配置Informacast並對其進行故障排除

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簡介

本檔案介紹思科呼叫伺服器產品(也稱為InformaCast),以及如何將其與思科整合通訊管理員 (CUCM)整合。本文檔將介紹該功能的用途、配置該功能、收集哪些資料以進行故障排除、資料示 例分析以及更多研究的相關資源。

必要條件

需求

思科建議您瞭解以下主題:

- 思科整合通訊管理員
- InformaCast
- SIP、CTI、Http和SNMP協定。

採用元件

本文中的資訊係根據以下軟體和硬體版本:

- InformaCast版本11.5.2 38
- CUCM版本11.5.1.14900-8
- CP-8811和CP-8861 sip88xx.12-0-1SR1-1
- •基本許可證

本文中的資訊是根據特定實驗室環境內的裝置所建立。文中使用到的所有裝置皆從已清除(預設))的組態來啟動。如果您的網路正在作用,請確保您已瞭解任何指令可能造成的影響。

背景資訊

功能的用途

思科呼叫伺服器是適用於數千台電話、揚聲器和其他裝置的尋呼/群發通知解決方案。這在具有即時 、預先錄製的音訊和/或文本通知的緊急情況下尤其有用。

根據與Singlewire(InformaCast供應商)簽訂的原始裝置製造商(OEM)協定,思科技術支援中心 (TAC)支援從8.3版以及CUCM 8.5版及更高版本開始的InformaCast。Cisco TAC支援的唯一模式是 基本尋呼。

基本與高級

基本尋呼模式僅支援每個接收方組最多50個電話的即時音訊廣播,不需要額外的許可證。作為 CUCM的一部分提供的InformaCast版本包括基本尋呼模式的許可證。需要補充功能的客戶可以升級 到高級通知模式並受Singlewire支援。

高級尋呼許可證允許無限尋呼組。它還可以實現其他高級功能,包括呼叫到額外開銷的模擬和IP揚 聲器、鈴聲排程、使用呼叫插入選項區分緊急通知優先順序、預錄製和純文字檔案頁面、與社群媒 體網站整合以通知功能、電子郵件和簡訊服務(SMS)批次通知和全數字監控、緊急服務警報以及與 Cisco Jabber客戶端整合。安裝InformaCast後,您可以啟用「高級通知」模式的試用版。

使用的協定

思科呼叫伺服器使用SIP、SNMP、AXL和CTI與Unified CM通訊,並且從思科呼叫伺服器9.0.1開始 ,HTTP或JTAPI可用於與電話通訊。

思科呼叫伺服器使用SNMP查詢其他Unified CM節點以及註冊到每個集群成員的電話清單。 SNMP通訊完成後,思科呼叫伺服器使用AXL確定有關每個註冊電話的其他資訊,例如裝置名稱、 說明、裝置池、呼叫搜尋空間、目錄號和位置。此資訊可用於構建電話的邏輯組,稱為收件人組。 如前所述,在具有基本許可證的思科呼叫伺服器中,收件人組最多可以包含50部電話。

附註:每個Unified CM集群支援單個思科呼叫伺服器。

HTTP與JTAPI

9.x以前的InformaCast版本都使用HTTP啟用電話。在HTTP模式下,思科呼叫伺服器向每個IP電話 HTTP伺服器傳送命令和憑證。IP電話驗證這些憑證,然後執行命令。 在廣播傳送時 ,InformaCast通過HTTP直接通過XML服務介面(XSI)聯絡他們。

在JTAPI模式下,思科尋呼通過Unified CM向每台電話傳送命令。思科呼叫伺服器不需要為每個請求傳送憑證,因此每個電話不必啟用其Web伺服器,命令執行速度更快。此外,CTI模式允許更快 速地檢查並啟用繁忙電話。

無論與CUCM的整合型別(SIP或CTI),都可以使用HTTP或JTAPI。請記住,在非英語語言環境的 電話上,JTAPI比HTTP更有效。若要確認使用者區域設定,請檢視電話網頁。

cisco		Network setup Cisco IP Phone CP-8861 (SEP2C3124C9F8E1)
Device information	MAC address	2C3124C9F8E1
Network setup	Host name	SEP2C3124C9F8E1
Network statistics	Domain name	
Ethernet information	DHCP server	10.1.61.10
Access	BOOTP server	No
Network	DHCP	Yes
Device logs	IP address	10.1.61.12
Console logs	User locale	English_United_States
Core dumps	Network locale	United_States
Status messages	User locale version	11.0.0.0(1)
Debug display	Network locale version	11.0.0.0(1)

附註:要使用JTAPI,請考慮CUCM版本必須是9.1.2或更高版本,並且不支援Cisco 3905、7902、7905、7912電話。

SIP與CTI

Informacast可以通過CTI和/或SIP接收呼叫。對於CTI,呼叫在CTI路由點上處理(思科尋呼伺服器 不需要CTI埠來應答入站呼叫)。 對於SIP,呼叫在SIP中繼上離開Unified CM。CTI和SIP均有效且受支援。但是,思科建議通過 CTI進行SIP呼叫流,因為SIP整合故障排除比CTI容易得多。

組態

網路圖表



- 1. 呼叫方(尋呼發起方)撥打了Unified CM中的預定義號碼。例如7777。
- 2. Unified CM通過SIP中繼或CTI路由點將呼叫路由至思科尋呼伺服器。
- 3. 思科呼叫伺服器應答呼叫。
- 4. 來電者聽到低聲阻音。當思科呼叫伺服器播放此提示音時,會通過HTTP或JTAPI將指令傳送到收件人組中的每台電話以加入組播組。
- 5. 所有電話加入組播組後,思科呼叫伺服器會播放高音。當呼叫方聽到此音時,表示思科呼 叫伺服器已準備好接收音訊並將其傳送到組播IP和埠。
- 當呼叫者發言時,媒體從呼叫者的電話傳送到思科尋呼伺服器,然後從尋呼伺服器傳送到 組播IP地址和埠,最後從組播IP傳送到接收電話。

7. 當呼叫者掛斷時,指令將傳送到每個IP電話,這次將離開組播組,廣播結束。

InformaCast使用JTAPI庫和電腦電話整合(CTI)管理器與Cisco Call Manager整合時,它使用TCP上的快速緩衝區編碼(QBE)協定,如下圖所示。



對於SIP整合,InformaCast使用TCP上的SIP協定和埠5060與Call Manager進行通訊,如下圖所示



配置呼叫管理器

步驟1. 啟用服務,導航到Cisco Unified Serviceability > Tools > Service Activation並啟用以下服務:

- Cisco CallManager
- Cisco CTIManager
- Cisco AXL Web服務
- Cisco CallManager SNMP服務

提示:在所有節點上啟用SNMP,在群集中至少一個節點上啟用AXL,在至少一個運行Call Manager服務的節點上啟用CTI Manager(或者為冗餘目的啟用更多節點)。

步驟2.設定SNMP(版本2或版本3)

對於SNMP v2

- 導覽至Cisco Unified Serviceability > SNMP > v1/v2。
- •使用只讀訪問許可權配置社群字串名稱。
- •如果可能,請選中「應用到所有節點」覈取方塊,然後按一下儲存。

Status : Ready	
erver* 10.1.61.158CUCM Voice/Video	Ŧ
Community String Information	
Community String Name* ICVA	
Host IP Addresses Information	
Accept SNMP Packets from any host	 Accept SNMP Packets only from these hosts Host IP Address
	Insert
	Host IP Addresses
	· ·
	Remove
Access Privileges	
Access Privileges* ReadOnly	T
(I) Matify access privileges is required in orde	r to configure Notification Destinations.

對於SNMP v3

- 導航到Cisco Unified Serviceability > SNMP > V3 > User,建立名為ICVA的使用者。
- 啟用Authentication Required覈取方塊,輸入身份驗證密碼並選擇SHA單選按鈕。
- 啟用Privacy Required覈取方塊, 輸入隱私密碼並選擇AES128單選按鈕。
- 從Access Privileges下拉選單中選擇ReadOnly,並選中Apply To All Nodes覈取方塊(如果可能),然後按一下Save。

Status							
Status : Ready							
Server* 10.1.61.158CUCM Voice/Video	Ŧ						
User Information							
User Name* ICVA							
Authentication Information							
Authentication Required							
Password •••••	Reenter Password ••••••	F	Protocol	0	MD5	۲	SHA
Privacy Information							
Privacy Required							
Password •••••	Reenter Password	F	Protocol	0	DES	۲	AES128
Host IP Addresses Information							
Accept SNMP Packets from any host	Accept SNMP Packets only from these host	ts					
	Host IP Address						
	Insert						
	Host IP Addresses						
		-					
	Remove						
Access Privileges							
Access Privileges* ReadOnly	T						
ONotify access privilege is required in order	er to configure Notification Destinations.						

2	Apply	То	All	Nodes

Save Clear All Cancel

步驟3.將預設編解碼器設定為G.711

- 定位至CM Administration > System > Region Information > Region,然後建立一個新區域,例 如ICVA。
- •在「區域」區域中選擇所有區域,並將64kbps(G.722, G.711)配置為最大音訊位元率。
- 選擇Max Video Call Bit Rate中的None單選按鈕,然後按一下Save。

egion Configuration							Relate	d Links: Back To Find/Lis	
🚽 Save 🗙 Delete	省 Reset 🥒 Apply Config 🛛	Add New							
Name* ICVA									
Region Relationships-									
Region	Audio Codec Prefe	rence List	Maximum Ra	Audio Bit te	Maximum Session B Call	it Rate for Video	Maximum S	ession Bit Rate for Immersiv Calls	e Vide
Default	Use System Default (Fac loss)	tory Default low	64 kbps G.7	(G.722, 11)	Non	e		None	
ICVA	Use System Default (Fac	tory Default low	64 kbps G.7	(G.722, 11)	Non	•		None	
Mex	Use System Default (Fac	tory Default low	64 kbps G.7	(G.722, 11)	Non	•		None	
SanJose	Use System Default (Fac loss)	tory Default low	64 kbps G.7	(G.722, 11)	Non	•		None	
NOTE: Regions not displayed	Use System D	efault	Use Syste	m Default	Use System	Default		Use System Default	
odify Relationship to	other Regions								
	Regions	Audio Codec Pre	ference List	Maxims	m Audio Bit Rate	Maximum Sessio for Video C	n Bit Rate Calls	Maximum Session Bit Ra Immersive Video Cal	te for Is
Default ICVA Mex SanJose	* *	Keep Current Se	etting 🔻	* 64 kbps (0	8.722, G.711) ¥	Keep Current Use System (None	Setting Default	 Keep Current Setting Use System Default None 	

附註:組播媒體流始終使用G.711 mu-law編解碼器。不允許或支援其他編解碼器。使用其它 編解碼器到達Informacast的呼叫必須轉碼。

步驟4.建立裝置池

- 導航到CM Administration > System > Device Pool,然後建立裝置池。例如,將其命名為 ICVA_DP。
- 將剛建立的ICVA區域新增到其中。
- 從SRST Reference下拉選單中選擇Disable。
- 從Join Across Lines下拉選單中選擇On,然後按一下Save。

Device Pool Configuration			
🕞 Save 🗶 Delete 📔 Copy	Preset ,	🖉 Apply Config 🚦 Add New	
Device Pool Settings			
Device Pool Name*		ICVA_DP	
Cisco Unified Communications Mar	nager Group*	Default	~
Calling Search Space for Auto-reg	istration	< None >	~
Adjunct CSS		< None >	~
Reverted Call Focus Priority		Default	~
Intercompany Media Services Enro	olled Group	< None >	~
- Poaming Sensitive Settings-			
Date/Time Group*	Chilleseel		
Design *	CMLOCAL		
Neglon	ICVA		
Media Resource Group List	< None >	`	1
Location	< None >	·	1
Network Locale	< None >	· · · · · · · · · · · · · · · · · · ·	·
SRST Reference*	Disable	· · · · · · · · · · · · · · · · · · ·	
Connection Monitor Duration***			
Single Button Barge*	Default	· · · · · · · · · · · · · · · · · · ·	
Join Across Lines*	On		
Physical Location	< None >		
Device Mobility Group	< None >		4
Wireless LAN Profile Group	< None >		View Details

步驟5.建立路由分割槽,例如ICVA_PT。

步驟6.建立呼叫搜尋空間,例如ICVA_CSS。包括ICVA_PT。

步驟7.建立訪問控制組(AXL)。

- 導航到CM Admin > User Management > User Settings> Access Control Group,然後建立訪 問控制組,例如ICVA User Group。
- 向其添加「標準AXL API訪問」角色。

附註:您可能已經有一個名為「標準AXL API訪問」的訪問控制組,並且已經新增了「標準 AXL API訪問」角色,您也可以使用該訪問控制組。

步驟8.建立應用程式使用者

- 導航到CM Admin > User Management > Application User,然後點選Add New。將應用程式使 用者命名為ICVA_InformaCast,並分配以下角色:
- 1. 已啟用標準CTI
- 2. ICVA使用者組(或標準AXL API訪問)
- 3. 標準CTI允許控制支援連線的交換機和會議。

Standard CTI Allow Control of All Devices

Standard CTI Enabled

Standard CTI Allow Control of Phones supporting Conn Standard CTI Allow Control of Phones supporting Rollo

- 4. 標準CTI允許對支援翻滾模式的電話進行控制
- 5. 標準CTI允許控制所有裝置

Application User Configu	ration	
🔚 Save 🗶 Delete [Copy 🕂 Add New	
Application User Informat	ion	
User ID*	ICVAInformacast	Edit Credential
Password	•••••	
Confirm Password	•••••	
Digest Credentials		
Confirm Digest Credentials		
BLF Presence Group*	Standard Presence group ~	
User Rank*	1-Default User Rank v	
- Permissions Information		
Groups ICVA User Group		
Standard CTI Allow Co	Add to Access C	ontrol Group
Standard CTI Allow Co Standard CTI Allow Co	ntrol of Phones supporting C ntrol of Phones supporting R	cess Control Group
Standard CTI Enabled	View Details	

View Details

警告:每個缺陷 CSCve47332 ,建議不要對應用程式使用者ID使用空格。

步驟9.使用SIP或CTI將通訊管理器與Informacast整合。

對於SIP整合,請建立SIP配置檔案、SIP中繼和路由模式。

- 導航到CM Admin> Device > Device Settings > SIP Profile,然後點選Standard SIP Profile,然 後點選Copy
- 將配置檔案命名為ICVA SIP Profile,然後選擇Best Effort(no MTP inserted)。 按一下Save。
- 導航至CM Admin > Device > Trunk,然後點選Add New
- 從中繼型別下拉選單中選擇SIP Trunk。按一下Next,然後輸入您的SIP中繼的名稱。
- 選擇裝置池ICVA_DP,向下滾動到SIP資訊區域,然後在Destination Address中輸入 InformaCast伺服器的IP地址
- 確保Destination Port欄位中的值為5060,選擇**Non Secure SIP Trunk Profile**,然後從SIP Profile下拉選單分配之前建立的SIP配置檔案。按一下**Save**。

🕞 Save 🗙 Delete 省 Reset 🕂	Add New				
Device Information					
Product:		SIP Trunk			
Device Protocol:		SIP			
Trunk Service Type		None(Default)			
Device Name*		ICVA_SipTrunk			
Description		10.1.61.118			
Device Pool*		ICVA_DP		~	
Common Device Configuration		< None >		~	
Call Classification*		Use System Defaul	t	~	
Media Resource Group List		< None >		~	
Location*		Hub None		~	
AAR Group		< None >		~	
Tunneled Protocol*		None		~	
OSIG Variant*		No Changes		~	
ASN.1 ROSE OID Encoding*		No Changes		~	
Packet Capture Mode*		None		~	
Packet Capture Duration					1
		0			
Media Termination Point Required					
SIP Information					
C Destination					
Destination Address is an SRV					
Destination Add	dress	Destination A	ddress IPv6	Destination P	ort
1* 10.1.61.118				5060	
MTP Preferred Originating Codec*	711.02		1		
BLE Presence Group*	/11ulaw	×	1		
SIP Truck Security Profile*	Standard Presence group	*]		
Reputing Calling Search Space	Non Secure SIP Trunk Profile	*]		
Out-Of-Dialog Refer Calling Search Space	< None >	Ť]		
SUBSCRIBE Calling Search Space	< None >	*]		
SIP Profile*	ICVA SIP Profile	· ·	View Details		
DTMF Signaling Method*	No Preference	Ť	1 THEN DOLONS		
	Inversionence		1		

- 建立路由模式,導航到CM Admin > Call Routing > Route Hunt > Route pattern,然後點選Add New。
- 輸入路由模式(例如7777)並配置可從電話訪問的分割槽(例如ICVA_PT)。
- 從Gateway/Route List下拉選單中選擇剛剛建立的SIP中繼。
- 選擇Route This Pattern和OnNet單選按鈕。

• 取消選中Provide Outside Dial Tone覈取方塊,然後按一下Save。 對於CTI整合,請建立CTI路由點並關聯到步驟8中建立的應用程式使用者。

- 導航到CM Administration > Device > CTI Route Point, 然後點選Add new。
- 輸入名稱,例如ICVA_CTI_RP(或您喜歡的任何名稱)。
- 分配裝置池ICVA_DP並點選Save。
- 選擇第1行,輸入目錄編號(例如7778),然後分配最近建立的分割槽(ICVA_PT)。
- •根據需要配置其餘資訊,然後按一下Save。

將CTI路由點新增為ICVA應用使用者配置上的受控裝置。

Controlled Devices	ICVA_CTI_RP	^
		~

附註:如果在通訊管理器中建立多個CTI路由點並將其與InformaCast應用程式使用者相關聯 ,則InformaCast可以支援這些CTI路由點。

提示:您無需為每個需要撥入的號碼建立CTI路由點,還可以向單個CTI路由點新增多條線路 。另一種選擇是使用萬用字元模式匹配一系列數字。

步驟10.為Cisco IP電話啟用Web Access,以便使用HTTP控制電話。

- •Web訪問可以按裝置、按公共裝置配置檔案或在企業電話配置中配置系統範圍。
- 若要應用企業電話配置中的更改,請導航到CM Admin > System > Enterprise Phone Configuration,向下滾動到Web Access下拉選單,然後選擇Enabled。按一下Save。

• 里直電詰以應用史改。			
Enterprise Phone Configuration			
Save			
Web Access*	Enabled	~	

步驟11.設定身份驗證URL。

更改身份驗證URL以將身份驗證請求從IP電話傳送到InformaCast。所有非InformaCast身份驗證請 求都重定向回預設CUCM身份驗證URL。

- 導航到CM管理>系統>企業引數。
- 在「URL Authentication」欄位和「Secure Authentication URL」中輸入http://<InformaCast Virtual Appliance IP Address>:8081/InformaCast/phone/auth。
- 按一下「Save」、「Apply config」和「Reset」電話。

nterprise Parameters Configuration	
🚽 Save 🤌 Setto Default 🌯 Reset 🥒 Apply Config	
Phone URL Parameters	
URL Authentication	http://10.1.01.118.8081/3nformaCast/phone/auth
URL Directories	http://10.1.61.158:8080/comcip/weikdrectory.jsp
utt. title	
JRL John Time	0
381. Information	http://10.1.61.158.0000/comcip/GetTelecasterHeipText.js
JRL Messages	
P. Phone Proxx Address	
URL Services	http://10.1.61.158-8080/comcip/petservicesmenu.jsp
Secure Phone URI. Parameters	
Secure Authentication URL	http://10.1.61.118:8081/3nformaCast/phone/auth
lecure Directory URL (104L)	https://10.1.61.158.8443/comdp/unideedtory.jsp
ecure Contact Search URL (UDS)	https://10.1.41.158:8443/cuom-uds/users
ieare Ide UR.	
Secure Information URL	https://10.1.61.158.8443/comdp/GetTelecasterHelpText.)
Secure Messages URL	
Secure Services URL	https://10.1.61.158:8443/com/p/pitserv/cesmenu.jsp

附註:URL區分大小寫,因此請確保InformaCast一詞中的I和C大寫。安全身份驗證URL和身 份驗證URL必須設定為相同的值,即HTTP URL。

步驟12.設定API瀏覽器訪問的身份驗證方法。

• 如果您使用的是Unified Communications Manager 11.5.1及更高版本,請向下滾動頁面到「安全引數」區域,然後從API瀏覽器訪問的身份驗證方法下拉選單中選擇Basic。

步驟13.測試您的電話,例如,撥打7777(SIP整合)或7778(CTI整合)。

附註:如果在混合模式下運行Unified Communications Manager,請確保對InformaCast的呼 叫不使用加密介質。

配置Informacast

步驟1.在Informacast中配置Communications Manager群集。

- 登入到Informacast並導航到Admin > Telephony > Unified Communications Manager Cluster。 按一下Edit。
- 輸入您在步驟8中建立的應用程式使用者的使用者名稱和密碼。
- 確保選中Use Application User for AXL覈取方塊,這意味著在構建InformaCast的電話快取時使 用您的應用程式使用者憑據。

附註:如果將此欄位留空,InformaCast將嘗試在運行CallManager服務的伺服器中查詢運行 AXL服務的伺服器。

- 在「**Communications Manager IP Address(es)」欄位中輸入**Unified Communications Manager伺服器的IP地址。使用數字IP地址而不是DNS名稱。
- 選擇SNMP v2或SNMP v3單選按鈕。輸入在CUCM中配置的相同資訊。按一下Update。

			L	to Out And	ication Adm	instato
Admin Telephony Cisco Unified Con Configuration	mmunications	Manager (Cluster E	Edit Tele	phony	
·	elephony Config	uration				
Unified Communications Manager Cluster Description	CUCM	(require	d)			
Unified Communications Manager Application User	CVPInformacast	(require	d)			
Unified Communications Manager Application Password						
Confirm Application Password						
	Use Application	User for A30,				
Alt, IP. Address(es)	10.1.61.158					
Unified Communications Manager IP. Address(es)	10.1.61.158	(require	4			
Chorase SMMP version	SHMP v2 (requ	ired)				
SMMP v2 Community Name						
Confirm SMMP v2 Community Name						
XML Push Authentication If you are not using JTAPI to activate phones during broad parameter for the Unified Communications Manager in the Parameters page) is set to the following value.	lcasts or if this is no is cluster (found in th	t your primary clu e Phone URL Pr	ster, make su trameters se	re the URL ction of the	Authentica System Er	tion terprise
http://10.1.6	1.118:8081/1686	ensCast/phone	/wuth			
Optionally, you can also tell InformaCast where to send a need to do this if, before installing InformaCast, you had such cases, copy the current Unified Communications M	uthentication reques set this Unified Com lanager setting into t	ts for commands munications Mana he field below, bef	that aren't co ger paramete bre changing i	ming from 8 r to a non s t to the vak	normaCast. 1 tandard value se shown abo	You only a. In ove.
Next Authentication URL:						
If empty, non-informaCast authentication requests from p authentication page, http://l0.1.41.153/comctp/	hones in this cluster authenticate.p	will be sent to the	default Unifie	d Commun	ications Mar	ager

步驟2.配置收件人組。

• 導航到Recipients > Edit recipient Groups,然後點選Update以顯示在CUCM中註冊並由 InformaCast發現的所有電話。

Inform	aCast [®] asic paging	All N		ution 2007 Levers	1	Recorder	8	CO Ipenalters	800 B	*	O Flagen	() 140
8	Recipients E	Edit Recip over curren uming), v Defunct Pr	ient Gro t IP phone	NUDS Reci	ipient gro	oup memb Cisco Un	ified Com	led municatio	ns Manaj	per (may l	be time	
									4	44 O		
	A Name								Phones	Action		
	(All Recipients)								2	>	ann O (a	am 🕑

• 若要建立新的收件人組,請按一下Add,寫入名稱,然後按一下Edit為此收件人組新增電話。將 電話新增到收件人後,按一下Submit。

Select	Individual Recipients			
Filter Avail	able Recipients (double click to select)		Sale	thed Recipients (double click to remove)
	Descriptive Text			Descriptive Text
•	Cisco IP Phone: Auto 111; DNs: 111; SEPF87B204EED99	Add Barnova	•	Cisco IP Phone: Auto 110; DNs: 110; SEP2C3124C9F8E1
<<1	st < prov 1 next > last >>			Submit Canta

• 要儲存更改,請按一下Update。

Inform	aCast ^e asic paging	Advanced () Lay	olification 20	Anna R	Recorders	Bacqueents	CO Speakers	2	2 	Propies	? •**
								to	g Our Appl	ication Adv	iristato
	Recipients Ed	lit Recipient	Groups I	Edit Rec	ipient C	Group					
	Name Mex			(require	ed)						
	Tege MEx			Add A	Tag 🖂						
	Select Recipients										
	🖂 industrially (🔤 🔽										
	Cisco IP Phone: Auto 110; DNs: 110; SEP2C312AOF8E1										
	Ether with Recipiers	.Groves 💷 🔽									
	Filter with Rules										
	Exclusions are only	available when the	Recipient Grox	p is Fitered	by Recipie	ent Groups	or Rules.				
				Castan (D (111						

步驟3.允許/禁用對InformaCast的SIP訪問。

導航到Admin > SIP > SIP Access。預設情況下,所有SIP呼叫均被拒絕。
選擇Allow單選按鈕允許所有SIP呼叫,或按一下Add允許此允許的例外。

Inform	aCast			cettos 200 Learn	formal sector	Dessage	8 Nacional	Speature	No.	en e	Pages	? Ndp
4	Admin SIP S	SIP Aco	988							ng 0-4 App	6(2)(n A3	nan stratig
	Controls access of inbound SIP calls to InformaCast. Click to restore to default settings (more C)											
	a host exception											

提示:定義例外時,請確保指定直接向InformaCast傳送INVITE請求的主機。如果代理位於 InformaCast和呼叫主機之間,則此伺服器可能是SIP代理伺服器。

步驟4.新增廣播撥號配置

• 轉至Admin > DialCast > Dialing Configurations,點選Add

- 根據在CUCM中建立的路由模式(用於SIP整合)或CTI路由點(用於CTI整合),在Dialing Pattern欄位中輸入撥號模式(例如7777、7778)。
- 從清單中選擇收件人組,然後按一下Update。

Inform	aCast ^e asic paging		٩	ation 20 Learn		Recorder.	83 Recipients	C) Spendares	80 H	e constante da la constante da	Property	() 140
a S	Admin DialCas	t Dial	ling Con	figurati Daling	ons E Pattern	dit Broa	dcast D	ialing C	Configu	ration	elanan win	
				Broadca Recipient	Groups	(Al Recip Men SarJose	ients) ~					
				CARGE				-0				

步驟5.配置廣播引數。

- 導航到Admin > Broadcast Parameters
- •配置組播的IP。通常使用預設IP(239.0.1.2)。
- 如果要將命令以JTAPI形式傳送到電話,請啟用JTAPI覈取方塊,否則將使用HTTP消息。

Inform	aCast [®]	0	()	adice 200			8	۲	2	2 3	O	2
										og Out App	Ication Ad	ninistrator
а <mark>с</mark> з	Admin Broad	Icast Par s to Phones I phony Termin	tameter by JTAPI: E als for all Phones E	s 3 3								
	Enders	Multicast IP	Address B	29.0.1.2		(mage)	(but					
	201124	Muto	cast.TTL; []	lee <u>shttp://w</u>	iniana. Ared)	orgʻassignme	nts/mattic	ant address	<u>892</u> .			
				Cancia D			840	uu 🔕				

確保此範圍對應於您的網路基礎設施設定並涵蓋所有收件人組。在多站點部署中,Singlewire和思 科建議使用一系列地址。此範圍應足夠大,以便為每個同時廣播處理一個地址。

附註:建議通過HTTP使用JTAPI,因為它可以更好地監視電話的狀態並適用於更多區域設定 。

提示:網路介面的預設設定將在五分鐘後註銷。**導航到Admin > Network Parameters >** Session Timeouts, 然後將General Session Timeout(seconds)欄位從300更改為新值。

配置網路中的組播

如果思科呼叫伺服器和IP電話位於不同的IP子網中,則這兩個子網之間的路由器必須配置為組播路 由。

思科呼叫伺服器不需要任何特定的組播路由方法(SM、DM、S-DM、SSM等)。 某些廣域網環境 不支援組播路由。對於這些環境,可以在站點之間構建GRE隧道並用於傳輸組播。 在您的環境中設計和配置組播不屬於本文檔的範圍,但您可能會發現以下資源很有用:

• <u>組播白皮書</u>

•<u>多點傳送測試工具</u>

附註:如果您使用的是Meraki交換機,則預設情況下它們啟用了IGMP監聽。這可能導致問題 ,需要由Meraki禁用。一旦您聯絡他們並讓他們禁用IGMP監聽,請再次測試尋呼。

驗證

目前沒有適用於此組態的驗證程序。

疑難排解

常見問題

電話未啟用

考慮到Informacast會在廣播發生時跳過任何正在使用(忙)的電話。

InformaCast根據您向電話傳送消息的方式(HTTP或JTAPI)使用不同的忙碌檢測方法。

HTTP:繁忙檢測僅適用於運行英文載入的電話區域設定

CTI:適用於非英語電話區域設定

忙碌檢測根據協定、線路型別和線路狀態也可以不同地工作。

線路狀態	CTI忙檢測	HTTP忙檢測
共用線路,另一部電話上的呼叫正在使用 ,無保留的呼叫	空閒	空閒
摘機,收集數字	作亡	不忙
通話、活動呼叫	作亡	忙
保留,共用線路上的非活動呼叫	作亡	不忙
保留,唯一線路上的非活動呼叫	作亡	不忙

附註:如果嘗試同時廣播,Informacast首先播放第一個廣播(第二個廣播被顛覆)。

對未啟用的電話進行故障排除時,應收集以下資料:

- 來自Informacast的效能日誌。
- 電話的控制檯日誌(PRT)。

未發現電話

InformaCast只發現已註冊的電話。如果IP電話已註冊但未發現,請檢查Informacast中的SNMP服務 配置以及電話註冊到的CUCM節點。應為啟用呼叫管理器服務的所有節點配置SNMP服務和社群字 串。

SNMP錯誤無法建立收件人組: java.lang.Exception

8	X
2	\sim
<u>م</u>	

Recipients Edit Recipient Groups	
Recipient group members updated	
Unable to build incipient groups: java.lang.Exception: Unable to obtain phone information via 1 cluster. Make sure the SNNP service is started on all servers in the cluster and that the provide [Show details]	AMP from all servers in the 'CUCM' d SNMP community name is correct.
(PMLS) Discover current IP phone information from Cisco Unified Communications Manag	per (may be time consuming).
(INVIALO) Show Defunct Phones	
All the Store State Store Stor	(ALD C)
A Name	Phones Action
(All Radgierts)	3 (BE 🔽 (BEE) (BEE)
Mex	1 00 000 000
SanJose	1 (00 🔽 (000) (000)
TELEVISION Property of 1 (1997) Annual Strategy (1997) (1997) West With a strategy property	

- 1. 此錯誤意味著,由於DNS連線或解析,SNMP無法及時響應查詢。
- 2. 確認沒有阻止UDP埠161從InformaCast伺服器到所有Unified Communications Manager群集 節點。
- 3. 確認SNMP資訊是否正確。如果可能,導航到Admin > Telephony > Unified Communications Manager Cluster,然後鍵入新的SNMP字串。在CUCM中配置新字串。
- 4. 您可能還正在使用超出社群字串最大字元數的社群字串。如果要從CUCM複製社群字串並將其 貼上到Informacast配置中,請嘗試將其鍵入以檢視是否可以鍵入整個字串。在Informacast版 本11中,最大字元數為18。
- 5. 檢查CUCM上的DNS配置是否正確,並確認您與缺陷CSCtb70375不匹配。

目標電話無音訊

如果電話亮起但不播放音訊,則問題很可能與組播路由有關,而與CUCM伺服器或IP電話無關。

要收集的資料

在對Informacast進行故障排除時,您應收集以下資料:

- 1. 來自Informacast的效能日誌。
- 2. 從Informacast捕獲資料包。
- 3. 從電話捕獲資料包。
- 4. 從CUCM捕獲資料包。
- 5. 來自CUCM的SDL日誌
- 6. PRT(控制檯日誌)

效能日誌

有兩種方法可以從Informacast獲取效能日誌。

方法1

- 1. 導航至https://<Informacast IP>:8444/InformaCast/logs/performance.log
- 2. 複製並將日誌儲存到.txt檔案中。

方法2

- 1. 在Web瀏覽器中開啟Informacast IP(https://<informacast_IP>),然後選擇Informacast。
- 2. 使用您的憑據登入並導航到**幫助>支援**。



You currently have InformaCast Basic Paging installed for use. Click the Try and Buy

3.按一下「Tools(工具)」部分下的「Performance Logs」,如下圖所示。

Tools

These links help carry out steps mentioned in the documentation, or suggested by technical support.

API Log Shows requests made to the InformaCast REST API.

Calling Terminal Diagnostics Shows the CTI ports and route points registered with InformaCast.

Call Detail Records Directory Shows the directory containing the call detail records.

InformaCast Logs Directory Shows the directory containing the InformaCast logs.

Log Tool Collects and analyzes Singlewire log files for errors.

Performance Log Ontains information logged by InformaCast.

SIP Stack Log Contains information logged by the SIP stack.

Summary Log Contains a summary of broadcasts sent by InformaCast.

封包擷取

從Informacast

有三種方法可以從Informacast獲取資料包捕獲。

方法1

- 1. 通過SSH連線到Informacast框的CLI
- 2. 執行命令sudo capturePackets test.cap開始捕獲並建立名為test.cap的檔案
- 3. 呼叫不工作的電話
- 4. 按下Ctrl + C以結束pcap
- 5. 執行ls以確保資料包捕獲在機箱中
- 6. 使用SFTP或安全複製(SCP)將檔案傳輸到您的PC

admin@singlewire:~\$ sudo capturePackets test.cap tcpdump: listening on eth0, link-type EN10MB (Ethernet), capture size 1514 bytes ^C34 packets captured 36 packets received by filter 0 packets dropped by kernel admin@singlewire:~\$ sftp cisco@10.1.61.20 Authenticated with partial success. cisco@10.1.61.20's password: Hello, I'm freeFTPd 1.0Connected to 10.1.61.20. sftp> sftp> sftp> put test.cap Uploading test.cap to /test.cap test.cap sftp>

方法2

- 1. 從Web下載並安裝InformaCast LogTool。
- 執行軟體並選擇選項[5]。如圖所示,寫入Informacast的IP、登入憑據以及資料包捕獲應運行 的秒數。

Administrator: Singlewire Software: InformaCast Log Tool - 20150707	
Singlewire Software	4
[1] Gather Logs From InformaCast Server [2] Uncompress Gathered Logs [3] Parse Logs For Errors and Solutions [4] Turn JTAPI Debugging On/Off [5] Network Traffic Capture [9] Exit	
Menu Choice: 5	
Server IP: 172.16.3.221 Username: admin Password:	
Attempting to Access 172.16.3.221 via SSH	
Seconds Capture Should Run [1-300]: _	_

 捕獲不會立即啟動,因此您可以準備測試環境。 就緒後,選擇選項[1]並按Enter開始捕獲資料 包,如下圖所示。

```
Seconds Capture Should Run [1-300]: 60
[1] Start 60 Second Capture
[9] Return to Main Menu
Menu Choice.....: 1
Iraffic Capture Started: 60 Seconds
00:00:51
```

- 4. 該工具將顯示一個倒計時計時器,其中包含捕獲的未完成持續時間。在此時間以及捕獲倒數達 到零時複製問題,捕獲完成並停止。
- 5. 該工具將資料包捕獲和所有日誌繫結到.tgz檔案,並將其傳輸到您的工作站。 這與選項1收集 日誌相同,但也包括網路流量捕獲。
- 6. 該工具將在Informacast_LogTool.exe的基目錄下建立一個包含資料包捕獲的資料夾,如下圖所示。

InformaCast_LogTool.exe
InformaCast_LogTool_Logs_201809231605.tgz

方法3(12.0.1及更高版本中提供)

- 1. 登入<Informacast_IP>:10000
- 2. 導覽至System > Capture Network Traffic。



3. 按一下「Start a new packet capture」,然後複製問題,如下圖所示。



- 4. 當問題完全複製時按一下Stop Packet Capture,或在捕獲33,000個資料包後自行停止。
- 5. 導航到System > Collect Logs, 輸入問題的簡短說明, 然後按一下Collect a new set logs。
- 6. 若要儲存日誌,請按一下Download to Your Computer,如下圖所示。

Legix admin	Module Config Collect Logs
System Bootop and Shutdows Capture Network Traffic Change Prasowerds Caffect Lags Disk and National Fibersystems Running Processes System Lags Upgrade or Switch Versions Networking Networking Network Search	Collect a New Set of Logs for a Problem Report This process will produce a package of logs for ste by technical support Collect New Log Set Problem description to include in report Singlewire support contract number, if known Do not automatically and the log collection to Singlewire Support Collect a new set of logs
🚖 System Information 📵 Lagout	The log collection from 2016-04-08 21:10:52 Send to Singlevine Support must be downloaded and sent to Singlevine Support. Download to Your Computer

方法4(12.0.1及更新版本中提供)

在12.0.1及更高版本中,不再需要使用sudo命令。若要執行封包擷取,請使用命令**capture-packets**<name of the file> <number of packets>,如範例所示:

admin@informacast:~\$ capture-packets test Saving up to 33000 packets to /var/log/capture-packets/test tcpdump: listening on eth0, link-type EN10MB (Ethernet), capture size 1514 bytes ^C13 packets captured 15 packets received by filter 0 packets dropped by kernel **附註**:GUI方法比CLI更好,因為不需要依賴SFTP伺服器,而且您可以從網頁啟動、停止和下 載資料包捕獲。

來自CUCM

根據部署定義需要從何處獲取資料包捕獲。集群中只能有一個CUCM節點或多個CUCM。

•如果您有一個CUCM節點,請獲取資料包捕獲,如下圖所示。





- 1. 為需要捕獲的節點開啟SSH會話
- 2. 運行命令utils network capture eth0 size all count 1000000 file Test以開始資料包捕獲。
- 3. 複製問題
- 4. 使用Ctrl + C停止資料包捕獲
- 5. 要確認資料包捕獲已儲存,請運行命令file list activelog platform/cli/*

₽	10.1.61.158 - PuTT	Y	 • •	×
admin: admin:utils network capture Executing command with optic	eth0 size all count	1000000 file Test		^
size=ALL com src= des ip=	unt=100000 st=	interface=eth0 port=		
Control-C pressed				
admin:file list activelog pl	latform/cli/*			
Test.cap				
admin:	1			~

6. 使用命令file get activelog platform/cli/Test.cap將資料包捕獲傳送到SFTP伺服器。或者,要收 集伺服器上儲存的所有.cap檔案,請使用file get activelog

```
platform/cli/*.cap
admin:file get activelog platform/cli/*.cap
Please wait while the system is gathering files info ...done.
Sub-directories were not traversed.
Number of files affected: 7
Total size in Bytes: 658062
Total size in Kbytes: 642.6387
Would you like to proceed [y/n]? y
SFTP server IP: 14.48.27.201
SFTP server port [22]:
User ID: administrator
Password: *******
Download directory: /
......
Transfer completed.
admin:
```

7. 如果無法使用SFTP伺服器,請使用RTMT。導覽至**System > Trace & Log Central > Collect** Files。按一下「Next」,然後啟用「Packet capture logs」覈取方塊,如下圖所示。

Select System Services/Applications									
Select all Services on all Servers									
Name	All Servers	ccm8pub							
Cisco WebDialerRedirector Web Service									
Cron Logs									
Event Viewer-Application Log									
Event Viewer-System Log									
Host Resources Agent									
PT Platform CLI Created Reports									
PT Platform CLI Logs									
PT Platform Cert Monitor Logs									
PT Platform CertMgr Logs									
PT Platform Cluster Manager Logs									
PT Platform GUI Logs									
PT Platform IPSecMgmt Logs									
PT Platform RemoteSupport Logs									
nstall File Signing									
nstall and Upgrade Logs									
IIB2 Agent									
fail Logs									
/getty Logs									
ITP Logs									
letdump Logs									
Packet Capture Logs									
Prog Logs									
SAR Logs									
SNMP Master Agent									
Security Logs									
Service Manager									
Spooler Logs									
System Application Agent									

8. 按一下**下一步**,選擇一個下載檔案目錄,然後按一下**完成**。

9. 使用file delete activelog platform/cli/Test.cap命令刪除資料包

通過電話

- 1. 啟用SPAN到PC埠。導航到CM Admin page > Device > phone,然後查詢報告有問題的電話。
- 2. 在「產品特定配置佈局」部分下,找到Span to PC Port,然後從下拉選單中選擇Enable。按 一下Save,然後按一下Apply config。
- 3. 將筆記型電腦連線到電話的PC埠。
- 4. 在筆記型電腦中運行資料包分析器軟體。您可以使用Wireshark(或其他封包擷取軟體)。
- 5. 重現問題。
- 6. 完全複製問題時,繼續停止資料包捕獲。
- 您可以在以下連結中找到更多詳細資訊
- : <u>https://supportforums.cisco.com/document/44741/collecting-packet-capture-cisco-ip-phone</u>

範例分析



SDL跟蹤

用於SIP整合和由JTAPI控制的電話

CUCM:10.1.61.158

Informacast:10.1.61.118

電話A

DN:110

型號:CP-8861

韌體版本:sip88xx.12-0-1SR1-1

電話A IP地址:10.1.61.12

MAC SEP2C3124C9F8E1

電話B

DN:111

型號:CP-8811

韌體版本:sip88xx.12-0-1SR1-1

電話B IP地址: 10.1.61.11

MAC SEPF87B204EED99

撥號號碼:7777

CUCM receives the invite from Phone A

71439050.002 |19:00:35.206 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.1.61.12 on port 51600 index 25770 with 1791 bytes: [431528,NET] INVITE sip:7@10.1.61.158;user=phone SIP/2.0 Via: SIP/2.0/TCP 10.1.61.12:51600;branch=z9hG4bK18a14280

```
From: "PhoneA" <sip:110@10.1.61.158>;tag=2c3124c9f8e10c3c246b7956-5c62fa57
To: <sip:7@10.1.61.158>
Call-ID: 2c3124c9-f8e1000d-00337209-0547bb10@10.1.61.12
Max-Forwards: 70
Date: Tue, 10 Sep 2019 00:00:37 GMT
CSeq: 101 INVITE
User-Agent: Cisco-CP8861/12.0.1
Contact: <sip:142b9f25-7f2b-48a8-9ff9-
377f616f3084@10.1.61.12:51600;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP2C3124C9F8E1"
Expires: 180
Accept: application/sdp
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE, SUBSCRIBE, INFO
Remote-Party-ID: "PhoneA" <sip:110@10.1.61.158>;party=calling;id-
type=subscriber;privacy=off;screen=yes
Supported: replaces, join, sdp-anat, norefersub, resource-priority, extended-refer, X-cisco-
callinfo,X-cisco-serviceuri,X-cisco-escapecodes,X-cisco-service-control,X-cisco-srtp-fallback,X-
cisco-monrec,X-cisco-config,X-cisco-sis-7.0.0,X-cisco-xsi-8.5.1
Allow-Events: kpml, dialog
Recv-Info: conference
Recv-Info: x-cisco-conference
Content-Length: 548
Content-Type: application/sdp
Content-Disposition: session; handling=optional
v=0
o=Cisco-SIPUA 11811 0 IN IP4 10.1.61.12
s=STP Call
b=AS:4064
t = 0 0
m=audio 22018 RTP/AVP 114 9 124 0 8 116 18 101
c=IN IP4 10.1.61.12
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114 maxplaybackrate=16000;sprop-
maxcapturerate=16000;maxaveragebitrate=64000;stereo=0;sprop-stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
### CUCM performs digit analysis for the dialed digits (dd="7777")
71439203.000 |19:00:36.580 |Sdlsig
                                    DaReq
                                                                            wait
Da(1,100,216,1)
                                  |Cdcc(1,100,224,6)|
1,100,14,1368.16^10.1.61.12^*
                                         [R:N-H:0,N:0,L:0,V:0,Z:0,D:0] CI=19282342
Fqdn=ti=1nd=110pi=0si1 Cgpn=tn=0npi=0ti=1nd=110pi=1si1
DialedNum=tn=0npi=1ti=1nd=7777User=7777Host=10.1.61.158Port=5060PassWord=Madder=Transport=4mDisp
layName=RawUrl=sip:7@10.1.61.158;user=phoneOrigPort=0pi=0sil requestID=0
DigitAnalysisComplexity=1 CallingUser= IgnoreIntercept=0 callingDeviceName=SEP2C3124C9F8E1
71439203.001 |19:00:36.580 |AppInfo |Digit Analysis: star_DaReq:
daReq.partitionSearchSpace(8653f609-05a7-5914-819b-3a89680af6a2:),
filteredPartitionSearchSpaceString(Informacast_PT:phone_pt),
partitionSearchSpaceString(Informacast_PT:phone_pt)
71439203.002 |19:00:36.580 |AppInfo |Digit Analysis: Host Address=10.1.61.158 MATCHES this
node's IPv4 address.
71439203.003 |19:00:36.580 |AppInfo |Digit Analysis: star_DaReq: Matching SIP URL, Numeric
User, user=7777
```

71439203.012 |19:00:36.588 |AppInfo |Digit analysis: match(pi="2", fqcn="110", cn="110",plv="5", pss="Informacast_PT:phone_pt", TodFilteredPss="Informacast_PT:phone_pt", dd="7777",dac="1") 71439203.013 |19:00:36.588 |AppInfo |Digit analysis: analysis results 71439203.014 |19:00:36.588 |AppInfo ||PretransformCallingPartyNumber=110 CallingPartyNumber=110 |DialingPartition=Informacast_PT |DialingPattern=7777 |FullyQualifiedCalledPartyNumber=7777 DialingPatternRegularExpression=(7777) |DialingWhere= |PatternType=Enterprise PotentialMatches=NoPotentialMatchesExist |DialingSdlProcessId=(0,0,0) |PretransformDigitString=7777 |PretransformTagsList=SUBSCRIBER |PretransformPositionalMatchList=7777 CollectedDigits=7777 UnconsumedDigits= |TagsList=SUBSCRIBER PositionalMatchList=7777 VoiceMailbox= VoiceMailCallingSearchSpace= VoiceMailPilotNumber= |RouteBlockFlag=RouteThisPattern RouteBlockCause=0 |AlertingName= UnicodeDisplayName= [CallableEndPointName=[ddef6b78-6232-f5eb-b286-79292be99bb5]

CUCM determines call must stay on the same node, then it sends the call to SIP Trunk PID=SIPD(1,100,84,12)

71439207.001 |19:00:36.588 |AppInfo |Digit analysis: wait_DmPidRes- Partition=[107a02ea-a384-5219-3670-ba9d14b9d094] Pattern=[7777] Where=[],cmDeviceType=[Unknown], OutsideDialtone =[0], DeviceOverride=[0], PID=SIPD(1,100,84,12),CI=[19282342],Sender=Cdcc(1,100,224,6)

CUCM extends the call to the Informacast SIP Trunk

71439248.001 |19:00:36.643 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.1.61.118 on port 5060 index 25758 [431545,NET] INVITE sip:7777@10.1.61.118:5060 SIP/2.0 Via: SIP/2.0/TCP 10.1.61.158:5060;branch=z9hG4bK1996d1e0c5e3e From: "PhoneA" <sip:110@10.1.61.158>;tag=229417~7cc9781e-f7e3-4c51-a2b9-de353a4e7d6f-19282343 To: <sip:7777@10.1.61.118> Date: Tue, 10 Sep 2019 00:00:36 GMT Call-ID: 2d72f80-d761e7a4-1996c-9e3d010a@10.1.61.158 Supported: timer, resource-priority, replaces Min-SE: 1800 User-Agent: Cisco-CUCM11.5 Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY CSeq: 101 INVITE Expires: 180 Allow-Events: presence, kpml Supported: X-cisco-srtp-fallback,X-cisco-original-called Call-Info: ;method="NOTIFY;Event=telephone-event;Duration=500" Call-Info: ;x-cisco-video-traffic-class=DESKTOP Cisco-Guid: 0047656832-0000065536-000000001-2654798090 Session-Expires: 1800 P-Asserted-Identity: "PhoneA" <sip:110@10.1.61.158> Remote-Party-ID: "PhoneA" <sip:110@10.1.61.158>;party=calling;screen=yes;privacy=off Contact: <sip:110@10.1.61.158:5060;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP2C3124C9F8E1" Max-Forwards: 69

Content-Type: application/sdp Content-Length: 552 v=0o=CiscoSystemsCCM-SIP 229417 1 IN IP4 10.1.61.158 s=STP Call c=IN IP4 10.1.61.12 b=TIAS:64000 b=AS:64 t = 0 0m=audio 22018 RTP/AVP 114 9 124 0 8 116 18 101 b=TIAS:64000 a=rtpmap:114 opus/48000/2 a=fmtp:114 maxplaybackrate=16000;spropmaxcapturerate=16000;maxaveragebitrate=64000;stereo=0;sprop-stereo=0;usedtx=0 a=rtpmap:9 G722/8000 a=rtpmap:124 iSAC/16000 a=rtpmap:0 PCMU/8000 a=rtpmap:8 PCMA/8000 a=rtpmap:116 iLBC/8000 a=maxptime:20 a=fmtp:116 mode=20 a=rtpmap:18 G729/8000 a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-15 #### Informacast replies with 200 OK (Call established using codec PCMU) 71439316.004 |19:00:36.849 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.1.61.118 on port 5060 index 25758 with 889 bytes: [431549,NET] SIP/2.0 200 OK CSeq: 101 INVITE Call-ID: 2d72f80-d761e7a4-1996c-9e3d010a@10.1.61.158 From: "PhoneA" <sip:110@10.1.61.158>;tag=229417~7cc9781e-f7e3-4c51-a2b9-de353a4e7d6f-19282343 To: <sip:7777@10.1.61.118>;tag=2c9be8b4 Via: SIP/2.0/TCP 10.1.61.158:5060;branch=z9hG4bK1996d1e0c5e3e;rport=43802 Content-Type: application/sdp Contact: "InformaCast" <sip:7777@10.1.61.118;transport=tcp> Allow: ACK, BYE, CANCEL, INFO, INVITE, OPTIONS, NOTIFY Accept: application/sdp Accept-Encoding: identity Accept-Language: en Supported: Call-Info: <sip:7777@10.1.61.118:5060>;method="NOTIFY;Event=telephone-event;Duration=500" Content-Length: 248 v=0o=SinglewireInformaCast-SIP 1568074182370 1 IN IP4 10.1.61.118 s=STP Call c=IN IP4 10.1.61.118 b=TIAS:64000 b=AS:64 t=0 0 m=audio 32070 RTP/AVP 0 101 a=rtpmap:0 PCMU/8000 a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-15 a=ptime:20 #### ACK from CUCM to Informacast 71439319.001 |19:00:36.850 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.1.61.118 on port 5060 index 25758

[431550,NET]

ACK sip:7777@10.1.61.118;transport=tcp SIP/2.0

Via: SIP/2.0/TCP 10.1.61.158:5060;branch=z9hG4bK1996e72237022
From: "PhoneA" <sip:110@10.1.61.158>;tag=229417~7cc9781e-f7e3-4c51-a2b9-de353a4e7d6f-19282343
To: <sip:7777@10.1.61.118>;tag=2c9be8b4
Date: Tue, 10 Sep 2019 00:00:36 GMT
Call-ID: 2d72f80-d761e7a4-1996c-9e3d010a@10.1.61.158
User-Agent: Cisco-CUCM11.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Length: 0

CUCM sends 200 OK to Phone A with codec PCMU

71439437.001 |19:00:36.884 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.1.61.12 on port 51600 index 25770 [431551,NET] SIP/2.0 200 OK Via: SIP/2.0/TCP 10.1.61.12:51600;branch=z9hG4bK18a14280 From: "PhoneA" <sip:110@10.1.61.158>;tag=2c3124c9f8e10c3c246b7956-5c62fa57 To: <sip:7@10.1.61.158>;tag=229414~7cc9781e-f7e3-4c51-a2b9-de353a4e7d6f-19282342 Date: Tue, 10 Sep 2019 00:00:35 GMT Call-ID: 2c3124c9-f8e1000d-00337209-0547bb10@10.1.61.12 CSeq: 101 INVITE Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY Allow-Events: presence Supported: replaces Server: Cisco-CUCM11.5 Call-Info: ; security= NotAuthenticated; orientation= to; qci= 1-15008; isVoip; call-instance= 1 Send-Info: conference, x-cisco-conference Remote-Party-ID: <sip:7777@10.1.61.158>;party=called;screen=no;privacy=off Session-ID: ddef6b786232f5ebb2867929ab229417;remote=712c9e1f00105000a0002c3124c9f8e1 Remote-Party-ID: <sip:7777@10.1.61.158;user=phone>;party=x-cisco-original-called;privacy=off Contact: <sip:7@10.1.61.158:5060;transport=tcp> Content-Type: application/sdp Content-Length: 235

v=0

o=CiscoSystemsCCM-SIP 229414 1 IN IP4 10.1.61.158
s=SIP Call
c=IN IP4 10.1.61.118
b=AS:64
t=0 0
m=audio 32070 RTP/AVP 0 101
b=TIAS:64000
a=ptime:20
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

ACK from Phone A to CUCM

71439438.002 |19:00:36.950 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.1.61.12 on port 51600 index 25770 with 692 bytes: [431552,NET] ACK sip:7@10.1.61.158:5060;transport=tcp SIP/2.0 Via: SIP/2.0/TCP 10.1.61.12:51600;branch=z9hG4bK20553712 From: "PhoneA" <sip:110@10.1.61.158>;tag=2c3124c9f8e10c3c246b7956-5c62fa57 To: <sip:7@10.1.61.158>;tag=229414~7cc9781e-f7e3-4c51-a2b9-de353a4e7d6f-19282342 Call-ID: 2c3124c9-f8e1000d-00337209-0547bb10@10.1.61.12 Max-Forwards: 70 Session-ID: 712c9e1f00105000a0002c3124c9f8e1;remote=ddef6b786232f5ebb2867929ab229417 Date: Tue, 10 Sep 2019 00:00:39 GMT CSeq: 101 ACK User-Agent: Cisco-CP8861/12.0.1 Remote-Party-ID: "PhoneA" <sip:110@10.1.61.158>;party=calling;idtype=subscriber;privacy=off;screen=yes Content-Length: 0 Recv-Info: conference Recv-Info: x-cisco-conference

Since integration is with JTAPI, CUCM sends REFER to the phone with instructions to join to the IP and port of multicast 71439541.002 |19:00:38.199 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.1.61.11 on port 51784 index 25768 [431557.NET] REFER sip:e2881942-2853-4eab-a0d9-96228c79d062@10.1.61.11:51784;transport=tcp SIP/2.0 Via: SIP/2.0/TCP 10.1.61.158:5060;branch=z9hG4bK19970687ccd2b From: <sip:111@10.1.61.158>;tag=1598606730 To: <sip:111@10.1.61.11> Call-ID: 4085c80-d761e7a6-1996d-9e3d010a@10.1.61.158 CSeq: 101 REFER Max-Forwards: 70 Contact: <sip:111@10.1.61.158:5060;transport=tcp> User-Agent: Cisco-CUCM11.5 Expires: 30 Refer-To: cid:1234567890@10.1.61.158 Content-Id: <1234567890@10.1.61.158> Content-Type: multipart/mixed;boundary=uniqueBoundary Mime-Version: 1.0 Referred-By: <sip:111@10.1.61.158> Content-Length: 682 --uniqueBoundary Content-Type:application/x-cisco-remotecc-request+xml <x-cisco-remotecc-request> <datapassthroughreq> <applicationid>0</applicationid> <lineid>0</lineid> <transactionid>109</transactionid> <stationsequence>StationSequenceLast</stationsequence> <displaypriority>2</displaypriority> <appinstance>0</appintance> <routingid>0</routingid> <confid>0</confid> <featuredata></featuredata> </datapassthroughreq> </x-cisco-remotecc-request> --uniqueBoundary Content-Type:application/x-cisco-remote-cm+xml <CiscoIPPhoneExecute><ExecuteItem URL="RTPMRx:239.0.1.2:20480"/></CiscoIPPhoneExecute> --uniqueBoundary--

Phone B replies with 202 Accepted

```
#### Phone B sends a NOTIFY to indicate that it was activated (Data="Success")
71439548.004 |19:00:38.453 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from
10.1.61.11 on port 51784 index 25768 with 2006 bytes:
[431559,NET]
NOTIFY sip:111@10.1.61.158:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.1.61.11:51784;branch=z9hG4bK08ccf329
To: <sip:111@10.1.61.158>;tag=1598606730
From: <sip:111@10.1.61.11>;tag=f87b204eed990c3a4020c613-5969341f
Call-ID: 4085c80-d761e7a6-1996d-9e3d010a@10.1.61.158
Date: Tue, 10 Sep 2019 00:00:40 GMT
CSeq: 1000 NOTIFY
Event: refer
Subscription-State: terminated; reason=timeout
Max-Forwards: 70
Contact: <sip:e2881942-2853-4eab-a0d9-
96228c79d062@10.1.61.11:51784;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEPF87B204EED99"
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE, SUBSCRIBE
Content-Type: multipart/mixed; boundary=uniqueBoundary
Mime-Version: 1.0
Content-Length: 1199
--uniqueBoundary
Content-Type:application/x-cisco-remotecc-response+xml
Content-Disposition_session; handling=required
<?xml version=1.0" enconding="UTF-8"?>
<x-cisco-remotecc-response>
<response>
<code>200</code>
<reason></reason>
<applicationid>0</applicationid>
<transactionid>109</transactionid>
<stationsequence>StationSequenceLast</stationsequence>
<displaypriority>2</displaypriority>
<appinstance>0</appintance>
linenumber>0</linenumber>
<routingid>0</routingid>
<confid>0</confid>
<callid></callid>
<options_ind>
<combine max="0">
<service-control></service-control>
</combine>
<dialog usage="">
<unot></unot>
<sub></sub>
</dialog>
<presence usage="">
<unot></unot>
<sub></sub>
</presence>
</options_ind>
</response>
</x-cisco-remotecc-response>
--uniqueBoundary
Content-Type:application/x-cisco-remote-cm+xml
Csontent-Disposition:session;handling=required
<?xml version="1.0" encoding="utf-8"?>
<CiscoIPPhoneResponse>
<ResponseItem URL="RTPMRx:239.0.1.2:20480" Data="Success" Status="0"/>
```

</CiscoIPPhoneResponse>

--uniqueBoundary--

CUCM send a 200 OK for the NOTIFY received

71439556.001 |19:00:38.464 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.1.61.11 on port 51784 index 25768 [431560,NET] SIP/2.0 200 OK Via: SIP/2.0/TCP 10.1.61.11:51784;branch=z9hG4bK08ccf329 From: <sip:111@10.1.61.11>;tag=f87b204eed990c3a4020c613-5969341f To: <sip:111@10.1.61.158>;tag=1598606730 Date: Tue, 10 Sep 2019 00:00:38 GMT Call-ID: 4085c80-d761e7a6-1996d-9e3d010a@10.1.61.158 CSeq: 1000 NOTIFY Server: Cisco-CUCM11.5 Content-Length: 0

CUCM sends to the phone B a REFER to stop receiving multicast audio

71442357.002 |19:01:10.795 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.1.61.11 on port 51784 index 25768 [431582,NET] REFER sip:e2881942-2853-4eab-a0d9-96228c79d062@10.1.61.11:51784;transport=tcp SIP/2.0 Via: SIP/2.0/TCP 10.1.61.158:5060;branch=z9hG4bK199754588a6e3 From: <sip:111@10.1.61.158>;tag=928499252 To: <sip:111@10.1.61.11> Call-ID: 171b2c80-d761e7c6-19970-9e3d010a@10.1.61.158 CSeq: 101 REFER Max-Forwards: 70 Contact: <sip:111@10.1.61.158:5060;transport=tcp> User-Agent: Cisco-CUCM11.5 Expires: 30 Refer-To: cid:1234567890@10.1.61.158 Content-Id: <1234567890@10.1.61.158> Content-Type: multipart/mixed;boundary=uniqueBoundary Mime-Version: 1.0 Referred-By: <sip:111@10.1.61.158> Content-Length: 683 --uniqueBoundary Content-Type:application/x-cisco-remotecc-request+xml <x-cisco-remotecc-request> <datapassthroughreq> <applicationid>0</applicationid> <lineid>0</lineid> <transactionid>109</transactionid> <stationsequence>StationSequenceLast</stationsequence> <displaypriority>2</displaypriority> <appinstance>0</appintance> <routingid>0</routingid> <confid>0</confid> <featuredata></featuredata> </datapassthroughreq>

</x-cisco-remotecc-request>
--uniqueBoundary
Content-Type:application/x-cisco-remote-cm+xml
<CiscoIPPhoneExecute><ExecuteItem Priority="0" URL="**RTPMRx:Stop**"/></CiscoIPPhoneExecute>
--uniqueBoundary--

Phone B sends to CUCM a 202 Accepted

71442358.002 |19:01:10.802 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.1.61.11 on port 51784 index 25768 with 571 bytes: [431583,NET] SIP/2.0 202 Accepted

Via: SIP/2.0/TCP 10.1.61.158:5060;branch=z9hG4bK199754588a6e3 From: <sip:111@10.1.61.158>;tag=928499252 To: <sip:111@10.1.61.11>;tag=f87b204eed990c3e1c1bfe96-1d092704 Call-ID: 171b2c80-d761e7c6-19970-9e3d010a@10.1.61.158 Date: Tue, 10 Sep 2019 00:01:12 GMT CSeq: 101 REFER Server: Cisco-CP8811/12.0.1 Contact: <sip:e2881942-2853-4eab-a0d9-96228c79d062@10.1.61.11:51784;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEPF87B204EED99" Content-Length: 0 #### A NOTIFY is sent from the phone B to CUCM to indicate that it stopped receiving multicast audio 71442417.004 |19:01:11.069 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.1.61.11 on port 51784 index 25768 with 1994 bytes: [431584,NET] NOTIFY sip:111@10.1.61.158:5060;transport=tcp SIP/2.0 Via: SIP/2.0/TCP 10.1.61.11:51784; branch=z9hG4bK68d7f530 To: <sip:111@10.1.61.158>;tag=928499252 From: <sip:111@10.1.61.11>;tag=f87b204eed990c3e1c1bfe96-1d092704 Call-ID: 171b2c80-d761e7c6-19970-9e3d010a@10.1.61.158 Date: Tue, 10 Sep 2019 00:01:13 GMT CSeq: 1000 NOTIFY Event: refer Subscription-State: terminated; reason=timeout Max-Forwards: 70 Contact: <sip:e2881942-2853-4eab-a0d9-96228c79d062@10.1.61.11:51784;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEPF87B204EED99" Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE, SUBSCRIBE Content-Type: multipart/mixed; boundary=uniqueBoundary Mime-Version: 1.0 Content-Length: 1187 --uniqueBoundary Content-Type:application/x-cisco-remotecc-request+xml Content-Disposition:session;handling=required <?xml version="1.0" encoding="UTF-8"?> <x-cisco-remotecc-response> <response> <code>200</code> <reason></reason> <applicationid>0</applicationid> <transactionid>117</transactionid> <stationsequence>StationSequenceLast</stationsequence> <displaypriority>2</displaypriority> <appinstance>0</appinstance> linenumber>0</linenumber> <routingid>0</routingid> <confid>0</confid> <callid></callid> <options ind> <combine max="0"> <service-control></service-control> </combine> <dialog usage=""> <unot></unot> </dialog> <presence usage="">

<unot></unot>

</presence> </options_ind> </response> </x-cisco-remotecc-response> --uniqueBoundary Content-Type: application/x-cisco-remotecc-cm+xml Content-Disposition: session; handling=required <?xml version="1.0" encoding="utf-8"?> <CiscoIPPhoneResponse> <ResponseItem URL="RTPRx:Stop" Data="Success" Status="0" /> </CiscoIPPhoneResponse> --uniqueBoundary-### CUCM replies with 200 OK 71442425.001 |19:01:11.070 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.1.61.11 on port 51784 index 25768 [431585,NET] SIP/2.0 200 OK Via: SIP/2.0/TCP 10.1.61.11:51784;branch=z9hG4bK68d7f530 From: <sip:111@10.1.61.11>;tag=f87b204eed990c3e1c1bfe96-1d092704 To: <sip:111@10.1.61.158>;tag=928499252 Date: Tue, 10 Sep 2019 00:01:11 GMT Call-ID: 171b2c80-d761e7c6-19970-9e3d010a@10.1.61.158 CSeq: 1000 NOTIFY Server: Cisco-CUCM11.5 Content-Length: 0

用於CTI整合和由HTTP控制的電話

CUCM:10.1.61.158

Informacast:10.1.61.118

電話A

DN:110

型號:CP-8861

韌體版本:sip88xx.12-0-1SR1-1

電話A IP地址: 10.1.61.12

MAC:SEP2C3124C9F8E1

電話B

DN:111

型號:CP-8811

韌體版本:sip88xx.12-0-1SR1-1

電話B IP地址:10.1.61.11

MAC:SEPF87B204EED99

CUCM receives the INVITE from phone A (Call Manager SDL Log) 71531116.002 |19:15:32.972 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.1.61.12 on port 51600 index 25770 with 1791 bytes: [431985,NET] INVITE sip:7@10.1.61.158;user=phone SIP/2.0 Via: SIP/2.0/TCP 10.1.61.12:51600;branch=z9hG4bK112766fc From: "PhoneA" <sip:110@10.1.61.158>;tag=2c3124c9f8e10c541ed075c2-67793e32 To: <sip:7@10.1.61.158> Call-ID: 2c3124c9-f8e10011-0bb54030-57b0a7c8@10.1.61.12 Max-Forwards: 70 Date: Tue, 10 Sep 2019 00:15:35 GMT CSeq: 101 INVITE User-Agent: Cisco-CP8861/12.0.1 Contact: <sip:142b9f25-7f2b-48a8-9ff9-377f616f3084@10.1.61.12:51600;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP2C3124C9F8E1" Expires: 180 Accept: application/sdp Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE, SUBSCRIBE, INFO Remote-Party-ID: "PhoneA" <sip:110@10.1.61.158>;party=calling;idtype=subscriber;privacy=off;screen=yes Supported: replaces, join, sdp-anat, norefersub, resource-priority, extended-refer, X-ciscocallinfo,X-cisco-serviceuri,X-cisco-escapecodes,X-cisco-service-control,X-cisco-srtp-fallback,Xcisco-monrec,X-cisco-config,X-cisco-sis-7.0.0,X-cisco-xsi-8.5.1 Allow-Events: kpml,dialog Recv-Info: conference Recv-Info: x-cisco-conference Content-Length: 548 Content-Type: application/sdp Content-Disposition: session; handling=optional v=0o=Cisco-SIPUA 19108 0 IN IP4 10.1.61.12 s=STP Call b=AS:4064 t=0 0 m=audio 19104 RTP/AVP 114 9 124 0 8 116 18 101 c=IN IP4 10.1.61.12 b=TIAS:64000 a=rtpmap:114 opus/48000/2 a=fmtp:114 maxplaybackrate=16000;spropmaxcapturerate=16000;maxaveragebitrate=64000;stereo=0;sprop-stereo=0;usedtx=0 a=rtpmap:9 G722/8000 a=rtpmap:124 ISAC/16000 a=rtpmap:0 PCMU/8000 a=rtpmap:8 PCMA/8000 a=rtpmap:116 iLBC/8000 a=fmtp:116 mode=20 a=rtpmap:18 G729/8000 a=fmtp:18 annexb=yes a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-15 a=sendrecv #### Digit analysis for the dialed number 7778 71531367.000 |19:15:34.231 |Sdlsig DaReq wait Da(1,100,216,1) Cdcc(1,100,224,12) 1,100,14,1368.88^10.1.61.12^* |[R:N-H:0,N:0,L:0,V:0,Z:0,D:0] CI=19282358 Fqdn=ti=1nd=110pi=0si1 Cgpn=tn=0npi=0ti=1nd=110pi=1si1

DialedNum=tn=0npi=1ti=1nd=7778User=7778Host=10.1.61.158Port=5060PassWord=Madder=Transport=4mDisp

layName=RawUrl=sip:7@10.1.61.158;user=phoneOrigPort=0pi=0sil requestID=0

```
DigitAnalysisComplexity=1 CallingUser= IgnoreIntercept=0 callingDeviceName=SEP2C3124C9F8E1
71531367.001 |19:15:34.231 |AppInfo |Digit Analysis: star_DaReq:
daReq.partitionSearchSpace(8653f609-05a7-5914-819b-3a89680af6a2:),
filteredPartitionSearchSpaceString(Informacast_PT:phone_pt),
partitionSearchSpaceString(Informacast_PT:phone_pt)
71531367.002 |19:15:34.231 | AppInfo | Digit Analysis: Host Address=10.1.61.158 MATCHES this
node's IPv4 address.
71531367.003 |19:15:34.231 |AppInfo |Digit Analysis: star_DaReq: Matching SIP URL, Numeric
User, user=7778
71531367.004 |19:15:34.232 |AppInfo |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept
DAMR.sstype=[0], TPcount=[0], DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]
71531367.005 |19:15:34.232 |AppInfo |Digit Analysis: getDaRes - Remote Destination [] isURI[1]
71531367.006 |19:15:34.232 |AppInfo |Digit analysis: patternUsage=2
71531367.007 |19:15:34.232 |AppInfo |Digit analysis: match(pi="2", fqcn="110",
cn="110",plv="5", pss="Informacast_PT:phone_pt", TodFilteredPss="Informacast_PT:phone_pt",
dd="7778",dac="1")
71531367.008 |19:15:34.232 |AppInfo |Digit analysis: analysis results
71531367.009 |19:15:34.232 |AppInfo ||PretransformCallingPartyNumber=110
CallingPartyNumber=110
|DialingPartition=Informacast_PT
DialingPattern=7778
FullyQualifiedCalledPartyNumber=7778
|DialingPatternRegularExpression=(7778)
|DialingWhere=
|PatternType=Enterprise
PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=7778
PretransformTagsList=SUBSCRIBER
PretransformPositionalMatchList=7778
CollectedDigits=7778
UnconsumedDigits=
|TagsList=SUBSCRIBER
|PositionalMatchList=7778
VoiceMailbox=
VoiceMailCallingSearchSpace=
VoiceMailPilotNumber=
RouteBlockFlag=RouteThisPattern
RouteBlockCause=0
|AlertingName=InformacastCTIRP
|UnicodeDisplayName=InformacastCTIRP
|DisplayNameLocale=1
OverlapSendingFlagEnabled=0
WithTags=
WithValues=
|CallingPartyNumberPi=NotSelected
|ConnectedPartyNumberPi=NotSelected
CallingPartyNamePi=NotSelected
|ConnectedPartyNamePi=NotSelected
CallManagerDeviceType=NoDeviceType
PatternPrecedenceLevel=Routine
[CallableEndPointName=[4db482c3-64c3-5adf-33c5-allc890d96d0]
PatternNodeId=[4db482c3-64c3-5adf-33c5-a11c890d96d0]
AARNeighborhood=[]
[AARDestinationMask=[]
AARKeepCallHistory=true
AARVoiceMailEnabled=false
NetworkLocation=OnNet
Calling Party Number Type=Cisco Unified CallManager
Calling Party Numbering Plan=Cisco Unified CallManager
Called Party Number Type=Cisco Unified CallManager
Called Party Numbering Plan=Cisco Unified CallManager
ProvideOutsideDialtone=false
AllowDeviceOverride=false
```

IsEmergencyNumber=false
AlternateMatches=
TranslationPatternDetails=
ResourcePriorityNamespace=
PatternRouteClass=RouteClassDefault

CUCM extends the call to the Line control associated to the CTI Route Point ICVA_CTI_RP (Call Manager SDL Log)

71531370.001 |19:15:34.232 |AppInfo |Digit analysis: wait_DmPidRes- Partition=[107a02ea-a384-5219-3670-ba9d14b9d094] Pattern=[7778] Where=[],cmDeviceType=[UserDevice], OutsideDialtone =[0], DeviceOverride=[0], PID=LineControl(1,100,178,1306),CI=[19282358],Sender=Cdcc(1,100,224,12) 71531386.001 |19:15:34.233 |AppInfo |LineCdpc(20): -dispatchToAllDevices-, sigName=CcSetupReq, device=ICVA_CTI_RP

CUCM sends the CTI New call notify (Call Manager SDL Log)

71531404.000 |19:15:34.235 |SdlSig-0 |CtiNewCallNotify NA RemoteSignal UnknownProcessName(1,200,25,1) |StationCdpc(1,100,67,2) |1,100,14,1.33^*^* [R:N-H:0,N:4,L:0,V:0,Z:0,D:0] LH=1|47 GCH=1|15018 CH=1|19282359 Held CH=0|0 State=2(CtiOfferingState) Reason=1 Origin=1 DeviceName=ICVA_CTI_RP CGPN=[DN=110 uDN=110 NumPI=T Part=phone_pt VmBox= NumType=0 Name=PhoneA UniName=PhoneA NamePI=T Locale=1 PU=2 Device=SEP2C3124C9F8E1 GlblCgpn=110] CDPN=[DN=7778 uDN=7778 NumPI=T Part=Informacast_PT VmBox= NumType=0 Name=InformacastCTIRP UniName=InformacastCTIRP NamePI=T Locale=1 PU=2 Device=] LRP=[DN= uDN= NumPI=T Part= VmBox= NumType=0 Name= UniName= NamePI=T Locale=1] OCDPN=[DN=7778 uDN=7778 NumPI=T Part=Informacast_PT VmBox= NumType=0 Name=InformacastCTIRP UniName=InformacastCTIRP NamePI=T Locale=1] AuxData=T FarEndCMId=1 EndpointType=1 RIU=F Privacy=F CallPresent=T FeatPriority=1 Feature=137 AttrType=0 LineId [DN=110 Part=phone_pt] IPAddrMode=0 IsConsCallDueToRollover=F UniqCallRef=0000000000003AAA012639B700000000 CgpnIPv4Addr=c3d010a CgpnIPv6Addr= CallingMultiMediaCap=0F0 CalledMultiMediaCap=0F0 CallingPartyMultiMediaMask=3 CalledPartyMultiMediaMask=3 Session-ID: Device= 5ee92aa5415831d8b114c4ba19282359; Remote= 02023b9b00105000a0002c3124c9f8e1

CTI process receives the CtiNewCallNotify from CallManager process (CTI Manager SDL Trace) 04961495.000 |19:15:34.236 |SdlSig-I |CtiNewCallNotify CTIDeviceLineMgr(1,200,25,1) ready StationCdpc(1,100,67,2) |1,100,14,1.33^*^* [R:N-H:0,N:1,L:0,V:0,Z:0,D:0] LH=1|47 GCH=1|15018 CH=1|19282359 Held CH=0|0 State=2(CtiOfferingState) Reason=1 Origin=1 DeviceName=ICVA_CTI_RP CGPN=[DN=110 UDN=110 NumPI=T Part=phone_pt VmBox= NumType=0 Name=PhoneA UniName=PhoneA NamePI=T Locale=1 PU=2 Device=SEP2C3124C9F8E1 GlblCgpn=110] CDPN=[DN=7778 uDN=7778 NumPI=T Part=Informacast_PT VmBox= NumType=0 Name=InformacastCTIRP UniName=InformacastCTIRP NamePI=T Locale=1 PU=2 Device=] LRP=[DN= uDN= NumPI=T Part= VmBox= NumType=0 Name= UniName= NamePI=T Locale=1] OCDPN=[DN=7778 uDN=7778 NumPI=T Part=Informacast_PT VmBox= NumType=0 Name=InformacastCTIRP UniName=InformacastCTIRP NamePI=T Locale=1] AuxData=T FarEndCMId=1 EndpointType=1 RIU=F Privacy=F CallPresent=T FeatPriority=1 Feature=137 AttrType=0 LineId [DN=110 Part=phone_pt] IPAddrMode=0 IsConsCallDueToRollover=F UniqCallRef=0000000003AAA012639B700000000 CgpnIPv4Addr=c3d010a CgpnIPv6Addr= CallingMultiMediaCap=0F0 CalledMultiMediaCap=0F0 CallingPartyMultiMediaMask=3 CalledPartyMultiMediaMask=3 Session-ID: Device= 5ee92aa5415831d8b114c4ba19282359; Remote= 02023b9b00105000a0002c3124c9f8e1

CTI process sends the NewCallEvent to Informacast server (CTI Manager SDL Trace)

04961497.003 |19:15:34.236 |AppInfo |[CTI-APP] [CTIHandler::OutputCtiMessage 1 CTI NewCallEvent (LH=1|46 CH=1|19282359 CH=0|0 GCH=1|15018 lineHandleSpecified=1 state=2 origin=1 farEndpointSpecified=1 farEndpointCMID=1 endpointType=1 reason=1 remote in use=0 privacy=0 mediaResourceID= resource ID=0 deviceName=ICVA_CTI_RP cgpn=110 Presentation=1 cgpn NameInfo=locale: 1 pi: 1 Name: PhoneA UnicodeName: PhoneA cdpn=7778 Presentation=1 cdpn NameInfo=locale: 1 pi: 1 Name: InformacastCTIRP UnicodeName: InformacastCTIRP original cdpn=7778 Presentation=1 original cdpn NameInfo=locale: 1 pi: 1 Name: InformacastCTIRP UnicodeName: InformacastCTIRP LRP= Presentation=1 LRP NameInfo=locale: 1 pi: 1 Name: UnicodeName: UserData= callingPartyDeviceName=SEP2C3124C9F8E1 mediaDeviceName= ucgpn=110 ucdpn=7778 unmodifiedOriginal cdpn=7778 uLRP= cgPnPartition=phone_pt cdPnPartition=Informacast_PT oCdPnPartition=Informacast_PT lrpPartition= CgpnIP=0xc3d010a IsConsultCallDueToRollover=0 apiCallReference=00000000003AAA012639B700000000 lineId.DN=110 lineId.part=phone_pt CallPresentable=1 FeaturePriority =1 globalizedCgPn=110 ipAddrMode=0 cgpnPU=2

cdpnPU=2CallingPartyMultiMediaBitMask=3CalledPartyMultiMediaBitMask=3 Session-ID: Device= 5ee92aa5415831d8b114c4ba19282359; Remote= 02023b9b00105000a0002c3124c9f8e1

CTI process receives the LineCallAcceptRequest from Informacast server (CTI Manager SDL Trace)

04961500.002 |19:15:34.242 |AppInfo |[CTI-APP] [CTIHandler::processIncomingMessage] CTI LineCallAcceptRequest (seq#=33 LH=1|46 CH=1|19282359 media resource ID= resource ID=0 media device name=)

 #### CTI process sends the answer to Call Manager process (CTI Manager SDL Trace)

 04961503.000 |19:15:34.242 |SdlSig-0 |CtiLineCallAcceptReq
 NA

 RemoteSignal
 |UnknownProcessName(1,100,66,16) |CTIDeviceLineMgr(1,200,25,1)

 1,200,13,90.89^10.1.61.118^ICVA_CTI_RP
 |[R:N-H:0,N:0,L:0,V:0,Z:0,D:0] AsyncResponse=124

 CH=1 |19282359 LH=1 |47 MediaDeviceName = MediaDevicePid = (0,0,0,0) resource ID=0

Call Manager process receives the answer from CTI process (Call Manager SDL Log)
71531414.000 |19:15:34.243 |SdlSig-I |CtiLineCallAcceptReq
|restart0 |StationD(1,100,66,16)
|CTIDeviceLineMgr(1,200,25,1) |1,200,13,90.89^10.1.61.118^ICVA_CTI_RP |[R:NH:0,N:0,L:0,V:0,Z:0,D:0] AsyncResponse=124 CH=1|19282359 LH=1|47 MediaDeviceName =
MediaDevicePid = (0,0,0,0) resource ID=0

CTI Process receives from Informacast the port to be used to receive the audio (CTI Manager SDL Trace)

04961525.002 |19:15:34.256 |AppInfo |[CTI-APP] [CTIHandler::processIncomingMessage] CTI DeviceSetRTPForCallRequest (seq#=35 DH=1|52 CH=1|19282359 RtpDestination=1983709450|32080)

CTI Process sends the port to Call manager process (CTI Manager SDL Trace)

04961528.000 |19:15:34.256 |SdlSig-O |CtiDeviceSetRTPForCallReq |NA RemoteSignal |UnknownProcessName(1,100,66,16) |CTIDeviceLineMgr(1,200,25,1) |1,200,13,90.91^10.1.61.118^ICVA_CTI_RP |[R:N-H:0,N:0,L:0,V:0,Z:0,D:0] AsyncResponse=126mCtiInterface(1,200,25,1) DH=1|53 CH=1|19282359 RtpDestination1983709450|32080

CUCM sends the 200 OK to the Phone A (Codec PCMU, IP and port of Informacast)

71531593.001 |19:15:34.258 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.1.61.12 on port 51600 index 25770 [432000,NET] SIP/2.0 200 OK Via: SIP/2.0/TCP 10.1.61.12:51600;branch=z9hG4bK112766fc From: "PhoneA" <sip:110@10.1.61.158>;tag=2c3124c9f8e10c541ed075c2-67793e32 To: <sip:7@10.1.61.158>;tag=229579~7cc9781e-f7e3-4c51-a2b9-de353a4e7d6f-19282358 Date: Tue, 10 Sep 2019 00:15:32 GMT Call-ID: 2c3124c9-f8e10011-0bb54030-57b0a7c8@10.1.61.12 CSeq: 101 INVITE Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY Allow-Events: presence Supported: replaces Server: Cisco-CUCM11.5 Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; gci= 1-15018; isVoip; call-instance= 1 Send-Info: conference, x-cisco-conference Session-ID: 5ee92aa5415831d8b114c4ba19282359;remote=02023b9b00105000a0002c3124c9f8e1 Remote-Party-ID: "InformacastCTIRP" <sip:7778@10.1.61.158>;party=called;screen=yes;privacy=off Contact: <sip:7@10.1.61.158:5060;transport=tcp> Content-Type: application/sdp Content-Length: 179 v=0o=CiscoSystemsCCM-SIP 229579 1 IN IP4 10.1.61.158 s=SIP Call c=IN IP4 10.1.61.118 b=AS:64 t=0 0 m=audio 32080 RTP/AVP 0

b=TIAS:64000
a=ptime:20
a=rtpmap:0 PCMU/8000

ACK from Phone A to CUCM

71531622.002 |19:15:34.473 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.1.61.12 on port 51600 index 25770 with 692 bytes: [432004,NET] ACK sip:7@10.1.61.158:5060;transport=tcp SIP/2.0 Via: SIP/2.0/TCP 10.1.61.12:51600;branch=z9hG4bK4fcbad6d From: "PhoneA" <sip:110@10.1.61.158>;tag=2c3124c9f8e10c541ed075c2-67793e32 To: <sip:7@10.1.61.158>;tag=229579~7cc9781e-f7e3-4c51-a2b9-de353a4e7d6f-19282358 Call-ID: 2c3124c9-f8e10011-0bb54030-57b0a7c8@10.1.61.12 Max-Forwards: 70 Session-ID: 02023b9b00105000a0002c3124c9f8e1;remote=5ee92aa5415831d8b114c4ba19282359 Date: Tue, 10 Sep 2019 00:15:37 GMT CSeq: 101 ACK User-Agent: Cisco-CP8861/12.0.1 Remote-Party-ID: "PhoneA" <sip:110@10.1.61.158>;party=calling;idtype=subscriber;privacy=off;screen=yes Content-Length: 0 Recv-Info: conference Recv-Info: x-cisco-conference

NOTE: At this point the call from phone A to Informacast has been established successfully. For this scenario the phones are activated using HTTP, hence there are no CUCM logs related to the phone activation.

效能日誌

對於SIP整合

Informacast receives an INVITE sent by CUCM

2019-09-09 19:09:42,323 [pool-41-thread-1] INFO ba [] - Received INVITE request; call ID 2d72f80-d76le7a4-1996c-9e3d010a@10.1.61.158; from "PhoneA" <sip:110@10.1.61.158; to <sip:7777@10.1.61.118>; contact <sip:110@10.1.61.158:5060;transport=tcp>; user-agent Cisco-CUCM11.5

Informacast sends a 200 OK to CUCM

2019-09-09 19:09:42,508 [pool-41-thread-1] INFO ba [] - Sent INVITE response; status OK (200); call ID 2d72f80-d761e7a4-1996c-9e3d010a@10.1.61.158; from "PhoneA" <sip:110@10.1.61.158; to <sip:7777@10.1.61.118; contact "InformaCast" <sip:7777@10.1.61.118; transport=tcp>

CUCM replies with ACK to Informacast

2019-09-09 19:09:42,527 [pool-41-thread-1] INFO ba [] - Received ACK request; call ID 2d72f80d761e7a4-1996c-9e3d010a@10.1.61.158; from "PhoneA" <sip:110@10.1.61.158>; to <sip:7777@10.1.61.118>; user-agent Cisco-CUCM11.5

Informacast provides the IP and port

2019-09-09 19:09:42,871 [pool-1264-thread-1] INFO u [] - providing address: 239.0.1.2 2019-09-09 19:09:42,885 [pool-1264-thread-1] INFO t [] - Gathering information required to send the message 2019-09-09 19:09:42,904 [pool-1264-thread-1] INFO t [] - Broadcast will be sent on port: 20480

Stream settings:

2019-09-09 19:09:43,556 [Signaler # 1 run 1] INFO Signaler [] - Stream settings: General info: User=dialcast(System User), BroadcastInitiator=10.1.61.12,

SourceType=CallingPhone, MessageKey=908, MessageType=Live Audio, MessageDescription=Basic Paging Live Broadcast, RecipientGroupDescription=SanJose, MaxIPPhones=50, MaxIPSpeakers=0, DeviceArbiter=null, CreatedOn=Mon Sep 09 19:09:42,849 CDT 2019, PauseLength=0, NumberOfRepetitions=1

Audio details: AudioFile=null, AudioFormat=ULAW 8000.0 Hz, 8 bit, mono, 1 bytes/frame, ,

RemoteAddress=239.0.1.2, RemotePort=20480, MessageVolume=As-Is, NonUrgent=true, Interrupt=false, Priority=2, LiveAudioSource=LiveBroadcastTriggerTask[callID=2d72f80-d761e7a4-1996c-9e3d010a@10.1.61.158, callMapper=CallMapper[dialedNumber=7777 isMapped=true messageId=908 recipientIds=[1714] dialcode=null dn=null], multicastAddress=null, multicastPort=0, triggerFailAudioFile=/usr/local/singlewire/InformaCast/web/sounds/ivr/broadcastTrigger/triggerFa il.ulaw.wav, preToneFile=null, postToneFile=null, recordedFile=null, recordingStarted=false, done=false], PreTone=null, PostTone=null, HasDynamicAudio=falseReplay=false Confirmation details: CollectConfirmations=false

Informacast sends the instruction message to 1 participant (SEPF87B204EED99)

2019-09-09 19:09:43,555 [Signaler # 1 run 1] INFO Signaler [] - Sending message to 1 participants 2019-09-09 19:09:43,643 [Push:10.1.61.11-pool-1269-thread-1] INFO i [1 run 1] - Started device instructor for phone PhoneDescription (deviceType=36670, deviceName=SEPF87B204EED99, description=Auto 111, devicePool=Default, callingSearchSpace=, address=10.1.61.11, ctiUser=ICVAInformacast, ctiPassword=[hidden], location=Hub_None, profileDescription=null, pbxDescription=CUCM)

Informacast received the response via JTAPI from the phone

2019-09-09 19:09:44,126 [Push:10.1.61.11-pool-1269-thread-1] INFO i [1 run 1] - The response from the phone SEPF87B204EED99 via JTAPI is: <?xml version="1.0" encoding="UTF-8"?> <CiscoIPPhoneResponse> <ResponseItem URL="RTPMRx:239.0.1.2:20480" Data="Success" Status="0" /> </CiscoIPPhoneResponse>

Informacast starts broadcasting

2019-09-09 19:09:44,151 [pool-1269-thread-1] INFO ah [] - Starting broadcast for inbound call 2d72f80-d761e7a4-1996c-9e3d010a@10.1.61.158 on multicast address /239.0.1.2 and port 20480

Informacast receives the BYE to end the paging

2019-09-09 19:10:15,222 [pool-41-thread-1] INFO ba [] - Received BYE request; call ID 2d72f80d761e7a4-1996c-9e3d010a@10.1.61.158; from "PhoneA" <sip:110@10.1.61.158>; to <sip:7777@10.1.61.118>; user-agent Cisco-CUCM11.5

Informacast sends to the phone the instruction to stop receiving audio

2019-09-09 19:10:16,403 [Push:10.1.61.11-pool-1269-thread-3] INFO i [1 run 1] - Pushing stop command to phone: PhoneDescription (deviceType=36670, deviceName=SEPF87B204EED99, description=PhoneB, devicePool=Default, callingSearchSpace=, address=10.1.61.11, ctiUser=ICVAInformacast, ctiPassword=[hidden], location=Hub_None, profileDescription=null, pbxDescription=CUCM)

Informacast receives the response from the phone

2019-09-09 19:10:16,732 [Push:10.1.61.11-pool-1269-thread-3] INFO i [1 run 1] - The response from the phone SEPF87B204EED99 via JTAPI is:

<?xml version="1.0" encoding="UTF-8"?>
<CiscoIPPhoneResponse>
<ResponseItem URL="RTPMRx:Stop" Data="Success" Status="0" />
</CiscoIPPhoneResponse>
Task ended
2019-09-09 19:10:19,357 [DeviceDeactivator-pool-1268-thread-1] INFO ah [1] - Canceling live
broadcast for inbound call 2d72f80-d761e7a4-1996c-9e3d010a@10.1.61.158
2019-09-09 19:11:45,250 [Timer-0] INFO JavaExchangeAdapter [] - Task Ended: checkpoint command

```
to compact the database
```



Informacast receives the request to route the call

2019-09-09 19:24:39,936 [RouteCall:15018/1Thread] INFO av [] - Route request for call [CiscoCallID=15018/1 callingDN=110 callingPartition=phone_pt callingTerminal=SEP2C3124C9F8E1 lastRedirectedDN=null modifiedCalledDN=7778 currentCalledDN=7778 calledDN=7778] on ICVA_CTI_RP,7778

Dialing pattern matches

2019-09-09 19:24:39,942 [ObserverThread(af@feaf7c)] INFO V [] - Dialing pattern "7778" matched dialed route point number 7778

Informacast provides the IP and port for multicast

2019-09-09 19:24:40,020 [pool-1287-thread-1] INFO u [] - providing address: 239.0.1.2 2019-09-09 19:24:40,020 [pool-1287-thread-1] INFO t [] - Gathering information required to send the message 2019-09-09 19:24:40,023 [pool-1287-thread-1] INFO t [] - Broadcast will be sent on port: 20486

Informacast sends the message to all devices in the recipient group, in this case to only 1 device

2019-09-09 19:24:40,262 [Signaler # 4 run 1] INFO Signaler [] - Sending message to 1 participants

Informacast starts the live broadcast over the IP and port

2019-09-09 19:24:40,263 [Signaler # 4 run 1] INFO ah [] - Starting live broadcast alert for inbound call 15018/1 on multicast address /239.0.1.2 and port 20486

Informacast sends the instruction activate the phone (SEPF87B204EED99) and join to the multicast audio

2019-09-09 19:24:40,278 [Push:10.1.61.11-pool-1269-thread-10] INFO i [4 run 1] - Started device instructor for phone PhoneDescription (deviceType=36670, deviceName=SEPF87B204EED99, description=PhoneB, devicePool=Default, callingSearchSpace=, address=10.1.61.11, ctiUser=ICVAInformacast, ctiPassword=[hidden], location=Hub_None, profileDescription=null, pbxDescription=CUCM)

Informacast receives the response from the phone

2019-09-09 19:24:40,624 [Push:10.1.61.11-pool-1269-thread-10] INFO i [4 run 1] - The response from the phone is:

Informacast starts the broadcast over the IP and port

2019-09-09 19:24:40,637 [pool-1269-thread-10] INFO ah [] - Starting broadcast for inbound call 15018/1 on multicast address /239.0.1.2 and port 20486

Informacast receives the notification that the call has ended

2019-09-09 19:25:21,253 [ObserverThread(af@feaf7c)] INFO af [] - RTP input stopped event received for inbound call 15018/1

Informacast sends the instruction to the phones in order to stop receiving audio

2019-09-09 19:25:21,865 [Push:10.1.61.11-pool-1269-thread-12] INFO i [4 run 1] - Pushing stop command to phone: PhoneDescription (deviceType=36670, deviceName=SEPF87B204EED99, description=PhoneB, devicePool=Default, callingSearchSpace=, address=10.1.61.11, ctiUser=ICVAInformacast, ctiPassword=[hidden], location=Hub_None, profileDescription=null, pbxDescription=CUCM)

Informacast receives the response from the phone

2019-09-09 19:25:22,123 [Push:10.1.61.11-pool-1269-thread-12] INFO i [4 run 1] - The response from the phone is:

Deactivation done

2019-09-09 19:25:22,134 [pool-1269-thread-12] INFO ah [] - Canceling live broadcast for inbound call 15018/1 2019-09-09 19:25:22,134 [pool-1269-thread-12] INFO Signaler [] - Notifying signaler that the deactivator is done

控制檯日誌(PRT)

The same IP and port for multicast provided by Informacast is shown in the console logs 5311 INF Sep 10 00:15:34.434302 (701:844) JAVA-PushThread|cip.push.PushThread:execute - Sleep for 100ms previous= current=RTPMRx:239.0.1.2:20486 i=0 total=1 5312 DEB Sep 10 00:15:34.535773 (701:832) JAVA-SIPCC-MSP: mp_create_rtp_session: scheme_specific=239.0.1.2:20486 direction=0 mcast=1 payloadtype=4 framesize=20 vadenable=0 5313 DEB Sep 10 00:15:34.535893 (701:832) JAVA-SIPCC-MSP: mp_create_rtp_session: precedence=0 mixingmode=0 mixingparty=0 channeltype=0 5314 DEB Sep 10 00:15:34.535980 (701:832) JAVA-SIPCC-MSP: mp_create_rtp_session: ipv4 address/port/type [-1382943496/20486/1].

Create receive session only

5315 DEB Sep 10 00:15:34.536032 (701:832) JAVA-SIPCC-MSP: mp_create_rtp_session: Create Rx only stream. 5316 NOT Sep 10 00:15:34.536151 (408:408) ms-MSAPI.ms_forceReserveMediaPort port 20486 5317 NOT Sep 10 00:15:34.536291 (701:832) JAVA-SIPCC-MED_API: 0/-1, mp_create_rx_session: MCAP 0:GRP -1:STRM -1: PT 4: PRD 20: PORT 20486: DTPT 0: MCAST 1 5320 DEB Sep 10 00:15:34.536489 (701:832) JAVA-mp_create_rx_session:type=1, addr=239.0.1.2, ip4=-285212414 5321 DEB Sep 10 00:15:34.536525 (701:832) JAVA-mp_create_rx_session:addr_str=239.0.1.2 5323 DEB Sep 10 00:15:34.536661 (701:832) JAVA-mp_create_rx_session:[ToMS] payload=4 dynpayload=0 pkt_period=20 local_addr=239.0.1.2 type=0 local_port=20486 5326 NOT Sep 10 00:15:34.537528 (408:408) ms-RTPSESSION.createRTPSession media [ipv4=239.0.1.2][port=20486][interface=NULL][mediatype=4][relayee=0][groupid=4294967295][callid= 4294967295]

Start RTCP

5385 NOT Sep 10 00:15:34.673264 (408:408) ms-RTCPMGR.rtcpm_startRtcp[A:6:5:8] [local IPv4:port=239.0.1.2:20487][remote IPv4:port=0.0.0.0:0]

Start RTP session RX

5388 NOT Sep 10 00:15:34.673917 (408:408) ms-RTPSESSION.ms_startRTPSessionRx[A:6] START RX [stream=5][mediaType(codec)=4][pkt size=20][P-IPv4=239.0.1.2][Port=20486][groupid=-1][callid=-1]

Release connection

5536 NOT Sep 10 00:16:16.173301 (701:832) JAVA-SIPCC-MED_API: mp_session_cmd: release local rtp port 20486 5537 NOT Sep 10 00:16:16.173396 (408:408) ms-MSAPI.ms_releaseRxPort : port 20486

封包擷取

從電話收集資料包捕獲並驗證InformaCast中的HTTP XSI命令。為了加入組播流,傳送網際網路組 管理協定(IGMP)消息。如果您在IGMP消息之後沒有看到組播即時傳輸協定(RTP)流,您可以從 InformaCast捕獲資料包,確認Informacast伺服器已將RTP傳送到IP和埠,然後檢查網路基礎架構 。

電話上的資料包捕獲(由HTTP控制)

- CUCM:10.1.61.158
- Informacast:10.1.61.118
- 電話B IP地址: 10.1.61.11
- 型號: CP-8811
- 韌體版本: sip88xx.12-0-1SR1-1
- eth.addr==SEPF87B204EED99

電話上收到的HTTP和IGMP消息如下圖所示。

File	Edit	liew Go	Capture	Analyze Statistics Telephony	Wireless Tools Help					
	■ ₫ 6		S X	Q + + = = + = = =	0,0,0,11					
	ttp igmp							Expression		
No.	1	'ime		Source	Destination	Protocol	Length Info			
	1771 6	0:24:39.	352999	10.1.61.22	173.36.89.68	HTTP	2295 CCM_POST /ccm_system/request HTTP/1.1 (text/plain)			
	1777 6	0:24:39.	404529	173.36.89.68	10.1.61.22	HTTP	394 HTTP/1.1 200 OK (text/plain)	Informacast sends IP and port for multicast		
	1905 0	0:24:49.	392163	10.1.61.118	10.1.61.11	HTTP	223 GET /StreamingStatisticsX?1 HTTP/1.1	internaces series in and port for mendess		
	1911 (0:24:49.	444329	10.1.61.11	10.1.61.118	HTTP/XML	1452 HTTP/1.1 200 OK	The share with a first south before seat		
+	1917 (0:24:49.	453245	10.1.61.118	10.1.61.11	HTTP	399 POST /CGI/Execute HTTP/1.1 (application/x-www-form-urlencoded)	The phone authenticates with informacast		
	1922 @	0:24:49.	479784	10.1.61.11	10.1.61.118	HTTP	457 GET /InformaCast/phone/auth?UserID=ICVAInformacast&Password=rtpavvid	J&devicename=SEPF878204EED99 HTTP/1.1		
	1926 @	0:24:49.	483773	10.1.61.118	10.1.61.11	HTTP	76 HTTP/1.1 200 OK (text/html) <	Informacast replies with 200 OK		
	1932 0	0:24:49.	610049	10.1.61.11	239.0.1.2	IGMPv2	60 Membership Report group 239.0.1.2	Internacional de la compactación de		
-	1941 0	0:24:49.	735551	10.1.61.11	10.1.61.118	HTTP/XML	474 HTTP/1.1 200 OK			
	1965 6	0:24:50.	999480	10.1.61.11	239.0.1.2	IGMPv2	60 Membership Report group 239.0.1.2 The	phone joins to the Membership (IP and port)		
	2070 0	0:24:58.	399886	10.1.61.11	239.0.1.2	IGMPv2	60 Membership Report group 239.0.1.2			
	2512 0	0:25:30.	985190	10.1.61.118	10.1.61.11	HTTP	404 POST /CGI/Execute HTTP/1.1 (application/x-www-form-urlencoded)	Informacast gives the order to leave		
	2516 0	0:25:31.	228042	10.1.61.11	224.0.0.2	IGMPv2	60 Leave Group 239.0.1.2			
	2518 0	0:25:31.	234468	10.1.61.11	10.1.61.118	HTTP/XML	462 HTTP/1.1 200 OK	The phone leaves the IGMP group		
<								, , , , , , , , , , , , , , , , , , ,		
~	Hyperte	xt Trans	fer Pro	tocol						
	> POST	/CGI/Exe	cute H	TTP/1.1\r\n						
	> Auth	rization	: Basi	c SUNWQUluZm9ybWFjYXN0OnJ	0cGF2dmlk\r\n					
	User	Agent: 3	lakarta	Commons-HttpClient/3.1\r	\n					
	Host	10.1.61	.11\r\	n						
	> Cont	ent-Lengt	h: 116	\r\n						
	Cont	ent-Type:	appli	cation/x-www-form-urlenco	ded\r\n					
	\r\n									
	[Ful	request	URI:	http://10.1.61.11/CGI/Exe	cute]					
	(HTT	P request	: 1/1]							
	[Res	oonse in	frame:	1941]						
	File	Data: 11	6 byte	s						
¥-	HTML Form URL Encoded: application/xwww.form.urlencoded									
	> Earm	Shame Hy		" of a control on a factor when a factor	and Them UDI - "OTOM		- 20406" (> < / Circle and TOOL and Francisco V			

電話上的資料包捕獲(由JTAPI控制)

• CUCM:10.1.61.158

- Informacast:10.1.61.118
- 電話B IP地址: 10.1.61.11
- 型號: CP-8811
- 韌體版本:sip88xx.12-0-1SR1-1
- MAC SEPF87B204EED99

如配置一節所述,電話可以由JTAPI控制,這表示啟用Send Commands to Phones by Jtapi,如下 圖所示。

5	Informa ba Provided by DEM Agri	accast® asic paging extended with Gisco	Adva (2) Buy	nced Notifica	ation >>> Learn
		Admin Broadcast Parameters Send Commands to Phones by JTAPI: Create Telephony Terminals for all Phones: Starting Multicast IP. Address: 239.0.1.2 (required) Ending Multicast IP. Address: 239.0.1.2 (required) See http://www.iana.org/assignments/multicast-addresses . Multicast TTL: 16 (required)		UPD	ATE 🖉

如果是這種情況,電話B通過SIP REFER從CUCM伺服器接收組播的IP和埠。您可以按一下SIP REFER訊息,然後按一下右鍵Message Body標頭並選擇Show Packet Bytes,如下圖所示。

Cal-10 == "79440380-d8a133a2-29151-9a34010a@30.1.61.158"				• Even		
	and a second brook before		Cal lined	- Copies		
61 15:28:04.551855 10.1.61.158 10.1 63 15:28:04.567280 10.1.61.158 10.1 68 15:28:04.768154 10.1.61.12 10.1 71 15:28:04.772605 10.1.61.158 10.1	6.1.12 SIP 1358 Request: REFER sip:e2881942-2853 781.136 31P 637 Status: 202 Actepted . .61.158 51P 624 Request: MOTIFY sip:lingi0.1.61. . .61.12 SIP 403 Status: 200 OK > .	-4eab-a0d9-96228/79d062@10.1.61.12:49340;transport=tcp (application/x-cisco-remotecc-request+xml) (application/x-cis 158:5060;transport=tcp (application/x-cisco-remotecc-response+xml) (application/x-cisco-remotecc-cm+xml)	co-remoted	20-08		
		🖉 🔏 Wireshark - Message Body (sip.msg_body) - PhoneB_capture_JTAPI.pcapng —		×		
<pre>rame 61: 1358 bytes on wire (10864 bits), 135 thermet II, scr: Wmare_0618515a (00150156) intermet Protocol Version 4, Src: 10.10.103 (resamission Control Protocol, Src Port: 500 (ession Initiation Protocol (8FFR)) Request-line: RFFR sip:e2851942-2853-4er > Message Meader > Message Body > MIDME Multipart Media Encapsulation, T</pre>	iš bytes captured (18064 bits) on interface 0 is5153), Dit Cisco_teredi99 (f8705:2014e:ed99) Dot: 10.1.61.12 Epand Subtrees Mayúsculas-Itquienda Eigand All Control-Derecha Collapse All Control-Lequienda Apply as Column Control-Mayúsculas-I Apply as Filter Prepare a Filter	uniqueBoundary Content-Type: application/x-cisco-remotecc-request+xml (x-cisco-remotecc-request> (datapassthroughreq) (applicationids/capplicationids) (displaysthroughreq) (displaysthroughreq) (displaypriority2/(displaypriority2) (displaypriority2/(displaypriority2) (displaypriority2/(displaypriority2) (displaypriority2/(displaypriority2) (displaypriority2/(displaypriority2)		^		
	Conversation Filter Colorize with Filter Follow Copy Copy	<contid=< contid="</contid=</contid=</contid=</contid=</contid=</contid=</contid=</contid=</contid=</contid=</contid=</contid=</contid=</contid=</td"></contid=<>				
	Show Packet Bytes Control+Mayúsculas+O Export Packet Bytes Control+Mayúsculas+X					
	Wiki Protocol Page	<pre><useupronexecute cuteitemux.actiones="" cuteitemux.actiones<="" cxecuteitemux.actiones="" files="" td=""><td></td><td>~</td></useupronexecute></pre>		~		
	Filter Field Reference					
	Filter Field Reference Protocol Preferences	name & Name & Name & Show as ASCII V	End 682	•		

電話收到指令後,會使用IGMP訊息加入多點傳送IP和連線埠。電話嘗試接收音訊的最大次數為三倍 。當尋呼結束時,接收方組中的電話傳送離開組消息以丟棄組播會話。

Phone8_capture_JTAPI.pcapng												
File	File Edit View Go Capture Analyze Statistics Telephony Wireless Tools Help											
4		1	0 📜		ି ବ୍	⊨ → 월 ∓ J	Ł 📜		Q. Q. 🕅			
	ional											
No.		т	ime	^	Source	Destination		Protocol	Length Info			
		66 1 157 1	5:28:04 5:28:09	4.690300	10.1.61	.12 239.0.1.2		IGMPv2 IGMPv2	60 Membership 60 Membership	Report group Report group	239.0.1.2	Join the multicast group
		320 1	5:28:18	8.960256	10.1.61	.12 239.0.1.2		IGMPv2	60 Membership	Report group	239.0.1.2	
		338 1	5:28:20	0.345639	10.1.61	.12 224.0.0.2	2	IGMPv2	60 Leave Group	p 239.0.1.2 ৰ		Leave multicast group
<												
>	> Frame 66: 60 bytes on wire (480 bits), 60 bytes captured (480 bits) on interface 0											
>	Ethe	rnet 1	II, Sro	: Cisco_	4e:ed:99 (f8:7b:20:4e:ed:9	9), Ds	t: IPv4m	cast_01:02 (01:00:	5e:00:01:02)		
>	Inte	rnet A	Protoco	l Versio	n 4, Src: :	10.1.61.12, Dst:	239.0	.1.2				
\sim	Inte	rnet (Group N	lanagement	t Protocol							
		[IGMP	Versio	n: 2]								
		Type:	Member	ship Repo	ort (0x16)							
		Max Re	sp Tim	e: 0.0 se	c (0x00)							
	Checksum: 0xf9fc [correct]											

故障排除工具

[Checksum Status: [...] Multicast Address: 239.0.1.2

<u>組播測試工</u>具將幫助您進一步排除SNMP故障。

InformaCast_LogTool將幫助您解決在網路中實施和維護InformaCast時遇到的常見問題。

高級許可證

Singlewire支援具有高級通知模式的客戶。如需其他支援,請聯絡<u>sales@singlewire.com</u>。

Sunglewire支援服務從上午7點到下午6點(CDT), 週一到週五提供+1 608.661.1140選項2。

密碼

在Informacast中,有幾種型別的密碼:

作業系統憑據:用於進入Webmin和控制中心(<u>https://x.x.x.x:10000</u>)以及使用SSH訪問 InformaCast虛擬裝置時。預設使用者為**admin**,而密碼為**changeMe**。

管理員密碼:用於登入到管理介面<u>(https://x.x.x.x8444/InformaCast/admin)。</u>

密碼短語:用於保護InformaCast虛擬裝置的備份。您必須記住此密碼短語。如果丟失 ,Singlewire支援人員無法為您恢復它。

Cisco出版書籍

對於思科呼叫伺服器12.5.1和轉發

: <u>https://www.singlewire.com/help/InformaCast/v12.5.1/advanced/cucm/index.htm#t=InformaCast</u> <u>Fusion%2FWebmin%2FRecover_the_Servers_Password.htm</u>

在Informacast中更新JTAPI

初次安裝InformaCast虛擬裝置或每次更改CUCM版本時,需要將InformaCast虛擬裝置使用的 JTAPI庫更新為CUCM伺服器使用的相同版本。 通過虛擬裝置更新JTAPI將更新所有使用JTAPI的Singlewire應用程式的JTAPI版本。

以下指南<u>https://community.cisco.com/t5/collaboration-voice-and-video/integrating-basic-cisco-paging-basic-informacast-with-cucm/ta-p/3161322的在Informacast中更新JTAPI一節中介紹了這些步驟</u>

常見缺陷

CSCve47332 Cisco IP Phone 69XX系列無法處理Application User for Informatacast中的空格

CSCuy56088 8800系列電話無組播音訊

CSCut91894 從FF37和Chrome到InformaCast的連線在FF/Chrome升級後失敗

CSCtb70375 SNMP需要提醒使用者存在DNS連線問題

相關資訊

- CUCM相容性矩陣:<u>https://www.singlewire.com/matrix/cisco-platforms</u>
- 電話矩陣: <u>https://www.singlewire.com/matrix/cisco-phones</u>
- 升級路徑: <u>https://www.singlewire.com/matrix/ic-upgrades</u>
- 伺服器平台: <u>https://www.singlewire.com/matrix/server-platforms</u>
- 硬體要求: <u>https://www.singlewire.com/informacast-hardware-requirements</u>
- 技術支援與檔案 Cisco Systems SRND:<u>https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12.pdf</u>
- CUCM與思科呼叫伺服器/InformaCast整合配置示例

: <u>https://www.cisco.com/c/en/us/support/docs/unified-communications/paging-server/117059-</u> <u>configure-informacast-00.html</u>

•思科呼叫伺服器快速入門手冊

: <u>https://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/cucm/cisco_paging_server/12_5_1/QSGInformaCastBasicPaging1251.pdf</u>