配置Jabber扩展和连接并修改主叫方显示

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简介

本文档介绍如何在Jabber中配置扩展和连接功能以及修改远程目标上显示的主叫方。

先决条件

思科统一通信管理器(CUCM)9.1或更高版本。

Jabber 9.1或更高版本。

要求

需要以前的经验和知识来配置使用Cisco Unified Communications Manager和IM and Presence Server的Jabber。

使用的组件

本文档中的信息基于以下软件版本:

- Jabber 11.8.2
- •思科统一通信管理器11.0.1.10000-10
- IM and Presence Server(IMP)11.0.1.10000-6

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。本文档中使用的所有设备最初均采用原 始(默认)配置。如果您的网络处于活动状态,请确保您了解任何配置的潜在影响。

配置

步骤1.为已配置Jabber的同一用户配置CTI远程设备(CTI RD)电话配置文件。

Phone Configuration	
🔜 Save 🗶 Delete 🗋 Copy 🌯 Reset 🧷 Apply Config 🕂	Add New
Status Status: Ready	
Association 1 •rm: Line [1] - 1001 in Phones 2 •rm: Line [2] - Add a new DN •rm: Line [2] - Add a new DN •rm: Line [2] - Add a new DN Real-time Device Status Registration: Registered with IPv4 Address:	Device
Device Information Device is Active Device is not trusted Active Remote Destination Owner User ID* Device Name* Description	3001 testuser1 CTIRDtestuser1

- 配置CTI RD时,关联到同一Jabber用户。线路配置与Jabber客户端服务框架(CSF)设备线路相同
- •需要正确配置重路由呼叫搜索空间,以使远程目标呼叫正常工作

步骤2.配置远程目标。

Remote Destination Conf	iguration			
Save 🗶 Delete 🗋	Copy 🕂 Add Ne	9W		
_ Status				
i Status: Ready				
CTI Remote Device		Remote Destination Information		
Line	Line Association	Name	JabberRD	
Line [1] - 1001 in Phones	\checkmark	Destination Number*	01	
		Owner User ID*	testuser1	T
		Enable Unified Mobility features		
		Remote Destination Profile*	Not Selected	•
		Single Number Reach Voicemail Policy*	Use System Default	Ŧ
		Enable Single Number Reach Ring this phone and my business phone at th	e same time when my business line(s) is dialed.	
		Enable Move to Mobile If this is a mobile phone, transfer active calls	to this phone when the mobility button on your Cise	co IP Phone is p
✓ Enable Extend and Connect Allow this phone to be controlled by CTI applications (e.g. Jabber)				
		CTI Remote Device*	CTIRDtestuser1	Ŧ
		_ Timer Information		

• 在本例中,我使用3001作为远程目标号码。此远程目标号码应为外部号码(Jabber注册的 CUCM集群外部的号码,例如其他电话系统)

步骤3.将CTI RD配置文件关联到最终用户。

Device Information					
Controlled Devices	BOTTEST1 CIPCTEST1 CSFTEST1 CTIPDtecturer1	*			
	CTIRDlestuseri	÷			

步骤4.登录Jabber后,您将看到一个选项,用于设置Jabber电话服务以使用扩展和连接设备(使用 其他号码进行呼叫)。 使用"编辑编号"选项时,新编号应具有匹配的路由模式。

Re	ecents	✓ Other contacts		
_		a torturar2@circo.com	-	
	ŗ	Use my computer for calls		
		Use my desk phone for calls		
		Use other number for calls	30	01
	X	Disable phone services	Ed	lit number
	R.,	Forward calls to	Delete number	

• 将Jabber设置为使用扩展和连接设备后,Jabber上的电话图标将显示如下。

	testuser3@cisco.com
More	
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网络图

•出站Jabber扩展和连接呼叫的呼叫流程如下图所示



故障排除示例

在本示例中,当远程目标("其他号码")振铃时,它不显示主叫方号码。因此,他们无法使用扩展 和连接区分呼叫是来自外部方还是来自Jabber。使用扩展和连接时,CUCM会向远程设备发起呼叫 ,并且默认情况下不发送主叫方信息。

在以下Extend and Connect呼叫的数字分析摘要中,我们可以看到CallingPartyNumber字段为空。

```
16766318.007 |19:17:23.127 |AppInfo |Digit analysis: patternUsage=5
16766318.008 |19:17:23.127 |AppInfo |Digit analysis: match(pi="1", fqcn="", cn="",plv="5",
pss="test:Phones", TodFilteredPss="test:Phones", dd="3001",dac="0")
16766318.009 |19:17:23.127 |AppInfo |Digit analysis: analysis results
16766318.010 |19:17:23.127 |AppInfo |PretransformCallingPartyNumber=
|CallingPartyNumber=
|DialingPartition=Phones
|DialingPattern=3001
|FullyQualifiedCalledPartyNumber=3001
|DialingPatternRegularExpression=(3001)
|DialingWhere=
PatternType=Enterprise
PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=3001
```

|PretransformTagsList=SUBSCRIBER PretransformPositionalMatchList=3001 CollectedDigits=3001 |UnconsumedDigits= |TagsList=SUBSCRIBER |PositionalMatchList=3001 VoiceMailbox= VoiceMailCallingSearchSpace=Global Learned E164 Numbers:Directory URI:Phones VoiceMailPilotNumber=88800 RouteBlockFlag=RouteThisPattern RouteBlockCause=0 |AlertingName= UnicodeDisplayName= |DisplayNameLocale=1 OverlapSendingFlagEnabled=0 WithTags= 在SIP INVITE中,主叫方号码可在sip之后看到:标记。

在下面的节选中,可以看到主叫方号码未包含在INVITE中(sip:10.66.87.195),并且发送的主叫方名称显示为VoiceConnect。

16766935.001 |19:17:25.831 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.66.87.204 on port 5060 index 1146 [1276581,NET] INVITE sip:3001@10.66.87.204:5060;transport=tcp SIP/2.0 Via: SIP/2.0/TCP 10.66.87.195:5060;branch=z9hG4bK6dae5b551945 From: "VoiceConnect"

;tag=634549~59c9c4bc-724d-e1f0-017a-a8992d4fc521-19395629 To: <sip:3001@10.66.87.204>;tag=325889~2a8670d1-cf49-4a53-ae8f-36c41a8e75cf-23913736 Date: Thu, 18 May 2017 09:17:25 GMT Call-ID: cbe81900-91d166a3-6d704-c357420a@10.66.87.195 Supported: timer, resource-priority, replaces User-Agent: Cisco-CUCM10.5 Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY CSeq: 105 INVITE Max-Forwards: 70 Expires: 180 Allow-Events: presence Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=VIDEO_UNSPECIFIED Supported: X-cisco-srtp-fallback Supported: Geolocation Session-Expires: 1800;refresher=uas Min-SE: 1800 P-Asserted-Identity: <sip:1003@10.66.87.195> Remote-Party-ID: <sip:1003@10.66.87.195>;party=calling;screen=yes;privacy=off Contact: <sip:10.66.87.195:5060;transport=tcp> Content-Length: 0

要在远程设备上接收主叫方号码,需要将其配置为以下之一:

- 中继配置上的主叫方转换掩码
- 路由模式上的主叫方转换掩码
- 思科网关上的语音转换规则

当在路由模式(主叫方转换掩码)上配置中继直接拨入(DID)号码时,数字分析显示 CallingPartyNumber字**段已**更新。 16759993.008 |19:12:08.414 |AppInfo |Digit analysis: match(pi="1", fqcn="", cn="",plv="5", pss="test:Phones", TodFilteredPss="test:Phones", dd="3001",dac="0") 16759993.009 |19:12:08.414 |AppInfo |Digit analysis: analysis results 16759993.010 |19:12:08.414 |AppInfo ||PretransformCallingPartyNumber= CallingPartyNumber=777777 DialingPartition=Phones DialingPattern=3001 |FullyQualifiedCalledPartyNumber=3001 DialingPatternRegularExpression=(3001) |DialingWhere= PatternType=Enterprise PotentialMatches=NoPotentialMatchesExist DialingSdlProcessId=(0,0,0) |PretransformDigitString=3001 |PretransformTagsList=SUBSCRIBER PretransformPositionalMatchList=3001 CollectedDigits=3001 UnconsumedDigits= TagsList=SUBSCRIBER PositionalMatchList=3001 |VoiceMailbox= VoiceMailCallingSearchSpace=Global Learned E164 Numbers:Directory URI:Phones VoiceMailPilotNumber=88800 RouteBlockFlag=RouteThisPattern RouteBlockCause=0 |AlertingName= UnicodeDisplayName= DisplayNameLocale=1 OverlapSendingFlagEnabled=0 WithTags= 到远程目的地的SIP INVITE将主叫方号码显示为TRUNK DID。这会导致当CTI RD振铃时,中继

16484506.001 |18:32:10.720 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.66.87.204 on port 5060 index 951 [1255331,NET] INVITE sip:3001@10.66.87.204:5060 SIP/2.0 Via: SIP/2.0/TCP 10.66.87.195:5060;branch=z9hG4bK6bd621bee81d7 From: "VoiceConnect"

DID显示为主叫方号码。

ag=624206~59c9c4bc-724d-e1f0-017a-a8992d4fc521-19395539 To: <sip:3001@10.66.87.204> Date: Wed, 17 May 2017 08:32:10 GMT Call-ID: 506b6680-91c10a8a-6ba4d-c357420a@10.66.87.195 Supported: timer, resource-priority, replaces Min-SE: 1800 User-Agent: Cisco-CUCM10.5 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: <sip:10.66.87.195:5060>;method="NOTIFY;Event=telephone-event;Duration=500" Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=VIDEO_UNSPECIFIED Cisco-Guid: 1349215872-0000065536-0000000144-3277275658 Session-Expires: 1800 P-Asserted-Identity: "VoiceConnect" <sip:777777@10.66.87.195> Remote-Party-ID: "VoiceConnect" <sip:77777@10.66.87.195>;party=calling;screen=yes;privacy=off Contact: <sip:777777@10.66.87.195:5060;transport=tcp>;isFocus
Max-Forwards: 70
Content-Length: 0