配置Informacast并排除故障

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简介

本文档介绍思科寻呼服务器产品(也称为InformaCast),以及如何将其与思科统一通信管理器 (CUCM)集成。本文档将介绍功能的用途、功能配置、要收集哪些数据进行故障排除、数据示例分 析以及相关资源,以供其他研究。

先决条件

要求

Cisco 建议您了解以下主题:

- Cisco Unified Communications Manager
- InformaCast
- SIP、CTI、Http和SNMP协议。

使用的组件

本文档中的信息基于以下软件和硬件版本:

- 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
- •基本许可证

本文档中的信息都是基于特定实验室环境中的设备编写的。用于本文的所有设备始于初始(默认)配置。如果您使用的是真实网络,请确保您已经了解所有命令的潜在影响。

背景信息

功能的用途

思科寻呼服务器是面向成千上万部电话、扬声器和其他设备的寻呼/群发通知解决方案。这在具有实 时、预录音频和/或文本通告的紧急情况下尤其有用。

根据与Singlewire(InformaCast供应商)签订的原始设备制造商(OEM)协议,思科技术支持中心 (TAC)支持从8.3版到CUCM 8.5版及更高版本的InformaCast。思科TAC支持的唯一模式是基本寻呼 。

基本与高级

基本寻呼模式仅支持每个收件人组最多50部电话的实时音频广播,无需额外许可证。作为CUCM的 一部分提供的InformaCast版本包括基本寻呼模式的许可证。需要补充功能的客户可以升级到高级通 知模式并受Singlewire支持。

高级寻呼许可证允许无限寻呼组。它还可实现其他高级功能,包括呼叫到高开销模拟和IP扬声器、 铃声调度、使用呼叫插入选项排定紧急通知的优先顺序、预录和纯文本页面、与社交媒体站点集成 以进行通知、电邮和短信服务(SMS)群发通知和全号监控、紧急服务警报,以及与Cisco Jabber客 户端集成。安装InformaCast后,您可以启用高级通知模式试用。

使用的协议

思科寻呼服务器**使用SIP、SNMP、AXL和CTI与Unified CM通**信,从思科寻呼服务器9.0.1开始 ,HTTP或JTAPI可用于与电话通信。

思科寻呼服务器使用SNMP查找其他Unified CM节点以及注册到每个集群成员的电话列表。 SNMP通信完成后,思科寻呼服务器使用AXL来确定有关每个注册电话的其他信息,如设备名称、 说明、设备池、呼叫搜索空间、目录号码和位置。此信息可用于构建电话的逻辑组,称为收件人组 。如前所述,在具有基本许可证的思科寻呼服务器中,收件人组最多可以包含50部电话。

注意:每个Unified CM集群支持一个思科寻呼服务器。

HTTP与JTAPI

9.x之前的InformaCast版本都使用HTTP进行电话激活。在HTTP模式下,思科寻呼服务器向每个 IP电话HTTP服务器发送命令和凭证。IP电话验证这些凭证,然后执行命令。在广播发送时 ,InformaCast通过HTTP直接与XML服务接口(XSI)联系。

在JTAPI模式下,思科寻呼通过Unified CM向每台电话发送命令。思科寻呼服务器不需要随每个请 求发送凭证,因此每部电话都不必激活其Web服务器,命令的执行速度也更快。此外,CTI模式允 许更快检查繁忙电话并激活它们。

无论与CUCM的集成类型(SIP或CTI)如何,您都可以使用HTTP或JTAPI。请记住,在非英语区域 设置的电话上,JTAPI比HTTP更有效。要确认用户区域设置,请查看电话网页。

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
<u>Network</u>	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)

注意:要使用JTAPI,请考虑CUCM版本必须为9.1.2或更高版本,并且不支持Cisco 3905、7902、7905、7912电话。

SIP与CTI

Informacast可以通过CTI和/或SIP接收呼叫。在CTI的情况下,呼叫在CTI路由点上进行服务(思科 寻呼服务器不需要CTI端口来应答入站呼叫)。 对于SIP,呼叫在SIP中继上离开Unified CM。CTI和SIP均有效且受支持。但是,思科建议通过 CTI进行SIP呼叫流,因为排除SIP集成故障比CTI容易得多。

配置

网络图



- 1. 主叫方(寻呼发起方)在Unified CM中拨打预定义号码。例如7777。
- 2. Unified CM通过SIP中继或CTI路由点将呼叫路由到思科寻呼服务器。
- 3. 思科寻呼服务器应答呼叫。
- 4. 主叫方听到低停音。当思科寻呼服务器播放此音时,指令通过HTTP或JTAPI发送到收件人 组中的每部电话,以加入组播组。
- 5. 所有电话加入组播组后,思科寻呼服务器将播放高提前音。当主叫方听到此音时,它表示 思科寻呼服务器已准备好接收音频并将其发送到组播IP和端口。
- 6. 当主叫方发言时,媒体从主叫方的电话发送到思科寻呼服务器,然后从寻呼服务器发送到 组播IP地址和端口,最后从组播IP发送到接收电话。

7. 当主叫方挂断时,指示将发送到每部IP电话,这次将离开组播组,广播结束。

当InformaCast使用JTAPI库和计算机电话集成(CTI)管理器与Cisco Call Manager集成时,它使用 TCP上的快速缓冲编码(QBE)协议,如图所示。



对于SIP集成,InformaCast使用TCP上的SIP协议和端口5060与Call Manager通信,如图所示。



配置Call Manager

步骤1.激活服务,导航至Cisco Unified Serviceability > Tools > Service Activation并启用以下服务:

- Cisco CallManager
- Cisco CTIManager
- Cisco AXL Web Service
- Cisco CallManager SNMP Service

提示:在所有节点上激活SNMP,在集群中至少一个节点上激活AXL,在运行Call Manager服 务的至少一个节点上激活CTI管理器(或更多节点,用于冗余)。

步骤2.配置SNMP(版本2或版本3)

对于SNMP v2

- 导航至Cisco Unified Serviceability > SNMP > v1/v2。
- 使用ReadOnly的访问权限配置社区字符串名。
- •如果可能,应用到所有节点复选框,然后单击保存。

Server* 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 Accept SNMP Packets Accept SNMP Packets Accept SNMP Packets Accept SNMP Accept Accept SNMP Accept Accept Accept Accept SNMP Accept Accept	Accept SNMP Packets only from these hosts Host IP Address Host IP Addresses Remove
Access Privileges Access Privileges* ReadOnly Notify access privilege is required in orde Apply To All Nodes	▼ er to configure Notification Destinations.

对于SNMP v3

- 导航至Cisco Unified Serviceability > SNMP > V3> User并创建名为ICVA的用户。
- 启用"需要身份验证"复选框,输入身份验证密码并选择SHA单选按钮。
- 启用"需要隐私"复选框,输入隐私密码并选择AES128单选按钮。
- 从"访问权限"下拉菜单中选择"只读",并选中"应用到所有节点"复选框(如果可能),然后单击 "保存"。

Status							
Status : Ready							
erver* 10.1.61.158CUCM Voice/Video	Ŧ						
User Information							
User Name* ICVA							
Authentication Information							
Authentication Required							
Password •••••	Reenter Password	•••••		Protocol	○ MD	5 🖲	SHA
Privacy Information							
Privacy Required							
Password •••••	Reenter Password	•••••		Protocol	O DES	; •	AES128
Host IP Addresses Information							
Accept SNMP Packets from any host	Accept SNMP	Packets only from	these hosts				
	Host IP Add	ress]		
	Insert						
	Host IP Add	resses					
			<u></u>				
			~				
	Remove						
Access Privileges							
Access Privileges* ReadOnly	Ŧ						
ONotify access privilege is required in ord	er to configure Notifica	tion Destinations.					

Appl Appl	y To All Noo	les
Save	Clear All	Cancel

步骤3.将默认编解码器设置为G.711

- 导航至CM管理>System >区域信息>区域并创建新区域,例如ICVA。
- 在Regions区域中选择所有区域,并将**64kbps(G.722、G.711)配**置为Maximum Audio Bit Rate。
- 在Max Video Call Bit Rate中选择None单选按钮,然后单击Save。

Region Configuration							Relate	d Links: Back To Find/List	
📊 Save 🗙 Delete 🧣	🎦 Reset 🥜 Apply Config 🛛	Add New							
Name* ICVA									
Region Relationships									
Region	Audio Codec Prefer	ence List	Maximum Ra	Audio Bit te	Maximum Session B Call	iit Rate for Video	Maximum	Session Bit Rate for Immersive Calls	Video
Default	Use System Default (Fact loss)	ory Default low	64 kbps G.7	(G.722, 11)	Non	e		None	
ICVA	Use System Default (Fact	ory Default low	64 kbps G.7	(G.722,	Non	•		None	
Mex	Use System Default (Fact	tory Default low	64 kbps	(G.722,	Non	e		None	
SanJose	Use System Default (Fact loss)	tory Default low	64 kbps G.7	(G.722, 11)	Non	e		None	
NOTE: Regions not displayed	Use System D	afault	Use Syste	m Default	Use System	Default		Use System Default	
Modify Relationship to o	ther Regions								
R	tegions	Audio Codec Pre	ference List	Maxim	ım Audio Bit Rate	Maximum Sessio for Video C	n Bit Rate alls	Maximum Session Bit Rate Immersive Video Calls	for
Default ICVA Mex SanJose	*	Keep Current S	etting 🔻	* 64 kbps (0	5.722, G.711) ¥	 Keep Current Use System I None 	Setting Default	 Keep Current Setting Use System Default None 	

注意:组播媒体流始终使用G.711 mu-law编解码器。不允许或支持其他编解码器。使用其他 编解码器到达Informacast的呼叫必须进行转码。

步骤4.创建设备池

- 导航至CM Administration > System > Device Pool并创建设备池。例如,将其命名为 ICVA_DP。
- 将您刚创建的ICVA区域添加到该区域。
- •从SRST Reference下拉菜单中选择Disable。
- •从"跨行**连接**"下**拉菜单中**选择"开",然后单**击"保存**"。

Device Pool Configuration							
🕞 Save 🗶 Delete 🗋 Copy	Preset ,	🖉 Apply Config 👍 Add New					
Device Pool Settings							
Device Pool Name*		ICVA_DP					
Cisco Unified Communications Ma	nager Group*	Default	~				
Calling Search Space for Auto-reg	istration	< None >	~				
Adjunct CSS		< None >	~				
Reverted Call Focus Priority		Default	~				
Intercompany Media Services Enr	olled Group	< None >	~				
- Poaming Sensitive Settings							
Date/Time Group*	CMI acal						
Region*	LCV/A						
Nedia Resource Group List	< None >		<u>의</u> 기				
Location	< None >						
Network Locale	< None >						
SRST Reference*	Disable		7				
Connection Monitor Duration***							
Single Button Barge*	Default						
Join Across Lines*	On		~				
Physical Location	< None >		×				
Device Mobility Group	< None >		~				
Wireless LAN Profile Group	< None >		View Details				

步骤5.创建路由分区,例如ICVA_PT。

步骤6.创建呼叫搜索空间,例如ICVA_CSS。包括ICVA_PT。

步骤7.创建访问控制组(AXL)。

- 导航至CM Admin > User Management > User Settings > Access Control Group并创建访问控 制组,例如ICVA User Group。
- 将标准AXL API访问角色添加到该角色。

注意:您可能已经添加了名为Standard AXL API Access的访问控制组,并添加了Standard AXL API Access角色,您也可以使用该角色。

步骤8.创建应用用户

- 导航至CM Admin > User Management > Application User, 然后点击Add New。将应用程序用 户命名为ICVA_InformaCast并分配以下角色:
- 1. Standard CTI Enabled
- 2. ICVA用户组(或标准AXL API访问)
- 3. 标准CTI允许控制支持Connected Xfer和Conf的电话。
- 4. 标准 CTI 允许控制支持回滚模式的电话
- 5. Standard CTI Allow Control of All Devices

Application User Configura	ition	
🔚 Save 🗶 Delete 🗋	Copy 🕂 Add New	
Application User Informatio	n	
User ID*	CVAInformacast	Edit Credential
Password .	•••••••••••	
Confirm Password	•••••••••••	Ξ.
Digest Credentials		Ξ́
Confirm Digest Credentials		
BLF Presence Group*	Standard Presence group ~	
User Rank*	1-Default User Rank V	
Groups ICVA User Group Standard CTI Allow Cont Standard CTI Allow Cont Standard CTI Allow Cont	trol of All Devices trol of Phones supporting C Remove from	s Control Group Access Control Group

	Standard CTI Enabled	<u>Vie</u>	<u>ew Details</u>
Roles	Standard AXL API Access	~	
	Standard CTI Allow Control of All Devices		
	Standard CTI Allow Control of Phones supporting Co	nn	
	Standard CTI Allow Control of Phones supporting Ro	lo	
	Standard CTI Enabled	\sim	View Deta

警告:每个缺陷 <u>CSCve47332</u>,建议不要为应用用户ID使用空格。

步骤9.使用SIP或CTI将Communications Manager与Informacast集成。

对于SIP集成,请创建SIP配置文件、SIP中继和路由模式。

- 导航至CM Admin> Device > Device Settings > SIP Profile,**然**后点击标**准**SIP配置文件,然后 点击Copy
- 将配置文件命名为ICVA SIP配置文件,然后选择Best Effort(不插入MTP)。单击"Save(保存)"。
- 导航至CM Admin > Device > Trunk, 然后点击Add New
- •从中继**类型下**拉菜单中选择SIP中继。单击Next并输入SIP中继的名称。
- •选择设备池ICVA_DP,向下滚动到SIP Information区域,并在Destination Address中输入 InformaCast服务器的IP**地址**
- •确保Destination Port字段中的值为5060,选择**Non Secure SIP Trunk Profile**,并从SIP Profile下拉菜单中分配之前创建的SIP配置文件。单击"**Save**(保**存**)"。

Device Information		
Product:	SIP Trunk	
Device Protocol:	SIP None(Default)	
Device Name*	ICI/A SinTruck	
Description		
Device Beel*	10.1.61.118	
Device Pool	ICVA_DP	
Common Device Configuration	< None >	~
Call Classification*	Use System Default	~
Media Resource Group List	< None >	~
Location*	Hub_None	~
AAR Group	< None >	~
Tunneled Protocol*	None	~
QSIG Variant*	No Changes	~
ASN.1 ROSE OID Encoding*	No Changes	~
Packet Capture Mode*	None	~
Packet Capture Duration	0	
	1	

Destination			
Destination Address is an SRV			
Destination Add	ress	Destination Address IPv6	Destination Port
1* 10.1.61.118			5060
MTP Preferred Originating Codec*	711ulaw	~	
BLF Presence Group*	Standard Presence group	~	
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile	~	
Rerouting Calling Search Space	< None >	~	
Out-Of-Dialog Refer Calling Search Space	< None >	~	
SUBSCRIBE Calling Search Space	< None >	~	
SIP Profile*	ICVA SIP Profile	View Details	
DTMF Signaling Method*	No Preference	~	

- 创建路由模式,导航至CM Admin > Call Routing > Route Hunt > Route pattern,然后点击Add New。
- 输入路由模式(例如7777),并配置可从电话(例如ICVA_PT)访问的分区。
- •从Gateway/Route List下拉菜单中选择您刚创建的SIP中继。

•选择Route This Pattern和OnNet单选按钮。

• 取消选中Provide Outside Dial Tone复选框,然后单击Save。

对于CTI集成,请创建CTI路由点并关联到步骤8中创建的应用用户。

- 导航至CM Administration > Device > CTI Route Point,然后点击Add new。
- 输入名称,例如ICVA_CTI_RP(或您喜欢的任何名称)。
- 分配设备池ICVA_DP,然后单击Save。
- •选择行1,输入目录编号(例如7778),并分配最近创建的分区(ICVA_PT)。
- 根据需要配置其余信息,然后单击"Save"。

在ICVA应用用户配置中将CTI路由点添加为受控设备。

Controlled Devices	ICVA_CTI_RP	^
		~

注意:如果InformaCast是在Communications Manager中创建并与InformaCast应用用户关联 的,则它可以支持多个CTI路由点。

提示:您也可以将多条线路添加到单个CTI路由点,而不是为拨号广播所需的每个号码创建 CTI路由点。另一种选择是使用通配符模式匹配一系列数字。

步骤10.启用Web Access for Cisco IP电话以使用HTTP控制电话。

- 在企业电话配置中,可以按设备、通用设备配置文件或系统范围配置Web访问。
- 要应用企业电话配置中的更改,请导航至CM Admin > System > Enterprise Phone Configuration,向下滚动到Web Access下拉菜单并选择Enabled。单击"Save(保存)"。
 重置电话以应用更改。

Enterprise Phone Configuration			
Save			
Web Access*	Enabled	~	

步骤11.设置身份验证URL。

更改身份验证URL,以便将身份验证请求从IP电话发送到InformaCast。所有非InformaCast身份验证请求都重定向回默认CUCM身份验证URL。

- 导航至CM Administration > System > Enterprise parameters。
- 在URL Authentication (URL身份验证)字段和Secure Authentication URL (安全身份验证 URL)中输入http://<InformaCast Virtual Appliance IP Address>:8081/InformaCast/phone/auth。
- 单击Save、Apply config和Reset the phones。

sterprise Parameters Configuration	
🚽 Save 🤣 Setto Default 🌯 Reset 🥒 Apply Config	
Phone URL Parameters	
URL Authentication	http://10.1.61.118:0001/5nformaCast/phone/auth
JRL Directories	http://10.1.61.158:8080/comcip/emidirectory.jsp
RL Ide	
Rs. Idle Time	0
Ri. Information	http://10.1.01.158.0080/comcip/GetTelecasterHeipText.js
RL Messages	
P. Phone Proxy Address	
JRL Services	http://10.1.61.158:8080/comcip/getservicesmenu.jsp
Secure Phone URI. Parameters	
ecure Authentication URL	http://10.1.61.118:8081/InformaCast/phone/auth
ecure Directory URL DPAL)	https://10.1.41.158.8443/condp/unideedbry.jsp
ecure Contact Search URL (UDS)	https://10.1.41.158:8443/cuom-uds/users
ieare Ide UIL	
Secure Information URL	https://10.1.61.158.8443/comcip/GetTelecasterHelpText.)
iecure Messages URL	
Secure Services URL	https://10.1.61.158:8443/comdp/petserv/cesmenu.jsp

注意:URL区分大小写,因此请确保InformaCast一词中的I和C大写。安全身份验证URL和身 份验证URL必须设置为相同的值,即HTTP URL。

步骤12.设置API浏览器访问的身份验证方法。

 如果您使用Unified Communications Manager 11.5.1及更高版本,请向下滚动页面到Security Parameters区域,然后从Authentication Method for API Browser Access下拉菜单中选择 Basic。

步骤13.测试您的电话,例如拨号7777(用于SIP集成)或7778(用于CTI集成)。

注意:如果您以混合模式运行Unified Communications Manager,请确保InformaCast的呼叫和来自InformaCast的呼叫不使用加密介质。

配置Informacast

步骤1.在Informacast中配置Communications Manager集群。

- 登录Informacast并导航至Admin > Telephony > Unified Communications Manager Cluster。单击"编辑"。
- 输入您在第8步中创建的应用用户的用户名和密码。
- 确保选中Use Application User for AXL复选框,这表示在构建InformaCast的电话缓存时使用您的应用程序用户凭据。

注意:如果将此字段留空,InformaCast将尝试在运行CallManager服务的服务器中查找运行 AXL服务的服务器。

- 在Communications Manager IP Address(es)字段中输入Unified Communications Manager服务
 器的IP地址。使用数字IP地址而不是DNS名称。
- •选择SNMP v2或SNMP v3单选按钮。输入在CUCM中配置的相同信息。单击"Update(更新)"。

Ad Co Solution Solution	Imin Telephony (Infiguration Unfied Communications Ma Unfied Communications Mana Cont	Cisco Unified Cor nager Cluster Description Vanager Application Diser, per Application Passevort, mit Application Passevort, mit Application Passevort,	mmunication elephony Config CUCM (CVPMtormacas)	s Mana	(required)	ster E	g Ovi Appi	ication Adm	in stato
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		where we are an and the second							
			Use Applicatio	n User for A	AVG.				
		All IP Address(es)	10.1.61.158						
	Unified Communications	Manager IP, Address(es),	10 1 61 158		(required)				
		Choose SNMP version	● SNMP v2 ○ SNMP v3 (req	sired)					
	50.	MP v2 Community Name							
	Confirm SN	MP v2 Community Name.							
XM	IL Push Authentication	tivate phones during broad	scants or if this is n	ot your prin	nary cluster	make sur	e the URL	Authentical	tion
Par	ameter for the Unified Comm nameters page) is set to the	nunications Manager in thi following value:	is cluster (found in t	the Phone	URL Parar	neters per	tion of the	System En	desprise
		http://10.1.6	1.118:0001/164	OFFICART.	/phone/a	4.54			
Opt nee suc	tionally, you can also tell inf ed to do this if, before install ch cases, copy the current U	ormaCast where to send a ing InformaCast, you had s Inified Communications M	uthentication reque set this Unified Con lanager setting into	sts for com munication the field be	vmands that ns Manager ilow, before	aren't con parameter changing it	ting from in to a non si to the valu	formaCast. tandard value e shown ab	You only a. In ove.
		Next Authentication URL:							
if er aut	ngty, non-informaCast auth hentication page, http://	entication requests from pl 10.1.41.159/ccmcsp/	hones in this cluste authenticate.;	r will be se inp	nt to the de	Gault Unifie	d Communi	cations Mar	nager
		Course T			ALL PA				

步骤2.配置收件人组。

• 导航至**Recipients > Edit Recipient Groups**,然后单击**Update**以显示在CUCM中注册并由 InformaCast发现的所有电话。

Inform	aCast ^a			(ation 22) Livere		Recorder	E Receptor	C) Spendarers	ere ere	****	O Plagan	(?) 1440
8	Recipients	Edit Recip scover current nsuming). now Defunct Pr	ient Gro IP phone nones	oups Reci informat	ipient gro	oup memb Cisco Un	ers updat	ted municatio	ns Mana	<u>per (may </u>	leation Adm be time	nini strator
										0-10		
	A Name								Phones	Action		
	(All Recipients)								2	?)	ant O (a	an 😒

• 要创建新的收件人组,请单击Add,写下名称,然后单击Edit,为此收件人组添加电话。将电话 添加到收件人后,单击Submit。

riect	Individual Recipients			
ilter valla	citar able Recipients (double click to select)		Sales	cted Recipients (double click to remove)
	Descriptive Text	1		Descriptive Text
•	Cisco IP Phone: Auto 111; DNs: 111; SEPF87B204EED99		٠	Cisco IP Phone: Auto 110; DNs: 110; SEP2C3124C9F8E1
		Add		
		Bamova		
	ni e anno 11 anni e distance		-	
	a sprav i mou z mou zz			_
				Submit Can

• 要保存更改,请单击"更**新"**。

Inform	aCast ^o	Advanced No Control of Control o	ification 2000 Learn	feet a	Nessages	Bocqueents	Speakers	2	8 	O Program	() 140
	Recipients Ec	dit Recipient C	Groups E	Edit Rec	sipient C	Sroup		to	g Our Appi	ication Adr	inistrator
<u> </u>	Nerre Mex Tegs MEx			(requir (Add /	ed) «Tag –						
	Select Recipients										
	Cisco IP Phone: A	luto 110; DNis: 110; S	EP2C312NC9	FBE1							
	Elter with Recipient Elter with Rules Exclusions (m) Exclusions are only	available when the F	ecipient Grou	g is Filtere	d by Recipie	ent Groups (or Rules.				
			and the second	Cases	D (111						

步骤3.允许/禁用对InformaCast的SIP访问。

- 导航至Admin > SIP > SIP Access。默认情况下,所有SIP呼叫都被拒绝。
- 选择"允许"单选按钮允许所有SIP呼叫,或单击"添加"以允许此允许的例外。

Inform	aCast [®]	Adver CO Day		alice 200 Loarn	forme	Ressager	83 1 Hore points	Speakers	Se .	e. Anna	Pagers	? 1489
а <mark>а</mark> с	Admin SIP S	SIP Acce	oss c	ontrols ac	cess of ini	bound SIP	calls to Inf	ormaCast	Li	ιg Out Αρρ	lication Adr	ninistrator
	Click to restore to default settings when O Deny incoming SP calls											
				GANDR	o	a host ex	ception (ana					

提示:在定义例外时,请确保指定直接将INVITE请求发送到InformaCast的主机。如果代理位于InformaCast和主叫主机之间,则此服务器可能是SIP代理服务器。

步骤4.添加广播拨号配置

• 转到Admin > DialCast > Dialing Configurations,单击Add

- 根据在CUCM中创建的路由模式(用于SIP集成)或CTI路由点(用于CTI集成),在拨号模式
 字段中输入拨号模式(例如7777、7778)。
- •从列表中选择收件人组,然后单击"更新"。

Informa	aCast ^e Isic paging		٩	ation 20 Learn	() 1 m	Recessor	E Recuents	() Speedners	S) Defis	e Admin	() Puges	? Help
â.	Admin DialCas	st I Dial	ing Cor	figurati	ons I F	dit Broa	dcast D)ialing (Configu	ration	lication Adr	nini strator
103	Plantin Diaroa	n bride	ing con	Daing	Pattern:	7777	FURDINE D		soringe			
				Broadca	st Recip	ients						
				Recipient	Groups;	(All Recip Mex SanJose	pients) ~					
							9					
				Cancel.			101					

步骤5.配置广播参数。

- 导航至Admin > Broadcast Parameters
- 配置组播的IP。通常使用默认IP(239.0.1.2)。
- 如果要将命令以JTAPI的形式发送到电话,请启用JTAPI复选框,否则将使用HTTP消息。

Inform	aCast [®]	0		2000 2000	0	- Carlos	8	0	2	2 3	0	2
									L	ng Out App	Ecution Ad	nestato
â	Admin Broad	icast Par	rameters	s								
0	Send Command	s to Phones I	by JTAPI: 12	3								
1	Create Tele	phony Termin	phones	3								
	Stating	Multicest P.	Address 2	29012		(requi	(bar					
	Ending	Multicast IP	Address: 2	29.0.1.2		(redry	rød)					
			8	ee <u>shttp://w</u>	wiana.	org/assignme	ents/multic	ast address	852.			
		Multi	cast TTL; 1	6 (req	(berku							
				Canoli I	3		843	ui 🕘				

确保此范围与您的网络基础设施设置相对应,并覆盖所有收件人组。在多站点部署中 ,Singlewire和Cisco建议使用一系列地址。此范围应足够大,以便处理每个同步广播的一个地址。

注意:建议在HTTP上使用JTAPI,因为它可以更好地监控电话的状态,并适用于更多区域设置。

提示:Web界面的默认设置将在五分钟后注销。**导航至Admin > Network Parameters >** Session Timeouts,并将General Session Timeout(seconds)字段从300更改为新值。

在网络中配置组播

如果思科寻呼服务器和IP电话位于不同的IP子网上,则必须为组播路由配置这两个子网之间的路由 器。

思科寻呼服务器不需要任何特定的组播路由方法(SM、DM、S-DM、SSM等)。 某些广域网环境 不支持组播路由。对于这些环境,GRE隧道可以在站点之间构建,并用于传输组播。 在您的环境中设计和配置组播不在本文档的讨论范围之内,但您可能会发现以下资源很有帮助:

•<u>组播白皮书</u>

<u>组播测试工具</u>

注意:如果您使用Meraki交换机,则默认情况下它们启用IGMP监听。这可能会导致问题,需 要由Meraki禁用。联系他们并让他们禁用IGMP监听后,请再次测试寻呼。

验证

当前没有可用于此配置的验证过程。

故障排除

常见问题

电话未激活

考虑到广播发生时,Informacast会跳过所有正在使用(忙)的电话。

InformaCast使用不同的忙检测方法,具体取决于您向电话(HTTP或JTAPI)发送消息的方式。

HTTP:忙线检测仅适用于运行英语负载的电话区域设置

CTI:与非英语电话区域设置配合使用

根据协议以及线路类型和线路状态,忙线检测的工作方式也不同。

线路状态	CTI忙检测	HTTP忙检测
在另一部电话上使用呼叫的共享线路,无 呼叫处于保留状态	空闲	空闲
摘机,收集数字	忙碌	不忙
通话、活动呼叫	忙碌	忙碌
保留,共享线路上的非活动呼叫	忙碌	不忙
处于保留状态,在唯一线路上处于非活动	忙碍	不忙
状态的呼叫	I∟ ₩¥K	111

注意:如果尝试同时广播,Informacast会先播放第一个广播(第二个广播会被颠覆)。

排除未激活的电话故障时,应收集以下数据:

•从Informacast发送性能日志。

•从电话中记录控制台日志(PRT)。

未发现电话

InformaCast只发现注册电话。如果IP电话已注册但未发现,请检查Informacast和电话注册到的 CUCM节点中的SNMP服务配置。应为激活Call Manager服务的所有节点配置SNMP服务和社区字 符串。

SNMP错误无法生成收件人组: java.lang.exception

Recipients Edit Recipient Groups	
Recipient group members updated	
Unable to build inclusion groups: java.lang.Exception: Unable to obtain phone information via 58 cluster. Make sure the SMMP service is started on all servers in the cluster and that the provided [Show details]	MP from all servers in the 'CUCM' SNMP community name is correct.
PINE Discover current IP phone information from Cisco Unified Communications Manage	r (may be time consuming).
(monaul) Show Defunct Phones	
Process @ Page 1 of 1 (617 @ Jump to page (61 @ Show 50 results per page 178e; ()	(ALL C)
A Name	Phones Action
(All Recigients)	3 (101 🔽 (101 🖉 101 🔽
Mex	1 002 000 000
SanJose	1
PERCENT OF Page 1 of 1 (1011 OF Avery Surgage (101 OF Street SO Interests per page	

- 1. 该错误意味着SNMP由于DNS连接或解析而无法及时响应查询。
- 2. 确认没有阻止从InformaCast服务器到所有Unified Communications Manager集群节点的 UDP端口161。
- 3. 确认SNMP信息正确。导航至Admin > Telephony > Unified Communications Manager Cluster,并键入新的SNMP字符串(如果可能)。在CUCM中配置新字符串。
- 4. 您也可能使用的社区字符串超过了社区字符串的最大字符数。如果从CUCM复制社区字符串并 将其粘贴到Informacast配置中,请尝试键入该字符串以查看是否可以键入整个字符串。在 Informacast版本11中,最大字符数为18。
- 5. 检查CUCM上的DNS配置是否正确,并确认您与缺陷CSCtb70375不匹配。

目标电话上没有音频

如果电话亮起但不播放音频,则问题很可能与组播路由有关,而与CUCM服务器或IP电话无关。

要收集的数据

排除Informacast故障时,应收集以下数据:

- 1. 从Informacast发送性能日志。
- 2. 从Informacast捕获数据包。
- 3. 从电话捕获数据包。
- 4. 从CUCM捕获数据包。
- 5. 来自CUCM的SDL日志
- 6. PRT(控制台日志)

性能日志

从Informacast获取性能日志有两种方法。

方法 1

1. 导航至https://<Informacast IP>:8444/InformaCast/logs/performance.log

方法 2

- 1. 在Web浏览器中打开Informacast IP, https://<informacast_IP>并选择Informacast。
- 2. 使用您的凭证登录并导航至"帮助">**"支持"**。



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

3.单击"工**具"部**分下的"性能日志",如图所示。

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.

数据包捕获

从Informacast

从Informacast获取数据包捕获有三种方法。

方法 1

- 1. 通过SSH连接到Informacast框的CLI
- 2. 执行命令sudo capturePackets test.cap以开始捕获并创建名为test.cap的文件
- 3. 向不工作的电话分页
- 4. 按Ctrl + C结束pcap
- 5. 执行ls以确保数据包捕获在机箱中
- 6. 使用SFTP或安全复制(SCP)将文件传输到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

0

- 1. 从Web下<u>载并安</u>装InformaCast_LogTool。
- 执行软件并选择选项[5]。写下Informacast的IP、登录凭证和数据包捕获应运行的秒数,如图 所示。

Administrator: Singlewire Software: InformaCast Log Tool - 20150707	<u>_0×</u>
Singlewire Software	-
	_
[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]: _	•

 捕获不会立即开始,这样您就可以准备测试环境。 准备就绪后,选择选项[1]并按Enter开始捕 获数据包,如图所示。

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

- 4. 该工具将显示一个倒计时计时器,其中显示捕获的未完成持续时间。在此时间复制问题,当捕获倒计时达到零时,捕获完成并停止。
- 5. 该工具将数据包捕获和所有日志捆绑到.tgz文件中,并将其传输到工作站。 这与用于收集日志 的选项1相同,但也包括网络流量捕获。
- 6. 该工具将在Informacast_LogTool.exe的基目录中创建一个包含数据包捕获的文件夹,如图所示

InformaCast_LogTool.exe
InformaCast_LogTool_Logs_201809231605.tgz

方法3(在版本12.0.1及更高版本中提供)

- 1. 登录<Informacast_IP>:10000
- 2. 导航至System > Capture Network Traffic。



3. 单击"Start a new packet capture(开始新数据包捕获)",然后复制问题,如图所示。



- 4. 当问题被完全复制时,单击Stop Packet Capture(停止数据包捕获),或者在捕获33,000个数据包后单击Stop Packet Capture(停止数据包捕获)。
- 5. 导航至System > Collect Logs, 输入问题的简短说明, 然后单击Collect a new set logs。

6. 要保存日志	5,请单击"Download to Your Computer (下载到您的计 算机)" ,如图所示
Legix admin	Module Config Collect Logs
** System Bootup and Stutdown Capture Network Traffic	Collect a New Set of Logs for a Problem Report
Change Passwords Collect Legs Disk and Network Filesystems	Collect New Log Set
Running Processes System Logs Upgrade or Switch Versions	Singlewire support contract number, if known
> Netvoling > Hadvare	Do not automatically send the log collection to Singlewire Support
Search	Log Actions
(B Lageut	The log collection from 2016-04-07-1923 Send to Singlevire Support must be downloaded and sent to Singlevire Support Deveload to Year Computer

方法4(在版本12.0.1及更高版本中提供)

在版本12.0.1及更高版本中,不再需要sudo命令。要运行数据包捕获,请使用**capture-packets <文** 件名> <数据包数>命令,如示例所示:

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 **注意:**GUI方法优于CLI,因为它不依赖SFTP服务器,而且您可以从网页启动、停止和下载数 据包捕获。

Жсисм

根据部署定义从何处获取数据包捕获。集群中只能有一个CUCM节点或多个CUCM。

• 如果您有一个CUCM节点,请获取数据包捕获,如图所示。



- 1. 为需要捕获的节点打开SSH会话
- 2. 运行命令utils network capture eth0 size all count 1000000 file Test以启动数据包捕获。
- 3. 复制问题
- 4. 使用Ctrl + C停止数据包捕获
- 5. 要确认数据包捕获已保存,请运行命令file list activelog platform/cli/*

P	>	ĸ		
admin: admin:utils network capture Executing command with opti	eth0 size all count ons:	1000000 file Test		^
size=ALL co src= de ip=	unt=100000 st=	interface=eth0 port=		
Control-C pressed				
admin:file list activelog p	latform/cli/*			
Test.cap				
admin:	T			~

6. 使用命令file get activelog platform/cli/Test.cap将数据包捕获发送到SFTP服务器。或者,要收集存储在服务器上的所有.cap文件,请使用file get activelog



7. 如果无法使用SFTP服务器,请使用RTMT。导航**至System > Trace & Log Central > Collect** Files。单击"下**一步"**并启用"**数据包捕获日**志"复选框,如图所示。

Collect Files	_						
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							
IPT Platform CLI Created Reports							
IPT Platform CLI Logs							
IPT Platform Cert Monitor Logs							
IPT Platform CertMgr Logs							
IPT Platform Cluster Manager Logs							
IPT Platform GUI Logs							
IPT Platform IPSecMgmt Logs							
IPT Platform RemoteSupport Logs							
Install File Signing							
Install and Upgrade Logs							
MIB2 Agent							
Mail Logs							
Mgetty Logs							
NTP Logs							
Netdump Logs							
Packet Capture Logs							
Prog Logs							
SAR Logs							
SNMP Master Agent							
Security Logs							
Service Manager							
Spooler Logs							
System Application Agent							
< Back Next	> Finish (Cancel					

8. 单击"Next(下一步)",选择下载文件目录,然后单击"Finish (完成)"。

9. 使用命令file delete activelog platform/cli/Test.cap删除数据包

从电话

- 1. 激活SPAN到PC端口。导航至**CM Admin page > Device > phone**,并查找报告有问题的电话 。
- 2. 在"产品特定配置布局"部**分下,找到**Span to PC Port,然后从下拉**菜**单中选择"启用"。单击"保 存",然后单击"应用配置"。
- 3. 将笔记本电脑连接到电话的PC端口。
- 4. 在笔记本电脑中运行数据包分析器软件。您可以使用Wireshark(或其他数据包捕获软件)。
- 5. 复制问题。
- 6. 当问题完全复制时,继续停止数据包捕获。
- 您可以在以下链接中找到更多详细信息
- : <u>https://supportforums.cisco.com/document/44741/collecting-packet-capture-cisco-ip-phone</u>

示例分析



SDL跟踪

用于SIP集成和由JTAPI控制的电话

CUCM:10.1.61.158

Informacast:10.1.61.118

电话A

DN:110

型号:CP-8861

固件版本:sip8xx.12-0-1SR1-1

电话A的IP地址:10.1.61.12

MAC SEP2C3124C9F8E1

电话B

DN:111

型号:CP-8811

固件版本:sip8xx.12-0-1SR1-1

电话B的IP地址:10.1.61.11

MAC SEPF87B204EED99

拨号播号: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

用于CTI集成和由HTTP控制的电话

CUCM:10.1.61.158

Informacast:10.1.61.118

电话A

DN:110

型号:CP-8861

固件版本:sip8xx.12-0-1SR1-1

电话A的IP地址:10.1.61.12

MAC :SEP2C3124C9F8E1

电话B

DN:111

型号:CP-8811

固件版本:sip8xx.12-0-1SR1-1

电话B的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=0si1 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.

性能日志

用于SIP集成

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

控制台日志(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

数据包捕获

从电话收集数据包捕获并检验InformaCast的HTTP XSI命令。发送互联网组管理协议(IGMP)消息以 加入组播流。如果在IGMP消息后未看到组播实时传输协议(RTP)流,则可以从InformaCast捕获数 据包,确认Informacast服务器已将RTP发送到IP和端口,然后检查网络基础设施。

电话上的数据包捕获(由HTTP控制)

- CUCM:10.1.61.158
- Informacast:10.1.61.118
- 电话B的IP地址: 10.1.61.11
- 型号: CP-8811
- 固件版本: sip8xx.12-0-1SR1-1
- eth.addr==SEPF87B204EED99

电话上收到的HTTP和IGMP消息显示在图像中。

File	e Edit View Go Capture A	Analyze Statistics Teleph	ony Wireless Tools Help)				
4	II 2 0 I G X C 9	🕻 🗰 🗰 🐺 🛓 🛄	📕 Q, Q, Q, 🎹					
	http][igmp						8 =	Expressi
No.	Time	Source	Destination	Protocol	Length Info			
	1771 00:24:39.352999	10.1.61.22	173.36.89.68	HTTP	2295 CCM POST /ccm system/request HTTP/1.1 (text/plain)			
	1777 00:24:39,404529	173.36.89.68	10.1.61.22	HTTP	394 HTTP/1.1 200 OK (text/plain)		Informacest conds ID and port for multicast	
	1905 00:24:49.392163	10.1.61.118	10.1.61.11	HTTP	223 GET /StreamingStatisticsX?1 HTTP/1.1		informacast sends iP and port for multicast	
	1911 00:24:49.444329	10.1.61.11	10.1.61.118	HTTP/XML	1452 HTTP/1.1 200 OK		where the set of the s	
+	1917 00:24:49.453245	10.1.61.118	10.1.61.11	HTTP	399 POST /CGI/Execute HTTP/1.1 (application/x-www-form-uplencoded))	The phone authenticates with Informacast	
	1922 00:24:49.479784	10.1.61.11	10.1.61.118	HTTP	457 GET /InformaCast/phone/auth?UserID=ICVAInformacast&Password=rtg	pavvidð	devicename=SEPF878204EED99 HTTP/1.1	_
	1926 00:24:49.483773	10.1.61.118	10.1.61.11	HTTP	76 HTTP/1.1 200 OK (text/html) <		Informacast replies with 200 OK	
	1932 00:24:49.610049	10.1.61.11	239.0.1.2	IGMPv2	60 Membership Report group 239.0.1.2		informacastrepiles with 200 OK	
	1941 00:24:49.735551	10.1.61.11	10.1.61.118	HTTP/XML	474 HTTP/1.1 200 OK			-
	1965 00:24:50.999480	10.1.61.11	239.0.1.2	IGMPv2	60 Membership Report group 239.0.1.2	The p	hone joins to the Membership (IP and port)	
	2070 00:24:58.399886	10.1.61.11	239.0.1.2	IGMPv2	60 Membership Report group 239.0.1.2	_		-
	2512 00:25:30.985190	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 00:25:31.228042	10.1.61.11	224.0.0.2	IGMPv2	60 Leave Group 239.0.1.2	L		
	2518 00:25:31.234468	10.1.61.11	10.1.61.118	HTTP/XML	462 HTTP/1.1 200 OK		The share have the LCLUD server	
1							The phone leaves the IGMP group	>
~						_		
~	Hypertext Transfer Protoc	col						
	> POST /CGI/Execute HTTP	P/1.1\r\n						
	> Authorization: Basic 5	SUNWQU1uZm9ybWFjYXN0	OnJ0cGF2dm1k\r\n					
	User-Agent: Jakarta Co	ommons-HttpClient/3.	1\r\n					
	Host: 10.1.61.11\r\n							
	> Content-Length: 116\r\	\n						
	Content-Type: applicat	tion/x-www-form-urle	ncoded\r\n					
	\r\n							
	[Full request URI: htt	tp://10.1.61.11/CGI/	Execute					
	[HIP request 1/1]							
	Insponse in frame: 1941]							
	File Data: 116 bytes		e unlanded.					
ľ	Free iters "We - "(Circation/X-MM-top	-urtencoded	···· 220 0 1 2	20406" / \ / (/			
	> FORM ITEM: "XML = - CC	LISCOIPPHONEEXecute>	cexecuteitem OKF=KIN	MX:239.0.1.2	20480 / >			

电话上的数据包捕获(由JTAPI控制)

- CUCM:10.1.61.158
- Informacast:10.1.61.118

- •电话B的IP地址:10.1.61.11
- 型号: CP-8811
- 固件版本: sip8xx.12-0-1SR1-1
- MAC SEPF87B204EED99

如配置部分所述,电话可以由JTAPI控制,这意味着**Send Commands to Phones by Jtapi已启**用 ,如图所示。

6	Informa ba Provided by GEM Agre	asic paging sement with Gaco	Adva (2) Buy	nced Notific	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 🖉

如果是这样,电话B通过SIP REFER从CUCM服务器接收组播的IP和端口。您可以单击SIP REFER**消息**,然后右键单击Message Body报头并选择**Show Packet Bytes**,如图所示。

PhoneB_capture_JTAPI.pcapng Edit View Go Capture Applyce Statistics Tele	ohony Wester Toole Melo		-	- 0
p.Cal-1D == "79d40280-d8a133a2-29151-9e3d010a@10.1.61.158"				* Egress
Time Source Detir	ation Protocol Length Info			
61 15:28:04.551855 10.1.61.158 10.1	.61.12 SIP 1358 Request: REFER sip:e2881942-	2853-4eab-a0d9-96228c79d062@10.1.61.12:49340:transport=tcp (application/x-cisco-remotecc-request+xml) (application/x-	cisco-remoter	cc-cm
63 15:28:04.567280 10.1.01.12 10.1	.01.130 SIP 037 Status: 202 Accepted			-308
68 15:28:04.768154 10.1.61.12 10.1	.61.158 SIP 624 Request: NOTIFY sip:111@10.1	.61.158:5060;transport=tcp (application/x-cisco-remotecc-response+xml) (application/x-cisco-remotecc-cm+xml)		
71 15:28:04.772605 10.1.61.158 10.1	.61.12 SIP 403 Status: 200 OK			
		Vireshark - Message Body (sip.msg_body) - Phone8_capture_JTAPI.pcapng -	- 0	×
rame 61: 1358 bytes on wire (18864 bits), 135	8 bytes captured (10864 bits) on interface 0			
thernet II, Src: Vmware_96:a5:5a (00:50:56:96	ia5:5a), Dst: Cisco_4e:ed:99 (f8:7b:20:4e:ed:99)	uniqueBoundary		^
sternet Protocol Version 4, Src: 10.1.61.158.	Dst: 10.1.61.12	Content-Type: application/x-cisco-remotecc-request+xml		
ansmission Control Protocol, Src Port: 5004	expand subtrees mayusculas+ Derecha			
Request-Line: REFER sip:e2881942-2853-4ea	Collapse Subtrees Mayúsculas+Izquierda IP	/2.0 <x-cisco-remotecc-request></x-cisco-remotecc-request>		
Nessage Header	Expand All Control+Derecha	(detanasthroughpan)		
Nessage Body	Collapse All Control+Izquierda	<applicationid>@</applicationid>		6 B
> MIME Multipart Media Encapsulation, T	Analysis Column Control Manfronders I	(lineid)		
	Apply as Column Control+ Mayusculas+1	<pre><transactionio>aoz/(transactionio> </transactionio></pre> <pre></pre> <pre><!--</td--><td></td><td></td></pre>		
	Apply as Filter	<displaypriority>2</displaypriority>		
	Prepare a Filter	<appinstance>%/appinstance></appinstance>		
	Conversation Filter	<pre>cconfid>0</pre>		
	Coloring with Filter	<featuredata></featuredata>		
	Contract With Printer	 vietebessiin orden ads. 		
	Follow			
	Copy			
	Show Packet Bytes Control+Mavúsculas+O	uniqueBoundary		
	Evont Packet Putes Controla Maujsculasa X	Content-Type: application/x-cisco-remotecc-cm+xml		
	control of the second of the s	<ciscoipphoneexecute><executeitem url="RTPMRx:239.0.1.2:20488"></executeitem>:/CiscoIPPhoneExecute></ciscoipphoneexecute>		
	Wiki Protocol Page			
	Filter Field Reference	**uniqueboundary**		~
	Protocol Preferences	Frame 62, Mossage Body (siguring, body), 682 byte(s).		
		Decode as None V Show as ASCII V Start 0	C End 682	\$
	Decode As	54	-	
0d 0a 0d 0a 2d 2d 75 6e 69 71	Go to Linked Packet	Pres	Find Next	
64 61 72 79 0d 0a 43 6f 6e 74	Show Linked Packet in New Window	Print Copy Save as Cerrar	Ayuda	
70 65 3a 20 61 70 70 6c 69 63 c	1 /4 09 0T 0P /T			

电话收到指令后,会使用IGMP消息加入组播IP和端口。电话尝试的接收音频次数是最大尝试次数的 三倍。当寻呼结束时,收件人组中的电话发送离开组消息以丢弃组播会话。

	PhoneB_capture_JTAPI.pcapng								
File	le Edit View Go Capture Analyze Statistics Telephony Wireless Tools Help								
4	🔳 🖉 💿 📜 🗎 🗙 🏹	। 🤇 🖛 🏓 🖀 🗿 💆	2 📃 📃 Q, Q, Q, 🎹						
	lampl								
No.	Time S	iource Destination	Protocol Length Info						
	66 15:28:04.690300 10 157 15:28:09.140169 10 320 15:28:18.960256 10 338 15:28:20.345639 10	0.1.61.12 239.0.1.2 0.1.61.12 239.0.1.2 0.1.61.12 239.0.1.2 0.1.61.12 239.0.1.2 0.1.61.12 224.0.0.2	IGMPv2 60 Membership Report group 239.0.1.2 Join the multicast group IGMPv2 60 Membership Report group 239.0.1.2 Join the multicast group IGMPv2 60 Membership Report group 239.0.1.2 Leave multicast group IGMPv2 60 Leave Group 239.0.1.2 Leave multicast group	, 					
<									
>>>>	Frame 66: 60 bytes on wire (480 bits), 60 bytes captured (480 bits) on interface 0 Ethernet II, Src: Cisco_4e:ed:99 (f8:7b:20:4e:ed:99), Dst: IPv4mcast_01:02 (01:00:5e:00:01:02) Internet Protocol Version 4, Src: 10.1.61.12, Dst: 239.0.1.2 Internet Group Management Protocol								
	[IGNP Version: 2] Type: Membership Report (0x16) Nax Resp Time: 0.0 sec (0x00) Checksum: 0xf9fc [correct]								

故障排除工具

[Checksum Status: food] Multicast Address: 239.0.1.2

组播测试工具将帮助您进一步排除SNMP故障。

InformaCast_LogTool将帮助您排除在网络中实施和维护InformaCast时遇到的常见问题。

高级许可证

Singlewire支持具有高级通知模式的客户。请联系<u>sales@singlewire.com以获</u>得更多支持。

Sunglewire支持服务于CDT(星期一至星期五)上午7:00至下午6:00(电话:+1 608.661.1140选项 2)。

密码

在Informacast中,有几种密码类型:

操作系统凭证:用于进入Webmin和控制中心(<u>https://x.x.x.x:10000</u>),以及使用SSH访问 InformaCast虚拟设备时。默认用户为**admin,**而密码为**changeMe**。

Admin 密码:用于登录管理员界面(<u>https://x.x.x.x:8444/InformaCast/admin)。</u>

密码:用于保护InformaCast虚拟设备的备份。您必须记住此密码。如果丢失,Singlewire支持人员 无法为您恢复。

密码 恢复

对于思科寻呼服务器12.5.1并转发

: <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

最初安装InformaCast虚拟设备时,或每次更改CUCM版本时,需要将InformaCast虚拟设备使用的

JTAPI库更新为CUCM服务器使用的相同版本。

通过虚拟设备更新JTAPI将更新使用JTAPI的所有Singlewire应用的JTAPI版本。

这些步骤在以下指南https://community.cisco.com/t5/collaboration-voice-and-video/integratingbasic-cisco-paging-basic-informacast-with-cucm/ta-p/3161322的Update JTAPI In Informacast部分 中有所<u>描述</u>

常见缺陷

CSCve47332 Cisco IP电话69XX系列无法处理Informacast应用用户中的空格

CSCuy56088 8800系列电话无组播音频

CSCut91894 FF37和Chrome到InformaCast的连接在FF/Chrome更新后失败

CSCtb70375 SNMP需要向用户发出DNS连接问题的警报

相关信息

- CUCM兼容性列表:<u>https://www.singlewire.com/matrix/cisco-platforms</u>
- 电话矩阵: <u>https://www.singlewire.com/matrix/cisco-phones</u>
- •升级路径:<u>https://www.singlewire.com/matrix/ic-upgrades</u>
- 服务器平台:<u>https://www.singlewire.com/matrix/server-platforms</u>
- 硬件要求:<u>https://www.singlewire.com/informacast-hardware-requirements</u>
- 技术支持和文档 Cisco Systems SRND:<u>https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12.pdf</u>
- CUCM与思科分页服务器/InformaCast集成配置示例

 https://www.cisco.com/c/en/us/support/docs/unified-communications/paging-server/117059
- configure-informacast-00.html
- •思科寻呼服务器 快速入门指南
 - : <u>https://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/cucm/cisco_paging_server/12_5</u> _1/QSGInformaCastBasicPaging1251.pdf