ID	Title	Description	Call Component Flow	Status	Defects
CO10.CCM.081	Single Number Reach Interoperability with Cisco Unity Voice Mail	Verify the ability to operate a single voice mail box for each Single Number Reach user.	Unified CallManager->, Phone->, SNR server->, Unity voicemail	Passed	
CO10.CCM.082	Single Number Reach Interoperability with Cisco Unified MeetingPlace	Verify the ability to attend a Cisco Unified MeetingPlace meeting and exchange the call between a mobile phone and a desk phone during the meeting.	Unified CallManager->, MeetingPlace, Phone->, SNR server->	Passed	
CO10.CCM.086	Join Conference from Single Number Reach Desk Phone	Verify the ability to join a conference call from a Single Number Reach desk phone.	Unified CallManager->, Phone, Phone-> (SNR)	Passed	
CO10.CCM.087	Join Conference from Single Number Reach Mobile Phone	Verify the ability to join a meeting from a mobile phone in an Single Number Reach group (requires enterprise dial tone feature for access).	Unified CallManager->, cell Phone->, meeting, SNR server-> (IVR)	Passed	
CO10.CCM.102	Intra-Cluster Call from Analog Phone on VG248 to SCCP Phone with Shared Line	Verify that an analog phone connected to a VG248 gateway can make a call to an SCCP phone with shared lines.	Analog Phone->VG248->Unified CallManager->SCCP (shared-line, group-pickup)->Transfer->Unified CallManager->VG224->Analog Phone.	Passed	
CO10.CCM.158	PSTN-to-SIP Intra-Cluster Call over Nortel DMS-100 Trunk Released by Calling Party	Verify the ability to make a successful basic intra-cluster call over a PSTN trunk configured for Nortel DMS-100 signaling to a SIP phone that is then released by the calling party.	Analog Phone->DMS100->PRI->CMM->CMM->SIP Phone.	Passed	
CO10.CCM.159	PSTN to SIP Intra-Cluster Call over AT&T 5ESS Trunk Released by Called Party	Verify the ability to make a successful basic intra-cluster call over a PSTN trunk configured for AT&T 5ESS signaling to a SIP phone that is then released by the called party.	Analog Phone->5ESS->PRI->CMM->CMM->SIP Phone.	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO10.CCM.180	Other Group Pickup Multiple Calls	Verify the ability to answer the longest unanswered call with one key or answer another call in the other pickup group by using a softkey.	SIP Phone A->Unified CallManager1->Unified CallManager2->SIP Phone B->(Group Pickup) SIP Phone D	Passed	
CO10.CCM.195	FXS-to-SIP Encrypted Phone Call	Verify the ability to make a call from a foreign exchange station (FXS) phone to an encrypted SIP phone.	Analog Phone VG224->Unified CallManager->SIP Phone	Passed	
CO10.CCM.196	PSTN-to-SIP Encrypted Phone Call	Verify the ability to make a call from a PSTN line to an encrypted SIP phone.	Analog Phone->PSTN->Unified CallManager->SIP Phone	Passed	
CO10.CCM.198	Call Forward All Inter-Cluster Calls Between Encrypted SIP Phones	Verify the ability to call forward all (CFA) inter-cluster calls between encrypted SIP phones across a H.323 ICT.	SIP Phone 1(encrypted)->Unified CallManager->ICT->Unified CallManager->SIP Phone->(encrypted)->CFA->SIP Phone 4.	Passed	
CO10.CCM.201	Blind Transfer Inter-Cluster Calls Between Encrypted SIP Phones	Verify the ability to perform a blind transfer on inter-cluster calls between encrypted SIP phones across a H.323 ICT.	SIP Phone 1-> Unified CallManager->ICT->Unified CallManager->SIP Phone 2->blind Transfer->SIP Phone 3	Passed	CSCse19135
CO10.CCM.202	Consult Transfer Inter-Cluster Calls Between Encrypted SIP Phones	Verify the ability to perform a consultative transfer on inter-cluster calls between encrypted SIP phones across a H.323 ICT.	SIP Phone 1-> Unified CallManager->ICT->Unified CallManager->SIP Phone 2->consult Transfer->SIP Phone 3	Passed	
CO10.CRS.000	H.323 Gateway Call Transferred to Unified Contact Center Express Server in Different Cluster	Verify a successful call from an H.323 gateway to an IP phone that has call pickup activated, which is then consultative transferred to a Unified Contact Center Express server in a different Unified CallManager cluster and received by an available SIP TNP agent who parks the call.	H.323 Gateway->Unified CallManager->IP Phone (call pickup)->IP Phone->(consult transfer)->Unified CallManager->Gatekeeper ICT->Unified CallManager->CRA->SIP TNP Agent->(call park)->Unified CallManager->Gatekeeper ICT->Unified CallManager (call park DN)->IP Phone	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO10.CRS.004	PSTN MGCP Gateway Call to SIP TNP Agent Conferencing with Remote Cluster	Verify the ability to complete a call from a MGCP gateway to any available SIP TNP agent served by a Unified Contact Center Express server in a different Unified CallManager cluster. The SIP TNP agent conferences in a remote cluster directory number, consults with the remote party and the caller, and then continues the call with the caller alone.	EP->MGCP Gateway->Unified CallManager->ICT->Unified CallManager->CRA->SIP TNP Agent->(conference)->Unified CallManager->ICT->IP DN	Passed	
CO10.CRS.014	Call from Cisco Unity Auto Attendant Transferred to Customer Response Solution SIP TNP Agent	Verify the ability to complete a call from a Cisco Unity auto attendant this is transferred to a SIP TNP agent.	cm->crs->SIP TNP Agent., Phone->unity aa->	Passed	
CO10.CTP.000	Centralized TFTP interoperability with current and previous release of Centralized TFTP servers with SIP Legacy phones	Multi cluster Centralized TFTP server with mixed Unified CallManager versions Clusters where Unified CallManager 4.1 is the master Centralized TFTP server		Passed	
CO10.CTP.002	Centralized TFTP interoperability with current and previous release of Centralized TFTP servers with SCCP phones	Multi cluster Centralized TFTP server with mixed Unified CallManager versions Clusters where Unified CallManager 4.1 is the master Centralized TFTP server		Passed	
CO10.CTP.004	Centralized TFTP failover to backup Local TFTP server with SCCP Phones	Multi cluster Centralized TFTP server with mixed Unified CallManager 5.0 and 4.1 Clusters where Unified CallManager 4.1 is the master Centralized TFTP server. Fail the primary Unified CallManager 5.0 Local TFTP server while Master Centralized TFTP server is still available.		Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO10.FAM.004	T.38 Fax Relay Through IOS Gateway to Cisco Fax Server	Verify the ability to send a T.38 fax through an IOS gateway to a Cisco Fax Server using SIP for signaling and T.38 for media.	Cisco Fax server, FaxLab->, IOS gateway->	Passed	
CO10.INT.004	Analog-to-IP Call from Unified CallManager Release 4.1.x SCCP Analog Phone to Unified CallManager Release 5.0 SCCP Phone over H.323 ICT	Verify the ability to make and answer an audio call from a Unified CallManager Release 4.1.x SCCP analog phone connected to a VG224 to a Unified CallManager Release 5.0 SCCP phone over an H.323 inter-cluster trunk.	Phone 1->Unified CallManager->ICT ->Unified CallManager->Phone2	Passed	
CO10.INT.011	IP-to-IP Inter-Cluster Call with Hardware Transcoding on IOS Gateway	Verify the ability to make an audio call from a Unified CallManager Release 5.0 SCCP phone to a Unified CallManager Release 4.1.x SCCP phone over a gatekeeper-controlled H.323 inter-cluster trunk with hardware transcoding performed by a IOS gateway in the Unified CallManager Release 4.1.x cluster.	Phone 1->Unified CallManager->ICT ->Unified CallManager->Phone2	Passed	
CO10.IPP.046	AutoRegister over TCP	Verify that a phone continually attempts to auto-register until successful.	TNP SIP Phone->Unified CallManager	Passed	
CO105.CUP.001	Bulk list of buddies in multiple IP phones and check the phone status on SCCP phones.	Configure bulk list of buddies(20) on preferably 6 real phones and check the Status of the phones are correctly reflected.	Phones--<html>--Unified CallManager	Failed	CSCse03676
CO105.CUP.002	Bulk list of buddies in multiple IP phones and check the phone status on SIP phones.	Configure bulk list of buddies(20) on preferably 6 real SIP legacy and TNP IP phones check the presence status of the buddies are correctly reflected.	Phones--<html>--Unified CallManager	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO105.CUP.003	Same buddies on multiple IP phones, check the phone presence status.	Configure 5 buddies Executive buddies) on preferably 1000 simulated phones and 2 real phones and check the Status of the executive buddies are correctly reflected in Very Large Campus with Clustering over the WAN site-RFD cluster	Phones--<html>--Unified CallManager	Passed	CSCse03676
CO105.CUP.004	Same buddies on multiple IP phones, check the phone presence status.	Configure 5 buddies Executive buddies) on preferably 1000 simulated phones and 2 real phones and check the Status of the executive buddies are correctly reflected in Large SIP site cluster	Phones--<html>--Unified CallManager	Passed	
CO105.CUP.005	Bulk list of buddies, check the Instant Messager status on multiple IP phones behind SRST Gateway in SJC cluster.	Configure bulk list of buddies on preferably 4 real phones behind SRST GW and check the presence status of the phones are correctly reflected.	Phones--<html>--Unified CallManager	Passed	
CO105.CUP.006	Mixed list of buddies like video phones, Secure IP phones, and analog end points check the Instant Messager status on multiple IP phones in SJC-RFD cluster	Configure H323, SCCP video IP phones, secure IP phones are as buddies in 4 SCCP legacy and TNP phones in Very Large Campus with Clustering over the WAN site cluster, check for presence status.	Phones--<html>--Unified CallManager	Passed	
CO105.CUP.007	Mixed list of buddies like video phones, Secure SIP phones, check the Instant Messager status on multiple IP phones in DFW cluster	Configure SIP video IP phones, secure SIP phones are as buddies in 4 SIP legacy and TNP phones in Large SIP site cluster, check for presence status.	Phones--<html>--Unified CallManager	Passed	
CO105.CUP.008	To check for different Presence CSS	To check for different Presence CSS		Failed	CSCse22635
CO105.CUP.009	To check for different Presence CSS involving extension mobility	To check for different Presence CSS involving extension mobility in Large SIP site cluster		Failed	CSCse22635

ID	Title	Description	Call Component Flow	Status	Defects
CO105.CUP.016	To check for new Instant Messager indication on a TNP SIP phone	To check for new IM indication on a TNP SIP phone	Phones->CUPS->-Unified CallManager.	Passed	
CO105.CUP.017	To check Cisco Unified Presence Server Admin can broadcast Instant Messager to bulk number of phones	To check the Unified Presence Server admincan can broadcast IM to bulk number of phones to few hundreds of phones.	Phones->CUPS->-Unified CallManager.	Failed	CSCse24102
CO105.CUP.018	To check Instant Messager between IP phones behind SRST Gateway and SIP TNP ip phone.	To check IM between IP phones behind SRST GW and SIP TNP ip phone.	Phones->CUPS->-Unified CallManager.	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CE10.CME.101	Call from Unified CallManager Controlled MGCP Gateway to Local Unified CallManager Express Phone can be Forwarded to Local Unified CallManager Express Phone and Forwarded on No Answer to Unified CallManager Cisco IP SoftPhone Which Forwards Call Back to the Local Unified CallManager Express Phone	Verify that a call from a Unified CallManager controlled MGCP Gateway to a local Unified CallManager Express phone can be call forwarded to a local Unified CallManager Express phone and forwarded on no answer (CFNA) to a Unified CallManager IP SoftPhone which forwards the call back to the local Unified CallManager Express phone.	Phone 1->MGCP Gateway->Unified CallManager->Unified CallManager Express 1->Unified CallManager Express->Phone 1->Unified CallManager->Unified CallManager Express->Phone 1	Passed	
CO10.INT.003	IP-to-IP Call from Unified CallManager Release 4.1.x SCCP Phone over SIP ICT to Unified CallManager Release 5.0 SIP Phone Forwarded to Another Extension and Voice Mail	Verify the ability to make and answer an audio call from a Unified CallManager Release 4.1.x SCCP IP phone over a SIP ICT to a Unified CallManager Release 5.0 SIP phone which forwards all calls (CFA) to another extension and then to voice mail.	Phone 1->Unified CallManager->ICT ->Unified CallManager->Phone2	Passed	
CO10.IPP.024	Call Forward on Busy for Inter-Cluster Call from 3rd Party SIP Phone (Polycom) to SIP Cisco Unified IP Phone	Verify that an inter-cluster call from a 3rd party SIP phone (Polycom) can be call forwarded on busy (CFB) to a SIP Cisco Unified IP Phone.	Cisco SIP Phone->Unified CallManager->3rd Party SIP Phone->Call Forward->ICT->Cisco SIP Phone	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CE10.CME.156	Call from Unified CallManager Controlled SIP Gateway to Local Unified CallManager Express Phone can be Transferred to Remote Unified CallManager Express Phone and Forwarded on Busy or Always Forwarded to Remote Unified CallManager Express Phone	Verify that a call from a Unified CallManager controlled SIP gateway to a local Unified CallManager Express phone can be transferred to a remote Unified CallManager Express phone and call forwarded on busy (CFB) or always forwarded (CFA) to a remote Unified CallManager Express phone.	SIP Phone->Unified CallManager->Unified CallManager Express->?transfer->Unified CallManager Express->? Unified CallManager3->Phone	Passed	
CE10.CME.138	Call from Unified CallManager Registered H.323 Phone to Local Unified CallManager Express Phone can be Consultative Transferred to Unified CallManager Controlled SIP Phone	Verify that a call from a Unified CallManager registered H.323 phone to a local Unified CallManager Express phone can be consultative transferred to a Unified CallManager controlled SIP phone.	Registered H.323 Phone->Unified CallManager->Unified CallManager Express->Consult Transfer->Unified CallManager->SIP Phone	Passed	
CO10.IPP.007	Intra-Cluster Call from 3rd Party SIP Phone to SIP Cisco Unified IP Phone with Blind Transfer	Verify the ability to make an intra-cluster call from a 3rd party SIP phone to a SIP Cisco Unified IP Phone which performs a successful blind call transfer to another SIP Cisco Unified IP Phone; confirm that the secondary call gets answered.	3rd Party SIP Phone->Unified CallManager->SIP Phone->Blind Trfr->Cisco SIP Phone	Passed	
CO10.IPP.038	Intra-Cluster Call from 3rd Party SIP Phone (Sipura) to SIP Cisco Unified IP Phone with Blind Transfer over ICT	Verify the ability to make a call from a 3rd party SIP phone (Sipura) over an inter-cluster trunk to a SIP Cisco Unified IP Phone which performs a successful blind call transfer to another SIP Cisco Unified IP Phone; confirm that the secondary call gets answered.	3rd Party SIP Phone->Unified CallManager->SIP Phone->Blind Trfr->ICT->Cisco SIP Phone	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO10.CER.020	Call Back from PSAP Using ELIN Reaches Original Emergency Caller	Verify that a public safety answering point (PSAP) can call back the original caller in case the emergency call gets disconnected.	PSAP->Unified CallManager->CER->911 Caller	Passed	
GB31.ER.276	Cisco Emergency Responder Phone Discovery and IP-to-IP Cisco Emergency Responder Calls	Cisco Emergency Responder phone discovery and IP-to-IP Cisco Emergency Responder.		Passed	
GB31.ER.282	IP-to-IP Intra-Cluster Cisco Emergency Responder Calls using Extension Mobility	IP - IP intra-cluster Cisco Emergency Responder calls using extension mobility.		Passed	
GB40.ER.009	IP-to-IP Intra-Cluster Cisco Emergency Responder Calls using Authenticated Phones	Verify that Cisco Emergency Responder can properly discover phones that have been configured for authentication and that calls are routed to the PSAP with proper automatic number identification.		Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CE10.CME.183	Ad Hoc Conference Set Up by Local Unified CallManager Express Phone with Unified CallManager Controlled H.323 Gateway and Unified CallManager Controlled SIP Phone	Verify that an ad hoc conference can be established by a local Unified CallManager Express phone with a Unified CallManager controlled H.323 gateway and a Unified CallManager controlled SIP phone.	Stage 1: Phone->? Unified CallManager Express->Unified CallManager->H.323 Gateway->Phone 2, Stage 2: Phone->? conference softkey->Unified CallManager Express->Unified CallManager->SIP Phone, Stage 3: Phone->? conference softkey->Phone->and SIP Phone	Passed	
CE10.CME.222	Ad Hoc Conference Set Up by Local Unified CallManager Express Phone with Unified CallManager Controlled SIP Phone and Another Unified CallManager Controlled SIP Phone	Verify that an ad hoc conference can be established by a local Unified CallManager Express phone with a Unified CallManager controlled SIP phone and another Unified CallManager controlled SIP phone.	Stage 1: Phone->? Unified CallManager Express->Unified CallManager->SIP Phone, Stage 2: Phone->? conference softkey->Unified CallManager Express->Unified CallManager->SIP Phone 2, Stage 3: Phone->? conference softkey->SIP Phone and SIP Phone 2	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
GB31.CM.157	Extension Mobility	IP to IP Intra-cluster Extension Mobility.	IP to IP Intra-cluster Extension Mobility.		Passed

ID	Title	Description	Call Component Flow	Status	Defects
CO10.CER.010	ERL Reconfiguration While Secondary Cisco Emergency Responder was Active Gets Discovered by Restored Primary	Verify that an emergency response location (ERL) reconfiguration which occurred while the secondary Cisco Emergency Responder was active is discovered when the primary becomes active again.	CER Admin->ERL Reconfiguration->Failover Primary	Passed	
CO105.CUP.020	To check after failure the recovery of the Cisco Unified Presence Server is successful	Verify after failure the recovery of the Unified Presence Server is successful	Not Applicable	Failed	CSCse20572

ID	Title	Description	Call Component Flow	Status	Defects
CO10.CME.010	Blind Conference Between Unified CallManager and Unified CallManager Express which Forwards All Calls to Another Unified CallManager Express Without Tandem Gateway	Verify the ability to pre-answer a conference call (press the Conference softkey before the called party answers the call) that is originated by Unified CallManager to a Unified CallManager Express that forwards all calls (CFA) to another Unified CallManager Express without a tandem gateway.	SIP IP Phone->Unified CallManager->IP Phone->Unified CallManager Express->-Unified CallManager Express->IP Phone	Passed	
CO10.CME.011	Blind Conference Between Unified CallManager and Unified CallManager Express which Forwards All Calls to Another Unified CallManager Express with Tandem Gateway	Verify the ability to pre-answer a conference call (press the Conference softkey before the called party answers the call) that is originated by Unified CallManager to a Unified CallManager Express that forwards all calls (CFA) to another Unified CallManager Express using a tandem gateway.	SIP IP Phone->Unified CallManager->IP Phone->Gateway->Unified CallManagerE->Unified CallManagerE->-IP Phone	Passed	
CO10.CME.018	Call on Secure/Non-Secure H.225 Trunk Between Unified CallManager and Unified CallManager Express Phones	Verify the ability to make a call between a Unified CallManager phone and Unified CallManager Express phone over both secure and non-secure H.225 trunks.	SIP IP Phone->Unified CallManager->H.225 Trunk->-Unified CallManagerE->sccp IP Phone	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO10.CCM.119	Inter-Cluster SCCP Phone Call to Hunt List with Forward on No Answer to Cisco Unity Voice Mail	Verify that a call made from a SCCP phone to a hunt list pilot number (configured with call forward no answer) is successfully forwarded to a Cisco Unity voice mail when no member of the hunt list answers the call and the timer expires.	SCCP Phone 1->Unified CallManager->ICT->Unified CallManager->Hunt Pilot->Hunt list (SCCP Phone 2, SCCP Phone 3)->timeout->Forward Hunt No Answer->Unified CallManager->Unity Voice Mail	Passed	
CO10.CCM.122	PSTN Phone Call to Hunt List with Forward Hunt Busy to SCCP Phone	Verify that a call made from a PSTN phone to a hunt list pilot number (configured for call forward busy) is successfully forwarded to a destination SCCP phone configured when all members of the hunt list are busy.	FXS Phone->PSTN->Gateway->Unified CallManager->Hunt Pilot->Hunt list (SCCP Phone 2, SCCP Phone 3)->Forward Hunt Busy->Unified CallManager->ICT->Unified CallManager->SCCP Phone 1.	Passed	
CO10.CCM.210	Hunt List Forward to Route Pattern with FAC	Verify the Unified CallManager server's ability to process an inbound call to a hunt list with Broadcast distribution, then hunt list forward the call to a route pattern with a forced authorization code (FAC).	SIP Phone->Unified CallManager->SIP Phone->CFB->Unified CallManager/HL->PSTN Phone	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO105.CUP.022	Rebooting of Primary Cisco Unified Presence Server with large number of active subscriptions with IPSec link to CCM	Setup IPSec between Unified CallManager to Unified Presence Server for Database and PE/Proxy. Setup around large buddy list in 2 real phones and engage them in a call. Reboot the Primary Unified Presence Server under this situation.	Not Applicable	Failed	CSCse20572

ID	Title	Description	Call Component Flow	Status	Defects
CO10.CCM.039	40,000 BHCA in Unified CallManager	Verify that 99.99% calls complete at 40,000 busy hour call attempts (BHCA).		Failed	CSCse26087
CO105.CUP.011	Bulk IPPM traffic to multiple real IP phones.	Generate bulk IPPM traffic from simulated Phones to preferably 10 real IP phones	Unified CallManager->Unified Presence Server	Passed	
CO105.CUP.012	Bulk IPPM traffic for long duration involving multiple phones .	Generate bulk IM traffic involving preferably 100 simulated phones distributed over multiple DPs.	Unified CallManager->Unified Presence Server	Passed	
CO105.CUP.013	Bulk generation of buddy list large number of sessions between CCM and Cisco Unified Presence Server.	This is for testing high volume of sessions established between Unified CallManager and Unified Presence Server	Unified CallManager->Unified Presence Server	Passed	
CO105.CUP.014	Bulk addition deletion of buddy list.	From 10 real phones add 200 buddies total and delete them. Repeat this exercise for preferably 10 times	Unified CallManager->Unified Presence Server	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO10.IPP.049	SIP Phone Early Attended Transfer with Music on Hold	Verify the ability of a SIP phone to perform an early attended transfer with Music on Hold (MoH).	TNP SIP Phone->SIP Phone->Transfer->SIP Phone	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO10.NME.008	Alerts and Activities Display, Case 1	Existing AAD functionality will continue to be available. All operations under the topology display can be launched under the tools pulldown menu. The event property detail launched under the "Alert Detail View" shall show the treshold value and the latest polled value.	Not Applicable	Passed	
CO10.NME.033	Phone Reachability Testing, Case 2	Verify Phone Reachability Test in operational mode. Users will pick phones from a phone report and include them to be monitoring for reachability.	Not Applicable	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO10.QOS.000	Transfer Failure Due to Not Enough Bandwidth Condition when Policy is Mandatory	Verify that call transfers do not complete when the RSVP policy is mandatory and not enough bandwidth is available for the called party.	A calls->B, B transfer->C	Passed	
CO10.QOS.002	Under Limited Bandwidth Condition Voice Calls Succeed But Video Calls Fail	Verifies how under limited bandwidth condition where there is only enough bandwidth for voice, and when the policy is Mandatory Video Desired, the voice call succeed and the video call fails.	A -video-call-->B,->-video-Conference-> C	Passed	
CO10.QOS.004	RSVP Reservation retries after initial Failure due to lack of bandwidth	Verifies the retry feature in a complex conference setup.	A -Conference-> B,->-Conference-> C, A -Conference-> D,->-Conference-> E, A -Conference-> F,->-Conference-> G	Passed	
CO10.QOS.005	Reservations will be installed for Shared Lines and torn down the ones Not used after the Call is accepted by one of the Endpoints	Verify that RSVP reservations are installed and torn properly.	A -call-> B,->-Blind Transfer-> (C, D, E, F -answer the call, F)	Passed	
CO10.QOS.015	Call diversion using AAR when reservation fails.	Calls will be diverted to AAR when the reservation fails	A -call-> B	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO10.CCM.066	Analog Phone to SIP Phone Shared Line, Hunt Group, Consultative Transfer, Blind Transfer, and Call Forward No Answer to Cisco Unity Voice Mail	Verify hunt groups, consultative and blind transfers, and call forward on no answer (CFNA) to a Cisco Unity voice mail box for a SIP phone with shared lines.	EP->VG 224/248->Unified CallManager->SIP Phone->(shared-line, hunt group)->consult Transfer->SIP Phone->(non-shared-line)->blind Transfer->SIP Phone 3->CFNA->Unity voice mail.	Passed	
CO10.CCM.103	SCCP Phone Shared Line (Multiple Phones) Hunt Group, Call Park, and Call Pickup Features	Verify hunt group, call park and call pickup features on a shared line across SCCP phones.	call pickup), hunt group)->blind transfer->Gatekeeper ICT->SCCP Phone 4 (shared-line)->hold->resume->call park->SCCP Phone5 (shared-line, SCCP Phone 1-> ICT->SCCP Phone->(shared-line	Passed	
CO10.CCM.111	SCCP Phone Shared Line Group Pickup of Multiple Calls	Verify group pickup of multiple calls on a SCCP phone with shared lines.	SCCP Phone 1->Unified CallManager->ICT->Unified CallManager->SCCP Phone->(shared line)->CFA->SCCP Phone 3->group pickup SCCP Phone 4 (non-shared-line)., While call is active SCCP Phone->calls Phone 3/4 shared-line and the call is picked up by SCCP Phone 4 (shared-line). Phone 4 CC Phone->and Phone 2.	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO10.CCM.138	16 SRST Sites (Financial Cluster) 1 Hour 6K BHCA Load Run	Generate traffic at 6k BHCA for 1 hour across 16 SRST sites. The call flows are as follows: 1. 4-digit Intra-Site Dialing 2. 8-digit Inter-Site Dialing 3. 7-digit or 10-digit Local PSTN Dialing 4. 11-digit Long Distance PSTN Dialing These call flows includes basic, Hold, Blind Transfer, Supervised Transfer, and Conferencing	Unified CallManager Load	Passed	
CO10.CCM.139	48 SRST Sites (Financial Cluster) 1 Hour 18K BHCA Load Run	Generate traffic at 18k BHCA for 1 hour across 48 SRST sites. The call flows are as follows: 1. 4-digit Intra-Site Dialing 2. 8-digit Inter-Site Dialing 3. 7-digit or 10-digit Local PSTN Dialing 4. 11-digit Long Distance PSTN Dialing These call flows includes basic, Hold, Blind Transfer, Supervised Transfer, and Conferencing	Unified CallManager Load	Passed	
CO10.CCM.141	Registration of 100 PSTN MGCP Gateways to Unified CallManager Cluster	Verify that 100 PSTN MGCP gateways, including Cisco Catalyst 6000 series switches with 6608 modules and Communication Media Modules supporting both ISDN PRI and channel associated signaling (CAS) trunks, can successfully register to a very large single-site Unified CallManager cluster.	Unified CallManager->MGCP Gateway	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO10.CCM.045	WebDialer Call from SIP Phone across Gatekeeper ICT to Another SIP Phone in Call Pickup Group, Call Parked and Retrieved from Third SIP Phone	Verifies WebDialer Make and End call features with non-encrypted SIP phone using the following process: WebDialer initiates a non-encrypted call across a gatekeeper ICT to a SIP phone in call pickup group, the call is parked and then retrieved by another SIP phone, and then WebDialer ends the call.	WebDialer->SIP Phone 1->Unified CallManager->Gatekeeper ICT->Unified CallManager->SIP Phone->(call pickup group)->call park->SIP Phone->call retrieve->WebDialer (end call).	Failed	CSCse09848
CO10.CCM.090	WebDialer Call from SIP Phone across ICT to SCCP Phone in Hunt Group, Blind Transfer Call to Another SCCP Phone with Call Forward No Answer to Cisco Unity Voice Mail	Verify WebDialer Make and End call features with a SIP phone using the following process: WebDialer initiates a call from the SIP phone across an ICT to an SCCP phone in a hunt group, and then the call is blind transferred to another SCCP phone configured for call forward on no answer (CFNA) to Cisco Unity voice mail before WebDialer ends the call.	WebDialer->SIP Phone 1->Unified CallManager->ICT->Unified CallManager->SCCP Phone->(hunt group)->blind Transfer->SCCP Phone 3->CFNA->Unity Voice Mail->WebDialer (end call).	Failed	CSCse16204
CO10.CCM.220	Voice Mail after Logging into Unified CallManager Assistant	Verify the ability to leave voice mail after logging into the Cisco Unified CallManager Extension Mobility.	SIP Phone 2->Unified CallManager->SIP Phone->(EM)->CFA->Unity Voice Mail	Failed	CSCse12199

ID	Title	Description	Call Component Flow	Status	Defects
CO10.MPL.001	Audio and Data Conference on Unified MeetingPlace with Mixed Secure, Unsecured, Intra- and Inter-Cluster IP Phones Outdial via SIP Gateway	Verify the ability to hold an audio and application-sharing data conference on Unified MeetingPlace with mixed secure and unsecured SCCP IP phones (dial in) and SIP phones (outdial via SIP gateway) that originate both from within the same cluster and across inter-cluster trunks.	Stage 1: SCCP IP Phone->Unified CallManager->Gatekeeper->Cisco MeetingPlace, Stage 2: SCCP IP Phone->Unified CallManager->Gatekeeper-ICT->Unified CallManager->Gatekeeper->Cisco MeetingPlace, Stage 3: Cisco MeetingPlace->Unified CallManager->SIP IP Phone, Stage 4: Cisco MeetingPlace->Unified CallManager->SIP Trunk->Unified CallManager->SIP IP Phone	Passed	
GB40.MP.025.09	Attend Basic Mixed Audio and Video Meeting: Remote Out-dial, Gatekeeper ICT, SIP MeetingPlace IP Gateway	Verify the ability to attend and control a multi-endpoint mixed audio and video conference with all calls remote via gatekeeper ICT trunk, no encryption, no call treatment, calls originate via outbound dialing using SIP MeetingPlace IP Gateway.	Refer to Original EDCS document	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CE10.MPE.137	Unified MeetingPlace Express Functionality: Cisco IP Communicator User Invokes Meeting Extension, Case 2	Verify that an end user with Cisco IP Communicator can invoke an extension for a meeting with the following participants: low-end, Cisco Catalyst 6000 WS-6624 (MGCP), Unified CallManager Express (H.323), and MGCP gateway.	low-end->SCCP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express; 6624->MGCP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express; Phone->H323->Unified CallManager Express->Gatekeeper->Unified MeetingPlace Express; PSTN->Gateway->MGCP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express	Passed	
CE10.MPE.169	Unified MeetingPlace Express Functionality: Cisco IP Communicator User Initiates Recording, Case 1	Verify that an end user with Cisco IP Communicator can start recording a meeting with the following participants: Cisco Unified IP Phone 7936, video endpoint, Unified CallManager Express (H.323), and H.323 gateway.	7936->Unified CallManager->Gatekeeper->Unified MeetingPlace Express; Video->Unified CallManager->Gatekeeper->Unified MeetingPlace Express; Phone->H323->Unified CallManager Express->Gatekeeper->Unified MeetingPlace Express; PSTN->Gateway->H323->Unified CallManager->Gatekeeper->Unified MeetingPlace Express	Passed	
CE10.MPE.204	Unified MeetingPlace Express Feature Interaction: Calling Line ID Presentation (CLIP)/Caller Name (CNAM), Case 3	Verify the Calling Line ID Presentation (CLIP) for the following meeting participants: high-end, Cisco Unified IP Phone 7920, Cisco VG248 analog gateway (SCCP), Unified CallManager Express (H.323), and H.323 gateway.	high-end->Unified CallManager->Gatekeeper->Unified MeetingPlace Express; 7920->SCCP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express; VG248->SCCP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express; Phone->H323->Unified CallManager Express->Gatekeeper->Unified MeetingPlace Express; PSTN->Gateway->H323->Unified CallManager->Gatekeeper->Unified MeetingPlace Express	Failed	CSCsc88441

ID	Title	Description	Call Component Flow	Status	Defects
CE10.MPE.229	Unified MeetingPlace Express Feature Interaction: Call Forward on Busy, Case 2	Verify a meeting where all participants have the following call forwarding on busy settings: (a) from Cisco Unified IP Phone 7936 to Cisco Unity; (b) from high-end-new to Cisco VG248 analog gateway (SCCP); and (c) from Cisco SIP Proxy Server to MGCP gateway.	part-A-from 7936->Unified CallManager->Gatekeeper->Unified MeetingPlace Express to Unity; part-B-from high-end->Unified CallManager->Gatekeeper->Unified MeetingPlace Express-new to VG248->SCCP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express; part-C-from CSPC->SIP->Unified CallManager-> Gatekeeper->Unified MeetingPlace Express to PSTN->Gateway->MGCP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express	Passed	
CE10.MPE.244	Unified MeetingPlace Express Feature Interaction: Unconditional Call Forward, Case 4	Verify a meeting where all participants have the following unconditional call forwarding settings: (a) from Cisco Unified IP Phone 7936 to Cisco Unity; (b) from Cisco IP Communicator to Cisco VG248 analog gateway (SCCP); and (c) from Unified SRST (SCCP) to SIP gateway.	part-A-from 7936->Unified CallManager->Gatekeeper->Unified MeetingPlace Express to Unity; part-B-from CIPC->SCCP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express to VG248->SCCP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express; part-C-from 79XX->SCCP->SRST/Unified CallManager-> Gatekeeper->Unified MeetingPlace Express to PSTN->Gateway->SIP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO10.UCA.034	Cisco Unified Personal Communicator in softphone mode behind IPSEC VPN listen to VM messages thru Instant MessagerAP (use different domain)	Bring up Unified Personal Communicator behind IPSEC VPN. Click on the message waiting and down load the VM from Voicemail server. Listen to the VM		Passed	
CO10.UCA.035	4 incoming calls to Cisco Unified Personal Communicator in softphone mode almost instantaneously.	Set the Maximum calls to Unified Personal Communicator phone to 4 in Unified CallManager. Call Unified Personal Communicator from different sites across SIP, H323 trunks and local calls almost simultaneously. Check that Unified Personal Communicator won't crash and at least 2 calls are successful in Unified Personal Communicator first Release.		Passed	
CO10.UCA.036	Create a LDAP failure and double click a contact to make a call	Create a LDAP authentication failure. Double click a contact and try to establish a call. Restore it and create a LDAP IP connectivity failure. Double click a contact to establish a call.		Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO10.UCA.037	Isolate the Cisco Unified Presence Server server during a call and check the call last for the duration.	Establish at least 2 calls from 2 Unified Personal Communicator end points, one from softphone and one from phone associate mode. Create a IP connection failure to Unified Presence Server. Check for the status of the buddies after few minutes. Verify that Unified Personal Communicator shows errors a which indicate Unified Presence Server connection failure. Disconnect the call and establish a call from the desk phone associated with Unified Personal Communicator.		Passed	
CO10.UCA.038	Presence: Cisco Unified Personal Communicator in softphone mode, set the presence status of Cisco Unified Personal Communicator to busy and check that it is updated in other Cisco Unified Personal Communicator, and Phone IPPM.	From a Unified Personal Communicator in softphone mode set it's presence status to busy. Verify that other Unified Personal Communicator end points and IPPM phones display the correct status. Establish a call to this Unified Personal Communicator and verify that the Unified Personal Communicator alert the user and a way to decline the call is displayed.		Failed	CSCse35706
CO10.UCA.039	Presence status: Check for Presence of the phone under Received Calls	Check the Presence status of the the buddies which comprises of IPPM phones, Unified Personal Communicators and all are configured in Unified Presence Server. Verify that correct statuses are displayed.		Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO10.UCA.040	Cisco Unified Personal Communicator as a IPCC Express phone	Configure Unified Personal Communicator as one of the IPCC express phone(auto attendant). Establish an incoming call from PSTN to this Unified Personal Communicator. Verify that the PSTN caller hear the IVR prompts and it punch the extension number of Unified Personal Communicator phone. Verify that call is established to Unified Personal Communicator		Passed	
CO10.UCA.041	Cisco Unified Personal Communicator in Softphone mode incoming video call from H323 video end point	Bring up Unified Personal Communicator in Softphone mode. Establish a video call from a H323 video end point from a different cluster. Verify that the call is established and both parties can see the local and remote video. Turn off the video during the call and turn it on again after a minute.		Failed	CSCse30001, CSCse34224
CO10.UCA.042	Cisco Unified Personal Communicator in Softphone mode, merge an incoming Video call from 7985 and out going video call from IP Communicator	Set up a video call from 7985 video phone 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.		Failed	CSCsd74434

ID	Title	Description	Call Component Flow	Status	Defects
CO10.UCA.043	Cisco Unified Personal Communicator in softphone mode merge a outgoing video call to SCCP video endpoint and incoming from H323 video endpoint.	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 H323 video end point thru ICT. Answer the call. Verify that the first call is fed with Music On Hold and no Video. Click on the Merge button and estabish a 3 way conference call. Check for audio and video of this conference call.		Failed	CSCse30001, CSCse34224
CO10.UCA.044	Cisco Unified Personal Communicator in softphone mode behind IPSEC VPN call a CVTA video end point through SIP trunk	Bring up Unified Personal Communicator thru a IPSEC VPN connection. Originate a call from Unified Personal Communicator to a CVTA end point thru SIP ICT. Bring up video during the call. Check the call again for audio and video connection		Passed	
CO10.UCA.045	Cisco Unified Personal Communicator behind IPSEC VPN in softphone mode, Hold and Resume a Video call to CVTA .	Bring up Unified Personal Communicator thru a IPSEC VPN connection. Originate a call from Unified Personal Communicator to a CVTA end point thru 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	
CO105.UCA.001	Unified Personal Communicator in SJC cluster, blind transfer(if supported) an incoming PSTN call to a IP phone in DFW cluster over SIP trunk	Unified Personal Communicator running behind SRST GW NOT in SRST mode, blind transfer an incoming PSTN call to a IP phone in another cluster.	H323->Unified CallManager->Unified Personal Communicator->SIPTrunk	Failed	CSCse08772

ID	Title	Description	Call Component Flow	Status	Defects
CO105.UCA.002	Unified Personal Communicator in RFD Cluster. Set up Conference call with CME SIP and CCM SCCP phones using Unified MeetingPlace Express, sharing of documents.	Unified Personal Communicator is running in RFD cluster. Set up Conference call using MP express, sharing of documents.	Unified Personal Communicator->-Unified CallManager->Unified MeetingPlace Express->Unified CallManager Express->SIPTrunk	Failed	CSCse15895
CO105.UCA.003	Bring up Unified Personal Communicator though IPSEC VPN, this Unified Personal Communicator is one of the phone in hunt group. Answer the call when it is rung.	Unified Personal Communicator is connected though IPSEC VPN, is one of the phone in hunt group. Answer the call when it is rung.		Passed	
CO105.UCA.005	Unified Personal Communicator phone is used as ipccx agent in phone associate mode(sccp phone)	Unified Personal Communicator phone is used as ipccx agent in phone associate mode. Make a call to agent RP to reach an agent	Phone->Unified CallManager->Unified Personal Communicator(ipccx agent)	Passed	
CO105.UCA.006	Presence Status of large number of buddies(~100). Unified Personal Communicator is running in laptop connected IPSEC VPN.	Unified Personal Communicator is running on a windows server which is connected through a IPSEC VPN to the test bed. Add buddy using Active Directory and check the status of buddies.		Passed	CSCse15922
CO105.UCA.007	Presence Status of large number of buddies(~100). Unified Personal Communicator is running in a laptop connected through Cisco VPN Client 3002.	Unified Personal Communicator is running on a laptop which is connected to the test bed through a VPN using Cisco VPN Client 3002. Add buddy using Active Directory and check the status of buddies.		Passed	CSCse15922

ID	Title	Description	Call Component Flow	Status	Defects
CO105.UCA.008	Unified Personal Communicator on a laptop through VPN IPSEC connection. Establish a conference call by gettting two endpoints in different clusters.	Unified Personal Communicator on a laptop through VPN IPSEC connection. Establish a conference call by gettting two endpoints in different clusters. One leg is established through H323 trunk from Unified CallManager and second leg using SIP trunk from Unified CallManager	Unified Personal Communicator->Unified CallManager->SCCP/SIP IP Phone.	Passed	CSCse16081
CO105.UCA.009	Unified Personal Communicator is connected through Cisco VPN Client 3002 connection. Engage in multiparty(5) conference call using Unified MeetingPlace Express, sharing of documents etc.	Unified Personal Communicator in associated mode connected through Cisco VPN client, 3002 VPN connection, set up a conference call using MP express, sharing of documents etc.	Unified Personal Communicator->Unified CallManager->SCCP/SIP IP Phone.	Failed	CSCse15895
CO105.UCA.010	Unified Personal Communicator is connected through IPSEC VPN connection, call forward on NO Answer to intercluster SIP IP phone.	Bring up a Unified Personal Communicator on a windows-xp based platform behind IPSEC VPN, set up CFNA.	Unified Personal Communicator->Unified CallManager->SCCP/SIP IP Phone.	Passed	
CO105.UCA.012	Unified Personal Communicator is set for send to VM and an incoming PSTN call through H323 Gateway is sent to VM.	At Unified Personal Communicator send to voice mail is configured. Check that incoming PSTN call through H323 GW is sent to VM.	Unified Personal Communicator->Unified CallManager->SCCP/SIP IP Phone.	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO105.UCA.014	Answer the incoming PSTN call through SIP trunk at hard phone then transfer the call to a intercluster SIP phone.	An incoming PSTN call though SIP trunk to an Unified Personal Communicator in associated mode. Answer the call at hard phone and move the call to softphone. Check for moving the call back and forth between hard phone and soft phone. Finally from the hardphone, blind transfer to a intercluster SIP end point. Check for correct display of caller-id, relaying of DTMF digits, and satisfactory establishment of bearer path.	Unified Personal Communicator->Unified CallManager->SCCP/SIP IP Phone.	Failed	CSCsd96312
CO105.UCA.018	Check the Presence Status of the bulk number of buddies on Unified Personal Communicator	From the Search list pick 75 buddies into a group. Make the end points to go offhook and onhook and check that the presece status at Unified Personal Communicator for those end points are updated. Check for the incoming NOTIFY messages from Unified Personal Communicator.		Failed	CSCse35706
CO105.UCA.019	Unified Personal Communicator talking to Active directory using LDAP, Active Directory is running at SJC and Unified Personal Communicator at RFD, click to dial a contact.	Get a contact using LDAP and click to dial	Unified Personal Communicator->LDAP->Unified CallManager	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO105.UCA.021	Unified Personal Communicator running behind SRST Gateway come back to active state after WAN failure	An Unified Personal Communicator is running behind a SRST GW. Establish a incoming call to Unified Personal Communicator. Create a WAN failure. Verify that Unified Personal Communicator application tear down the call. Bring up the WAN connection and setup the call again		Passed	
CO105.UCA.022	Unified Personal Communicator interacting with VM	Unified Personal Communicator is connected through IPSEC VPN. Dial the VM and listen for Voice Mail and delete the messages.		Passed	
CO105.UCA.023	Secure TLS links to Cisco Unified Presence Server.	Unified Personal Communicator's links to Unified Presence Server thru TLS. CEO Unified Personal Communicator is in phone associated mode. Bring up the Unified Personal Communicator		Passed	
CO105.UCA.024	Secure TLS links to Cisco Unified Presence Server, Unified Personal Communicator behind IPSEC VPN connection	Unified Personal Communicator's links to Unified Presence Server thru TLS. CEO Unified Personal Communicator is in soft phone mode. Bring up the Unified Personal Communicator		Passed	
CO105.UCA.025	Unified Personal Communicator in phone associated mode behind a VPN concentrator CVPN 3002, and Unified Personal Communicator is a MLPP subscriber.	Unified Personal Communicator's links to Unified Presence Server thru TLS, The Unified Personal Communicator is a CEO number and it's status is monitored by around 1000 users. The Unified Personal Communicator is behind a hardware CVPN 3002. Unified Personal Communicator is in a MLPP domain. Bring up the Unified Personal Communicator and make a call.		Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO10.CCM.007	Multiple Line Registration from H.323 Gateway During SRST Fallback	Verify multiple line registration from a H.323 gateway to an SRST gateway.	Phone->SRST Gateway->SIP Phone, Phone->Unified CallManager->SIP Phone	Passed	
CO10.CCM.016	Secure SIP Phones Behind Unified SRST Gateway	Verify that secure SIP endpoints can register with a Unified SRST gateway.	Phone->SRST Gateway->SIP IP Phone, Phone->Unified CallManager->SIP Phone	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO10.VID.001	SCCP Non-Video to SCCP Video Call Transfer to SCCP Video Endpoint Inter-Cluster H.323 Gatekeeper Controlled	Verify a call transfer from SCCP non-video endpoint, transferring a SCCP Video endpoint to another SCCP Video Capable endpoint inter-cluster H.323 gatekeeper Controlled	Stage 1: SCCP Audio->Unified CallManager->SCCP Video, Stage2: SCCP Audio Transfer->Unified CallManager->ICT->Unified CallManager->SCCP Video	Passed	
CO10.VID.002	SCCP Non-Video to SCCP Video Call Transfer to SCCP Video Endpoint Inter-Cluster SIP Trunk	Verify a call transfer from SCCP non-video endpoint, transferring a SCCP Video endpoint to another SCCP Video Capable endpoint inter-cluster SIP Trunk	Stage 1: SCCP Audio->Unified CallManager->SCCP Video, Stage2: SCCP Audio Transfer->Unified CallManager->ICT->Unified CallManager->SCCP Video	Passed	
CO10.VID.005	SIP Non-Video to SCCP Video Call Transfer to SCCP Video Endpoint Inter-Cluster SIP Trunk	Verify a call transfer from SIP (TNP) non-video endpoint, transferring a SCCP Video endpoint to another SCCP Video Capable endpoint, inter-cluster SIP Trunk	Stage 1: SIP Audio->Unified CallManager->SCCP Video, Stage2: SIP Audio Transfer->Unified CallManager->ICT->Unified CallManager->SCCP Video	Passed	
CO10.VID.013	SIP Non-Video to H.323 Video Call Transfer to H.323 Video Endpoint Inter-Cluster H.323 Gatekeeper Controlled	Verify a call transfer from SIP (TNP) non-video endpoint, transferring a H.323 Video endpoint to a H.323 Video endpoint, inter-cluster H.323 gatekeeper Controlled	Stage 1: SIP TNP Audio->Unified CallManager->H.323 Video, Stage2: SIP TNP Audio Transfer->Unified CallManager->ICT->Unified CallManager->H.323 Video	Failed	
CO10.VID.031	SCCP Tandberg/7985 to H.323 Tandberg Video Call with H.264 codec Inter-Cluster SIP Trunk	Video call using SCCP 7985 and SCCP Tandberg video endpoints to H.323 Tandberg Video endpoints H.264 codec, inter-cluster SIP Trunk	SCCP Tandberg/7985->Unified CallManager->ICT->Unified CallManager->H.323 Endpoints	Passed	
CO10.VID.033	H.264 Video calls between H.323 Endpoints across SIP Trunks	Intercluster IP-IP Video call between H.323 Video endpoints across SIP Trunks	H.323 Video Endpoint->Gatekeeper->Unified CallManager->SIP->Unified CallManager->Gatekeeper->H.323 Video Endpoint	Passed	
CO10.VID.036	H.323 Tandberg to Polycom VSX Gatekeeper Controlled, Inter-Cluster SIP Trunk	H.323 Tandberg T1000 Video endpoint to Polycom VSX 7000 Video endpoint with gatekeeper control and utilizing dynamic H.323 addressing feature, inter cluster SIP Trunk	H.323 Tandberg T1000->Gatekeeper->Unified CallManager->ICT->Unified CallManager->Gatekeeper->Polycom VSX 7000	Passed	

ID	Title	Description	Call Component Flow	Status	Defects
CO10.VID.045	SCCP Video Endpoint to H.323 Video Endpoint Call Transfer to SCCP Video Endpoint, Inter-Cluster SIP Trunk	Call transfer of a H.323 Video Endpoint between SCCP Video Endpoints inter-cluster SIP Trunk	Stage 1: SCCP Video Endpoint->Unified CallManager->Gatekeeper->H.323 Video Endpoint, Stage2: SCCP Video Endpoint->Transfer->Unified CallManager->ICT->Unified CallManager->SCCP Video Endpoint	Passed w/ Exception	

ID	Title	Description	Call Component Flow	Status	Defects
CO10.UNC.012	Inter-Site Message Delivery, Retrieval, Deletion and MWI from SIP and SCCP Endpoint to Cisco Unity Connection PDL over Dual Integration with G.729 Codec	Verify that Cisco Unity Connection can support message delivery, retrieval, and deletion when calls are made between Unified CallManager cluster and Cisco SIP Proxy Server using a SIP and SCCP phones over a SIP integration.	SCCP Phone->Unified CallManager->SIP Trunk->->CSPS->SIP Phone->Unity Connection	Passed	
CO10.UNC.024	Calls Forwarded to Cisco Unity Connection Through Intercluster Trunk (SCCP Integration)	Verify that calls forwarded to Cisco Unity Connection through an intercluster trunk are successful.	SCCP Phone->Unified CallManager->ICT->Unified CallManager->SIP Phone->Unity Connection	Passed	
CO10.UNC.027	Direct Calls to Cisco Unity Connection Through Intercluster Trunk (SIP Integration)	Verify that direct calls to Cisco Unity Connection through an intercluster trunk are successful.	SCCP Phone->Unified CallManager->ICT->Unified CallManager->SIP Phone->Unity Connection	Passed	
CO10.UNC.036	Calls Forwarded to Unity Connection Through MGCP Gateway over SIP Integration to Unified CallManager	Verify that calls forwarded to Unity Connection from a PSTN phone through an MGCP gateway are successful.	PSTN Phone->MGCP Gateway->Unified CallManager->SIP Phone->Unity Connection->SIP Phone	Passed	
CO10.UNC.037	Direct calls to Unity Connection Through H.323 Gateway over SIP Integration to Unified CallManager	Verify that calls forwarded to Unity Connection from a PSTN phone through an H.323 gateway are successful.	PSTN Phone->MGCP Gateway->Unified CallManager->SIP Phone->Unity Connection->SIP Phone	Passed	
CO10.UNI.001	Multiple Cluster Integration Outside Caller Call	Verify that Cisco Unity can integrate with Unified CallManager servers in different clusters.	SCCP/SIP Phone->Unified CallManager->Unity->Microsoft Exchange	Passed	
CO10.UNI.012	Inter-Site Message Delivery, Retrieval and Deletion from SCCP Endpoints to Cisco Unity over a SIP Cisco Unity-Cisco SIP Proxy Server Integration Forwarded Call to Voice Mail	Verify that Cisco Unity can support message delivery, retrieval, and deletion when calls are made between Unified CallManager cluster and Cisco SIP Proxy Server using a SCCP phones over a SIP integration.	SCCP Phone->Unified CallManager->SIP Trunk->CSPS-> Unity->Microsoft Exchange->SIP Phone	Failed	CSCse19135

ID	Title	Description	Call Component Flow	Status	Defects
CO10.UNI.054	Internet Mail Access Protocol Client	Verify IMAP client support in Windows and Macintosh.	IP Phone->Unified CallManager->Unity->Microsoft Exchange ->IMAP Client	Passed	
CO10.UNI.073	Message Notification with SIP Integration	Verify that message notification feature works as expected with SIP integration to Unified CallManager.	SIP/SCCP Phone->Unified CallManager->Unity	Failed	CSCsd26874
CO105.UNI.000	Direct calls to Unity through a SIP gateway	Verify that direct calls to Unity from a PSTN through a SIP gateway is successful with the codec type negotiated as OOB NOTIFY.	PSTN Phone->SIP Gateway->SIPT->Unified CallManager->SIPT-> Unity	Failed	CSCse22729
GB31.CX.268	CUE subscriber mail box features, message deposit, retrieve, delete etc.	CUE subscriber mail box features, message deposit, retrieve, delete etc.		Passed	
GB40.CX.001.07	Cisco Unified CallManager Express Overlay: Directory Number and Called Directory Number Display Feature	Verify that when a phone is configured for overlay-directory numbers (DNs), multiple DNs are mapped to a single physical line on a phone.		Passed	

ID	Title	Description	Call Component Flow	Status	Defects
GB31.WL.746	Three-party Ad Hoc Conference Call Between Wireless and Desktop IP Phones Started by Cisco Unified IP Phone 7920 Registered to Unified CallManager Express	Verify the ability to establish a 3-party ad hoc conference call from a Cisco Unified IP Phone 7920 registered to Unified CallManager Express that involves other Unified IP Phone 7920 phones and desktop IP phones.		Passed	
GB31.WL.747	Meet-Me Conference Call Between Wireless and Desktop IP Phones Started by Cisco Unified IP Phone 7920 Registered to Unified CallManager	Verify the ability to establish a Meet-Me conference call from a Cisco Unified IP Phone 7920 registered to Unified CallManager Express that involves other Unified IP Phone 7920 phones and desktop IP phones.		Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CE10.CME.101	Call Forward	Call from Unified CallManager Controlled MGCP Gateway to Local Unified CallManager Express Phone can be Forwarded to Local Unified CallManager Express Phone and Forwarded on No Answer to Unified CallManager Cisco IP SoftPhone Which Forwards Call Back to the Local Unified CallManager Express Phone	Verify that a call from a Unified CallManager controlled MGCP Gateway to a local Unified CallManager Express phone can be call forwarded to a local Unified CallManager Express phone and forwarded on no answer (CFNA) to a Unified CallManager IP SoftPhone which forwards the call back to the local Unified CallManager Express phone.	Phone 1->MGCP Gateway->Unified CallManager->Unified CallManager Express 1->Unified CallManager Express->Phone 1->Unified CallManager->Unified CallManager Express->Phone 1	Passed	
CE10.CME.138	Call Transfer	Call from Unified CallManager Registered H.323 Phone to Local Unified CallManager Express Phone can be Consultative Transferred to Unified CallManager Controlled SIP Phone	Verify that a call from a Unified CallManager registered H.323 phone to a local Unified CallManager Express phone can be consultative transferred to a Unified CallManager controlled SIP phone.	Registered H.323 Phone->Unified CallManager->Unified CallManager Express->Consult Transfer->Unified CallManager->SIP Phone	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CE10.CME.156	Call Forward / Transfer	Call from Unified CallManager Controlled SIP Gateway to Local Unified CallManager Express Phone can be Transferred to Remote Unified CallManager Express Phone and Forwarded on Busy or Always Forwarded to Remote Unified CallManager Express Phone	Verify that a call from a Unified CallManager controlled SIP gateway to a local Unified CallManager Express phone can be transferred to a remote Unified CallManager Express phone and call forwarded on busy (CFB) or always forwarded (CFA) to a remote Unified CallManager Express phone.	SIP Phone->Unified CallManager->Unified CallManager Express->? transfer->Unified CallManager Express->? Unified CallManager3->Phone	Passed	
CE10.CME.183	Conference	Ad Hoc Conference Set Up by Local Unified CallManager Express Phone with Unified CallManager Controlled H.323 Gateway and Unified CallManager Controlled SIP Phone	Verify that an ad hoc conference can be established by a local Unified CallManager Express phone with a Unified CallManager controlled H.323 gateway and a Unified CallManager controlled SIP phone.	Stage 1: Phone->? Unified CallManager Express->Unified CallManager->H.323 Gateway->Phone 2, Stage 2: Phone->? conference softkey->Unified CallManager Express->Unified CallManager->SIP Phone, Stage 3: Phone->? conference softkey->Phone->and SIP Phone	Passed	
CE10.CME.222	Conference	Ad Hoc Conference Set Up by Local Unified CallManager Express Phone with Unified CallManager Controlled SIP Phone and Another Unified CallManager Controlled SIP Phone	Verify that an ad hoc conference can be established by a local Unified CallManager Express phone with a Unified CallManager controlled SIP phone and another Unified CallManager controlled SIP phone.	Stage 1: Phone->? Unified CallManager Express->Unified CallManager->SIP Phone, Stage 2: Phone->? conference softkey->Unified CallManager Express->Unified CallManager->SIP Phone 2, Stage 3: Phone->? conference softkey->SIP Phone and SIP Phone 2	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CE10.MPE.137	Unified MeetingPlace Express	Unified MeetingPlace Express Functionality: Cisco IP Communicator User Invokes Meeting Extension, Case 2	Verify that an end user with Cisco IP Communicator can invoke an extension for a meeting with the following participants: low-end, Cisco Catalyst 6000 WS-6624 (MGCP), Unified CallManager Express (H.323), and MGCP gateway.	low-end->SCCP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express; 6624->MGCP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express; Phone->H323->Unified CallManager Express->Gatekeeper->Unified MeetingPlace Express; PSTN->Gateway->MGCP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express	Passed	
CE10.MPE.169	Unified MeetingPlace Express	Unified MeetingPlace Express Functionality: Cisco IP Communicator User Initiates Recording, Case 1	Verify that an end user with Cisco IP Communicator can start recording a meeting with the following participants: Cisco Unified IP Phone 7936, video endpoint, Unified CallManager Express (H.323), and H.323 gateway.	7936->Unified CallManager->Gatekeeper->Unified MeetingPlace Express; Video->Unified CallManager->Gatekeeper->Unified MeetingPlace Express; Phone->H323->Unified CallManager Express->Gatekeeper->Unified MeetingPlace Express; PSTN->Gateway->H323->Unified CallManager->Gatekeeper->Unified MeetingPlace Express	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CE10.MPE.204	Unified MeetingPlace Express	Unified MeetingPlace Express Feature Interaction: Calling Line ID Presentation (CLIP)/Caller Name (CNAM), Case 3	Verify the Calling Line ID Presentation (CLIP) for the following meeting participants: high-end, Cisco Unified IP Phone 7920, Cisco VG248 analog gateway (SCCP), Unified CallManager Express (H.323), and H.323 gateway.	high-end->Unified CallManager->Gatekeeper->Unified MeetingPlace Express; 7920->SCCP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express; VG248->SCCP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express; Phone->H323->Unified CallManager Express->Gatekeeper->Unified MeetingPlace Express; PSTN->Gateway->H323->Unified CallManager->Gatekeeper->Unified MeetingPlace Express	Failed	CSCsc88441
CE10.MPE.229	Unified MeetingPlace Express	Unified MeetingPlace Express Feature Interaction: Call Forward on Busy, Case 2	Verify a meeting where all participants have the following call forwarding on busy settings: (a) from Cisco Unified IP Phone 7936 to Cisco Unity; (b) from high-end-new to Cisco VG248 analog gateway (SCCP); and (c) from Cisco SIP Proxy Server to MGCP gateway.	part-A-from 7936->Unified CallManager->Gatekeeper->Unified MeetingPlace Express to Unity; part-B-from high-end->Unified CallManager->Gatekeeper->Unified MeetingPlace Express-new to VG248->SCCP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express; part-C-from CSPC->SIP->Unified CallManager-> Gatekeeper->Unified MeetingPlace Express to PSTN->Gateway->MGCP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CE10.MPE.244	Unified MeetingPlace Express	Unified MeetingPlace Express Feature Interaction: Unconditional Call Forward, Case 4	Verify a meeting where all participants have the following unconditional call forwarding settings: (a) from Cisco Unified IP Phone 7936 to Cisco Unity; (b) from Cisco IP Communicator to Cisco VG248 analog gateway (SCCP); and (c) from Unified SRST (SCCP) to SIP gateway.	part-A-from 7936->Unified CallManager->Gatekeeper->Unified MeetingPlace Express to Unity; part-B-from CIPC->SCCP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express to VG248->SCCP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express; part-C-from 79XX->SCCP->SRST/Unified CallManager-> Gatekeeper->Unified MeetingPlace Express to PSTN->Gateway->SIP->Unified CallManager->Gatekeeper->Unified MeetingPlace Express	Passed	
CO10.CCM.007	Unified SRST	Multiple Line Registration from H.323 Gateway During SRST Fallback	Verify multiple line registration from a H.323 gateway to an SRST gateway.	Phone->SRST Gateway->SIP Phone, Phone->Unified CallManager->SIP Phone	Passed	
CO10.CCM.016	Unified SRST	Secure SIP Phones Behind Unified SRST Gateway	Verify that secure SIP endpoints can register with a Unified SRST gateway.	Phone->SRST Gateway->SIP IP Phone, Phone->Unified CallManager->SIP Phone	Passed	
CO10.CCM.039	Load	40,000 BHCA in Unified CallManager	Verify that 99.99% calls complete at 40,000 busy hour call attempts (BHCA).		Failed	CSCse26087

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.CCM.045	Unified CallManager Applications	WebDialer Call from SIP Phone across Gatekeeper ICT to Another SIP Phone in Call Pickup Group, Call Parked and Retrieved from Third SIP Phone	Verifies WebDialer Make and End call features with non-encrypted SIP phone using the following process: WebDialer initiates a non-encrypted call across a gatekeeper ICT to a SIP phone in call pickup group, the call is parked and then retrieved by another SIP phone, and then WebDialer ends the call.	WebDialer->SIP Phone 1->Unified CallManager->Gatekeeper ICT->Unified CallManager->SIP Phone->(call pickup group)->call park->SIP Phone->call retrieve->WebDialer (end call).	Failed	CSCse09848
CO10.CCM.066	Shared Line	Analog Phone to SIP Phone Shared Line, Hunt Group, Consultative Transfer, Blind Transfer, and Call Forward No Answer to Cisco Unity Voice Mail	Verify hunt groups, consultative and blind transfers, and call forward on no answer (CFNA) to a Cisco Unity voice mail box for a SIP phone with shared lines.	EP->VG 224/248->Unified CallManager->SIP Phone->(shared-line, hunt group)->consult Transfer->SIP Phone->(non-shared-line)->blind Transfer->SIP Phone 3->CFNA->Unity voice mail.	Passed	
CO10.CCM.081	Basic Call Flow	Single Number Reach Interoperability with Cisco Unity Voice Mail	Verify the ability to operate a single voice mail box for each Single Number Reach user.	Unified CallManager->, Phone->, SNR server->, Unity voicemail	Passed	
CO10.CCM.082	Basic Call Flow	Single Number Reach Interoperability with Cisco Unified MeetingPlace	Verify the ability to attend a Cisco Unified MeetingPlace meeting and exchange the call between a mobile phone and a desk phone during the meeting.	Unified CallManager->, MeetingPlace, Phone->, SNR server->	Passed	
CO10.CCM.086	Basic Call Flow	Join Conference from Single Number Reach Desk Phone	Verify the ability to join a conference call from a Single Number Reach desk phone.	Unified CallManager->, Phone, Phone-> (SNR)	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.CCM.087	Basic Call Flow	Join Conference from Single Number Reach Mobile Phone	Verify the ability to join a meeting from a mobile phone in an Single Number Reach group (requires enterprise dial tone feature for access).	Unified CallManager->, cell Phone->, meeting, SNR server-> (IVR)	Passed	
CO10.CCM.090	Unified CallManager Applications	WebDialer Call from SIP Phone across ICT to SCCP Phone in Hunt Group, Blind Transfer Call to Another SCCP Phone with Call Forward No Answer to Cisco Unity Voice Mail	Verify WebDialer Make and End call features with a SIP phone using the following process: WebDialer initiates a call from the SIP phone across an ICT to an SCCP phone in a hunt group, and then the call is blind transferred to another SCCP phone configured for call forward on no answer (CFNA) to Cisco Unity voice mail before WebDialer ends the call.	WebDialer->SIP Phone 1->Unified CallManager->ICT->Unified CallManager->SCCP Phone->(hunt group)->blind Transfer->SCCP Phone 3->CFNA->Unity Voice Mail->WebDialer (end call).	Failed	CSCse16204
CO10.CCM.102	Basic Call Flow	Intra-Cluster Call from Analog Phone on VG248 to SCCP Phone with Shared Line	Verify that an analog phone connected to a VG248 gateway can make a call to an SCCP phone with shared lines.	Analog Phone->VG248->Unified CallManager->SCCP (shared-line, group-pickup)->Transfer->Unified CallManager->VG224->Analog Phone.	Passed	
CO10.CCM.103	Shared Line	SCCP Phone Shared Line (Multiple Phones) Hunt Group, Call Park, and Call Pickup Features	Verify hunt group, call park and call pickup features on a shared line across SCCP phones.	call pickup), hunt group)->blind transfer->Gatekeeper ICT->SCCP Phone 4 (shared-line)->hold->resume->call park->SCCP Phone5 (shared-line, SCCP Phone 1-> ICT->SCCP Phone->(shared-line	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.CCM.111	Shared Line	SCCP Phone Shared Line Group Pickup of Multiple Calls	Verify group pickup of multiple calls on a SCCP phone with shared lines.	SCCP Phone 1->Unified CallManager->ICT->Unified CallManager->SCCP Phone->(shared line)->CFA->SCCP Phone 3->group pickup SCCP Phone 4 (non-shared-line)., While call is active SCCP Phone->calls Phone 3/4 shared-line and the call is picked up by SCCP Phone 4 (shared-line). Phone 4 CC Phone->and Phone 2.	Passed	
CO10.CCM.119	Hunt Groups	Inter-Cluster SCCP Phone Call to Hunt List with Forward on No Answer to Cisco Unity Voice Mail	Verify that a call made from a SCCP phone to a hunt list pilot number (configured with call forward no answer) is successfully forwarded to a Cisco Unity voice mail when no member of the hunt list answers the call and the timer expires.	SCCP Phone 1->Unified CallManager->ICT->Unified CallManager->Hunt Pilot->Hunt list (SCCP Phone 2, SCCP Phone 3)->timeout->Forward Hunt No Answer->Unified CallManager->Unity Voice Mail	Passed	
CO10.CCM.122	Hunt Groups	PSTN Phone Call to Hunt List with Forward Hunt Busy to SCCP Phone	Verify that a call made from a PSTN phone to a hunt list pilot number (configured for call forward busy) is successfully forwarded to a destination SCCP phone configured when all members of the hunt list are busy.	FXS Phone->PSTN->Gateway->Unified CallManager->Hunt Pilot->Hunt list (SCCP Phone 2, SCCP Phone 3)->Forward Hunt Busy->Unified CallManager->ICT->Unified CallManager->SCCP Phone 1.	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.CCM.138	Unified CallManager	16 SRST Sites (Financial Cluster) 1 Hour 6K BHCA Load Run	Generate traffic at 6k BHCA for 1 hour across 16 SRST sites. The call flows are as follows: 1. 4-digit Intra-Site Dialing 2. 8-digit Inter-Site Dialing 3. 7-digit or 10-digit Local PSTN Dialing 4. 11-digit Long Distance PSTN Dialing These call flows includes basic, Hold, Blind Transfer, Supervised Transfer, and Conferencing	Unified CallManager Load	Passed	
CO10.CCM.139	Unified CallManager	48 SRST Sites (Financial Cluster) 1 Hour 18K BHCA Load Run	Generate traffic at 18k BHCA for 1 hour across 48 SRST sites. The call flows are as follows: 1. 4-digit Intra-Site Dialing 2. 8-digit Inter-Site Dialing 3. 7-digit or 10-digit Local PSTN Dialing 4. 11-digit Long Distance PSTN Dialing These call flows includes basic, Hold, Blind Transfer, Supervised Transfer, and Conferencing	Unified CallManager Load	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.CCM.141	Unified CallManager	Registration of 100 PSTN MGCP Gateways to Unified CallManager Cluster	Verify that 100 PSTN MGCP gateways, including Cisco Catalyst 6000 series switches with 6608 modules and Communication Media Modules supporting both ISDN PRI and channel associated signaling (CAS) trunks, can successfully register to a very large single-site Unified CallManager cluster.	Unified CallManager->MGCP Gateway	Passed	
CO10.CCM.158	Basic Call Flow	PSTN-to-SIP Intra-Cluster Call over Nortel DMS-100 Trunk Released by Calling Party	Verify the ability to make a successful basic intra-cluster call over a PSTN trunk configured for Nortel DMS-100 signaling to a SIP phone that is then released by the calling party.	Analog Phone->DMS100->PRI->CMM->CMM->SIP Phone.	Passed	
CO10.CCM.159	Basic Call Flow	PSTN to SIP Intra-Cluster Call over AT&T 5ESS Trunk Released by Called Party	Verify the ability to make a successful basic intra-cluster call over a PSTN trunk configured for AT&T 5ESS signaling to a SIP phone that is then released by the called party.	Analog Phone->5ESS->PRI->CMM->CMM->SIP Phone.	Passed	
CO10.CCM.180	Basic Call Flow	Other Group Pickup Multiple Calls	Verify the ability to answer the longest unanswered call with one key or answer another call in the other pickup group by using a softkey.	SIP Phone A->Unified CallManager1->Unified CallManager2->SIP Phone B->(Group Pickup) SIP Phone D	Passed	
CO10.CCM.195	Basic Call Flow	FXS-to-SIP Encrypted Phone Call	Verify the ability to make a call from a foreign exchange station (FXS) phone to an encrypted SIP phone.	Analog Phone VG224->Unified CallManager->SIP Phone	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.CCM.196	Basic Call Flow	PSTN-to-SIP Encrypted Phone Call	Verify the ability to make a call from a PSTN line to an encrypted SIP phone.	Analog Phone->PSTN->Unified CallManager->SIP Phone	Passed	
CO10.CCM.198	Basic Call Flow	Call Forward All Inter-Cluster Calls Between Encrypted SIP Phones	Verify the ability to call forward all (CFA) inter-cluster calls between encrypted SIP phones across a H.323 ICT.	SIP Phone 1(encrypted)->Unified CallManager->ICT->Unified CallManager->SIP Phone->(encrypted)->CFA->SIP Phone 4.	Passed	
CO10.CCM.201	Basic Call Flow	Blind Transfer Inter-Cluster Calls Between Encrypted SIP Phones	Verify the ability to perform a blind transfer on inter-cluster calls between encrypted SIP phones across a H.323 ICT.	SIP Phone 1-> Unified CallManager->ICT->Unified CallManager->SIP Phone 2->blind Transfer->SIP Phone 3	Passed	CSCse19135
CO10.CCM.202	Basic Call Flow	Consult Transfer Inter-Cluster Calls Between Encrypted SIP Phones	Verify the ability to perform a consultative transfer on inter-cluster calls between encrypted SIP phones across a H.323 ICT.	SIP Phone 1-> Unified CallManager->ICT->Unified CallManager->SIP Phone 2->consult Transfer->SIP Phone 3	Passed	
CO10.CCM.210	Hunt Groups	Hunt List Forward to Route Pattern with FAC	Verify the Unified CallManager server's ability to process an inbound call to a hunt list with Broadcast distribution, then hunt list forward the call to a route pattern with a forced authorization code (FAC).	SIP Phone->Unified CallManager->SIP Phone->CFB->Unified CallManager/HL->PSTN Phone	Passed	
CO10.CCM.220	Unified CallManager Applications	Voice Mail after Logging into Unified CallManager Assistant	Verify the ability to leave voice mail after logging into the Cisco Unified CallManager Extension Mobility.	SIP Phone 2->Unified CallManager->SIP Phone->(EM)->CFA->Unity Voice Mail	Failed	CSCse12199
CO10.CER.010	Failover	ERL Reconfiguration While Secondary Cisco Emergency Responder was Active Gets Discovered by Restored Primary	Verify that an emergency response location (ERL) reconfiguration which occurred while the secondary Cisco Emergency Responder was active is discovered when the primary becomes active again.	CER Admin->ERL Reconfiguration->Failover Primary	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.CER.020	Cisco Emergency Responder	Call Back from PSAP Using ELIN Reaches Original Emergency Caller	Verify that a public safety answering point (PSAP) can call back the original caller in case the emergency call gets disconnected.	PSAP->Unified CallManager->CER->911 Caller	Passed	
CO10.CME.010	Feature Termination	Blind Conference Between Unified CallManager and Unified CallManager Express which Forwards All Calls to Another Unified CallManager Express Without Tandem Gateway	Verify the ability to pre-answer a conference call (press the Conference softkey before the called party answers the call) that is originated by Unified CallManager to a Unified CallManager Express that forwards all calls (CFA) to another Unified CallManager Express without a tandem gateway.	SIP IP Phone->Unified CallManager->IP Phone->Unified CallManager Express->-Unified CallManager Express->IP Phone	Passed	
CO10.CME.011	Feature Termination	Blind Conference Between Unified CallManager and Unified CallManager Express which Forwards All Calls to Another Unified CallManager Express with Tandem Gateway	Verify the ability to pre-answer a conference call (press the Conference softkey before the called party answers the call) that is originated by Unified CallManager to a Unified CallManager Express that forwards all calls (CFA) to another Unified CallManager Express using a tandem gateway.	SIP IP Phone->Unified CallManager->IP Phone->Gateway->Unified CallManagerE->Unified CallManagerE->-IP Phone	Passed	
CO10.CME.018	Feature Termination	Call on Secure/Non-Secure H.225 Trunk Between Unified CallManager and Unified CallManager Express Phones	Verify the ability to make a call between a Unified CallManager phone and Unified CallManager Express phone over both secure and non-secure H.225 trunks.	SIP IP Phone->Unified CallManager->H.225 Trunk->-Unified CallManagerE->sccp IP Phone	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.CRS.000	Basic Call Flow	H.323 Gateway Call Transferred to Unified Contact Center Express Server in Different Cluster	Verify a successful call from an H.323 gateway to an IP phone that has call pickup activated, which is then consultative transferred to a Unified Contact Center Express server in a different Unified CallManager cluster and received by an available SIP TNP agent who parks the call.	H.323 Gateway->Unified CallManager->IP Phone (call pickup)->IP Phone->(consult transfer)->Unified CallManager->Gatekeeper ICT->Unified CallManager->CRA->SIP TNP Agent->(call park)->Unified CallManager->Gatekeeper ICT->Unified CallManager (call park DN)->IP Phone	Passed	
CO10.CRS.004	Basic Call Flow	PSTN MGCP Gateway Call to SIP TNP Agent Conferencing with Remote Cluster	Verify the ability to complete a call from a MGCP gateway to any available SIP TNP agent served by a Unified Contact Center Express server in a different Unified CallManager cluster. The SIP TNP agent conferences in a remote cluster directory number, consults with the remote party and the caller, and then continues the call with the caller alone.	EP->MGCP Gateway->Unified CallManager->ICT->Unified CallManager->CRA->SIP TNP Agent->(conference)->Unified CallManager->ICT->IP DN	Passed	
CO10.CRS.014	Basic Call Flow	Call from Cisco Unity Auto Attendant Transferred to Customer Response Solution SIP TNP Agent	Verify the ability to complete a call from a Cisco Unity auto attendant this is transferred to a SIP TNP agent.	cm->crs->SIP TNP Agent., Phone->unity aa->	Passed	
CO10.CTP.000	Basic Call Flow	Centralized TFTP interoperability with current and previous release of Centralized TFTP servers with SIP Legacy phones	Multi cluster Centralized TFTP server with mixed Unified CallManager versions Clusters where Unified CallManager 4.1 is the master Centralized TFTP server		Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.CTP.002	Basic Call Flow	Centralized TFTP interoperability with current and previous release of Centralized TFTP servers with SCCP phones	Multi cluster Centralized TFTP server with mixed Unified CallManager versions Clusters where Unified CallManager 4.1 is the master Centralized TFTP server		Passed	
CO10.CTP.004	Basic Call Flow	Centralized TFTP failover to backup Local TFTP server with SCCP Phones	Multi cluster Centralized TFTP server with mixed Unified CallManager 5.0 and 4.1 Clusters where Unified CallManager 4.1 is the master Centralized TFTP server. Fail the primary Unified CallManager 5.0 Local TFTP server while Master Centralized TFTP server is still available.		Passed	
CO10.FAM.004	Basic Call Flow	T.38 Fax Relay Through IOS Gateway to Cisco Fax Server	Verify the ability to send a T.38 fax through an IOS gateway to a Cisco Fax Server using SIP for signaling and T.38 for media.	Cisco Fax server, FaxLab->, IOS gateway->	Passed	
CO10.INT.003	Call Forward	IP-to-IP Call from Unified CallManager Release 4.1.x SCCP Phone over SIP ICT to Unified CallManager Release 5.0 SIP Phone Forwarded to Another Extension and Voice Mail	Verify the ability to make and answer an audio call from a Unified CallManager Release 4.1.x SCCP IP phone over a SIP ICT to a Unified CallManager Release 5.0 SIP phone which forwards all calls (CFA) to another extension and then to voice mail.	Phone 1->Unified CallManager->ICT ->Unified CallManager->Phone2	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.INT.004	Basic Call Flow	Analog-to-IP Call from Unified CallManager Release 4.1.x SCCP Analog Phone to Unified CallManager Release 5.0 SCCP Phone over H.323 ICT	Verify the ability to make and answer an audio call from a Unified CallManager Release 4.1.x SCCP analog phone connected to a VG224 to a Unified CallManager Release 5.0 SCCP phone over an H.323 inter-cluster trunk.	Phone 1->Unified CallManager->ICT ->Unified CallManager->Phone2	Passed	
CO10.INT.011	Basic Call Flow	IP-to-IP Inter-Cluster Call with Hardware Transcoding on IOS Gateway	Verify the ability to make an audio call from a Unified CallManager Release 5.0 SCCP phone to a Unified CallManager Release 4.1.x SCCP phone over a gatekeeper-controlled H.323 inter-cluster trunk with hardware transcoding performed by a IOS gateway in the Unified CallManager Release 4.1.x cluster.	Phone 1->Unified CallManager->ICT ->Unified CallManager->Phone2	Passed	
CO10.IPP.007	Call Transfer	Intra-Cluster Call from 3rd Party SIP Phone to SIP Cisco Unified IP Phone with Blind Transfer	Verify the ability to make an intra-cluster call from a 3rd party SIP phone to a SIP Cisco Unified IP Phone which performs a successful blind call transfer to another SIP Cisco Unified IP Phone; confirm that the secondary call gets answered.	3rd Party SIP Phone->Unified CallManager->SIP Phone->Blind Trfr->Cisco SIP Phone	Passed	
CO10.IPP.024	Call Forward	Call Forward on Busy for Inter-Cluster Call from 3rd Party SIP Phone (Polycom) to SIP Cisco Unified IP Phone	Verify that an inter-cluster call from a 3rd party SIP phone (Polycom) can be call forwarded on busy (CFB) to a SIP Cisco Unified IP Phone.	Cisco SIP Phone->Unified CallManager->3rd Party SIP Phone->Call Forward->ICT->Cisco SIP Phone	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.IPP.038	Call Transfer	Intra-Cluster Call from 3rd Party SIP Phone (Sipura) to SIP Cisco Unified IP Phone with Blind Transfer over ICT	Verify the ability to make a call from a 3rd party SIP phone (Sipura) over an inter-cluster trunk to a SIP Cisco Unified IP Phone which performs a successful blind call transfer to another SIP Cisco Unified IP Phone; confirm that the secondary call gets answered.	3rd Party SIP Phone->Unified CallManager->SIP Phone->Blind Trfr->ICT->Cisco SIP Phone	Passed	
CO10.IPP.046	Basic Call Flow	AutoRegister over TCP	Verify that a phone continually attempts to auto-register until successful.	TNP SIP Phone->Unified CallManager	Passed	
CO10.IPP.049	Music on Hold	SIP Phone Early Attended Transfer with Music on Hold	Verify the ability of a SIP phone to perform an early attended transfer with Music on Hold (MoH).	TNP SIP Phone->SIP Phone->Transfer->SIP Phone	Passed	
CO10.MPL.001	Unified MeetingPlace	Audio and Data Conference on Unified MeetingPlace with Mixed Secure, Unsecured, Intra- and Inter-Cluster IP Phones Outdial via SIP Gateway	Verify the ability to hold an audio and application-sharing data conference on Unified MeetingPlace with mixed secure and unsecured SCCP IP phones (dial in) and SIP phones (outdial via SIP gateway) that originate both from within the same cluster and across inter-cluster trunks.	Stage 1: SCCP IP Phone->Unified CallManager->Gatekeeper->Cisco MeetingPlace, Stage 2: SCCP IP Phone->Unified CallManager->Gatekeeper-ICT->Unified CallManager->Gatekeeper->Cisco MeetingPlace, Stage 3: Cisco MeetingPlace->Unified CallManager->SIP IP Phone, Stage 4: Cisco MeetingPlace->Unified CallManager->SIP Trunk->Unified CallManager->SIP IP Phone	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.NME.008	Network Management	Alerts and Activities Display, Case 1	Existing AAD functionality will continue to be available. All operations under the topology display can be launched under the tools pulldown menu. The event property detail launched under the "Alert Detail View" shall show the treshold value and the latest polled value.	Not Applicable	Passed	
CO10.NME.033	Network Management	Phone Reachability Testing, Case 2	Verify Phone Reachability Test in operational mode. Users will pick phones from a phone report and include them to be monitoring for reachability.	Not Applicable	Passed	
CO10.QOS.000	RSVP	Transfer Failure Due to Not Enough Bandwidth Condition when Policy is Mandatory	Verify that call transfers do not complete when the RSVP policy is mandatory and not enough bandwidth is available for the called party.	-->-call-> 2,->2 -Transfer-> 3	Passed	
CO10.QOS.002	RSVP	Under Limited Bandwidth Condition Voice Calls Succeed But Video Calls Fail	Verifies how under limited bandwidth condition where there is only enough bandwidth for voice, and when the policy is Mandatory Video Desired, the voice call succeed and the video call fails.	A -video-call-->B,->-video-conf-> C	Passed	
CO10.QOS.004	RSVP	RSVP Reservation retries after initial Failure due to lack of bandwidth	Verifies the retry feature in a complex conference setup.	A -conf-> B,->-conf-> C, A -conf-> D,->-conf-> E, A -conf-> F,->-conf-> G	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.QOS.005	RSVP	Reservations will be installed for Shared Lines and torn down the ones Not used after the Call is accepted by one of the Endpoints	Verify that RSVP reservations are installed and torn properly.	A -call-> B,->-blind_Transfer-> (C, D, E, F -answer the call, F)	Passed	
CO10.QOS.015	RSVP	Call diversion using AAR when reservation fails.	Calls will be diverted to AAR when the reservation fails	A -call-> B	Passed	
CO10.UCA.034	Unified Personal Communicator	Cisco Unified Personal Communicator in softphone mode behind IPSEC VPN listen to VM messages thru Instant MessagerAP (use different domain)	Bring up Unified Personal Communicator behind IPSEC VPN. Click on the message waiting and down load the VM from Voicemail server. Listen to the VM		Passed	
CO10.UCA.035	Unified Personal Communicator	4 incoming calls to Cisco Unified Personal Communicator in softphone mode almost instantaneously.	Set the Maximum calls to Unified Personal Communicator phone to 4 in Unified CallManager. Call Unified Personal Communicator from different sites across SIP, H323 trunks and local calls almost simultaneously. Check that Unified Personal Communicator won't crash and at least 2 calls are successful in Unified Personal Communicator first Release.		Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.UCA.036	Unified Personal Communicator	Create a LDAP failure and double click a contact to make a call	Create a LDAP authentication failure. Double click a contact and try to establish a call. Restore it and create a LDAP IP connectivity failure. Double click a contact to establish a call.		Passed	
CO10.UCA.037	Unified Personal Communicator	Isolate the Cisco Unified Presence Server server during a call and check the call last for the duration.	Establish at least 2 calls from 2 Unified Personal Communicator end points, one from softphone and one from phone associate mode. Create a IP connection failure to Unified Presence Server. Check for the status of the buddies after few minutes. Verify that Unified Personal Communicator shows errors a which indicate Unified Presence Server connection failure. Disconnect the call and establish a call from the desk phone associated with Unified Personal Communicator.		Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.UCA.038	Unified Personal Communicator	Presence: Cisco Unified Personal Communicator in softphone mode, set the presence status of Cisco Unified Personal Communicator to busy and check that it is updated in other Cisco Unified Personal Communicator, and Phone IPPM.	From a Unified Personal Communicator in softphone mode set it's presence status to busy. Verify that other Unified Personal Communicator end points and IPPM phones display the correct status. Establish a call to this Unified Personal Communicator and verify that the Unified Personal Communicator alert the user and a way to decline the call is displayed.		Failed	CSCse35706
CO10.UCA.039	Unified Personal Communicator	Presence status: Check for Presence of the phone under Received Calls	Check the Presence status of the the buddies which comprises of IPPM phones, Unified Personal Communicators and all are configured in Unified Presence Server. Verify that correct statuses are displayed.		Passed	
CO10.UCA.040	Unified Personal Communicator	Cisco Unified Personal Communicator as a IPCC Express phone	Configure Unified Personal Communicator as one of the IPCC express phone(auto attendant). Establish an incoming call from PSTN to this Unified Personal Communicator. Verify that the PSTN caller hear the IVR prompts and it punch the extension number of Unified Personal Communicator phone. Verify that call is established to Unified Personal Communicator		Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.UCA.041	Unified Personal Communicator	Cisco Unified Personal Communicator in Softphone mode incoming video call from H323 video end point	Bring up Unified Personal Communicator in Softphone mode. Establish a video call from a H323 video end point from a different cluster. Verify that the call is established and both parties can see the local and remote video. Turn off the video during the call and turn it on again after a minute.		Failed	CSCse30001, CSCse34224
CO10.UCA.042	Unified Personal Communicator	Cisco Unified Personal Communicator in Softphone mode, merge an incoming Video call from 7985 and out going video call from IP Communicator	Set up a video call from 7985 video phone 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.		Failed	CSCsd74434
CO10.UCA.043	Unified Personal Communicator	Cisco Unified Personal Communicator in softphone mode merge a outgoing video call to SCCP video endpoint and incoming from H323 video endpoint.	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 H323 video end point thru ICT. Answer the call. Verify that the first call is fed with Music On Hold and no Video. Click on the Merge button and estabish a 3 way conference call. Check for audio and video of this conference call.		Failed	CSCse30001, CSCse34224

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.UCA.044	Unified Personal Communicator	Cisco Unified Personal Communicator in softphone mode behind IPSEC VPN call a CVTA video end point through SIP trunk	Bring up Unified Personal Communicator thru a IPSEC VPN connection. Originate a call from Unified Personal Communicator to a CVTA end point thru SIP ICT. Bring up video during the call. Check the call again for audio and video connection		Passed	
CO10.UCA.045	Unified Personal Communicator	Cisco Unified Personal Communicator behind IPSEC VPN in softphone mode, Hold and Resume a Video call to CVTA .	Bring up Unified Personal Communicator thru a IPSEC VPN connection. Originate a call from Unified Personal Communicator to a CVTA end point thru 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	
CO10.UNC.012	Voice Mail	Inter-Site Message Delivery, Retrieval, Deletion and MWI from SIP and SCCP Endpoint to Cisco Unity Connection PDL over Dual Integration with G.729 Codec	Verify that Cisco Unity Connection can support message delivery, retrieval, and deletion when calls are made between Unified CallManager cluster and Cisco SIP Proxy Server using a SIP and SCCP phones over a SIP integration.	SCCP Phone->Unified CallManager->SIP Trunk->->CSPS->SIP Phone->Unity Connection	Passed	
CO10.UNC.024	Voice Mail	Calls Forwarded to Cisco Unity Connection Through Intercluster Trunk (SCCP Integration)	Verify that calls forwarded to Cisco Unity Connection through an intercluster trunk are successful.	SCCP Phone->Unified CallManager->ICT->Unified CallManager->SIP Phone->Unity Connection	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.UNC.027	Voice Mail	Direct Calls to Cisco Unity Connection Through Intercluster Trunk (SIP Integration)	Verify that direct calls to Cisco Unity Connection through an intercluster trunk are successful.	SCCP Phone->Unified CallManager->ICT->Unified CallManager->SIP Phone->Unity Connection	Passed	
CO10.UNC.036	Voice Mail	Calls Forwarded to Unity Connection Through MGCP Gateway over SIP Integration to Unified CallManager	Verify that calls forwarded to Unity Connection from a PSTN phone through an MGCP gateway are successful.	PSTN Phone->MGCP Gateway->Unified CallManager->SIP Phone->Unity Connection->SIP Phone	Passed	
CO10.UNC.037	Voice Mail	Direct calls to Unity Connection Through H.323 Gateway over SIP Integration to Unified CallManager	Verify that calls forwarded to Unity Connection from a PSTN phone through an H.323 gateway are successful.	PSTN Phone->MGCP Gateway->Unified CallManager->SIP Phone->Unity Connection->SIP Phone	Passed	
CO10.UNI.001	Voice Mail	Multiple Cluster Integration Outside Caller Call	Verify that Cisco Unity can integrate with Unified CallManager servers in different clusters.	SCCP/SIP Phone->Unified CallManager->Unity->Microsoft Exchange	Passed	
CO10.UNI.012	Voice Mail	Inter-Site Message Delivery, Retrieval and Deletion from SCCP Endpoints to Cisco Unity over a SIP Cisco Unity-Cisco SIP Proxy Server Integration Forwarded Call to Voice Mail	Verify that Cisco Unity can support message delivery, retrieval, and deletion when calls are made between Unified CallManager cluster and Cisco SIP Proxy Server using a SCCP phones over a SIP integration.	SCCP Phone->Unified CallManager->SIP Trunk->CSPS-> Unity->Microsoft Exchange->SIP Phone	Failed	CSCse19135
CO10.UNI.054	Voice Mail	Internet Mail Access Protocol Client	Verify IMAP client support in Windows and Macintosh.	IP Phone->Unified CallManager->Unity->Microsoft Exchange ->IMAP Client	Passed	
CO10.UNI.073	Voice Mail	Message Notification with SIP Integration	Verify that message notification feature works as expected with SIP integration to Unified CallManager.	SIP/SCCP Phone->Unified CallManager->Unity	Failed	CSCsd26874

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.VID.001	Video	SCCP Non-Video to SCCP Video Call Transfer to SCCP Video Endpoint Inter-Cluster H.323 Gatekeeper Controlled	Verify a call transfer from SCCP non-video endpoint, transferring a SCCP Video endpoint to another SCCP Video Capable endpoint inter-cluster H.323 gatekeeper Controlled	Stage 1: SCCP Audio->Unified CallManager->SCCP Video, Stage2: SCCP Audio Transfer->Unified CallManager->ICT->Unified CallManager->SCCP Video	Passed	
CO10.VID.002	Video	SCCP Non-Video to SCCP Video Call Transfer to SCCP Video Endpoint Inter-Cluster SIP Trunk	Verify a call transfer from SCCP non-video endpoint, transferring a SCCP Video endpoint to another SCCP Video Capable endpoint inter-cluster SIP Trunk	Stage 1: SCCP Audio->Unified CallManager->SCCP Video, Stage2: SCCP Audio Transfer->Unified CallManager->ICT->Unified CallManager->SCCP Video	Passed	
CO10.VID.005	Video	SIP Non-Video to SCCP Video Call Transfer to SCCP Video Endpoint Inter-Cluster SIP Trunk	Verify a call transfer from SIP (TNP) non-video endpoint, transferring a SCCP Video endpoint to another SCCP Video Capable endpoint, inter-cluster SIP Trunk	Stage 1: SIP Audio->Unified CallManager->SCCP Video, Stage2: SIP Audio Transfer->Unified CallManager->ICT->Unified CallManager->SCCP Video	Passed	
CO10.VID.013	Video	SIP Non-Video to H.323 Video Call Transfer to H.323 Video Endpoint Inter-Cluster H.323 Gatekeeper Controlled	Verify a call transfer from SIP (TNP) non-video endpoint, transferring a H.323 Video endpoint to a H.323 Video endpoint, inter-cluster H.323 gatekeeper Controlled	Stage 1: SIP TNP Audio->Unified CallManager->H.323 Video, Stage2: SIP TNP Audio Transfer->Unified CallManager->ICT->Unified CallManager->H.323 Video	Failed	
CO10.VID.031	Video	SCCP Tandberg/7985 to H.323 Tandberg Video Call with H.264 codec Inter-Cluster SIP Trunk	Video call using SCCP 7985 and SCCP Tandberg video endpoints to H.323 Tandberg Video endpoints H.264 codec, inter-cluster SIP Trunk	SCCP Tandberg/7985->Unified CallManager->ICT->Unified CallManager->H.323 Endpoints	Passed	
CO10.VID.033	Video	H.264 Video calls between H.323 Endpoints across SIP Trunks	Intercluster IP-IP Video call between H.323 Video endpoints across SIP Trunks	H.323 Video Endpoint->Gatekeeper->Unified CallManager->SIP->Unified CallManager->Gatekeeper->H.323 Video Endpoint	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO10.VID.036	Video	H.323 Tandberg to Polycom VSX Gatekeeper Controlled, Inter-Cluster SIP Trunk	H.323 Tandberg T1000 Video endpoint to Polycom VSX 7000 Video endpoint with gatekeeper control and utilizing dynamic H.323 addressing feature, inter-cluster SIP Trunk	H.323 Tandberg T1000->Gatekeeper->Unified CallManager->ICT->Unified CallManager->Gatekeeper->Polycom VSX 7000	Passed	
CO10.VID.045	Video	SCCP Video Endpoint to H.323 Video Endpoint Call Transfer to SCCP Video Endpoint, Inter-Cluster SIP Trunk	Call transfer of a H.323 Video Endpoint between SCCP Video Endpoints inter-cluster SIP Trunk	Stage 1: SCCP Video Endpoint->Unified CallManager->Gatekeeper->H.323 Video Endpoint, Stage2: SCCP Video Endpoint->Transfer->Unified CallManager->ICT->Unified CallManager->SCCP Video Endpoint	Passed w/ Exception	
CO105.CUP.001	Basic Call Flow	Bulk list of buddies in multiple IP phones and check the phone status on SCCP phones.	Configure bulk list of buddies(20) on preferably 6 real phones and check the Status of the phones are correctly reflected.	Phones--<html>--Unified CallManager	Failed	CSCse03676
CO105.CUP.002	Basic Call Flow	Bulk list of buddies in multiple IP phones and check the phone status on SIP phones.	Configure bulk list of buddies(20) on preferably 6 real SIP legacy and TNP IP phones check the presence status of the buddies are correctly reflected.	Phones--<html>--Unified CallManager	Passed	
CO105.CUP.003	Basic Call Flow	Same buddies on multiple IP phones, check the phone presence status.	Configure 5 buddies Executive buddies) on preferably 1000 simulated phones and 2 real phones and check the Status of the executive buddies are correctly reflected in Very Large Campus with Clustering over the WAN site-RFD cluster	Phones--<html>--Unified CallManager	Passed	CSCse03676

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO105.CUP.004	Basic Call Flow	Same buddies on multiple IP phones, check the phone presence status.	Configure 5 buddies Executive buddies) on preferably 1000 simulated phones and 2 real phones and check the Status of the executive buddies are correctly reflected in Large SIP site cluster	Phones--<html>--Unified CallManager	Passed	
CO105.CUP.005	Basic Call Flow	Bulk list of buddies, check the Instant Messager status on multiple IP phones behind SRST Gateway in SJC cluster.	Configure bulk list of buddies on preferably 4 real phones behind SRST GW and check the presence status of the phones are correctly reflected.	Phones--<html>--Unified CallManager	Passed	
CO105.CUP.006	Basic Call Flow	Mixed list of buddies like video phones, Secure IP phones, and analog end points check the Instant Messager status on multiple IP phones in SJC-RFD cluster	Configure H323, SCCP video IP phones, secure IP phones are as buddies in 4 SCCP legacy and TNP phones in Very Large Campus with Clustering over the WAN site cluster, check for presence status.	Phones--<html>--Unified CallManager	Passed	
CO105.CUP.007	Basic Call Flow	Mixed list of buddies like video phones, Secure SIP phones, check the Instant Messager status on multiple IP phones in DFW cluster	Configure SIP video IP phones, secure SIP phones are as buddies in 4 SIP legacy and TNP phones in Large SIP site cluster, check for presence status.	Phones--<html>--Unified CallManager	Passed	
CO105.CUP.008	Basic Call Flow	To check for different Presence CSS	To check for different Presence CSS		Failed	CSCse22635
CO105.CUP.009	Basic Call Flow	To check for different Presence CSS involving extension mobility	To check for different Presence CSS involving extension mobility in Large SIP site cluster		Failed	CSCse22635

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO105.CUP.011	Load	Bulk IPPM traffic to multiple real IP phones.	Generate bulk IPPM traffic from simulated Phones to preferably 10 real IP phones	Unified CallManager->Unified Presence Server	Passed	
CO105.CUP.012	Load	Bulk IPPM traffic for long duration involving multiple phones .	Generate bulk IM traffic involving preferably 100 simulated phones distributed over multiple DPs.	Unified CallManager->Unified Presence Server	Passed	
CO105.CUP.013	Load	Bulk generation of buddy list large number of sessions between CCM and Cisco Unified Presence Server.	This is for testing high volume of sessions established between Unified CallManager and Unified Presence Server	Unified CallManager->Unified Presence Server	Passed	
CO105.CUP.014	Load	Bulk addition deletion of buddy list.	From 10 real phones add 200 buddies total and delete them. Repeat this exercise for preferably 10 times	Unified CallManager->Unified Presence Server	Passed	
CO105.CUP.016	Basic Call Flow	To check for new Instant Messager indication on a TNP SIP phone	To check for new IM indication on a TNP SIP phone	Phones->CUPS->-Unified CallManager.	Passed	
CO105.CUP.017	Basic Call Flow	To check Cisco Unified Presence Server Admin can broadcast Instant Messager to bulk number of phones	To check the Unified Presence Server admincan can broadcast IM to bulk number of phones to few hundreds of phones.	Phones->CUPS->-Unified CallManager.	Failed	CSCse24102
CO105.CUP.018	Basic Call Flow	To check Instant Messager between IP phones behind SRST Gateway and SIP TNP ip phone.	To check IM between IP phones behind SRST GW and SIP TNP ip phone.	Phones->CUPS->-Unified CallManager.	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO105.CUP.020	Failover	To check after failure the recovery of the Cisco Unified Presence Server is successful	To check after failure the recovery of the Unified Presence Server is successful		Failed	CSCse20572
CO105.CUP.022	Interoperability	Rebooting of Primary Cisco Unified Presence Server with large number of active subscriptions with IPSec link to CCM	Setup IPSec between Unified CallManager to Unified Presence Server for Database and PE/Proxy. Setup around large buddy list in 2 real phones and engage them in a call. Reboot the Primary Unified Presence Server under this situation.		Failed	CSCse20572
CO105.UCA.001	Unified Personal Communicator	Unified Personal Communicator in SJC cluster, blind transfer(if supported) an incoming PSTN call to a IP phone in DFW cluster over SIP trunk	Unified Personal Communicator running behind SRST GW NOT in SRST mode, blind transfer an incoming PSTN call to a IP phone in another cluster.	H323->Unified CallManager->Unified Personal Communicator->SIPTrunk	Failed	CSCse08772
CO105.UCA.002	Unified Personal Communicator	Unified Personal Communicator in RFD Cluster. Set up Conference call with CME SIP and CCM SCCP phones using Unified MeetingPlace Express, sharing of documents.	Unified Personal Communicator is running in RFD cluster. Set up Conference call using MP express, sharing of documents.	Unified Personal Communicator->-Unified CallManager->Unified MeetingPlace Express->Unified CallManager Express->SIPTrunk	Failed	CSCse15895

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO105.UCA.003	Unified Personal Communicator	Bring up Unified Personal Communicator though IPSEC VPN, this Unified Personal Communicator is one of the phone in hunt group. Answer the call when it is rung.	Unified Personal Communicator is connected though IPSEC VPN, is one of the phone in hunt group. Answer the call when it is rung.		Passed	
CO105.UCA.005	Unified Personal Communicator	Unified Personal Communicator phone is used as ipccx agent in phone associate mode(sccp phone)	Unified Personal Communicator phone is used as ipccx agent in phone associate mode. Make a call to agent RP to reach an agent	Phone->Unified CallManager->Unified Personal Communicator(ipccx agent)	Passed	
CO105.UCA.006	Unified Personal Communicator	Presence Status of large number of buddies(~100). Unified Personal Communicator is running in laptop connected IPSEC VPN.	Unified Personal Communicator is running on a windows server which is connected through a IPSEC VPN to the test bed. Add buddy using Active Directory and check the status of buddies.		Passed	CSCse15922
CO105.UCA.007	Unified Personal Communicator	Presence Status of large number of buddies(~100). Unified Personal Communicator is running in a laptop connected through Cisco VPN Client 3002.	Unified Personal Communicator is running on a laptop which is connected to the test bed through a VPN using Cisco VPN Client 3002. Add buddy using Active Directory and check the status of buddies.		Passed	CSCse15922

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO105.UCA.008	Unified Personal Communicator	Unified Personal Communicator on a laptop through VPN IPSEC connection. Establish a conference call by gettting two endpoints in different clusters.	Unified Personal Communicator on a laptop through VPN IPSEC connection. Establish a conference call by gettting two endpoints in different clusters. One leg is established through H323 trunk from Unified CallManager and second leg using SIP trunk from Unified CallManager	Unified Personal Communicator->Unified CallManager->SCCP/SIP IP Phone.	Passed	CSCse16081
CO105.UCA.009	Unified Personal Communicator	Unified Personal Communicator is connected through Cisco VPN Client 3002 connection. Engage in multiparty(5) conference call using Unified MeetingPlace Express, sharing of documents etc.	Unified Personal Communicator in associated mode connected through Cisco VPN client, 3002 VPN connection, set up a conference call using MP express, sharing of documents etc.	Unified Personal Communicator->Unified CallManager->SCCP/SIP IP Phone.	Failed	CSCse15895
CO105.UCA.010	Unified Personal Communicator	Unified Personal Communicator is connected through IPSEC VPN connection, call forward on NO Answer to intercluster SIP IP phone.	Bring up a Unified Personal Communicator on a windows-xp based platform behind IPSEC VPN, set up CFNA.	Unified Personal Communicator->Unified CallManager->SCCP/SIP IP Phone.	Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO105.UCA.012	Unified Personal Communicator	Unified Personal Communicator is set for send to VM and an incoming PSTN call through H323 Gateway is sent to VM.	At Unified Personal Communicator send to voice mail is configured. Check that incoming PSTN call through H323 GW is sent to VM.	Unified Personal Communicator->Unified CallManager->SCCP/SIP IP Phone.	Passed	
CO105.UCA.014	Unified Personal Communicator	Answer the incoming PSTN call through SIP trunk at hard phone then transfer the call to a intercluster SIP phone.	An incoming PSTN call though SIP trunk to an Unified Personal Communicator in associated mode. Answer the call at hard phone and move the call to softphone. Check for moving the call back and forth between hard phone and soft phone. Finally from the hardphone, blind transfer to a intercluster SIP end point. Check for correct display of caller-id, relaying of DTMF digits, and satisfactory establishment of bearer path.	Unified Personal Communicator->Unified CallManager->SCCP/SIP IP Phone.	Failed	CSCsd96312
CO105.UCA.018	Unified Personal Communicator	Check the Presence Status of the bulk number of buddies on Unified Personal Communicator	From the Search list pick 75 buddies into a group. Make the end points to go offhook and onhook and check that the presece status at Unified Personal Communicator for those end points are updated. Check for the incoming NOTIFY messages from Unified Personal Communicator.		Failed	CSCse35706

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO105.UCA.019	Unified Personal Communicator	Unified Personal Communicator talking to Active directory using LDAP, Active Directory is running at SJC and Unified Personal Communicator at RFD, click to dial a contact.	Get a contact using LDAP and click to dial	Unified Personal Communicator->LDAP->Unified CallManager	Passed	
CO105.UCA.021	Unified Personal Communicator	Unified Personal Communicator running behind SRST Gateway come back to active state after WAN failure	An Unified Personal Communicator is running behind a SRST GW. Establish a incoming call to Unified Personal Communicator. Create a WAN failure. Verify that Unified Personal Communicator application tear down the call. Bring up the WAN connection and setup the call again		Passed	
CO105.UCA.022	Unified Personal Communicator	Unified Personal Communicator interacting with VM	Unified Personal Communicator is connected through IPSEC VPN. Dial the VM and listen for Voice Mail and delete the messages.		Passed	
CO105.UCA.023	Unified Personal Communicator	Secure TLS links to Cisco Unified Presence Server.	Unified Personal Communicator's links to Unified Presence Server thru TLS. CEO Unified Personal Communicator is in phone associated mode. Bring up the Unified Personal Communicator		Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
CO105.UCA.024	Unified Personal Communicator	Secure TLS links to Cisco Unified Presence Server, Unified Personal Communicator behind IPSEC VPN connection	Unified Personal Communicator's links to Unified Presence Server thru TLS. CEO Unified Personal Communicator is in soft phone mode. Bring up the Unified Personal Communicator		Passed	
CO105.UCA.025	Unified Personal Communicator	Unified Personal Communicator in phone associated mode behind a VPN concentrator CVPN 3002, and Unified Personal Communicator is a MLPP subscriber.	Unified Personal Communicator's links to Unified Presence Server thru TLS, The Unified Personal Communicator is a CEO number and it's status is monitored by around 1000 users. The Unified Personal Communicator is behind a hardware CVPN 3002. Unified Personal Communicator is in a MLPP domain. Bring up the Unified Personal Communicator and make a call.		Passed	
CO105.UNI.000	Voice Mail	Direct calls to Unity through a SIP gateway	Verify that direct calls to Unity from a PSTN through a SIP gateway is successful with the codec type negotiated as OOB NOTIFY.	PSTN Phone->SIP Gateway->SIPT->Unified CallManager->SIPT-> Unity	Failed	CSCse22729
GB31.CM.157	Extension Mobility	IP to IP Intra-cluster Extension Mobility.	IP to IP Intra-cluster Extension Mobility.		Passed	
GB31.CX.268	Voice Mail	CUE subscriber mail box features, message deposit, retrieve, delete etc.	CUE subscriber mail box features, message deposit, retrieve, delete etc.		Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
GB31.ER.276	Cisco Emergency Responder	Cisco Emergency Responder Phone Discovery and IP-to-IP Cisco Emergency Responder Calls	Cisco Emergency Responder phone discovery and IP-to-IP Cisco Emergency Responder.		Passed	
GB31.ER.282	Cisco Emergency Responder	IP-to-IP Intra-Cluster Cisco Emergency Responder Calls using Extension Mobility	IP - IP intra-cluster Cisco Emergency Responder calls using extension mobility.		Passed	
GB31.WL.746	Wireless	Three-party Ad Hoc Conference Call Between Wireless and Desktop IP Phones Started by Cisco Unified IP Phone 7920 Registered to Unified CallManager Express	Verify the ability to establish a 3-party ad hoc conference call from a Cisco Unified IP Phone 7920 registered to Unified CallManager Express that involves other Unified IP Phone 7920 phones and desktop IP phones.		Passed	
GB31.WL.747	Wireless	Meet-Me Conference Call Between Wireless and Desktop IP Phones Started by Cisco Unified IP Phone 7920 Registered to Unified CallManager	Verify the ability to establish a Meet-Me conference call from a Cisco Unified IP Phone 7920 registered to Unified CallManager Express that involves other Unified IP Phone 7920 phones and desktop IP phones.		Passed	
GB40.CX.001.07	Voice Mail	Cisco Unified CallManager Express Overlay: Directory Number and Called Directory Number Display Feature	Verify that when a phone is configured for overlay-directory numbers (DNs), multiple DNs are mapped to a single physical line on a phone.		Passed	

ID	Features Tested	Title	Description	Call Component Flow	Status	Defects
GB40.ER.009	Cisco Emergency Responder	IP-to-IP Intra-Cluster Cisco Emergency Responder Calls using Authenticated Phones	Verify that Cisco Emergency Responder can properly discover phones that have been configured for authentication and that calls are routed to the PSAP with proper automatic number identification.		Passed	
GB40.MP.025.09	Unified MeetingPlace	Attend Basic Mixed Audio and Video Meeting: Remote Out-dial, Gatekeeper ICT, SIP MeetingPlace IP Gateway	Verify the ability to attend and control a multi-endpoint mixed audio and video conference with all calls remote via gatekeeper ICT trunk, no encryption, no call treatment, calls originate via outbound dialing using SIP MeetingPlace IP Gateway.		Passed	