Configure and Troubleshoot Informacast

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Introduction

This document describes the Cisco Paging Server product (also known as InformaCast) and how to integrate it with Cisco Unified Communications Manager (CUCM). This document will cover the purpose of the feature, configuration of the feature, what data to collect for troubleshooting, example analysis of the data, and related resources for additional research.

Prerequisites

Requirements

Cisco recommends that you have knowledge of these topics:

- Cisco Unified Communications Manager
- InformaCast
- SIP, CTI, Http and SNMP protocol.

Components Used

The information in this document is based on these software and hardware versions:

- InformaCast Version 11.5.2 38
- CUCM Versions 11.5.1.14900-8
- CP-8811 and CP-8861 sip88xx.12-0-1SR1-1
- Basic License

The information in this document was created from the devices in a specific lab environment. All the devices used in this document started with a cleared (default) configuration. If your network is live, make sure that you understand the potential impact of any command.

Background Information

Purpose of the Feature

The Cisco Paging Server is a paging/mass notification solution for thousands of phones, speakers, and other devices. This is especially useful in emergency situations with live, prerecorded audio, and/or text announcements.

Upon Original Equipment Manufacturer (OEM) agreement with Singlewire (InformaCast vendor), Cisco Technical Assistance Center (TAC) supports InformaCast from Version 8.3 together with CUCM Version 8.5 and later. The only mode supported by the Cisco TAC is Basic Paging.

Basic vs Advanced

Basic Paging mode supports live audio broadcast only for up to 50 phones per recipient group and require no additional license. The InformaCast version provided as part of CUCM includes a license for Basic Paging mode. Customers who need supplementary functionality can upgrade to Advanced Notification mode and be supported by Singlewire.

An advanced paging license allows unlimited paging groups. It also makes possible other advanced functions, including paging to overhead analog and IP speakers, bell scheduling,

prioritizing emergency notifications with the call-barge option, prerecorded and text-only pages, integration with social media sites for notification, email and Short Message Service (SMS) mass notification and all-number monitoring, Emergency Services alerting, and integration with Cisco Jabber clients. After the installation of InformaCast, you can enable a trial of Advanced Notification mode.

Protocols Used

The Cisco Paging Server communicates with Unified CM using **SIP**, **SNMP**, **AXL** and **CTI** and beginning with Cisco Paging Server 9.0.1, either **HTTP or JTAPI** can be used to communicate with phones.

The Cisco Paging Server uses SNMP to find the other Unified CM nodes as well as a list of phones registered to each cluster member. Once the SNMP communications are complete, the Cisco Paging Server uses AXL to determine additional information regarding each registered phone, such as device name, description, device pool, calling search space, directory number, and location. This information can be used to build logical groups of phones, called recipient groups. As mentioned before, in the Cisco Paging Server with basic license, recipient groups can contain a maximum of 50 phones.

Note: A single Cisco Paging Server per Unified CM cluster is supported.

HTTP vs JTAPI

InformaCast versions prior to 9.x all used HTTP for phone activation. In HTTP mode, Cisco Paging Server sends commands and credentials to each IP phone HTTP server. IP phones validate these credentials and then execute the commands. At broadcast send time, InformaCast contacts them directly with the XML Services Interface (XSI) over HTTP.

In JTAPI mode, Cisco Paging sends commands to each phone via Unified CM. Cisco Paging Server does not need to send credentials with each request, so each phone does not have to activate its web server, and commands are executed more quickly. In addition, CTI mode allows faster checking of busy phones and activate them.

You can use HTTP or JTAPI regardless the type of integration (SIP or CTI) with CUCM. Keep in mind that JTAPI works better than HTTP on phones with non-English locale. In order to confirm the User locale take a look at the phone web page.

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)

Note: In order to use JTAPI, take into consideration that CUCM version must be 9.1.2 or above, and Cisco 3905, 7902, 7905, 7912 phones are not supported.

SIP vs CTI

Informacast can receive calls through CTI and/or SIP. In the case of CTI, calls are serviced on a CTI Route Point (the Cisco Paging Server does not require CTI ports to answer inbound calls).

In the case of SIP, calls depart Unified CM on a SIP trunk. Both CTI and SIP are valid and supported. However, Cisco recommends SIP call flows over CTI because troubleshooting SIP integrations is much easier than CTI.

Configurations

Network Diagram



- 1. The caller (paging originator) dials a predefined number in Unified CM. E.g. 7777.
- 2. Unified CM routes the call to the Cisco Paging Server over either a SIP trunk or CTI route point.
- 3. The Cisco Paging Server answers the call.
- 4. The caller hears a low stall tone. While the Cisco Paging Server plays this tone, instructions are sent via HTTP or JTAPI to each phone in the recipient group to join to the multicast group.
- 5. Once all phones have joined the multicast group, the Cisco Paging Server plays a high go-ahead tone. When the caller hears this tone, it indicates that the Cisco Paging Server is ready to receive and sent the audio to the multicast IP and port.
- 6. When the caller speaks, the media is sent from the caller's phone to the Cisco Paging server, then from the Paging Server to the multicast IP address and port, and eventually from the multicast IP to the receiving phones.
- 7. When the caller hangs up, the instruction is sent to each IP phone, this time to leave the multicast group, and the broadcast is over.

When InformaCast is integrated with Cisco Call Manager using the JTAPI library and Computer Telephony Integration (CTI) Manager it uses Quick Buffer Encoding (QBE) protocol over TCP as shown in the image.



For SIP integrations, InformaCast uses SIP protocol over TCP and port 5060 to communicate with Call Manager as shown in the image.



Configure Call Manager

Step 1. Activate services, navigate to **Cisco Unified Serviceability > Tools > Service Activation** and enable the following services:

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

Tip: Activate SNMP on all nodes, AXL on at least one node in the cluster, and CTI Manager on at least one node running the Call Manager service (or more for redundancy purposes).

Step 2. Configure SNMP (version 2 or version 3)

For SNMP v2

- Navigate to **Cisco Unified Serviceability > SNMP > v1/v2**.
- Configure the community string name with access privilege of ReadOnly.
- Apply to All Nodes checkbox, if possible and click on Save.

Status	
Status : Ready	
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 only from these hosts Host IP Address Insert Host IP Addresses Remove
Access Privileges Access Privileges* ReadOnly Notify access privilege is required in order Apply To All Nodes	▼ r to configure Notification Destinations.

For SNMP v3

- Navigate to **Cisco Unified Serviceability > SNMP > V3 > User** and create a user named ICVA.
- Enable the **Authentication Required** checkbox, enter an authentication password and select the **SHA** radio button.
- Enable the **Privacy Required** checkbox, enter a privacy password and select the **AES128** radio button.
- Select **ReadOnly** from the Access Privileges dropdown menu and select the **Apply To All Nodes** checkbox, if possible and click on **Save**.

Status				
Status : Ready				
Gerver* 10.1.61.158CUCM Voice/Video	Ŧ			
User Information				
User Name* ICVA				
Authentication Information				
Authentication Required Password	Reenter Password	Protocol	MD5	SHA
Privacy Information				
Privacy Required Password	Reenter Password	Protocol	O DES	AES12
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 SNMP Packets only from these hosts Host IP Address Host IP Addresses Remove]	
Access Privileges				
Access Privileges* ReadOnly	¥			
Notify access privilege is required in order	er to configure Notification Destinations.			



Save Clear All Cancel

Step 3. Set the Default Codec to G.711

- Navigate to CM Administration > System > Region Information > Region and create a new region, e.g. ICVA.
- Select all your regions in the Regions area, and configure **64kbps (G.722, G.711)** as the Maximum Audio Bit Rate.
- Select the **None** radio button in the Max Video Call Bit Rate and click on **Save**.

Region Configuration							Related	Links: Back To Find/List	•
🔚 Save 🗙 Delete 🕻	🎦 Reset 🏒 Apply Config 🛛	🚰 Add New							
Name* ICVA									
Region Relationships									_
Region	Audio Codec Prefe	rence List	Maximum Ra	Audio Bit	Maximum Session I Call	Nt Rate for Video	Maximum Ses	sion Bit Rate for Immersive Vi Calls	deo
Default	Use System Default (Fac	tory Default low	64 kbps G.7	(6.722,	Nor	e		None	
ICVA	ICVA Use System Default (Factory Default low		64 kbps	(G.722,	Nor		None		
Mex	Use System Default (Fac	Use System Default (Factory Default low		64 kbps (G.722, None		2		None	
SanJose	Use System Default (Fac loss)	tory Default low	64 kbps G.7	(G.722, 11)	Nor			None	
NOTE: Regions not displayed	Use System D	efault	Use Syste	m Default	Use System	n Default		Use System Default	
- Modify Relationship to o	other Regions								
	Regions	Audio Codec Pre	ference List	Maxim	um Audio Bit Rate	Maximum Sessio for Video C	n Bit Rate Calls	Maximum Session Bit Rate fo Immersive Video Calls	*
Default ICVA Mex SanJose	*	Keep Current S	etting 🔻	e 64 kbps (i	5.722, G.711) V	 Keep Current Use System I None 	: Setting G Default G	Keep Current Setting Use System Default None	

Note: The multicast media streams always use the G.711 mu-law codec. No other codecs are allowed or supported. Calls arriving to Informacast using other codecs must be transcoded.

Step 4. Create a Device Pool

- Navigate to CM Administration > System > Device Pool and create a device pool. E.g. Name it ICVA_DP.
- Add the ICVA region you just created to it.
- Select **Disable** from the **SRST Reference** dropdown menu.
- Select On from the Join Across Lines dropdown menu and click on Save.

Device Pool Configuration									
🔚 Save 🗶 Delete 🗋 Co	py 🎦 Reset ,	🖉 Apply Config 🚽 Add New							
Device Pool Settings									
Device Pool Name*		ICVA_DP							
Cisco Unified Communications Manager Group*		Default	~						
Calling Search Space for Auto-	registration	< None >	~						
Adjunct CSS		< None >	~						
Reverted Call Focus Priority		Default	~						
Intercompany Media Services I	Inrolled Group	< None >	~						
Date/Time Group* Region* Media Resource Group List Location Network Locale SRST Reference* Connection Monitor Duration**	CMLocal ICVA < None > < None > Disable	> > > > > >							
Single Button Barge* Join Across Lines*	Default	~							
Physical Location	< None >	v							
Device Mobility Group	< None >	~							
Wireless LAN Profile Group	< None >	 Viet 	w Details						

Step 5. Create a Route Partition, e.g. ICVA_PT.

Step 6. Create a Calling Search Space, e.g. ICVA_CSS. Include the ICVA_PT.

Step 7. Create an Access Control Group (AXL).

- Navigate to **CM Admin > User Management > User Settings > Access Control Group** and create an access control group, e.g. ICVA User Group.
- Add the Standard AXL API Access role to it.

Note: You may already have an access control group named Standard AXL API Access with the Standard AXL API Access role added to it, which you can also use.

Step 8. Create an Application User

- Navigate to **CM Admin > User Management > Application User** and click on **Add New**. Name the application user as **ICVA_InformaCast** and assign these roles:
- 1. Standard CTI Enabled
- 2. ICVA User Group (or Standard AXL API Access)

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

Standard CTI Enabled

- 3. Standard CTI Allow Control of Phones Supporting Connected Xfer and Conf.
- 4. Standard CTI Allow Control of Phones Supporting Rollover Mode
- 5. Standard CTI Allow Control of All Devices

Application User Configu	ration	
🔚 Save 🗶 Delete [Copy 🕂 Add New	
Application User Informat	ion	
User ID*	ICVAInformacast	Edit Credential
Password		1
Confirm Password	••••••	1
Digest Credentials		1
Confirm Digest Credentials	C	Ĩ
BLF Presence Group*	Standard Presence group ~	
User Rank*	1-Default User Rank ~	
Permissions Information —		
Groups ICVA User Group Standard CTI Allow Co Standard CTI Allow Co Standard CTI Allow Co Standard CTI Allow Co Standard CTI Enabled	entrol of All Devices entrol of Phones supporting C entrol of Phones supporting R View Details	Control Group Access Control Group
Roles Standard AXL API Acce Standard CTI Allow Co	ess ontrol of All Devices	

Warning: Per defect <u>CSCve47332</u>, it is recommended not to use spaces for the application User ID.

View Details

Step 9. Integrate Communications Manager with Informacast using SIP or CTI.

For SIP integration, create a SIP profile, a SIP Trunk and a Route Pattern.

- Navigate to CM Admin> Device > Device Settings > SIP Profile and click on the Standard SIP Profile then click on the Copy
- Name the profile as ICVA SIP Profile and select Best Effort (no MTP inserted). Click on Save.
- Navigate to CM Admin > Device > Trunk and click on the Add New
- Select SIP Trunk from the trunk type dropdown menu. Click on Next and enter a name for your SIP trunk.
- Select the device pool ICVA_DP, scroll down to the SIP Information area and enter the IP address of your InformaCast server in the Destination Address
- Ensure that the value in the Destination Port field is 5060, select the **Non Secure SIP Trunk Profile**, and assign the SIP profile you created before from the SIP Profile dropdown menu. Click on **Save**.

Trunk Configuration			
🕞 Save 🗙 Delete 省 Reset 🕂	Add New		
Device Information			
Product: Device Protocol: Trunk Service Type Device Name*		SIP Trunk SIP None(Default) ICVA_SipTrunk	
Description		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	
Media Termination Point Required			
SIP Information			
Destination			
Destination Address is an SRV			
Save Debte Preduct: SIP Product: SIP Trunk Device Protocol: SIP Trunk Service Type None(Defaulk) Description 10.1.61.113 Device Pool* [CVA_OP 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 Mode* None Obestination Address is an SRV Destination Address IPv6 Destination Point Required			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 >	~	

• Create the route pattern, navigate to CM Admin > Call Routing > Route Hunt > Route pattern, click on Add New.

View Details

< None >

ICVA SIP Profile

No Preference

SUBSCRIBE Calling Search Space

DTMF Signaling Method*

SIP Profile*

- Enter a route pattern, e.g. 7777 and configure a partition that is reachable from the phones, e.g. ICVA_PT.
- Select the SIP trunk you just created from the Gateway/Route List dropdown menu.
- Select the Route This Pattern and the OnNet radio buttons.
- Uncheck the Provide Outside Dial Tone checkbox and click on Save.

For CTI integration, create a CTI route point and associate to the Application User created in step 8.

- Navigate to CM Administration > Device > CTI Route Point and click on Add new.
- Enter a name, e.g. ICVA_CTI_RP (or whatever you prefer).
- Assign the device pool ICVA_DP and click on Save.
- Select the line 1, enter a directory number, e.g. 7778, and assign the recently created partition (ICVA_PT).
- Configure the rest of information as desired and click on **Save**.

Add the CTI route point(s) as controlled devices on the ICVA application user's configuration.

Controlled Devices	ICVA_CTI_RP	^
		~

Note: InformaCast can support multiple CTI route points if they are created in Communications Manager and associated to the InformaCast application user.

Tip: Instead of creating a CTI route point for every number you need for DialCasts, you could also add multiple lines to a single CTI route point. Another option would be to use wild card patterns to match a range of numbers.

Step 10. Enable Web Access for Cisco IP Phones to use HTTP to control the phones.

- Web access can be configured per device, per common device profile, or system-wide in the Enterprise Phone Configuration.
- In order to apply the change in Enterprise phone configurations, navigate to CM Admin > System > Enterprise Phone Configuration, scroll down to the Web Access dropdown menu and select Enabled. Click on Save.
- Reset the phones to apply the changes.

Enterprise Phone Configuration			
Save			
Web Access*	Enabled	~	2

Step 11. Set the Authentication URL.

Change the authentication URL in order to send authentication requests from IP phones to InformaCast. All non-InformaCast authentication requests are redirected back to the default CUCM authentication URL.

- Navigate to **CM Administration > System > Enterprise parameters**.
- Enter http://<InformaCast Virtual Appliance IP Address>:8081/InformaCast/phone/auth

in the URL Authentication field and Secure Authentication URL.

• Click on Save, Apply config and Reset the phones.

🚽 Save 🤌 Setto Default 🌯 Reset 🥒 Apply Conto	
Phone URL Parameters	
URL Authentication	http://10.1.01.118:8081/3nformaCast/phone/auth
URL Directories	http://10.1.61.158:8080/comcip/vmkdirectory.jsp
URL INFe	
URL Idle Time	0
URL Information	http://10.1.01.158.0000/comcip/GetTelecasterHeigText.js
JRL Messages	
P. Phone Provx Address	
URL Services	http://10.1.61.150:0000/comcip/petservicesmenu.jsp
Secure Phone URI. Parameters	
Secure Authentication URL	http://10.1.61.118:8081/InformaCaot/phone/auth
lecure Directory URL (199L)	Mtps://10.1.61.158.8443/comop/vnidrectory.jsp
Secure Contact Search URL (UDS)	https://10.1.61.159:8443/com-uds/users
ieoure Idie URL	
Secure Information URL	https://10.1.61.158:8443/concip/GetTalecasterNelpText.j
Secure Messages URL	
Secure Services URL	https://10.1.61.150.8443/com/ac/ostserv/ceamenu.lsp

Note: The URL is case sensitive, so make sure that the I and C in the word InformaCast are capitalized. Both the secure authentication URL and the authentication URL must be set to the same value, the HTTP URL.

Step 12. Set the Authentication Method for API Browser Access.

 If you're using Unified Communications Manager 11.5.1 and later, scroll down the page to the Security Parameters area and select **Basic** from the **Authentication Method for API Browser Access** dropdown menu.

Step 13. Test your phones, e.g. dial 7777 (for SIP integration) or 7778 (for CTI integration).

Note: If you are running Unified Communications Manager in mixed mode, ensure that calls to and from InformaCast are not using encrypted media.

Configure Informacast

Step 1. Configure the Communications Manager Cluster in Informacast.

- Log in to Informacast and navigate to Admin > Telephony > Unified Communications Manager Cluster. Click on Edit.
- Enter the application user's username and password for the Application User that you created in step 8.
- Make sure the **Use Application User for AXL** checkbox is selected, meaning that your application user credentials are used when building InformaCast's phone cache.

Note: If you leave this field blank, InformaCast will attempt to find a server running the AXL service among those servers running the CallManager service.

- Enter the IP address of the Unified Communications Manager server(s) in the Communications Manager IP Address(es) field. Use the numeric IP addresses rather than DNS names.
- Select the SNMP v2 or SNMP v3 radio button. Enter the same information configured in

CUCM. Click on Update.							
nformaCast ^o 🧕 💮 🚬	terne Hessapes	8 Recipients	٩	800 E	None Address	() Pagins	(?) 140
			_	L.	ig Out App	lication Adn	nnistrato
Admin Telephony Cisco Unified Cor Configuration	nmunication	is Manag	ger Clu	ster E	dit Tele	phony	
	elephony Config	guration					
Unified Communications Manager Cluster Description	CUCM	0	required)				
Unified Communications Manager Application User	(CVRInformacast		required)				
Unified Communications Manager Application Password,	•••••						
Confirm Application Password	•••••						
	Use Applicatio	n User for Al	8				
ASIL IP. Address(ss).	10.1.61.158						
Unified Communications Manager IP, Address(es);	10.1.61.158	0	required)				
Storest SMMP version.	SHMP v2 (req SHMP v3 (req	(brid)					
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 this Parameters page) is set to the following value:	casts or if this is n s cluster (found in t	ot your prim the Phone U	ary cluster RL Paran	make su neters sec	e the URL tion of the	Authentica System E	tion nterpris
http://10.1.6	.118:0081/Ind	lormaCast/	phone/w	410			
Optionally, you can also tell InformaCast where to send a need to do this if, before installing InformaCast, you had a such cases, copy the current Unified Communications M	uthentication reque at this Unified Con anager setting into	ests for communications the field belo	nands that Manager w, before	aren't con parameter changing it	ting from in to a non si to the valu	rformaCast. tandard valu re shown ab	You only a. In ove.
Next Authentication URL:							
If empty, non-informaCast authentication requests from ph authentication page, http://10.1.41.159/comcts/	ones in this cluste authenticate.;	er will be sen SKP	t to the de	fault Unifie	d Commun	ications Ma	nager
CANCEL T	•		n Ø				

Step 2. Configure the Recipient Group.

• Navigate to **Recipients > Edit recipient Groups** and click on **Update** in order to show all the phones registered in CUCM and discovered by InformaCast.

InformaCast [®]				ution 200 Learn	1	Recorder	8	C) Spendaren	See.	Admin 1	O Pagen	? 140
8	Recipients	Edit Recip	ient Gro	neci	ipient gro	oup memb	ers upda	ted		90.01629	lication Adv	ene stration
	Unset O Discover current IP phone information from Cisco United Communications Manager (may be time consuming).											
	A Name (All Recipients)								Phones 2	Action	an 0 a	

• In order to create a new Recipient Group click on **Add**, write a name and then click on **Edit** to add the phones for this recipient group. Once the phones were added to the recipient, click on **Submit**.

Select	Individual Recipients				×
Filter Availa	citar ble Recipients (double click to select)		Salec	ted Recipients (double click to remove)	
	Descriptive Text			Descriptive Text	
٠	Cisco IP Phone: Auto 111; DNs: 111; SEPF87B204EED99		٠	Cisco IP Phone: Auto 110; DNs: 110; SEP2C3124C9F8E1	
		Add			
		Bamova			
	ected cher for and a last sa				_
	n - post i total e ann e e			Submit Can	cel

• To save the changes click on Update.

Inform	aCast ^e esic paging	Adver CO Buy	٩	etice 2000 Learn	formal Research	Nessage	Bacqueents	Speakers	2	8000 Anno	O Pugins	() 140
8	Recipients Ed	lit Recip	ient Gro	ups E	dit Ro (requi	cipient (rd) K Teg =	Group		LO	g Out App	ication Adm	in strato
	Cisco IP Phone A Filter with Recipiert Filter with Rules Exclusions are only	uto 110; DNA .Gransk 👜 available wh	s 110: SEP	pient Grou	FIE1	d by Recipi	ent Groups	or Rules.				

Step 3. Allow/Disable SIP Access to InformaCast.

- Navigate to Admin > SIP > SIP Access. By default, all SIP calls are denied.
- Select the **Allow** radio button allows all SIP calls or click on **Add** to allow exceptions to this allowance.

	Tarreet											
Inform	aCast ^a asic paging	Advi O Day		adion 2017 Learn	former Name	Dessage	8	Speakers	See.	Admin	() Pisgins	? Nelp
ф. С	Admin SIP S	SIP Acce	oss c	ontrols ac	cess of in	bound SIP	calls to Inf	ormaCast.				
, Ó	Click to restore to	default se	ttings 🕞	1100 O	New O	Deny inco	ming SIP ca	ilis				
					0	a host ex	rception					
				GANER			(875	ALE 🚱				

Tip: When defining exceptions, make sure to specify the host that directly sends the INVITE request to InformaCast. This may be a SIP proxy server if proxies stand between InformaCast and the calling host.

- Go to Admin > DialCast > Dialing Configurations, click the Add
- Enter a dialing pattern (e.g. 7777, 7778) in the **Dialing Pattern** field according to the Route pattern (for SIP integration) or CTI Route Point (for CTI integration) created in CUCM.
- Select the recipient groups from the list and click on Update.

Inform	aCast ^e asic paging		٩	anton 2000 Learn	() 1000	Receipter	ES Recipients	Constant Sector	S) Defi	e Admin	0 Phopen	(?) Help
а <mark>ф</mark>	Admin DialCa	st Dia	ling Cor	Dairg	ions E	dit Broa	dcast D	Dialing C	Configu	ration	Let and a second	
				Recipient	Groups;	(All Recip Mer SanJose	ients) -					
				CLOCK,	B			11 O				

Step 5. Configure the broadcast parameters.

- Navigate to Admin > Broadcast Parameters
- Configure the IP for multicast. The default IP (239.0.1.2) is commonly used.
- Enable the JTAPI checkbox if you want to send the commands to the phones as JTAPI, otherwise HTTP messages will be used.

Inform	aCast*	0	<u>چ</u>	aton 200 Learn	formal Research	Recenter of	Bacquierts	Speakers	800 M	Sec.	() Plugins	(?) 1449
									L	ig Out App	Ication Adv	nnstats
*	Admin Broad	icast Par to Phones b shany Termini Multicast P /	ameters y JTAPI: 2 dis for all Phones: 2 Address: 2	5] 29.0.1.2		franki	nd					
	Ending	Multicast IP /	Address: 37	29.0.1.2		(requi	(bar					
	ever a	0000000000	\$	ee shttp://w	miana.	erg/assignm	ents/multica	ant address	652.			
		Multic	ast TTL; 1	6 (req	(berk							
				Canoli D	3		100	uu 🔕				

Ensure that this range corresponds to your network infrastructure settings and covers all recipient groups. In multisite deployments, Singlewire and Cisco recommend that a range of addresses be used. This range should be large enough in order to handle one address for each simultaneous broadcast.

Note: The use of JTAPI is recommended over HTTP since it better monitors the status of phones and works with more locales.

Tip: The default settings for the web interface will log you out after five minutes. Navigate to Admin > Network Parameters > Session Timeouts and change the General Session Timeout (seconds) field from 300 to the new value.

Configure Multicast in the Network

If the Cisco Paging Server and IP phones are on separate IP subnets, the routers in between those two subnets must be configured for multicast routing.

The Cisco Paging Server does not require any particular method of multicast routing (SM, DM, S-DM, SSM, and so forth). Some wide area network environments do not support multicast routing. For those environments, GRE tunnels may be built between sites and used to transport multicast.

The design and configuration of multicast in your environment is outside the scope of this document, but you may find the following resources helpful:

- <u>Multicast whitepaper</u>
- <u>Multicast Testing Tool</u>

Note: If you are using Meraki switches, they have IGMP snooping enabled by default. This can cause issues and needs to be disabled by Meraki. Once you contact them and have them disable IGMP snooping, test the paging again.

Verify

There is currently no verification procedure available for this configuration.

Troubleshoot

Common Issues

Phones not Activated

Take into consideration that Informacast skips any phones that are in use (busy) when the broadcast occurs.

InformaCast uses different busy detection methods depending on how you send messages to the phones (HTTP or JTAPI).

HTTP: Busy detection only works with phone locales running English loads

CTI: Works with non-English phone locales

Busy detection also works differently according to protocol as well as line type and line state.

Line status	CTI busy detection	HTTP busy detection
Shared line with call in use on another phone, no call on hold	Idle	Idle
Off hook, collect digits	Busy	Not busy
Talking, active call	Busy	Busy
On hold, inactive call on shared line	Busy	Not busy
On hold, inactive call on unique line	Busy	Not busy

Note: If there are simultaneous broadcasts attempted, Informacast plays the first broadcast first (the second broadcast is bumped).

When troubleshooting a phone not being activated you should collect the following data:

- Performance logs from Informacast.
- Console logs (PRT) from the phone.

Phones not Discovered

Only registered phones are discovered by InformaCast. If an IP phone is registered but not discovered, check the SNMP service configuration in Informacast and the CUCM node where the phone is registered to. The SNMP service and community string should be configured for all nodes where the Call Manager service is activated.

SNMP Error Unable to build recipient groups: java.lang.Exception



- 1. The error means that SNMP fails to respond to queries in a timely manner due to DNS connectivity or resolution.
- 2. Confirm that nothing is blocking UDP port 161 from the InformaCast server to all Unified Communications Manager cluster nodes.
- 3. Confirm that SNMP information is correct. Navigate to Admin > Telephony > Unified Communications Manager Cluster and type a new SNMP string if possible. Configure the new string in CUCM.
- 4. You may also be using a community string that exceeds the maximum number of characters for the community string. If you are copying the community string from CUCM and pasting it into the Informacast configuration, try typing it in to see if you can type the whole string. In Informacast version 11 the maximum number of characters is 18.
- 5. Check your DNS configuration on CUCM is correct and confirm you are not matching the defect <u>CSCtb70375</u>.

No Audio on the Destination Phones

If phones light up but don't play the audio the issue is most likely related with multicast routing and not with your CUCM server or IP phones.

Data to Collect

When troubleshooting Informacast you should collect the following data:

- 1. Performance logs from Informacast.
- 2. Packet capture from Informacast.
- 3. Packet capture from the phones.
- 4. Packet capture from CUCM.
- 5. SDL logs from CUCM
- 6. PRT (console logs)

Performance Logs

There are two methods to get the performance logs from Informacast.

Method 1

- 1. Navigate to https://<Informacast IP>:8444/InformaCast/logs/performance.log
- 2. Copy the and save the log into a .txt file.

Method 2

- 1. Open the Informacast IP in a web browser, https://<informacast_IP> and select **Informacast**.
- 2. Use your credentials to log in and navigate to Help > Support.

	Adva () Buy	anced Notific	ation >>>> Learn	Home	Messages	Recipients	Speakers	Bell	k Admin	() Plugins	? Help
									InformaCast	t User Guid	enistrator
									Frequently /	Asked Ques	stions
									Troubleshoo	oting Guide	
									API Docume	entation	
Inf	ormaCast	Basic Pa	ging - P	rovided	by OEM	Agreeme	nt with (Cisco	API Quick S	tart Guide	
You	currently ha	ve InformaC	ast Basic F	Paging inst	alled for us	e. Click the	Try and Bu	y int.	Support	>	

unlock a 60-day free trial or upgrade to InformaCast Advanced Notification.

3. Click on **Performance Logs** under the Tools section as shown in the image.

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.

Packet Capture

From Informacast

There are three methods to get a packet capture from Informacast.

Method 1

- 1. Connect to the CLI of the Informacast box via SSH
- 2. Execute the command **sudo capturePackets test.cap** to start capturing and create a file named **test.cap**
- 3. Page out to the phones that aren't working
- 4. Hit Ctrl + C to end the pcap
- 5. Execute Is to ensure the packet capture is on the box

6. use SFTP or Secure Copy (SCP) in order to transfer the file to your 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>
```

Method 2

- 1. Download and install <u>InformaCast_LogTool</u> from the web.
- 2. Execute the software and select the option **[5]**. Write the IP of Informacast, the login credentials and the seconds that the packet capture should run as shown in the image.

```
Administrator: Singlewire Software: InformaCast Log Tool - 20150707

III Gather Logs From InformaCast Server
III Uncompress Gathered Logs
III Parse Logs For Errors and Solutions
III Turn JTAPI Debugging On/Off
ISI Network Traffic Capture
I91 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]: ______
```

3. The capture will not start immediately, this allows you to prepare your test environment. When ready, select option [1] and press Enter to start the capture of packets as shown in the



image.00:00:51

- 4. The tool will display a countdown timer with the outstanding duration of the capture. Replicate the issue during this time and when the capture countdown reaches zero the capture is complete and stops.
- 5. The tool bundles the packet capture and all the logs into a **.tgz** file and transfers it to your workstation. This is the same as option 1 to gather logs, but also includes the network traffic capture.
- 6. The tool will create a folder with the packet capture in the base directory of the Informacast_LogTool.exe as shown in the image.

```
InformaCast_LogTool.exe
```

InformaCast_LogTool_Logs_201809231605.tgz

Method 3 (Available in versions 12.0.1 and above)

- 1. Log into <Informacast_IP>:10000
- 2. Navigate to System > Capture Network



3. Click on **Start a new packet capture** and replicate the issue as shown in the image.



- 4. Click on **Stop Packet Capture** when the issue is totally replicated, or it stops by itself after capturing 33,000 packets.
- 5. Navigate to **System > Collect Logs**, enter a short description of the problem and click on **Collect a new set of logs**.
- 6. In order to save the logs click on **Download to Your Computer** as shown in the image.



Method 4 (Available in version 12.0.1 and above)

In version 12.0.1 and later sudo command is no longer required. In order to run a packet capture use the command **capture-packets <name of the file> <number of packets>** as shown in the example:

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 Interrupt signaled. Cleaning up.

Note: The GUI method is better than the CLI since there is no dependency on an SFTP server, and you can start, stop and download the packet capture from the web page.

From CUCM

Define from where you need to get packet capture according to your deployment. You can have only one CUCM node or multiple CUCM in the cluster.

• If you have one CUCM node, get the packet capture as shown in the image.



• If you have a CUCM cluster and one node is communicating with Informacast but another is communicating with the phones, then get the packet capture as shown in the image.



- 1. Open a SSH session for the node where you need to capture
- 2. Run the command **utils network capture eth0 size all count 1000000 file Test** to start the packet capture.
- 3. Replicate the issue
- 4. Stop the packet capture with Ctrl + C
- 5. In order to confirm that the packet capture was save, run the command **file list activelog platform/cli/***

P	10.1.61.158 - PuTT	γ	 x
admin:			^
admin:utils network capture e	th0 size all count	1000000 file Test	
Executing command with option	3:		
size=ALL coun	t=100000	interface=eth0	
src= dest	=	port=	
ip=			
Control-C pressed			
admin:file list activelog pla	tform/cli/*		
Test.cap			
dir count = 0, file count = 1			
admin:			\sim

6. Use the command **file get activelog platform/cli/Test.cap** to send the packet capture to a SFTP server. Alternatively, to collect all .cap files stored on the server, use **file get activelog platform/cli/*.cap**

admin:file get activelog platform/cli/*.cap
Please wait while the system is gathering files infodone.
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. Use RTMT in case you are not able to use an SFTP server. Navigate to System > Trace & Log Central > Collect Files. Click on Next and enable the Packet capture logs checkbox as shown in the

Select System Services/Applications	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		

- 8. Click on Next, select a download file directory and click on Finish.
- 9. Delete the packet with the command file delete activelog platform/cli/Test.cap

From the Phone

- 1. Activate the SPAN to PC port. Navigate to **CM Admin page > Device > phone** and find the phone reported with issues.
- 2. Under Product Specific Configuration Layout section, find Span to PC Port and select Enable from the drop-down menu. Click on Save and then on Apply config.
- 3. Connect a laptop to the PC-port of the phone.
- 4. Run the packet analyzer software in the laptop. You can use Wireshark (or other packet capture software).
- 5. Replicate the issue.
- 6. When the issue is totally replicated proceed to stop the packet capture.

You can find more details in the following link:

https://supportforums.cisco.com/document/44741/collecting-packet-capture-cisco-ip-phone

Example Analysis



SDL Traces

For SIP integration and phones controlled by JTAPI

CUCM: 10.1.61.158

Informacast: 10.1.61.118

Phone A

DN: 110

Model: CP-8861

Firmware version: sip88xx.12-0-1SR1-1

Phone A IP address: 10.1.61.12

MAC SEP2C3124C9F8E1

Phone B

DN: 111

Model: CP-8811

Firmware version: sip88xx.12-0-1SR1-1

Phone B IP address: 10.1.61.11

MAC SEPF87B204EED99

Dialcast number: 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;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 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;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

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=0si1 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=SIP 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=0

o=SinglewireInformaCast-SIP 1568074182370 1 IN IP4 10.1.61.118
s=SIP 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; gci= 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 $v_z = 0$

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>
<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

<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
```

<x-cisco-remotecc-request>

```
<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>
   <sub></sub>
</dialog>
 <presence usage="">
       <unot></unot>
  <sub></sub>
 </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
```

For CTI integration and phones controlled by HTTP

CUCM: 10.1.61.158

Informacast: 10.1.61.118

Phone A

DN: 110

Model: CP-8861

Firmware version: sip88xx.12-0-1SR1-1

Phone A IP address: 10.1.61.12

MAC: SEP2C3124C9F8E1

Phone B

DN: 111

Model: CP-8811

Firmware version: sip88xx.12-0-1SR1-1

Phone B IP address: 10.1.61.11

MAC: SEPF87B204EED99

Dialcast number: 7778

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=SIP Call b=AS:4064 t = 0 0m=audio 19104 RTP/AVP 114 9 124 0 8 116 18 101

```
c=IN IP4 10.1.61.12
b=TTAS: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
#### Digit analysis for the dialed number 7778
71531367.000 |19:15:34.231 |Sdlsig
                                   DaReq
                                Da(1,100,216,1)
wait
                                  1,100,14,1368.88^10.1.61.12^*
Cdcc(1,100,224,12)
                                                                            [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
|ready |CTIDeviceLineMgr(1,200,25,1)
|StationCdpc(1,100,67,2) |1,100,14,1.33^*** |[R:NH: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=000000000003AAA012639B70000000 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 CTT 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: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

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-0
 |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=STP Call c=IN IP4 10.1.61.118 b=AS:64 t = 0 0m=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.

Performance Logs

For SIP Integration

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-d761e7a4-1996c-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-d76le7a4-1996c-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

For CTI Integration



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

Console Logs (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

Packet capture

Collect a packet capture from the phone and verify the HTTP XSI commands from InformaCast. An Internet Group Management Protocol (IGMP) message is sent in order to join the multicast stream. If you do not see a Multicast Real-Time Transport Protocol (RTP) stream after the IGMP message, you can take a packet capture from InformaCast, confirm that Informacast server is sent the RTP to the IP and port and then inspect your network infrastructure.

Packet capture on the phone (controlled by HTTP)

- CUCM: 10.1.61.158
- Informacast: 10.1.61.118
- Phone B IP address: 10.1.61.11
- Model: CP-8811
- Firmware version: sip88xx.12-0-1SR1-1
- eth.addr==SEPF87B204EED99

The HTTP and IGMP messages received on the phones are shown in the image.

Fil	e Edit View Go Capture A	analyze Statistics Teleph	ony Wireless Tools Help				
4	0 0 1 0 1 0 0	*****	0,0,0,1				
	http://igmp					🛛 🗔 🔹 Expr	ressi
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)	Informacast sends IP 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	into initiacast sentas in and port for interteast	
	1911 00:24:49.444329	10.1.61.11	10.1.61.118	HTTP/XML	1452 HTTP/1.1 200 OK	The share outliest could be farmoust	
+	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=rtpavvi	d&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		
-	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	phone 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		
	2518 00:25:31.234468	10.1.61.11	10.1.61.118	HTTP/XML	462 HTTP/1.1 200 OK	The phone leaves the IGMP group	
<						The phone leaves the form group	>
~	Hypertext Transfer Protoc	col					
	> POST /CGI/Execute HTTP	P/1.1\r\n					
	> Authorization: Basic S	SUNWQUluZm9ybWFjYXN0	OnJ0cGF2dmlk\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]				
	[HTTP request 1/1]						
	[Response in frame: 19	941]					
	File Data: 116 bytes						
¥	HTML Form URL Encoded: an	nnlication/x-wew-for	m-urlencoded				
	<pre>> Form item: "XML" = "<c< pre=""></c<></pre>	CiscoIPPhoneExecute>	<executeitem url="RTPM#</td><td>x:239.0.1.2</td><td>:20486"></executeitem> "				

Packet capture on the phone (controlled by JTAPI)

- CUCM: 10.1.61.158
- Informacast: 10.1.61.118
- Phone B IP address: 10.1.61.11
- Model: CP-8811
- Firmware version: sip88xx.12-0-1SR1-1
- MAC SEPF87B204EED99

As discussed in the configuration section, phones can be controlled by JTAPI, that means that the **Send Commands to Phones by Jtapi** is enabled as shown in the image.

InformaCast® basic paging Provided by CEM Agreement with Clace		Adva (Rec) Buy	anced Notific	ation Solution Learn
Admin Broadcast Parame	ers nd Commands to Phones by JTAPI: Telephony Terminals for all Phones: Starting Multicast IP Address: 239.0.1.2 Ending Multicast IP Address: 239.0.1.2 (required) See <http: assignments="" multicast-addresses="" www.iana.org=""> Multicast TTL: 16 (required)</http:>			

If that is the case, the phone B receives from the CUCM server the IP and port of multicast through a SIP REFER. You can click on the **SIP REFER** message, then right click on te **Message Body** header and select **Show Packet Bytes** as shown in the image.

PhoneB_capture_JTAPI.pcapng				>
File Edit View Go Capture Analyze Statistics Telep	phony Wireless Tools Help			
🚄 🔳 🖉 📵 📕 🛅 🗙 🙆 🔍 🗰 🗯 🚟 🤻	F 🖢 🚍 🔲 Q, Q, Q, 🕅			
sip.Call-1D == "79d40280-d8a133a2-29151-9e3d010a@10.1.61.158"				Expession
No. Time Source Destin	ation Protocol Length Info			
61 15:28:04.551855 10.1.61.158 10.1 63 15:28:04.567280 10.1.01.12 10.1 65 15:28:04.768154 10.1.61.12 10.1 71 15:28:04.772605 10.1.61.12 10.1	.61.12 SIP 1358 Request: REFER sip:e2883 .61.158 SIP 637 Status: 202 mcLepted [.61.158 SIP 624 Request: NOTFY sip:111(.61.12 SIP 403 Status: 200 OK [942-2853-	<pre>4eab-a0d9-96228C796062@10.1.61.12:49348;transport=tcp (application/x-cisco-remotecc-request+xml) (applic 58:5960;transport=tcp (application/x-cisco-remotecc-response+xml) (application/x-cisco-remotecc-cm+xml)</pre>	ntion/x-cisco-remotecc-cm -xml)
<			📕 Wireshark - Message Body (sip.msg_body) - PhoneB_capture_JTAPI.pcapng	- 0 ×
> Frame 61: 1356 bytes on wire (10864 bits), 135 5 Ethernet II, Src: Wang-Sist355 (08:80:56:56:96) Internet Protocol Version 4, Src: 10.1.61.184 U Transmission Control Protocol, Src Poet: 500 Segueration Protocol (SEFER) > Resuge Header > Pessage Meder > Pessage Body > MUHE Hultipart Media Encapsulation, T	8 bytes captured (18864 bits) on interface 8 inisisa), Dist. (16:0; 4:e1099) (81:7bi28:44:e1099) Dati. 19, 1.61.12 Epand Subtres: Mayüscular-Derecha Collapse Subtres: Mayüscular-Derecha Collapse All Control-Derecha Collapse All Control-Inquierda Apply as Column Control-Mayüscular-I Apply as Filter Prepare a Filter Conversation Filter Colonize with Filter Follow Copy Show Roket Bytes Control-Mayüscular-X Suport Roket Bytes Control-Mayüscular-X Suport Roket Bytes Control-Mayüscular-X Wai Protocol Page Filter Field Reference Protocol Preferences	19/2.0	<pre>uniqueBoundary Content-Type: application/x-cisco-remotecc-request*xml (x-cisco-remotecc-request> (datagassithroughreq) capitalistionEdb*//policationIdb citationsequenceStationSequenceIsst(/stationsequence> citationsequenceStationSequenceIsst(/stationsequence> citationsequenceStationSequenceIsst(/stationsequence> citationsequenceStationSequenceIsst(/stationsequence> citationsequenceStationSequenceIsst(/stationsequence> citationsequenceStationSequenceIsst(/stationsequence> citationsequenceStationSequenceIsst(/stationsequence> citationsequenceStationSequenceIsst(/stationsequence> citationsequenceStationSequenceIsst(/stationsequence> citationsequenceStationSequence) crowtingBible/routingBible citationsequenceStationSequenceStationSequence> citationsequenceStationSequenceStationSequence> citationsequenceStationSequenceStationSequence> citationsequenceStationSequenceStationSequence> citationsequenceStationSequenceStationSequence> citationSequenceStationSequenceStationSequence> citationSequenceStationSequenceStationSequenceSequence> citationSequenceStationSequenceStationSequenceStationSequence> citationSequenceStationSequenceStationSequenceSequence> citationSequenceStationSequenceStationSequence</pre>	Start (C) End (dd; (2)
02a0 0d 0a 0d 0a 2d 2d 75 6e 69 71 02b0 64 61 72 79 0d 0a 43 6f 6e 74 02c0 70 65 3a 20 61 70 70 6c 69 63 a 02d0 78 2d 63 60 73 63 66 2d 72 65 6	Decode As Go to Linked Packet Show Linked Packet in New Window	-	Finds Copy Serve M C	Find Next

Once the phone receives the instruction, it joins to the multicast IP and port with an IGMP message. The phone attempts three times as maximum to start receiving audio. When the paging ends, the phones in the recipient group sends a Leave Group message to drop the multicast session.

Phone8_capture_JTAPI.pcapng														
File Edit View Go Capture Analyze Statistics Telephony Wireless Tools Help														
🚄 🔳 🖉 🐵 📕 🛅 🕱 🗳 🤤 🐐 🏝 🚁 👱 🔜 🔍 Q, Q, Q, 11														
l ond														
No.		Time	^	Source	Destination	Protocol	Length Info							
	66	15:28:04	.690300	10.1.61.12	239.0.1.2	IGMPv2	60 Membershi	p Report group	239.0.1.2	Join the multicast aroun				
	157	15:28:09	.140169	10.1.61.12	239.0.1.2	IGMPv2	60 Membershi	p Report group	239.0.1.2	som the monocast group				
	320	15:28:18	345630	10.1.61.12	239.0.1.2	TGMPv2	60 Membershi	p Report group	239.0.1.2	Losus multisast scoup				
	550	13:20:20	. 343039	10.1.01.12	224.0.0.2	10/19/2	oo ceave oro	up 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	l Version	4, Src: 10.1.61	.12, Dst: 23	9.0.1.2								
~	Internet	Group Ma	anagement	Protocol										
	[IGM	P Version	: 2]	(0.40)										
Type: Membership Report (0x16)														
	Chec	ksum: 0xf	9fc [corr	ectl										
Checksum Status: food														
	Multicast Address: 239.0.1.2													

Troubleshooting tools

<u>Multicast Testing Tool</u> will help you to troubleshoot SNMP further.

<u>InformaCast LogTool</u> will help you troubleshoot common issues experienced with implementing and maintaining InformaCast on your network.

Advance License

Customers with Advanced Notification mode are supported by Singlewire. Contact <u>sales@singlewire.com</u> for additional support.

Sunglewire support is available from 7 a.m. to 6 p.m. CDT, Monday through Friday at +1

608.661.1140 option 2.

Passwords

In Informacast, there are several type of passwords:

OS credentials: Used to enter Webmin and Control Center (<u>https://x.x.x.x:10000</u>) and when using SSH to access the InformaCast Virtual Appliance. The default user is **admin** while the password is **changeMe**.

Admin password: Used to log into the admin interface (https://x.x.x.x:8444/InformaCast/admin).

Passphrase: Used to secure your backups of the InformaCast Virtual Appliance. You must remember this passphrase. Singlewire Support personnel cannot recover it for you if it's lost.

Password recovery

For Cisco paging server 12.5.1 and forward: https://www.singlewire.com/help/InformaCast/v12.5.1/advanced/cucm/index.htm#t=InformaCast_F usion%2FWebmin%2FRecover_the_Servers_Password.htm

Update JTAPI in Informacast

When you initially install InformaCast Virtual Appliance or whenever you change versions of CUCM, you need to update the JTAPI library used by InformaCast Virtual Appliance to the same version used by your CUCM server.

Updating JTAPI through the Virtual Appliance will update the JTAPI version for all of the Singlewire applications that use JTAPI.

The steps are described in the section **Update JTAPI In Informacast** in the following guide <u>https://community.cisco.com/t5/collaboration-voice-and-video/integrating-basic-cisco-paging-basic-informacast-with-cucm/ta-p/3161322</u>

Common Defects

<u>CSCve47332</u> Cisco IP Phone 69XX Series cannot handle spaces in Application User for Informacast

CSCuy56088 8800 Series phone no multicast audio

CSCut91894 Connections from FF37 & Chrome to InformaCast fail after FF/Chrome updt

<u>CSCtb70375</u> SNMP needs to alert user of DNS connectivity issues

Related Information

- CUCM compatibility matrix: <u>https://www.singlewire.com/matrix/cisco-platforms</u>
- Phone matrix: <u>https://www.singlewire.com/matrix/cisco-phones</u>

- Upgrade paths: <u>https://www.singlewire.com/matrix/ic-upgrades</u>
- Server platforms: https://www.singlewire.com/matrix/server-platforms
- Hardware requirements: <u>https://www.singlewire.com/informacast-hardware-requirements</u>
- Technical Support & Documentation Cisco Systems SRND: <u>https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12.pdf</u>
- CUCM Integration with Cisco Paging Server/InformaCast Configuration Example: <u>https://www.cisco.com/c/en/us/support/docs/unified-communications/paging-server/117059-</u> <u>configure-informacast-00.html</u>
- Cisco Paging Server -Quick Start Guide : <u>https://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/cucm/cisco_paging_server/12_5_1/QSGInformaCastBasicPaging1251.pdf</u>