配置Jabber擴展、連線並修改主叫方顯示

目錄

<u>簡介</u> <u>必要條件</u> <u>需求</u> <u>採用元件</u> <u>設定</u> <u>網路圖表</u> 疑難排解範例

簡介

本文檔介紹如何在Jabber中配置擴展和連線功能,以及如何修改遠端目標上顯示的主叫方。

必要條件

Cisco Unified Communications Manager(CUCM)9.1或更高版本。

Jabber 9.1或更高版本。

需求

需要有關使用Cisco Unified Communications Manager和IM and Presence Server配置Jabber的先前經驗和知識。

採用元件

本檔案中的資訊是根據以下軟體版本:

- Jabber 11.8.2
- 思科整合通訊管理員11.0.1.10000-10
- •即時消息和狀態伺服器(IMP)11.0.1.10000-6

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組態可能造成的影響。

設定

步驟1.為已配置Jabber的同一使用者配置CTI遠端裝置(CTI RD)電話配置檔案。

Phone Configuration				
🔜 Save 🗶 Delete 🗋 Copy 🌯 Reset 🧷 Apply Config 🕂	Add New			
Status Status: Ready				
Association 1 •mit Line [1] - 1001 in Phones Phone Type Product Type: CTI Remote Device				
2 emi Line [2] - Add a new DN Registration: Registered with Cisco Unified Communications Manager IPv4 Address:				
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時,請關聯到同一Jabber使用者。線路配置將與Jabber客戶端服務框架(CSF)裝置 線路相同
- •需要正確配置重新路由呼叫搜尋空間,遠端目標呼叫才能正常工作

步驟2.配置遠端目標。

Remote Destination Confi	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*	3001	
		Owner User ID*	testuser1	•
		Enable Unified Mobility features		
		Remote Destination Profile*	Not Selected	•
		Single Number Reach Voicemail Policy*	Use System Default	-
		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 Cisco	IP Phone is p
	Enable Extend and Connect Allow this phone to be controlled by CTI applications (e.g. Jabber)			
		CTI Remote Device*	CTIRDtestuser1	T
		- Timer Information		

• 在本例中,我使用3001作為遠端目的地號碼。此遠端目標號碼應為外部號碼(註冊了Jabber的 CUCM群集外部的號碼,例如另一個電話系統)

步驟3.將CTI RD配置檔案關聯到終端使用者。

BOTTEST1	*
CSFTEST1	
CTIRDtestuser1	-
	BOTTEST1 CIPCTEST1 CSFTEST1 CTIRDtestuser1

步驟4.登入到Jabber後,您將看到一個選項,用於將Jabber電話服務設定為使用Extend and Connect裝置(使用其他號碼進行呼叫)。 使用「編輯號碼」選項時,應該為新號碼設定匹配的路 由模式。

Re	ecents	✓ Other contacts			
_		e torturor2@circo.com			
	Ċ	Use my computer for calls	•		
		Use my desk phone for calls			
		Use other number for calls		3001	
	X	Disable phone services		Edit number	
	R>	Forward calls to	•	Delete number	

	90月10日10日11月11日11日11日11日11日11日11日11日11日11日11日11日
	• testuser3@cisco.com
More	
*	

• 將Jabber設定為使用擴展和連線裝置後,電話圖示將顯示在Jabber上,如下所示。

網路圖表

•出站Jabber擴展和連線呼叫的呼叫流程如下圖所示



疑難排解範例

在此示例中,當遠端目標(「其他號碼」)振鈴時,它不會顯示主叫方號碼。因此,他們無法使用 Extend and Connect區分來自外部方還是來自Jabber。使用Extend and Connect時,CUCM會向遠 端裝置發起呼叫,預設情況下不會傳送呼叫方資訊。

在下面的Extend and Connect呼叫的數字分析摘錄中可以看到,CallingPartyNumber欄位為空。

```
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,可以看到sip後面的主叫方編號:From標頭中的標簽。

在下面的摘錄中,可以看到INVITE(sip:10.66.87.195)中未包含主叫方號碼,並且正在傳送的主叫方 名稱顯示為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

要在遠端裝置上接收主叫方號碼,需要將其配置為以下其中一項:

• 中繼配置上的呼叫方轉換掩碼

- 路由模式上的主叫方轉換掩碼
- 思科網關上的語音轉換規則

當在路由模式(主叫方轉換掩碼)上配置中繼直接撥入(DID)號碼時,數字分析顯示 CallingPartyNumber欄位已更新。 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= 遠端目標的SIP INVITE將主叫方號碼顯示為中繼DID。這會導致在CTI RD振鈴時中繼的DID顯示為

<u>主叫方號碼。</u>

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"

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