瞭解CUSP術語和路由邏輯

目錄

簡介 必要條件 需求 技術 定義 網路拓撲 通話範例 基本呼叫路由 組態 關鍵配置元素 完整配置 <u>疑難排解</u> <u> 跟蹤級別配置</u> 跟蹤收集 <u> 跟蹤順序</u> <u> 觸發條件跟蹤示例</u> <u>路由跟蹤示例</u> <u>SIP-Wire-Log跟蹤示例</u> 架構參考

簡介

本檔案將說明思科整合SIP代理(CUSP)通話路由邏輯。

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必要條件

需求

思科建議您瞭解以下主題:

- •作業階段啟始通訊協定(SIP)一般知識
- 語音網路部署中CUSP的概念理解

技術

定義

字詞 定義

SIP網路是本地介面的邏輯集合,在一般路由用途中可以將其視為同一介面。

從<<u>http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusp/rel9_1/gui_configuration/en_Us</u> 網路在邏輯上定義網路的區域。可以使用CUSP裝置上的介面定義網路,也可以使用特定埠提 ,可以配置單獨的偵聽埠。

網路 (範例: 偵聽埠14.50.245.9:**5060**、14.50.245.9:**5062**、14.50.245.9:**5065**可以使用單個CUSP 網路在邏輯上定義後,便可用於根據網路配置觸發器。

> **附註**:如果設定了偵聽埠,請確保將流量傳送到CUSP的裝置使用正確的埠。如果為CU 14.50.245.9:**5065**,則必須確保CUCM將流量傳送到埠5065,而不是預設的5060。

觸發器 可以設定觸發器以標識傳入消息。

觸發器可以識別入站網路、本地埠和遠端網路等。 伺服器組定義Cisco Unified SIP Proxy系統為每個網路互動的元素。 從

伺服器組 <<u>http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusp/rel9_1/gui_configuration/en_US/c</u>
ml>

伺服器組和路由組都可以用作路由表中的目標。 伺服器組通常用於相同型別的冗餘裝置。 CU 例。

路由組允許您指定網關和中繼的選擇順序。它允許您為傳出中繼選擇確定網關和埠清單的優先 從

<<u>http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusp/rel9_1/gui_configuration/en_US/c</u>

路由組

伺服器組和路由組都可以用作路由表中的目標。 路由組通常定義到達同一裝置的加權組目標。 直接到CUCM的SIP中繼和到PSTN網關的SIP中繼以到達CUCM是路由組的典型示例。 到CU(,而PSTN路由則是備份。

您可以配置路由表,將SIP請求定向到其相應的目的地。每個路由表都包含一組基**於查**找策略近 從

<<u>http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusp/rel9_1/gui_configuration/en_US/c</u>>

CUSP中的路由表類似於第3層路由表。CUSP路由表由與第3層路由表中的網路類似的金鑰組成

在CUSP路由表中鍵可以對映到以下路由型別以路由SIP消息:

路由表 <u>destination</u>:可以將特定主機或本地配置的伺服器組配置為目標 route-group:具有一個或多個元素的本地配置的路由組 route-policy:路由策略可用於在路由表之間移動,類似CUCM中的轉換模式 回應:CUSP可以傳送特定響應來終止呼叫嘗試,而不是路由SIP消息 <u>default-sip</u>: 遵循RFC 3263的簡單路由。

附註:如果將**Key**對映到route-policy,請注意邏輯環路。

- 路由表名稱:"FromCUCM105-RT"
- 路由策略 查詢鍵匹配項:"Prefix-Longest-Match" 查詢鍵:"SIP報頭:「收件人」電話」 通過將Key的定義與Key的配置值相分離,同一路由表能以不同方式使用。 例如,一個路由策 TO的前綴:報頭,而另一個路由策略可以將路由表的金鑰定義為FROM的字首:標題。
- 路由觸發器 邱田胸發語府胸發語匡相到邱田來唱。 從邏輯上說,如果SIP消息與觸發器匹配,則使用配置的路由策略。

總而言之,根據SIP偵聽埠使用Network標籤SIP消息。 Network可用於匹配Trigger。 Route Policy然後根據Trigger確定要使用的Route Table,並定義在何處查詢Key。 Route Table然後使用 Key查詢將SIP消息路由到的位置(路由型別)。 路由型別(主機、**伺服器組、路由組**等)將用於將

網路拓撲



通話範例

從CUCM115上的PSTN 1001到2003的呼叫

基本呼叫路由

- 傳入網路:"PSTN"
- 觸發程序: "From-PSTN-Trigger"

如果傳入消息與網路「PSTN」匹配,則觸發

路由觸發器: "FromPSTN-RPolicy" "From-PSTN-Trigger"

連結「From-PSTN-Trigger」至「FromPSTN-RPolicy」

路由策略: "FromPSTN-RPpolicy"

指定路由表「PSTN-RT」

指定查詢鍵匹配「Prefix-Longest-Match」

指定查詢金鑰為「SIP報頭:「收件人」電話」

路由表:"PSTN-RT"

包含要轉至路由組「<u>CUCM115</u>RG」的金鑰「2」

路由組(或伺服器組):"CUCM115_RG"

包含元素14.50.245.20:5065

這些配置結合起來形成邏輯語句:

對於來自PSTN的呼叫(電話號碼字首為2),路由到14.50.245.20:5065

組態

PSTN - 2XXX和5XXX呼叫通過CUBE和vCUBE傳送到CUSP

CUCM 10.5 - 1XXX和2XXX通過SIP中繼傳送到CUSP

CUCM 11.5 - 1XXX和5XXX通過SIP中繼傳送到CUSP

附註:使用GUI時,必須提交某些配置才能在其他配置部分獲得這些配置。 這些標籤有 ###Commit Configuration

GUI配置

關鍵配置元素

CLI組態

建立網路 配置>>網路>>新增 Network Name: PSTN Type: standard + Allow Outbound Connections Enable 🥺 Disable 🤇 SIP Header Hiding Hide VIA: UDP Settings Maximum Packet Size: 1500 TCP Settings TCP Connection Setup Timeout (ms): 1000 TLS Certificate Verification Setting; Verify Client Certificate: V Verify Server Certificate; Add Cancel 定義偵聽埠以標識網路「PSTN」 配置>>網路>> [網路名稱] >> SIP偵聽點>>新增

sip listen PSTN udp 14.50.245.9 5060

sip網路PSTN標準

	Network 'PSTN' Listen Point
	Listen Point O IP Address: 14.50.245.9 O Port: 5060 O Transport Type: udp
入 站網路「 觸發條件From-PSTN-Trigger 序列1 網路內^\QPSTN\E\$ 結束序列 結束觸發條件	Materian PSTN」的觸發器 配置>>/觸發器>>新增 配置觸發器名稱 Trigger (New) Name: :m-PSTN-Trigger Inger Rates Image: Cancel
	Trigger Condition Inbound Network Is exactly PSTN Add Trigger Conditions Condition Inbound Network is exactly PSTN Remove Cancel
指定「CUC	M115_RG」的目標 配置>>路由組>>新增(### Commit 配置) 配置路由組名稱 Route Group (New)
路由組CUCM115_RG element target-destination 14.50.245.20:5065:udp CUCM115 q-value 0.0 failover-codes 502 - 503 重量50 end元素 結束路由	Name CUCNIIIS_RC Options Enable time of day routing: Enable weight based routing National Cancel 按一下「元素」列下的「按一下此處」,然後按一 新增」 輸入要素目標

Route Group 'CUCM115_RG' Element (New)			
Target Destination 💿 Ne	ext Hop 🔘		
Target Destination			
O Host / Server Group:	14.50.245.20		
Port	5060		
Transport Type:	udp -		
Next Hop SIP URI:			
O Network	CUCM115 -		
O Q-Value:	1		
o Weight	50		
Time Policy:	None -		
Failover Response C	Codes: 502,503		

Add Cancel

定義路由表並將一個鍵關聯到目標

配置>>	▶路由表>	>新增(###	Commit配置)
	мних-		

配置路由表名稱	
Route Tables	
Route Table	
O Name: PSTN-RT	
Add Cancel	
輸入金鑰和目標	
Route Table 'PSTN-RT' Route (New)	
Candidate Value	
G Key 2	
Route Type route-group -	
Route Group CUCM115_RG	
Add Cancel	

在**路由表**中將**路由組**配置為目的地時,不要新增埠 輸型別。 通過新增埠和/或傳輸型別,您指示CUSF DNS主機條目Cubestack:5060:UDP,而不是查詢本 要的伺服器組配置。

路由表PSTN-RT 金鑰2組CUCM115_RG key 5組CUCM105_RG 結束路由表

			_
	Candidate value		
	Key*		
	Route Type destination •		
	Target Destination Next Hop	Both O	
	Target Destination		
	Host/Server Group: Cubestack		
	Port		
	Transport Type: none 👻		
	Network PSTN		
定義「FromF	PSTN-RPolicy」 的金鑰 配置>>路由策略>>新增(#) 配置路由策略名稱 Route Policy (New)	/#Commit配置)	
	, (,		
	Name: omPSTN-RPolicy		
	Route Policy Steps		
	C State		Key
	No data to display	1	
	Add Remove Revert	∧ Move to ∨	
	點選Add新增策略步驟		
	Route Policy Step (New)		
	Route Table		
	Name:	PSTN-RT -	
	Lookup Key Matches:	Prefix-Longest-Match +	
	Case Sensitive:		
	Route Table Lookup Key		
	Lookup Key:	SIP Header + To	Pt
	Lookup Key Modifiers		
	Regular Expression Match:		
	Regular Expression Replace:		
	Remove leading '+' symbol:		
	Remove separator characters:		

■Add ■ Cancel 策略步驟將定義金鑰的使用方式。 在這種情況下, 將在To:上查詢匹配的最長電話號碼SIP報頭中的欄

將「From-PSTN-Trigger」連結到「FromPSTN-RPolicy」

配置>>路由觸發器>>新增

選擇要連結到觸發器的路由策略

Routing	g Trigg	er (New)
---------	---------	---------	---

Routing Policy: Trigger:	FromPSTN-RPolicy From-PSTN-Trigger	•
Add Cancel		

來自PSTN-RPpolicy的策略查詢 序列100 PSTN-RT報頭到uri元件電話 規則字首 結束序列 結束策略

觸發路由序列2策略FromPSTN-RPpolicy condition From-PSTN-Trigger

附註:show configuration active verbose將顯示包括路由表在內的整個配置。

```
josmeado-CUSP(cusp)# show configuration active verbose
Building CUSP configuration...
server-group sip global-load-balance weight
server-group sip retry-after 250
server-group sip element-retries udp 2
server-group sip element-retries tls 1
server-group sip element-retries tcp 1
sip dns-srv
enable
no naptr
end dns
1
no sip header-compaction
no sip logging
1
sip max-forwards 70
sip network CUCM105 standard
no non-invite-provisional
allow-connections
no tls verify
retransmit-count invite-client-transaction 3
retransmit-count invite-server-transaction 5
retransmit-count non-invite-client-transaction 3
retransmit-timer T1 500
retransmit-timer T2 4000
retransmit-timer T4 5000
retransmit-timer TU1 5000
retransmit-timer TU2 32000
retransmit-timer clientTn 64000
retransmit-timer serverTn 64000
tcp connection-setup-timeout 1000
tls handshake-timeout 3000
udp max-datagram-size 1500
end network
1
sip network CUCM115 standard
no non-invite-provisional
allow-connections
no tls verify
retransmit-count invite-client-transaction 3
retransmit-count invite-server-transaction 5
retransmit-count non-invite-client-transaction 3
retransmit-timer T1 500
retransmit-timer T2 4000
retransmit-timer T4 5000
retransmit-timer TU1 5000
retransmit-timer TU2 32000
retransmit-timer clientTn 64000
retransmit-timer serverTn 64000
tcp connection-setup-timeout 1000
tls handshake-timeout 3000
udp max-datagram-size 1500
end network
Ţ.
sip network PSTN standard
```

```
no non-invite-provisional
allow-connections
no tls verify
retransmit-count invite-client-transaction 3
retransmit-count invite-server-transaction 5
retransmit-count non-invite-client-transaction 3
retransmit-timer T1 500
retransmit-timer T2 4000
retransmit-timer T4 5000
retransmit-timer TU1 5000
retransmit-timer TU2 32000
retransmit-timer clientTn 64000
retransmit-timer serverTn 64000
tcp connection-setup-timeout 1000
tls handshake-timeout 3000
udp max-datagram-size 1500
end network
1
sip overload reject retry-after 0
1
no sip peg-counting
!
sip privacy service
sip queue message
drop-policy head
low-threshold 80
size 2000
thread-count 20
end queue
1
sip queue radius
drop-policy head
low-threshold 80
size 2000
thread-count 20
end queue
!
sip queue request
drop-policy head
low-threshold 80
size 2000
thread-count 20
end queue
!
sip queue response
drop-policy head
low-threshold 80
size 2000
thread-count 20
end queue
!
sip queue st-callback
drop-policy head
low-threshold 80
size 2000
thread-count 10
end queue
!
sip queue timer
drop-policy none
low-threshold 80
size 2500
thread-count 8
 end queue
```

```
I.
sip queue xcl
drop-policy head
low-threshold 80
size 2000
thread-count 2
end queue
1
route recursion
1
sip tcp connection-timeout 30
sip tcp max-connections 256
1
no sip tls
!
sip tls connection-setup-timeout 1
1
trigger condition From-CUCM105-Trigger
sequence 1
 in-network ^\QCUCM105\E$
 end sequence
end trigger condition
1
trigger condition From-CUCM115-Trigger
sequence 1
 in-network ^\QCUCM115\E$
 end sequence
end trigger condition
!
trigger condition From-PSTN-Trigger
sequence 1
 in-network ^\QPSTN\E$
 end sequence
end trigger condition
1
trigger condition mid-dialog
sequence 1
 mid-dialog
 end sequence
end trigger condition
!
accounting
no enable
no client-side
no server-side
end accounting
1
server-group sip group Cubestack PSTN
element ip-address 14.50.245.6 5060 udp q-value 0.0 weight 1
element ip-address 14.50.245.7 5060 udp q-value 0.0 weight 1
failover-resp-codes 503
lbtype weight
ping
end server-group
!
route group CUCM105_RG
element target-destination 14.50.245.25:5062:udp CUCM105 q-value 0.0
 failover-codes 510
 weight 50
  end element
 end route
!
route group CUCM115_RG
 element target-destination 14.50.245.20:5065:udp CUCM115 q-value 0.0
```

```
failover-codes 502 - 503
  weight 50
  end element
 end route
Ţ
route table FromCUCM105-RT
key * target-destination Cubestack PSTN
key 2 group CUCM115_RG
end route table
1
route table FromCUCM115-RT
key 1 target-destination Cubestack PSTN
key 5 group CUCM105_RG
end route table
Ţ
route table PSTN-RT
key 2 group CUCM115_RG
key 5 group CUCM105_RG
end route table
1
policy lookup FromCUCM105-RPolicy
 sequence 100 FromCUCM105-RT header to uri-component phone
 rule prefix
 end sequence
end policy
1
policy lookup FromCUCM115-RPolicy
sequence 100 FromCUCM115-RT header to uri-component phone
 rule prefix
 end sequence
end policy
1
policy lookup FromPSTN-RPolicy
sequence 100 PSTN-RT header to uri-component phone
 rule prefix
  end sequence
end policy
1
trigger routing sequence 1 by-pass condition mid-dialog
trigger routing sequence 2 policy FromPSTN-RPolicy condition From-PSTN-Trigger
trigger routing sequence 3 policy FromCUCM115-RPolicy condition From-CUCM115-Trigger
trigger routing sequence 4 policy FromCUCM105-RPolicy condition From-CUCM105-Trigger
1
server-group sip global-ping
1
no server-group sip ping-503
1
sip cac session-timeout 720
sip cac PSTN 14.50.245.6 5060 udp limit -1
sip cac PSTN 14.50.245.7 5060 udp limit -1
1
no sip cac
1
sip listen CUCM105 udp 14.50.245.9 5062
sip listen CUCM115 udp 14.50.245.9 5065
sip listen PSTN udp 14.50.245.9 5060
1
call-rate-limit 100
Т
end
```



跟蹤級別配置

在CUSP GUI中, 導航至故障排除>> Cisco Unified SIP Proxy >> Trace

Trigger-Conditions - Level:debug:這將顯示哪些觸發器匹配以啟動呼叫路由。

Routing - Level:debug:這將顯示呼叫路由過程中完成的操作。 已匹配哪個金鑰、選擇了哪個目標 等。

SIP-Wire-Log - Level:debug:這將顯示接收和傳送的SIP消息。

跟蹤收集

通過GUI

在CUSP GUI中, 導航至故障排除>> Cisco Unified SIP Proxy >> Trace

選擇下載日誌檔案

您還可以清除日誌

通過FTP客戶端

預設情況下,沒有具有FTP許可權的帳戶。 要啟用具有FTP許可權的帳戶,請將使用者新增到 PFS組。

josmeado-CUSP# user platformadmin group ?
Administrators System administrators group
pfs-privusers PFS privileged users group
pfs-readonly PFS read only group
josmeado-CUSP# user platformadmin group pfs

·通過FTP客戶端連線到CUSP。 檔案路徑:cusp >> log >> trace >> trace.log

跟蹤順序

- 1. SIP-Wire-Log 傳入SIP邀請
- 2. SIP-Wire-Log 返回100嘗試
- 3. Trigger-Condition 確定網路和觸發路由策略
- 4. 路由 有關詳細資訊,請參閱下面的路由跟蹤部分
- 5. SIP-Wire-Log 向目標傳送邀請
- 6. SIP-Wire-Log 繼續正常的SIP事務,直到每個呼叫段都出現200 Ok消息

觸發條件跟蹤示例

13:24:36:987 08:17:2017 vCUSP,9.1.5,josmeado-CUSP,14.50.245.9,trace.log [REQUESTI.7] DEBUG 2017.08.17 13:25:03:006 conditions.RegexCondition - inNetwork='PSTN' [REQUESTI.7] DEBUG 2017.08.17 13:25:03:006 conditions.RegexCondition - IN_NETWORK: PSTN [REQUESTI.7] DEBUG 2017.08.17 13:25:03:006 conditions.AbstractRegexCondition pattern(^\QPSTN\E\$), toMatch(PSTN) returning true [REQUESTI.7] DEBUG 2017.08.17 13:25:03:006 triggers.ModuleTrigger - ModuleTrigger.eval()
action<FromPSTN-RPolicy> actionParameter<>
[REQUESTI.7] DEBUG 2017.08.17 13:25:03:006 triggers.ModuleTrigger - ModuleTrigger.eval() got the
policy, executing it ...

在上方範例中,我們看到網路是以PSTN相符的,這用在路由原則「FromPSTN-RPolicy」中。

路由跟蹤示例

13:29:13:453 08:17:2017 vCUSP,9.1.5,josmeado-CUSP,14.50.245.9,trace.log
[REQUESTI.7] DEBUG 2017.08.17 13:29:33:987 nrs.XCLNRSShiftRoutes - Entering
ShiftAlgorithms.execute()
[REQUESTI.7] DEBUG 2017.08.17 13:29:33:987 modules.XCLLookup - Entering execute()
[REQUESTI.7] DEBUG 2017.08.17 13:29:33:987 nrs.XCLPrefix - Entering getKeyValue()
[REQUESTI.7] DEBUG 2017.08.17 13:29:33:987 nrs.FieldSelector - getToUri: To header obtained To:

[REQUESTI.7] DEBUG 2017.08.17 13:29:33:987 nrs.FieldSelector - getUriPart: URI sip:2003@14.50.245.9 part 1 [REQUESTI.7] DEBUG 2017.08.17 13:29:33:987 nrs.FieldSelector - Requested field 52 [REQUESTI.7] DEBUG 2017.08.17 13:29:33:987 nrs.FieldSelector - Returning key 2003 [REQUESTI.7] DEBUG 2017.08.17 13:29:33:987 nrs.XCLPrefix - Leaving getKeyValue() [REQUESTI.7] DEBUG 2017.08.17 13:29:33:987 modules.XCLLookup - table=PSTN-RT, key=2003 [REQUESTI.7] INFO 2017.08.17 13:29:33:987 modules.XCLLookup - table is PSTN-RT [REQUESTI.7] DEBUG 2017.08.17 13:29:33:987 routingtables.RoutingTable - Entering lookup() [REQUESTI.7] DEBUG 2017.08.17 13:29:33:987 routingtables.RoutingTable - Looking up 2003 in table PSTN-RT with rule prefix and modifiers=none [REQUESTI.7] DEBUG 2017.08.17 13:29:33:987 routingtables.RoutingTable - Entering applyModifiers() [REQUESTI.7] DEBUG 2017.08.17 13:29:33:987 routingtables.RoutingTable - Leaving applyModifiers(), returning 2003 [REQUESTI.7] DEBUG 2017.08.17 13:29:33:988 routingtables.RoutingTable - Leaving lookup() [REQUESTI.7] INFO 2017.08.17 13:29:33:988 nrs.XCLPrefix - NRS Routing decision is: RouteTable:PSTN-RT, RouteKey:2, RouteGroup:CUCM115_RG [REQUESTI.7] DEBUG 2017.08.17 13:29:33:988 loadbalancer.LBFactory - Entering createLoadBalancer() [REQUESTI.7] INFO 2017.08.17 13:29:33:988 loadbalancer.LBFactory - lbtype is 3(call-id) [REQUESTI.7] DEBUG 2017.08.17 13:29:33:988 loadbalancer.LBFactory - Leaving createLoadBalancer() [REQUESTI.7] DEBUG 2017.08.17 13:29:33:988 nrs.XCLPrefix - Stored NRSAlgResult=isFound=true, isFailure=false, Response=-1, Routes=[Ruri: 14.50.245.20:5065:udp, Route: null, Network: CUCM115, q-value=0.0radvance=[502, 503]], PolicyAdvance=null [REQUESTI.7] DEBUG 2017.08.17 13:29:33:988 nrs.NRSAlgResult - set policyAdvance as specified in route=RouteTable:PSTN-RT, RouteKey:2, RouteGroup:CUCM115_RG [REQUESTI.7] DEBUG 2017.08.17 13:29:33:988 nrs.NRSAlgResult - no policyAdvance specified in route [REQUESTI.7] DEBUG 2017.08.17 13:29:33:988 nrs.NRSAlgResult - set policyAdvance as specified in algorithm={lookuprule=1, lookupfield=52, lookuplenght=-1, lookuptable=PSTN-RT, sequence=100, algorithm=1} [REQUESTI.7] DEBUG 2017.08.17 13:29:33:988 nrs.NRSAlgResult - no policyAdvance specified in algorithm [REQUESTI.7] DEBUG 2017.08.17 13:29:33:988 modules.XCLLookup - Leaving execute() [REQUESTI.7] DEBUG 2017.08.17 13:29:33:988 nrs.XCLNRSShiftRoutes - Entering ShiftRoutes.execute() [REQUESTI.7] DEBUG 2017.08.17 13:29:33:988 loadbalancer.LBBase - Entering getServer() [REQUESTI.7] DEBUG 2017.08.17 13:29:33:988 loadbalancer.LBBase - Entering initializeDomains() [REQUESTI.7] DEBUG 2017.08.17 13:29:33:988 nrs.NRSRoutes - routes before applying time policies:

[Ruri: 14.50.245.20:5065:udp, Route: null, Network: CUCM115, q-value=0.0radvance=[502, 503]] [REQUESTI.7] DEBUG 2017.08.17 13:29:33:988 nrs.NRSRoutes -routes after applying time policies: [Ruri: 14.50.245.20:5065:udp, Route: null, Network: CUCM115, q-value=0.0radvance=[502, 503]] [REQUESTI.7] DEBUG 2017.08.17 13:29:33:988 loadbalancer.LBBase - Leaving initializeDomains() [REQUESTI.7] INFO 2017.08.17 13:29:33:988 loadbalancer.LBHashBased - list of elements in order on which load balancing is done : Ruri: 14.50.245.20:5065:udp, Route: null, Network: CUCM115, qvalue=0.0radvance=[502, 503], [REQUESTI.7] DEBUG 2017.08.17 13:29:33:988 loadbalancer.LBBase - Server group route-sg selected Ruri: 14.50.245.20:5065:udp, Route: null, Network: CUCM115, q-value=0.0radvance=[502, 503] [REQUESTI.7] DEBUG 2017.08.17 13:29:33:988 loadbalancer.LBBase - Leaving getServer() [REQUESTI.7] DEBUG 2017.08.17 13:29:33:988 loadbalancer.LBBase - Leaving getServer()

1. CUSP獲得TO:中的Key值標題

2. CUSP將金鑰標識為2003

- 3. CUSP在路由表中查詢金鑰
- 4. CUSP匹配路由表中的條目並標識目標RouteGroup:CUCM115_RG
- 5. CUSP在RouteGroup內應用負載均衡
- 6. CUSP標識它要向其傳送SIP消息的RouteGroup中的特定元素
- 7. CUSP應用時間策略(如果適用)

8. CUSP最終確定它將向其傳送SIP消息的元素

SIP-Wire-Log跟蹤示例

13:48:26:669 08:17:2017 vCUSP,9.1.5, josmeado-CUSP,14.50.245.9, trace.log [DsTransportListener-2] DEBUG 2017.08.17 13:48:52:221 DsSipLlApi.Wire - Received UDP packet on 14.50.245.9:5060 ,source 14.50.245.6:50683 INVITE sip:2003@14.50.245.9:5060 SIP/2.0 Via: SIP/2.0/UDP 14.50.245.6:5060; branch=z9hG4bK2A5763 Remote-Party-ID: <sip:1001@14.50.245.6>;party=calling;screen=no;privacy=off From: <sip:1001@14.50.245.6>;tag=4E329FEC-A9F To: <sip:2003@14.50.245.9> Date: Thu, 17 Aug 2017 13:48:52 GMT Call-ID: 2A7BE22B-82AB11E7-83AEAE0B-F940DC75@14.50.245.6 Supported: 100rel, timer, resource-priority, replaces, sdp-anat Min-SE: 1800 Cisco-Guid: 0350227076-2191790567-2162465606-1670485135 User-Agent: Cisco-SIPGateway/IOS-15.5.3.S4b Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER CSeq: 101 INVITE Timestamp: 1502992132 Contact: <sip:1001@14.50.245.6:5060> Expires: 180 Allow-Events: telephone-event Max-Forwards: 69 Content-Type: application/sdp Content-Disposition: session; handling=required Content-Length: 266 v=0o=CiscoSystemsSIP-GW-UserAgent 7317 4642 IN IP4 14.50.245.6 s=SIP Call

c=IN IP4 14.50.245.6 $t = 0 \quad 0$ m=audio 8266 RTP/AVP 18 127 c=IN IP4 14.50.245.6 a=rtpmap:18 G729/8000 a=fmtp:18 annexb=no a=rtpmap:127 telephone-event/8000 a=fmtp:127 0-16 a=ptime:20 --- end of packet ---[REQUESTI.7] DEBUG 2017.08.17 13:48:52:223 DsSipLlApi.Wire - Sending UDP packet on 14.50.245.9:32789, destination 14.50.245.6:5060 SIP/2.0 100 Trying Via: SIP/2.0/UDP 14.50.245.6:5060;branch=z9hG4bK2A5763 To: <sip:2003@14.50.245.9> From: <sip:1001@14.50.245.6>;tag=4E329FEC-A9F Call-ID: 2A7BE22B-82AB11E7-83AEAE0B-F940DC75@14.50.245.6 CSeq: 101 INVITE Timestamp: 1502992132 Content-Length: 0 [REQUESTI.7] DEBUG 2017.08.17 13:48:52:225 DsSipLlApi.Wire - Sending UDP packet on 14.50.245.9:32790, destination 14.50.245.20:5065 INVITE sip:2003@14.50.245.20:5065;transport=udp SIP/2.0 Via: SIP/2.0/UDP 14.50.245.9:5065; branch=z9hG4bKM3X51yKL9BEW5v0Kudc5Dw~~128 Via: SIP/2.0/UDP 14.50.245.6:5060;branch=z9hG4bK2A5763 Max-Forwards: 68 To: <sip:2003@14.50.245.9> From: <sip:1001@14.50.245.6>;tag=4E329FEC-A9F Contact: <sip:1001@14.50.245.6:5060> Expires: 180 Remote-Party-ID: <sip:1001@14.50.245.6>;party=calling;screen=no;privacy=off Call-ID: 2A7BE22B-82AB11E7-83AEAE0B-F940DC75@14.50.245.6 CSeq: 101 INVITE Content-Length: 266 Date: Thu, 17 Aug 2017 13:48:52 GMT Supported: 100rel, timer, resource-priority, replaces, sdp-anat Min-SE: 1800 Cisco-Guid: 0350227076-2191790567-2162465606-1670485135 User-Agent: Cisco-SIPGateway/IOS-15.5.3.S4b Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER Timestamp: 1502992132 Allow-Events: telephone-event Content-Type: application/sdp Content-Disposition: session; handling=required v=0o=CiscoSystemsSIP-GW-UserAgent 7317 4642 IN IP4 14.50.245.6 s=SIP Call c=IN IP4 14.50.245.6 t=0 0 m=audio 8266 RTP/AVP 18 127 c=IN IP4 14.50.245.6 a=rtpmap:18 G729/8000 a=fmtp:18 annexb=no a=rtpmap:127 telephone-event/8000 a=fmtp:127 0-16 a=ptime:20

14.50.245.9:5065 ,source 14.50.245.20:5065 SIP/2.0 100 Trying Via: SIP/2.0/UDP 14.50.245.9:5065; branch=z9hG4bKM3X51yKL9BEW5v0Kudc5Dw~~128, SIP/2.0/UDP 14.50.245.6:5060;branch=z9hG4bK2A5763 From: <sip:1001@14.50.245.6>;tag=4E329FEC-A9F To: <sip:2003@14.50.245.9> Date: Thu, 17 Aug 2017 17:48:52 GMT Call-ID: 2A7BE22B-82AB11E7-83AEAE0B-F940DC75@14.50.245.6 CSeq: 101 INVITE Allow-Events: presence Content-Length: 0 --- end of packet ---[DsTransportListener-3] DEBUG 2017.08.17 13:48:52:284 DsSipLlApi.Wire - Received UDP packet on 14.50.245.9:5065 ,source 14.50.245.20:5065 SIP/2.0 180 Ringing Via: SIP/2.0/UDP 14.50.245.9:5065; branch=z9hG4bKM3X51yKL9BEW5v0Kudc5Dw~~128, SIP/2.0/UDP 14.50.245.6:5060;branch=z9hG4bK2A5763 From: <sip:1001@14.50.245.6>;tag=4E329FEC-A9F To: <sip:2003@14.50.245.9>;tag=93896~37db7c49-96d4-4c4c-a223-626b2c74c16a-16919968 Date: Thu, 17 Aug 2017 17:48:52 GMT Call-ID: 2A7BE22B-82AB11E7-83AEAE0B-F940DC75@14.50.245.6 CSeq: 101 INVITE Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY Allow-Events: presence Server: Cisco-CUCM11.5 Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=DESKTOP Supported: X-cisco-srtp-fallback Supported: Geolocation Session-ID: 1e6e772300105000a00084b517ae1a83;remote=c07cdfa83b8f7c373757cf842ab93896 P-Asserted-Identity: "Alerting JM1 - 2003" <sip:2003@14.50.245.20> Remote-Party-ID: "Alerting JM1 - 2003" <sip:2003@14.50.245.20>;party=called;screen=yes;privacy=off Contact: <sip:2003@14.50.245.20:5065>;+u.sip!devicename.ccm.cisco.com="SEP84B517AE1A83" Content-Length: 0 --- end of packet ---[CT_CALLBACK.15] DEBUG 2017.08.17 13:48:52:285 DsSipLlApi.Wire - Sending UDP packet on 14.50.245.9:32789, destination 14.50.245.6:5060 SIP/2.0 180 Ringing Via: SIP/2.0/UDP 14.50.245.6:5060;branch=z9hG4bK2A5763 To: <sip:2003@14.50.245.9>;tag=93896~37db7c49-96d4-4c4c-a223-626b2c74c16a-16919968 From: <sip:1001@14.50.245.6>;tag=4E329FEC-A9F Contact: <sip:2003@14.50.245.20:5065>;+u.sip!devicename.ccm.cisco.com="SEP84B517AE1A83" Remote-Party-ID: "Alerting JM1 - 2003" <sip:2003@14.50.245.20>;party=called;screen=yes;privacy=off Call-ID: 2A7BE22B-82AB11E7-83AEAE0B-F940DC75@14.50.245.6 CSeq: 101 INVITE Content-Length: 0 Date: Thu, 17 Aug 2017 17:48:52 GMT Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY Allow-Events: presence Server: Cisco-CUCM11.5 Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=DESKTOP Supported: X-cisco-srtp-fallback Supported: Geolocation Session-ID: 1e6e772300105000a00084b517ae1a83; remote=c07cdfa83b8f7c373757cf842ab93896 P-Asserted-Identity: "Alerting JM1 - 2003" <sip:2003@14.50.245.20>

[DsTransportListener-3] DEBUG 2017.08.17 13:48:54:292 DsSipLlApi.Wire - Received UDP packet on 14.50.245.9:5065 ,source 14.50.245.20:5065 SIP/2.0 200 OK Via: SIP/2.0/UDP 14.50.245.9:5065; branch=z9hG4bKM3X51yKL9BEW5v0Kudc5Dw~~128, SIP/2.0/UDP 14.50.245.6:5060;branch=z9hG4bK2A5763 From: <sip:1001@14.50.245.6>;tag=4E329FEC-A9F To: <sip:2003@14.50.245.9>;tag=93896~37db7c49-96d4-4c4c-a223-626b2c74c16a-16919968 Date: Thu, 17 Aug 2017 17:48:52 GMT Call-ID: 2A7BE22B-82AB11E7-83AEAE0B-F940DC75@14.50.245.6 CSeq: 101 INVITE Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY Allow-Events: presence, kpml Supported: replaces Server: Cisco-CUCM11.5 Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=DESKTOP Supported: X-cisco-srtp-fallback Supported: Geolocation Session-Expires: 1800;refresher=uas Require: timer Session-ID: 1e6e772300105000a00084b517ae1a83;remote=c07cdfa83b8f7c373757cf842ab93896 P-Asserted-Identity: "CLID JM1 - 2003" <sip:2003@14.50.245.20> Remote-Party-ID: "CLID JM1 - 2003" <sip:2003@14.50.245.20>;party=called;screen=yes;privacy=off Contact: <sip:2003@14.50.245.20:5065>;+u.sip!devicename.ccm.cisco.com="SEP84B517AE1A83" Content-Type: application/sdp Content-Length: 258 v=0o=CiscoSystemsCCM-SIP 93896 1 IN IP4 14.50.245.20 s=SIP Call c=IN IP4 14.50.245.254 b=TIAS:8000 b = AS:8t=0 0 m=audio 16502 RTP/AVP 18 101 a=ptime:20 a=rtpmap:18 G729/8000 a=fmtp:18 annexb=no a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-15 --- end of packet ---[CT_CALLBACK.15] DEBUG 2017.08.17 13:48:54:293 DsSipLlApi.Wire - Sending UDP packet on 14.50.245.9:32789, destination 14.50.245.6:5060 SIP/2.0 200 OK Via: SIP/2.0/UDP 14.50.245.6:5060;branch=z9hG4bK2A5763 To: <sip:2003@14.50.245.9>;tag=93896~37db7c49-96d4-4c4c-a223-626b2c74c16a-16919968 From: <sip:1001@14.50.245.6>;tag=4E329FEC-A9F Contact: <sip:2003@14.50.245.20:5065>;+u.sip!devicename.ccm.cisco.com="SEP84B517AE1A83" Require: timer Remote-Party-ID: "CLID JM1 - 2003" <sip:2003@14.50.245.20>;party=called;screen=yes;privacy=off Call-ID: 2A7BE22B-82AB11E7-83AEAE0B-F940DC75@14.50.245.6 CSeq: 101 INVITE Content-Length: 258 Date: Thu, 17 Aug 2017 17:48:52 GMT Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY Allow-Events: presence, kpml Supported: replaces Supported: X-cisco-srtp-fallback Supported: Geolocation Server: Cisco-CUCM11.5 Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=DESKTOP Session-Expires: 1800; refresher=uas Session-ID: 1e6e772300105000a00084b517ae1a83;remote=c07cdfa83b8f7c373757cf842ab93896

P-Asserted-Identity: "CLID JM1 - 2003" <sip:2003@14.50.245.20> Content-Type: application/sdp v=0 o=CiscoSystemsCCM-SIP 93896 1 IN IP4 14.50.245.20 s=SIP Call c=IN IP4 14.50.245.254 b=TIAS:8000 b=AS:8 t=0 0 m=audio 16502 RTP/AVP 18 101 a=ptime:20 a=rtpmap:18 G729/8000 a=fmtp:18 annexb=no a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-15

SIP-Wire-Log顯示通常的SIP消息,最大為200 Okay,對於兩個呼叫段。



架構參考