配置Informacast並對其進行故障排除

目錄

簡介

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

必要條件

需求

思科建議您瞭解以下主題：

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

採用元件

本文中的資訊係根據以下軟體和硬體版本：

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

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

背景資訊

功能的用途

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

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

基本與高級

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

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

使用的協定

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

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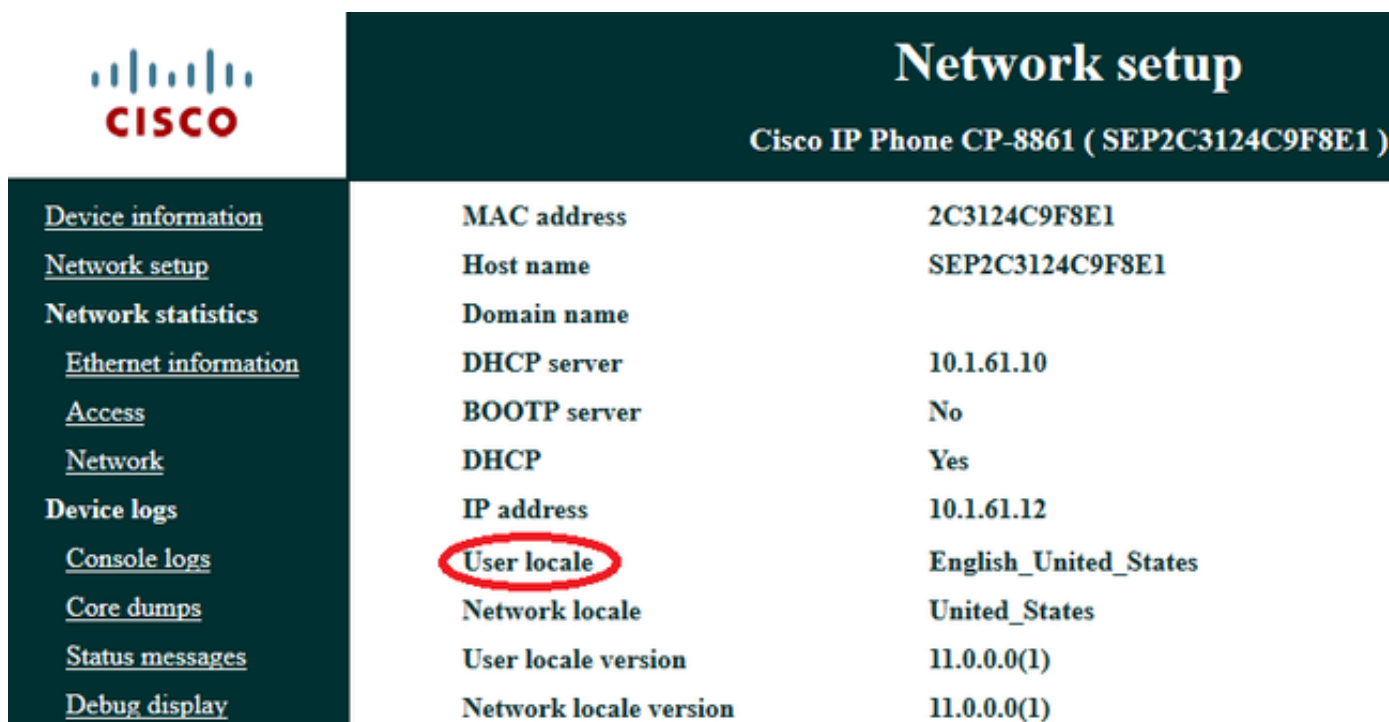
> **附註**：每個Unified CM集群支援單個思科呼叫伺服器。

HTTP與JTAPI

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

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

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

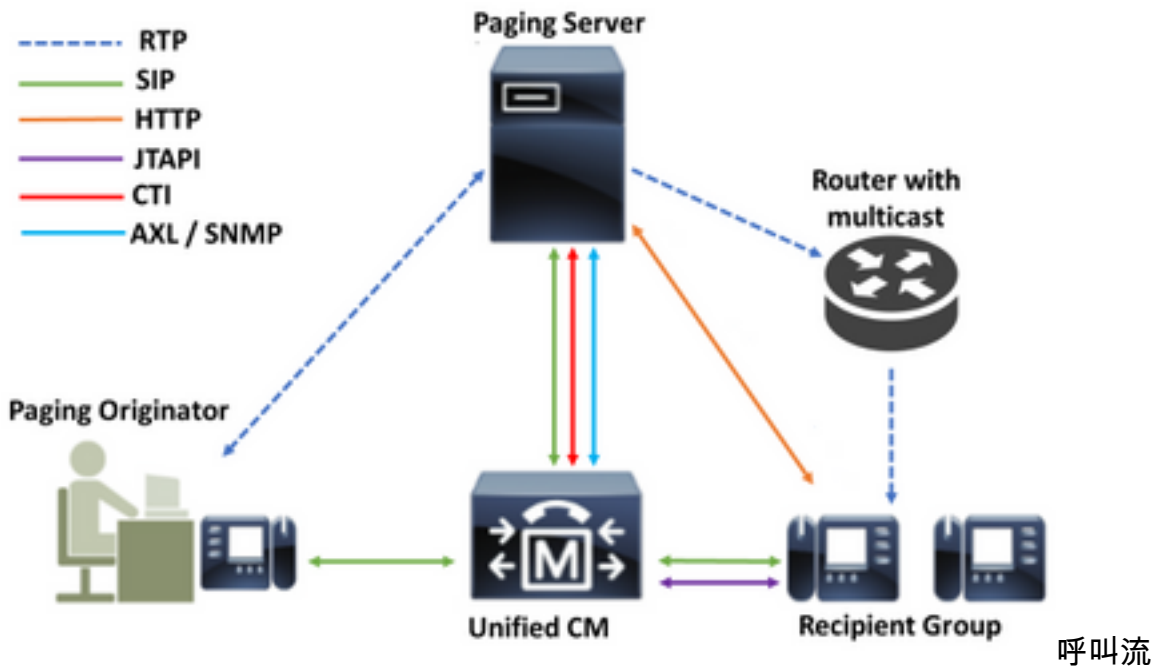> **附註**：要使用JTAPI，請考慮CUCM版本必須是9.1.2或更高版本，並且不支援Cisco 3905、7902、7905、7912電話。

SIP與CTI

Informacast可以通過CTI和/或SIP接收呼叫。對於CTI，呼叫在CTI路由點上處理（思科尋呼伺服器不需要CTI埠來應答入站呼叫）。

對於SIP，呼叫在SIP中繼上離開Unified CM。CTI和SIP均有效且受支援。但是，思科建議通過CTI進行SIP呼叫流，因為SIP整合故障排除比CTI容易得多。

組態

網路圖表

呼叫流

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

InformaCast使用JTAPI庫和電腦電話整合(CTI)管理器與Cisco Call Manager整合時，它使用TCP上的快速緩衝區編碼(QBE)協定，如下圖所示。
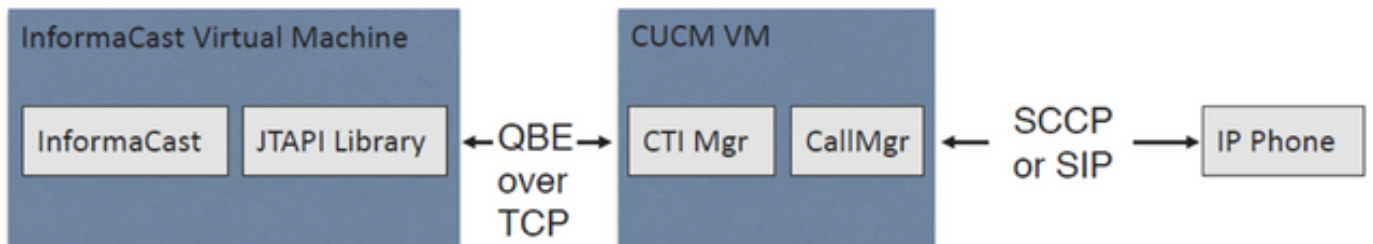
對於SIP整合，InformaCast使用TCP上的SIP協定和埠5060與Call Manager進行通訊，如下圖所示。
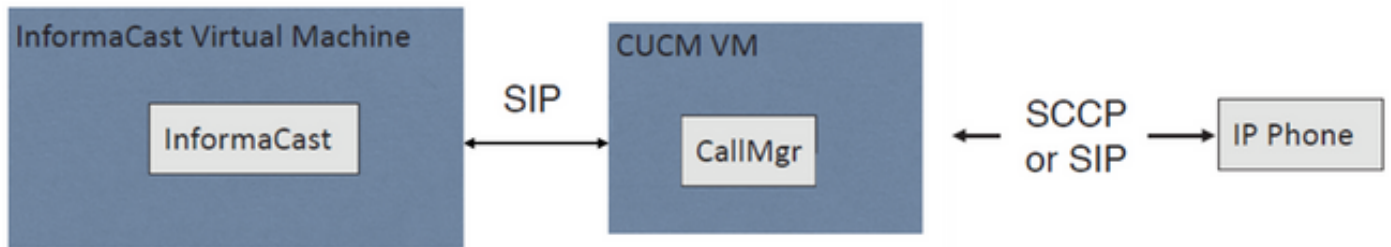
配置呼叫管理器

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

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

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

步驟2.設定SNMP（版本2或版本3）

對於SNMP v2

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

對於SNMP v3

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

Status

ⓘ Status : Ready

Server* | 10.1.61.158--CUCM Voice/Video ▼ |

User Information

User Name* | ICVA |

Authentication Information

☑ Authentication Required

Password | •••••••• | Reenter Password | •••••••• | Protocol ○ MD5 ● SHA

Privacy Information

☑ Privacy Required

Password | •••••••• | Reenter Password | •••••••• | Protocol ○ DES ● AES128

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 to configure Notification Destinations.

☑ Apply To All Nodes

[Save] [Clear All] [Cancel]

步驟3.將預設編解碼器設定為G.711

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

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

步驟4.建立裝置池

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

步驟5.建立路由分割槽，例如ICVA_PT。

步驟6.建立呼叫搜尋空間，例如ICVA_CSS。包括ICVA_PT。

步驟7.建立訪問控制組(AXL)。

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

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

步驟8.建立應用程式使用者

- 導航到CM Admin > User Management > Application User，然後點選Add New。將應用程式使 用者命名為ICVA_InformaCast，並分配以下角色：

1. 已啟用標準CTI
2. ICVA使用者組（或標準AXL API訪問）
3. 標準CTI允許控制支援連線的交換機和會議。
4. 標準CTI允許對支援翻滾模式的電話進行控制
5. 標準CTI允許控制所有裝置

Application User Configuration

💾 Save ❌ Delete 📄 Copy ➕ Add New

┌─ Application User Information ─────────

User ID* `ICVAInformacast` **Edit Credential**

Password `●●●●●●●●●●●●●●●●●●●●●●●●●●●●●●●`

Confirm Password `●●●●●●●●●●●●●●●●●●●●●●●●●●●●●●●`

Digest Credentials `[            ]`

Confirm Digest Credentials `[            ]`

BLF Presence Group* `Standard Presence group ⌄`

User Rank* `1-Default User Rank ⌄`

┌─ Permissions Information ─────────

Groups
```
ICVA User Group
Standard CTI Allow Control of All Devices
Standard CTI Allow Control of Phones supporting C
Standard CTI Allow Control of Phones supporting R
Standard CTI Enabled
```
View Details

Add to Access Control Group
Remove from Access Control Group

Roles
```
Standard AXL API Access
Standard CTI Allow Control of All Devices
Standard CTI Allow Control of Phones supporting Conn
Standard CTI Allow Control of Phones supporting Rollo
Standard CTI Enabled
```
View Details

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

步驟9.使用SIP或CTI將通訊管理器與Informacast整合。

對於SIP整合，請建立SIP配置檔案、SIP中繼和路由模式。

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

Trunk Configuration

💾 Save ❌ Delete 🔄 Reset ➕ Add New

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	ICVA_SipTrunk
Description	10.1.61.118
Device Pool*	ICVA_DP
Common Device Configuration	< None >
Call Classification*	Use System 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

	Destination Address	Destination Address IPv6	Destination Port
1*	10.1.61.118		5060

MTP Preferred Originating Codec*	711ulaw
BLF Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	ICVA SIP Profile View Details
DTMF Signaling Method*	No Preference

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

- 取消選中**Provide Outside Dial Tone**覈取方塊，然後按一下**Save**。

對於CTI整合，請建立CTI路由點並關聯到步驟8中建立的應用程式使用者。

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

將CTI路由點新增為ICVA應用使用者配置上的受控裝置。

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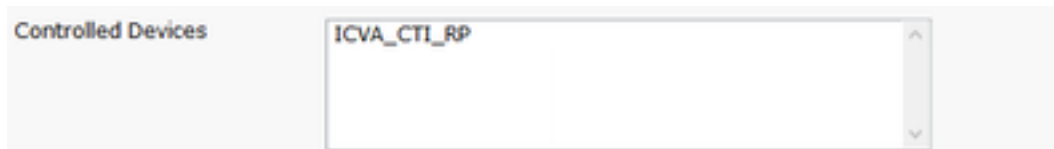
> **附註**：如果在通訊管理器中建立多個CTI路由點並將其與InformaCast應用程式使用者相關聯，則InformaCast可以支援這些CTI路由點。

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

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

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

Enterprise Phone Configuration

Save

Web Access*	Enabled	☑
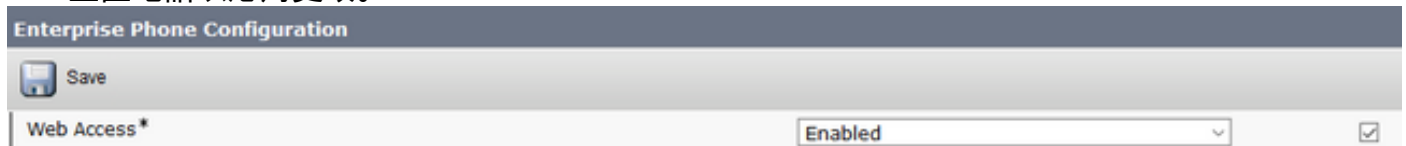
步驟11.設定身份驗證URL。

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

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

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

步驟12.設定API瀏覽器訪問的身份驗證方法。

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

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

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

配置Informacast

步驟1.在Informacast中配置Communications Manager群集。

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

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

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

步驟2.配置收件人組。

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

- 若要建立新的收件人組，請按一下Add，寫入名稱，然後按一下Edit為此收件人組新增電話。將 電話新增到收件人後，按一下Submit。
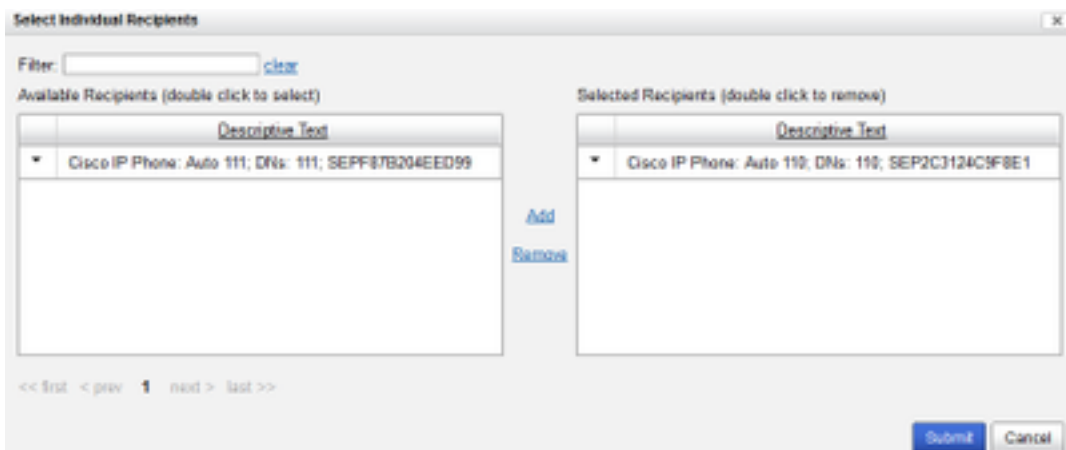
- 要儲存更改，請按一下Update。
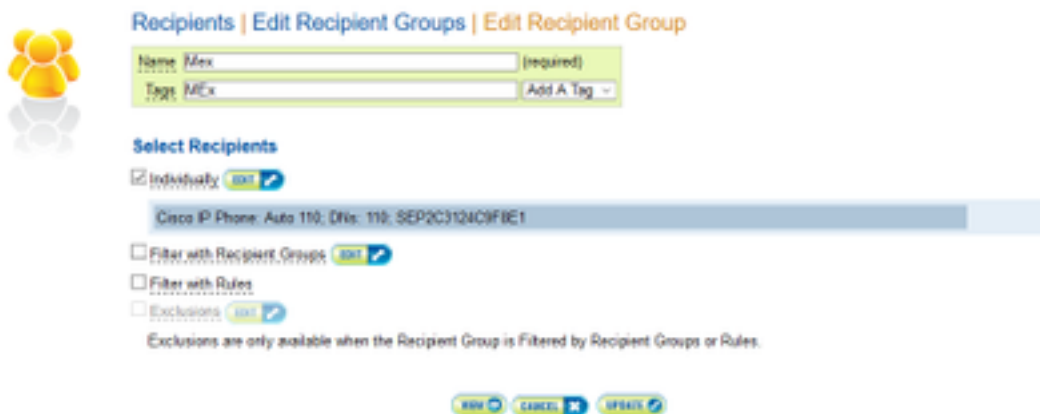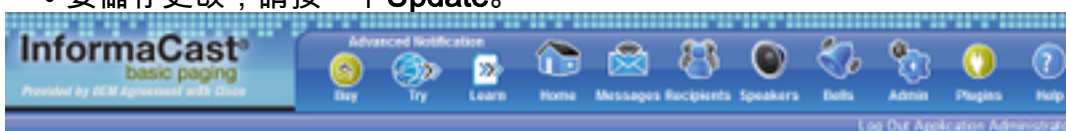
步驟3.允許/禁用對InformaCast的SIP訪問。

- 導航到Admin > SIP > SIP Access。預設情況下，所有SIP呼叫均被拒絕。
- 選擇Allow單選按鈕允許所有SIP呼叫，或按一下Add允許此允許的例外。
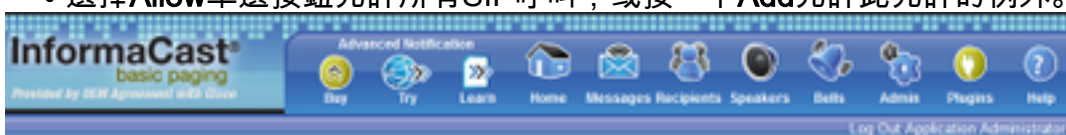
> 提示：定義例外時，請確保指定直接向InformaCast傳送INVITE請求的主機。如果代理位於InformaCast和呼叫主機之間，則此伺服器可能是SIP代理伺服器。

步驟4.新增廣播撥號配置

- 轉至Admin > DialCast > Dialing Configurations，點選Add

- 根據在CUCM中建立的路由模式（用於SIP整合）或CTI路由點（用於CTI整合），在**Dialing Pattern**欄位中輸入撥號模式（例如7777、7778）。
- 從清單中選擇收件人組，然後按一下**Update**。
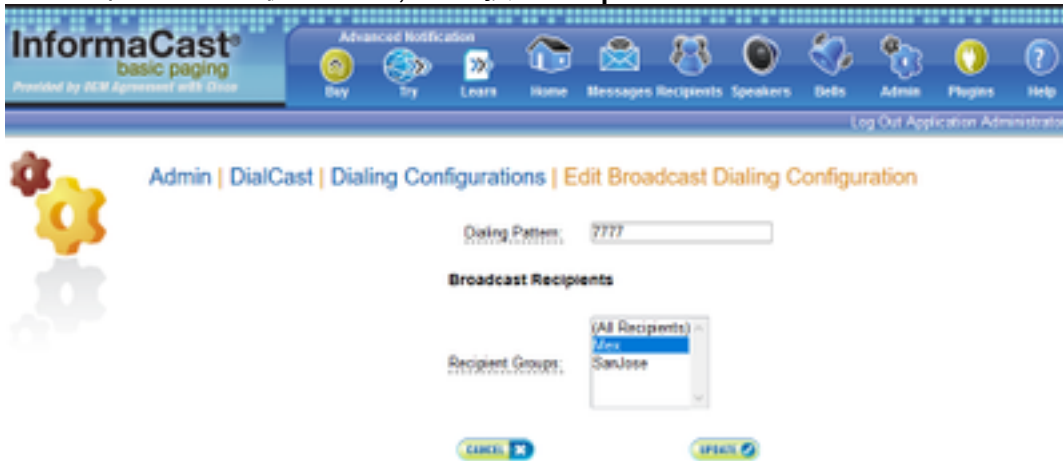
步驟5.配置廣播引數。

- 導航到**Admin > Broadcast Parameters**
- 配置組播的IP。通常使用預設IP(239.0.1.2)。
- 如果要將命令以JTAPI形式傳送到電話，請啟用JTAPI覈取方塊，否則將使用HTTP消息。
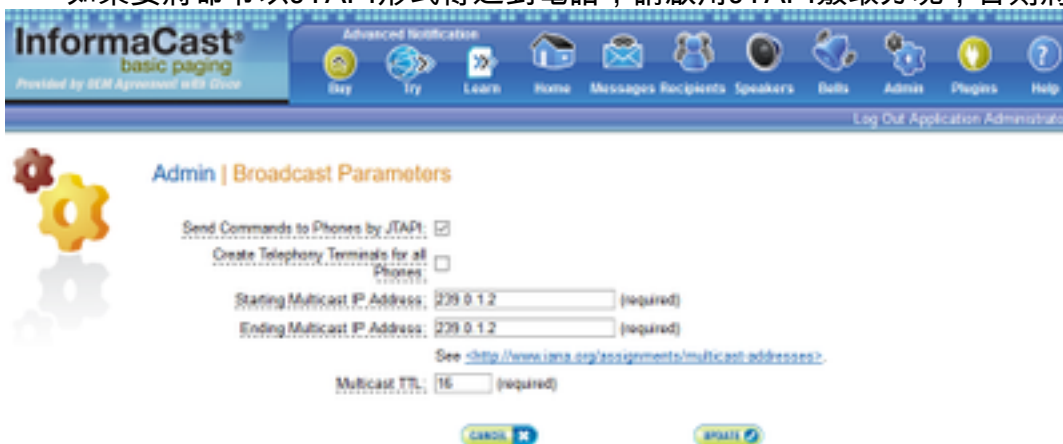
確保此範圍對應於您的網路基礎設施設定並涵蓋所有收件人組。在多站點部署中，Singlewire和思科建議使用一系列地址。此範圍應足夠大，以便為每個同時廣播處理一個地址。

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

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

配置網路中的組播

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

思科呼叫伺服器不需要任何特定的組播路由方法（SM、DM、S-DM、SSM等）。 某些廣域網環境不支援組播路由。對於這些環境，可以在站點之間構建GRE隧道並用於傳輸組播。

在您的環境中設計和配置組播不屬於本文檔的範圍，但您可能會發現以下資源很有用：

- [組播白皮書](#)
- [多點傳送測試工具](#)

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

驗證

目前沒有適用於此組態的驗證程序。

疑難排解

常見問題

電話未啟用

考慮到Informacast會在廣播發生時跳過任何正在使用（忙）的電話。

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

HTTP:繁忙檢測僅適用於運行英文載入的電話區域設定

CTI:適用於非英語電話區域設定

忙碌檢測根據協定、線路型別和線路狀態也可以不同地工作。

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

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

對未啟用的電話進行故障排除時，應收集以下資料：

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

未發現電話

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

SNMP錯誤無法建立收件人組：java.lang.Exception

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

目標電話無音訊

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

要收集的資料

在對Informacast進行故障排除時，您應收集以下資料：

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

效能日誌

有兩種方法可以從Informacast獲取效能日誌。

方法1

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

方法2

1. 在Web瀏覽器中開啟Informacast IP(https://<informacast_IP>)，然後選擇**Informacast**。
2. 使用您的憑據登入並導航到**幫助>支援**。
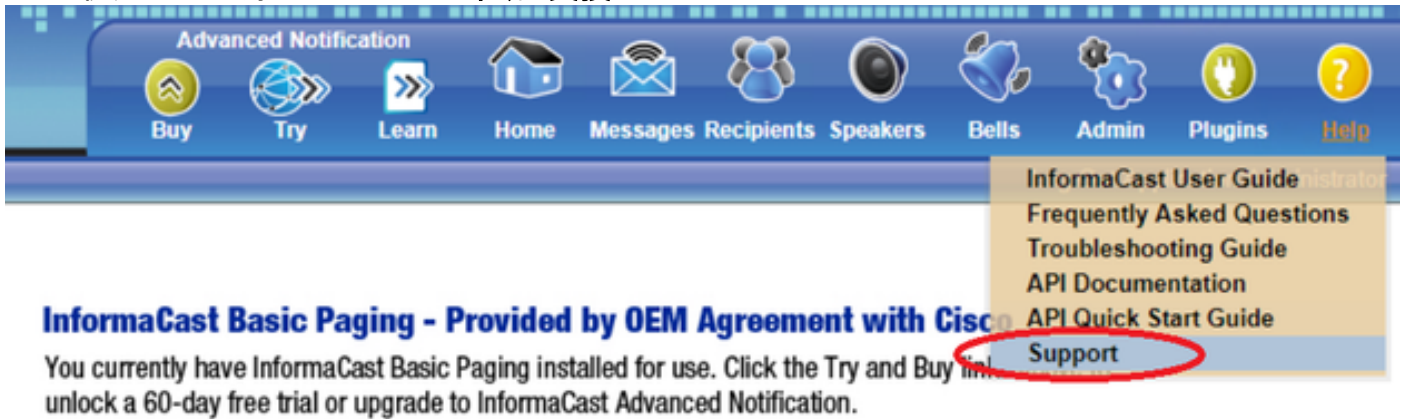
3.按一下「Tools（工具）」部分下的「**Performance Logs**」，如下圖所示。

封包擷取

從Informacast

有三種方法可以從Informacast獲取資料包捕獲。

方法1

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

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

方法2

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

```
Administrator: Singlewire Software: InformaCast Log Tool - 20150707          _ □ X

Singlewire Software


          [1] Gather Logs From InformaCast Server
          [2] Uncompress Gathered Logs
          [3] Parse Logs For Errors and Solutions
          [4] Turn JTAPI Debugging On/Off
          [5] Network Traffic Capture
          [9] Exit


Menu Choice......: 5

Server IP: 172.16.3.221
Username: admin
Password:

Attempting to Access 172.16.3.221 via SSH

Seconds Capture Should Run [1-300]: _
```

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

```
Seconds Capture Should Run [1-300]: 60

              [1] Start 60 Second Capture
              [9] Return to Main Menu

Menu Choice......: 1
Traffic Capture Started:  60 Seconds

00:00:51
```

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

InformaCast_LogTool.exe

InformaCast_LogTool_Logs_201809231605.tgz

方法3（12.0.1及更高版本中提供）

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

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

4. 當問題完全複製時按一下Stop Packet Capture，或在捕獲33,000個資料包後自行停止。
5. 導航到System > Collect Logs，輸入問題的簡短說明，然後按一下Collect a new set logs。
6. 若要儲存日誌，請按一下Download to Your Computer，如下圖所示。
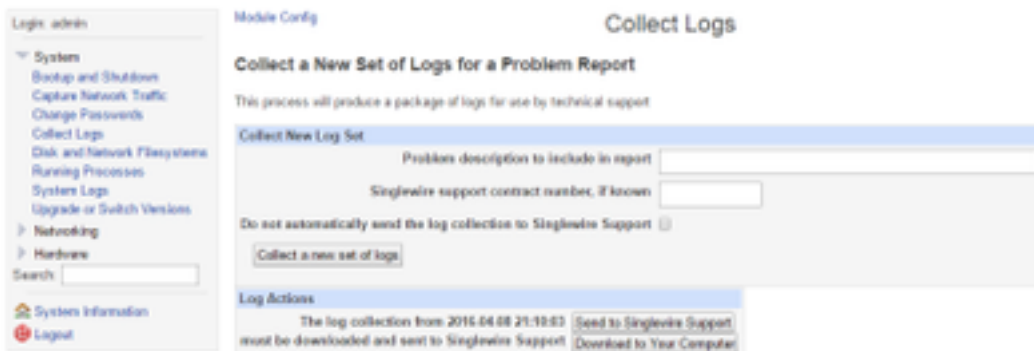
方法4（12.0.1及更新版本中提供）

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

```
admin@informacast:~$ capture-packets test
Saving up to 33000 packets to /var/log/capture-packets/test
tcpdump: listening on eth0, link-type EN10MB (Ethernet), capture size 1514
bytes
^C13 packets captured
15 packets received by filter
0 packets dropped by kernel
```

```
Interrupt signaled. Cleaning up.
```

附註：GUI方法比CLI更好，因為不需要依賴SFTP伺服器，而且您可以從網頁啟動、停止和下載資料包捕獲。

來自CUCM

根據部署定義需要從何處獲取資料包捕獲。集群中只能有一個CUCM節點或多個CUCM。
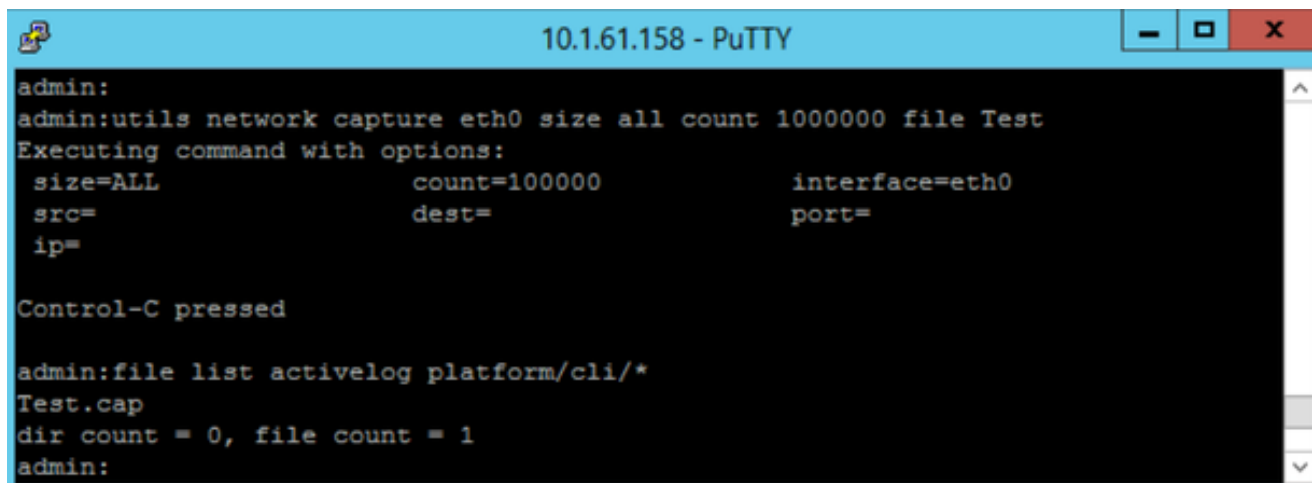
- 如果您有一個CUCM節點，請獲取資料包捕獲，如下圖所示。
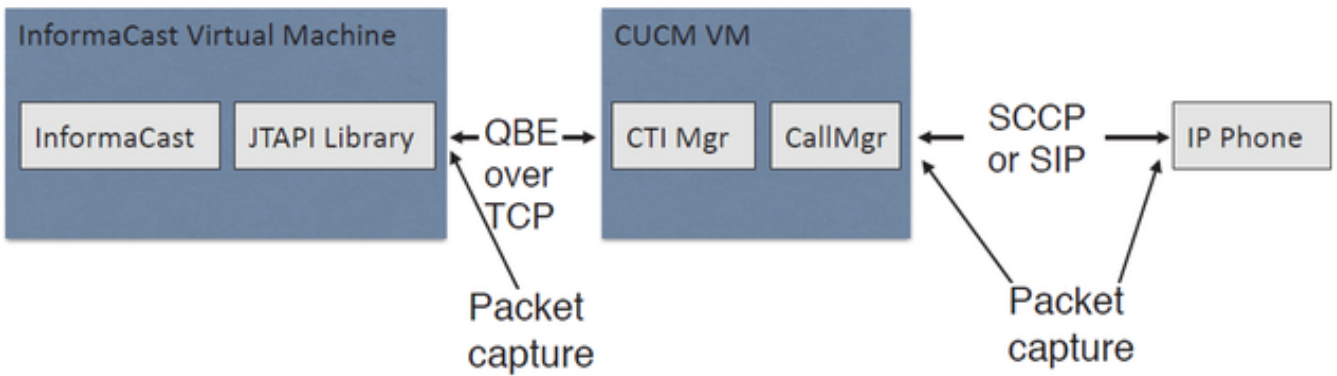
- 如果您有CUCM集群，並且一個節點正在與Informacast通訊，但另一個節點正在與電話通訊，則獲取資料包捕獲，如圖所示。
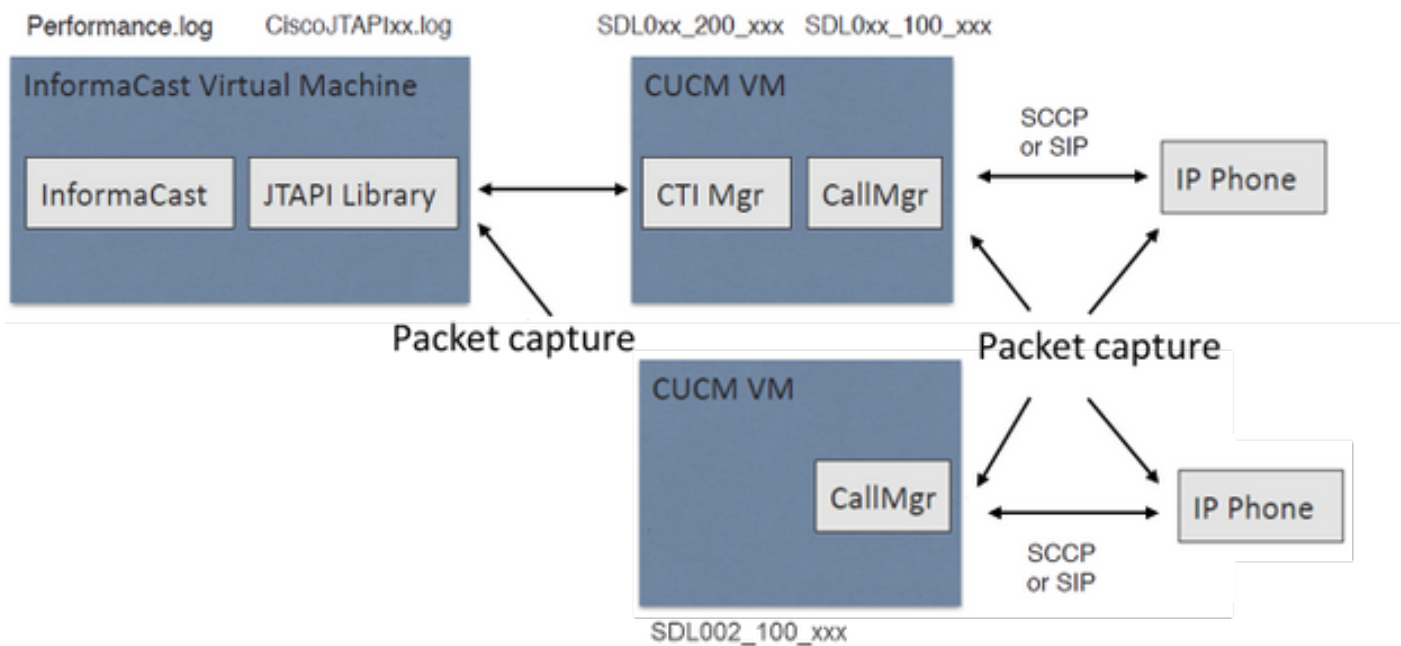
1. 為需要捕獲的節點開啟SSH會話
2. 運行命令**utils network capture eth0 size all count 1000000 file Test**以開始資料包捕獲。
3. 複製問題
4. 使用**Ctrl + C**停止資料包捕獲
5. 要確認資料包捕獲已儲存，請運行命令**file list activelog platform/cli/***

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


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

Download directory: /


.......
Transfer completed.
admin:
```

7. 如果無法使用SFTP伺服器，請使用RTMT。導覽至System > Trace & Log Central > Collect Files。按一下「Next」，然後啟用「Packet capture logs」覈取方塊，如下圖所示。
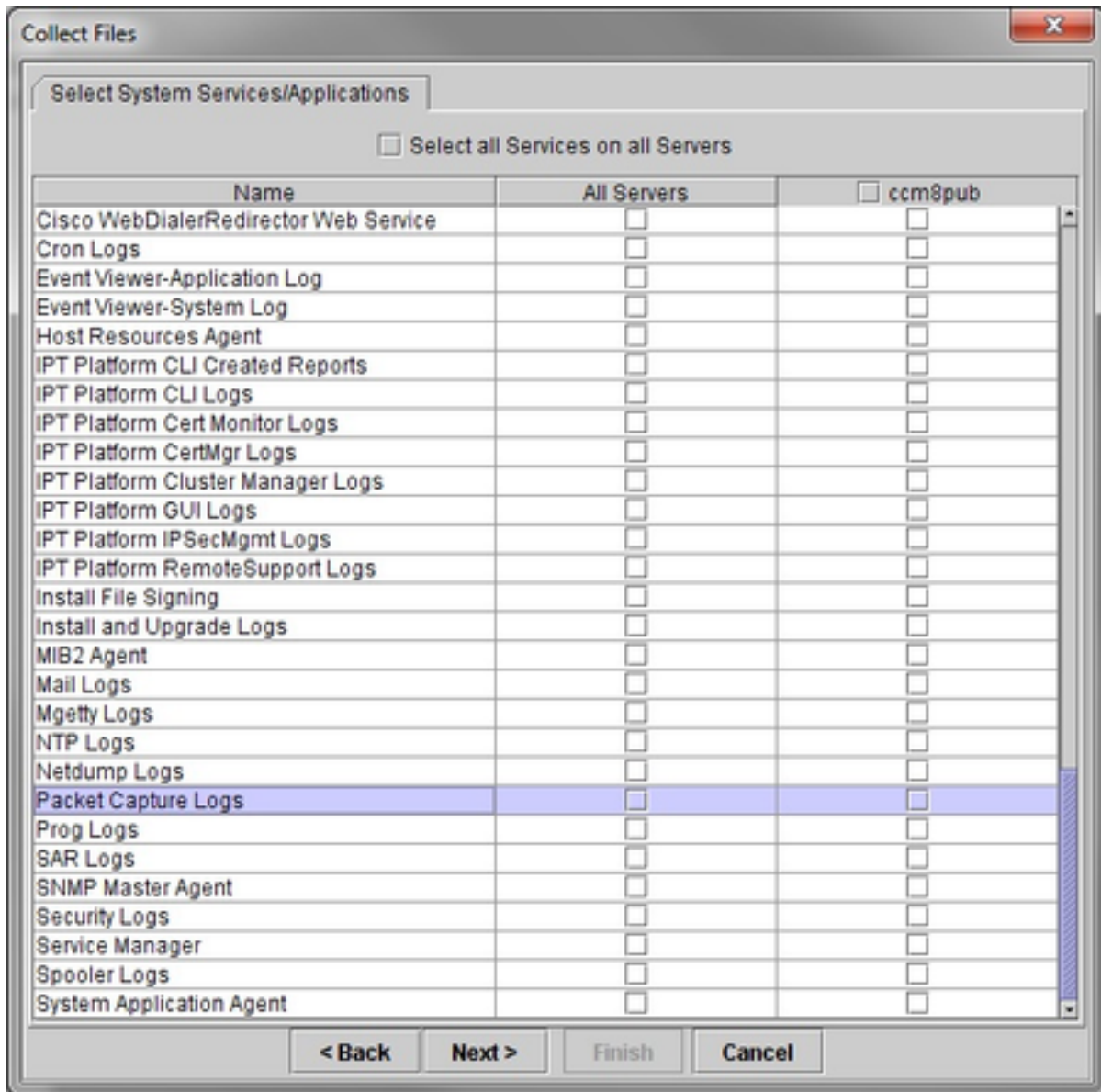
8. 按一下**下一步**，選擇一個下載檔案目錄，然後按一下**完成**。
9. 使用**file delete activelog platform/cli/Test.cap**命令刪除資料包

通過電話

1. 啟用SPAN到PC埠。導航到**CM Admin page > Device > phone**，然後查詢報告有問題的電話。
2. 在「產品特定配置佈局」部分下，找到**Span to PC Port**，然後從下拉選單中選擇**Enable**。按一下**Save**，然後按一下Apply config。
3. 將筆記型電腦連線到電話的PC埠。
4. 在筆記型電腦中運行資料包分析器軟體。您可以使用Wireshark（或其他封包擷取軟體）。
5. 重現問題。
6. 完全複製問題時，繼續停止資料包捕獲。

您可以在以下連結中找到更多詳細資訊
：https://supportforums.cisco.com/document/44741/collecting-packet-capture-cisco-ip-phone

範例分析
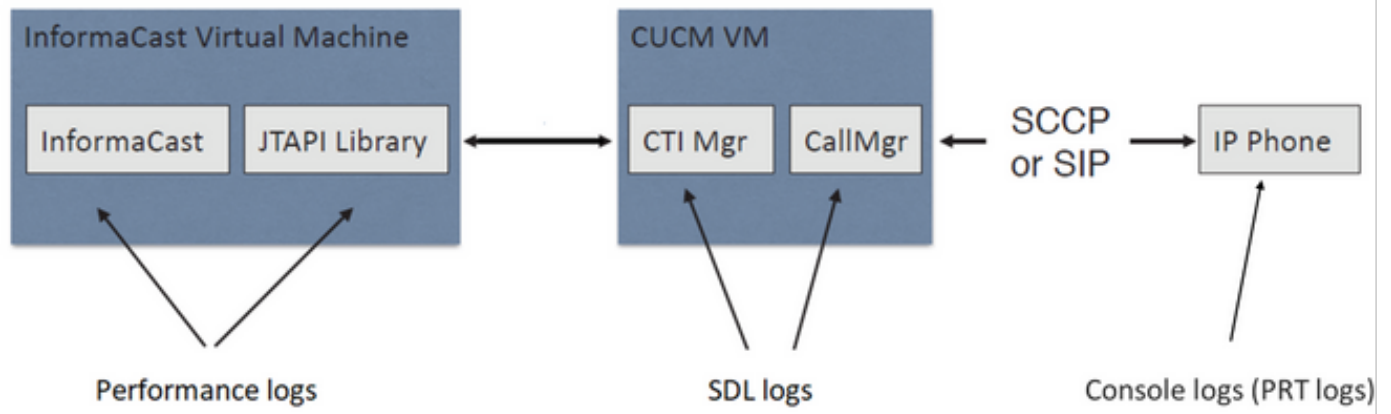
SDL跟蹤

用於SIP整合和由JTAPI控制的電話

CUCM:10.1.61.158

Informacast:10.1.61.118

電話A

DN:110

型號：CP-8861

韌體版本：sip88xx.12-0-1SR1-1

電話A IP地址：10.1.61.12

MAC SEP2C3124C9F8E1

電話B

DN:111

型號：CP-8811

韌體版本：sip88xx.12-0-1SR1-1

電話B IP地址：10.1.61.11

MAC SEPF87B204EED99

撥號號碼：7777

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

```
From: "PhoneA" <sip:110@10.1.61.158>;tag=2c3124c9f8e10c3c246b7956-5c62fa57
To: <sip:7@10.1.61.158>
Call-ID: 2c3124c9-f8e1000d-00337209-0547bb10@10.1.61.12
Max-Forwards: 70
Session-ID: 712c9e1f00105000a0002c3124c9f8e1;remote=00000000000000000000000000000000
Date: Tue, 10 Sep 2019 00:00:37 GMT
CSeq: 101 INVITE
User-Agent: Cisco-CP8861/12.0.1
Contact: <sip:142b9f25-7f2b-48a8-9ff9-
377f616f3084@10.1.61.12:51600;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP2C3124C9F8E1"
Expires: 180
Accept: application/sdp
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "PhoneA" <sip:110@10.1.61.158>;party=calling;id-
type=subscriber;privacy=off;screen=yes
Supported: replaces,join,sdp-anat,norefersub,resource-priority,extended-refer,X-cisco-
callinfo,X-cisco-serviceuri,X-cisco-escapecodes,X-cisco-service-control,X-cisco-srtp-fallback,X-
cisco-monrec,X-cisco-config,X-cisco-sis-7.0.0,X-cisco-xsi-8.5.1
Allow-Events: kpml,dialog
Recv-Info: conference
Recv-Info: x-cisco-conference
Content-Length: 548
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 11811 0 IN IP4 10.1.61.12
s=SIP Call
b=AS:4064
t=0 0
m=audio 22018 RTP/AVP 114 9 124 0 8 116 18 101
c=IN IP4 10.1.61.12
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114 maxplaybackrate=16000;sprop-
maxcapturerate=16000;maxaveragebitrate=64000;stereo=0;sprop-stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

CUCM performs digit analysis for the dialed digits (dd="7777")

```
71439203.000 |19:00:36.580 |SdlSig   |DaReq                                 |wait
|Da(1,100,216,1)              |Cdcc(1,100,224,6)
|1,100,14,1368.16^10.1.61.12^*            |[R:N-H:0,N:0,L:0,V:0,Z:0,D:0]  CI=19282342
Fqdn=ti=1nd=110pi=0si1 Cgpn=tn=0npi=0ti=1nd=110pi=1si1
DialedNum=tn=0npi=1ti=1nd=7777User=7777Host=10.1.61.158Port=5060PassWord=Madder=Transport=4mDisp
layName=RawUrl=sip:7@10.1.61.158;user=phoneOrigPort=0pi=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
Session-ID: 712c9e1f00105000a0002c3124c9f8e1;remote=00000000000000000000000000000000
Cisco-Guid: 0047656832-0000065536-0000000001-2654798090
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=0
o=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 0
m=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;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=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=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;id-
type=subscriber;privacy=off;screen=yes
```

```
Content-Length: 0
Recv-Info: conference
Recv-Info: x-cisco-conference
```

Since integration is with JTAPI, CUCM sends REFER to the phone with instructions to join to the IP and port of multicast

```
71439541.002 |19:00:38.199 |AppInfo   |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to
10.1.61.11 on port 51784 index 25768
[431557,NET]
REFER sip:e2881942-2853-4eab-a0d9-96228c79d062@10.1.61.11:51784;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.1.61.158:5060;branch=z9hG4bK19970687ccd2b
From: <sip:111@10.1.61.158>;tag=1598606730
To: <sip:111@10.1.61.11>
Call-ID: 4085c80-d761e7a6-1996d-9e3d010a@10.1.61.158
CSeq: 101 REFER
Max-Forwards: 70
Contact: <sip:111@10.1.61.158:5060;transport=tcp>
User-Agent: Cisco-CUCM11.5
Expires: 30
Refer-To: cid:1234567890@10.1.61.158
Content-Id: <1234567890@10.1.61.158>
Content-Type: multipart/mixed;boundary=uniqueBoundary
Mime-Version: 1.0
Referred-By: <sip:111@10.1.61.158>
Content-Length: 682


--uniqueBoundary
Content-Type:application/x-cisco-remotecc-request+xml
<x-cisco-remotecc-request>
<datapassthroughreq>
<applicationid>0</applicationid>
<lineid>0</lineid>
<transactionid>109</transactionid>
<stationsequence>StationSequenceLast</stationsequence>
<displaypriority>2</displaypriority>
<appinstance>0</appintance>
<routingid>0</routingid>
<confid>0</confid>
<featuredata></featuredata>
</datapassthroughreq>

</x-cisco-remotecc-request>
--uniqueBoundary
Content-Type:application/x-cisco-remote-cm+xml
<CiscoIPPhoneExecute><ExecuteItem URL="RTPMRx:239.0.1.2:20480"/></CiscoIPPhoneExecute>
--uniqueBoundary--
```

Phone B replies with 202 Accepted

```
71439542.002 |19:00:38.215 |AppInfo   |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from
10.1.61.11 on port 51784 index 25768 with 571 bytes:
[431558,NET]
SIP/2.0 202 Accepted
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>;tag=f87b204eed990c3a4020c613-5969341f
Call-ID: 4085c80-d761e7a6-1996d-9e3d010a@10.1.61.158
Session-ID: f9d4984b00105000a000f87b204eed99;remote=00000000000000000000000000000000
Date: Tue, 10 Sep 2019 00:00:40 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
```

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
Session-ID: f9d4984b00105000a000f87b204eed99;remote=00000000000000000000000000000000
Date: Tue, 10 Sep 2019 00:00:40 GMT
CSeq: 1000 NOTIFY
Event: refer
Subscription-State: terminated; reason=timeout
Max-Forwards: 70
Contact: <sip:e2881942-2853-4eab-a0d9-
96228c79d062@10.1.61.11:51784;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEPF87B204EED99"
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE
Content-Type: multipart/mixed; boundary=uniqueBoundary
Mime-Version: 1.0
Content-Length: 1199

--uniqueBoundary
Content-Type:application/x-cisco-remotecc-response+xml
Content-Disposition_session;handling=required

<?xml version=1.0" enconding="UTF-8"?>
<x-cisco-remotecc-response>
<response>
<code>200</code>
<reason></reason>
<applicationid>0</applicationid>
<transactionid>109</transactionid>
<stationsequence>StationSequenceLast</stationsequence>
<displaypriority>2</displaypriority>
<appinstance>0</appintance>
<linenumber>0</linenumber>
<routingid>0</routingid>
<confid>0</confid>
<callid></callid>

<options_ind>
<combine max="0">
<service-control></service-control>
</combine>

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

</options_ind>
</response>
</x-cisco-remotecc-response>
--uniqueBoundary
Content-Type:application/x-cisco-remote-cm+xml
Csontent-Disposition:session;handling=required
<?xml version="1.0" encoding="utf-8"?>
<CiscoIPPhoneResponse>
<ResponseItem URL="RTPMRx:239.0.1.2:20480" Data="Success" Status="0"/>
```

```
</CiscoIPPhoneResponse>
--uniqueBoundary--
```

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

CUCM sends to the phone B a REFER to stop receiving multicast audio
```
71442357.002 |19:01:10.795 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to
10.1.61.11 on port 51784 index 25768
[431582,NET]
REFER sip:e2881942-2853-4eab-a0d9-96228c79d062@10.1.61.11:51784;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.1.61.158:5060;branch=z9hG4bK199754588a6e3
From: <sip:111@10.1.61.158>;tag=928499252
To: <sip:111@10.1.61.11>
Call-ID: 171b2c80-d761e7c6-19970-9e3d010a@10.1.61.158
CSeq: 101 REFER
Max-Forwards: 70
Contact: <sip:111@10.1.61.158:5060;transport=tcp>
User-Agent: Cisco-CUCM11.5
Expires: 30
Refer-To: cid:1234567890@10.1.61.158
Content-Id: <1234567890@10.1.61.158>
Content-Type: multipart/mixed;boundary=uniqueBoundary
Mime-Version: 1.0
Referred-By: <sip:111@10.1.61.158>
Content-Length: 683

--uniqueBoundary
Content-Type:application/x-cisco-remotecc-request+xml
<x-cisco-remotecc-request>
<datapassthroughreq>
<applicationid>0</applicationid>
<lineid>0</lineid>
<transactionid>109</transactionid>
<stationsequence>StationSequenceLast</stationsequence>
<displaypriority>2</displaypriority>
<appinstance>0</appintance>
<routingid>0</routingid>
<confid>0</confid>
<featuredata></featuredata>
</datapassthroughreq>

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

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

```
Via: SIP/2.0/TCP 10.1.61.158:5060;branch=z9hG4bK199754588a6e3
From: <sip:111@10.1.61.158>;tag=928499252
To: <sip:111@10.1.61.11>;tag=f87b204eed990c3e1c1bfe96-1d092704
Call-ID: 171b2c80-d761e7c6-19970-9e3d010a@10.1.61.158
Session-ID: f9d4984b00105000a000f87b204eed99;remote=00000000000000000000000000000000
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
Session-ID: f9d4984b00105000a000f87b204eed99;remote=00000000000000000000000000000000
Date: Tue, 10 Sep 2019 00:01:13 GMT
CSeq: 1000 NOTIFY
Event: refer
Subscription-State: terminated; reason=timeout
Max-Forwards: 70
Contact: <sip:e2881942-2853-4eab-a0d9-
96228c79d062@10.1.61.11:51784;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEPF87B204EED99"
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE
Content-Type: multipart/mixed; boundary=uniqueBoundary
Mime-Version: 1.0
Content-Length: 1187


--uniqueBoundary
Content-Type:application/x-cisco-remotecc-request+xml
Content-Disposition:session;handling=required

<?xml version="1.0" encoding="UTF-8"?>
<x-cisco-remotecc-response>
<response>
<code>200</code>
<reason></reason>
<applicationid>0</applicationid>
<transactionid>117</transactionid>
<stationsequence>StationSequenceLast</stationsequence>
<displaypriority>2</displaypriority>
<appinstance>0</appinstance>
<linenumber>0</linenumber>
<routingid>0</routingid>
<confid>0</confid>
<callid></callid>
<options_ind>
 <combine max="0">
  <service-control></service-control>
</combine>
 <dialog usage="">
  <unot></unot>
    <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
```

用於CTI整合和由HTTP控制的電話

CUCM:10.1.61.158

Informacast:10.1.61.118

電話A

DN:110

型號:CP-8861

韌體版本:sip88xx.12-0-1SR1-1

電話A IP地址:10.1.61.12

MAC:SEP2C3124C9F8E1

電話B

DN:111

型號:CP-8811

韌體版本:sip88xx.12-0-1SR1-1

電話B IP地址:10.1.61.11

MAC:SEPF87B204EED99

撥號號碼：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
Session-ID: 02023b9b00105000a0002c3124c9f8e1;remote=00000000000000000000000000000000
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;id-
type=subscriber;privacy=off;screen=yes
Supported: replaces,join,sdp-anat,norefersub,resource-priority,extended-refer,X-cisco-
callinfo,X-cisco-serviceuri,X-cisco-escapecodes,X-cisco-service-control,X-cisco-srtp-fallback,X-
cisco-monrec,X-cisco-config,X-cisco-sis-7.0.0,X-cisco-xsi-8.5.1
Allow-Events: kpml,dialog
Recv-Info: conference
Recv-Info: x-cisco-conference
Content-Length: 548
Content-Type: application/sdp
Content-Disposition: session;handling=optional
v=0
o=Cisco-SIPUA 19108 0 IN IP4 10.1.61.12
s=SIP Call
b=AS:4064
t=0 0
m=audio 19104 RTP/AVP 114 9 124 0 8 116 18 101
c=IN IP4 10.1.61.12
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114 maxplaybackrate=16000;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
|wait |Da(1,100,216,1)
|Cdcc(1,100,224,12) |1,100,14,1368.88^10.1.61.12^* |[R:N-
H:0,N:0,L:0,V:0,Z:0,D:0] CI=19282358 Fqdn=ti=1nd=110pi=0si1 Cgpn=tn=0npi=0ti=1nd=110pi=1si1
DialedNum=tn=0npi=1ti=1nd=7778User=7778Host=10.1.61.158Port=5060PassWord=Madder=Transport=4mDisp
layName=RawUrl=sip:7@10.1.61.158;user=phoneOrigPort=0pi=0si1 requestID=0

```
DigitAnalysisComplexity=1 CallingUser= IgnoreIntercept=0 callingDeviceName=SEP2C3124C9F8E1
71531367.001 |19:15:34.231 |AppInfo  |Digit Analysis: star_DaReq:
daReq.partitionSearchSpace(8653f609-05a7-5914-819b-3a89680af6a2:),
filteredPartitionSearchSpaceString(Informacast_PT:phone_pt),
partitionSearchSpaceString(Informacast_PT:phone_pt)
71531367.002 |19:15:34.231 |AppInfo  |Digit Analysis: Host Address=10.1.61.158 MATCHES this
node's IPv4 address.
71531367.003 |19:15:34.231 |AppInfo  |Digit Analysis: star_DaReq: Matching SIP URL, Numeric
User, user=7778
71531367.004 |19:15:34.232 |AppInfo  |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept
DAMR.sstype=[0], TPcount=[0], DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]
71531367.005 |19:15:34.232 |AppInfo  |Digit Analysis: getDaRes - Remote Destination [] isURI[1]
71531367.006 |19:15:34.232 |AppInfo  |Digit analysis: patternUsage=2
71531367.007 |19:15:34.232 |AppInfo  |Digit analysis: match(pi="2", fqcn="110",
cn="110",plv="5", pss="Informacast_PT:phone_pt", TodFilteredPss="Informacast_PT:phone_pt",
dd="7778",dac="1")
71531367.008 |19:15:34.232 |AppInfo  |Digit analysis: analysis results
71531367.009 |19:15:34.232 |AppInfo  ||PretransformCallingPartyNumber=110
|CallingPartyNumber=110
|DialingPartition=Informacast_PT
|DialingPattern=7778
|FullyQualifiedCalledPartyNumber=7778
|DialingPatternRegularExpression=(7778)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=7778
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=7778
|CollectedDigits=7778
|UnconsumedDigits=
|TagsList=SUBSCRIBER
|PositionalMatchList=7778
|VoiceMailbox=
|VoiceMailCallingSearchSpace=
|VoiceMailPilotNumber=
|RouteBlockFlag=RouteThisPattern
|RouteBlockCause=0
|AlertingName=InformacastCTIRP
|UnicodeDisplayName=InformacastCTIRP
|DisplayNameLocale=1
|OverlapSendingFlagEnabled=0
|WithTags=
|WithValues=
|CallingPartyNumberPi=NotSelected
|ConnectedPartyNumberPi=NotSelected
|CallingPartyNamePi=NotSelected
|ConnectedPartyNamePi=NotSelected
|CallManagerDeviceType=NoDeviceType
|PatternPrecedenceLevel=Routine
|CallableEndPointName=[4db482c3-64c3-5adf-33c5-a11c890d96d0]
|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-O |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:N-
H:0,N:1,L:0,V:0,Z:0,D:0]  LH=1|47 GCH=1|15018 CH=1|19282359 Held CH=0|0
State=2(CtiOfferingState) Reason=1 Origin=1 DeviceName=ICVA_CTI_RP CGPN=[ DN=110 uDN=110 NumPI=T
Part=phone_pt VmBox= NumType=0 Name=PhoneA UniName=PhoneA NamePI=T Locale=1 PU=2
Device=SEP2C3124C9F8E1 GlblCgpn=110] CDPN=[ DN=7778 uDN=7778 NumPI=T Part=Informacast_PT VmBox=
NumType=0 Name=InformacastCTIRP UniName=InformacastCTIRP NamePI=T Locale=1 PU=2 Device=] LRP=[
DN= uDN= NumPI=T Part= VmBox= NumType=0 Name= UniName= NamePI=T Locale=1] OCDPN=[ DN=7778
uDN=7778 NumPI=T Part=Informacast_PT VmBox= NumType=0 Name=InformacastCTIRP
UniName=InformacastCTIRP NamePI=T Locale=1] AuxData=T FarEndCMId=1 EndpointType=1 RIU=F
Privacy=F CallPresent=T FeatPriority=1 Feature=137 AttrType=0 LineId [DN=110 Part=phone_pt]
IPAddrMode=0 IsConsCallDueToRollover=F UniqCallRef=0000000000003AAA012639B700000000
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      ]    CTI
NewCallEvent  ( LH=1|46 CH=1|19282359  CH=0|0  GCH=1|15018 lineHandleSpecified=1 state=2
origin=1 farEndpointSpecified=1 farEndpointCMID=1 endpointType=1 reason=1 remote in use=0
privacy=0 mediaResourceID= resource ID=0 deviceName=ICVA_CTI_RP cgpn=110 Presentation=1 cgpn
NameInfo=locale: 1 pi: 1 Name: PhoneA UnicodeName: PhoneA cdpn=7778 Presentation=1 cdpn
NameInfo=locale: 1 pi: 1 Name: InformacastCTIRP UnicodeName: InformacastCTIRP original cdpn=7778
Presentation=1 original cdpn NameInfo=locale: 1 pi: 1 Name: InformacastCTIRP UnicodeName:
InformacastCTIRP LRP= Presentation=1 LRP NameInfo=locale: 1 pi: 1 Name:  UnicodeName:  UserData=
callingPartyDeviceName=SEP2C3124C9F8E1 mediaDeviceName= ucgpn=110 ucdpn=7778 unmodifiedOriginal
cdpn=7778 uLRP= cgPnPartition=phone_pt cdPnPartition=Informacast_PT
oCdPnPartition=Informacast_PT lrpPartition= CgpnIP=0xc3d010a IsConsultCallDueToRollover=0
apiCallReference=0000000000003AAA012639B700000000 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-O |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-O |CtiDeviceSetRTPForCallReq                 |NA
RemoteSignal                  |UnknownProcessName(1,100,66,16)  |CTIDeviceLineMgr(1,200,25,1)
|1,200,13,90.91^10.1.61.118^ICVA_CTI_RP    |[R:N-H:0,N:0,L:0,V:0,Z:0,D:0]
AsyncResponse=126mCtiInterface(1,200,25,1) DH=1|53 CH=1|19282359 RtpDestination1983709450|32080
```

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

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

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

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

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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 2d72f80-
d761e7a4-1996c-9e3d010a@10.1.61.158; from "PhoneA" <sip:110@10.1.61.158>; to
<sip:7777@10.1.61.118>; user-agent Cisco-CUCM11.5

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

Stream settings:
2019-09-09 19:09:43,556 [Signaler # 1 run 1] INFO Signaler [] - Stream settings:
  General info: User=dialcast(System User), BroadcastInitiator=10.1.61.12,
SourceType=CallingPhone, MessageKey=908, MessageType=Live Audio, MessageDescription=Basic Paging
Live Broadcast, RecipientGroupDescription=SanJose, MaxIPPhones=50, MaxIPSpeakers=0,
DeviceArbiter=null, CreatedOn=Mon Sep 09 19:09:42,849 CDT 2019, PauseLength=0,
NumberOfRepetitions=1
  Audio details: AudioFile=null, AudioFormat=ULAW 8000.0 Hz, 8 bit, mono, 1 bytes/frame, ,

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

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

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

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

Informacast receives the BYE to end the paging
2019-09-09 19:10:15,222 [pool-41-thread-1] INFO ba [] - Received BYE request; call ID 2d72f80-
d761e7a4-1996c-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" />
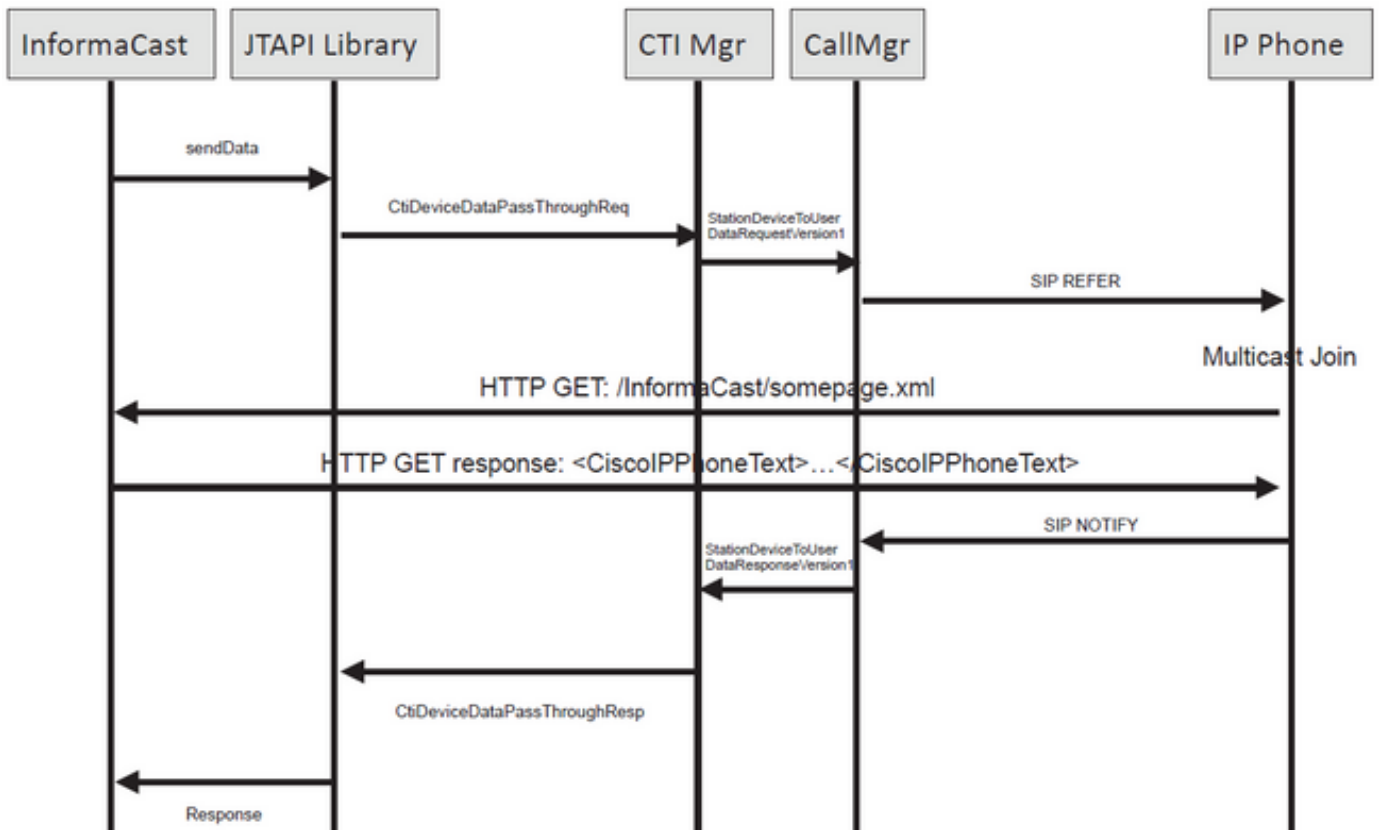</CiscoIPPhoneResponse>
Task ended
2019-09-09 19:10:19,357 [DeviceDeactivator-pool-1268-thread-1] INFO ah [1] - Canceling live
broadcast for inbound call 2d72f80-d761e7a4-1996c-9e3d010a@10.1.61.158
2019-09-09 19:11:45,250 [Timer-0] INFO JavaExchangeAdapter [] - Task Ended: checkpoint command
to compact the database

CTI整合

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

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

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

Informacast sends the message to all devices in the recipient group, in this case to only 1
device
2019-09-09 19:24:40,262 [Signaler # 4 run 1] INFO Signaler [] - Sending message to 1
participants

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

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

Informacast receives the response from the phone

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

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

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

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

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

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

控制檯日誌(PRT)

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

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

```
5326 NOT Sep 10 00:15:34.537528 (408:408) ms-RTPSESSION.createRTPSession media
[ipv4=239.0.1.2][port=20486][interface=NULL][mediatype=4][relayee=0][groupid=4294967295][callid=
4294967295]
```

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

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

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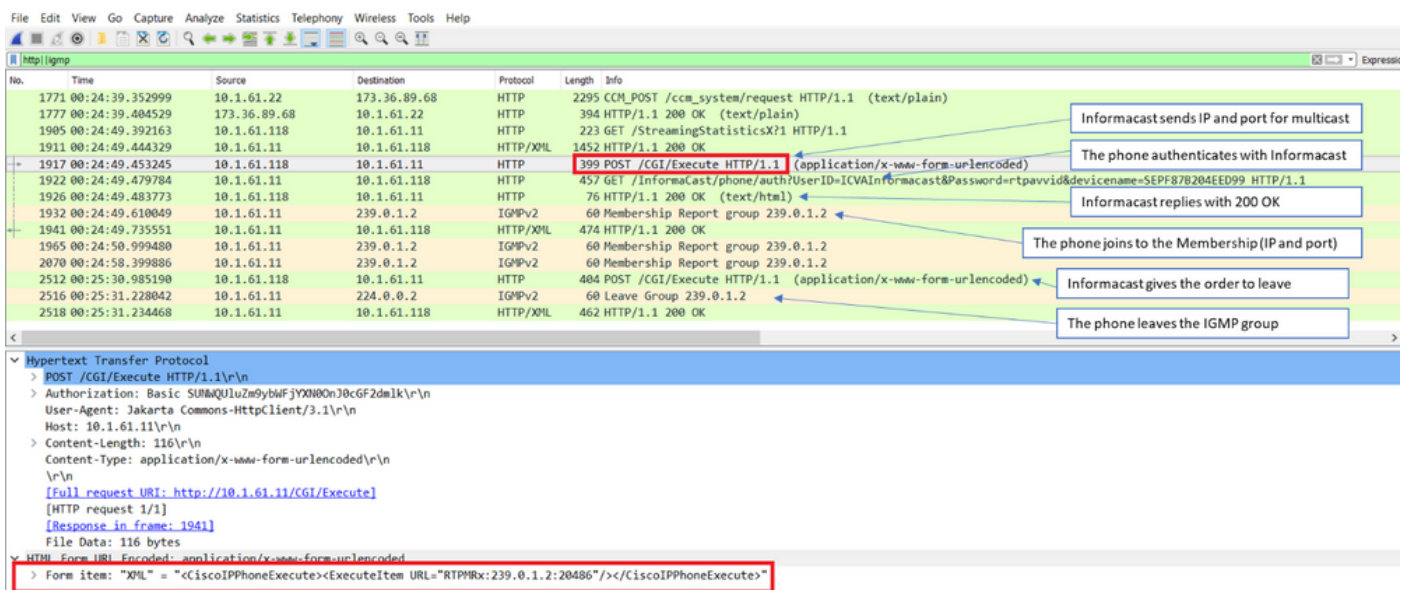
封包擷取

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

電話上的資料包捕獲(由HTTP控制)

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

電話上收到的HTTP和IGMP消息如下圖所示。

電話上的資料包捕獲(由JTAPI控制)

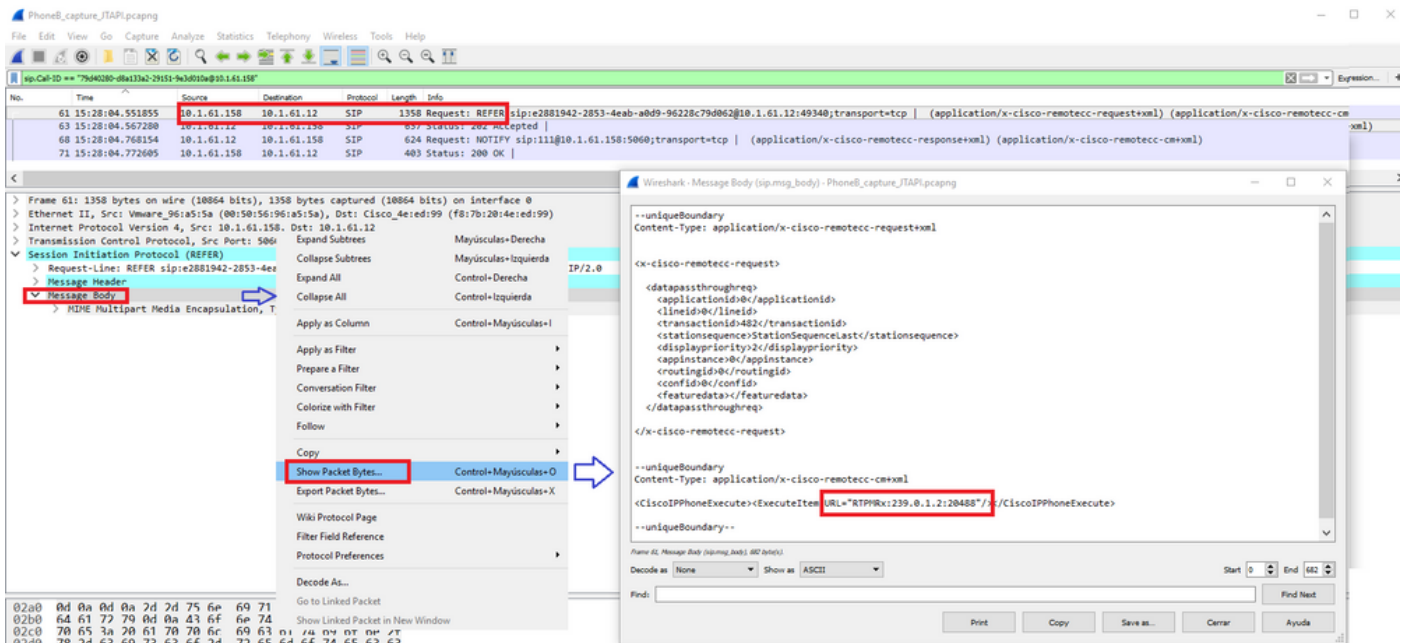- CUCM:10.1.61.158

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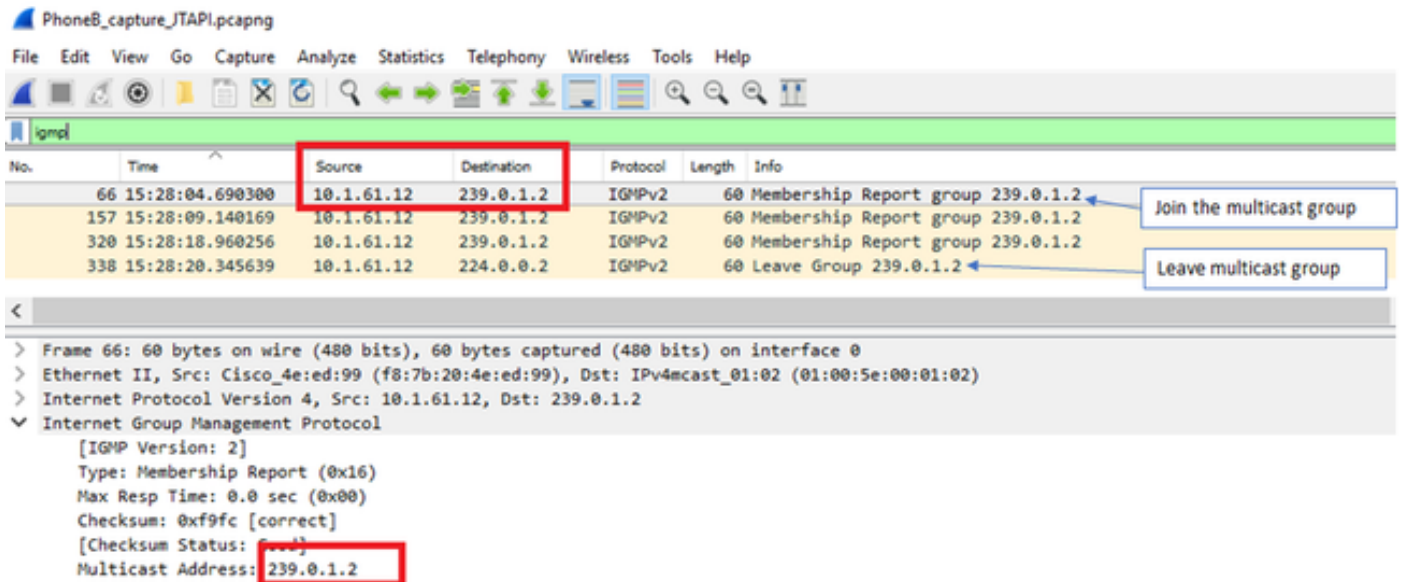- 型號：CP-8811
- 韌體版本：sip88xx.12-0-1SR1-1
- MAC SEPF87B204EED99

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

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

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

故障排除工具

組播測試工具將幫助您進一步排除SNMP故障。

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

高級許可證

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

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

密碼

在Informacast中，有幾種型別的密碼：

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

管理員密碼：用於登入到管理介面(https://x.x.x.x:8444/InformaCast/admin)。

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

Cisco出版書籍

對於思科呼叫伺服器12.5.1和轉發
：https://www.singlewire.com/help/InformaCast/v12.5.1/advanced/cucm/index.htm#t=InformaCast_Fusion%2FWebmin%2FRecover_the_Servers_Password.htm

在Informacast中更新JTAPI

初次安裝InformaCast虛擬裝置或每次更改CUCM版本時，需要將InformaCast虛擬裝置使用的JTAPI庫更新為CUCM伺服器使用的相同版本。

通過虛擬裝置更新JTAPI將更新所有使用JTAPI的Singlewire應用程式的JTAPI版本。

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

常見缺陷

[CSCve47332](#) Cisco IP Phone 69XX系列無法處理Application User for Informatacast中的空格

[CSCuy56088](#) 8800系列電話無組播音訊

[CSCut91894](#) 從FF37和Chrome到InformaCast的連線在FF/Chrome升級後失敗

[CSCtb70375](#) SNMP需要提醒使用者存在DNS連線問題

相關資訊

- CUCM相容性矩陣：https://www.singlewire.com/matrix/cisco-platforms
- 電話矩陣：https://www.singlewire.com/matrix/cisco-phones
- 升級路徑：https://www.singlewire.com/matrix/ic-upgrades
- 伺服器平台：https://www.singlewire.com/matrix/server-platforms
- 硬體要求：https://www.singlewire.com/informacast-hardware-requirements
- 技術支援與檔案 — Cisco Systems
  SRND:https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12.pdf
- CUCM與思科呼叫伺服器/InformaCast整合配置示例
  ：https://www.cisco.com/c/en/us/support/docs/unified-communications/paging-server/117059-configure-informacast-00.html
- 思科呼叫伺服器快速入門手冊
  ：https://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/cucm/cisco_paging_server/12_5_1/QSGInformaCastBasicPaging1251.pdf