAF Advertisements that SRST1 no longer reachable via SAF check the SRST1 route removed on the SRST2 router)	Passed	

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

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.005	Service Advertisement Framework	Unified CM Advertises SRST PSTN Prefix Information to SRST and Remote sites in SAF Network	Verify if a PSTN call can be made from SRST1 site to Remote site (in Gateway mode) when WAN connectivity between SRST1 and Unified UCM cluster is down. (Assuming there is no WAN connectivity issue between Remote Site to Unified CM and as well SAF CCD on Unified CM advertised the SRST sites DN pattern and its corresponding PSTN prefix information to the SRST sites in the SAF network).	Ph 1->SRST 1 (using the PSTN route learned via SAF)->PSTN->Remote Site (Gateway mode)->Unified CM1->Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.006	Service Advertisement Framework	Unified CM1 Advertises DN Information to other SAF Clients in the SAF Network	Verify the SAF CCD on Unified CM1 advertises its own site DN pattern and its reachability information to Unified CM2 cluster sites via SAF enabled SIP trunk between Unified CM1 and Unified CM2. Similarly verify the SAF CCD on Unified CM2 advertises its own site DN pattern and its reachability information to Unified CM1 cluster sites via SAF enabled SIP trunk between Unified CM2 and Unified CM1. Verify by making a VOIP call from Unified CM2 to Unified CM1 and second VOIP call from Unified CM1 to Unified CM2. Also check the End-to End RSVP reservation in the above scenarios.	Stage 1:Ph 1->Unified CM 2 -> SIP Precondition enabled SAF Trunk->Unified CM 1 ->Ph 2 Stage 2:Ph 1 ->Unified CM 1->SIP Precondition enabled SAF Trunk->Unified CM 2->Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.007	Service Advertisement Framework	Unified CM1 Advertises DN Information to Other Clients in the SAF Network in Load-Balancing mode	Verify the following: 1. The SAF CCD on Unified CM1 on SAF network advertises its own site DN pattern and its reachability information to Unified CM2 cluster sites via SAF enabled SIP Trunk and H.225 enabled SAF between the Unified CM1 and Unified CM2 . 2. The SAF CCD on Unified CM2 on SAF network advertises its own DN pattern and its reachability information to Unified CM1 cluster via SAF enabled SIP trunk. 3. The Unified CM1 SAF trunks are used in load balancing mode by making first VOIP call from Unified CM2 to Unified CM1, by making second VOIP call from Unified CM2 to Unified CM 1, by making third VOIP call from the Unified CM2 to Unified CM1, and all calls are successful. 4. Also verify the End-to-End RSVP reservation in the above scenarios.	Stage 1:Ph 1->Unified CM2-> SIP precondition enabled SAF Trunk ->Unified CM1 ->Ph 2 Stage 2:Ph 1->Unified CM2 ->H.225 Trunk (SAF enabled Trunk)->Unified CM1 ->Phone2 Stage 3:Ph 1->Unified CM2->SIP precondition enabled SAF Trunk->Unified CM1 ->Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.008	Service Advertisement Framework	Interworking of SAF Learnt Route and Static Route	Verify the following: 1. The SAF CCD on Unified CM2 advertises the DN pattern and its reachability information to Unified CM1 via SAF enabled SIP trunk between the Unified CM2 and Unified CM1 and also configure static SIP trunk on Unified CM1 to route the calls from Unified CM1 to Unified CM2 . 2. Verify Inter working of SAF learnt route and Static route on the Unified CM1 by making a VOIP call from Phone 1(its corresponding CSS first preference set to Static partition) in Unified CM1 call to Phone 2 in Unified CM2 and also verify by making a second call from Phone 2 (its corresponding CSS first preference set to SAF learned partition) in Unified CM1 call to Phone 3 in Unified CM2, both the calls are successful.	Stage 1:Phone 1(CSS to static partition) ->Unified CM1-> SIP ICT trunk -> Unified CM 2->Phone 2 Stage 2: Phone 2 (CSS to SAF learned partition)->Unified CM1-> SAF enabled SIP trunk ->Unified CM2->Phone 3	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.009	Service Advertisement Framework	Unified CM Client Service Advertisements in Redundancy Mode	Verify the following: 1. The SAF CCD on the Unified CM1 advertising its own DN pattern and reachability information twice in the network by associating the same SAF enabled SIP trunk created on the Unified CM1 to the Node1 and Node2. Bring the Unified CM1 Node 2 down, modify the DN pattern on the Unified CM1 and advertise the modified DN pattern by the SAF CCD service in the network. 2. Verify if Unified CM node 1 can advertise the service on the network even if node 2 is down by making a VOIP call (modified DN) from Unified CM2 to Unified CM1 (node2).(Assuming the SAF CCD on the Unified CM2 advertising its own DN Pattern and reachability information to Unified CM1).	Stage 1:Ph 1->Unified CM2->SAF -SIP trunk->Unified CM1(node 1)->Ph 2;Stage 2:Ph 2->Unified CM2->SAF-SIP trunk->Unified CM1(node 2)->Ph 2;Stage 3:Ph 1->Unified CM2->SAF-SIP Trunk->Unified CM 1(node 1)->Ph 2;Stage 4:Ph 1->Unified CM2->SAF-SIP Trunk->Unified CM 1 (node 1)->Ph 2	Passed	
UC802EF.SAF.010	Service Advertisement Framework	Unified CM Client Service Advertisements Passing through Non-SAF to SAF Network	Verify the SAF CCD on Unified CM1 advertisement reaches to SAF client Unified CM2 via SAF unaware network by making a VOIP call from the Unified CM1 to Unified CM2.	Stage 1: Ph 1->Unified CM1->SIP Trunk (SAF enabled)->SAF unaware Network->Unified CM 2->Ph 2Stage 2: Ph 1->Unified CM1->SIP Trunk (SAF enabled)->SAF unaware Network->Unified CM 2->Ph 2	Passed	

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

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

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.013	Service Advertisement Framework	Unified CM1 Advertises Information to Other Clients in the SAF Network and Checks RSVP Application ID	Verify the following: The SAF CCD on Unified CM1 advertises its own site DN pattern and its reachability information to Unified CM2 via SAF enabled SIP trunk between Unified CM1 and Unified CM2. 2. Verify the RSVP Application ID functionality on Unified CM1 by making a Video call from Phone 1 in Unified CM1 to Phone 2 in Unified CM2 and then by making an audio call from Phone 2 in Unified CM1 to Phone 3 in Unified CM2. Now, de-escalate the video call to audio call by pressing MUTE on Phone 1 in Unified CM1.	Stage 1:Video Ph 1->Unified CM1->SIP Trunk (RSVP-SAF)->Unified CM2->Video Ph 2 Stage 2:Ph 1->Unified CM1->SIP Trunk (RSVP-SAF)->Unified CM 2->Ph 2 Stage 3:Video ph1(mute ON)->Unified CM1->SIP Trunk(RSVP-SAF)->Unified CM 2->Video ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.014	Service Advertisement Framework	Unified CM1 Advertises DN Information to Other Clients in SAF Network	Verify the following: 1. The SAF CCD on Unified CM1 advertises its own site DN pattern and "To PSTN" prefix information to Unified CM2 via SAF enabled SIP trunk between Unified CM1 and Unified CM2. 2. Verify by making a VOIP call from the Unified CM2 to Unified CM1, and then bring down the WAN connection between the Unified CM1 and Unified CM2 down. Verify by making a PSTN call from central Phone 1 in Unified CM2 to Phone 1 in SRST Site. Also verify by making a PSTN call from the SRST (in Unified CM1 site) to Unified CM2 .	Stage 1:Ph 1->Unified CM 1->SIP Trunk (RSVP enabled SAF)->Unified CM2->Ph 2 Stage 2:Ph 1->Unified CM 2->MGCPGW->PSTN->MGCP GW->Unified CM 1->Ph 1Stage 3: Ph 1->Unified CM1->MGCP Gateway->PSTN->MGCP Gateway->Unified CM2->Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.015	Service Advertisement Framework	Unified CM1 SAF Traffic Load-Balanced Across Call-Manager Nodes	Verify the following: 1. On Unified CM2, create two SAF enabled SIP Trunks, associate first trunk to CCD Advertising service and Unified CM1 group consisting of Node1, Node2 and Node3. Similarly associate the second trunk to CCD requesting service and Unified CM1 group consisting of Node4, Node5, Node 6 on Unified CM1 Cluster. By CCD advertising service on Unified CM1 advertise its DN pattern in the Network. 2. By SAF CCD on Unified CM2 advertise the DN Pattern it serves in the Network via SAF enabled SIP Trunk created between Unified CM2 and Unified CM1. 3. Verify that on Unified CM1,the calls from Unified CM2 is load-balanced across Unified CM1 nodes by making four consecutive VOIP calls from Unified CM2 to Unified CM1. Make sure the first call from Unified CM2 is served by Unified CM1 node1, second call is served by Unified CM1 Node 2, third Call by Unified CM1 Node	Stage 1:Ph 1to4->Unified CM2->SAF enabled SIP Trunk->Unified CM1 Node1-3 1->Ph 5-6 Stage 2:Ph 1to4->Unified CM1Node 4-6 4->SAF enabled SIP trunk->Unified CM2->Ph 5-6	Passed	CSCtd71187

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.016	Service Advertisement Framework	SAF - Forwarder Failure Results in Alternate PSTN Routing	Verify the SAF Forwarder failure results in PSTN routing: 1. On Unified CM1 by SAF CCD service advertise its DN pattern and "To PSTN" reachability information via SAF enabled SIP Trunk created between the Unified CM1 and Unified CM2. 2. Assuming SAFF1 is acting for the Service Advertisement for Unified CM1, and SAFF2 is acting for the Service Advertisement for the Unified CM2, Verify by making first VOIP call from Unified CM2 to Unified CM1, then bring down SAFF1 in the network. Ensure that on Unified CM2 it marks the DN pattern it learned as down and starts the age-out timer. After the Age-out Timer of DN pattern is learnt, make second call from Unified CM2 to Unified CM1, and it should be routed via the PSTN network. Then Bring back SAFF1 in Unified CM1 site and modify the DN Pattern on Unified CM1 and advertise it in the SAF network. Verify by making a VOIP call from Unified CM2 to Unified CM1.	Stage 1:Ph 1->Unified CM2->SAF enabled SIP Trunk->Unified CM 1->Ph 2 Stage 2:Ph 1 - Unified CM2->MGCP GW->PSTN->MGCP GW->Unified CM 1->Ph 2 Stage 3:Ph 1->Unified CM2->SAF enabled SIP Trunk->Unified CM 1->Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.017	Service Advertisement Framework	Join Across Lines over SAF Enabled SIP Trunk	Verify if the SAF CCD on Unified CM1 advertises its own site DN pattern and its reachability information to Unified CM2 cluster sites via SAF enabled SIP trunk between Unified CM1 and Unified CM2. Similarly verify if the SAF CCD on Unified CM2 advertises its own site DN pattern and its reachability information to Unified CM1 cluster sites via SAF enabled SIP trunk between Unified CM2 and Unified CM1. Also verify if the Join Across Lines over SAF enabled SIP Trunk works fine.	Stage 1:Phone 1(first line) ->Unified CM2->SAF enabled SIP trunk->Unified CM1 (Phone 2: Stage 2:Phone 1->Unified CM2->SAF enabled SIP Trunk (Unified CM1->Phone 3	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.018	Service Advertisement Framework	Location Based CAC on Calling Cluster Results in Alternate Routing	Verify the following: 1. On the Unified CM1 restrict location based CAC bandwidth setting for one audio call, then by SAF CCD service on Unified CM1 and Unified CM2 advertise DN pattern and its "To PSTN" information via SAF enabled SIP trunks, to inform each Unified CM the other Unified CM's DN pattern and reachability information. 2. Verify the location based CAC on the Calling Cluster (Unified CM1) by making VOIP call from the Unified CM1 to Unified CM2; the calls should be connected successfully. Then make the consult transfer from Unified CM1 to Unified CM2. Locations based CAC bandwidth settings on the Unified CM1, will reject the transfer, and should get routed over the PSTN and get connected successfully to Unified CM2.	Stage 1:Phone 1->Unified CM1->SAF enabled SIP Trunk->Unified CM2->Phone 2 Stage 2: Phone 1->Unified CM1->MGCP->PSTN->MGCP (Unified CM2->Phone 3	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.019	Service Advertisement Framework	Location Based CAC on Calling Cluster Results in Alternate Routing	Verify the following: 1. On Unified CM2, restrict the location based CAC bandwidth setting for one audio call, then by SAF CCD service on Unified CM1 and Unified CM2 advertise DN pattern and its "To PSTN" information via SAF enabled SIP trunks, to publicize each Unified CM about the other's DN pattern and reachability information. 2. Verify the location based CAC on the called cluster (Unified CM2) by making VOIP call from Unified CM1 to Unified CM2; the calls should be connected successfully. Make a consult transfer from Unified CM1 to Unified CM2. Locations based CAC bandwidth settings on the Unified CM2 will reject the transfer, and verify if it gets routed over the PSTN successfully to Unified CM2.	Stage 1:Phone 1->Unified CM1->SAF enabled SIP Trunk->Unified CM2->Phone 2 Stage 2:Phone 1->Unified CM1->MGCP->PSTN->MGCP (Unified CM2->Phone 3	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.020	Service Advertisement Framework	SAF Working on SRST During Connection Monitor Period	Verify SAF working on SRST during connection monitor period, assuming that SAF CCD on the Unified CM advertises the SRST sites DN ranges and its corresponding 'To PSTN' prefix to the SRST sites via SAF enabled SIP trunk in the network. Verify by making a PSTN call from SRST 1 to Central Phone 1 in Unified CM1 when WAN connectivity from the SRST 1 to Unified CM is down. Bring back the WAN connectivity, Verify by making a call in SRST1 site phone to Central Phone 1 in Unified CM1 and before the expiry of connection monitor, the call should be routed via PSTN.	Stage 1:Phone 1->SRST 1->PSTN->Unified CM ->Central Phone 2;Stage 2:Phone 1->SRST 1->PSTN->Unified CM (Central Phone 2	Passed	
UC802EF.SAF.021	Service Advertisement Framework	Age-Out Timer Expiry and PSTN Flush Out Timer Expiry	Verify if the learned DN Patterns are marked down when connectivity to SAFF is lost. DN or patterns are flushed out after age-out timer expiry and PSTN routes are also deleted after PSTN age-out expiry.	Stage 1:Ph 1->Unified CM2->SAF enabled SIP Trunk->Unified CM1->Ph 2;Stage 2: Ph 1->Unified CM2->SAF enabled SIP trunk->Unified CM1->Ph 1 Stage 3: Ph 1->Unified CM1->MGCP->PSTN->MGCP->Unified CM2->Ph 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.022	Service Advertisement Framework	Unified CM Client Advertises its Own DN Pattern to Unified CME Client in SAF Network	Verify if Unified CM on Service Advertisement Framework (SAF) network advertises its own site DN pattern and its reachability information to other Unified CME sites through SIP preconditions enabled SAF Trunk by making a VOIP call from the Unified CME to Unified CM.	Variation 1 :Phone 1->Unified CME->SIP Precondition enabled SAF Trunk->Unified CM->Phone 2;Variation 2: Phone 1->Unified CM->SIP Precondition enabled SAF Trunk->Unified CME->Phone 2	Passed	
UC802EF.SAF.023	Service Advertisement Framework	Unified CM Client Advertises Modified DN Pattern to Unified CME Client in SAF Network	Verify if Unified CM on SAF network advertises its own site modified DN pattern and its reachability information to other Unified CME cluster sites through SIP preconditions enabled SAF Trunk by making a VOIP call from the Unified CME to Unified CM.	Variation 1 :Phone 1->Unified CME->SIP Precondition enabled SAF Trunk->Unified CM->Phone 2;Variation 2: Phone 1->Unified CM->SIP Precondition enabled SAF Trunk->Unified CME->Phone 2	Passed	
UC802EF.SAF.024	Service Advertisement Framework	Interworking of SAF Learnt Route and Static Route	Verify if static route interworks with the route learnt from the Service Advertisement Framework network for the same destined service on a Unified CM by making VOIP call from the Unified CM to Unified CME by varying the learnt partition and CSS on the Unified CM. Also check for the interworking of SAF learnt route and static route on the Unified CME to Unified CM (vice versa call flow) as well.	Variation1 : Phone 1->Unified CM->SIP Precondition enabled SAF Trunk->Unified CME->Phone 2;Variation2: Phone 1->Unified CME->SIP Precondition enabled SAF Trunk->Unified CM->Phone 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.025	Service Advertisement Framework	Unified CME Client Advertises Modified DN Pattern to Unified CM Cluster2 Client in the SAF Network	Verify that the newly added DN Pattern in Unified CME gets advertised to Cluster2 through SAF network directly.	Variation 1: Phone 1->Unified CM Cluster2->SIP precondition enabled SAF Trunk ->Unified CME->Phone 2;Variation 2: Phone 1->Unified CME->SIP precondition enabled SAF Trunk ->Unified CM Cluster2->Phone 2	Passed	
UC802EF.SAF.026	Service Advertisement Framework	Unified CME Client Advertises PBX DN Pattern to other Unified CM Clients in the SAF Network	Verify if Unified CM Cluster can learn the PBX route pattern from the Unified CME Client and route the call when the already associated static CME PBX route pattern in the Unified CM cluster has been deleted.	Variation1:PBXPh1->QSIG PRI->Unified CM->SIP Precond enabled SAFTrunk->CME->QSIG PRI->PBXPh2;Variation2: IPPh1->Unified CM->SIP Precond enabled SAFTrunk->CME->QSIG PRI->PBX Ph2	Passed	
UC802EF.SAF.027	Service Advertisement Framework	Unified CME Client Advertises DN Pattern to Unified CME Client in the SAF Network Through SIP and H.225	Verify the following: 1. A call can be made from a Unified CME registered to Unified SIP Proxy to a Unified CME client that is associated with H.225 Trunk in the Unified CM cluster. 2. When Unified CME1 is registered to Unified SIP Proxy and Unified CME2 to a CME gatekeeper, a call made fails initially since there is no route pattern in Unified CME1 to reach Unified CME2. 3. After Unified CME1 advertises its modified DN range to Unified CME2 Cluster, a call made from Unified CME1 to Unified CME2 should work.	Variation1 :Ph1->Unified CME1->SAF trunk->Unified CME2->Ph2 (advertisement);Variation 2: Ph1->Unified CME2->H225 trunk->Unified CME1->Ph2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.028	Service Advertisement Framework	Unified CM Client Advertises its Remote DN Pattern and "TO PSTN" Prefix Information to Other Unified CME Clients in SAF Network	Verify if Unified CM on Service Advertisement Framework (SAF) network advertises its remote site DN pattern and "TO PSTN" prefix information to other Unified CME client through SIP preconditions enabled SAF Trunk by making a VOIP call from the Unified CME to remote DN.	Variation 1 : Phone 1->Unified CME->SIP Precondition enabled SAF Trunk->Unified CM->Rem Phone 2;Variation 2: Rem Phone 1->Unified CM->SIP Precondition enabled SAF Trunk->Unified CME->Phone 2	Passed	
UC802EF.SAF.029	Service Advertisement Framework	Unified CM Client Advertises its Modified Remote DN Pattern and "TO PSTN" prefix information to Other Unified CME Clients in the SAF Network	Verify if Unified CM on Service Advertisement Framework network advertises its remote site modified DN pattern and "TO PSTN" prefix information to other Unified CME client through SIP preconditions enabled SAF Trunk by making a VOIP call from the Unified CME to Unified CM.	Variation 1 : Ph1->Unified CME->SIP Precondition enabled SAF Trunk->Unified CM1->Rem Ph2;Variation 2 : Rem Ph1->Unified CM1->SIP Precondition enabled SAF Trunk->Unified CME->Ph2	Passed	
UC802EF.SAF.030	Service Advertisement Framework	Unified CME Client Advertises its own DN pattern and "To PSTN" prefix to SRST Client in the SAF Network	Verify if Unified CME on Service Advertisement Framework (SAF) network advertises its own site DN pattern and "To PSTN" prefix and its reachability information to SRST client in the SAF network.	Unified CM is down: Variation 1:Phone 1->Unified CME->PSTN->SRST->Phone 2; Variation 2: Phone1->SRST->PSTN->Unified CME->Phone2	Passed	
UC802EF.SAF.031	Service Advertisement Framework	Unified CME Client Advertises Modified DN Pattern and "To PSTN" prefix to SRST client in the SAF Network	Verify if Unified CME on Service Advertisement Framework network advertises its own site modified DN pattern and "To PSTN" prefix and its reachability information to SRST client in the SAF network.	Unified CM is down: Variation 1:Phone 1->Unified CME->PSTN->SRST->Phone 2;Variation 2: Phone1->SRST->PSTN->Unified CME->Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.SAF.032	Service Advertisement Framework	Load balancing of Calls to Remote Unified CM Cluster Advertising Same HostedDN from Unified CME	Verify when a trunk in the advertising cluster is assigned to two Unified CM, Unified CME receives the same pattern twice, once for each Unified CM node. Calls from CME to the advertised DN should alternate between the two Unified CM nodes.	Variation 1 :Phone 1->Unified CME->SIP Precondition enabled SAF Trunk->Unified CM->Phone 2;Variation 2: Phone 1->Unified CM->SIP Precondition enabled SAF Trunk->Unified CME->Phone 2	Passed	
UC802EF.SAF.033	Service Advertisement Framework	Calls to Cisco Unity Express Registered to Unified CM over SAF Trunks	Verify that route points can be advertised and calls can be placed successfully over SAF trunk.	Phone 1->Unified CM->SIP Precondition enabled SAF Trunk->Unified CM->Phone 2->JTAPI->CUE	Passed	
UC802EF.SAF.034	Service Advertisement Framework	Calls to Cisco Unity Express Registered to Unified CME over SAF Trunks	Verify that route points can be advertised and calls can be placed successfully over SAF trunk.	Phone 1->Unified CM->SIP Precondition enabled SAF Trunk->Unified CME->Phone 2->JTAPI->CUE	Passed	
UC802EF.SAF.035	Service Advertisement Framework	End to End RSVP over SIP- SAF Trunks and check for RSVP Application IDs	Verify that the SAF CCD on Unified CM1 advertisement reaches Unified CME SAF enabled trunk between Unified CM1 and Unified CME. Check the End-to-End RSVP reservation in this scenario.	Stage 1:Ph 1->Unified CM1->SIP Trunk (SIP precond enabled SAF Trunk)->Unified CME->Ph 2 Stage 2: Ph 3->Unified CM1->SIP Trunk(SIP precond enabled SAF Trunk)->Unified CME-> Ph 4	Passed	
UC802EF.SAF.036	Service Advertisement Framework	BLF Indication Available for Unified CM1 Phone after Learning Unified CM2's DN Pattern and Busy Information	Verify that SAF CCD on Unified CM1 advertises its own site DN pattern and its reachability information to Unified CM2 via SAF enabled SIP trunk between Unified CM1 and Unified CM2. Also verify that the BLF indication is available for Unified CM1 Phone when Unified CM2's Phone is busy on another call.	Stage1 : Ph 2->Unified CM2->Ph3 ; Stage 2: Ph 1->Unified CM1->SIP Trunk (RSVP- SAF)->Unified CM2->Ph 2 (Ph1 Should get the BLF indication)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.UNC.001	Unity Connection	Migration of Cisco Unity 7.x to Unity Connection 8.0	Verify the migration of Unity 7.x high-availability setup to Unity Connection 8.0.		Passed	
UC802EF.UNC.002	Unity Connection	Visual Voicemail with Unity Connection	Verify visual voicemail capability on Unified IP Phones (8961/9951/9971) with Unity Connection	IP Phones (8961/9951/9971)->CCM->Unity Connection	Passed	
UC802EF.UNC.003	Unity Connection	Enhanced MWI with Unity Connection	Verify enhanced MWI capability on Unified IP Phones (8961/9951/9971) with Unity Connection.	IP Phone 99xx/89xx->CCM->Unity Connection	Passed	
UC802EF.UNC.004	Unity Connection	Voicemail for SIP IP Phones (8961/9951/9971) in Unified CM 8.0 Cluster	Verify deposit and retrieval of a message for a Unified IP Phones (8961/9951/9971) in Unified CM 8.0 cluster.	Phone->PSTN GW->CCM->IP Phone 99xx/89xx->CFNA->Unity Connection	Passed	
UC802EF.UNC.005	Unity Connection	Voicemail for SCCP Unified IP Phones 6921/6941/6961 in Unified CM 8.0 cluster	Verify deposit and retrieval of a message for Unified IP Phones 6921/6941/6961 in Unified CM 8.0 cluster.	Phone->PSTN GW->CCM->IP Phone 69xx->CFNA->Unity Connection	Passed	
UC802EF.UNC.006	Unity Connection	Voicemail for QSIG PBX Phone in Unified CM 8.0 Cluster	Verify deposit and retrieval of a message for a QSIG PBX phone in Unified CM 8.0 cluster.	Phone->CCM->QSIG PBX->PBX Phone->CFNA->Unity Connection	Passed	
UC802EF.UNC.007	Unity Connection	Voicemail for IP Phone 99xx/89xx (SIP) in CCM 7.x Cluster	Verify deposit and retrieval of a message for a RT phone in CCM 7.x cluster	Phone->PSTN GW->CCM 7.x->IP Phone 99xx/89xx->CFNA->Unity Connection	Passed	
UC802EF.UNC.008	Unity Connection	Voicemail for SCCP Unified IP Phones 6921/6941/6961 in Unified CM 7.x cluster	Verify deposit and retrieval of a message for Unified IP Phones 6921/6941/6961 in Unified CM 7.x cluster.	Phone->PSTN GW->CCM 7.x->IP Phone 99xx/89xx->CFNA->Unity Connection	Passed	
UC802EF.UNC.009	Unity Connection	Voicemail for QSIG PBX Phone in Unified CM 7.x Cluster	Verify deposit and retrieval of a message for a QSIG PBX in Unified CM 7.x cluster.	Phone ->CCM 7.x->QSIG PBX->PBX Phone->CFNA->Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.UNC.010	Unity Connection	Voicemail for Unified IP Phones in Unified CME site	Verify deposit and retrieval of a message for Unified IP Phones in Unified CME site over H.323 network.	Phone->PSTN GW->CME->Phone->CFNA->Unity Connection	Passed w/ Exception	This testcase was executed with TNP phones and not with IP Phones 99xx/89xx series as some of these phones are not supported with Unified CME.
UC802EF.UNC.011	Unity Connection	Voicemail for QSIG PBX Phone in Unified CME Site	Verify deposit and retrieval of a message for a QSIG PBX phone in Unified CME site over H.323 network.	Phone->CME->QSIG PBX->PBX Phone->CFNA->Unity Connection	Passed	
UC802EF.UNC.012	Unity Connection	Voicemail Server Redundancy/Negative testing	Verify Unity Connection voicemail server redundancy.		Passed	
UC802EF.UNC.013	Unity Connection	Supervised Transfer in Unity Connection	Verify operation of supervised transfer in Unity connection.		Passed	
UC802EF.UNC.014	Unity Connection	Directory Handlers in Unity Connection	Verify operation of directory handlers in Unity connection.		Passed	
UC802EF.UNC.015	Unity Connection	Interview Handlers in Unity connection	Verify operation of interview handlers in Unity connection.		Passed	
UC802EF.UNC.016	Unity Connection	Voicemail for Remote Phones Over H.323/FXO PSTN Gateway	Verify voicemail deposit and retrieval for remote phones over H.323/FXO PSTN gateway.		Passed	
UC802EF.UNC.016	Unity Connection	Voicemail for Remote Phones Over MGCP/FXO PSTN Gateway	Verify voicemail deposit and retrieval for remote phones over MGCP/FXO PSTN gateway.		Passed	
UC802EF.UNC.016	Unity Connection	Voicemail for Remote Phones Over MGCP/PRI PSTN Gateway	Verify voicemail deposit and retrieval for remote phones over MGCP/PRI PSTN gateway.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.UNC.016	Unity Connection	Voicemail for Remote Phones Over MGCP BRI PSTN Gateway	Verify voicemail deposit and retrieval for remote phones over MGCP BRI PSTN gateway.		Passed	
UC802EF.UNC.016	Unity Connection	Voicemail for Remote Phones Over SIP/BRI PSTN Gateway	Verify voicemail deposit and retrieval for remote phones over SIP/BRI PSTN gateway.		Passed	
UC802EF.UNC.016	Unity Connection	Voicemail for Remote Phones Over Various PSTN Gateways	Verify voicemail deposit and retrieval for remote phones over various PSTN gateways.		Passed	
UC802EF.UNC.016	Unity Connection	Voicemail for Remote Phones Over H.323/PRI PSTN Gateway	Verify voicemail deposit and retrieval for remote phones over H.323/PRI PSTN gateway.		Passed	
UC802EF.UNC.016	Unity Connection	Voicemail for Remote Phones Over SIP/PRI PSTN Gateway	Verify voicemail deposit and retrieval for remote phones over SIP/PRI PSTN gateway.		Passed	
UC802EF.UNC.017	Unity Connection	Voicemail for DPNSS PBX Phone with Unity Connection	Verify deposit and retrieval of messages for a DPNSS PBX phone registered to a Unified CM 8.0 cluster.	IP Phone->Unified CM->VG 30D->DPNSS PBX->Unified CM->DPNSS PBX Phone->CFNA->Unity Connection	Passed	
UC802EF.UNC.018	Unity Connection	Voicemail for a Remote Unified IP Phones 8961/9951/9971 Registered to Unified CM 7.x Cluster	Verify deposit and retrieval of messages for a Remote Unified IP Phones 8961/9951/9971 registered to Unified CM 7.x cluster.	PSTN Phone->PSTN GW->Rem IP Phone 99xx/89xx->CCM 7.x->Unity Connection	Passed	
UC802EF.UUE.001	Voicemail	Reply to Voicemail with Call or IM or Voicemail	Verify if Unified IP Phone 9971 receives a voice mail from SIP phone and can reply through voicemail, or IM or call.	stage 1: SIP Ph 1->Unified CM->9971 Ph->Voice Mail;Stage 2:9971 Ph->Unified CM->IM->SIP Ph1 ; Stage 3 :9971 Ph->Unified CM->SIP ph1;Stage 4:9971 Ph->Unified CM->SIP Ph 1->Voice mail.	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.UUE.002	Voicemail	Display of Text Transcript of Voicemail in Context of Visual Voicemail	Verify if Unified Personal Communicator receives a voice mail from the SCCP Phone and if the text transcript of voice mail is available in the context of Visual voice mail.	SCCP Ph 1->Unified CM->CUPC 1->Voice mail	Passed w/ Exception	Text Transcript feature is not supported in VVM for UC 8.0.
UC802EF.UUE.003	Visual Voicemail	Inline Playback of Visual Voicemail	Verify when Unified Personal Communicator receives a voice mail from the SIP Phone, toast pop up of visual voice mail indication is received and playback of visual voice mail is done with fast forward and rewind options.	SIP Ph 1->Unified CM-> CUPC 1->Voice mail	Passed w/ Exception	Text Transcript feature is not supported in VVM for UC 8.0.
UC802EF.UUE.004	Presence	Reply to Voicemail with Call or IM or Voicemail	Verify if Unified Personal Communicator receives a voice mail from SIP phone and can reply through Instant messaging or calling or through voicemail.	stage 1: SIP Ph 1->Unified CM->CUPC 1->Voice Mail;Stage 2:CUPC 1->Unified CM->IM->SIP Ph1 ; Stage 3 :CUPC 1->Unified CM->SIP ph1;Stage 4:CUPC 1->Unified CM->SIP Ph 1->Voice mail.	Passed w/ Exception	CSCtd33303
UC802EF.UUE.005	Presence	Real-time Presence Status in Unified Personal Communicator	Verify when Unified Personal communicator is idle and when an incoming call from SIP phone is answered in, the presence status changes from idle to Busy.	SIP Ph->Unified CM-> CUPC	Passed	
UC802EF.UUE.006	Buddy Lists and Contact	Realtime Presence Status in Unified IP Phone 9971	Verify when a Unified IP Phone 9971 presence status is idle and when an incoming call from SIP phone is answered, the presence status changes from idle to Busy.	SIP Ph->Unified CM->9971 Ph	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EF.UUE.007	Buddy Lists and Contact	Click to Add Add Participant to Voice Session in Unified Personal Communicator	Verify if Unified Personal Communicator can add a participant from contact list to conference with active calls.	Stage 1: SIP Ph 1->Unified CM->CUPC ;Stage 2 : CUPC->Unified CM->Conference->SIP Ph 2.	Passed w/ Exception	"Click to Add" participants to IM sessions or call is not available for UC 8.0. Only Click to add into contacts is available.
UC802EF.UUE.008	Buddy Lists and Contact	Click to Add Add Participant to Voice Session in IP Phone 9971	Verify if Unified IP Phone 9971 can add a participant from contact list to conference with active calls.	Stage 1: SIP Ph 1->Unified CM->9971 Ph ;Stage 2 : 9971 Ph->Unified CM->Conference->SIP Ph 2.	Passed w/ Exception	"Click to Add" participants to IM sessions or call is not available for UC 8.0. Only Click to add into contacts is available.
UC802EL.PER.001	Reliability	Basic IP to IP Intra Cluster Calls for One Day Load Run	Verify that IP to IP basic calls and supplementary services calls in Cisco Unified Communications Manager Business Edition (Unified CMBE) with 1530 BHCA are successful for one day load run.	SCCP Phone1->Unified CMBE->SCCP Phone2	Passed	
UC802EL.PER.002	Reliability	Basic IP to IP Intra Cluster Calls for Three Day Load Run	Verify that IP to IP basic calls and supplementary services in Unified CMBE with 1530 BHCA successful for three day load run.	SCCP Phone1->Unified CMBE->SCCP Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EL.PER.003	Reliability	IP to IP Calls Over PSTN in Unified CMBE for One Day Load Run	Verify that IP to PSTN-to IP basic calls and supplementary services in Unified CMBE with 486 BHCA successful for one day load run.	SCCP Phone1->Unified CMBE->MGCP g/w->PSTN->MGCP g/w->Unified CMBE->SCCP Phone2	Passed	
UC802EL.PER.0038	Reliability	Basic IP to IP Intra cluster Calls	Verify that IP to IP calls in a large size cluster of 1692 BHCA are successful for 24 hours.	SCCP Phone1->Unified CM->SCCP Phone2	Passed	
UC802EL.PER.0039	Reliability	Basic IP to IP Intra cluster Calls	Verify that IP to IP calls in a large size cluster of 1692 BHCA are successful for 72 hours.	SCCP Phone1->Unified CM->SCCP Phone2	Passed	
UC802EL.PER.004	Reliability	IP to IP Calls Over PSTN in Unified CMBE for Three Day Load Run	Verify that IP to PSTN-to IP basic calls and supplementary services in Unified CMBE with 486 BHCA are successful for three day load run.	SCCP Phone1->Unified CMBE->MGCP g/w->PSTN->MGCP g/w->Unified CMBE->SCCP Phone2	Passed	
UC802EL.PER.0040	Reliability	IP to IP Calls Over PSTN	Verify that IP to IP calls over PSTN in a large size cluster of 8460 BHCA are successful for 24 hours.	SCCP Phone 1->Unified CM->MGCP g/w ->PSTN->MGCP g/w ->Unified CM->SCCP Phone2;SIP Phone1->Unified CM->MGCP g/w->PSTN->MGCP g/w->Unified CM->SIP Phone2	Passed	
UC802EL.PER.0041	Reliability	IP to IP Calls Over PSTN	Verify that IP to IP calls over PSTN in a large size cluster of 8460 BHCA are successful for 72 hours.	SCCP Phone 1->Unified CM->MGCP g/w ->PSTN->MGCP g/w ->Unified CM->SCCP Phone2;SIP Phone1->Unified CM->MGCP g/w->PSTN->MGCP g/w->Unified CM->SIP Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EL.PER.0042	Reliability	IP to IP Calls Over Inter Cluster Trunk	Verify that IP to IP Inter Cluster Trunk (ICT) calls with supplementary services (hold, bxfer, cxfer) in a large size cluster 2520 BHCA are successful.	SCCP Phone1->Unified CM->ICT Trunk1 (QSIG ICT)->Unified CM->SCCP Phone2	Passed	
UC802EL.PER.0044	Reliability	IP to IP Calls Over Inter Cluster Trunk with Unified CM 7.x	Verify that IP to IP Inter Cluster Trunk (ICT) calls with supplementary services (hold, bxfer, cxfer) in a large size cluster 2520 BHCA are successful for 24 hours.	SCCP Phone1->Unified CM->ICT Trunk1 (QSIG ICT)->Unified CM(Release 7.x)->SCCP Phone2	Passed	
UC802EL.PER.0045	Reliability	IP to IP Calls Over Inter Cluster Trunk with Unified CM 7.x	Verify that IP to IP Inter Cluster Trunk (ICT) calls with supplementary services (hold, bxfer, cxfer) in a large size cluster 2520 BHCA are successful for 72 hours.	SCCP Phone1->Unified CM->ICT Trunk1 (QSIG ICT)->Unified CM(Release 7.x)->SCCP Phone2	Passed	
UC802EL.PER.0046	Reliability	IP to IP Calls Between Unified CM Express and Unified CM Over H.323 Gatekeeper Controller Trunk	Verify that IP to IP calls between Unified CM Express to Unified CM of 3960 BHCA are successful for 24 hrs.	SCCP Phone1->Unified CM->GateKeeper 1 ->IPIPGW 1->GateKeeper 1-> GK 2-> CME ->SCCP Phone2	Passed	
UC802EL.PER.0047	Reliability	IP to IP Calls Between Unified CM Express and Unified CM Over H.323 Gatekeeper Controller Trunk	Verify that IP to IP calls between Unified CM Express to Unified CM of 3960 BHCA are successful for 72 hrs.	SCCP Phone1->Unified CM->GateKeeper 1 ->IPIPGW 1->GateKeeper 1-> GK 2-> CME ->SCCP Phone2	Passed	
UC802EL.PER.0048	Reliability	Calls to Unity Connection	Verify that Cisco Unity Connection calls of 3600 BHCA are successful for 24 hrs.	SCCP Phone1->Unified CM->SCCPPhone2->CFNA->UNITY; SCCPPhone2->UNITY->RETRIEVAL	Passed	
UC802EL.PER.0049	Reliability	Calls to Unity Connection	Verify that Cisco Unity Connection calls of 3600 BHCA are successful for 72 hrs.	SCCP Phone1->Unified CM->SCCPPhone2->CFNA->UNITY ;SCCPPhone2->UNITY->RETRIEVAL	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EL.PER.005	Reliability	IP to IP Calls Over Remote PSTN in Unified CMBE for One Day Load Run	Verify that IP to PSTN-to IP basic calls in Unified CMBE with 324 BHCA are successful for one day load run.	SCCP Phone1->Unified CMBE->MGCP g/w->PSTN->MGCP g/w-> Unified CMBE->SCCP Phone2	Passed	
UC802EL.PER.0050	Reliability	IP to IP RSVP WAN calls	Verify that RSVP WAN calls of 4788 BHCA are successful for 24 hrs.	REM SCCP Phone1->Unified CM->RSVP Agent 1 (Remote) -> RSVP Agent 2(Central site)->Unified CM-> SCCP Phone2;SCCP Phone1->Unified CM->RSVP Agent 2(Central site)-> RSVP Agent 1(Remote)->Unified CM-> REM SCCP Phoneone2	Passed	
UC802EL.PER.0051	Reliability	IP to IP RSVP WAN calls		REM SCCP Phone1->Unified CM->RSVP Agent 1 (Remote) -> RSVP Agent 2(Central site)->Unified CM-> SCCP Phone2;SCCP Phone1->Unified CM->RSVP Agent 2(Central site)-> RSVP Agent 1(Remote)->Unified CM-> REM SCCP Phone2	Passed	
UC802EL.PER.0052	Reliability	Remote IP to Remote IP Over PSTN Calls	Verify that remote IP to remote IP over PSTN calls of 8460 BHCA are successful for 24 hrs.	REM SCCP Phone 1->Unified CM->MGCP/H.323/SIP g/w ->PSTN->MGCP/H.323/SIP g/w->Unified CM->REM SCCP Phone2;REM SIP Phone1->Unified CM->MGCP/H.323/SIP g/w->PSTN->MGCP/H.323/SIP g/w->Unified CM->REM SIP Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EL.PER.0053	Reliability	Remote IP to Remote IP Over PSTN Calls	Verify that remote IP to remote IP over PSTN calls of 8460 BHCA are successful for 72 hrs.	REM SCCP Phone 1->Unified CM->MGCP/H.323/SIP g/w ->PSTN->MGCP/H.323/SIP g/w->Unified CM->REM SCCP Phone2;REM SIP Phone1->Unified CM->MGCP/H.323/SIP g/w->PSTN->MGCP/H.323/SIP g/w->Unified CM->REM SIP Phone2	Passed	
UC802EL.PER.006	Reliability	IP to IP Calls Over Remote PSTN in Unified CMBE for Three Day Load Run	Verify that IP to PSTN-to IP basic calls in Unified CMBE with 324 BHCA are successful for three day load run.	SCCP Phone1->Unified CMBE->MGCP g/w->PSTN->MGCP g/w-> Unified CMBE->SCCP Phone2	Passed	
UC802EL.PER.007	Reliability	RSVP Calls in Unified CMBE for One Day Load Run	Verify that IP to IP calls in Unified CMBE (RSVP configured between Unified CMBE & Remotes) with 360 BHCA are successful for one day load. run	SCCP Phone1->Unified CMBE->SCCP Phone2	Passed	
UC802EL.PER.008	Reliability	RSVP Calls in Unified CMBE for Three Day Load Run	Verify that IP to IP calls in Unified CMBE (RSVP configured between Unified CMBE & Remotes) with 360 BHCA are successful for one day load run.	SCCP Phone1->Unified CMBE->SCCP Phoneone2	Passed	
UC802EL.PER.009	Reliability	Calls to Unity Connection Voicemail for One Day Load Run	Verify that Unity Connection calls in Unified CMBE of 300 BHCA are successful for one day load run.	SCCP Phone1->Unified CM1->SCCPPhone2->CFNA->VOICE MAIL; SCCPPhone2->VOICE MAIL->RETRIEVAL	Passed	
UC802EL.PER.010	Reliability	Calls to Unity Connection Voicemail for Three Day Load Run	Verify that Unity Connection calls in Unified CMBE of 300 BHCA are successful for three day load run.	SCCP Phone1->Unified CM1->SCCPPhone2->CFNA->VOICE MAIL; SCCPPhone2->VOICE MAIL->RETRIEVAL	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EL.PER.023	Reliability	Basic IP to IP Intra Cluster Calls for One Day Load Run	Verify that IP to IP calls in a medium size cluster of 1740 BHCA are successful.	SCCPPhone1->Unified CM->SCCP Phone2	Passed	
UC802EL.PER.024	Reliability	IP to IP Calls Over PSTN for One Day Load Run	Verify that IP to PSTN-to IP calls in a medium size cluster of 5700 BHCA are successful.	SCCP Phone1->Unified CM->MGCP g/w->Unified CM->SCCP Phone2; SIP Phone1->Unified CM->MGCP g/w->Unified CM->SIP Phone2	Passed	
UC802EL.PER.025	Reliability	IP to IP Calls Over Inter-Cluster Trunk for One Day	Verify that IP to IP inter cluster calls with supplementary services (hold, bxfer, cxfer) in a medium size cluster 720 BHCA are successful.	SCCP Phone1->Unified CM->ICT->Unified CM->SCCP Phone2	Passed	
UC802EL.PER.026	Reliability	IP to IP Calls Between Unified CME and Unified CM Over Inter-Cluster Trunk for One Day	Verify that IP to IP calls between Unified CME to Unified CM of 1800 BHCA are successful.	SCCP Phone1->Unified CM->GateKeeper1->IPIPGW->GateKeeper1->GateKeeper2->CME->SCCP Phone2	Passed	
UC802EL.PER.027	Reliability	Calls to Unity Connection Voicemail for One Day	Verify that Unity Connection calls in a medium size cluster of 600 BHCA are successful.	Stage1: SCCP Phone1->Unified CM1->SCCPPhone2->CFNA->VOICE MAIL Stage2: SCCPPhone2->VOICE MAIL->RETRIEVAL	Passed	
UC802EL.PER.028	Reliability	IP to IP ICT Call Going Through Tandem Cluster for One Day	Verify that IP to IP inter cluster calls with supplementary services (hold, bxfer, cxfer) involving third cluster of 720 BHCA are successful.	SCCP Phone1->Unified CM1->ICT 1->Unified CM2->ICT 2->Unified CM3->SCCP Phone2	Passed	
UC802EL.PER.029	Reliability	IP to IP ICT Call Checking Interoperability with Previous Version of Unified CM (6.1.3) for One Day	Verify that IP to IP inter cluster calls with supplementary services (hold, bxfer, cxfer) in a medium size cluster 720 BHCA are successful.	SCCP Phone1->Unified CM->ICT 1->Unified CM->SCCP Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EL.PER.030	Reliability	Basic IP to IP Intra Cluster Calls for Three Days Load Run	Verify that IP to IP calls in a medium size cluster of 1740 BHCA are successful.	SCCPPhone1->Unified CM->SCCP Phone2	Passed	
UC802EL.PER.031	Reliability	IP to IP Calls Over PSTN for Three Days	Verify that IP to PSTN-to IP calls in a medium size cluster of 5700 BHCA are successful.	SCCP Phone1->Unified CM->MGCP g/w->Unified CM->SCCP Phone2; SIP Phone1->Unified CM->MGCP g/w->Unified CM->SIP Phone2	Passed	
UC802EL.PER.032	Reliability	IP to IP Calls Over Inter-Cluster Trunk for Three Days	Verify that IP to IP inter cluster calls with supplementary services (hold, bxfer, cxfer) in a medium size cluster 720 BHCA are successful.	SCCP Phone1->Unified CM->ICT->Unified CM->SCCP Phone2	Passed	
UC802EL.PER.033	Reliability	IP to IP calls between Unified CME and Unified CM over Inter-Cluster Trunk for three day	Verify that IP to IP calls between Unified CME to Unified CM of 1800 BHCA are successful.	SCCP Phone1->Unified CM->GateKeeper1->IPIPGW->GateKeeper1->GateKeeper2->CME->SCCP Phone2	Passed	
UC802EL.PER.034	Reliability	Calls to Unity Connection Voicemail for Three Days	Verify that Unity Connection calls in a medium size cluster of 600 BHCA are successful.	Stage1: SCCP Phone1->Unified CM1->SCCPPhone2->CFNA->VOICE MAIL Stage2: SCCPPhone2->VOICE MAIL->RETRIEVAL	Passed	
UC802EL.PER.035	Reliability	IP to IP ICT Call Going Through Tandem Cluster for Three Day	Verify that IP to IP inter cluster calls with supplementary services (hold, bxfer, cxfer) involving third cluster of 720 BHCA are successful.	SCCP Phone1->Unified CM1->ICT 1->Unified CM2->ICT 2->Unified CM3->SCCP Phone2	Passed	
UC802EL.PER.036	Reliability	IP to IP ICT Call Checking Interoperability with Previous Version of Unified CM (6.1.3) for Three Days	Verify that IP to IP inter cluster calls with supplementary services (hold, bxfer, cxfer) in a medium size cluster 720 BHCA are successful.	SCCP Phone1->Unified CM->ICT 1->Unified CM->SCCP Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802EL.PER.043	Reliability	IP to IP Calls Over Inter Cluster Trunk	Verify that IP to IP Inter Cluster Trunk (ICT) calls with supplementary services (hold, bxfer, cxfer) in a large size cluster 2520 BHCA are successful.	SCCP Phone1->Unified CM->ICT Trunk1 (QSIG ICT)->Unified CM->SCCP Phone2	Passed	
UC802IA.ISAC.501	Codec Protocols	iSAC Codec: Verify Calls Between Two Endpoints	Verify if phone A can call Phone B and also if Phone B can call Phone A.		Passed	
UC802IA.ISAC.502	Codec Protocols	iSAC Codec: Verify Hold and Resume	Verify if during a call between Phone A and B, can hold/resume either Phone A or B three times.		Passed	
UC802IA.ISAC.503	Codec Protocols	iSAC Codec: Verify Calls Between Two Endpoints	Verify if phone A can call Phone B and also if Phone B can call Phone A.		Passed	
UC802IA.ISAC.601	Codec Protocols	iSAC Codec: Verify Multiple Calls on a single Line	Verify if EpAsstnt1 (assistant endpoint 1) makes first call to epFeature (feature applied endpoint) and EpAsstnt2 (assistant endpoint 2) makes second call to epFeature on the same line. EpFeature can switch the first and second call correctly.		Passed	
UC802IA.ISAC.602	Codec Protocols	iSAC Codec: Verify Multiple Calls on Multiple Lines on Same Phone	Verify if EpAsstnt1 (assistant endpoint 1) makes first call to epFeature (feature applied endpoint) and EpAsstnt2 (assistant endpoint 2) makes second call to epFeature on a different line. EpFeature can switch the first and second calls correctly.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IA.ISAC.603	Codec Protocols	iSAC Codec: Verify Call Forward All	Verify if epAsstnt1 (assistant endpoint 1) calls epFeature (feature applied endpoint) and epFeature forwards all call to epAsstnt2 (assistant endpoint 2).		Passed	
UC802IA.ISAC.604	Codec Protocols	iSAC codec: Verify Call Forward Busy	Verify if epAsstnt1 (assistant endpoint 1) calls epFeature (feature applied endpoint). when epFeature is busy the call is forwarded to epAsstnt2 (assistant endpoint 2).		Passed	
UC802IA.ISAC.605	Codec Protocols	iSAC Codec: Verify Call Forward No Answer	Verify if epAsstnt1 (assistant endpoint 1) calls epFeature (feature applied endpoint), epFeature is ringing but there is no answer, then the call is forwarded to epAsstnt2 (assistant endpoint 2).		Passed	
UC802IA.ISAC.606	Codec Protocols	iSAC Codec: Verify Same Group <Normal> Pickup by Pickup Softkey	Verify if a user can setup epFeature (feature applied endpoint) and epAsstnt2 (assistant endpoint 2) in the same pickup group and Unified CM service parameter <Auto Call Pickup Enabled> as false. EpAsstnt1 (assistant endpoint 1) calls epAsstnt2. During epAsstnt2 ringing EpFeature push PickUp softkey. After the ringing EpFeature goes off-hook to connect the call.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IA.ISAC.607	Codec Protocols	iSAC Codec: Verify <Normal> Group Pick up by GPickUp softkey	Verify if a user can setup epFeature (feature applied endpoint) and epAsstnt2 (assistant endpoint 2) in the different pickup group and <Auto Call Pickup Enabled> as false in Unified CM. EpAsstnt1 (assistant endpoint 1) calls epAsstnt2. During epAsstnt2 ringing. EpFeature push GPickUp softkey and dials the pick code. After ringing epFeature goes off hook and connect the call.		Passed	
UC802IA.ISAC.608	Codec Protocols	iSAC Codec: Verify <Normal> Other Group Pick Up by OPickUp softkey	Verify is a user can setup epAsstnt2 (assistant endpoint 2) in an associate pickup group with epFeature (feature applied endpoint) and Unified CM service parameter <Auto Call Pickup Enabled> as false. EpAsstnt1 (assistant endpoint 1) calls epAsstnt2. During epAsstnt2 ringing EpFeature pushes OPickUp softkey. After the ringing EpFeature goes off-hook to connect the call.		Passed	
UC802IA.SRST.501	Unified SRST	Unified SRST: Verify Calls Between Two Endpoints	Verify if Phone A can call Phone B and also if Phone B can call Phone A when either or both are in Unified SRST mode.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IA.SRST.502	Unified SRST	Unified SRST: Verify Hold and Resume When Phone is in SRST Mode	Verify if during a call between Phone A and Phone B (either or both are in SRST mode), hold/resume either Phone A or B three times.		Passed	
UC802IA.SRST.503	Unified SRST	Unified SRST: Verify Hold and Resume When Phone is in SRST Mode	Verify if during a call between Phone A and Phone B (either or both are in SRST mode), hold/resume either Phone A or B three times.		Passed	
UC802IA.SRST.601	Unified SRST	Unified SRST: Verify Multiple Calls on Multiple Lines of a Phone in SRST mode	Verify if EpAsstnt1 (assistant endpoint 1) makes first call to epFeature (feature applied endpoint). EpAsstnt2 (assistant endpoint 2) makes second call to epFeature on a different line. EpFeature can switch the first and second calls correctly.		Passed	
UC802IA.SRST.602	Unified SRST	Unified SRST: Verify Multiple Calls on Multiple Lines of a phone in SRST mode	Verify if EpAsstnt1 (assistant endpoint 1) makes first call to epFeature (feature applied endpoint), EpAsstnt2 (assistant endpoint 2) makes second call to epFeature on a different line and if EpFeature can switch the first and second calls correctly.		Passed	
UC802IA.SRST.603	Unified SRST	Unified SRST: Verify Call Forward All by Softkey When Phone is SRST mode	Verify when epFeature (feature applied endpoint) is the phone pushing the softkey; epAsstnt1 (assistant endpoint 1) is the phone making calls; epAsstnt2 (assistant endpoint 2) is the target of call forward to.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IA.SRST.604	Unified SRST	Unified SRST: Verify Call Forward Busy When Phone is SRST Mode	Verify when epAsstnt1 (assistant endpoint 1) calls epFeature (feature applied endpoint) and when epFeature is busy the call is forwarded to epAsstnt2 (assistant endpoint 2).		Passed	
UC802IA.SRST.605	Unified SRST	Unified SRST: Verify Call Forward No Answer When Phone is SRST mode	Verify when epAsstnt1 (assistant endpoint 1) calls epFeature (feature applied endpoint), and if epFeature is ringing but no answer, then the call is forwarded to epAsstnt2 (assistant endpoint 2).		Passed	
UC802IA.SRST.606	Unified SRST	Unified SRST: Verify Call Forward All by Softkey When Phone is SRST mode	Verify when epFeature (feature applied endpoint) is the phone pushing the softkey; epAsstnt1 (assistant endpoint 1) is the phone making calls; epAsstnt2 (assistant endpoint 2) is the target of call forward to.		Passed	
UC802IA.SRST.607	Unified SRST	Unified SRST: Verify Call Forward Busy When Phone is SRST mode	Verify when epAsstnt1 (assistant endpoint 1) calls epFeature (feature applied endpoint), when epFeature is busy the call is forwarded to epAsstnt2 (assistant endpoint 2).		Passed	
UC802IA.SRST.608	Unified SRST	Unified SRST: Verify Call Forward No Answer When Phone is SRST mode	Verify when epAsstnt1 (assistant endpoint 1) calls epFeature (feature applied endpoint), epFeature is ringing but no answer, then the call is forwarded to epAsstnt2 (assistant endpoint 2).		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CCM.021	Unified CM	iPhone Client Joining Unified MeetingPlace Internal Meeting by Receiving VOIP Call and Handoff Call to Deskphone	To verify that iPhone client can join a Unified MeetingPlace hosted meeting by answering call back. When meeting in progress iPhone handoff the call to deskphone.	iPhone end point (Unified CM(Unified MeetingPlace	Passed	
UC802IF.CCM.109	SCCP Adhoc Conference	Adhoc Conference Using Unified MeetingPlace via H.225 Trunk	Verify if video conference can be established between 3 parties with H.225 trunk between Unified CM and Unified CME.	7985 -CME--- H225 ---7985 -- Unified CM --- Conference 7985 ---h225 --- Unified CM	Passed	
UC802IF.CCM.170	Unified CM	E911 Call Handling When Users Logs in Across Cluster Using EMCC Feature	To verify if the E911 call from users logged in using EMCC feature in visiting Unified CM cluster is routed to local PSAP.	IP Phone->ASA->Unified CM->Emergency Responder->PSAP	Passed	
UC802IF.CCM.200	Unified CM	iPhone Client Joins MeetMe Conference with Handoff Call to GSM	Verify if iPhone client is joining a conference by dialing meetme number and while in conference going out of Wifi coverage, continue the call in GSM.	iPhone end point (Unified CM	Passed	
UC802IF.CCM.203	Unified CM	iPhone Client Receiving a Cisco IME Call and Setting Up Adhoc Conference	To verify that iPhone client can answer an incoming IME call and then setup an adhoc conference.	iPhone end point (Unified CM	Passed	
UC802IF.CCM.204	Unified CM	iPhone Client Setting Up an IME Call and Transferring the Call to PSTN	To verify that iPhone client can set up an inter enterprise IME call and then transfer the call to a PSTN destination through H323 gateway.	iPhone end point (Unified CM(H323GW	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CCM.205	Unified CM	Visual VM Indication at iPhone Client, Downloading and Playing VM Using Unity Connection	To verify that iPhone client can receive Visual VM indication and can dial into Unity Connection VM server and check VM, and the DTMF is set for kpml.	iPhone end point (Unified CM(Unity Connection	Passed	
UC802IF.CCM.206	Unified CM	Visual VM Indication at iPhone client and Downloading and Playing VM Messages When Unity Connection VM Servers Active-Active	To verify iPhone clients get VM alerts and see visual VM indication and it can download and play the VM messages.	iPhone end point (Unified CM(Unity Connection	Passed	
UC802IF.CCM.207	Unified CM	iPhone Client Downloading and Playing VPIM Forwarded and Replied VM Messages When VM Server is Cisco Unity Connection	To verify iPhone clients get VM alerts for VPIM forwarded and replied voicemails and see visual VM indication and it can download and play these VM messages.	iPhone end point (Unified CM(Unity Connection	Passed	
UC802IF.CCM.208	Unified CM	iPhone Client Receiving Visual VM Notification and Downloading and Playing of Forwarded VM Messages	To verify iPhone client gets voicemail alert and can see visual VM indication and it can download and play VM messages from Unity VM servers in active-standby configuration and when active is down.	iPhone client end point(Unified CM(Unity	Passed	
UC802IF.CCM.209	Unified CM	Layer 2/3 Roaming When iPhone Client is in a Call	To verify iPhone client in a call can roam from one AccessPoint to another seamlessly and maintain the call.	iPhone client endpoint (Unified CM(Wireless	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CCM.211	Unified CM	Call Pick Up for iPhone Client	To verify if iPhone client and Unified IP Phones 9971, 9951, and 8961 are in same call pickup group. Verify that Unified IP Phones 9971, 9951, and 8961 can pick the call for iPhone client and vice versa (iPhone Client can pick up the call for Unified IP Phones 9971, 9951, and 8961 if supported).	iPhone Client endpoint (Unified CM	Passed	
UC802IF.CCM.300	Unified CM	BLF Speed Dial for Unified IP Phone 9971 With Three CKEM Expansion Modules	To verify and monitor the presence status of 108 BLF-Speed dials for three Unified IP Phone 9971 CKEM expansion modules.		Passed	
UC802IF.CCM.301	Cisco Unity Connection	eMWI on Shared Lines With Unity Connection-Unified CM SIP Integration	Verify that MWI count is seen on Unified IP Phone (8900 and 9900) series when a new voicemail is left for the subscriber in Unity Connection integrated to Unified CM using SIP.	IP Phone->Unified CM->SIP->Unity Connection	Passed	
UC802IF.CCM.302	Cisco Unity Connection	eMWI on Shared Lines With Unity Connection-Unified CM SCCP Integration	Verify that MWI count is seen on Unified IP Phone (8900 and 9900) series when a new voicemail is left for the subscriber in Unity Connection integrated to Unified CM using SIP.	IP Phone->Unified CM->SIP->Unity Connection	Passed	
UC802IF.CCM.303	Cisco Unity Connection	eMWI for Extension Mobility Across Cluster	Verify that eMWI works for Extension Mobility across cluster.	Unity Connection->Unified CM->SIPT->Unified CM->IP Phone 9971/9951/8961 (Extension Mobility)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CCM.304	Cisco Unity Connection	Support of eMWI Over Inter Cluster SIP Trunks	Verify that eMWI works over inter cluster SIP trunks.	Unity Connection (MWI)->Unified CM->SIPT->Unified CM->IP Phone 9971/9951/8961	Passed	
UC802IF.CCM.305	Cisco Unity Connection	Support of eMWI During Failover and Bulk Re-Synchronization in Unity Connection	Verify that eMWI works when primary VM server is down and the secondary server is the acting primary.		Passed	
UC802IF.CCM.500.1	Unified CM	Unified CM CoW Deployment Model	To verify and validate the CoW Deployment Model.	Unified CM->ASA->WAN->ASA->Unified CM	Passed	
UC802IF.CCM.501	Unified CM	Validate ASA TLS Proxy Feature	To verify and validate non-secure, encrypted and authenticate phone registration to Unified CM via ASA TLS Proxy.	Phone->ASA TLS Proxy->Unified CM	Passed w/ Exception	CSCtc06130
UC802IF.CCM.520	Unified IP Phone	Unified IP Phone 6911 Joining Unified MeetingPlace Meeting	To verify that Unified IP Phone 6911 can join a Unified MeetingPlace meeting.	Phone->Unified CM -->Unified MeetingPlace	Passed	
UC802IF.CCM.521	Unified IP Phone	Unified Personal Communicator Working in Phone Associated Mode to IP Phone 6911	To verify that Unified Personal Communicator can work in phone associated mode to IP Phone 6911.	Phone->Unified CM -->Unified MeetingPlace	Passed	
UC802IF.CCM.522	Unified IP Phone	IP Phone 6911 Dials Emergency Number	To verify that Unified IP Phone 6911 can dial emergency number.	Phone->Unified CM -->Unified MeetingPlace	Passed	
UC802IF.CCM.523	Unified IP Phone	Support for Different Codecs Including ILBC Codec by IP Phone 6911	To verify that Unified IP Phone 6911 can support codecs, g711u, g729r8, g729br8, ILBC codecs.	Phone->Unified CM -->Unified MeetingPlace	Passed	
UC802IF.CCM.524	Unified IP Phone	IP Phone 6911 Working as Agent Phone in Unified CCX Network	To verify that Unified IP Phone 6911 can be used as an agent phone in Unified CCX.	Phone->Unified CM -->Unified CCX	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CCM.525	Unified IP Phone	IP Phone 6911 and Unified Video Advantage Interworking	To verify that Unified Ip Phone 6911 can interwork with Unified Video Advantage and use as a video phone.	Phone->Unified CM -->Unified Video Advantage	Passed	
UC802IF.CCM.526	Unified IP Phone	IP Phone 6901 Joining Unified MeetingPlace Meeting	To verify that Unified IP Phone 6901 can join Unified MeetingPlace scheduled meetings.	Phone->Unified CM -->Unified MeetingPlace	Passed	
UC802IF.CCM.527	Unified IP Phone	Support for Different Codecs Including ILBC Codec by IP Phone 6901	To verify that Unified IP Phone 6901 can support codecs, g711u, g729r8, g729br8, ILBC codecs.	Phone->Unified CM -->Unified MeetingPlace	Passed	
UC802IF.CCM.600	Unified CM Extension Mobility Cross Cluster	Extension Mobility Cross Cluster Attend Home Cluster MeetingPlace Conference, Primary Home Unified CM Failure	Verify user is able to log into a visiting cluster using Extension Mobility Cross Cluster and attend a meeting place conference hosted on the home cluster and the home cluster's primary Unified CM fails.	EMCC IP Phone->ASA->Unified CM->MeetingPlace	Passed	
UC802IF.CCM.601	Unified CM Extension Mobility Cross Cluster	Extension Mobility Cross Cluster Calls MGCP PSTN Phone Using Local Route Group, Signaling WAN Link Failure	Verify if user is able to log into a visiting cluster using Extension Mobility Cross Cluster and call an external number through an MGCP PSTN gateway configured using Local Route Group and there is a Signaling WAN link failure during the call.	EMCC IP Phone->ASA->Unified CM->MGCP PSTN->PSTN Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CCM.602	Unified CM Extension Mobility Cross Cluster	Extension Mobility Cross Cluster Ad-Hoc Video Conference With Second Incoming Call, Local WAN Failover to Unified SRST	Verify if user is able to log into a visiting cluster using Extension Mobility Cross Cluster and initiate a video ad-hoc conference on a Unified IP Phone 9900 series in the Home cluster, a Unified IP 7985 phone in a remote Unified CME site, and a Cisco UC Integration for Microsoft Office Communicator client in the visiting cluster. A second incoming call also occurs during the conference. Local WAN failure causes endpoints to failover to Unified SRST.	EMCC IP Phone->ASA->Unified CM->ConfBridge->ICT->CME->IP Phone7985; EMCC IP Phone->ASA->Unified CM->ConfBridge ->ICT->ASA->Unified CM->UC Integration™ for Microsoft Office Communicator;	Passed	
UC802IF.CCM.603	Unified CM Extension Mobility Cross Cluster	Extension Mobility Cross Cluster Video ICT Call, Gatekeeper Based CAC Reverts the Call to Audio-Only	Verify if user is able to log into a visiting cluster using Extension Mobility Cross Cluster and attempt to Pickup an incoming video call in the user's Pickup Group. Call is reverted to audio-only due to Gatekeeper Based CAC bandwidth restrictions.	EMCC IP Phone->ASA->Unified CM->GateKeeper ICT->CME->IP Phone7985;	Passed	
UC802IF.CCM.604	Unified Computing System	Unified Computing System Blade Failover and Recover	Verify the applications running on a Unified Computing System Blade are able to properly failover and recover after the blade has failed.	NA	Passed	
UC802IF.CCM.605	Unified CM	TCP Connection Reuse With Unified CM SIP Trunk	Verify that TCP connection reuse behavior does not cause any adverse effect and calls succeed over SIP trunks.	Phn1 -->Unified CM1 -->SIPT -->Unified CM2 -->Unified CM2 -->Phn2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CCM.606	Unified CM	SIP Trunks With CODEC G729/G729a and MTP Required	Verify that calls over SIP trunks with MTP required are successful when the codec preference on the SIP trunk is G.729/G.729a.	Phn1 -->Unified CM1 -->SIPT (MTP G729 G729a) -->Unified CM2 -->Unified CM2 -->Phn2	Passed	
UC802IF.CCM.607	Unified Computing System	Unified Computing System Blade Reboot	Verify the applications running on a Unified Computing System Blade are able to properly recover after the Unified Computing System blade is rebooted.	NA	Passed	
UC802IF.CCM.651	IPv6	Dual Stack SIP Trunk With Trunk Codec Set to G729	Verify that MTP is invoked when the trunk has been configured for early offer. And to verify that calls works over DS SIP Trunks with codec preference set to G729/G729a.	SCCP (v6/v4) -->Unified CM -->SIPT (v6/v4) (v6 media/v4 sig pref) (ANAT on) (MTP) -->Unified CM -->IP Phone	Passed	
UC802IF.CCM.700	Unified CM	Clock in Various Components Fallback to Standard Time in Fall 2009	Verify that the clocks in various components fallback to Standard time in Fall 2009.		Passed	
UC802IF.CCM.701	Unified CM	Clock in Various Components Spring Forward to North American Daylight Savings Time	Verify if the Clocks in various components spring forward to North American Daylight Savings Time.		Passed	
UC802IF.CCM.702	Unified CM	Daylight Savings Time Regulation Using COP File	Verify support for any update in Daylight Savings Time Regulation using COP file.		Passed	
UC802IF.CCM.809	Video	Call Transfer Through Cisco IME Trunk	To verify if an audio call can escalate to video when call is transferred from audio capable endpoint to video capable endpoint.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CER.101	Cisco Emergency Responder	JTAPI Over WAN	Verify the validation of Cisco Emergency Responder in CoW deployment such as the Cisco Emergency Responder node registering CTI ports and route points to Unified CM over WAN.	Phone->Emergency Responder->ASA->WAN->ASA->Unified CM->Gateway->PSAP	Passed	
UC802IF.CER.102	Cisco Emergency Responder	Cisco Emergency Responder Database Replication	Verify to ensure Cisco Emergency Responder database replication reliability during network instability or Subscriber node going offline for few hours.	Phone->Emergency Responder->ASA->WAN->ASA->Unified CM->Gateway->PSAP	Passed	
UC802IF.CME.103	Cisco VTIII Support	Unified CME Interop With Cisco VTIII Camera	Verifies if Cisco VTIII camera behind Ip Communicator registered to Unified CME calls Cisco VTIII endpoint.	IP Communicator/Unified Video Advantage---- CME --- H323 --- Unified CM -- IP Communicator/Unified Video Advantage	Passed	
UC802IF.CME.901	Unified CME	Unified CME Stream Multicast MOH	Verify whether Unified CME can stream multicast MOH when SNR phone is put on hold.	IP Phone 6921/6941/6961 --- Unified CM ---- SIP T -----Unified Border Element ----- SIPT ---- CME --- SNR -Hold---- MMOH --- Unified CM IP Phone	Passed	
UC802IF.CME.902	Video Over SIP Trunk	Video Over SIP Trunk Using Endpoints	Verify the video over SIP trunk using Unified IP Phones 8900, 9900, and 6900 series.	IP Phone 6921/6941/6961 -CME 1--- SIP T - CME 2 ---IP Phone 9971/9951/8961 ----Conference ---Unified CM 1---IP Phone 3	Passed	
UC802IF.CME.903	Video	Video Call Resumed During Hold/Resume	Verify whether video call is resumed during hold/resume and transfer over SIP trunk.	IP Phone 6921/6941/6961 -CME 1--- SIP T - CME 2 ---IP Phone 9971/9951/8961 ----Transfer->Unified CM 1---IP Phone 3	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CRS.100	Unified Contact Center Express	High Availability over WAN Installation With Each Unified CCX Node Located in Different Time Zones	To Verify that Unified CCX can be installed successfully in high availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.CRS.100	Unified CCX	Unified Personal Communicator and VTIII	To Verify that Unified CCX can be installed successfully in High Availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.CRS.100	Unified CCX	Unified IP Phone 6911 and VTIII	To Verify that Unified CCX can be installed successfully in High Availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.CRS.100	Unified CCX	Unified IP Phone 6900 series and VTIII	To Verify that Unified CCX can be installed successfully in High Availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.CRS.100	Unified CCX	Unified IP Phone 7916 and VTIII	To Verify that Unified CCX can be installed successfully in High Availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.CRS.100	Unified CCX	IP Communicator and VTIII	To Verify that Unified CCX can be installed successfully in High Availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CRS.100	Unified CCX	Unified CME, Unified Video Advantage, and VTIII	To Verify that Unified CCX can be installed successfully in High Availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.CRS.100	Unified CCX	UC Integration™ for Microsoft Office Communicator and VTIII	To Verify that Unified CCX can be installed successfully in High Availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.CRS.100	Unified CCX	Unified Personal Communicator 8.0 and VTIII	To Verify that Unified CCX can be installed successfully in High Availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.CRS.100	Unified CCX	Unified CCX Installed Successfully in High Availability Over WAN	To Verify that Unified CCX can be installed successfully in High Availability over WAN environment with each Unified CCX located in different time zones.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.CRS.101	Unified Contact Center Express	Remote Agents Functionality Where an Agent and Primary Unified CCX Node are Separated by WAN	To verify that agents located in one campus location get service from Unified CCX node located in different campus locations separated by WAN.	Agent->ASA->Unified CM->ASA->WAN->ASA->Unified CM->Unified CCX	Passed	
UC802IF.CRS.102	Unified Contact Center Express	PSTN Call Gets Anchored on Voice Gateway in One Data Center With Primary Unified CCX in Another Data Center	To verify that Unified CM invokes transcoder to allow Unified CCX to play Unified IP IVR when incoming call uses non G.711 codec.	PSTN Caller->Voice Gateway->Unified CM->ASA->WAN (G.729)->ASA->Unified CM->Unified CCX->ASA->Agent	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CRS.103	Unified Contact Center Express	Unified CCX Active/Master Node and Primary/Active CTI Manager Separated by WAN	Verify that Unified CCX can be integrated with Unified CM separated by WAN.	PSTN Caller->Voice Gateway->Unified CM->ASA->WAN (G.729)->ASA->Unified CM->Unified CCX->ASA->Agent	Passed	
UC802IF.CRS.104	Unified Contact Center Express	System Behavior During Unified CCX Node Failover and CTI Manager Failover	To validate the failover behavior of Unified CCX.	1: PSTN Caller->Voice Gateway->Unified CM->ASA->WAN(G.729)->ASA->Unified CM->Unified CCX->ASA->Agent. 2: PSTN Caller->Voice Gateway->Unified CM->Unified CCX->ASA->Agent	Passed	
UC802IF.CRS.105.1	Unified Contact Center Express	Validate Island Mode Operation of Unified CCX	To verify and validate the island mode moderation of Unified CCX deployed in High Availability over WAN when Unified CCX node is active due to WAN link failure.	PSTN Caller->Voice Gateway->Unified CM->Unified CCX->ASA->Agent	Passed	
UC802IF.CRS.105.2	Unified Contact Center Express	Validate Island Mode Operation of Unified CCX	To validate island mode moderation of Unified CCX deployed in High Availability over WAN when Unified CCX node is active due to WAN link failure. (test case: When WAN link is flapping, to make sure Unified CCX Primary Unified CCX node remains in stable operating condition and there is no loss of call processing and the system remains in stable island mode).	PSTN Caller->Voice Gateway->Unified CM->Unified CCX->ASA->Agent	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CRS.105.3	Unified Contact Center Express	Validate Island Mode Operation of Unified CCX After WAN Link Recovery	To validate island mode moderation of Unified CCX deployed in High Availability over WAN and to ensure that the Unified CCX nodes and database is setup properly after a WAN link recovery.	PSTN Caller->Voice Gateway->Unified CM->Unified CCX->ASA->Agent	Passed	
UC802IF.CRS.105.4	Unified Contact Center Express	Island Mode Operation of Unified CCX When CAD and Agent Phones are Initially Register to Different Sides on Datacenters	To validate island mode moderation of Unified CCX deployed in High Availability over WAN when Unified CCX node is active due to WAN link failure.	PSTN Caller->Voice Gateway->Unified CM->Unified CCX->ASA->Agent	Passed	
UC802IF.CRS.106.1	Unified Contact Center Express	Validate Agents with Multi-line Unified IP Phones and Single Line Unified IP Phones	To verify that Unified IP Phone series (9971, 9951, and 8961) and Unified IP Phone (6900 series) models can be used as agents.	Caller->ASA->Unified CM->>Unified CCX->ASA->Agent	Passed	
UC802IF.CRS.106.2	Unified Contact Center Express	Validate Agents with Multi-line Unified IP Phones and Single Line Unified IP Phones	To verify that Unified IP Phone series (9971, 9951, and 8961) and Unified IP Phone (6900 series) models can be used as agents.	Caller->ASA->Unified CM->Unified CCX->ASA->Agent	Passed	
UC802IF.CRS.106.3	Unified Contact Center Express	Validate Agents with Multi-line Unified IP Phones and Single Line Unified IP Phones [Test case 3]	To verify that Unified IP Phone series (9971, 9951, and 8961) and Unified IP Phone (6900 series) models can be used as agents.	Caller->ASA->Unified CM->Unified CCX->ASA->Agent	Passed	
UC802IF.CRS.107.1	Unified Contact Center Express	JTAL And DTAL Feature Support on Unified IP Phones With 4 Lines	To verify that Unified CCX can monitor multiple lines of Unified IP Phone models (9971, 9951, and 8961).	Caller->ASA->Unified CM->Unified CCX->ASA->Agent -1 (JAL/DTAL)->Agent-2	Passed	
UC802IF.CRS.107.2	Unified Contact Center Express	JTAL And DTAL Feature Support on Unified IP Phone with 4 Lines	To verify if Unified CCX can monitor multiple lines of Unified IP Phone (9971, 9951, and 8961) models.	Caller->ASA->Unified CM->Unified CCX->ASA->Agent -1 (JAL/DTAL)->Agent-2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CSF.001	Cisco UC Integration™ for Microsoft Office Communicator	SD Video call between UC Integration™ for Microsoft Office Communicator and H323 and SIP 3rd party video end points	To verify standard definition video call can be established between UC Integration™ for Microsoft Office Communicator and H323 and SIP 3rd party end points.	UC Integration™ for Microsoft Office Communicator->Firewall ASA->Unified CM->Firewall ASA->Sony/Tandberg/ Video end points	Failed	CSCtf33615
UC802IF.CSF.002	Cisco UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator in Soft Phone Mode Setting Up Adhoc Video Conference With Cisco Endpoints	To verify if adhoc video conference can be established using Unified MeetingPlace video conference bridge resource from UC Integration for MOC inviting Unified IP Phone models (9971, 9951, and 8961), Unified IP Phone 7985, Unified Personal Communicator, and SIP and H.323 3rd party video end points.	UC Integration™ for Microsoft Office Communicator->Firewall ASA->Unified CM->Firewall ASA->Unified MeetingPlace Video conference resouce+Sony+Tandberg Video end points+RT+7985	Passed	
UC802IF.CSF.003	Cisco UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator in Desk Phone Mode Setting Up Adhoc HD Video Conference	To verify if adhoc HD video conference can be established using Unified MeetingPlace Video conference bridge resource from Cisco UC Integration for Microsoft Office Communicator in desk phone mode associated to a multi-line Unified IP Phone (9971, 9951, and 8961) model inviting SIP intercluster Unified IP Phone (9971, 9951, and 8961) model and UC Integration for Microsoft Office Communicator clients.	UC Integration™ for Microsoft Office Communicator->Firewall ASA->Unified CM->Firewall ASA->Unified MeetingPlace Video conference resouce+Sony+Tandberg Video end points+RT+7985	Failed	CSCtf65671

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CSF.004	Cisco UC Integration™ for Microsoft Office Communicator	Cisco UC Integration™ for Microsoft Office Communicator in Desk Phone Mode	To verify Cisco UC Integration for Microsoft Office Communicator in remote Clustering over WAN (CoW) site can receive an intercluster SIP HD video call.	UC Integration™ for Microsoft Office Communicator1->Firewall ASA->Unified CM1->firewall ASA---<SIP>-(Firewall ASA->Unified CM2--(UC Integration™ for Microsoft Office Communicator2	Passed	
UC802IF.CSF.005	Cisco UC Integration™ for Microsoft Office Communicator	Cisco UC Integration™ for Microsoft Office Communicator in Unified SRST Mode	To verify if Cisco UC Integration for Microsoft Office Communicator in a remote site registered to a Unified SRST Gateway can retrieve voicemails from the Mainsite.	UC Integration™ for Microsoft Office Communicator->SRST	Passed w/ Exception	CSCte64265
UC802IF.CSF.009	Client Services Framework	Client Services Framework Joining Unified MeetingPlace Scheduled Video Conference	To verify Client Services Framework client can join a scheduled Unified MeetingPlace video conference.	CSF->Firewall ASA->Unified CM->FIREWALL ASA->Cisco IME	Passed w/ Exception	CSCte70278
UC802IF.CSF.010	Cisco UC Integration™ for Microsoft Office Communicator	Cisco UC Integration™ for Microsoft Office Communicator in Desk Phone Mode	To verify Cisco UC Integration for Microsoft Office Communicator can associate to an EM enabled Unified IP phone .	UC Integration™ for Microsoft Office Communicator->Firewall ASA->Unified CM	Passed	
UC802IF.CSF.012	Cisco UC Integration™ for Microsoft Office Communicator	Incoming Cisco IME Video Call Escalation	To verify Cisco UC Integration for Microsoft Office Communicator or CSF can carry out mid call escalation of incoming Cisco IME video calls to HD video and establish a HD Video conference inviting parties across Cisco IME trunk.	UC Integration™ for Microsoft Office Communicator1->Firewall ASA->Unified CM1->Firewall ASA->Cisco IME->Unified CM2-Firewall ASA->UC Integration™ for Microsoft Office Communicator2	Failed	CSCtf65671

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CSF.013	Cisco UC Integration™ for Microsoft Office Communicator	CSF Client Receiving And Transferring DVO Calls Through Cisco IME Trunk	To verify CSF client can receive DVO call from Unified Mobile Communicator client and transfer the call to another enterprise destination through Cisco IME trunk.	Mobile Cllient->Unified CM of Enterprise 1->CSF Client--XFER operation--Cisco IME Trunk->Unified CM of Enterprise 2->CSF client2	Passed	
UC802IF.CSF.014	Cisco UC Integration™ for Microsoft Office Communicator	HD/SD Video Call Between CSF and SCCP Video Endpoint	To verify CSF client can successfully establish a SD/HD video call to a SCCP video end point.	CSF client->Unified CM of Enterprise1->Cisco IME Trunk->Unified CM of Enterprise2-->SCCP Video endpoint	Passed	
UC802IF.CSF.015	Cisco UC Integration™ for Microsoft Office Communicator	Cisco UC Integration™ for Microsoft Office Communicator/CSF Call Processing Nodes Unavailable	To verify that Cisco UC Integration for Microsoft Office Communicator or CSF can perform normally when the active Unified CM server become unavailable, or the AD server become unavailable, or the OCS server become unavailable.	UC Integration™ for Microsoft Office Communicator->Firewall ASA->Unified CM	Failed	CSCtf57532
UC802IF.CSF.016	Cisco UC Integration™ for Microsoft Office Communicator	Cisco UC Integration™ for Microsoft Office Communicator/CSF in Desk Phone Receiving Cisco IME Call	To verify that Cisco UC Integration for Microsoft Office Communicator in desk phone mode to a Unified IP Phone 9971 can successfully terminate a HD video call from a video end point in a different enterprise through Cisco IME trunk.	UC Integration™ for Microsoft Office Communicator->SRST(Firewall ASA->Unified CM	Passed	
UC802IF.CSF.020	Unified Contact Center Express	Connection Monitor Timer Interaction With Cisco UC Integration™ for Microsoft Office Communicator Registered in Unified SRST	To verify that Cisco UC Integration for Microsoft Office Communicator in Unified SRST with flapping WAN link remains registered in Unified SRST mode.	UC Integration™ for Microsoft Office Communicator--→SRST----<WAN Link>---Unified CM	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CUB.807	Unified Border Element	SIP to H323 Interoperability Through Unified Border Element on ASR	To verify if a user can call a Unified CME Phone from Headquarter Phone through Unified Border element running on ASR platform and can also conference another Unified CM Phone through SIP Trunk.	IP Phone ---Unified CM (1)--- SIP T -Unified SIP Proxy --- SIPT- Unified Border Element --- H323 ---- IP Phone ---conference -SIP T--- IP Phone 3 --- Unified CM (1)	Passed	
UC802IF.CUBE.808	SIP Supplementary Services	SIP to SIP Audio Supplementary Services Through Unified Border Element on ASR	Verify if an audio call can be transferred through Unified Border Element to another Unified CM video phone and if the call can be established.	IP Phone --- SIP T -Unified SIP Proxy --- SIPT- Unified Border Element -SIP T---- IP Phone--- Unified CM 2 -Xfer --- IP Phone 3 --- Unified CM 2	Passed	
UC802IF.CUBE.809	Unified SRST	Failover Between Cisco ISR Unified Border Element and ASR 1006 Unified Border Element	Verifies the fail over between ISR Unified Border Element and ASR 1006 . Unified CM SRV record is configured with ISR and ASR Unified Border Element IP addresses.	Unified CM--SIP Trunk --- (ISR/ASR Unified Border Element) --- SIP Trunk --- CME	Passed	
UC802IF.CUCM.801	Voice Activated Conference	Voice Activated Conference Using Adhoc Software	Verify if a user can conference IP Phones listed under different Unified CM clusters, through Unified MeetingPlace software conference bridge.	IP Phone 1->Unified CM 1->SIP Trunk->Unified CM2 -IP Phone 2 ----Conference->Unified CM 1--- IP Phone 3	Passed	
UC802IF.CUCM.802	Video	Group Pick Up Video Call	To verify if a user can make a call from a <variable > to group pick number, take the call from video phone, and can establish two way video or audio.		Passed	
UC802IF.CUCM.807	Video	Video Over Cisco IME Trunk	To verify if a video call can be placed over PSTN and subsequent calls can go over Cisco IME trunk.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CUCM.808	Video	Video Call Failover to PSTN When WAN Link is Clogged	To verify whether Cisco IME call failover to PSTN when WAN is clogged between Unified CM Clusters and if the video call falls back to audio.		Passed	
UC802IF.CUE.100	Cisco Unity Express Visual Voicemail	Cisco Unity Express Subscriber Listens to Message via VoiceView Express	Verifies whether the user can listen to voicemail messages via VoiceView xpress feature of Cisco Unity Express.	CUE Subscriber --->VoiceView Express --->Message	Passed	
UC802IF.CUE.101	Cisco Unity Express Visual Voicemail	Cisco Unity Express Subscriber Sends Message via VoiceView Express	Verify whether Cisco Unity Express subscriber can send message to another subscriber via voice view express.	CUE Subscriber --->VoiceView Express --->Message	Passed	
UC802IF.CUE.102	Cisco Unity Express Visual Voicemail	Cisco Unity Express Subscriber Sends Broadcast Message	Verify whether Cisco Unity Express subscriber can send broadcast message via VoiceView Express.	CUE Subscriber --->VoiceView Express --->Broad cast Message	Passed	
UC802IF.CUE.103	Cisco Unity Express Visual Voicemail	Forward Fax Message to Blind Address Over VPIM With Recorded Greeting	Verifies whether voicemail express view shows fax attachment and play recorded greeting.	CUE Subscriber --->VoiceView Express --->Fax Message	Passed	
UC802IF.CUE.751	Cisco Unity Express-SRSV	SRSV-Unified Messaging Gateway Interaction With Active-Active Cisco Unity Connection	Verify that Survivable Remote Site Voicemail feature supports Cisco Unity Connection in Active-Active deployment.		Failed	CSCtf54209
UC802IF.CUE.752	Cisco Unity Express-SRSV	Using Multiple SRSV Unified Messaging Gateway to Support Multiple SRSV-Cisco Unity Express	Verify that multiple SRSV-Unified Messaging Gateway can be used with one Unified CM cluster to provision multiple SRSV-Cisco Unity Express locations where each SRSV-UMG is assigned with a subset of available SRSV-Cisco Unity Express.	IP Phone->SRST->SIP->Cisco Unity Express; IP Phone->Unified CM->SCCP/SIP->Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CUE.753	Cisco Unity Express-SRSV	SRSV Extension List for Users in Different Partition and CSS	To verify and validate that CSS and partition settings does not impact SRSV functionality and the configured restrictions in Cisco Unity Connection do not get ported in SRSV.	IP Phone->SRST->SIP->Cisco Unity Express; IP Phone->Unified CM->SCCP/SIP->Connection	Passed	
UC802IF.CUE.754	Cisco Unity Express-SRSV	Synchronization When Secondary Cisco Unity Connection Server Acts as Primary After WAN Link Restoration	Verify when Cisco Unity Connection is deployed in Active-Active mode, SRSV-Unified Messaging Gateway is able to interact with voicemail servers when they are in both primary and secondary mode.	IP Phone->SRST->SIP->Cisco Unity Express; IP Phone->Unified CM->SCCP/SIP->Connection1	Passed	
UC802IF.CUE.755	Cisco Unity Express-SRSV	Voicemail Upload Order After WAN Link Restoration From Cisco Unity Express to Cisco Unity Connection	Verify that voicemail order and state is maintained when voicemails are uploaded to Cisco Unity Connection.		Passed	
UC802IF.CUE.756	Cisco Unity Express-SRSV	Provisioning and Functioning for Unified SRST with SIP Phones	Verify that SRSV feature can operate seamlessly in SIP Unified SRST deployment.	SIP Phone->SRST->SIP->Cisco Unity Express; SIP Phone->Unified CM->Connection	Passed w/ Exception	CSCta76151, CSCta78369
UC802IF.CUE.757	Cisco Unity Express-SRSV	SRSV-Unified Messaging Gateway and SRSV-Cisco Unity Express Monitoring Using Unified Operations Manager	Verify that SRSV-UMG and SRSV-CUE can be monitored using Unified Operations Manager		Passed	
UC802IF.CUE.758	Cisco Unity Express-SRSV	Voicemail Deposit, Retrieval, Forwarding and MWI Status When in Fallback Mode	Verify that voicemail features are available when in fallback mode and MWI status is updated reflecting the current state.	IP Phone->SRST->SIP->Cisco Unity Express; IP Phone->Unified CM->SCCP/SIP->Connection1	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CUE.759	Cisco Unity Express-SRSV	SRSV-Unified Messaging Gateway Support for Multiple Unified SRST Locations	Verify that one SRSV-Unified Messaging Gateway can be used to support two Unified SRST-Cisco Unity Express deployment.	IP Phone->SRST->SIP->Cisco Unity Express	Passed	
UC802IF.CUE.761	Cisco Unity Express-SRSV	Alternate Greeting With End Date Reverting Back to Standard Greeting	Verify that alternate greeting is in effect after the fallback and SRSV-Cisco Unity Express switches automatically to standard greeting at the configured end date and time (as configured in Cisco Unity Connection).	IP Phone->SRST->SIP->Cisco Unity Express	Passed	
UC802IF.CUE.762	Cisco Unity Express-SRSV	Custom Holiday Schedule and Holiday Greeting	Verify that Holiday Greeting is played by SRSV-Cisco Unity Express when in fallback mode.	IP Phone->SRST->SIP->Cisco Unity Express	Passed	
UC802IF.CUE.763	Cisco Unity Express-SRSV	Cisco Unity Connection Call Action Set to "Hang Up"	Verify if SRSV-Cisco Unity Express hangs up after the greeting is played.	IP Phone->SRST->SIP->Cisco Unity Express	Passed	
UC802IF.CUE.765	Cisco Unity Express-SRSV	PSTN Caller Leaving a Voicemail in SRSV-Cisco Unity Express	Verify that PSTN callers can leave a voicemail in SRSV-Cisco Unity Express without any additional configuration.	PSTN Phone->SRST->SIP->SRSV-Cisco Unity Express	Passed	
UC802IF.CUE.766	Cisco Unity Express-SRSV	Addressing Messages to Distribution List Containing Members from Digitally Networked Cisco Unity Connection	Verify that messages can be addressed to distribution lists from SRSV-Cisco Unity Express and they are successfully delivered on WAN link restoration.	IP Phone->SRST->SIP->Cisco Unity Express; IP Phone->Unified CM->SCCP/SIP->Connection	Passed	CSCtf34050
UC802IF.CUE.767	Cisco Unity Express-SRSV	Addressing Messages to Subscriber From Auto Attendant	Verify that calls can be routed to Auto Attendant in SRSV-Cisco Unity Express and caller can lookup a subscriber by dialing the extension.	PSTN Phone->SRST->SIP->SRSV-CUE	Failed	CSCtf01694

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CUE.806	Unified SRST	Accessing DN Voicemail on Cisco Unity Express	To verify if a phone configured with + prefix can access voicemail.	IP Phone --- WAN Link --- Unified CM --- Fail WAN Link ---- IP Phone --- SRST ----Cisco Unity Express	Passed	
UC802IF.CUM.001	Unified Mobility	Transfer of Calls Between Proxy Deskphone, iPhone And Deskphone	To verify if a user can start a meeting from his phone proxy IP Phone, transfer the call to his iPhone using Cisco Unified Mobility feature while on commute and then again transfer the call to his desk Unified IP Phone by dialing *74 Move feature.	IPPhone->ASA Phone Proxy->FireWall ASA->Unified CM->Unified Mobility->DMZ ASA->iPhone	Passed	
UC802IF.CUM.002	Unified Mobility	Mobile Call From Proxy IP Phone Gets transferred to Another Desk IP Phone	To verify if user A can call user B on his mobile phone and during conversation User A can put the call on hold while user B reaches office and transfers the call to his desk Unified IP Phone by dialing *74 Move feature and then resume the call.	IPPhone->ASA Phone Proxy->Firewall ASA->Unified CM->Unified Mobility->DMZ ASA->Mobile Phone	Passed	
UC802IF.CUM.003	Unified Mobility	An iPhone User in Meeting Transfers Call to Desk IP Phone	Verify if a user can receive and attend a Unified MeetingPlace meeting on his iPhone and later transfer the call to his desk IP Phone by dialing *74.	iPhone->DMZ ASA->Firewall ASA->Unified Mobility->Unified CM->Firewall ASA->IP Phone	Passed	
UC802IF.CUM.008	Unified Mobility	Working of VoiceMail When One of the Active-Active Unity Connection Server is Down	Verify that Unified Mobile communicator clients can continue to get Voicemails when one of the Unity Connection servers is not reachable when they are in active-active configuration mode.	iPhone1->DMZ ASA->Firewall ASA->Unified Mobility->UnityCxn	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CUM.010	Unified Mobility	Directory Search and DVO to Intercluster Destination From Symbian Client	To verify that Symbian Unified Mobile Communicator client can reach an enterprise DN across a SIP trunk through a directory search.	iPhone-(Unified Mobility---Unified CM1---<sip secure>---Unified CM2---IPPhone	Passed	
UC802IF.CUM.011	Unified Mobility	Call Logs For Missed, Placed and Received Calls From Symbian Client	To verify that Symbian Unified Mobile Communicator client can see the call logs for Missed, Placed and Received calls and are updated correctly.	Symbian->Unified Mobility-->Unified CM	Passed	
UC802IF.CUM.012	Unified Mobility	Deactivate And Activate Users, Delete And Add Phones, Swap SIM Cards	To verify if Unified Mobility can demonstrate resiliency when working users are deactivated and activated, phones are deleted and added, and swapping of SIMs between different phones like WM to Symbian and vice versa.	Unified Mobility	Passed	
UC802IF.CUP.001	Unified Presence 8.0	Unified Presence Upgrade	To verify the upgrade of Unified Presence 7.x high availability capable to Unified Presence 8.0 without high availability.		Passed	
UC802IF.CUP.004	Unified Presence 8.0	Unified Personal Communicator Intracluster to XMPP Client (IM and Presence Status)	Verify flows and compatibility when multiple client types are connected to one multinode Unified Presence cluster.	Client->ASA->Unified Presence 8.0->ASA->Client	Passed	CSCtc68619
UC802IF.CUP.005	Unified Presence 8.0	Unified Personal Communicator Intercluster to XMPP Client (IM and Presence Status)	Verify mixed client intercluster communications using Cisco and third party clients.	Unified Personal Communicator->ASA->Unified Presence->Unified Presence->ASA->Client	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CUP.006	Unified Presence 8.0	Unified SIP IP Phone 9971, Deskphone Mode with Unified Personal Communicator	To verify if a user can define Unified Personal Communicator and Unified IP Phone 9971 in phone associated mode. Ensure deskphone mode works correctly.	Unified Personal Communicator/RT Pro->ASA _> Unified CM->ASA->IP phone. Unified Personal Communicator/RT Pro->ASA->Unified Presence 8.0	Passed	
UC802IF.CUP.007	Unified Presence 8.0	Unified Presence 8.0 Server Fails and Restarts	Verify after the Unified Presence server failure, can clients reconnect successfully as high availability is not supported.	Client->ASA->Unified Presence8.0->ASA->client (Clients reside on separate Unified Presence nodes)	Passed	
UC802IF.CUP.008	Unified Presence 8.0	Client Fails and Restarts	To verify that a client (Unified Personal Communicator, Unified Mobility Advantage/Unified Mobile Communicator, XMPP) can restart and connect after failure. Ensure that failed clients are removed and allowed to reconnect to the Unified Presence server.	Unified Personal Communicator->ASA->Unified Presence->Unified Mobility->ASA->client	Passed	
UC802IF.CUP.009	Unified Presence 8.0	Intercluster Communications Failure	Verify the communications breakdown between clusters.	Client->ASA->Unified Presence->SDNS->Unified Presence->ASA->client	Passed	
UC802IF.CUP.010	Unified Presence 8.0	Client Redirect	Verify when a client logins to a wrong server, if login gets redirected.	Client->ASA->Unified Presence node->Unified Presence node	Passed	
UC802IF.CUP.012	Unified Presence 8.0	Remote Clients Across WAN	Verify that clients located in remote sites are able to log in to centralized Unified Presence server and communicate with users in central and remote site across WAN with 80 ms delay.	Unified Personal Communicator->ASA->WAN->ASA->Unified Presence->Unified Personal Communicator	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.CUP.013	Unified Presence 8.0	Adhoc Text Conferencing	Verify the operation of XMPP ad-hoc text conferencing intra-cluster and inter-cluster.	XMPP->Unified Presence->XMPP	Passed	
UC802IF.CUP.014	Unified Presence 8.0	Off-Line Storage	Verify that offline IMs are received by users once they log in across multiple clusters.		Passed	
UC802IF.CUP.015	Unified Presence 8.0	User Rebalance	Verify the user rebalance.		Passed	
UC802IF.CUP.016	Unified Presence 8.0	Basic Compliancy Using Message Archiver	Verify the operation/integration of basic compliancy using an external PostgreSQL database for message archiving.		Passed	
UC802IF.CUP.017	Unified Presence 8.0	Inter-Cluster Persistent Chat	Verify the inter-cluster persistent chat functionality.		Passed	
UC802IF.CUP015	Unified Presence 8.0	Unified Presence Replication Intact During Extended Period of NTP Server Outage	To verify that the Unified Presence replication remains intact during extended period of NTP server outage.		Passed	
UC802IF.DP.100	Unified Analysis Manager	Unified Analysis Manager With Unified CM Subscribers Installed in Unified Computing System Environment	To verify if Unified Analysis Manager can be used with components running on Unified Computing System platform.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.EXC.001	Unified PC 7.x, Unified PC 8.0, Unified presence 8.0	Basic Presence/IM With Other Unified Presence Clients	Verify if the Unified Personal Communicator 8.0 client can search an LDAP database for contacts, add these contacts to their roster, and send P2P IMs with Unified Personal Communicator 7.x and third party XMPP clients on the same cluster and on different clusters.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.EXC.002	Unified PC 8.0	Unified Presence Adhoc Chatroom	Verify if the Unified Personal Communicator 8.0 client can initiate an adhoc text conferencing session with other Unified Personal Communicator 8.0/XMPP clients on an inter-cluster setup. Verify that the option to initiate a chat session with a Unified Personal Communicator 7.x user is not available.		Passed	
UC802IF.EXC.003	Unified PC 8.0	Unified Presence Persistent Chat Room	Verify if the Unified Personal Communicator 8.0 client can create a persistent chatroom, and verify that the client properly re-displays the persistent instant messages when it leaves and rejoins the room.		Passed w/ Exception	CSCtg66391
UC802IF.EXC.004	Unified PC 8.0	Unity: Visual Voicemail Capabilities for Nonsecure Messages	Verify for a Cisco Unity 8.0 subscriber that message notifications are received when a new nonsecure voicemail arrives, that the user can retrieve and listen to the voicemail, perform operations on the message, and then call the user back who left the voicemail.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.EXC.005	Unified PC 8.0	Unity: Visual Voicemail Capabilities for Secure Messages	Verify for a Cisco Unity 8.0 subscriber that message notifications are received when a new secure voicemail arrives, that the user can retrieve and listen to the voicemail, perform operations on the message, and then call the user back who left the voicemail.		Passed	
UC802IF.EXC.006	Unified PC 8.0	Unity Connection: Visual Voicemail Capabilities for Nonsecure Messages	Verify for a Unity Connection 8.0 subscriber that message notifications are received when a new nonsecure voicemail arrives, that the user can retrieve and listen to the voicemail, perform operations on the message, and then call the user back who left the voicemail.		Passed	
UC802IF.EXC.007	Unified PC 8.0	Unity Connection: Visual Voicemail Capabilities for Secure Messages	Verify for a Unity Connection 8.0 subscriber that message notifications are received when a new secure voicemail arrives, that the user can retrieve and listen to the voicemail, perform operations on the message, and then call the user back who left the voicemail.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.EXC.008	MeetingPlace, Unified CM, Unified PC 8.0, Unified Presence 8.0	Soft Phone Escalate Incoming Call to Video, Merge with Outgoing Video Call	Verify if a user can set up an audio call from a Tandberg H.323 video endpoint in another cluster via SIP trunk to Unified Personal Communicator 8.0 in soft phone mode. Escalate the call to video. Set up an outgoing video call from the same Unified Personal Communicator 8.0 client to SIP IP Phone 7985. Check for bi-directional voice path. Merge both the calls.		Failed	CSCtg71142
UC802IF.EXC.009	MeetingPlace, Unified CM, Unified PC 8.0, Unified Presence 8.0	Soft Phone Escalate Incoming Call to Video, Merge with Outgoing Video Call	Verify if a user can set up an audio call from a SIP IP Phone 9900 series video endpoint in another cluster via SIP trunk to Unified Personal Communicator 8.0 in soft phone mode. Escalate the call to video. Set up an outgoing video call from the same Unified Personal Communicator 8.0 client to SCCP (Unified Video Advantage) endpoint. Check for bi-directional voice path. Merge both the calls.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.EXC.010	MeetingPlace, Unified CM, Unified PC8.0, Unified Presence 8.0	Soft Phone Escalate Incoming Call to Video, Merge with Incoming Video Call	Verify if a user can set up an audio call from a SIP IP Phone 9900 series video endpoint in another cluster via H.323 trunk to Unified Personal Communicator 8.0 in soft phone mode. Escalate the call to video. Set up an incoming video call from another SIP endpoint (IP Phone 7985) to the same Unified Personal Communicator 8.0 client. Check for bi-directional voice path. Merge both the calls.		Passed	
UC802IF.EXC.011	MeetingPlace, Unified CM, Unified PC 8.0, Unified Presence 8.0	Soft Phone Escalate Incoming Call to Video, Merge with Incoming Video Call	Verify if a user can set up an audio call from a SCCP (Sony) video endpoint in another cluster via SIP trunk to Unified Personal Communicator 8.0 in soft phone mode. Escalate the call to video. Set up an incoming video call from a SIP/CSF (Unified Personal Communicator 8.0) endpoint to the same Unified Personal Communicator 8.0 client. Check for bi-directional voice path. Merge both the calls.		Passed	
UC802IF.EXC.012	Unified CM, Unified PC 8.0, Unified Presence 8.0	Desk Phone Mode Device Selection with Unified IP Phones (9900, 7900, and 6900 series)	Verify if a user is configured with 3 phones: (9900, 7900, and 6900 series) . Verify Unified Personal Communicator 8.0 is able to select between the 3 devices in desk phone mode.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.EXC.013	Unified CM, Unified PC 8.0, Unified Presence 8.0	Desk Phone Mode with Unified PC 8.0 Extension Mobility in Unified SRST Site	Verify a can user log in to an extension mobility enabled phone and have Unified Personal Communicator 8.0 control this phone in desk phone mode in a Unified SRST site.		Passed	
UC802IF.EXC.014	Unified CM, Unified Mobility, Unified PC 8.0, Unified presence 8.0	Dusting with Soft Phone Mode, Hand-Off to Mobility Device	Verify a call can be handed off to a mobility device from Unified Personal Communicator 8.0 soft phone.		Passed	
UC802IF.EXC.015	MeetingPlace, Unified CM, Unified PC 8.0, Unified Presence 8.0	Escalate Inter-Cluster Group IM Session to Audio Conference, and Then Video	Verify if a user can begin an adhoc IM group chat with users from multiple clusters. Escalate the group chat to an meeting place audio conference. Escalate the audio conference to video, then de-escalate the conference.		Passed	
UC802IF.EXC.016	MeetingPlace, Unified CM, Unified PC 8.0, Unified Presence 8.0	Escalate Inter-Cluster Group IM Session to MeetingPlace Web Share	Verify if a user can begin an adhoc IM group chat with users from multiple clusters. Escalate the group chat to a Meeting Place web share.		Passed w/ Exception	CSCtf84197
UC802IF.EXC.017	MeetingPlace, Unified CM, Unified PC 8.0, Unified Presence 8.0	Ad-hoc Audio Conference (Audio-Only Bridge) Escalate to Video Attempt	verify if a user can begin an adhoc audio conference with audio-only endpoints. Have 1 video-capable endpoint join later. If the video endpoint (Unified Personal Communicator 8.0) attempts to escalate to video, verify the request is rejected and the conference continues.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.EXC.018	MeetingPlace, Unified CM, Unified PC 8.0, Unified Presence 8.0	P2P Audio Escalate to Chat, then Web Share	Verify if a user can place a call to another endpoint. Escalate this call to an IM session, and from the IM session escalate to a web share session.		Passed w/ Exception	CSCtf84197
UC802IF.EXC.019	Unified CM, Unified PC 8.0	Unified PC 8.0 Through IPSEC VPN in Soft Phone Mode with Video Call	Verify if a user can bring up Unified PC 8.0 through an IPSEC VPN connection. Originate a call from Unified PC 8.0 to a Unified IP Phone through SIP ICT. Bring up video during the call. Hold and Resume the call. Turn off the video and turn on during the call. Check the call again for audio and video connection.		Passed	
UC802IF.EXC.020	Cisco IME, Unified CM, Unified PC 8.0	Softphone Mode B2B Audio Call flow Sanity	Verify if a user can use Unified PC 8.0 soft phone and verify various basic P2P phone flows in Cisco IME deployment.		Passed	
UC802IF.EXC.021	Cisco IME, Unified CM, Unified PC 8.0	Softphone Mode B2B Video Call flow Sanity	Verify if a user can use Unified PC 8.0 soft phone and verify various basic P2P video phone flows in Cisco IME deployment.		Passed w/ Exception	CSCte52965

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.EXC.022	Unified CM, Unified PC 8.0, Unified Presence 8.0	Soft Phone Escalate Incoming Call to Video, Merge with Incoming Video Call	Verify if a user can set up an audio call from a SIP video endpoint in another cluster via SIP trunk to Unified PC 8.0 in soft phone mode. Escalate the call to video. Set up an incoming video call from another SIP endpoint (7985) to the same Unified PC 8.0 client. Check for bi-directional voice path. Merge both the calls.		Passed	
UC802IF.EXC.023	Cisco IME, Unified CM, Unified PC 8.0, Unified Presence 8.0	Softphone Mode B2B Video Call to UC Integration™ for Microsoft Office Communicator	Verify operation of a video call over a Cisco IME link to a UC Integration for Microsoft Office Communicator client.		Passed	
UC802IF.EXC.024	MCU, Unified CM, Unified PC 8.0, Unified Presence 8.0	Ad-hoc Video Conference Using Software Media Bridge	Verify operation of an ad hoc inter-cluster video conference using a software media bridge with Unified PC 8.0 clients and Unified IP Phones 8900 and 9900 series.		Passed	
UC802IF.EXC.025	Unified CM, Unified PC 8.0, Unified Presence 8.0	Intercluster Scheduled Video Conference with Multiple Parties	Verify operation of an inter-cluster video conference using a Meeting Place hardware media conferencing bridge with Unified PC 8.0 clients and Unified IP Phones 8900 and 9900 series.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.EXC.027	Cisco IME, Unified CM, Unified PC 8.0, Unified Presence 8.0	PSTN Fallback with Unified PC 8.0 Client	Verify if a user can place a call over Cisco IME to another phone. Degrade the quality of the link such that PSTN fallback is initiated. Verify the Unified PC 8.0 client handles the fall back properly.		Passed	
UC802IF.EXC.028	Unified CM, Unified Mobility, Unified PC 8.0, Unified Presence 8.0	Dusting with Soft Phone Mode, Inbound From Mobile	Verify a call can be transferred from a mobility device to the Unified PC 8.0 soft phone.		Passed	
UC802IF.EXC.029	Unified CM, Unified PC 8.0, Unified Presence 8.0	LDAP Telephone Number Update	Verify if a user can change a contact's telephone number in LDAP. Verify Unified PC 8.0 receives new contact info for this user and click-to-call reaches the new number.		Failed	CSCtf17373
UC802IF.EXC.030	Unified CM, Unified PC 8.0	Unified PC 8.0 Making a Secure Call, Transfer to Nonsecure Phone	verify if a user can place a call using Unified PC 8.0 as a secure endpoint to another secure phone across a SIP trunk. Transfer the call to a non-secure endpoint and verify two-way audio remains.		Passed	
UC802IF.IME.201	Cisco IME	Secure Conferencing Over Unified B2B Trunks With SRTP Enabled Inside the Firewall	Verify that secure conferencing can be placed over Cisco IME trunks. Also verify that if SRTP is enabled on the inside, ASA supports pass through and inserts as media intermediary.	Phone1->Unified CM1->Secure ConfBridge; Phone2->Unified CM2->Cisco IME ASA->SIPT ->Off Path Cisco IME ASA->Unified CM->Conf Bridge; Phone3->Unified CM->PSTN->GW->Unified CM->Conf Bridge	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.IME.202	Cisco IME	Early Offer Over Unified B2B Trunks	Verify that Cisco IME trunks and Cisco IME aware ASA supports early offer.	Phone1->Unified CM1->Off Path Cisco IME ASA->MTP->SIPT (WAN)->Cisco IME ASA->Unified CM2->Phone2	Passed	
UC802IF.IME.203	Cisco IME	Video Call Over Unified B2B Trunk From TRP Enabled Endpoint	Verify that video calls are supported over Cisco IME trunks for a TRP enabled endpoint.	Phone1->Unified CM1->Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM2->TRP->Phone2	Passed	
UC802IF.IME.204	Cisco IME	Hold/Transfer and MoH Playback Over Unified B2B Trunks	Verify that Unified B2B calls can be placed on hold and transferred. Also verify that MoH is available when the call is placed on hold.	Phone1->Unified CM1->Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM2->Phone2;	Passed	
UC802IF.IME.205	Cisco IME	Password Mismatch Between Unified B2B and ASA	Verify that an incoming call fails after the expiry of a ticket and the call is also denied when there is a mismatch in the password between Cisco IME server and ASA.	Phone1->Unified CM1->ASA->Cisco IME ASA->SIPT (WAN)->Cisco IME ASA->Unified CM2->Phone2;	Passed	
UC802IF.IME.206	Cisco IME	Expired and Revoked Tickets for Calls Over Unified B2B Trunks	Verify that Cisco IME calls get rejected if the tickets are expired and also Cisco IME calls when the ticket is revoked.	Phone1->Unified CM1->Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM2->IP Phone2;	Passed	
UC802IF.IME.207	Cisco IME	PSTN Fallback With Unified B2B	Verify the fallback to PSTN feature of Unified B2B when the quality of the voice call deteriorates. Also verify the fallback under various sensitivity settings for different codec's using different call quality condition.	Phone1->Unified CM1->Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM2->Phone2; PSTN Fallback: IP Phone1->Unified CM->GW->PSTN->GW->	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.IME.212	Cisco IME	Unified CM Overlap Dialing Support and MTP Insertion for PSTN Fallback	Verify that Unified CM has support for overlap dialing with Unified B2B. Also verify that PSTN MTP insertion occurs when dynamically if there is a need during PSTN fallback.	Phone1->ASA->Unified CM1->Off-path Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM2->Phone2	Failed	CSCtf09331
UC802IF.IME.215	Cisco IME	PSTN Fallback for Conference Call Over a Unified B2B Link	Verify that a video conference call successfully falls back to an audio conference.	Phone1->Unified CM1->ConfBridge; Phone2->Unified CM2->Cisco IME ASA->SIPT->Off Path Cisco IME ASA->Unified CM->Conf Bridge; Phone2->Unified CM2->Cisco IME ASA->SIPT->Off Path Cisco IME ASA->Unified CM->Conf Bridge	Passed	
UC802IF.IME.216	Cisco IME	PSTN Fallback with Unified B2B Using SIP Gateway	Verify the fallback to PSTN feature of Unified B2B when the quality of the voice call deteriorates. Also to verify the fallback under various sensitivity settings for different codec's using different call quality condition.	Phone1->Unified CM1->Off path Cisco IME ASA->SIPT->Cisco IME ASA (MSP)->ASA->Unified CM (MSP)->Phone2; Phone3->Unified CM1->Off path Cisco IME ASA->SIPT->Cisco IME ASA (SEA)->Unified CM (SEA)->Phone4	Passed	
UC802IF.IME.217	Cisco IME	Unified B2B for a Phone in Unified SRST Site Phone	To verify that Unified B2B calls can be placed to a phone in Unified SRST mode.	Phone1->Unified CM1->Cisco IME ASA->Cisco IME SIPT->Off path Cisco IME ASA->Unified CM2->Call Fwd Unregister->GW->PSTN->SRST->Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.IME.218	Cisco IME	Scheduled Conferencing With WebEx and Unified MeetingPlace Using Software Conference Resource	To verify that when a scheduled conference is held over Unified B2B link, rich media experience is available.	Phone1->Unified CM1->Off Path Cisco IME ASA->Cisco IME SIPT->Cisco IME ASA->Unified CM2->SIPT->Unified MeetingPlace; Phone2->Unified CM1->Off Path Cisco IME ASA->Cisco IME SIPT->Cisco IME ASA->Unified CM2->SIPT->Unified MeetingPlace; Phone3->Unified CM->SIPT->Unified MeetingPlace	Passed	
UC802IF.IME.219	Cisco IME	Scheduled Conferencing With WebEx and Unified MeetingPlace Using Hardware Conference Resource	To verify that when a scheduled conference is held over Unified B2B link, rich media experience is available.	Phn1->Unified CM1->Off Path Cisco IME ASA->Cisco IME SIPT->Cisco IME ASA->Unified CM2->SIPT->Unified MeetingPlace; Phn2->Unified CM1->ASA->Off Path Cisco IME ASA->Cisco IME SIPT->Cisco IME ASA->Unified CM2->SIPT->Unified MeetingPlace; Phn3->Unified CM->SIPT->Unified MeetingPlace	Failed	CSCtf02177
UC802IF.IME.220	Cisco IME	Unified B2B Calls to Phone With Mobile Connect Enabled	To verify that Mobile Connect feature works with Unified B2B.	Pn1->ASA->Unified CM1->Cisco IME ASA->SIPT->ASA->Off Path Cisco IME ASA->Unified CM2->GW->PSTN	Passed	
UC802IF.IME.221	Cisco IME	Call Forward All and iDivert to Cisco Unity Over a Unified B2B link	To verify that callers can leave a voicemail in Cisco Unity when the call is over a Unified B2B link.	Phn1->ASA->Unified CM1->Off Path Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM->Phn2->CFA->Phn3->iDivert->Unity	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.IME.222	Cisco IME	Supervised Transfer With Cisco Unity Connection for a Call From Unified B2B Link	To verify that callers are successfully supervise transferred by Cisco Unity Connection when the call is from a Unified B2B link.	Phn1->Unified CM1->Cisco IME ASA->SIPT->Off Path Cisco IME ASA->Unified CM2->SIPT->Connection->Supervise Xfer->Unified CM2->Off Path Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM1->Phn2	Passed	
UC802IF.IME.223	Cisco IME	Dual Stack Trunks and Phones With Unified B2B Link	To verify that dual stack endpoints can place calls over the Unified B2B link where they communicate using IPv4 and if fallback, to communicate over the PSTN they use IPv6.	DS Phn1->Unified CM1->Off Path Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM2->DS Phn2; PSTN Fallback->DS Phn1->ASA->Unified CM1->DS SIPT->DS SIP GW->PSTN->DS SIP GW->Unified CM2->DS Phn2	Passed	
UC802IF.IME.224	Cisco IME	GPickUp Using Analog Phones and Dual stack VG224 GateWay With Unified B2B Link	To verify that GPickUp can be performed from an analog endpoint connected to dual stack VG224 for a call over the Unified B2B.	DS Phn1->Unified CM1->Cisco IME ASA->SIPT->Off Path Cisco IME ASA->Unified CM2->DS Phn2->GPickUp->DS VG224->Analog Phone	Passed	
UC802IF.IME.225	Cisco IME	Monitoring Unified B2B Link Using Unified Service Monitor	Verify the Unified B2B link and calls over Unified B2B link, which can be monitored using Unified Service Monitor.		Failed	CSCte76074
UC802IF.IME.226	Cisco IME	Locations Based Call Admission Control With Unified B2B link	Verify that calls through Unified B2B link follows the CAC settings in Locations. Also, confirm that the mid-call PSTN fallback adheres to Locations based CAC setting.	IP Phone1->ASA->Unified CM1->ASA->Off Path Cisco IME ASA->ASA->SIPT (WAN)->Cisco IME ASA->ASA->Unified CM2->IP Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.IME.227	Cisco IME	Voicemail Deposit and Retrieval From Cisco Unity Express Over Unified B2B Link	Verify that callers can deposit a voicemail in Cisco Unity Express and call Cisco Unity Express over the Cisco IME trunk to retrieve a voicemail.	Phn1->Unified CM1->Cisco IME ASA->SIPT->Off Path Cisco IME ASA->Unified CM2->IP Phone2->CFNA->Unified CM2->ASA->JTAPI->Cisco Unity Express	Passed	
UC802IF.IME.228	Cisco IME	Call Routing With Unified B2B Link When Primary Unified CM is Down	Verify that the calls can be connected successfully even when the primary Unified CM server on which Unified B2B link is down.	IP Phone1->ASA->Unified CM1->ASA->Cisco IME ASA->SIPT (WAN)->ASA->Off Path Cisco IME ASA->ASA->Unified CM2->IP Phone2	Passed	
UC802IF.IME.229	Cisco IME	Call Routing When Internet Connectivity on Calling Cluster is Down	Verify that calls can be routed successfully when the internet connectivity is down on the calling cluster.	IP Phone1->ASA->Unified CM1->ASA->Off Path Cisco IME ASA->ASA->SIPT (WAN)->Cisco IME ASA->ASA->Unified CM2->IP Phone2	Passed	
UC802IF.IME.230	Cisco IME	Call Routing Wth Unified B2B Link When the Primary Off Path ASA is Down	Verify that the calls can be connected successfully through the secondary ASA when the primary off path ASA on the called enterprise is down.	IP Phone1->ASA->Unified CM1->ASA->Cisco IME ASA->SIPT (WAN)->ASA->Off Path Cisco IME ASA (Secondary)->ASA->Unified CM2->IP Phone2	Passed	
UC802IF.IME.231	Cisco IME	Link Between Unified B2B Server and Unified CM Server Down	Verify the behavior when link between Unified CM and B2B server is down. Test if any changes to the advertised DN are updated after the link is restored.		Passed	
UC802IF.IME.232	Cisco IME	Unified B2B Link Availability When Primary Cisco IME Server is Unavailable	Verify that Cisco IME functionality is available and the secondary Unified B2B server provides the services when primary server is down.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.IME.233	Cisco IME	Unified B2B Link Interoperability With EMCC	Verify the Cisco IME functionality with EMCC.	IP Phone1->ASA->Unified CM1->ASA->Cisco IME ASA->SIPT (WAN)->ASA->Off Path Cisco IME ASA->ASA->Unified CM2->EMCC->IP Phone2	Passed	
UC802IF.IME.235	Cisco IME	Inter-Company Video Call With Various Supported Endpoints	Verify and validate inter-company video call with various supported endpoints.		Passed	
UC802IF.IME.250	Cisco IME	Scheduled Conferencing With WebEx and Unified MeetingPlace Using Hardware Conference Resource-Out Dial	To verify that when Unified Meeting Place out dials over a Unified B2B link, rich media experience is available.	Unified MeetingPlace->SIPT->Unified CM1->Cisco IME ASA->SIPT WAN->Off Path ASA->Unified CM2->Phn1; Unified MeetingPlace->SIPT->Unified CM1->Phn2; Unified MeetingPlace->SIPT->Unified CM1->Phn2	Failed	CSCtf02177
UC802IF.IME.996	Cisco IME	IME Number Normalization on Call Diversion	Verify for a phone called that has "iDivert" enabled over Cisco IME that the calling party sees the forwarded party's normalized number with "Connected Party" transformation configurations.	Phone1->Unified CM1->Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM2->TRP->Phone2	Passed	
UC802IF.IME.997	Cisco IME	Cisco IME Number Normalization on Call Transfer	Verify for a transferred call over Cisco IME that the calling party sees the newly connected called party's normalized number with "Connected Party" transformation configurations.	Phone1->Unified CM1->Cisco IME ASA->SIPT->Cisco IME ASA->Unified CM2->TRP->Phone2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.IME.998	Cisco IME	Cisco IME Number Normalization on Call Connect	Verify when a call is placed over Cisco IME that the calling party sees the called party's normalized number using "Connected Party" transformation configurations.	IP Phone1->ASA->Unified CM1->ASA->Cisco IME ASA->SIPT (WAN)->ASA->Off Path Cisco IME ASA->ASA->Unified CM2->IP Phone2	Passed	
UC802IF.IME.999	Cisco IME	Cisco IME Call When There is Congestion on Internet During Call Setup	To verify the behavior of Cisco IME when the packets are dropped or when there is network congestion on the Internet during call setup.	Phone1->Unified CM1->Off Path Cisco IME ASA->MTP->SIPT (WAN)->Cisco IME ASA->Unified CM2->Phone2	Failed	CSCtf14161
UC802IF.MID.001	Unified IP Phone	Visual Voicemail Installation	To verify the end-user installation guide and verify initial Visual Voicemail installation.	IP Phone->Unified CM; IP Phone->web server	Passed	
UC802IF.MID.002	Unified IP Phone	Version Upgrade	Verify process of upgrading a Visual Voicemail IP Phone Service.	IP Phone->Unified CM; IP Phone->web server	Passed	
UC802IF.MID.003	Unified IP Phone	Version Downgrade	Verify process of downgrading a Visual Voicemail IP Phone Service.	IP Phone->Unified CM; IP Phone->web server	Passed	
UC802IF.MID.004	Unified IP Phone	Incoming Call with Visual Voicemail Running	Verify the ability to receive an incoming call while a Visual Voicemail is running.	IP Phone->Unified CM; IP Phone->web server	Passed	
UC802IF.MID.005	Unified IP Phone	General Service Provisioning	Verify the ability to subscribe/unsubscribe the Visual Voicemail and enable/disable the Visual Voicemail service.	Unified CM->IP Phone	Passed	
UC802IF.MID.006	Unified IP Phone	Weather Application Ease-of-Use/Basic Function	Verify general ease-of-use and functionality of Weather Forecast Visual Voicemail.	IP Phone->Web Proxy->WAN	Passed	
UC802IF.MID.007	Unified IP Phone	Stock Application Ease-of-Use/Basic Function	Verify general ease-of-use and functionality of the Stocks Update Visual Voicemail.	IP Phone->Web Proxy->WAN	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.MID.008	Unified IP Phone	Calculator Ease-of-Use/Basic Function	Verify general ease-of-use and functionality of the Quick Calculator Visual Voicemail.	IP Phone	Passed	
UC802IF.MID.009	Unified IP Phone	Visual Voicemail Download Behavior with IP Phone Service Version	Verify the download behavior of a Visual Voicemail depending on IP phone service version.	IP Phone->Unified CM; IP Phone->web server	Passed	
UC802IF.MP.100.1	Unified MeetingPlace	Establish And Test Various Web Conference Features	Verify if a user can establish a Unified MeetingPlace conference with WebEx as the web conference provider and test the various meeting features such as joining the conference, network based recording, window/desktop sharing, changing roles and ending the meeting from various browser types.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	
UC802IF.MP.100.2	Unified MeetingPlace	Establish And Utilize Web Conference With Internal And External Participants	Verify if a user can establish a meeting place conference with WebEx as the web conference provider. Test various meeting features such as joining the conference, network based recording, window/desktop sharing, changing roles and ending the meeting from various browser types.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	
UC802IF.MP.101.1	Unified MeetingPlace	Establish And Test Web And Audio Conferences Through Different DTMF Modes	Verify is a user can establish meeting place conferences with WebEx as the web conference provider and dial in utilizing different DTMF modes.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.MP.102.1	Unified MeetingPlace	Test Video Endpoints With Hardware Media Server	Verify if a user can establish MeetingPlace conferences with WebEx as the web conference provider utilizing a hardware media server. Join the conferences using multiple endpoint models configured to either a standard or high video rate and different video codecs.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	
UC802IF.MP.103.1	Unified MeetingPlace	Test Video Endpoints With Software Media Server	Verify if a user can establish MeetingPlace conferences with WebEx as the web conference provider utilizing a software media server. Join the conference using multiple endpoint models configured to the available video modes and different video codecs.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	
UC802IF.MP.104	Unified MeetingPlace	Unified MeetingPlace Upgrade Support Existing Recordings	Verify if a user can establish and record multiple MeetingPlace meetings where Unified MeetingPlace is the web conference provider. Upgrade Unified MeetingPlace 7.0MR1 to Unified MeetingPlace 8.0 and ensure that existing recordings are still accessible and playable.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.MP.105	Unified MeetingPlace	AppServer Failover With WebEx Node Recovery on Hardware Media Server System	Verify is a user can perform Unified MeetingPlace Application Server failover while internal meetings are in progress on a Hardware Media Server based system. The WebEx node will try to connect to the active WebEx TSP connection on the backup Application Server. Once the backup server is up, users will be prompted to enter the phone number for dialout. The same meeting ID will be utilized.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	
UC802IF.MP.106.1	Unified MeetingPlace	Test Endpoints Audio CODECS With Software Media Server	Verify if a user can establish meeting place conferences with WebEx as the web conference provider utilizing a software media server. Join the conference using multiple endpoint models configured to different supported audio CODECS.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	
UC802IF.MP.107.1	Unified MeetingPlace	Test Endpoints Audio CODECS With Hardware Media Server	Verify is a user can establish a meeting place conference with WebEx as the web conference provider utilizing a hardware media server. Join the conference using multiple endpoint models configured to the different supported audio codecs.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.MP.108.1	Unified MeetingPlace	Create And Utilize a Web Conference Through Microsoft Outlook	To verify if a user can create and establish a meeting Place conference with WebEx as the web conference provider. The meeting should be created with Mcrosoft Outlook and attended by clicking the meeting link within the corresponding meeting email. Send the email to users whose email format is set to SMTP and Exchange, HTML and plain text.	Endpoint->ASA->Unified CM->SIP Trunk->Unified MeetingPlace	Passed	
UC802IF.MP.109	Unified MeetingPlace	Application Server Failover With WebEx Node Recovery on SMS System	To verify if Unified MeetingPlace Application Server can failover while internal meetings are in progress on a Software Media Server based system. The WebEx node will try to connect to the active WebEx TSP connection on the backup Application Server. Once the backup server is up, users will be prompted to enter the phone number for dialout. The same meeting ID will be used.		Passed	
UC802IF.MP.803	Video Conference	Video Conference Through Unified MeetingPlace	Verify is a user can establish an adhoc video conference, activate voice call, and schedule conference.	IP Phone 1 / IP Phone 2 / IP Phone 3->Unified CM 1->SIP Trunk->Unified MeetingPlace	Passed	
UC802IF.MP.804	Video	Transcoding Using Unified MeetingPlace Hardware Conference	Verify if transcoding between two Unified MeetingPlace participants having different video definitions 1.1 and 1.2 respectively is successful.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.MP.810	Video	Schedule Adhoc Conference With Various Video Definition Endpoints	To verify the conference with H.263, H.264 QCIF, CIF, 4CIF endpoints using Software Media Server.		Passed	
UC802IF.MP.811	Video	Adhoc Video Conference With Third Party Video Endpoints	To verify whether Third Party video endpoints (Sony, Tandberg , Polycom VSX) can attend a meeting place meeting hosted by Unified MeetingPlace application.		Passed	
UC802IF.SAF.101	Service Advertisement Framework	Load Balancing of Calls to Remote Unified CM Cluster Advertising Same HostedDN From Unified CME	Verify when a trunk in the advertising cluster is assigned to two Unified CM, Unified CME receives the same pattern twice, one for each Unified CM node. Calls from Unified CME to the advertised DN should alternate between the two Unified CM nodes.	IP Phone->Unified CME->H.225 Trunk->ASA->Unified CM->ASA->IP Phone	Passed	
UC802IF.SAF.102	Service Advertisement Framework	Load Balancing Over SIP and H225 Trunks for Learned Patterns and Alternate Routing Between Unified CM Nodes	Verify when a requesting service has both SIP and H.255 trunk attached to it, call to the learned pattern alternates over both SIP and H.225 trunk.	IP Phone->Unified CM->H.225/SIP Trunk->ASA->Unified CM->ASA->IP Phone	Passed	
UC802IF.SAF.103	Service Advertisement Framework	Co-Existing With Static Routes on Unified CME	Verify when Service Advertisement Framework is advertising a pattern that matches a statically configured dialpeer, then routes from Unified CME to the advertised DN should be prioritized appropriately.	IP Phone->Unified CME->H.225 Trunk->Unified CME->IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.SAF.104	Service Advertisement Framework	Join Across Lines Over Service Advertisement Framework Trunks	Verify that join across lines works successfully over Service Advertisement Framework trunks.	IP Phone->ASA->Unified CM->ASA->H.225 Trunk->ASA->Unified CM->ASA->IP Phone	Passed	
UC802IF.SAF.105	Service Advertisement Framework	Video Conference Over H225 Trunks	Verify that patterns can be blocked from being learnt based on the remote call control field.	IP Phone1->Unified CM->H.225 Trunk->Unified CM->IP Phone2; IP Phone1->Unified CM->H.225->Unified CME->IP Phone3; IP Phone1->Conference->IP Phone2 and IP Phone3	Passed	
UC802IF.SAF.106	Service Advertisement Framework	Geophysical Location Based Advertisement and Service Advertisement Framework Redundancy	Verify that patterns are advertised based on the HostedDN configuration in CoW. And also verify that redundancy is available for Service Advertisement Framework.	IP Phone->Unified CME->H.225 Trunk->Unified CM->IP Phone	Passed	
UC802IF.SAF.107	Service Advertisement Framework	PSTN Failover Due to Bandwidth Over Subscription on Calling Cluster	Verify the bandwidth control for calls over Service Advertisement Framework trunks are controlled using Locations based CAC. Verify that calls failover to PSTN when sufficient bandwidth is not available.	IP Phone1->Unified CM->H.225 Trunk->Unified CM->IP Phone2; IP Phone1->Xfer->Unified CM->PSTN->IP Phone3	Passed	
UC802IF.SAF.108	Service Advertisement Framework	PSTN Failover Due to Bandwidth Over Subscription on Called Cluster	Verify the bandwidth control for calls over Service Advertisement Framework trunks are controlled using Locations based CAC. Verify that calls failover to PSTN when sufficient bandwidth is not available.	IP Phone1->Unified CM->H.225 Trunk->Unified CM->IP Phone2; IP Phone3->Unified CM->PSTN->IP Phone4	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.SAF.109	Service Advertisement Framework	Service Advertisement Framework Aware H323 Gateway Providing Unified SRST Services	Verify that Unified SRST router when acting as a gateway can learn advertised DN's and route incoming calls based on the learned pattern. The same router should also route calls to PSTN when if goes into Unified SRST mode.	IP Phone1->Unified CME->PSTN->H.323 GW->Unified CM->IP Phone2	Passed	
UC802IF.SAF.110	Service Advertisement Framework	Service Advertisement Framework Functioning on Unified SRST Router	Verify that calls to central site phone uses the PSTN even when the WAN link is up but the phones are still registered to the Unified SRST router because of connection monitor timer.	IP Phone1->SRST->PSTN->Unified CM->IP Phone2	Passed	
UC802IF.SAF.111	Service Advertisement Framework	Lost Connectivity Between Service Advertisement Framework Forwarders	To verify the behavior when the connectivity between two Service Advertisement Framework forwarders is lost but clients are able to actively maintain connectivity with their forwarders.		Passed	
UC802IF.SAF.112	Service Advertisement Framework	Lost Connectivity Between Advertising Client and Service Advertisement Framework Forwarder	Verify the behavior when client looses connectivity to the Service Advertisement Framework (SAF) forwarders. Any change to the advertised DN is pushed to the Forwarder once connectivity is restored.	IP Phone->Unified CME->H.225 Trunk->Unified CM->IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.SAF.113	Service Advertisement Framework	Age-Out Timer Expiry and PSTN Flush Out Timer Expiry	Verify that the patterns are marked down when connectivity to SAF Forwarder is lost. DN or patterns are flushed out after age-out timer expiry and PSTN routes are also deleted after PSTN age-out expiry.	IP Phone->Unified CME->H.225 Trunk->Unified CM->IP Phone	Passed	
UC802IF.SAF.114	Service Advertisement Framework	Manual Summarization Using Unified Border Element	Verify that Unified Border Element can be used manually to summarize SAF Advertisements from one SAF AS and re-advertise these into another AS.	IP Phone->Unified CM->H.225 Trunk->Unified CM->IP Phone	Passed	
UC802IF.SAF.115	Service Advertisement Framework	Calls From Unified Mobile Communicator to SAF Learned Pattern	Verify that incoming calls from a mobility agent can be routed over SAF trunks. In case of failure the call also failover PSTN.	Unified Mobile Communicator->Unified CM->H.225 Trunk->Unified CM->IP Phone; Unified Mobile Communicator->Unified CM->GW->PSTN->GW->Unified CM->IP Phone	Passed	
UC802IF.SAF.116	Service Advertisement Framework	Calls to Cisco Unity Connection Over SAF Trunk	Verify that voicemail pilot can be advertised and calls can be placed successfully over SAF trunk.	IP Phone->Unified CM->H.225 Trunk->Unified CM->Unity Connection->Xfer->IP Phone	Passed	
UC802IF.SAF.117	Service Advertisement Framework	Scheduled Video Conference With Unified MeetingPlace Over SAF Trunk	Verify that invitees of Unified Meeting Place conference can call into Meeting Place over SAF trunk and video is enabled for those participants with video capable endpoints.	IP Phone->Unified CM->H.225 Trunk->Unified CM->SIPT->Unified MeetingPlace	Passed	
UC802IF.SAF.118	Service Advertisement Framework	Meeting Place Dial Out Over SAF Trunk	Verify that calls from Unified MeetingPlace to users in remote cluster are successful over SAF trunk.	Unified MeetingPlace->SIPT->Unified CM->H.225 (SAF)->Unified CM->IP Phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.SAF.119	Service Advertisement Framework	Calls to Unified CCX Over SAF trunks	Verify that route points can be advertised and calls can be placed successfully over SAF trunk.	IP Phone->Unified CM->H.225 Trunk (SAF)->Unified CM->JTAPI->Unified CCX->Xfer->IP Phone	Passed	
UC802IF.SAF.120	Service Advertisement Framework	Calls to Cisco Unity Express Registered to Unified CM Over SAF Trunks	Verify that route points can be advertised and calls can be placed successfully over SAF trunk.	IP Phone->Unified CM->H.225 Trunk (SAF)->Unified CM->JTAPI->Cisco Unity Express	Passed	
UC802IF.SAF.121	Service Advertisement Framework	Advertising DN Ranges Associated With Voice Gateway	Verify that calls are routed to the voice gateway based on SAF advertisements by the gateway.	IP Phone->Unified CM->H.225 Trunk (SAF)->Gateway->PSTN	Passed	
UC802IF.SAF.122	Service Advertisement Framework	Call Over SAF Trunk to a Busy Phone	Verify that when a call over SAF trunk is placed to a busy phone then the call does get rerouted over PSTN.	IP Phone1->Unified CM->H.225 Trunk->Unified CM->IP Phone2;	Passed	
UC802IF.SAF.123	Service Advertisement Framework	Incoming SAF Call Terminated on Remote Destination Phone and Dusted to Desk Phone	To Verify that an incoming call through SAF trunk can be answered on a mobile phone, it's remote destination. Move the call to Desk phone by disconnecting the call at mobile phone and resuming the call at Desk phone. Move call to and fro between desk phone and Mobile phone.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.SAF.124	Service Advertisement Framework	Incoming SAF Call is Call Forwarded to PSTN Through SIP Gateway	Verify if an incoming SAF call can be Call Forwarded All to PSTN destination through a SIP gateway. Verify if call is established successfully.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.SAF.125	Service Advertisement Framework	Incoming SAF Trunk Call Forwarded Through SAF Trunk to a Third Cluster	Verify if cluster 2 phone is unregistered and set for Call Forwarded Unregistered to a number (phone2) in cluster 3 or 1. Verify that incoming call through SAF trunk is established through another SAF trunk to phone 2.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.SAF.126	Service Advertisement Framework	Failure of Active Unified CM When SAF Calls are Active	Verify that when SAF calls are active, fail the active Unified CM by shutting down the port. Verify that the active calls stay and subsequent SAF calls are successful.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.SAF.127	Service Advertisement Framework	Reload Unified SRST Router Which is also the SAF Forwarder	Verify that Unified SRST router re-learns all the patterns after reloading.	Unified CM->SAFF->SAFF->SAFF/SRST	Passed	
UC802IF.SRST.905	Unified CME and SRST	Supplementary Services Through SIP Phones With Unified CME in Unified SRST Mode	To verify whether the user can perform Supplementary services through Unified SIP IP Phones in Unified SRST Mode.	RT SIP Phone --- WAN Link --- SRST -----Transfer/Hold Resume /Conference ----- PSTN ---HQ Phone	Passed	
UC802IF.SRST.906	Unified SRST	Unified SRST SCCP Phone Hears Multicast MOH	Verifies whether Unified SRST SCCP phone can hear multicast MOH when Central Site Unified CM phone puts the call on hold.	IP Phone 6921/6941/6961 --- Unified CM ---- WAN Link --- SRST Phone ----Unified CM-on Hold ---- MMOH --- SRST Phone	Passed	
UC802IF.SRST.907	Unified SRST	Unified SRST Phone Joins MeetingPlace Meeting Hosted by Central Site Unified CM User	Verify whether SIP Secured Unified SRST can join meeting via PSTN phone.	Unified CM--WAN Link---SRST --Secure ---PSTN --- Meetingplace	Passed	
UC802IF.SRST.908	Unified SRST	Call From Unified SRST to H323 Gateway	To verify whether Unified SRST Phone is able to place a PSTN Call to Unified CCX agent when the router is in fall back mode.	Unified CM -- WAN Link---SRST -- PSTN --- Unified CCX Agent	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.SRST.909	Unified SRST	Call From Unified SRST:SIP Gateway to Unified CM Phone	Verifies whether a phone can talk to Unity Connection through PSTN and is also able to deposit mail to Unified CM subscriber.	Unified CM -- WAN Link --- SIP --- SRST ---Fail over ----PSTN --- Unified CM Phone --Unity connection	Passed	
UC802IF.SRST.910	Unified SRST	MGCP Fallback to PSTN	Verifies whether Unified SRST phones can call PSTN when Unified SRST:MGCP gateway is in Fall back mode.	Unified CCX Node 1->ASA->Clustering over WAN->ASA->Unified CCX Node 2	Passed	
UC802IF.UNC.201	Cisco Unity Connection	Download Direct and Forwarded Voicemails From Unity Connection Server2 When Server1 is Down	To verify that Cisco UC Integration for Microsoft Office Communicator can successfully download direct and forwarded voicemail from currently active Unity Connection server when one of the Unity Connection servers in active-active cluster is down where the VM is originally deposited.	UC Integration™ for Microsoft Office Communicator->Unified CM->Unity Connection	Passed	
UC802IF.UNC.202	Cisco Unity Connection	Updating list, Message With attachment, and Sender's Presence Status With Visual Voicemail on Cisco UC Integration™ for Microsoft Office Communicator	To verify that visual voicemail list is updated on arrival of new message while list is displayed, ability to display messages with attachment along with the presence status of the sender.	Unity Connection->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	
UC802IF.UNC.203	Cisco Unity Connection	Message Actions With Visual Voicemail on Cisco UC Integration™ for Microsoft Office Communicator	Verify that message actions such as play, pause, rewind, mark as new, delete, reply, reply to IM can be performed.	Unity Connection->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.UNC.204	Cisco Unity Connection	Retrieve, Reply And Send a Secure Message Using Cisco UC Integration™ for Microsoft Office Communicator	Verify that messages that have been marked secure can be retrieved and replied to. Also if it is possible to send a voicemail and mark it secure.	Unity Connection->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	
UC802IF.UNC.205	Cisco Unity Connection	Securing Visual Voicemail With Cisco UC Integration™ for Microsoft Office Communicator	Verify that security (HTTPS) can be enabled for visual voicemail.	Unity Connection->Unified CM->UC Integration™ for Microsoft Office Communicator	Passed	
UC802IF.UNC.206	Cisco Unity Connection	Download Direct and Forwarded Voicemails From Unity Connection Server2 When Server1 is Down	To verify that Unified IP Phone 8900 and 9900 series can successfully download direct and forwarded voicemails from currently active Unity Connection server when the primary Unity Connection server in active-active cluster is down where the VM is originally deposited.	RT phone->Unified CM->Unity Connection	Passed	
UC802IF.UNC.207	Cisco Unity Connection	Updating list, Message With attachment, and Sender's Presence Status With Visual Voicemail on Unified IP Phone 8900 and 9900 series	To verify that visual voicemail list is updated on arrival of new message while list is displayed, ability to display messages with attachment along with the presence status of the sender.	Unity Connection->Unified CM->RT phone	Passed	
UC802IF.UNC.208	Cisco Unity Connection	Message Actions With Visual Voicemail on Unified IP Phone 8900 and 9900 series	Verify that message actions such as play, pause, rewind, mark as new, delete, reply, reply to IM can be performed.	Unity Connection->Unified CM->RT phone	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.UNC.209	Cisco Unity Connection	Retrieve, Reply And Send a Secure Message Using Unified IP Phone 8900 and 9900 series	Verify that messages that have been marked secure can be retrieved and replied to. Also if it is possible to send a voicemail and mark it secure.	Unity Connection->Unified CM->RT phone	Passed	
UC802IF.UNC.210	Cisco Unity Connection	Securing Visual Voicemail With Unified IP Phone 8900 and 9900 series	Verify that security (HTTPS) can be enabled for visual voicemail.	Unity Connection->Unified CM->RT phone	Passed	
UC802IF.UNC.211	Unity Connection	Short Duration Failure of Unity Connection Publisher During Light Traffic	To verify that Unity Connection Publisher comes back up and running after short duration failure and under light traffic.	UNC1(active(UNC2--<skinny>---Unified CM	Passed	
UC802IF.UNC.212	Unity Connection	Short Duration Failure of Unity Connection Subscriber During Light Traffic	To verify that Unity Connection Subscriber comes back up and running after short duration failure and under light traffic.	UNC1(active(UNC2--<skinny>---Unified CM	Passed	
UC802IF.UNC.213	Unity Connection	Long Duration Failure of Unity Connection Subscriber During Light Traffic	To verify that Unity Connection Subscriber comes back up and running after Long duration failure(>12 hours) and under light traffic.	UNC1(active(UNC2--<skinny>---Unified CM	Passed	
UC802IF.UNC.214	Unity Connection	Unity Connection Failover in Active-Active Setup When Critical Service is Stopped	To verify that Unity Connection subscriber server picks up the functionality of the other server in the cluster when it fails.	UNC1(active(UNC2--<skinny>---Unified CM	Passed	
UC802IF.UNC.251	Unity Connection	Replication of Objects From Remote Site to Unity Connection Digitally Networked	Verify that objects such as users, SDL and CSS on a remote connection (inter-site link) are replicated to a Unity Connection server that is digitally networked (intra-site link).	Node A <-IntraSite->Node B<-InterSite->NodeC	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.UNC.252	Unity Connection	Sending and Receiving Messages Across Network of Unity Connection Servers	Verify that messages can be sent to any user on inter-site or intra-site link.	Node A <-IntraSite->Node B<-InterSite->NodeC	Passed	
UC802IF.UNC.253	Unity Connection	Sending Messages to SDL	Verify that messages can be sent to a SDL that has members from all the networked servers (intra-site and inter-site links).	Node A <-IntraSite->Node B<-InterSite->NodeC	Passed	
UC802IF.UNC.254	Unity Connection	Non-Delivery Receipt for Messages Sent to Non-Existent Mailbox	Verify that Non-Delivery Receipts (NDR's) are received when messages are sent to a non-existent subscriber.	Node A <-IntraSite->Node B<-InterSite->NodeC	Passed	
UC802IF.UNC.255	Unity Connection	Cross Box Handoff to User in Remote Site	Verify that the cross box handoff is successful with CCI.	Node A <-IntraSite->Node B<-InterSite->NodeC	Passed	
UC802IF.UNC.256	Unity Connection	Deleting SDL Used by Directory Handler in Remote Site	Verify that deleting SDL used by Directory Handler does not cause any error condition.	Node A <-IntraSite->Node B<-InterSite->NodeC	Passed	
UC802IF.UNI.101	Cisco Unity	eMWI on Shared Lines With Cisco Unity-Unified CM SCCP Integration	Verify that MWI count is seen on Unified IP Phone 8900 and 9900 series when a new voicemail is left for the subscriber in Cisco Unity integrated to Unified CM using SCCP.	IP Phone->Unified CM->SCCP->Unity	Passed	
UC802IF.UNI.102	Cisco Unity	eMWI on Shared Lines With Unity-Unified CM SIP Integration	Verify that MWI count is seen on Unified IP Phone 8900 and 9900 series when a new voicemail is left for the subscriber in Cisco Unity integrated to Unified CM using SIP.	IP Phone->Unified CM->SIP->Unity	Passed	
UC802IF.UNI.103	Cisco Unity	eMWI for Extension Mobility Across Cluster	Verify that eMWI works for Extension Mobility across cluster.	Unity (MWI)->Unified CM->SIPT->Unified CM->IP Phone 9971/9951/8961 (Extension Mobility)	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.UNI.104	Cisco Unity	Support of eMWI Over Inter Cluster SIP Trunks	Verify that eMWI works over inter cluster CIP trunks.	Unity (MWI)->Unified CM->SIPT->Unified CM->IP Phone 9971/9951/8961	Passed	
UC802IF.UNI.105	Cisco Unity	Support of eMWI During Failover and Bulk Re-Synchronization in Cisco Unity	Verify that eMWI works when primary VM server is down and the standby server is providing VM service.		Passed	
UC802IF.UNI.110	Cisco Unity	Cisco Unity in UMR Mode - Partner Exchange Server Unavailable for Short Duration	Verify that Cisco Unity can handle calls when it looses connectivity to the partner exchange server.	IP Phone->Unified CM->SCCP->Unity	Passed	
UC802IF.UNI.111	Cisco Unity	Operation of Cisco Unity When Non-Partner Exchange Server is Offline	Verify that Cisco Unity can handle calls when it looses connectivity to the partner exchange server.	IP Phone->Unified CM->SCCP->Unity	Passed	
UC802IF.UNI.112	Cisco Unity	Unity in UMR mode - Partner Exchange Server Unavailable for Long Duration	Verify that Cisco Unity can handle calls when it looses connectivity to the partner exchange server for more than 24 hours.	IP Phone->Unified CM->SCCP->Unity	Passed	
UC802IF.UNI.113	Cisco Unity	Primary Unified CM Server Unavailable to Cisco Unity Server	Verify that Cisco Unity can handle calls when it looses connectivity to the primary Unified CM server.	IP Phone->Unified CM->SCCP->Unity	Passed	
UC802IF.UNI.114	Cisco Unity	Primary Global Catalog Server Unavailable	Verify that Cisco Unity can handle calls when it looses connectivity to the primary Global Catalog server.	IP Phone->Unified CM->SCCP->Unity	Passed	
UC802IF.UOM.001	Unified Operations Manager	Unified Operations Manager in Unified CM Cluster Monitor UC Components	Verify if Unified Operations Manager installed in Unified Communications Cluster can monitor UC components.	CUOM->SNMP-> UC Components	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802IF.UOM.002	Unified Operations Manager	Unified Service Monitor MOS Score Reporting Capability Within Unified Communications 8.0 Deployment	Validate Unified Service Monitor MOS score reporting capability within Unified Communications deployment model.	Unified CM->Download CDR->CUSM	Passed	
UC802IF.WIR.001	Wireless	Wireless IP Phone 9971 WMM SIP Snooping	Verify SIP snooping support on Unified IP Phone 9971 WiFi.	9971->AP->WLAN controller->Unified CM->IP phone	Passed	
UC802IF.WIR.010	Wireless	H-REAP/Unified SRST remote site, Flapping WAN link	Verify to establish wireless connections from devices at remote sites and force the remote sites into Unified SRST mode.	7925->AP->GW->Unified CM->IP phone	Passed	
UC802IF.WIR.011	Wireless	Redundant WLAN Controllers in Active-Active Mode	Verify the active/active WLAN controller redundancy.	Wireless phone->AP->WLAN controller->Unified CM->ip phone	Passed	
UC802L.CCM.001		Table Out of Sync Detection Service Parameter ON	Verify that no adverse affect of call processing occurs during DB table sync operations when Table Out of Sync Detection service parameter is enabled.		Passed	
UC802L.CCM.005		IP to IP Calls	Verify that calls from SFO-ORD IP phone to SFO-ORD IP phone are successful.		Passed	
UC802L.CCM.020		SJC/RFD Cluster Upgrade Times	Verify that when the tests upgrades with IO throttling is enabled, the system is under moderate load, no phones unregister, no code yellows occur, and the upgrade completes in a reasonable amount of time.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802L.CCM.108		IP to IP Calls	Verify that calls from AZO IP phone to AZO IP phone are successful.		Passed	
UC802L.CCM.120		AZO Cluster Upgrade Times	Verify that when the tests upgrades with io throttling is enabled, the system is under moderate load, no phones unregister, no code yellows occur, and the upgrade completes in a reasonable amount of time.		Passed	
UC802L.CCM.301		Table Out of Sync Detection Service Parameter ON	Verify that no adverse affect of call processing occurs during DB table sync operations when Table Out of Sync Detection service parameter is enabled.		Passed	
UC802L.CCM.305		IP to IP Calls	Verify that calls from SFO-ORD IP phone to SFO-ORD IP phone are successful.		Passed	
UC802L.CCM.320		DFW Cluster Upgrade Times	Verify that when the tests upgrades with io throttling is enabled, the system is under moderate load, no phones unregister, no code yellows occur, and the upgrade completes in a reasonable amount of time.		Passed	
UC802L.CME.002		IP to ICT to Communications Manager Express to IP Calls	Verify that calls from SFO-ORD IP phone to Unified CME IP phone over Inter-Cluster Trunk are successful.		Passed	
UC802L.CME.302		IP to ICT to Communications Manager Express to IP Calls	Verify that calls from SFO-ORD IP phone to Unified CME IP phone over Inter-Cluster Trunk are successful.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802L.GTW.003		IP to PSTN to IP Calls	Verify that calls from SFO-ORD IP phone to SFO-ORD IP phone over PSTN are successful.		Passed	
UC802L.GTW.105		IP to PSTN to IP Calls	Verify that calls from AZO IP phone to AZO IP phone over PSTN are successful.		Passed	
UC802L.GTW.303		IP to PSTN to IP Calls	Verify that calls from SFO-ORD IP phone to SFO-ORD IP phone over PSTN are successful.		Passed	
UC802L.MP.101		IP to MeetingCalls	Verify that calls from IP phone to Unified MeetingPlace are successful.		Passed	
UC802L.NME.001	Network Management	Cisco Unified Operations Manager	Verifies functionality of Cisco Unified Operations Manager during use with a large scale testbed supporting up to 30,000 devices and users.		Passed	
UC802L.NME.002	Network Management	Cisco Unified Service Monitor	Verify comprehensive voice quality measurements through the combination of Cisco 1040 Sensors and Cisco VTQ (Voice Transmission Quality) and alert generation sent to an upstream application (Cisco Unified Operations Manager) when an MOS threshold is violated. Verification is accomplished using testbeds supporting up to 30,000 devices.		Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UC802L.NME.003	Network Management	Cisco Unified Service Statistics Manager	Verify the ability to provide visibility into and usability of key metrics such as call volume, service availability, call quality, resource utilization, and capacity for a large enterprise testbed of up to 30,000 devices.		Passed	
UC802L.UNC.001	Cisco Unity Calls	IP to Unity Calls	Verify that calls to SCCP integrated Unity Voicemail are able to deposit and retrieve voicemail successfully.		Passed	
UC802L.UNC.301		IP to Unity Calls, SIP Integrated	Verify that calls to SIP integrated Unity Voicemail are able to deposit and retrieve voicemail successfully.		Passed	
UCS712IF.CCM.101	IPv6	Blind Transfer of IPv6 Call to Unified CCX Phone Agent	Verify the call from a remote cluster across a dual stack SIP trunk to dual stack SCCP phone. From the SCCP phone the call is blind transferred to Unified CCX agent.	SCCP (v6/v4) -->Unified CM -->SIPT (ANAT-on) (addressing mode v6/v4) (v6 sig pref/media pref) -->Unified CM -->SCCP (v6/v4) -->blind transfer -->IPCCX -->SCCP Phone (v6/v4) IP Phone Agent	Passed	
UCS712IF.CCM.103	IPv6	Video Call Over SIP Trunk with ANAT Enabled	Verify the call from Unified IP Phone 7985 over a SIP trunk with ANAT enabled to another IP Phone 7985. The call finally being transferred to a Unified Personal Communicator with Unified Video Advantage.	7985 (v4) -->Unified CM -->SIPT (ANAT-on) (addressing mode v6/v4) (v6 sig pref) (v6 media pref) -->Unified CM -->7985 (v4) -->Xfer -->Unified Personal Communicator	Passed	
UCS712IF.CCM.106	IPv6	PSTN Call to Dual Stack VG224 and Call Answered by Group PickUp.	Verify if a call from PSTN to dual stack VG224 is answered by another phone using Group Pick Up.	PSTN Phone -->SIP GW (v6/v4) -->SIPT (v6 sig/media pref)-->Unified CM -->VG224 GW (v6/v4) -->FXS phone -->GPickup -->FXSPhone.	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF.CCM.107	IPv6	Call Over a Dual Stack SIP Trunk Forwarded Over ICT to Unified CME	Verify when a call is placed from a dual stack phone to another dual stack phone where the SIP trunk has also been configured to support both IPv4 and IPv6, but the media and signaling preference has been set to IPv6. The call is forwarded on busy to Unified CME over ICT.	SCCP Phone (v4/v6) -->Unified CM -->SIPT (v4/v6) -->Unified CM -->SCCP Phone (v4/v6) -->CFB -->ICT -->CME -->SIP Phone	Passed	
UCS712IF.CCM.109	IPv6	Call Park Over SIP Trunk With Media And Signaling Preference Set to IPV6 And IPv4	To verify if a call can be placed from a dual stack phone to another dual stack phone where the SIP trunk is also dual stack and is configured to prefer IPv6 for signaling and media. The call is then parked and retrieved by a Unified IP 7925 phone.	SCCP (v6/v4) -->Unified CM -->SIPT (v4/v6) (v6 sig/media pref) -->SCCP (v6/v4) -->Call Park; SCCP (v6/v4) -->Unified CM -->SIPT (v4/v6) (v6 sig/media pref) -->7925	Passed	
UCS712IF.CCM.109	IPv6	Call Park Over a SIP Trunk With Media and Signaling Preference Set to IPv6	Verify if a user can place a call from a dual stack phone to a another dual stack phone where the SIPT is also dual stack and is configured to prefer IPv6 for signaling and media. The call is then parked and retrieved by a IP 7925 phone.	SCCP (v6/v4) -->CUCM -->SIPT (v4/v6) (v6 sig/media pref) -->SCCP (v6/v4) -->Call Park; SCCP (v6/v4) -->CUCM -->SIPT (v4/v6) (v6 sig/media pref) -->7925	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF.CCM.111	IPv6	Join Across Lines With IPv6 And IPv4 Media	To verify if a user can join calls across lines where the media on line 1 is IPv6 and media on line 2 is IPv4.	SCCP Phn1 (v6/v4) (secure) -->Unified CM -->SCCP Phn2 Line 1 (v6/v4) (secure) (v6 sig/media pref); SIP Phn3 (secure) -->Unified CM -->SIPT -->SCCP Phn2 Line 2 (v4/v6) (secure); SCCP Phn2 Line 2 -->Join across Lines -->Conf Bridge (secure)	Passed	
UCS712IF.CCM.114	IPv6	Dual Stack Phone to Cisco Unity Express AA Transferred to Another Dual Stack Phone	To verify that calls can be placed from a dual stack phone to CTI devices such as Cisco Unity Express.	SCCP (v6/v4) -->Unified CM -->CTI -->Cisco Unity Express -->AA -->Unified CM -->SCCP (v6/v4)	Passed	
UCS712IF.CCM.118	IPv6	Dual Stack SIP Trunk to TRP Enabled Endpoint	Verify that MTP is invoked when the endpoint has been configured with TRP. Also to verify that when the call is IPv6 then only one MTP is invoked for the call.	SCCP (v6/v4) -->Unified CM -->SIPT (v6/v4) (v6 media/sig pref) (ANAT on) (MTP) -->Unified CM -->MTP/TRP -->Unified Personal Communicator	Passed w/ Exception	CSCsa60566
UCS712IF.CSF.010	UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator Client in Phone Associated Mode	Verify that the UC Integration for Microsoft Office Communicator Client is in phone associated mode to a secure IP Phone 7916 and can set up a secure ad-hoc conference.	MOC+CSF1 --->CUCM-->conference <---- MOC+CSF2	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF.CSF.015	UC Integration™ for Microsoft Office Communicator	UC Integration™ for Microsoft Office Communicator Client on Desk Phone	Verify if the UC Integration for Microsoft Office Communicator client is associated to an Extension Mobility Unified IP 9900 or 8900 series desk phone. This client is part of a group pick up and group pick up an incoming DVO call set up by Unified Mobile Communicator.	MOC+CSF1->CUCM1(Extension Mobility+Group Pickup)	Passed	
UCS712IF.UNC.103	Unity Connection	Message Handling for Non-Existent Subscriber with Unified Messaging Gateway	To verify that a fax received by Cisco Gateway can be relayed to Unity Connection via Cisco Fax Server. And to verify that NDR is received by Connection from Unified Messaging Gateway when the message is forwarded to non-existent subscriber in Cisco Unity Express.	Fax -->PSTN -->IOS Gateway -->CFS -->Unity Connection <-->VPIM <-->UMG <-->VPIM <-->Cisco Unity Express	Passed	
UCS712IF.UNC.103	Messaging Gateway, Unity Connection, Unity Express	Message Handling for Non-Existent Subscriber with Unified Messaging Gateway	To verify that a fax received by Cisco Gateway can be relayed to Unity Connection via Cisco Fax Server. And to verify that NDR is received by Unity Connection from Unified Messaging Gateway when the message is forwarded to non-existent subscriber in Unity Express.	Fax -->PSTN -->IOS Gateway -->CFS -->Unity Connection <-->VPIM <-->UMG <-->VPIM <-->CUE	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF.UNC.201	Unity Connection	Forwarding a Fax Message From Unity Connection to Cisco Unity Express Over VPIM	To verify that a fax received by Cisco Gateway can be relayed to Unity Connection via Cisco Fax Server. And to verify that the Connection subscriber can forward the message to Cisco Unity Express over VPIM and Cisco Unity Express subscriber can in turn forward back the message to Connection subscriber over a VPIM network.	Fax -->PSTN -->IOS Gateway -->CFS -->Unity Connection <-->VPIM <-->Cisco Unity Express	Passed	
UCS712IF.UNC.201	Cisco Unity Express, Unity Connection	Forwarding a Fax message From Unity Connection to Cisco Unity Express over VPIM and back	To verify that a fax received by Cisco Gateway can be relayed to Unity Connection via Cisco Fax Server. And to verify that the Connection subscriber can forward the message to Unity Express over VPIM and Unity Express subscriber can in turn forward back the message to Unity Connection subscriber over a VPIM network.	Fax -->PSTN -->IOS Gateway -->CFS -->Unity Connection <-->VPIM <-->CUE	Passed	
UCS712IF.UNC.206	Unity Connection	Incoming Fax Message to Remote Contact	To verify that Unity Connection can receive a fax for Cisco Unity Express subscriber from CFS and send it to the remote contact.	Fax -->PSTN -->IOS Gateway -->CFS -->Unity Connection <-->VPIM <-->Cisco Unity Express	Passed	
UCS712IF.UNC.206	Messaging Gateway, Unity Connection, Unity Express	Incoming Fax Message to a Remote Contact	To verify that Unity Connection can receive a fax for Unity Express subscriber from CFS and send it to the remote contact.	Fax -->PSTN -->IOS Gateway -->CFS -->Unity Connection <-->VPIM <-->CUE	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF.UNC.301	Cisco Unity Express, Unified Messaging Gateway, Unity Connection	Forwarding Fax Messages from Cisco Unity Express to Cisco Unity Connection Over Unified Messaging Gateway	To verify that a fax received by Cisco Gateway can be relayed to Cisco Unity Express through Cisco Fax Server. And to verify that the Cisco Unity Express subscriber can forward the message to Unity Connection over Unified Messaging Gateway (VPIM) and Unity Connection subscriber can in turn forward the message back to Cisco Unity Express subscriber over a VPIM network with Unified Messaging Gateway in between.	Fax -->PSTN -->IOS Gateway -->CFS -->Cisco Unity Express <-->VPIM <-->UMG <-->VPIM <-->Unity Connection	Passed	
UCS712IF.UNC.302	Cisco Unity Express, Unified Messaging Gateway, Unity Connection	Connection Subscriber Replying to a Fax Message From Cisco Unity Express to Unity Connection Over Unified Messaging Gateway	To verify that a fax received by Unified Messaging Gateway can be relayed to Cisco Unity Express through Cisco Fax Server. And to verify that the Cisco Unity Express subscriber can forward the message to Unity Connection over Unified messaging Gateway (VPIM) and Unity Connection subscriber can in turn reply back to that message.	Fax -->PSTN -->IOS Gateway -->CFS -->Cisco Unity Express <-->VPIM <-->UMG <-->VPIM <-->Unity Connection	Passed	
UCS712IF.UNC.304	Unity Connection	Forwarding a Fax to a System Distribution List in Unified Messaging Gateway	To verify that fax message forwarded to SDL is received by each member in the SDL.	Fax -->PSTN -->IOS Gateway -->CFS -->Cisco Unity Express <-->VPIM <-->UMG <-->VPIM <-->Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF.UNC.304	Messaging Gateway, Unity Connection, Unity Express	Forwarding a Fax to a System Distribution List in Unified Messaging Gateway	To verify that fax message forwarded to SDL is received by each member in the SDL.	Fax -->PSTN -->IOS Gateway -->CFS -->CUE <-->VPIM <-->UMG <-->VPIM <-->Unity Connection	Passed	
UCS712IF.UNC.501	Unity Connection	Fax Call Forwarded to Unity Connection in Connect First Mode-Single Number Fax	To verify that fax message is successfully delivered to the subscriber when the call is forwarded to Unity Connection. The gateway is configured in Connection-first fax detection mode.	Fax -->PSTN -->IOS Gateway -->SMTP -->Unity Connection	Passed	
UCS712IF.UNC.501	Unity Connection	Fax Call Forwarded to Unity Connection in Connect First Mode - Single Number Fax	To verify that fax message is successfully delivered to the subscriber when the call is forwarded to Unity Connection. The gateway is configured in Connection-first fax detection mode.	Fax -->PSTN -->IOS Gateway -->SMTP -->Unity Connection	Passed	
UCS712IF.UNC.504	Unity Connection	Fax Call Disconnect by User-Single Number Fax Connect First Mode	To verify if the fax message transmission to the subscriber is canceled when the call is answered by the phone and is disconnected.	Fax -->PSTN -->IOS Gateway -->SMTP -->Unity Connection	Passed	
UCS712IF.UNC.504	Unity Connection	Fax Call Disconnect by User- Single Number Fax Connect First Mode	To verify if the fax message is successfully delivered to the subscriber when the call is answered by the phone and is disconnected.	Fax -->PSTN -->IOS Gateway -->SMTP -->Unity Connection	Passed	
UCS712IF.UNC.505	Unity Connection	Single Number Fax-Listen First Mode Fax call	To verify if the fax message is successfully delivered to the subscriber when the call is answered by the phone and is disconnected.	Fax -->PSTN -->IOS Gateway -->SMTP -->Unity Connection	Passed	

ID	Features Tested	Case Title	Description	Call Component Flow	Status	Defects
UCS712IF.UNC.505	Unity Connection	Single Number Fax - Listen First Mode - Fax Call	To verify if the fax message is successfully delivered to the subscriber when the call is answered by the phone and is disconnected.	Fax -->PSTN -->IOS Gateway -->SMTP -->Unity Connection	Passed	
UCS713-WIR-002	Wireless Unified IP Phone 9971	WiFi connected Unified IP Phone 9971 Remote Site (WAN Failure)	Verify the Wan failure on Unified SRST/H-REAP.	Phone->AP->LWAPP/WAN->WLAN controller->Unified CM	Passed	
UCS713-WIR-002	Wireless	WiFi Connected Unified IP Phones in Remote Site	Verify H-REAP Access Point failure in remote Unified SRST site.	IP Phone->Access Point->LWAPP/WAN->Wireless LAN Controller->Unified Communications Manager	Passed	
UCS713-WIR-003	Wireless Unified IP Phone 9971	WiFi Unified IP Phone 9971 Central Site	Verify forced roam due to Access Point failure.	Phone->AP->WLAN controller->Unified CM	Passed	
UCS713-WIR-008	Unified IP Phone 9971 (WiFi Connected)	WiFi Connected Unified IP Phone 9971 Central Site (Authentication)	Verify that Unified IP Phone 9971 in central site can perform TKIP/AES authentication with CCKM.	Phone->AP->LWAPP/WAN->WLAN controller->Unified CM	Passed	

Project Features Tested	Total Test Cases		Passed		Pass w/ X		Failed	
	Number	% of Total	Number	% Pass	Number	% Pass w/ X	Number	% Failed
NA IPT Regression	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 UC 8.0(2)	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%

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