配置Informacast并排除故障

目录

简介

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

先决条件

要求

Cisco 建议您了解以下主题：

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

使用的组件

本文档中的信息基于以下软件和硬件版本：

- InformaCast版本11.5.2 - 38
- CUCM版本11.5.1.14900-8
- CP-8811和CP-8861 sip88xx.12-0-1SR1-1
- 基本许可证

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

背景信息

功能的用途

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

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

基本与高级

基本寻呼模式仅支持每个收件人组最多50部电话的实时音频广播，无需额外许可证。作为CUCM的一部分提供的InformaCast版本包括基本寻呼模式的许可证。需要补充功能的客户可以升级到高级通知模式并受Singlewire支持。

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

使用的协议

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

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

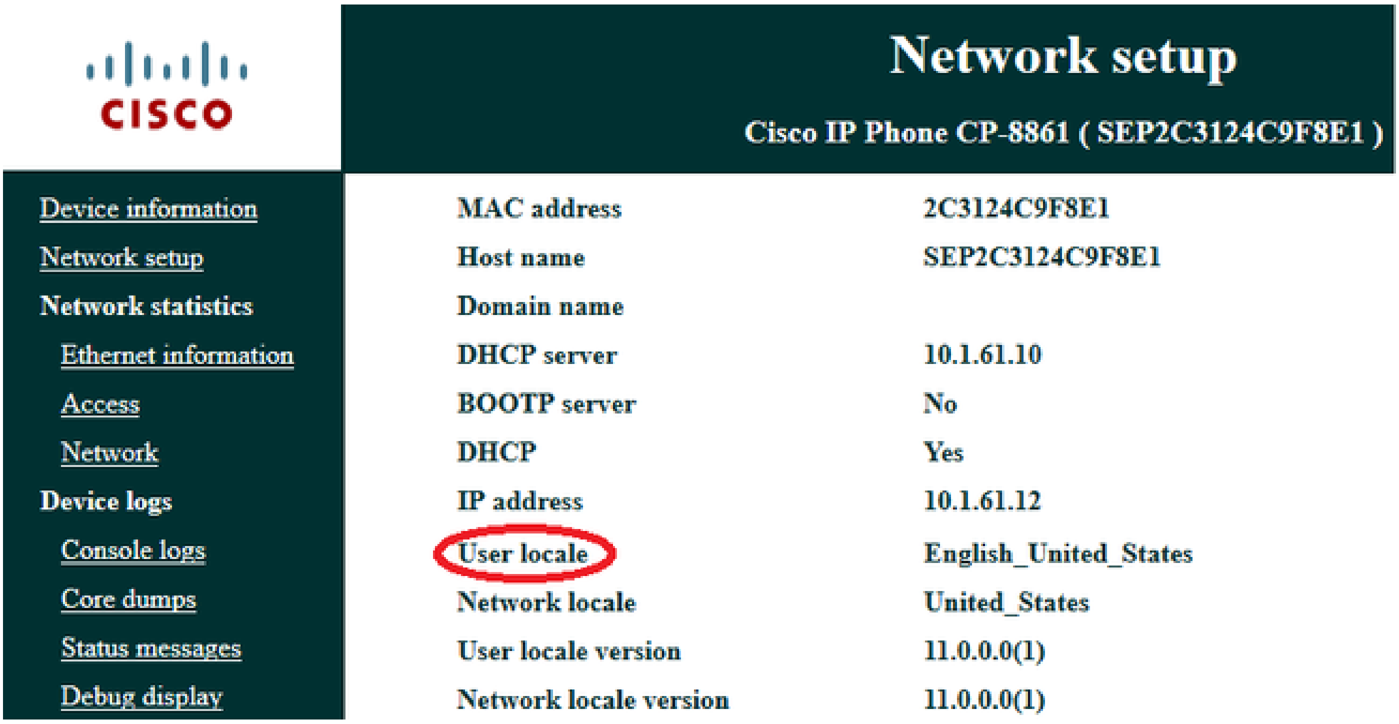
注意：每个Unified CM集群支持一个思科寻呼服务器。

HTTP与JTAPI

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

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

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



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

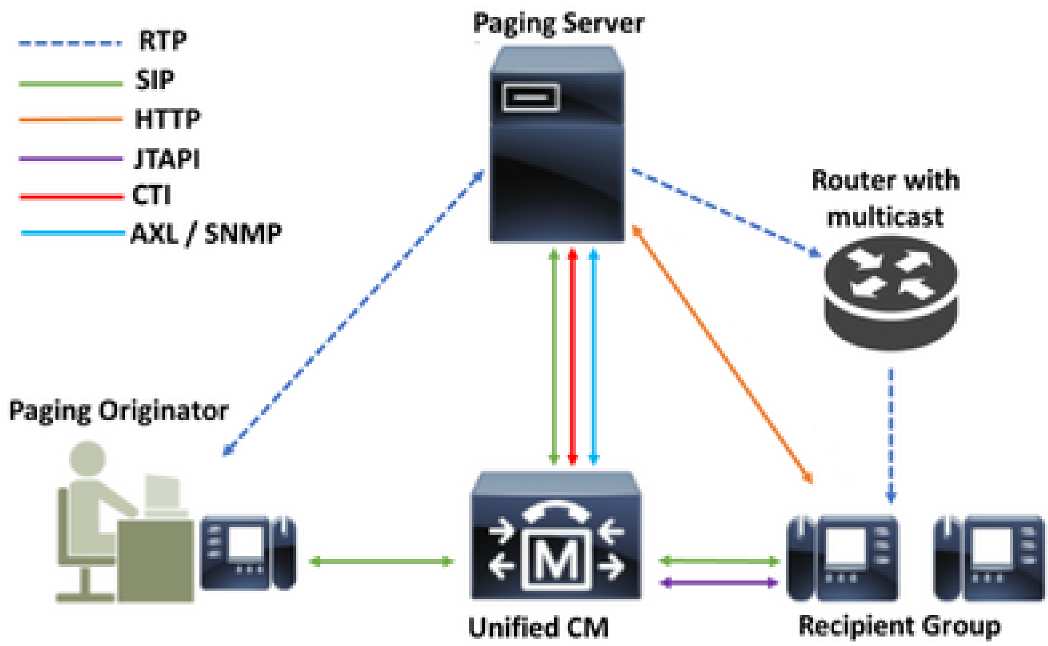
SIP与CTI

Informacast可以通过CTI和/或SIP接收呼叫。在CTI的情况下，呼叫在CTI路由点上进行服务（思科寻呼服务器不需要CTI端口来应答入站呼叫）。

对于SIP，呼叫在SIP中继上离开Unified CM。CTI和SIP均有效且受支持。但是，思科建议通过CTI进行SIP呼叫流，因为排除SIP集成故障比CTI容易得多。

配置

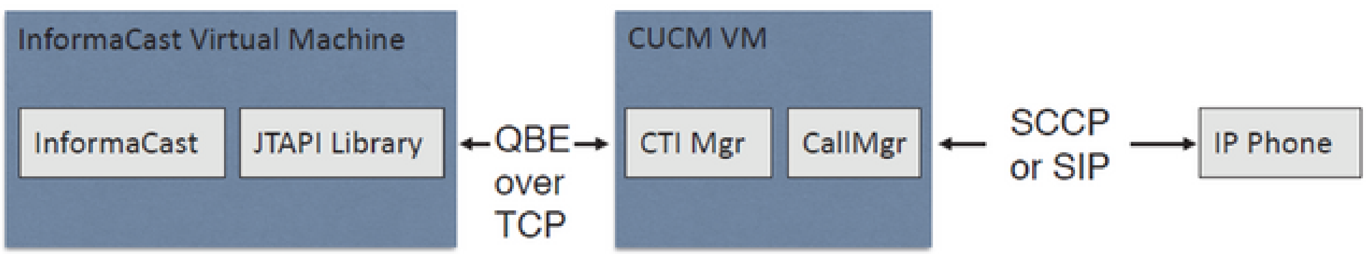
网络图



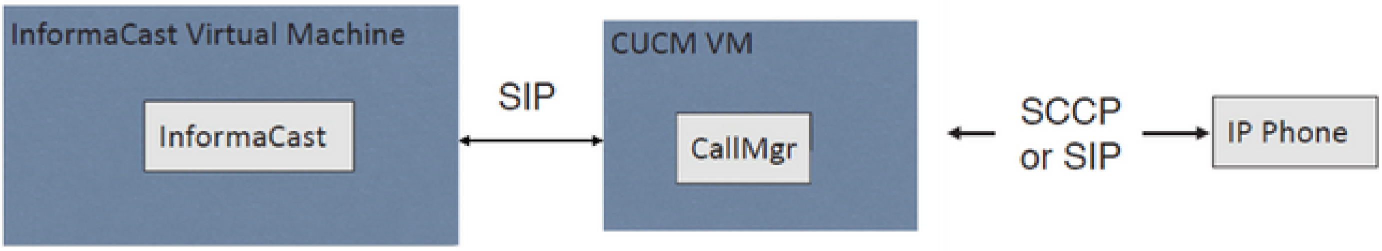
呼叫流程

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

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



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



配置Call Manager

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

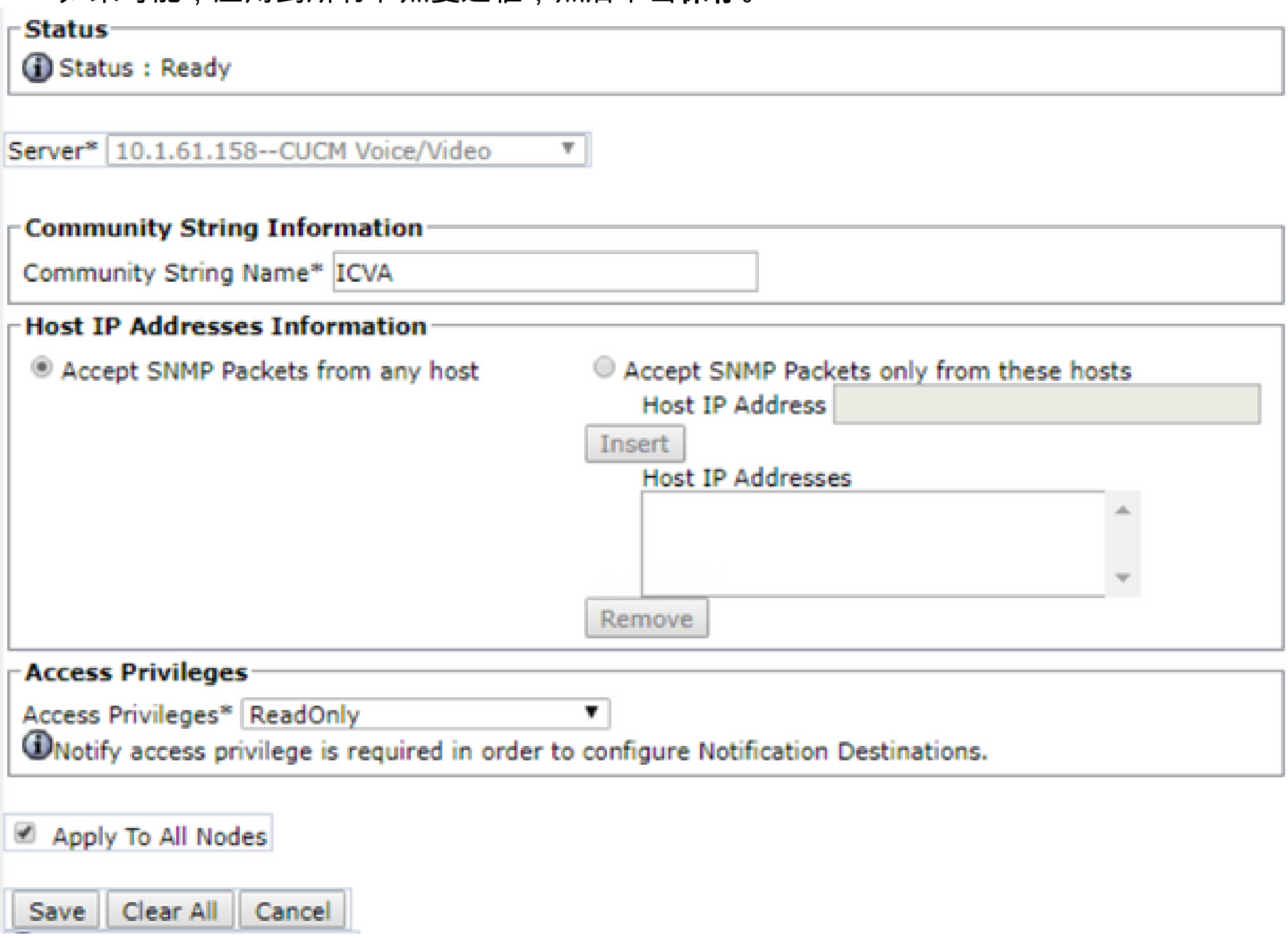
- Cisco CallManager
- Cisco CTIManager
- Cisco AXL Web Service
- Cisco CallManager SNMP Service

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

步骤2.配置SNMP（版本2或版本3）

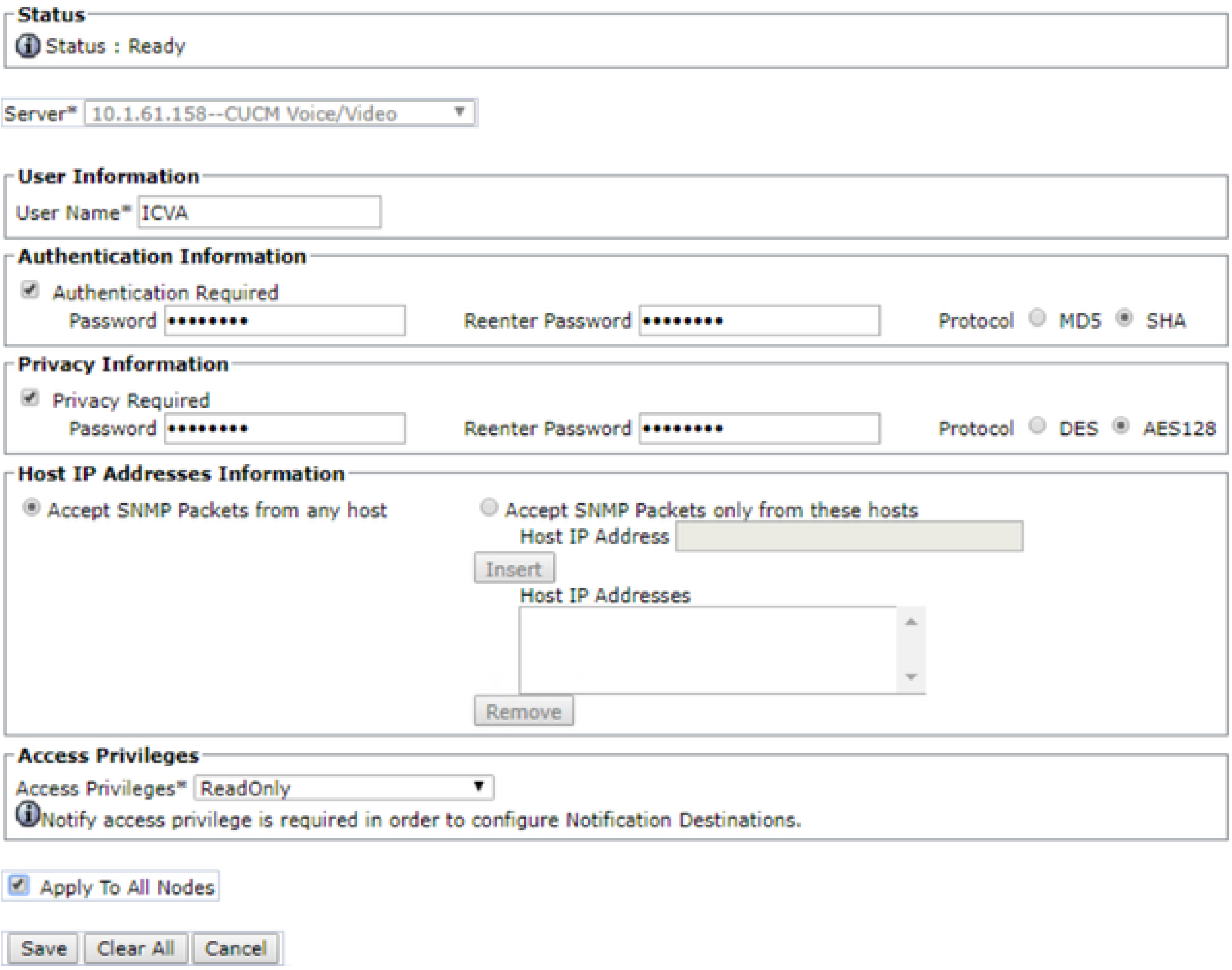
对于SNMP v2

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



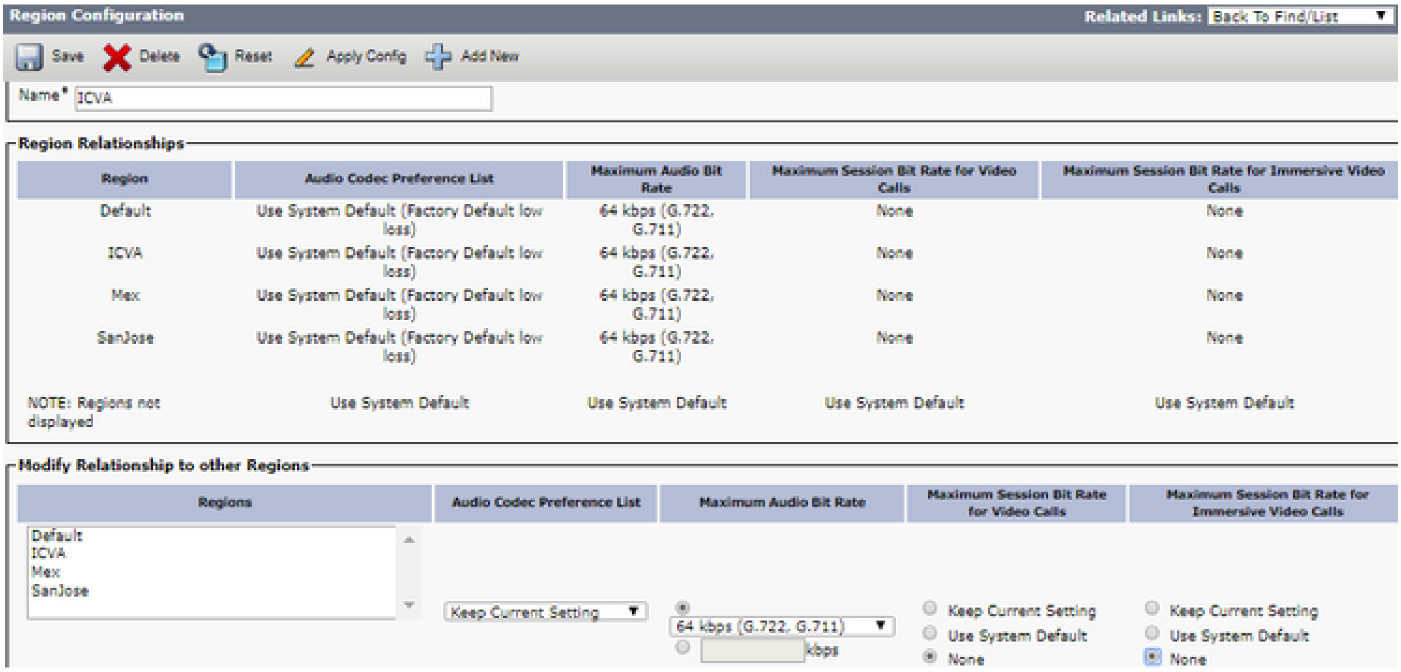
对于SNMP v3

- 导航至**Cisco Unified Serviceability > SNMP > V3> User**并创建名为ICVA的用户。
- 启用"需**要身份验**证"复选框，输入身份验证密码并选**择SHA**单选按钮。
- 启用"**需要隐**私"复选框，输入隐私密码并选**择AES128单**选按钮。
- 从"访**问权**限"下拉菜单中选择"只读"，并选**中"应用到所**有节点"复选框（如果可能），然**后单击 "保**存"。



步骤3.将默认编解码器设置为G.711

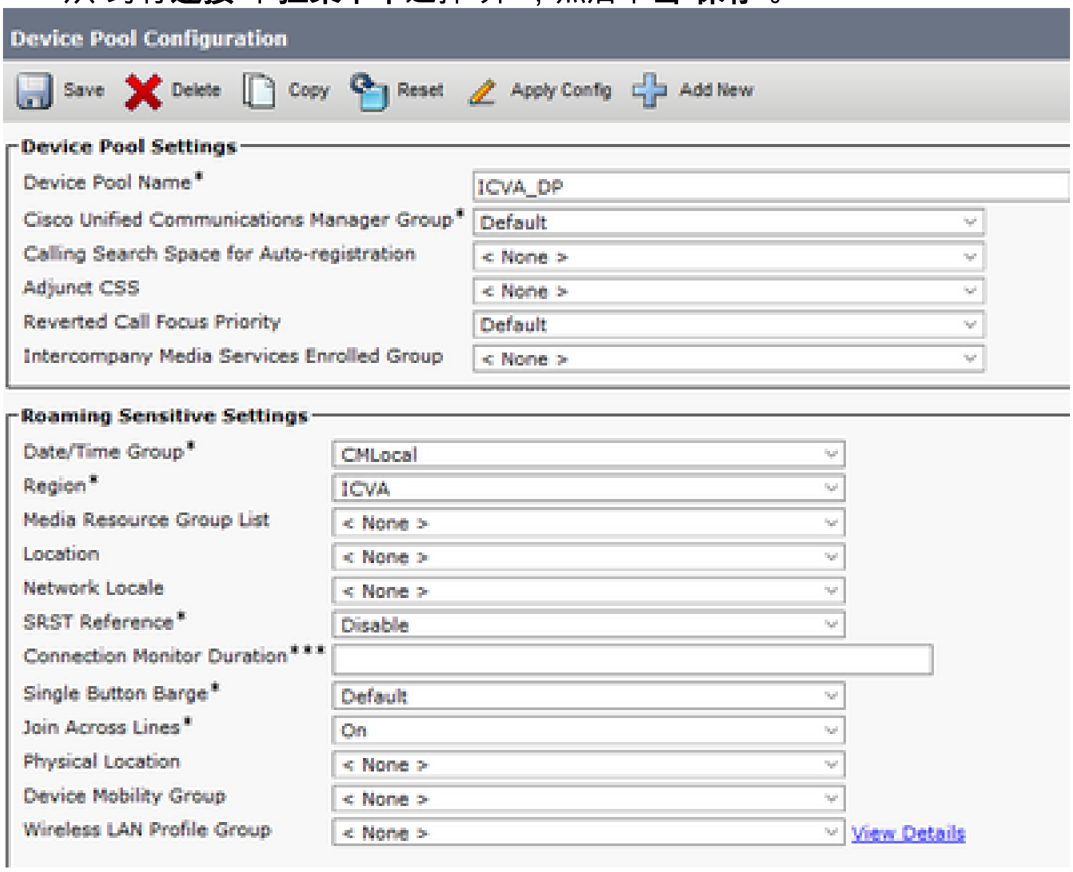
- 导航至**CM管理>System >区域信息>区域**并创建新区域，例如ICVA。
- 在Regions区域中选择所有区域，并将**64kbps(G.722、G.711)配**置为Maximum Audio Bit Rate。
- 在Max Video Call Bit Rate中**选择**None单选按钮，然后单击**Save**。



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

步骤4.创建设备池

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



步骤5.创建路由分区，例如ICVA_PT。

步骤6.创建呼叫搜索空间，例如ICVA_CSS。包括ICVA_PT。

步骤7.创建访问控制组(AXL)。

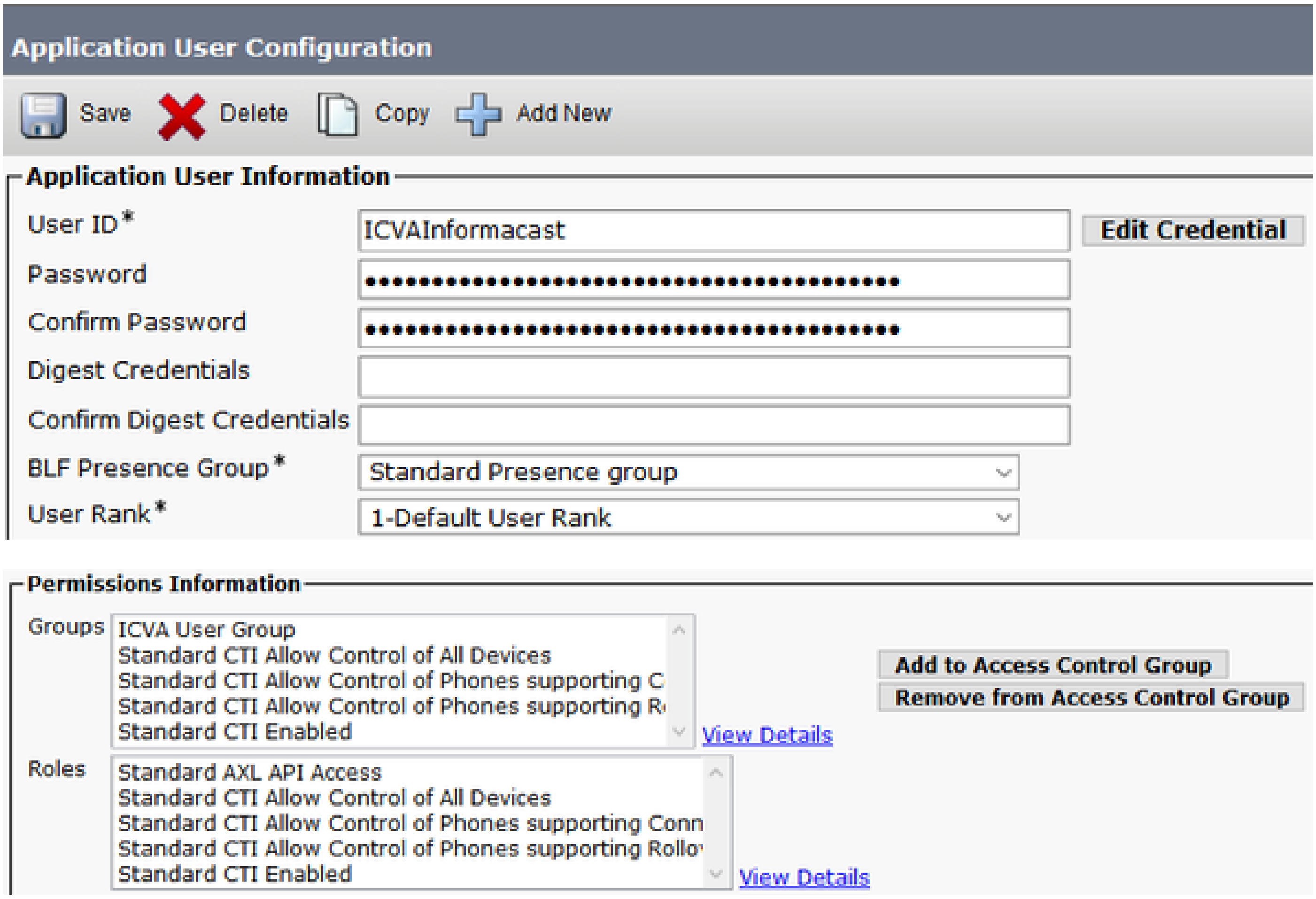
- 导航至**CM Admin > User Management > User Settings > Access Control Group**并创建访问控制组，例如ICVA User Group。
- 将标准**AXL API访问角**色添加到该角色。

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

步骤8.创建应用用户

- 导航至**CM Admin > User Management > Application User，然**后点击**Add New**。将应用程序用户命名为**ICVA_InformaCast**并分配以下角色：

1. Standard CTI Enabled
2. ICVA用户组（或标准AXL API访问）
3. 标准CTI允许控制支持Connected Xfer和Conf的电话。
4. 标准 CTI 允许控制支持回滚模式的电话
5. Standard CTI Allow Control of All Devices

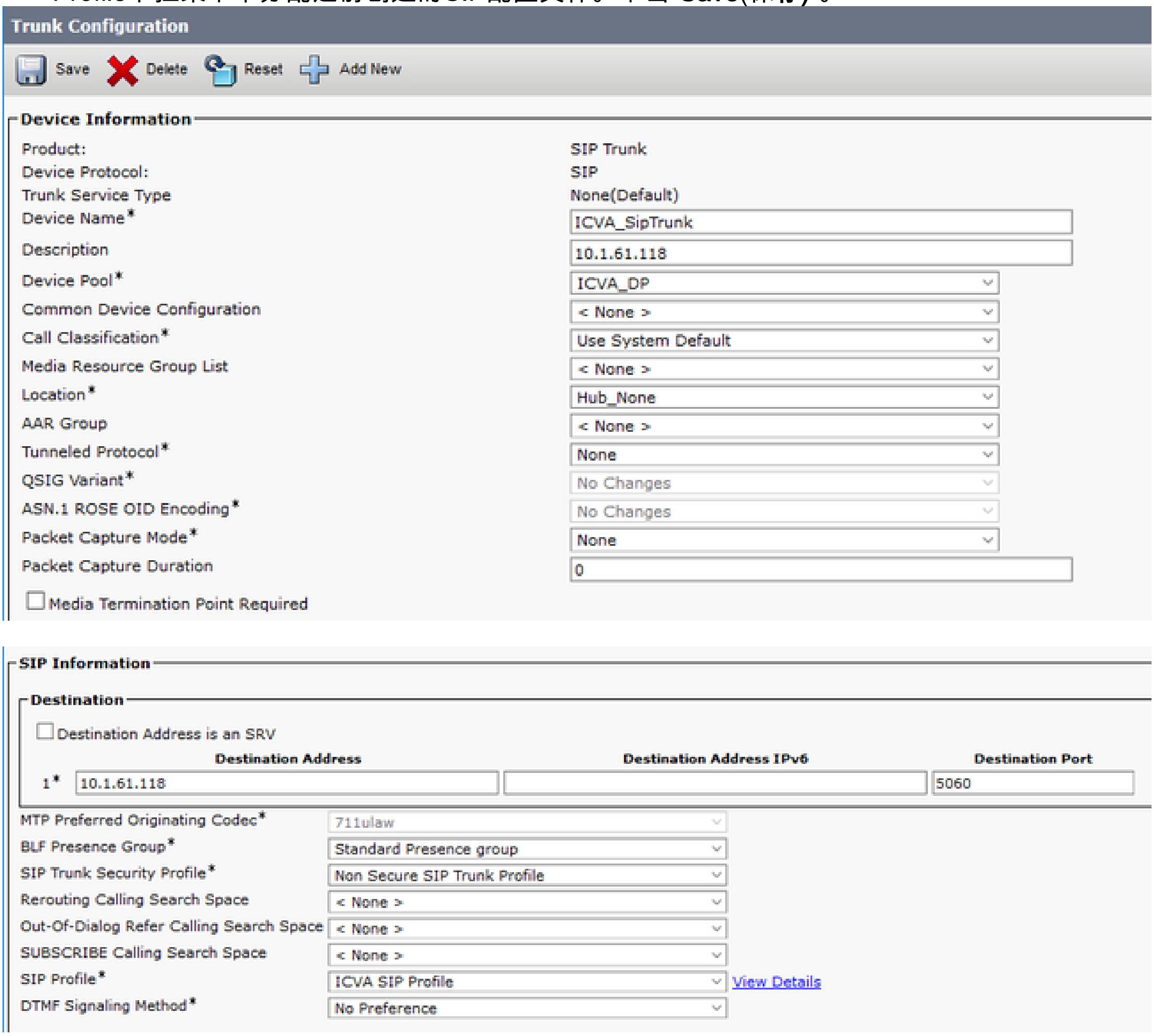


警告：每个缺陷 CSCve47332 ，建议不要为应用用户ID使用空格。

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

对于SIP集成，请创建SIP配置文件、SIP中继和路由模式。

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



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

- 选择**Route This Pattern和**OnNet单**选按**钮。
- 取消选中**Provide Outside Dial Tone复选**框，然后单击**Save**。

对于CTI集成，请创建CTI路由点并关联到步骤8中创建的应用用户。

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

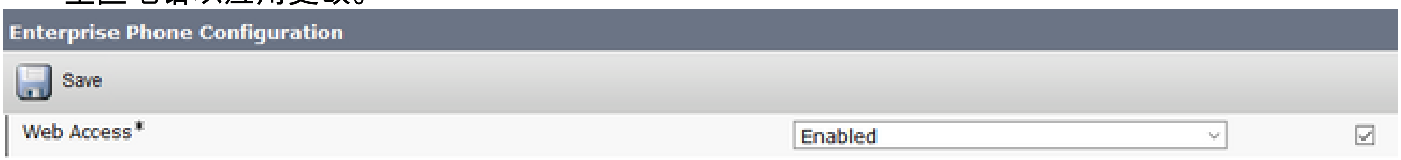
在ICVA应用用户配置中将CTI路由点添加为受控设备。

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

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

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

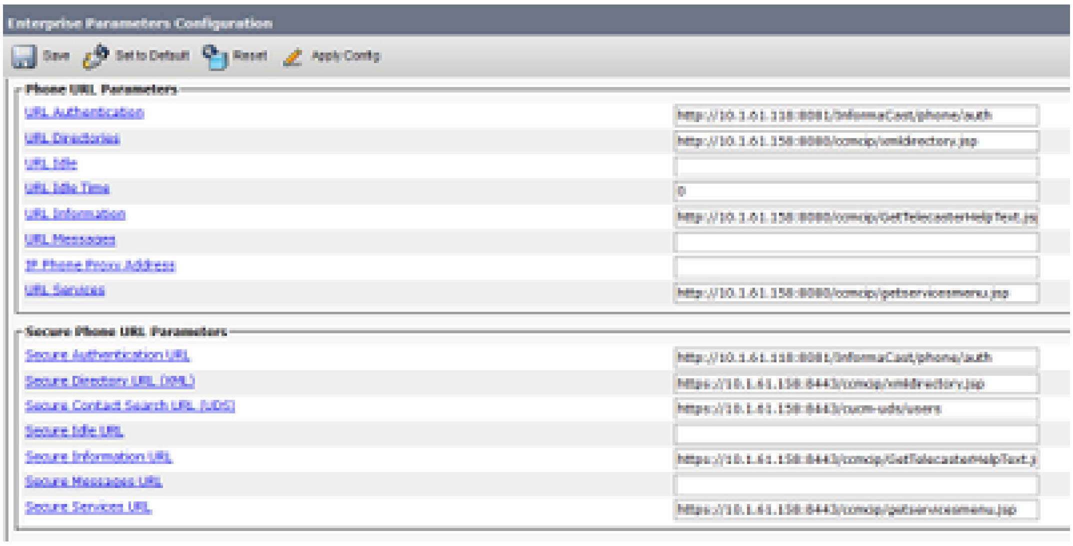
- 在企业电话配置中，可以按设备、通用设备配置文件或系统范围配置Web访问。
- 要应用企业电话配置中的更改，请导航至**CM Admin > System > Enterprise Phone Configuration**，向下滚动到**Web Access**下拉菜单并选择**Enabled**。单击“**Save(保存)**”。
- 重置电话以应用更改。



步骤11.设置身份验证URL。

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

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



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

步骤12.设置API浏览器访问的身份验证方法。

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

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

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

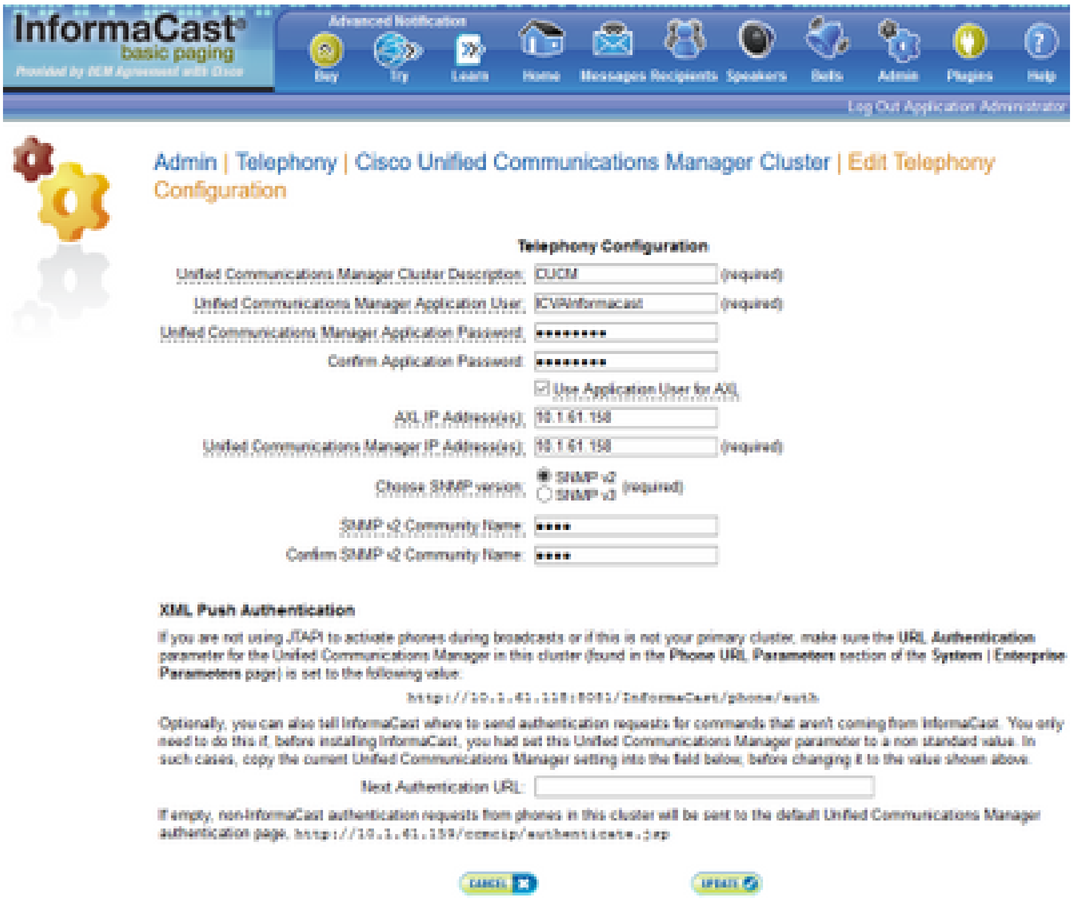
配置Informacast

步骤1.在Informacast中配置Communications Manager集群。

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

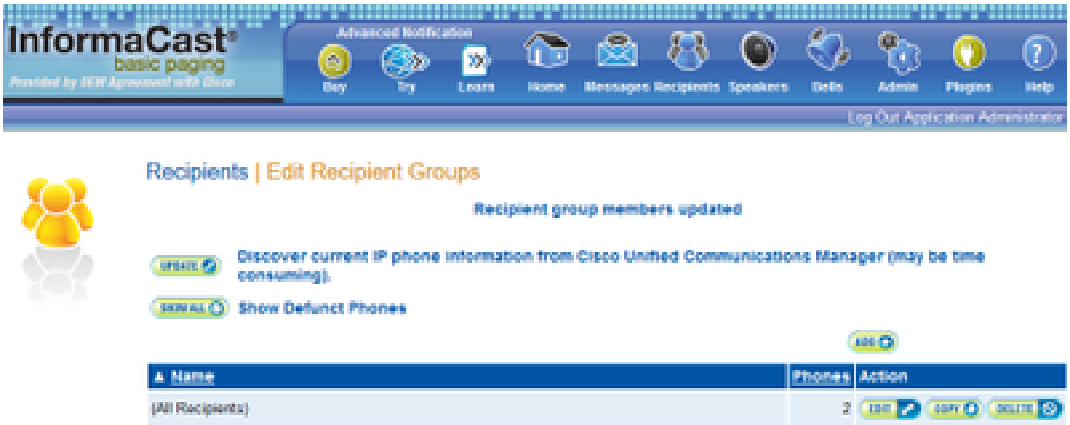
注意：如果将此字段留空，InformaCast将尝试在运行CallManager服务的服务器中查找运行AXL服务的服务器。

- 在Communications Manager IP Address(es)字段中输入Unified Communications Manager服**务器的IP地**址。使用数字IP地址而不是DNS名称。
- 选择**SNMP v2**或**SNMP v3单**选按钮。输入在CUCM中配置的相同信息。单击"**Update**(更**新)"**。

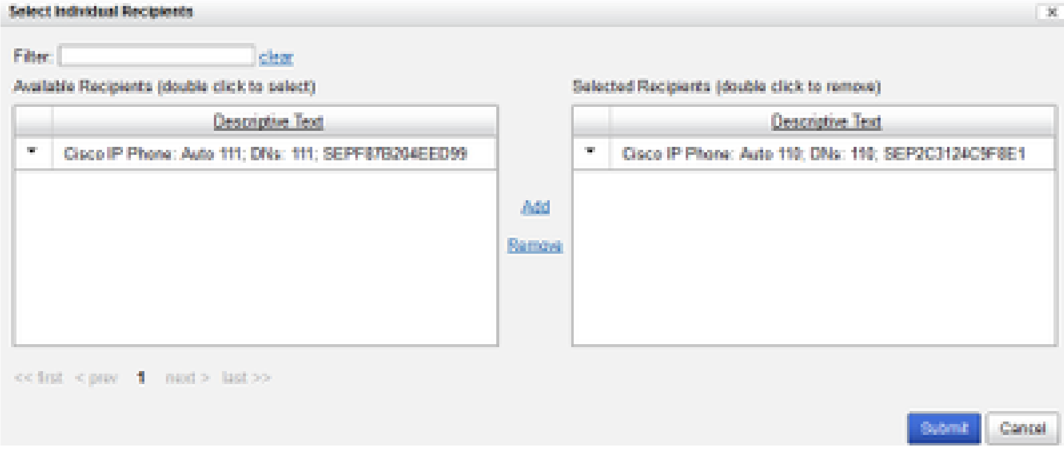


步骤2.配置收件人组。

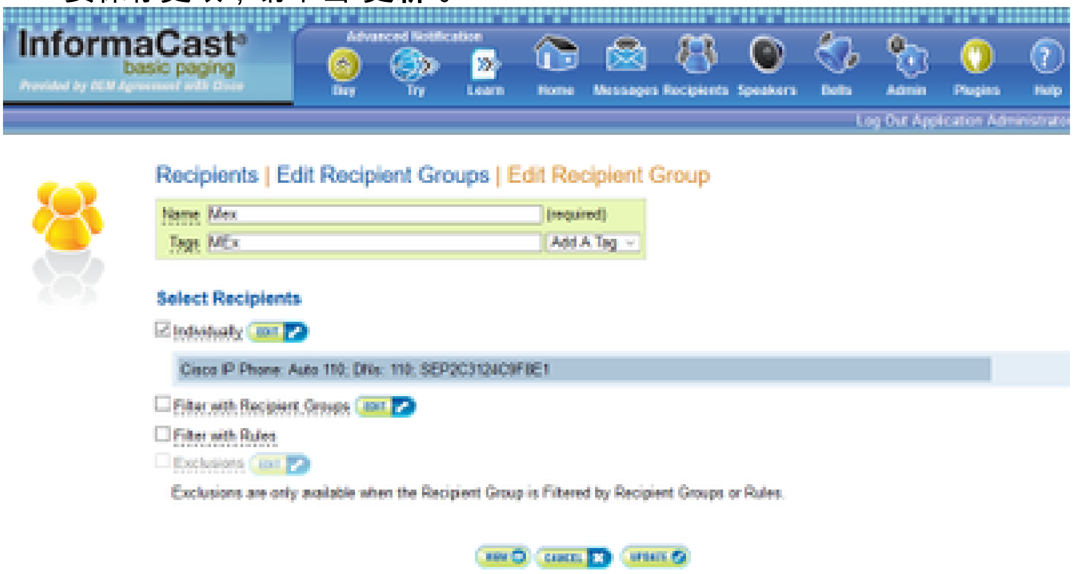
- 导航至**Recipients > Edit Recipient Groups**，然后单击**Update**以显示在CUCM中注册并由InformaCast发现的所有电话。



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

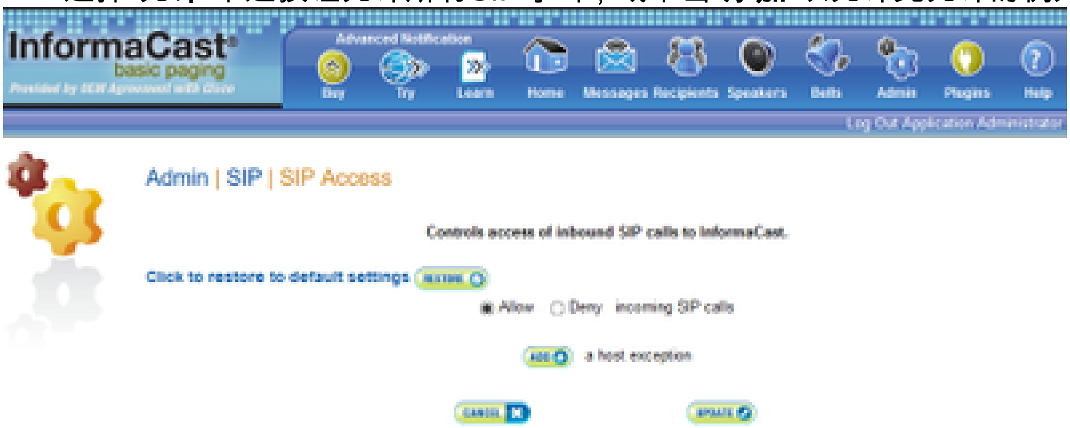


- 要保存更改，请单击“**更新**”。



步骤3.允许/禁用对InformaCast的SIP访问。

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

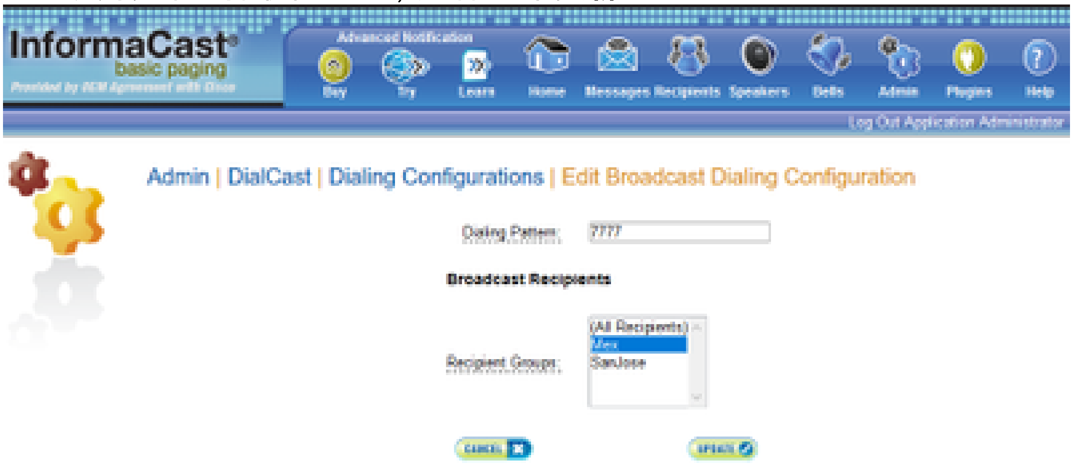


提示：在定义例外时，请确保指定直接将INVITE请求发送到InformaCast的主机。如果代理位于InformaCast和主叫主机之间，则此服务器可能是SIP代理服务器。

步骤4.添加广播拨号配置

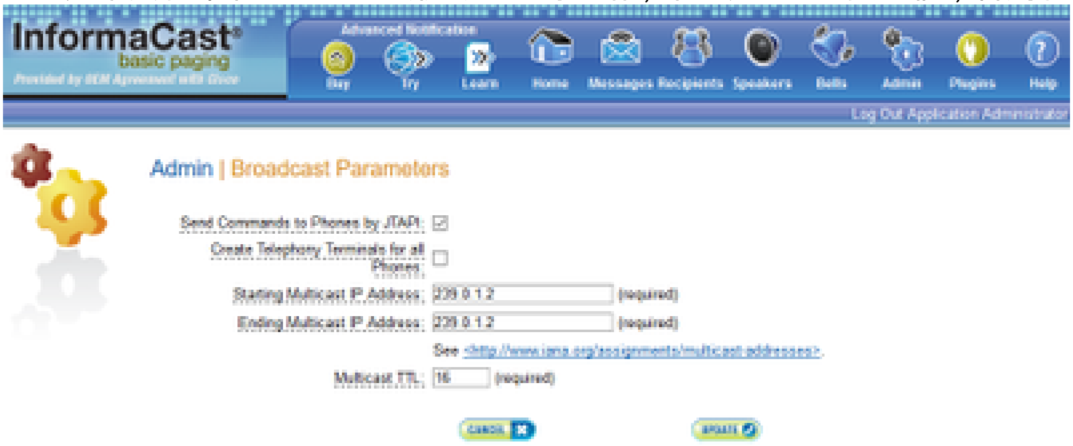
- 转到Admin > **DialCast > Dialing Configurations**，单击**Add**

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



步骤5.配置广播参数。

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



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

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

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

在网络中配置组播

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

思科寻呼服务器不需要任何特定的组播路由方法（SM、DM、S-DM、SSM等）。 某些广域网环境不支持组播路由。对于这些环境，GRE隧道可以在站点之间构建，并用于传输组播。

在您的环境中设计和配置组播不在本文档的讨论范围之内，但您可能会发现以下资源很有帮助：

- 组播白皮书
- 组播测试工具

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

验证

当前没有可用于此配置的验证过程。

故障排除

常见问题

电话未激活

考虑到广播发生时，Informacast会跳过所有正在使用（忙）的电话。

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

HTTP:忙线检测仅适用于运行英语负载的电话区域设置

CTI:与非英语电话区域设置配合使用

根据协议以及线路类型和线路状态，忙线检测的工作方式也不同。

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

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

排除未激活的电话故障时，应收集以下数据：

- 从Informacast发送性能日志。
- 从电话中记录控制台日志(PRT)。

未发现电话

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

符串。

SNMP错误无法生成收件人组：java.lang.exception

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

目标电话上没有音频

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

要收集的数据

排除Informacast故障时，应收集以下数据：

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

性能日志

从Informacast获取性能日志有两种方法。

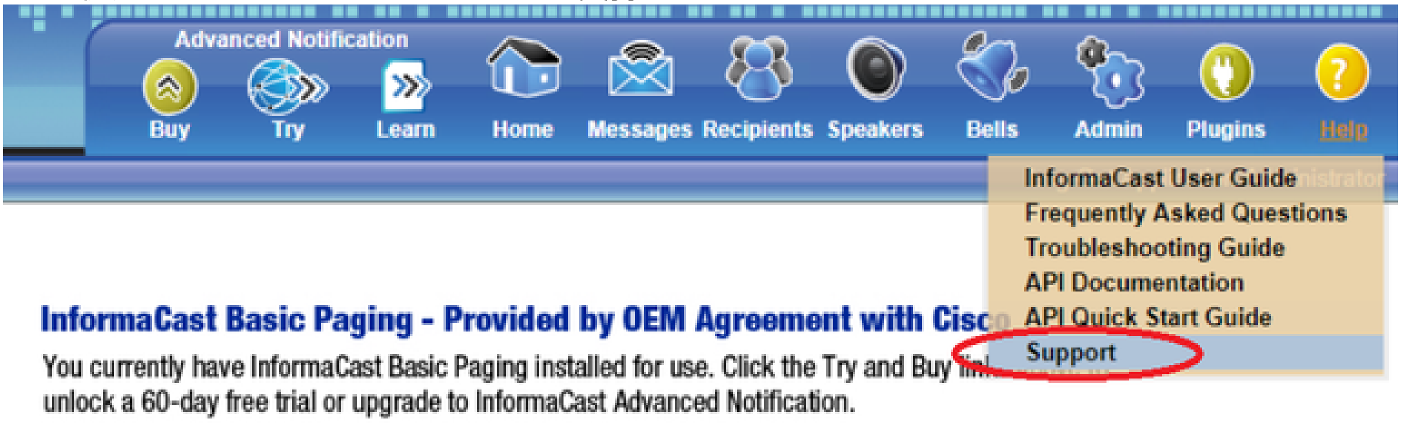
方法 1

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

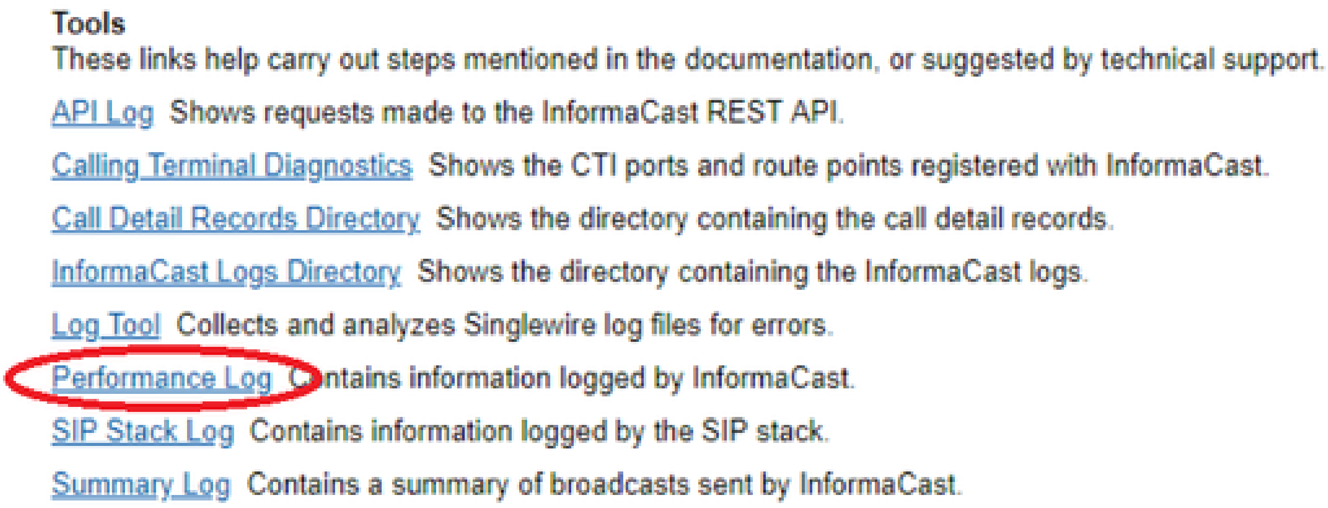
2. 复制并将日志保存到.txt文件。

方法 2

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



3.单击“工**具”部**分下的“性能日志”，如图所示。



数据包捕获

从Informacast

从Informacast获取数据包捕获有三种方法。

方法 1

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

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

方法 2

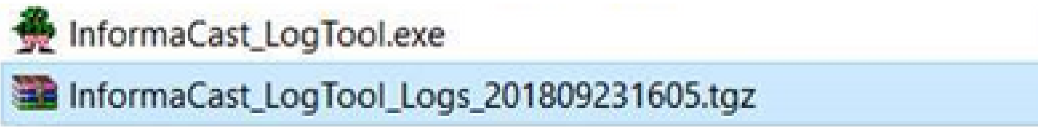
1. 从Web下载并安装InformaCast_LogTool。
2. 执行软件并选择选项[5]。写下Informacast的IP、登录凭证和数据包捕获应运行的秒数，如图所示。

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

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

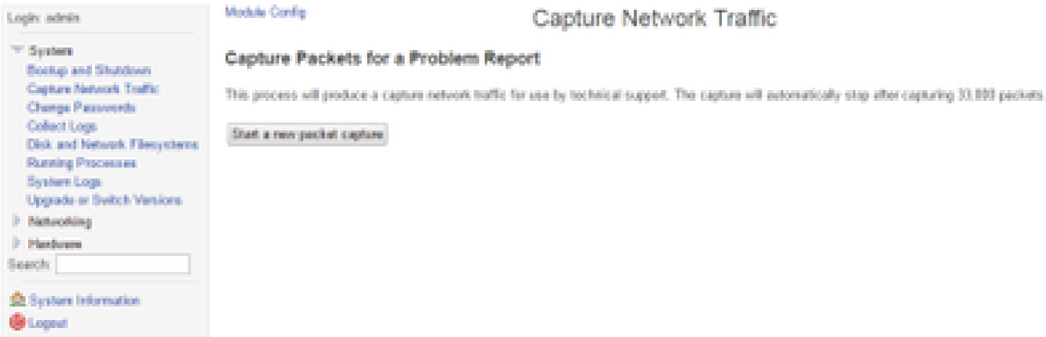
```
Seconds Capture Should Run [1-300]: 60

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

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

00:00:51
```

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

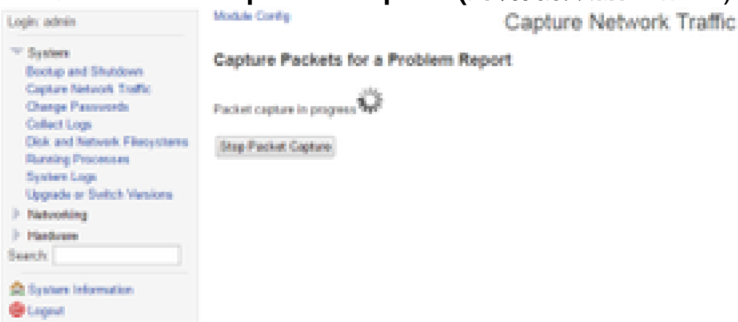


方法3（在版本12.0.1及更高版本中提供）

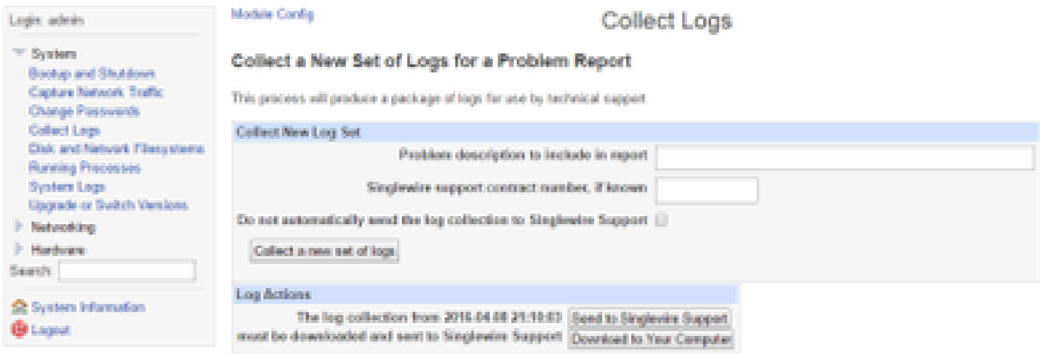
1. 登录<Informacast_IP>:10000
2. 导航至**System > Capture Network Traffic。**



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



4. 当问题被完全复**制时，单击Stop Packet Capture（停止数据包捕获），或者在捕获33,000个数据包后单击Stop Packet Capture（停止数据包捕获）。**
5. 导航至**System > Collect Logs**，输入问题的简短说明，然后单**击Collect a new set logs。**
6. 要保存日志，请单击"Download to Your Computer(**下载到您的计**算机)"，如图所示。



方法4（在版本12.0.1及更高版本中提供）

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

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

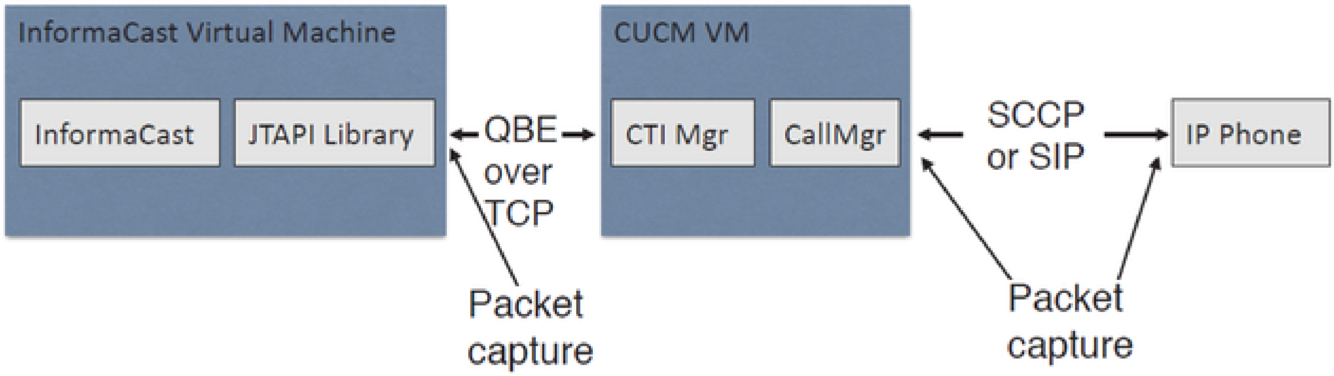
```
0 packets dropped by kernel
Interrupt signaled. Cleaning up.
```

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

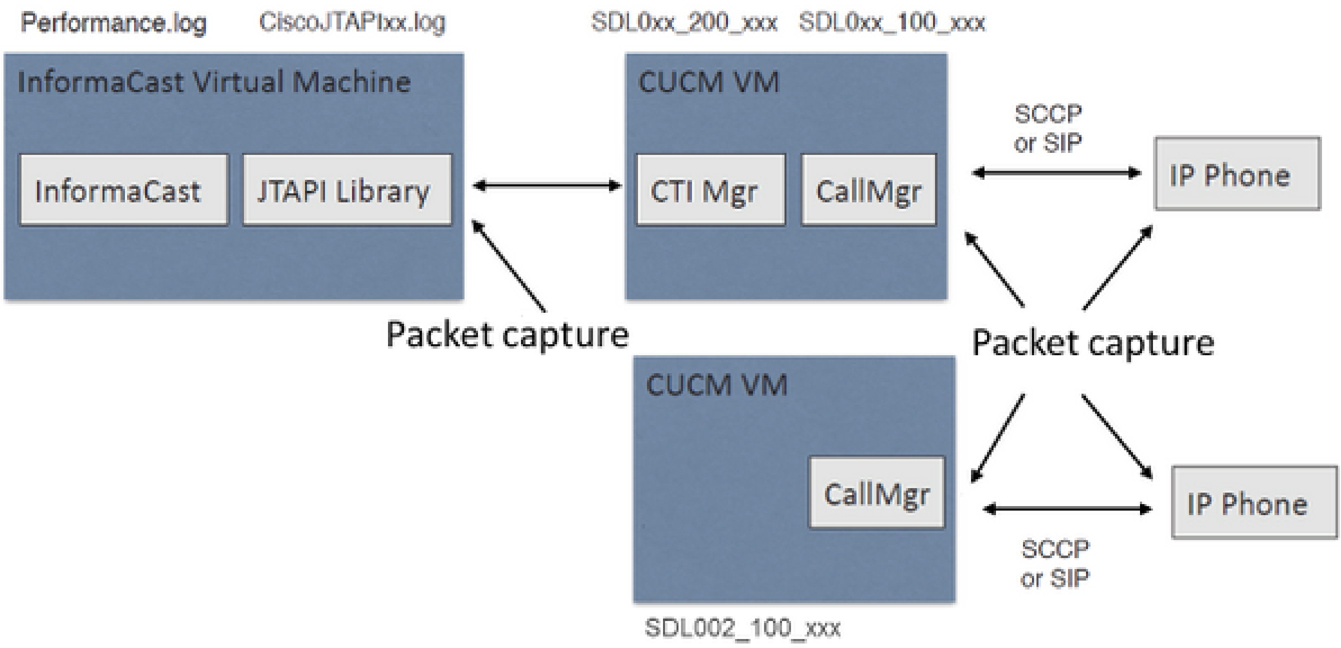
从CUCM

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

- 如果您有一个CUCM节点，请获取数据包捕获，如图所示。



- 如果您有CUCM群集，并且一个节点与Informacast通信，但另一个节点与电话通信，则获取数据包捕获，如图所示。



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

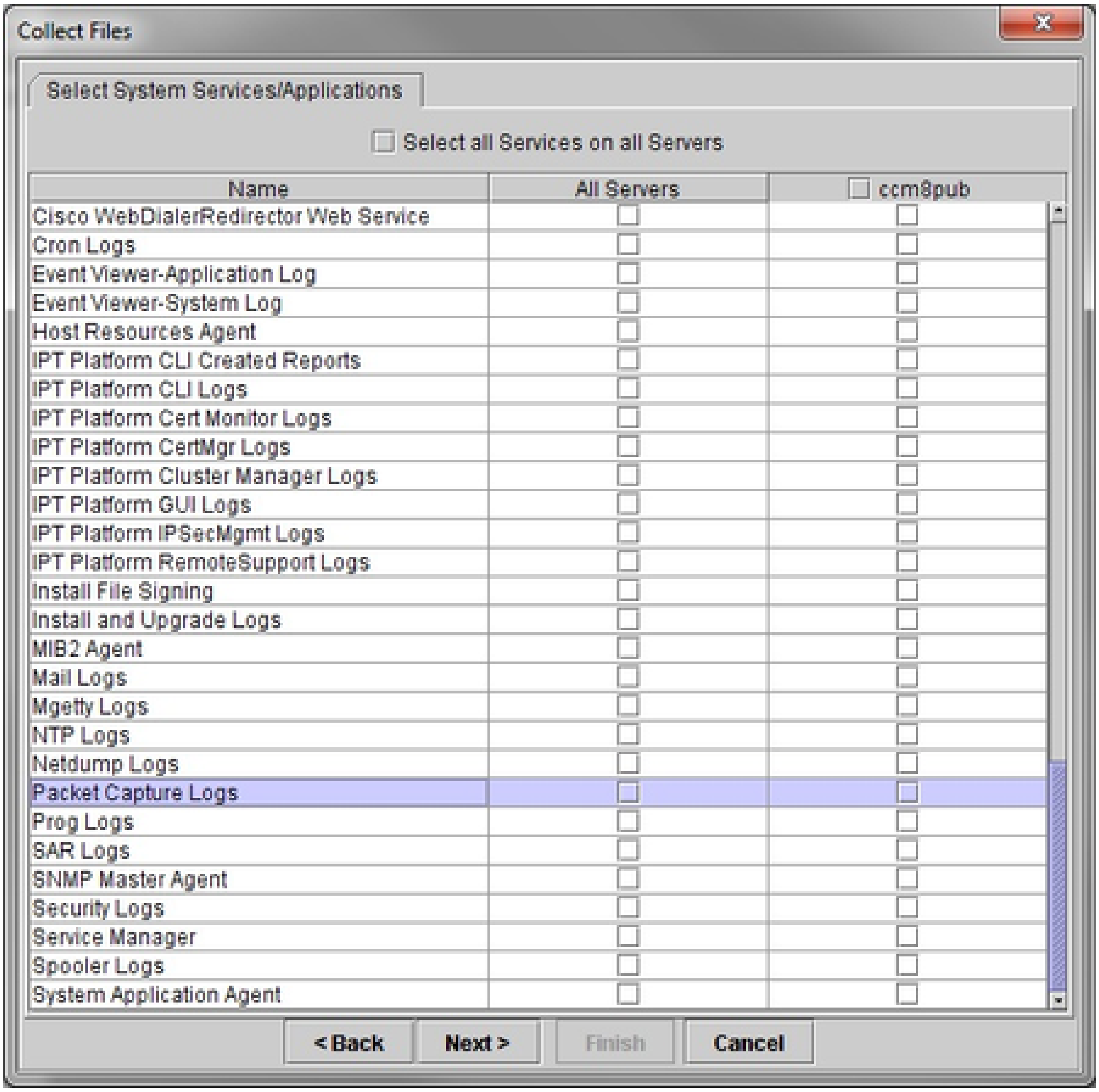
10.1.61.158 - PuTTY

```
admin:
admin:utils network capture eth0 size all count 1000000 file Test
Executing command with options:
 size=ALL                count=100000            interface=eth0
 src=                    dest=                   port=
 ip=

Control-C pressed

admin:file list activelog platform/cli/*
Test.cap
dir count = 0, file count = 1
admin:
```

6. 使用命令**file get activelog platform/cli/Test.cap**将数据包捕获发送到SFTP服务器。或者，要收集存储在服务器上的所有.cap文件，请**使用file get activelog 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. 如果无法使用SFTP服务器，请使用RTMT。导航**至System > Trace & Log Central > Collect Files**。单击“下**一步”**并启用“**数据包捕获日**志”复选框，如图所示。



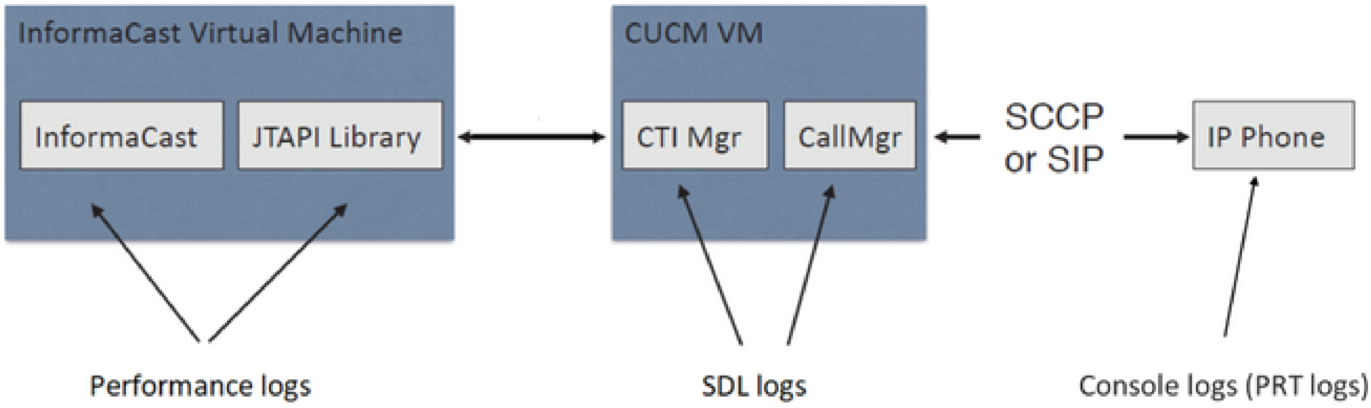
8. 单击“Next(下**一步**)”，选择下载文件目录，然后单击“**Finish（完成）**”。
9. 使用命令file delete activelog platform/cli/Test.cap删**除数据包**

从电话

1. 激活SPAN到PC端口。导航至**CM Admin page > Device > phone**，并查找报告有问题的电话。
2. 在“产品特定配置布局”部**分下，找到**Span to PC Port，然后从下拉**菜**单中选择“启用”。单击“保存”，然后单击“应用配置”。
3. 将笔记本电脑连接到电话的PC端口。
4. 在笔记本电脑中运行数据包分析器软件。您可以使用Wireshark（或其他数据包捕获软件）。
5. 复制问题。
6. 当问题完全复制时，继续停止数据包捕获。

您可以在以下链接中找到更多详细信息：https://supportforums.cisco.com/document/44741/collecting-packet-capture-cisco-ip-phone

示例分析



SDL跟踪

用于SIP集成和由JTAPI控制的电话

CUCM:10.1.61.158

Informacast:10.1.61.118

电话A

DN:110

型号：CP-8861

固件版本：sip8xx.12-0-1SR1-1

电话A的IP地址：10.1.61.12

MAC SEP2C3124C9F8E1

电话B

DN:111

型号：CP-8811

固件版本：sip8xx.12-0-1SR1-1

电话B的IP地址：10.1.61.11

MAC SEPF87B204EED99

拨号播号：7777

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

```
From: "PhoneA" <sip:110@10.1.61.158>;tag=2c3124c9f8e10c3c246b7956-5c62fa57
To: <sip:7@10.1.61.158>
Call-ID: 2c3124c9-f8e1000d-00337209-0547bb10@10.1.61.12
Max-Forwards: 70
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
```

用于CTI集成和由HTTP控制的电话

CUCM:10.1.61.158

Informacast:10.1.61.118

电话A

DN:110

型号：CP-8861

固件版本：sip8xx.12-0-1SR1-1

电话A的IP地址：10.1.61.12

MAC :SEP2C3124C9F8E1

电话B

DN:111

型号：CP-8811

固件版本：sip8xx.12-0-1SR1-1

电话B的IP地址：10.1.61.11

MAC :SEPF87B204EED99

拨号播号：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.
```

性能日志

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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
```

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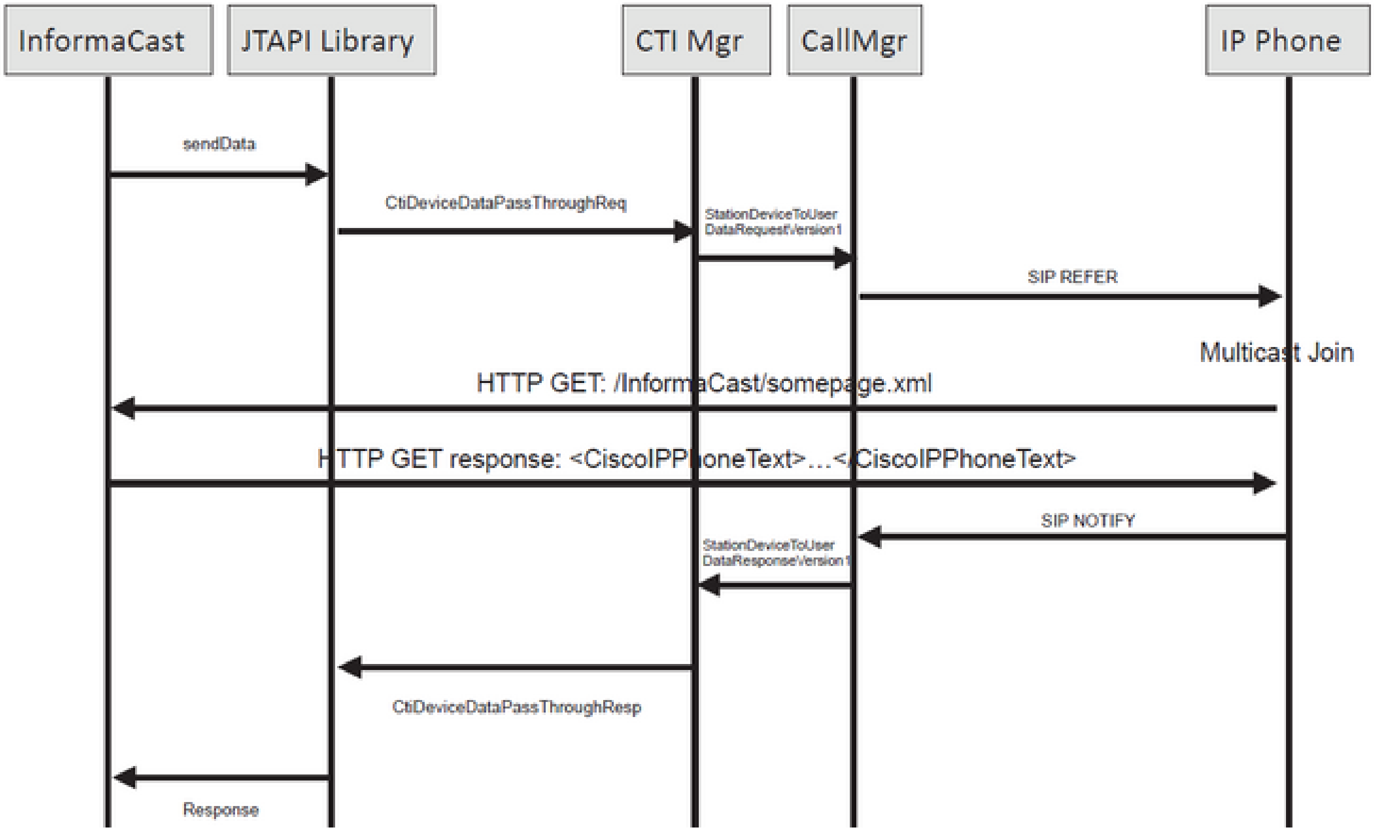


Informacast receives the request to route the call
2019-09-09 19:24:39,936 [RouteCall:15018/1Thread] INFO av [] - Route request for call [CiscoCallID=15018/1 callingDN=110 callingPartition=phone_pt callingTerminal=SEP2C3124C9F8E1 lastRedirectedDN=null modifiedCalledDN=7778 currentCalledDN=7778 calledDN=7778]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

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

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

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

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

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

Informacast receives the response from the phone

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

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

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

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

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

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

控制台日志(PRT)

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

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

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

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

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

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

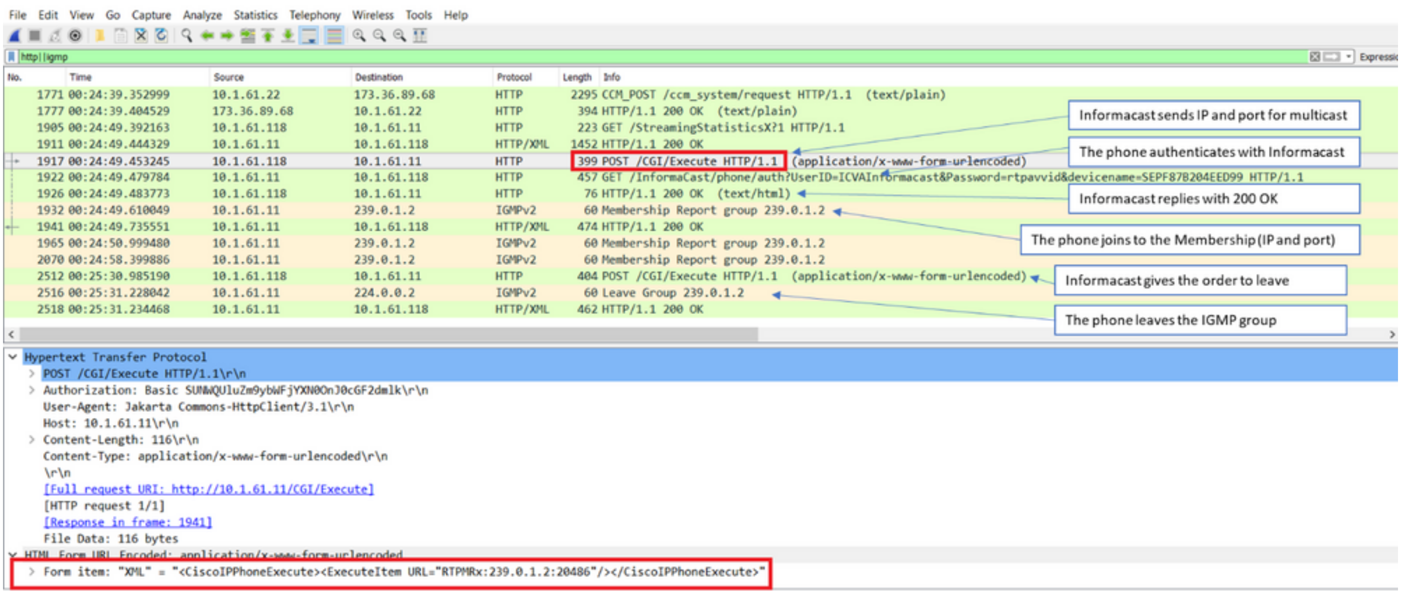
数据包捕获

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

电话上的数据包捕获（由HTTP控制）

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

电话上收到的HTTP和IGMP消息显示在图像中。

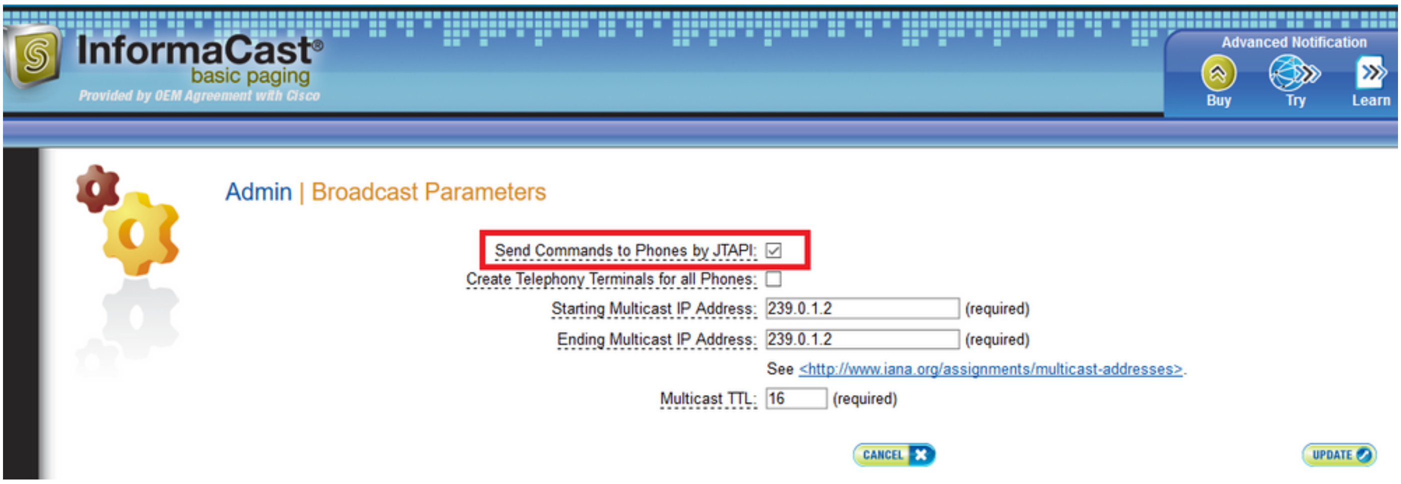


电话上的数据包捕获（由JTAPI控制）

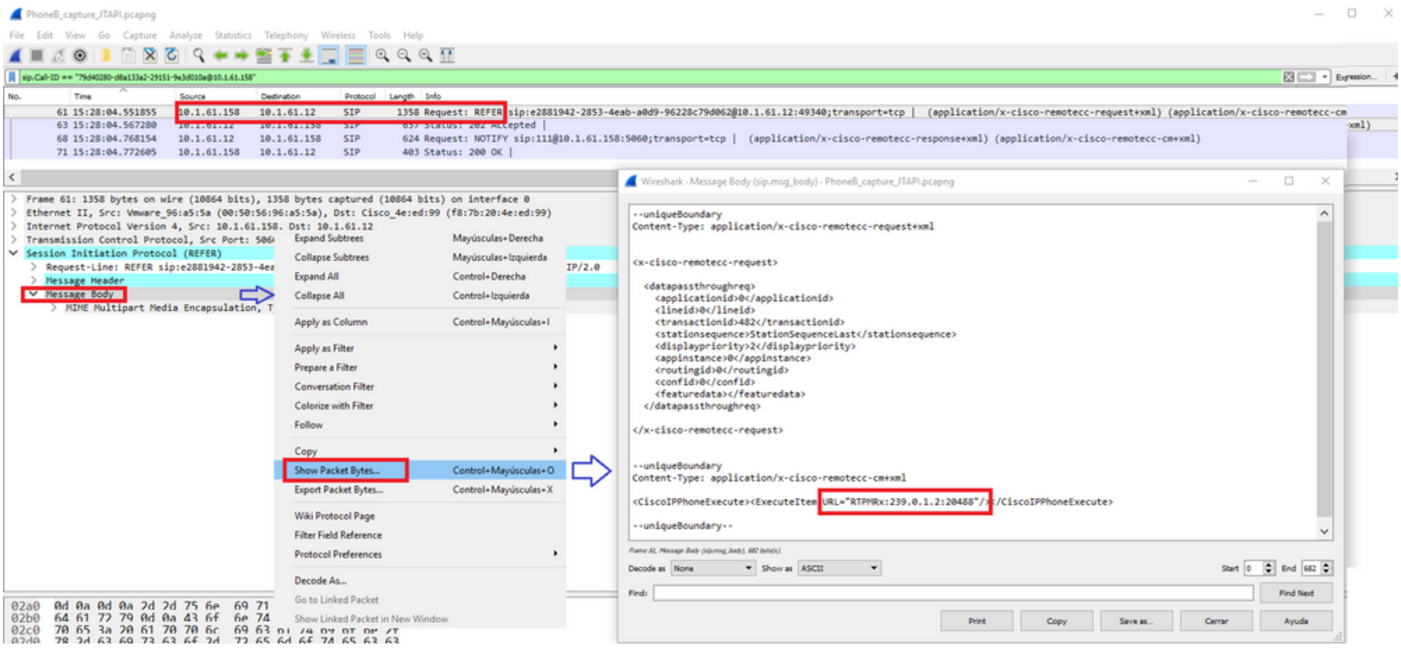
- CUCM:10.1.61.158
- Informacast:10.1.61.118

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

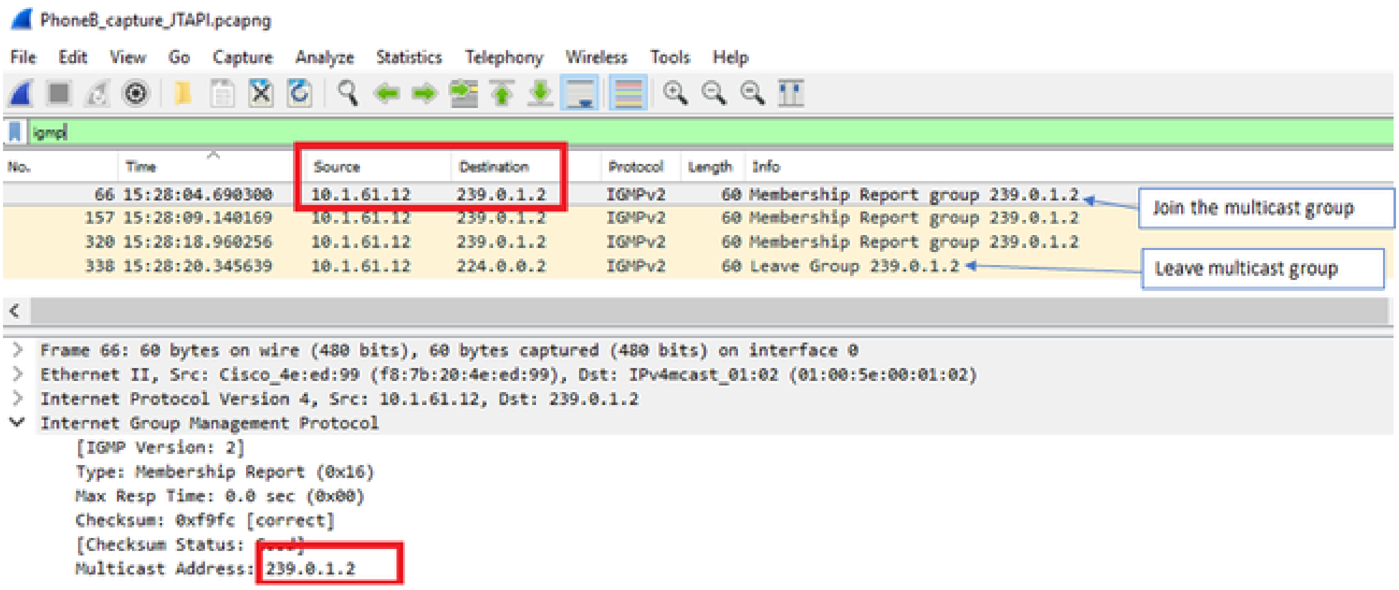
如配置部分所述，电话可以由JTAPI控制，这意味着**Send Commands to Phones by Jtapi已启**用，如图所示。



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



电话收到指令后，会使用IGMP消息加入组播IP和端口。电话尝试的接收音频次数是最大尝试次数的三倍。当寻呼结束时，收件人组中的电话发送离开组消息以丢弃组播会话。



故障排除工具

组播测试工具将帮助您进一步排除SNMP故障。

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

高级许可证

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

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

密码

在Informacast中，有几种密码类型：

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

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

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

密码 恢复

对于思科寻呼服务器12.5.1并转发：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

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

JTAPI库更新为CUCM服务器使用的相同版本。

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

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

常见缺陷

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

CSCuy56088 8800系列电话无组播音频

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

CSCtb70375 SNMP需要向用户发出DNS连接问题的警报

相关信息

- CUCM兼容性列表：https://www.singlewire.com/matrix/cisco-platforms
- 电话矩阵：https://www.singlewire.com/matrix/cisco-phones
- 升级路径:https://www.singlewire.com/matrix/ic-upgrades
- 服务器平台：https://www.singlewire.com/matrix/server-platforms
- 硬件要求:https://www.singlewire.com/informacast-hardware-requirements
- 技术支持和文档 — Cisco Systems SRND:https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12.pdf
- CUCM与思科分页服务器/InformaCast集成配置示例：https://www.cisco.com/c/en/us/support/docs/unified-communications/paging-server/117059-configure-informacast-00.html
- 思科寻呼服务器 — 快速入门指南：https://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/cucm/cisco_paging_server/12_5_1/QSGInformaCastBasicPaging1251.pdf