

了解CUSP术语和路由逻辑

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简介

本文档介绍思科统一SIP代理(CUSP)呼叫路由逻辑。

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先决条件

要求

思科建议您了解以下主题：

- 会话发起协议(SIP)的一般知识
- 对语音网络部署中CUSP的概念性了解

术语

定义

术语	定义
----	----

SIP网络是本地接口的逻辑集合，出于一般路由目的，可以对其进行相同处理。

从http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusp/rel9_1/gui_configuration/en_US网络在逻辑上定义网络区域。网络可以使用CUSP设备上的接口进行定义，或者特定端口可用分段，可以配置单独的侦听端口。

(示例： 侦听端口14.50.245.9:5060、 14.50.245.9:5062、 14.50.245.9:5065 可以使用单个逻辑网络)

网络

在逻辑上定义网络后，可以根据网络配置触发器。

注意：如果设置侦听端口，请确保将流量发送到CUSP的设备使用正确的端口。 如果为14.50.245.9:5065，则必须确保CUCM将流量发送到端口5065，而不是默认端口5060。

触发器

可以设置触发器来识别传入消息。

触发器可以识别入站网络、本地端口、远程网络等。

服务器组定义思科统一SIP代理系统与每个网络交互的元素。

从

服务器组

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusp/rel9_1/gui_configuration/en_US/c
[ml](http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusp/rel9_1/gui_configuration/en_US/c)>

服务器组和路由组均可用作路由表中的目标。服务器组通常用于同一类型的冗余设备。 CUB示例。

路由组允许您指定选择网关和中继的顺序。它允许您为传出中继选择确定网关和端口列表的优先

从

路由组

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusp/rel9_1/gui_configuration/en_US/c
[l](http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusp/rel9_1/gui_configuration/en_US/c)>

服务器组和路由组均可用作路由表中的目标。路由组通常定义到达同一设备的加权组目标。到CUCM的直接SIP中继和到PSTN网关以到达CUCM的SIP中继是路由组的一个很好的示例。

选方法，而PSTN路由是备份。

您可以配置路由表，将SIP请求定向到相应的目标。每个路由表都包含一组基于查找策略匹配的从

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusp/rel9_1/gui_configuration/en_US/c
>
CUSP中的路由表与第3层路由表类似。CUSP路由表由类似于第3层路由表中网络的密钥组成。

路由表

在CUSP路由表中，可将密钥映射到以下路由类型以路由SIP消息：

目标: 可以将特定主机或本地配置的服务器组配置为目标

route-group: 具有一个或多个元素的本地配置的路由组

route-policy: 路由策略可用于在路由表之间移动，类似于CUCM中的转换模式

回复: CUSP可以发送特定响应来终止呼叫尝试，而不是路由SIP消息

default-sip: 遵循RFC 3263的简单路由。

注意：如果将Key映射到路由策略，请注意逻辑环路。

路由策略指向路由表并定义如何使用该路由表中的密钥。

示例：

路由策略

路由表名称："FromCUCM105-RT"

查找密钥匹配："前缀最长匹配"

查找密钥："SIP报头："收件人"电话"

通过将Key的定义与Key的配置值分开，同一路由表可以以不同方式使用。例如，一个路由策

TO的前缀：报头，而另一个路由策略可以将路由表的密钥定义为FROM的前缀：标题。

路由触发器

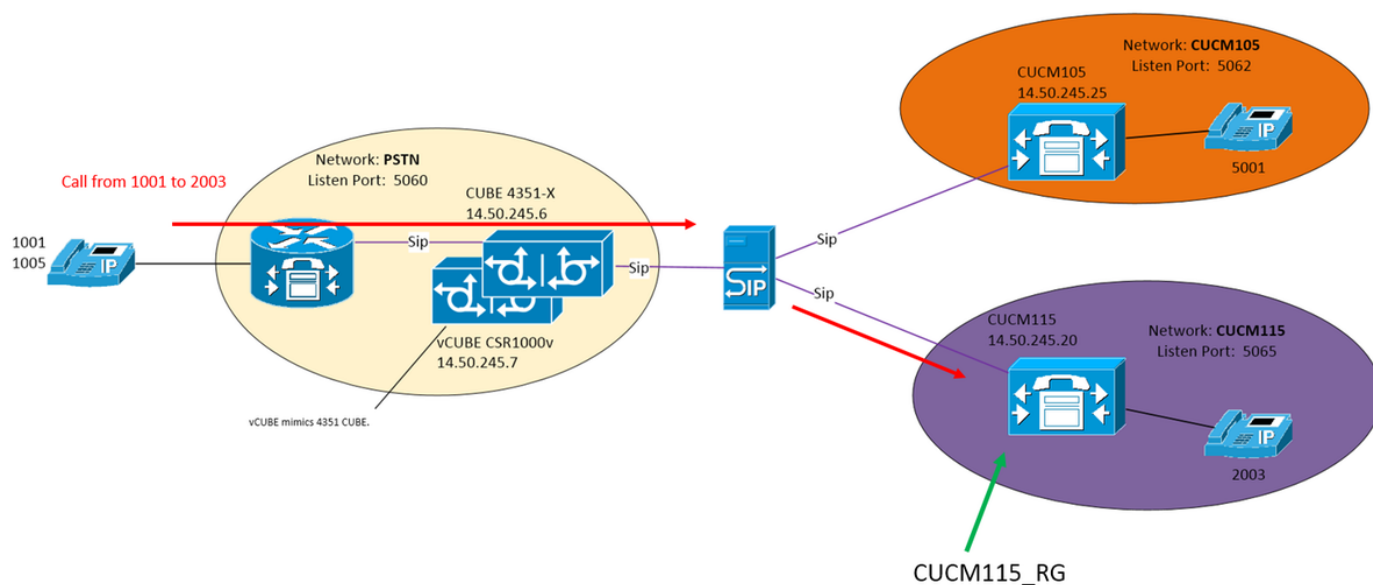
路由触发器将触发器链接到路由策略。

从逻辑上讲，如果SIP消息与触发器匹配，则使用已配置的路由策略。

总之，SIP消息基于SIP侦听端口用网络进行标记。网络可用于匹配触发器。然后，路由策略会根据触发器确定要使用的路由表，并定义查找密钥的位置。然后，路由表将使用Key查找在何处路由

SIP消息 (路由类型)。路由类型(主机、服务器组、路由组等)将用于将SIP消息发送到已配置的目标(元素)。

网络拓扑



呼叫示例

CUCM115上从PSTN 1001到2003的呼叫

基本呼叫路由

传入网络："PSTN"

触发器："从PSTN触发器"

如果传入消息与网络“PSTN”匹配，则触发

路由触发器："FromPSTN-RPolicy""From-PSTN-Trigger"

将“From-PSTN-Trigger”链接到“FromPSTN-RPolicy”

路由策略："FromPSTN-RPolicy"

指定路由表“PSTN-RT”

指定查找密钥匹配“前缀最长匹配”

指定查找密钥为“SIP报头：“收件人”电话”

路由表:"PSTN-RT"

包含要转到路由组“CUCM115_RG”的密钥“2”

路由组 (或服务器组) : "CUCM115_RG"

包含元素 14.50.245.20:5065

这些配置组合在一起可生成逻辑语句：

对于来自电话号码前缀为2的PSTN的呼叫，请路由到14.50.245.20:5065

配置

PSTN — 通过CUBE和vCUBE将2XXX和5XXX呼叫发送到CUSP

CUCM 10.5 - 1XXX和2XXX通过SIP中继发送到CUSP

CUCM 11.5 - 1XXX和5XXX通过SIP中继发送到CUSP

注意：使用GUI时，必须提交某些配置，才能在其他配置部分提供这些配置。 这些标有 **###Commit配置**

关键配置元素

CLI 配置


sip network PSTN标准

sip listen PSTN udp 14.50.245.9 5060

创建网络

GUI 配置

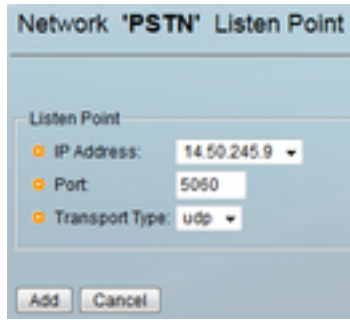
配置>>网络>>添加



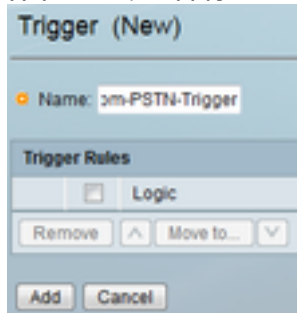
The screenshot shows the 'Network' configuration page in a GUI. The 'Name' field is set to 'PSTN' and the 'Type' is 'standard'. Under 'Allow Outbound Connections', 'Disable' is selected. Under 'SIP Header Hiding', 'Hide VIA' is unchecked. Under 'UDP Settings', 'Maximum Packet Size' is set to 1500. Under 'TCP Settings', 'TCP Connection Setup Timeout (ms)' is set to 1000. Under 'TLS Certificate Verification Setting', both 'Verify Client Certificate' and 'Verify Server Certificate' are checked. 'Add' and 'Cancel' buttons are at the bottom.

定义侦听端口以标识网络“PSTN”

配置>>网络>> [网络名称] >> SIP侦听点>>添加

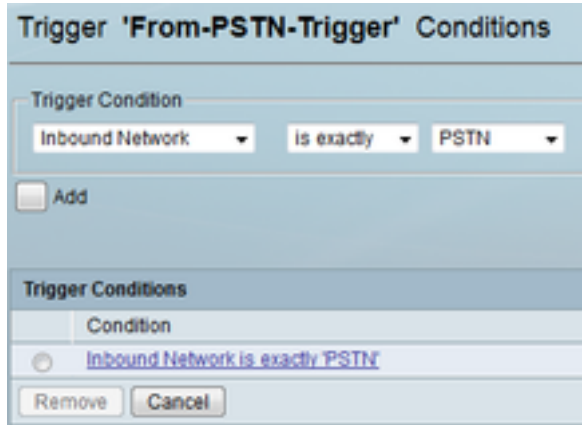


入站网络“PSTN”的触发器
配置>>触发器>>添加
配置触发器名称

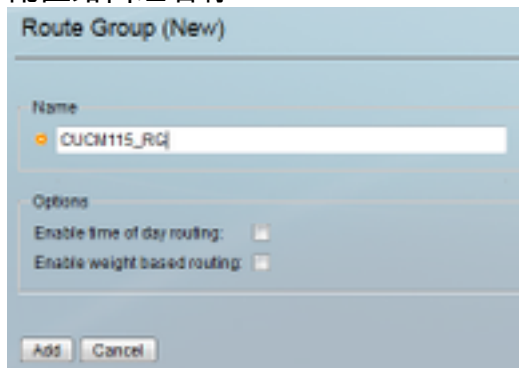


触发条件From-PSTN-Trigger
序列1
网内^\\QPSTN\\E\$
结束序列
结束触发条件

配置触发条件并点击添加



为“CUCM115_RG”指定目标
配置>>路由组>>添加(###Commit配置)
配置路由组名称



路由组CUCM115_RG
element target-destination 14.50.245.20:5065:udp
CUCM115 q-value 0.0
failover-codes 502 - 503
重量50
端元
结束路由

单击“元素列”下的“单击此处”，然后单击“添加”
输入要素目标

Route Group 'CUCM115_RG' Element (New)

Target Destination Next Hop

Target Destination

- Host / Server Group: 14.50.245.20
- Port: 5060
- Transport Type: udp

Next Hop

SIP URI:

Options

- Network: CUCM115
- Q-Value: 1
- Weight: 50
- Time Policy: None
- Fallover Response Codes: 502,503

Add Cancel

定义路由表并将密钥关联到目标

配置>>路由表>添加(###Commit配置)

配置路由表名称

Route Tables

Route Table

Name: PSTN-RT

Add Cancel

输入密钥和目标

Route Table 'PSTN-RT' Route (New)

Candidate Value

Key 2

Route Type route-group

Route Group CUCM115_RG

Add Cancel

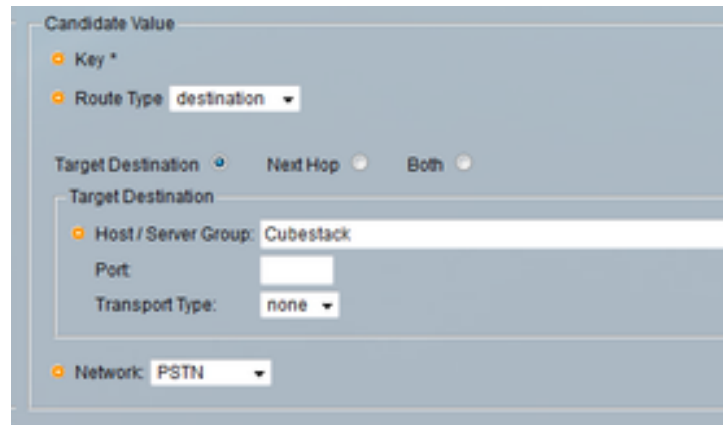
将路由组配置为路由表中的目标时，不要添加端口和传输类型。通过添加端口和/或传输类型，您要求查找DNS主机条目Cubestack:5060:UDP，而不是查找重要的服务器组配置。

路由表PSTN-RT

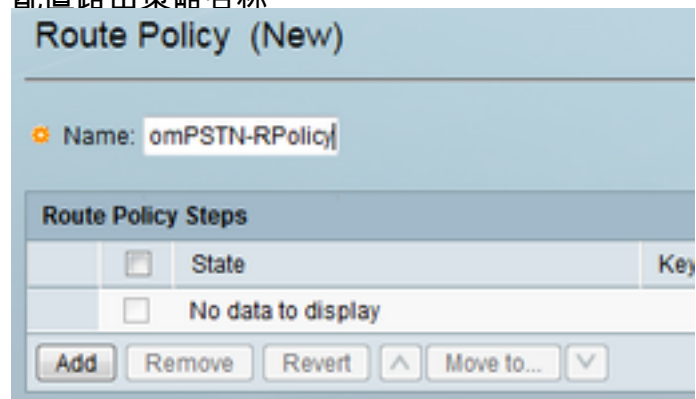
key 2 group CUCM115_RG

密钥5组CUCM105_RG

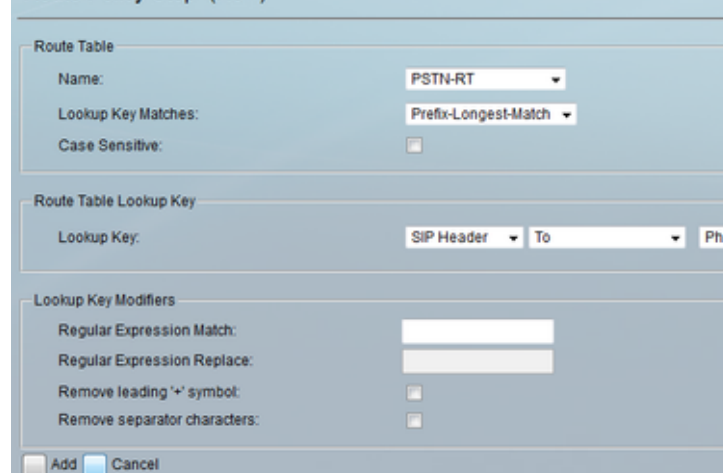
终端路由表



定义“FromPSTN-RPolicy”的密钥
 配置>>路由策略>>添加(###Commit配置)
 配置路由策略名称



点击Add以添加策略步骤
 Route Policy Step (New)

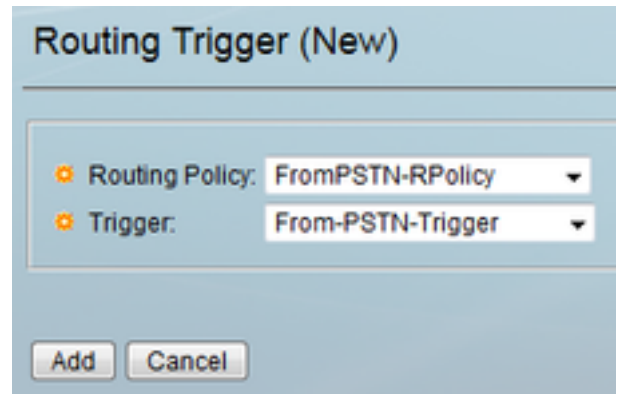


策略步骤将定义密钥的使用方式。在这种情况下，
 在收件人：(To:)上查找最长的电话号码匹配项SIP报头的
 的字段

将“From-PSTN-Trigger”链接到“FromPSTN-RPolicy”
 配置>>路由触发器>>添加
 选择要链接到触发器的路由策略

策略查找来自PSTN-RPolicy
 将100 PSTN-RT报头序列到uri-component电话
 规则前缀
 结束序列
 结束策略

触发路由序列2策略FromPSTN-RPolicy条件From-
 PSTN-Trigger



完整配置

注意： show configuration active verbose将显示包括路由表在内的整个配置。

```
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
!
no sip header-compaction
no sip logging
!
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
!
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
!
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
!
sip overload reject retry-after 0
!
no sip peg-counting
!
sip privacy service
sip queue message
drop-policy head
low-threshold 80
size 2000
thread-count 20
end queue
!
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
!
sip queue xcl
  drop-policy head
  low-threshold 80
  size 2000
  thread-count 2
end queue
!
route recursion
!
sip tcp connection-timeout 30
sip tcp max-connections 256
!
no sip tls
!
sip tls connection-setup-timeout 1
!
trigger condition From-CUCM105-Trigger
  sequence 1
  in-network ^\QCUCM105\E$
  end sequence
end trigger condition
!
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
!
trigger condition mid-dialog
  sequence 1
  mid-dialog
  end sequence
end trigger condition
!
accounting
  no enable
  no client-side
  no server-side
end accounting
!
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
!
route table FromCUCM105-RT
key * target-destination Cubestack PSTN
key 2 group CUCM115_RG
end route table
!
route table FromCUCM115-RT
key 1 target-destination Cubestack PSTN
key 5 group CUCM105_RG
end route table
!
route table PSTN-RT
key 2 group CUCM115_RG
key 5 group CUCM105_RG
end route table
!
policy lookup FromCUCM105-RPolicy
sequence 100 FromCUCM105-RT header to uri-component phone
rule prefix
end sequence
end policy
!
policy lookup FromCUCM115-RPolicy
sequence 100 FromCUCM115-RT header to uri-component phone
rule prefix
end sequence
end policy
!
policy lookup FromPSTN-RPolicy
sequence 100 PSTN-RT header to uri-component phone
rule prefix
end sequence
end policy
!
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
!
server-group sip global-ping
!
no server-group sip ping-503
!
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
!
```

```
no sip cac
!
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
!
call-rate-limit 100
!
end
```

故障排除

跟踪级别配置

在CUSP GUI中，导航至**Troubleshoot >> Cisco Unified SIP Proxy >> Traces**

触发条件 — 级别：调试：这将显示哪些触发器匹配以促进呼叫路由。

路由 — 级别：调试：这将显示呼叫路由过程中执行的操作。哪个密钥匹配，选择了什么目标，等等。

SIP-Wire-Log — 级别：debug:这将显示接收和发送的SIP消息。

跟踪收集

通过GUI

在CUSP GUI中，导航至**Troubleshoot >> Cisco Unified SIP Proxy >> Traces**

选择下载日志文件

您还可以清除日志

通过FTP客户端

默认情况下，没有具有FTP权限的帐户。要启用具有FTP权限的帐户，请将用户添加到PFS组。

```
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
```

通过FTP客户端，连接到CUSP。文件路径:**cusp >> log >> trace >> trace.log**

跟踪顺序

1. **SIP-Wire-Log** — 传入SIP邀请
2. **SIP-Wire-Log** — 返回100尝试
3. **Trigger-Condition** — 标识网络和触发路由策略
4. **路由** — 有关详细信息，请参阅下面的路由跟踪部分

5. SIP-Wire-Log — 向目标发送邀请

6. SIP-Wire-Log — 继续正常的SIP事务，直到每个呼叫段显示200 Ok消息

触发条件跟踪示例

```
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 ...
```

在上述示例中，我们看到网络与PSTN匹配，该PSTN用于路由策略“FromPSTN-RPolicy”。

路由跟踪示例

```
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 nrs.XCLNRSShiftRoutes - Leaving
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, lookuplength=-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, q-
value=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 nrs.XCLNRSShiftRoutes - Leaving ShiftRoutes.execute()

```

1. CUSP在TO:报头

2. CUSP将密钥标识为2003

3. CUSP在路由表中查找密钥

4. CUSP匹配路由表中的条目并标识目标路由组 : CUCM115_RG

5. CUSP在路由组内应用负载均衡

6. CUSP标识RouteGroup中将向其发送SIP消息的特定元素

7. CUSP适用时间策略 (如果适用)

8. CUSP将最终确定要向其发送SIP消息的元素

SIP-Wire-Log跟踪示例

```

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=0
o=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

--- 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=0

o=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

[DsTransportListener-3] DEBUG 2017.08.17 13:48:52:229 DsSipLlApi.Wire - Received UDP packet on 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=z9hG4bK3X51yKL9BEW5v0Kudc5Dw~~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=z9hG4bK3X51yKL9BEW5v0Kudc5Dw~~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: 1e6e772300105000a00084b517aela83;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-remotec: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-remotec: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=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

--- 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-remotecallinfo>;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显示两个呼叫段的正常SIP消息，最大值为200 Okay。

架构参考

