



日本市場向け Cisco Unified Communications 10.0 システム リリース テスト結果サマリ

2014 年 5 月
シスコ システムズ 合同会社

コンポーネント一覧 (1)

Applications	Component		Version	
Call Control	Cisco Unified Communications Manager	Version	10.0.1.10000-24	
		Locale	10.0.1.9902-70(JP)	
		Dial Plan	3-1-9.JP	
	Cisco Unified Communications Manager Express			10.0(1)
		IOS		15.4(1.24)T0
		Locale		10.0.2.7(JP)
	Cisco Unified Survivable Remote Site Telephony (SRST)			10.0(1)
		IOS		15.4(1.24)T0
	Cisco Telepresence Video Communication Server(VCS)			X8.1
		Locale		X7.2_LanguagePacks_BETA

コンポーネント一覧 (2)

Applications	Component		Version
Applications	Cisco Unified Attendant Console		10.0.1.10-63
	Cisco Unified Communications Manager IM and Presence Service		10.0.1.10000-26
		Locale	10.0.1.1000-1(JP)
Voice Mail and Unified Messaging	Cisco Unity Connection		10.0.1.10000-24
		Locale	10.0.0.1-1(JP)
End Point	Unified IP Phones 6921/41/45/61		9.4.1-3
	Cisco Unified IP Phones 7942/62/75		9-3-1SR4-1
	Cisco Unified IP Phone 7925		1.4(5)
	Cisco Unified IP Phone 8831		9-3-3-5
	Cisco Unified IP Phone 7821/41/61		10.1.1.9
	Cisco Unified IP Phones 8941/8945		9.4.1-8
	Cisco Unified IP Phones 9951/9971		9.4.1-9
	Cisco Desktop Collaboration Experience DX650		10-2-1JBT0-222
	Cisco UC Integration™ for Microsoft Lync		9.6.0.621

コンポーネント一覧 (3)

Applications	Component		Version
End Point	EX60 - Cisco TelePresence System EX60		TC 7.0.2
	EX90 - Cisco TelePresence System EX90		TC 7.0.2
	SX20 - Cisco TelePresence SX20 Quick Set		TC 7.0.2
	C20 - Cisco TelePresence System Quick Set C20		TC 7.0.2
	Cisco TelePresence System Integrator Package C90		TC 7.0.2
	500-32 Cisco Telepresence system 500-32		TX6.1.1(50)
	Cisco TX9000 -Cisco TelePresence System TX9000		TX6.1.1(50)

コンポーネント一覧 (4)

Applications	Component		Version
Communications Infrastructure	Gateways	IOS	15.4(1)T1
	Cisco Unified Border Element for ISR		10.0.0
	Fabric Interconnect PRIMARY	Cisco UCS 6140	2.1.1a
	Fabric Interconnect SUBORDINATE	Cisco UCS 6140	2.1.1a
	Fabric Cluster	Cisco UCS 6140	2.1.1a
	ESXi host		ESXi 5.1.0
	Vcenter Server		ESXi 5.1.0
	MDS switch	M9500	5.2(2a)
	Cisco Analog Telephone Adaptor	ATA 187	9.2.3
	Cisco 3750 PoE Switch		15.0.2-SE5
TelePresence	Cisco Telepresence Management Suite		14.3.2
	Cisco Telepresence MCU	4510	4.4(3.67)
		5310	4.4(3.67)
		Locale	MCU_4-3_UI_and_audio_JPNpackage
	Cisco Telepresence Server on VM		3.1(1.96)
	Cisco Telepresence Conductor		XC2.2.1
	Cisco Telepresence Server 7010		3.1(1.97)

コンポーネント一覧 (5)

Applications	Component		Version	
Wireless and Mobility	Wireless LAN Controller 2504		7.6.110.0	
	Wireless Access Point 1142		15.2	
	Wireless Access Point 35XX		15.2	
	Cisco Jabber for Mac		9.2.2.165297	
	Cisco Jabber for Windows		9.6.0.17088	
	Cisco Jabber for iPhone and iPad		9.6.1.163935	
		iPhone 5		Apple iOS 7.0.4
		iPad		Apple iOS 7.0.4
	Cisco Jabber for Android			9.6.0.165769
Galaxy SII and S4			Android OS 4.2.2	
Windows Client	Operating System	Windows 7 - SP1	Windows 7 - SP1 (Japanese)	
		Windows 8	Windows 8(Japanese)	
		Mac	10.8	

コンポーネント一覧 (6)

Applications	Component		Version
Client	Browser	IE	IE 9,10 (Supported Japanese language)
		Mozilla	Firefox 27.0.1 (Supported Japanese language)
		Chrome	Chrome 33.0 (Supported Japanese language)
	Microsoft Lync Client		2010
Server	Windows Server		Windows Server 2008 (R2 Enterprise - Japanese)
			Windows Server 2012
	Exchange Server		2010
	Lync Server		2010

テスト結果

対象コンポーネント	試験項目数	合格数	不合格数	不具合数
Cisco Unified Communications Manager	1001	992	9	9
Cisco Unity Connection	74	74	0	0
Cisco Unified Border Element	63	63	0	0
Cisco Unified Communications Manager Express	62	62	0	0
Cisco Unified Survivable Remote Site Telephony	32	32	0	0
Cisco Unified Attendant Console	142	142	0	0
Cisco Jabber for MAC	55	55	0	0
Cisco Jabber for Windows	145	145	0	0
Cisco Jabber for iPhone	96	96	0	0
Cisco Jabber for Android	132	132	0	0
Cisco Jabber for iPad	43	43	0	0
Cisco UC Integration™ for Microsoft Lync	165	165	0	0
Cisco TelePresence Multipoint Control Unit	62	62	0	0
Cisco TelePresence Management Suite	59	59	0	0
Cisco TelePresence Video Communication Server	55	49	6	6
Cisco TelePresence Conductor	497	496	1	1
Cisco Telepresence Server	157	157	0	0
Cisco Video Communication Server Expressway	27	27	0	0
Total	2867	2851	16	16

不具合一覽

Sl.No	Defect ID	Status
Cisco Unified Communications Manager		
1	CSCuj76467	N-New
2	CSCuj93496	N-New
3	CSCum79058	N-New
4	CSCum81613	N-New
5	CSCum81627	N-New
6	CSCum82008	N-New
7	CSCum81985	N-New
8	CSCun02402	N-New
9	CSCun27648	N-New

不具合一覽

Sl.No	Defect ID	Status
Cisco VCS		
1	CSCul13054	O-Opened
2	CSCul01896	O-Opened
3	CSCum74055	N-New
4	CSCun12798	A-Assigned
5	CSCun28040	N-New
6	CSCun08178	A-Assigned

不具合一覽

Sl.No	Defect ID	Status
Cisco Tele Presence Conductor		
1	CSCul04054	N-New

CSCuj76467: Audible Message Waiting indicator is not working in 8831 IP Phone(N-New)

Steps to Reproduce :

- Register 8831 IP phone in Unified CM
- Configure voicemail to 8831 IP Phone.
- Enable AMWI in Service Parameters.
- Enabled the visual message waiting Indicator and Audible Message Waiting Indicator options.
- Make a call to 8831 IP Phone and send the voicemail.

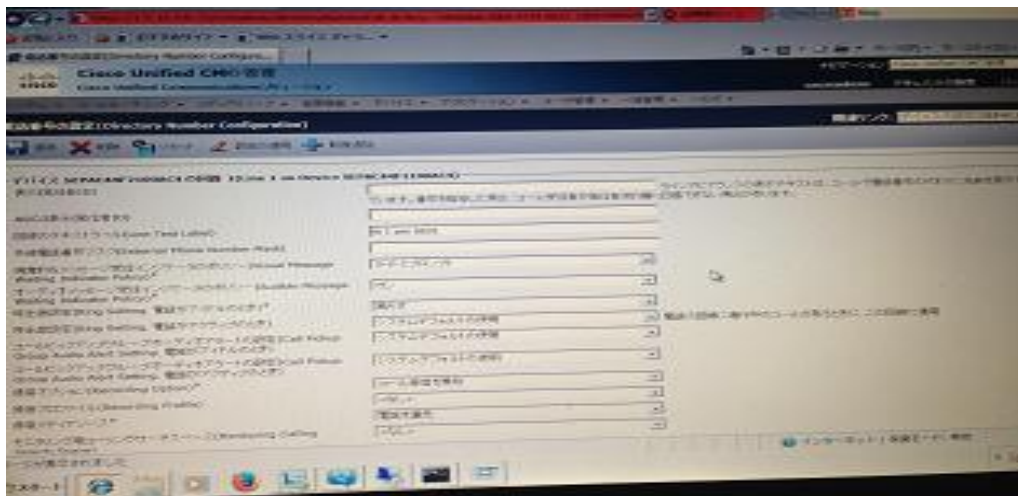
Problem Description : When receiving voicemail in 8831 IP Phone there is only message waiting indicator no Audible MWI.

Expected Behaviour : when receiving a voicemail both MWI and AMWI will be there

Environmental Matrix :

CUCM version : 10.0.0.97027-12

8831 Firmware : 9-3-3-5



CSCuj93496: "Enter the number to transfer" option is not found in 8831 (N-New)

Issue-Description:

During transfer, user is not prompted with option "Enter the number to transfer" after pressing the transfer softkey but the same is found in other IP Phones.

Environment:

Build details of Unified CM: 10.0.0.97028-14
Phone Firmware : sip8831.9-3-3-5

Steps to reproduce:

- 1) Register 8831 with Unified CM 10.0
- 2) Make a call from 8831 to any other IP Phone A
- 3) Answer the call at IP Phone A. 8831 and IP Phone A are connected now
- 4) Transfer the call from 8831 to IP Phone B by pressing the "Transfer" softkey in 8831.
- 5) "Enter the number to transfer" option is not displayed in 8831 once after pressing the Transfer softkey.

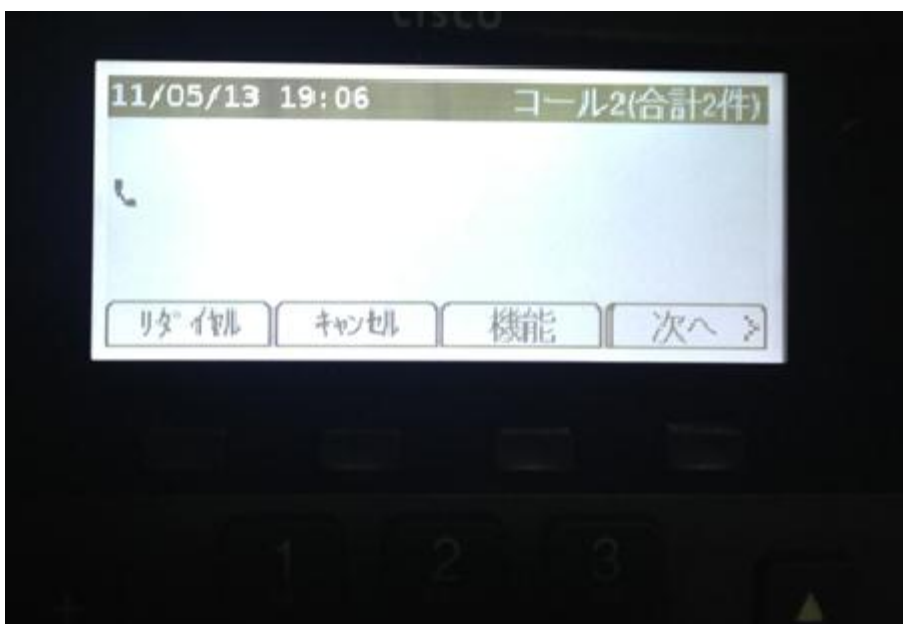
Actual Behaviour:

User is not prompted with the option "Enter the number to transfer" in 8831 after pressing the transfer softkey

Expected Behaviour:

The option should be prompted in order to guide the user while making transfer

Screen Shot1:



CSCum79058 : Details showing wrongly at the time of conference in DX650 (N-New)

Issue-Description:

Details showing wrongly at the time of conference in DX650.

Environment:

Unified CM Build: 10.0.1.10000-24

DX650 Build: sipdx650.10-1-1-78

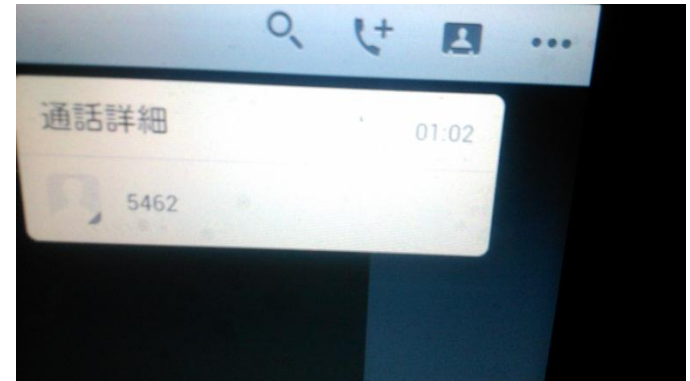
Step to reproduce:

- Install a Unified CM and register DX650 Phones and other Unified IP Phones.
- Let Phone C be DX650 and Phone A and Phone B be any other Unified IP Phones.
- Make a call from Phone A to Phone B. Answer the call in Phone B.
- Now hard-key conference button in Phone B and dial Phone C DN.
- Answer the call in Phone C and now again press conference in Phone B to complete the conference.

Actual Behavior: Now the three phones are in conference, and if hold is pressed in DX650 and if I press the "Details" means it is showing only the DN of Phone B. Phone A details is not displayed. Note: In the above scenario DX650 is not the initiator of conference, but if suppose DX650 is the initiator of the conference means(i.e. when the Phone A calls to Phone C and Phone C starts the conference and joins Phone B) at that time if hold is pressed, and in "Details" it is showing the details of all the three phones.

Expected Behavior:

At the time of conference if hold is pressed also it needs to show the details of all the three phones that are in conference.



CSCum81613 : FOR DN is displayed wrongly in DX650 when CFA is set. (N-New)

Issue-Description:

FOR DN is displayed wrongly in DX650 when CFA is set.

Environment:

Unified CM : 10.0.1.10000-24

DX650 load : Sipdx650 10.1.1.78

Step to reproduce:

- Install a Unified CM and Register Unified IP Phones and DX650 Phones in that Unified CM.
- Unified IP Phone A(5498), Unified IP Phone B(5462), Unified IP Phone C(5104), Unified IP Phone D(5469), DX650(5484)
- Set CFA in Phone B to Phone C, set CFA in Phone C to Phone D, set CFA in Phone D to DX650.
- Make a call from Phone A to Phone B. Since CFA is set the call is routed to DX650. So DX650 rings.
- In DX650 it is displaying calling phone DN as 5498 (From 5469, For 5462).
- Now answer the call in DX650 and end the call.

Actual Behavior:

After ending the call go to RECENT tab in DX650 and check the call history of this call under ALL calls. Calling DN will be 5498 in that From will be 5469 and For will be 5498. From DN is displaying correctly but the For DN it is wrongly displaying the DN of calling party DN.

Expected Behavior:

In that Recent call history For DN should be 5462. But it is wrongly displayed.



CSCum81627 : Conference members count is not decreasing in DX650 when in conference (N-New)

Issue-Description:

Conference members count is not decreasing in DX650 when in conference

Environment:

Unified CM: 10.0.1.10000-24

DX650 phone load: Sipdx650 10.1.1.78

Step to reproduce:

- Install an Unified CM and register Unified IP Phones and DX650 in that Unified CM.
- Unified IP Phone A(5462), Unified IP Phone B(5104), Unified IP Phone C(5469), DX650(5484)
- Make a call from Phone A to Phone B. Now initiate conference in Phone B to Phone C. Answer the call in Phone C.
- Now Phone A, B and Phone C are in conference. Now initiate conference in Phone C to DX650. Answer the call in DX650.
- Now all the four phones are in conference. In DX650 it is showing the conference initiator DN and 3 others in conference.

Actual Behavior:

Now end the call in Phone D. Still in DX650 it is showing as 3 others in conference. The count needs to be decreased and it should show as 2 others in conference.

Expected Behavior :

When a call is disconnected in conference, the conference members count in the DX650 should be decreased.



CSCum82008 : No conference timer on 6921/61 phones while pressing Details soft-key (N-New)

Issue-Description:

Display issue on 6921 and 6961 phones while pressing Details soft-key before answering conference call i.e, No Timer is displayed for Conference call on the phones when following the below procedure.

Environment:

Build details of Unified CM: 10.0.1.10000-24
Phone Firmware :SIP69xx.9-4-1-3

Steps to reproduce:

- 1) Make call from Phone A to Phone B. Initiate Conference call from Phone B to Phone C (69xx using SIP load).
- 2) Complete the conference at Phone B. Now in Phone C press the Details Soft-key before answering the conference call.
- 3) Next do not press the back soft-key wait for the notification No Caller Details Available to go off.
- 4) Now press Answer soft-key in Phone C to attend the conference call.
- 5) Check for the display of Phone C.

Screen Shot1:



Actual Behavior:

No Timer is displayed for Conference and improper display is found on phones

Expected Behavior:

The phone should display the timer for Conference call and the display should remain same as before.

CSCum81985: Unable to answer conference in SCCP load after pressing Details soft-key (N-New)

Issue-Description:

Unable to answer the conference call in SCCP phone load of 6921/6961 phones after pressing the Details soft-key

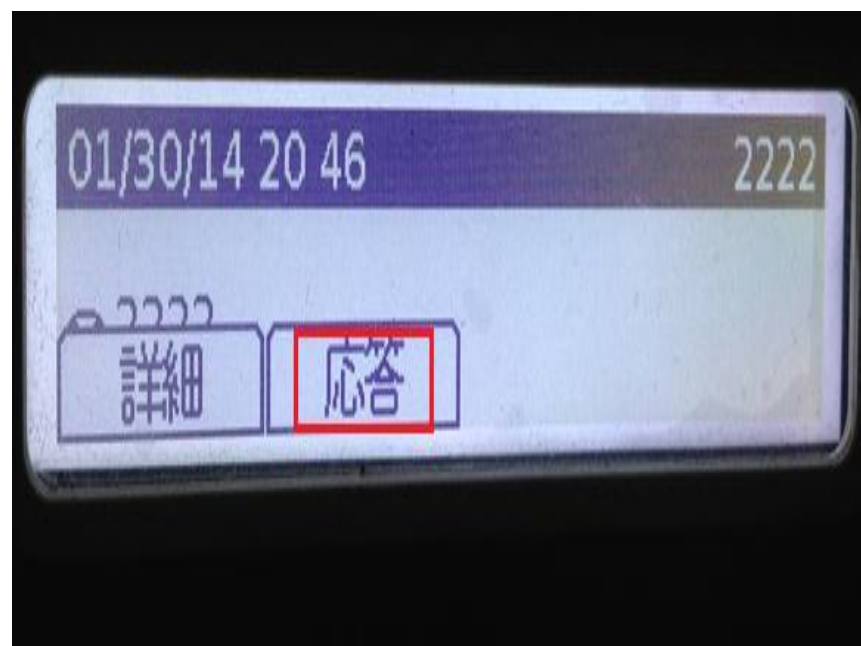
Environment:

Build details of Unified CM: 10.0.1.10000-24
Phone Firmware :SCCP69xx.9-4-1-3

Steps to reproduce:

- 1) Make call from Phone A to Phone B. Initiate Conference call from Phone B to Phone C (69xx using SCCP load).
- 2) Complete the conference at Phone B. Now in Phone C press the Details soft-key before answering the conference call.
- 3) Next do not press the back soft-key wait until the notification No caller Details available goes.
- 4) Once after the notification has gone, now press Answer soft-key in Phone C to attend the conference call. (Unable to answer the call)

Screen Shot1:



Actual Behavior:

Unable to answer the conference call in SCCP load after pressing the details soft-key

Expected Behavior:

User should be able to answer the conference call on SCCP load even after pressing the details soft-key

CSCun02402: Phone display language is not updated immediately to phones (N-New)

Issue-Description:

The phone display language is not updated immediately after applying the language to phones via 10.0.1 Self-Care portal

Environment:

Build details of Unified CM: 10.0.1.10000-24

Phone Firmware: SIP69xx.9-4-1-3

Japanese Locale: 10.0.1.9902-70

Steps to reproduce:

- 1) Assign a user to any one of the IP Phones in Unified CM
- 2) Open Self Care Portal using <http://UCMIP/ucmuser>
- 3) Login with owner user id and password
- 4) Navigate to General Settings -> Language. Under which select the display language either to be Japanese or English
- 5) Click Save to save the changes made. The notification Language settings have been successfully saved is also displayed. (Means, the Self-Care DB is updated)
- 6) Check for the display on the particular phone. Still the language remains the same with no change.

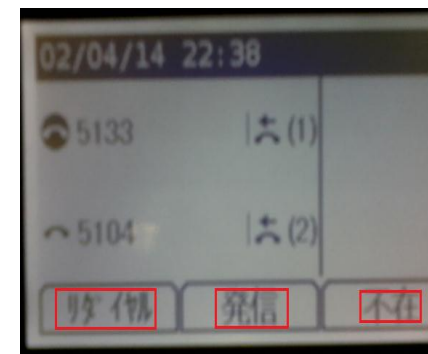
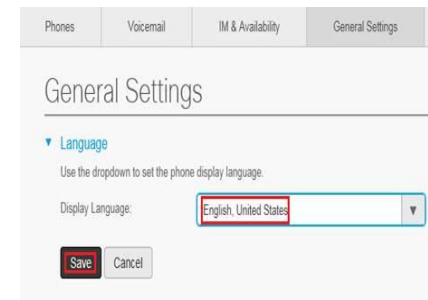
Actual Behavior:

Phone display language is not applied to phones immediately when done via 10.0.1 Self-Care

Expected Behavior:

It is expected for the phones to reflect the display language changes immediately

Screen Shots:



CSCun27648: 78xx:Phone do not get any ringtone when Default(Ring) is set to phones (N-New)

Issue-Description:

Phone does not get any ringtone when Default(Ring) is set to phones under When I'm on a call tab via Self-Care

Environment:

Unified CM : 10.0.1.10000-24

Phone Firmware : sip78xx.10.1.1.9

Steps to reproduce:

- 1) Login Self-care with right credentials
- 2) Navigate to Phones -> Phone Settings -> Ring Settings
- 3) Select Default(Ring) for Phone A (78xx) under When I'm on a call tab and save the changes
- 4) Make call from any phone to that particular phone (Phone A) to which the ring settings are applied
- 5) Answer the call at Phone A. Both the phones are connected now
- 6) Again make a call from any other IP Phone to Phone A.
- 7) Check for the ringing status on Phone A

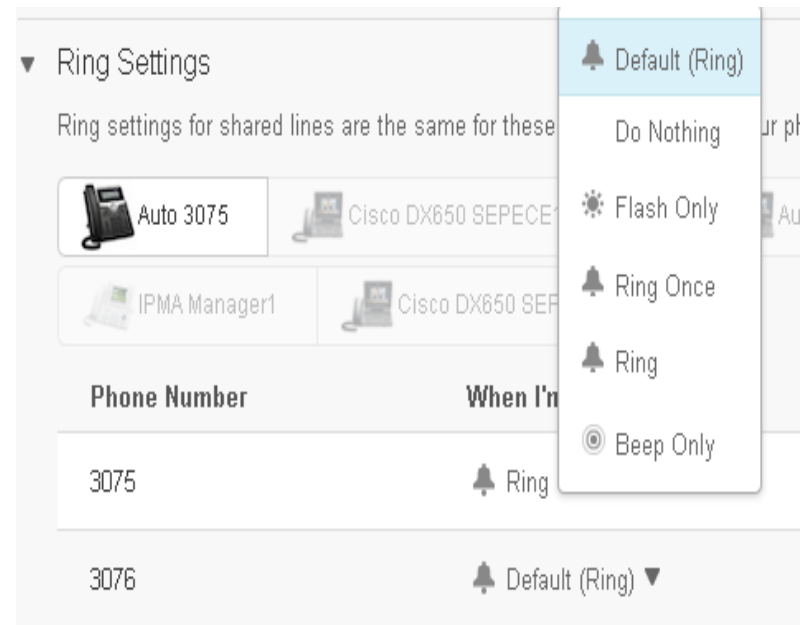
Actual Behavior:

Phone A is not getting any ring tone when Default(Ring) is set to phones via Self-Care and when on active call

Expected Behavior:

It is expected for the phones to get ringtone when on active call when Default(Ring) is set.

Screen Shot:



CSCu13054 : SX20 Quickset Touch-UI is Turning off After enabling Extended logging in SX20 Quickset Web-Gui (O-Opened)

Issue- Description: SX20 Quickset Touch-UI is Turning off After enabling Extended logging in SX20 Quickset Web-Gui

Environment:

SX20 Quickset:- TC6.3 RC3

CUCM: 10.0.1

Steps to Reproduce :

- SX20 Quick set registered with Unified CM
- login into SX20 Quick set Web Interface
- Goto log files under diagnostics tab
- Click Start Extended logging for 5 min's
- After enabling extended logging option 30 sec or 1 min SX20 Quick set Touch-UI is Turning off state and Turning up state after few seconds.

Expected Behavior:

SX20 Quick set Touch-UI Should not Turn off after enabling Extended logging option

CSCu01896 : Unable to display "Sign In Successful" message in C20 Touch UI for EM user(O-Opened)

Issue-Description: Unable to display "Sign In Successful" message in C20 Touch UI for EM user

Steps to Reproduce:

- Register C20 in CUCM
- Configure Extension Mobility in CUCM and associate with C20
- Extension Mobility User ->User locale Set to Japanese
- Login with Extension Mobility User in C20
- In C20 Touch UI it shows changing language and logged with EM User
- But unable to show "Sign In Successful " Message in C20 Touch UI

Environment:

C20 :TC6.3 RC2

CUCM: 10.0.1.10000-5

Expected Behavior :

“Sign In Successful “ message have to display in C20 Touch UI

CSCum74055: Unable to transfer call from EX90 registered in Cisco VCS (N-New)

Issue-Description: When there is call between EX90 registered in Cisco VCS and EX60(1) registered in Unified CM and EX90 transfers the call to another EX60(2) registered in Cisco VCS , the transfer fails and gives call is rejected message

Steps to Reproduce:

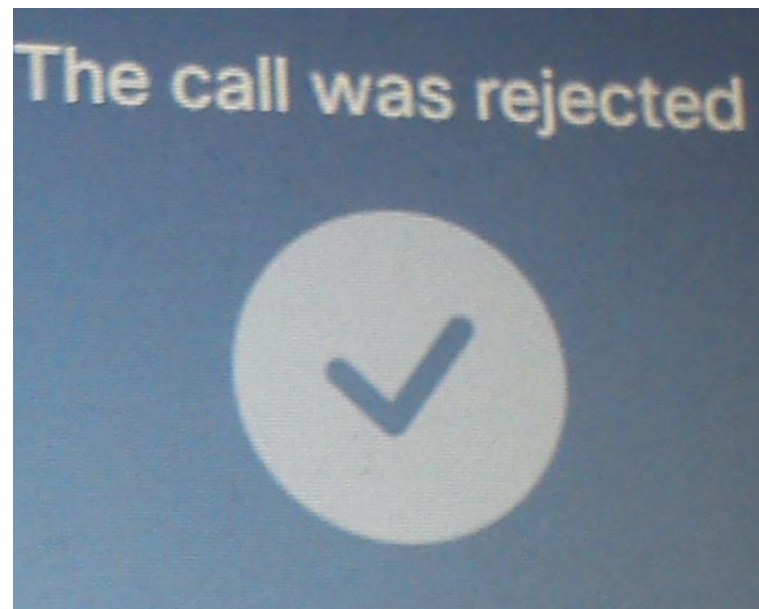
- Register EX90 and EX60(2) in Cisco VCS
- Register EX60(1) in Unified CM
- Make a call between EX90 and EX60(1)
- Press transfer and transfer the call to EX60(2)
- Transfer fails and call is rejected message is shown

Environment:

EX90:TC7.0.1, VCS:X8.1, 10.0.1.10000-24

Expected Behavior:

The transfer should work between Cisco VCS and Unified CM calls



CSCun12798 : Japanese display ID not coming in SX20 registered in Cisco VCS (A-Assigned)

Issue-Description: When EX60 phone with Japanese display ID registered in Unified CM calls SX20 Quickset registered in Cisco VCS, SX20 Quickset does not display the Japanese display ID of EX60. But if EX60 has English display ID, SX20 Quickset displays the display ID

Environment:

SX20 Quickset ->TC7.0.2, EX60 -> TC7.0.2, Unified CM -> 10.0, Cisco VCS ->X8.1

Steps to reproduce:

- Register SX20 Quickset in Cisco VCS
- Register EX60 in Unified CM
- In EX60 phone set Display ID in Japanese
- Make a call from EX60 to SX20
- SX20 does not display the Japanese display ID
- But if English display ID is set IN EX60, SX20 is able to display the display ID

Expected Behavior:

After call, SX20 Quickset should show the Japanese Display ID of EX60

CSCun28040 : SX20 transmitting video after setting default call rate as 64 kbps (N-New)

Issue-Description: When we set the default call rate as 64 Kbps in the web-gui of SX20 and now when we make a call from SX20 to EX90, SX20 is transmitting video.

Environment:

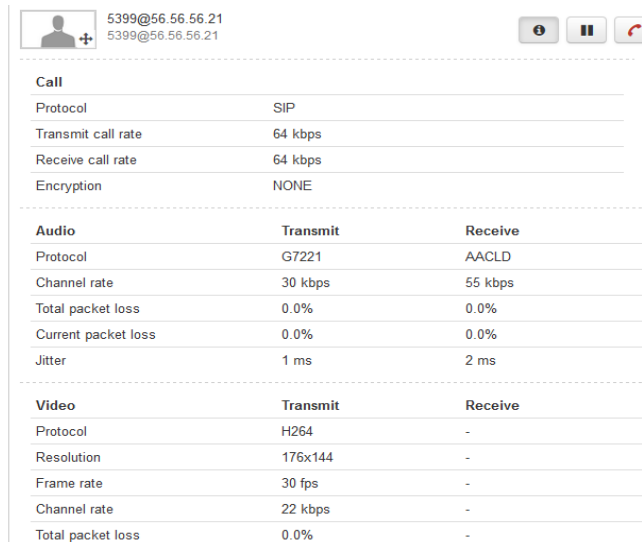
SX20 Quickset ->TC7.0.2, EX90->TC7.0.2, Unified CM -> 10.0

Steps to reproduce:

- Register SX20 Quickset in Unified CM
- In SX20 Quickset web-gui, go to Configuration -> System Configuration
- Go to Conference and change the default call rate to 64 Kbps
- Now in the Touch UI check the bandwidth rate for the call.
- The bandwidth rate shows 64 Kbps and audio, eventhough 64 Kbps is the lowest bandwidth for the call

Expected Behavior:

At 64KBPS only audio should be transmitted



The screenshot shows a call interface with a contact card at the top displaying the name '5399@56.56.56.21' and a plus icon. Below the contact card are three icons: a speaker, a pause button, and a red phone icon. The main content area is divided into three sections: 'Call', 'Audio', and 'Video', each with a table of metrics.

Call	
Protocol	SIP
Transmit call rate	64 kbps
Receive call rate	64 kbps
Encryption	NONE

Audio	Transmit	Receive
Protocol	G7221	AACLD
Channel rate	30 kbps	55 kbps
Total packet loss	0.0%	0.0%
Current packet loss	0.0%	0.0%
Jitter	1 ms	2 ms

Video	Transmit	Receive
Protocol	H264	-
Resolution	176x144	-
Frame rate	30 fps	-
Channel rate	22 kbps	-
Total packet loss	0.0%	-

CSCun08178: Complete Transfer option is not coming while transfer using Start new(A-Assigned)

Issue-Description: Complete Transfer option is not coming while transfer using Start new

Environment:

EX60 & EX90 : TC 7.0.2

Cisco VCS : X8.1

Steps to Reproduce :

- Register EX60 (1) ,EX60(2) and EX90 with Cisco VCS
- Make a call between EX60(1) and EX90
- Attend a call in EX90
- Press transfer option in EX90
- Press letter's Key pad Type DN of EX60(2) and press Start new option

Expected Behavior:

Complete transfer option should come while doing transfer using start new option

CSCul04054: 8941 IP Phones get reboot after transfer the conference call from EX/SX(N-New)

Issue- Description: 8941 IP Phones get reboot after transfer the conference call from EX/SX

Environment:

- CUCM -> 10.0, TP Conductor -> XC2.2, EX/C Series -> TC6.3, 8941 -> SCCP8941_8945.9-3-4-17

Steps to Reproduce :

- Register EX/C Series and IP Phone 8941 in CUCM
- Integrate CUCM with Telepresence Conductor
- Telepresence conductor is integrated with Media Resources (Cisco MCU/ Cisco TP Server)
- Configure rendezvous conference SIP Trunk from CUCM to Telepresence conductor
- Dial the rendezvous DN from EX/C Series End points
- Reach the conference media resource
- Then transfer the conference call from EX/C Series End points to 8941 IP Phone
- Press complete transfer on EX/C Series End points and check the conference status in 8941

Expected Behavior:

After transferring the call from EX/C Series End points to 8941 ip phones, IP Phone 8941 should be participate in the existing rendezvous conference.

Thank you.

