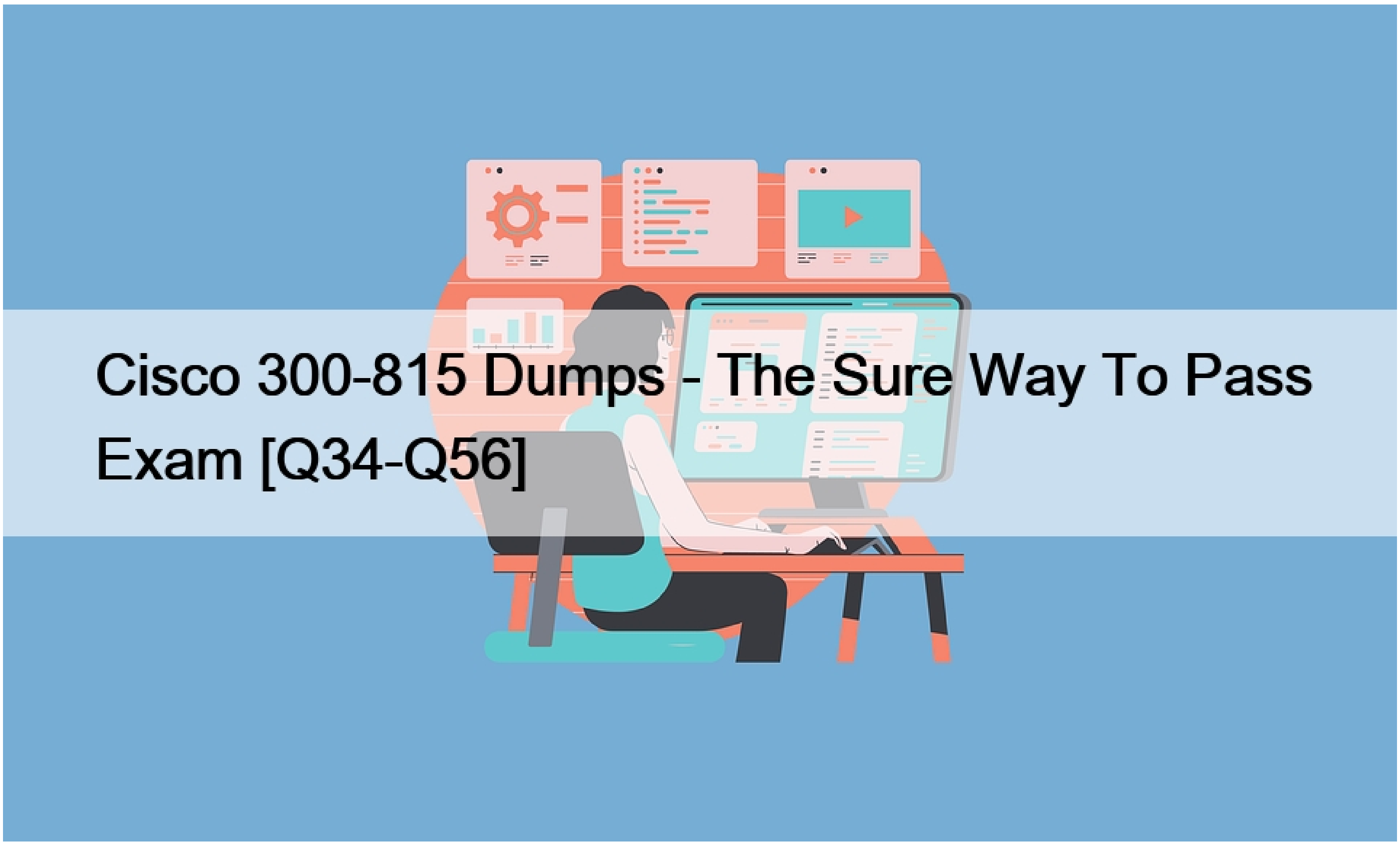
Cisco 300-815 Dumps - The Sure Way To Pass Exam [Q34-Q56



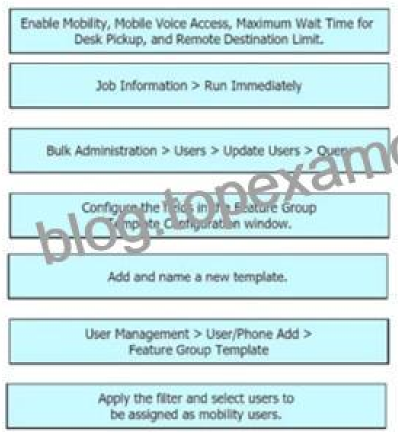
Cisco 300-815 Dumps - The Sure Way To Pass Exam
300-815 Exam Questions (Updated 2024) 100% Real Question Answers

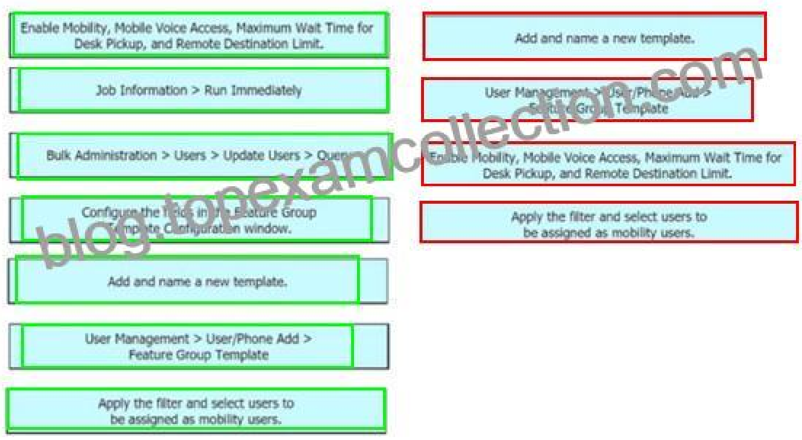
Q34. An administrator wants to use the Call Queuemg feature on Cisco UCM to allow new customers to wait in the queue while other agents are not available to take their call. Which configuration step enables this feature?
* Call Routing > Route/Hunt > Call Park
* Call Routing > Route/Hunt > Hunt Pilot
* Call Routing > Route/Hunt > Line Group
* Call Routing > Route/Hunt > Hunt List

Q35. Which two types of distribution algorithm are within a line group? (Choose two.)
* random
* circular
* highest preference
* top down
* bottom up

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmcfg/CUCM_BK_C DF59AFB_00_admin-guide-90/CUCM_BK_CDF59AFB_00_admin-guide_chapter_0100011.html

Q36. Drag and drop the steps from the left into the order to provision mobility users through LDAP on the right. Not all options are used.





Q37. An engineer must configure a secure SIP trunk with a remote provider, with a specific requirement to use port

5065 for inbound and otubound traffic. Which two items must be configured to complete this configuration?

(Choose two.)
* Incoming Port in SIP Information section of the SIP Trunk configuration.
* Incoming Port in Security Information of the SIP Profile configuration.
* Destination Port in SIP Information section of the SIP Trunk configuration
* Incoming Port in SIP Trunk Security Profile configuration
* Destination Port in SIP Trunk Security Profile configuration

Q38. An engineer is configuring a call park feature in Cisco Unified Communications Manager Express. Which command does the engineer use to ensure that the call is reverted to the user after 60 seconds?
* R2(config-ephone-dn)#park reservation-group 60
* R2(config-ephone-dn)#park-slot timeout 60 limit 2 recall alternate 3002
* R2(config-ephone-dn)#park reservation-group 1
* R2(config-ephone-dn)#park-slot timeout 30 limit 2 recall alternate 3002

Q39. Refer to the exhibit. An engineer is troubleshooting an issue where inbound Calls are failing after they transferred. The provider reports that update is not supported, and this is causing the calls to fail. Which command should resolve this issue?

```
Received
UPDATE sip:192.168.100.101:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 192.168.200.101:5060;branch=
From: "Amy" <sip:2001@192.168.100.101:5060;user=phone>;tag=
To: "Bob" <sip:2002@192.168.100.101:5060;user=phone>;tag=
Call-ID: abcd1234@192.168.200.101
Max-Forwards: 70
Timestamp: 123456789
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER,
SUBSCRIBE, NOTIFY, INFO, REGISTER
Cseq: 101 UPDATE
Contact: <sip:2001@192.168.200.101:5060>
Min-SE: 2000
P-Asserted-Identity: "Joe" <sip:3010@192.168.200.101>
Content-Length: 0
```

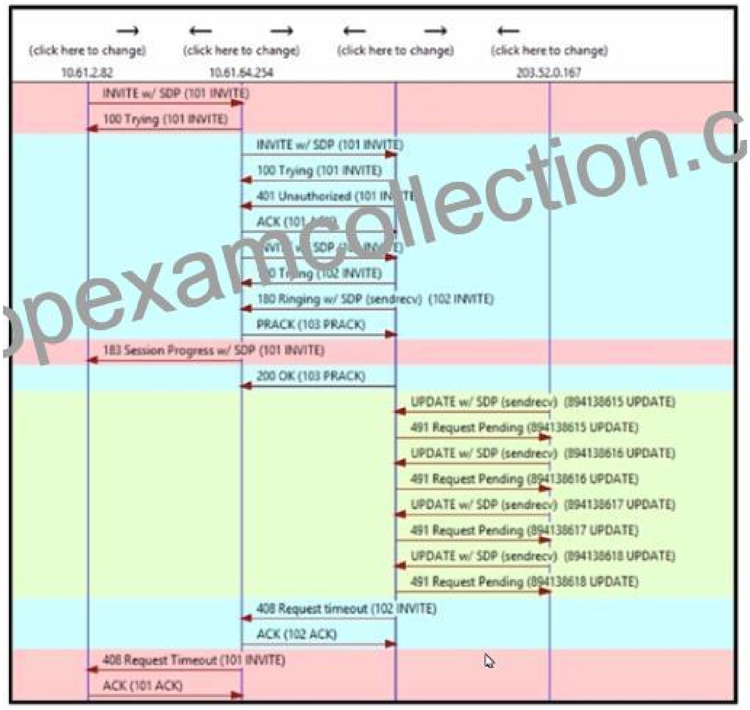
* no midcall-signaling passthru
* no update-callerId
* no contact-passig
* rel1xx require “100rel”

The exhibit shows a SIP message from a caller named “Amy” to a caller named “Bob”. The message contains an “UPDATE” request, which is used to modify the caller ID of an existing call.

However, the provider does not support the UPDATE message, so the call is failing.

The no update-callerId command will disable the sending of UPDATE messages. This will resolve the issue, as the call will no longer be failing due to the unsupported message.

Q40. Refer to the exhibit.



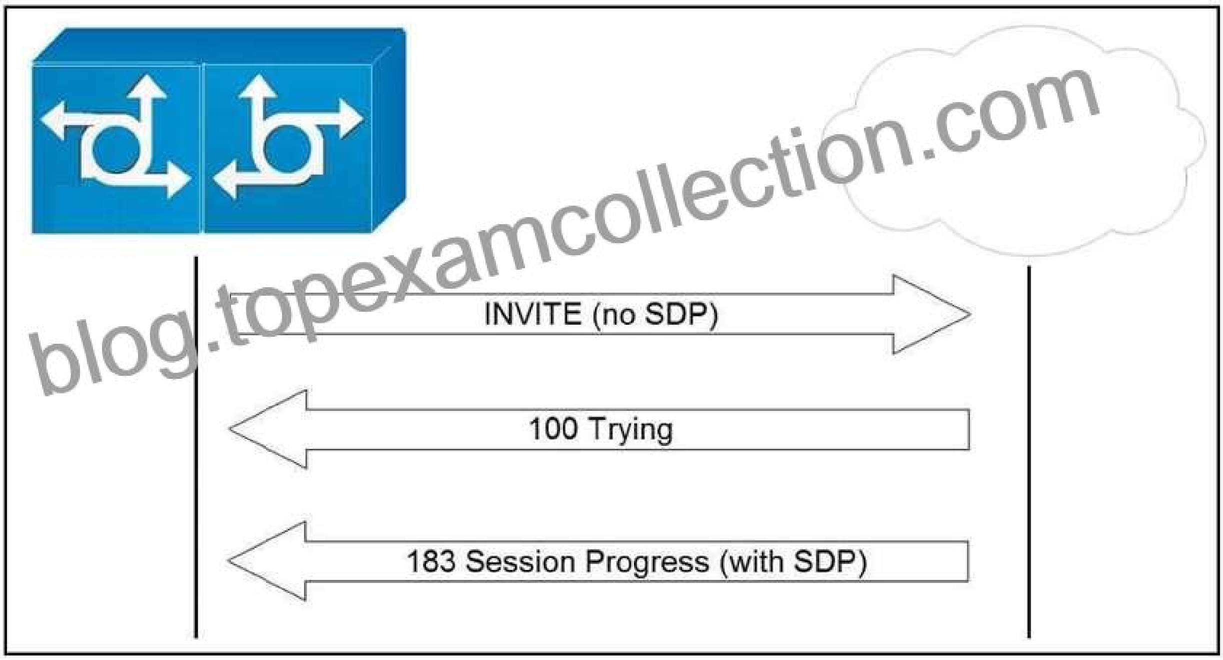
Refer to the exhibit. Calls from users to the PSTN in an organization get disconnected with a 408 Request Timeout when the called party is unavailable to pick up the call. Which solution must be used to resolve this challenge?
* Configure midcall-signaling preserve-codec.
* Choose “Send PRACK” if 1xx contains SDP in the SIP profile.
* Configure midcall-signaling passthru media-change
* Choose “Send PRACK for all 1xx Messages” in the SIP profile.

Q41. Which two configuration parameters are prerequisites to set Native Call Queuing on Cisco Unified Communications Manager? (Choose two.)
* Cisco IP Voice Media Streaming Service must be activated on at least one node in the cluster.
* A unicast music on hold audio source must be configured.
* Cisco RIS data collector service must be running on the same server as the Cisco CallManager service.
* The maximum number of callers allowed in queue must be 10.
* The phone button template must have the Queue Status Softkey configured.
Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_b_system-configuration-guide-1201/cucm_b_system-configuration-guide-1201_chapter_01001101.html#CUCM_RF_C960BC9A_00

Q42. A user reports when they press the services key they do not receive a user ID and password prompt to assign the phone extension. Which action resolves the issue?
* Create the default device profiles for all phone models that are used.
* Subscribe the phone to the Cisco Extension Mobility service.
* Create the end user and associate it to the device profile.
* Assign the extension as a mobile extension.

Q43.



Refer to the exhibit. An administrator is troubleshooting why users are not hearing audio when dialing long distance numbers across their Cisco Unified Border Element. The customer’s carrier has a requirement that dialing long distance requires an access code to be entered. Looking at the exhibit, what two actions can be taken to correct signaling? (Choose two.)

* Enanle PRACK.
* Enable Early Offer on the Cisco Unified Border Element.
* Enable the supplementary-service media-renegotiate command.
* Enable Media Flow Around
* Enable Mid-Call Signaling Consumption.

Section: Cisco Unified Border Element

Q44. What is first preference condition matched in a SIP-enabled incoming dial peer?

* incoming uri
* target carrier-id
* answer-address
* incoming called-number

Section: Signaling and Media Protocols

Explanation/Reference: https://www.cisco.com/c/en/us/support/docs/voice/ip-telephony-voice-over-ip-voip/211306-In-Depth-Explanation-of-Cisco-IOS-and-IO.html#anc8

Q45. Which two configuration parameters are prerequisites to set Native Call Queuing on Cisco Unified Communications Manager? (Choose two.)

* Cisco IP Voice Media Streaming Service must be activated on at least one node in the cluster.
* A unicast music on hold audio source must be configured.
* Cisco RIS data collector service must be running on the same server as the Cisco CallManager service.
* The maximum number of callers allowed in queue must be 10.
* The phone button template must have the Queue Status Softkey configured.

Q46. Calls are not working when sent from a Cisco Unified Border Element to a service provider. After investigating the logs, the engineer notes that the Cisco Unified Border Element is sending the extension only. How is the issue addressed in the configuration?

* voice class request sip-header diversion
* sip-header contact modify
* voice class request sip-header modify
* request invite sip-header diversion modify

Q47. Refer to the exhibit.

```
55697959.007 |12:20:50.913 |AppInfo  |RouteListCdrc::createPartyTransformedCcSetupReqMsg
- before DAapplyCdpnXform() preXformCdpn=[illegible] preTag=SUBSCRIBER prePos=11112222
crCdpnMask=33334444 crPrefixDigit=[illegible]
55697959.008 |12:20:50.913 |AppInfo  |RouteListCdrc::createPartyTransformedCcSetupReqMsg
- after  DAapplyCdpnXform() xformCdpn=33334444 xformTag=SUBSCRIBER xformPos=11112222
55697959.009 |12:20:50.913 |AppInfo  |RouteListCdrc::transformed cdpn (without unconsumpt
digits) = 33334444, unconsumed digit=
```

Which INVITE is sent to 10.10.100.123 as a result of this log?

A)

```
55698034.001 |12:20:50.922 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP
message to 10.10.100.123 on port 5060 index 41
[95992364,NET]
INVITE sip:11112222@10.10.100.123:5060 SIP/2.0
Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae
From: "11112222" <sip:11112222@10.122.200.50>;tag=32412716~41f7
To: <sip:11112222@10.10.100.123>
Date: Thu, 01 Apr 2021 17:20:50 GMT
Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM12.0
```

B)

```
55698034.001 |12:20:50.922 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP
message to 10.10.100.123 on port 5060 index 41
[95992364,NET]
INVITE sip:33334444@10.10.100.123:5060 SIP/2.0
Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae
From: "11112222" <sip:11112222@10.122.200.50>;tag=32412716~41f7
To: <sip:11112222@10.10.100.123>
Date: Thu, 01 Apr 2021 17:20:50 GMT
Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM12.0
```

C)

```
55698034.001 |12:20:50.922 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP
message to 10.10.100.123 on port 5060 index 41
[95992364,NET]
INVITE sip:33334444@10.10.100.123:5060 SIP/2.0
Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae
From: "1000" <sip:1000@10.122.200.50>;tag=32412716~41f7
To: <sip:33334444@10.10.100.123>
Date: Thu, 01 Apr 2021 17:20:50 GMT
Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM12.0
```

```
55698034.001 |12:20:50.922 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP
message to 10.10.100.123 on port 5060 index 41
[95992364,NET]
INVITE sip:11112222@10.10.100.123:5060 SIP/2.0
Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae
From: "1000" <sip:1000@10.122.200.50>;tag=32412716~41f7
To: <sip:11112222@10.10.100.123>
Date: Thu, 01 Apr 2021 17:20:50 GMT
Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM12.0
```


* Option A
* Option B
* Option C
* Option D

Q48. Where on Cisco Unified Communications Manager do you configure the standard local route group for a group of devices?
* System > Location Info
* Call Routing > Route/Hunt > Local Route Group Names
* System > Device Pool
* Call Routing > Emergency Location > Emergency Location (ELIN) Groups
Section: Call Control and Dial Planning

Explanation/Reference: https://www.uccollabing.com/configuring-standard-local-route-group-cucm/

Q49. In Cisco Unified Communications Manager, which tool do you use to check SIP traces?
* MTP
* CCSIP
* RTMT
* OS Administration Page

Q50. What is a function of the metadata carried in SIP sessions between the recording client and the recording server?
* It forks RTP media to the recorder.
* It provides advanced capabilities, such as speech analytics.
* It sets up a new SIP session
* It identifies the participant change due to transfers during the call.

Q51. Refer to the exhibit.

```
<sip:1155@10.2.2.13>;privacy=off;reason=unconditional;counter=1;screen=no

and from the Cisco CUBE the logs show :

<sip:1155@10.3.3.25>;privacy=off;reason=unconditional;counter=1;screen=no
```

Refer to the exhibit. Users report that outgoing calls do not work on the new SIP trunk for outgoing calls. The solution consists of a Cisco UCM Cluster linked to a Cisco Unified Border Element where the SIP trunk is terminated. The provider required 10 digits. The logs show a line going toward the Cisco Unified Border Element. Which code snippet must be added to the configuration to meet the requirement?

* request Invite sip-header modify “<sip:1(…)@” “<sip:91355511@” under the SIP translation profile configuration
* sip-header modify “<sip:1(…)@” “<sip:91355511@” under the voice translation profile configuration
* request Invite sip-header Diversion modify “<sip:1(…)@” “<sip:91355511@” under the SIP profile configuration
* request Invite sip-header modify “<sip:1(…)@” “<sip:91355511@” under the voice translation profile configuration

Q52. Refer to the exhibit. A call made through the Cisco Unified Border Element to pilot 2000 is failing.

What is causing the call to fail?

```
!
dial-peer voice 10 voip
 description Inbound
 session protocol sipv2
 incoming called-number 2000
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 20 voip
 description Outbound
 destination-pattern 2.
 session protocol sipv2
 session target ipv4:192.168.100.101
 voice-class sip options-keepalive
 dtmf-relay rtp-nte
!

CUBE#show dial-peer voice summary
dial-peer hunt 0
                AD                                    PRE   PASS   SESS-SER-GRP\    OUT
TAG    TYPE     MIN    OPER PREFIX    DEST-PATTERN    FER   THRU   SESS-TARGET      STAT
10     voip     up     up                              0    syst
20     voip     up     up             2.               0    syst   ipv4:192.168.100.101
```

* The Cisco Unified Border Element is not receiving a response to its OPTION keepalives.
* The destination pattern is incorrect for the dialed number.
* VAD was not disabled on the outgoing dial peer.
* No codecs are configured on the dial peers.

Q53. Refer to the exhibit.

Refer to the exhibit. An engineer deploys CAC to Cisco UCM. UptofiveG.711 calls must be supported on the WAN link between Site A and Site B. Users report that only four concurrent calls are possible between Site A and Site B. Why isn’t the filth concurrent call successful?

* The G.729 audio codec is negotiated between Site A and B instead of the G.711 codec.
* The QoS configuration on the WAN link causes the fifth call to be dropped.
* Cisco UCM is using alternative links because of the Weight value.
* The Audio Bandwidth value is undersized and must be set to 400 kbps.

Q54. Which two types of authentication are supported for the configuration of Intercluster Lookup Service? (Choose two.)

* TokenID
* username and secret key
* TLS certificates
* passwords
* FQDN of the servers defined in DNS

Q55. Refer to the exhibit. An administrator has configured a SIP trunk between two Cisco UCM clusters. For calls that should use the trunk, the calls fail with a fast busy. The administrator checks the Cisco CallManager SDL traces and found that the cluster to which the calling device is registered never sends an INVITE to the destination cluster. The administrator also verifies that all nodes from both clusters are powered on, and the CallManager service is running.

How is this issue resolved?

```
SIPHandler/ccbId=0/scbId=0/wait_SIPTimer: TimerExpired
type=SIP_TIMER_WAIT_CONNECT value=5000 retries=0
Stack/Transport/0x0xee9c8980/sipTransportPostInternalMsg: Posting Internal Msg
type=1
Stack/Transport/0x0/sipTransportPostCloseConnection: Posting TCP conn close for
addr=10.10.5.11, port=5060, connid=20
Stack/Transport/0x0/sipDeleteConnInstance: Deleted conn=0xe[illegible]c0, connid=20,
addr=10.10.5.11, port=5060, transport=ICP
Stack/Info/0x0/ccsip_process_sipspi_queue_event: ccsip_spi_get_msg_type returned:
2 (SIP_NETWORK_MSG), for event [illegible] (SIPSPI_EV_INTERNAL_MSG)
Stack/Error/0x0xee9c8950/sipTransportPostSendFailure: Posting send failure msg
with tcb: (n11) reason=[illegible]
Stack/Info/0x0/ccsip_process_sipspi_queue_event: ccsip_spi_get_msg_type
returned: 2 (SIP_NETWORK_MSG), for event 55 (SIPSPI_EV_SEND_FAILURE_MSG)
Stack/Info/0x0xee9c8980/ccsip_spi_process_event: Send Error for event
(0xee9cb8b0)
Stack/Error/0x0/act_idle_send_msg_failure: Send Error to 10.10.5.11:5060 for
transport TCP
Stack/Info/0x0xee9c8980/ccsip_set_oo_cause_for_spi_err: Categorized cause: 38,
category:186
Stack/Info/0x0xee9c8980/sipSPIInitiateDisconnect: Initiate call disconnect (38)
for outgoing call
SIPHandler/ccbId=22609/scbId=0/ccsip_api_call_disconnected: ccb->cc_disc_cause
(38): ccb->sip_disc_cause (503)
SIPHandler/ccbId=22609/scbId=0/findDevicePID: Routed to SIPD by ccbId/scbId
Stack/States/0x0xee9c8980/sipSPIChangeState: 0xee9c8980 : State change from
(STATE_IDLE, SUBSTATE_NONE) to (STATE_DISONNECTING, SUBSTATE_NONE)
```

* The administrator must associate the route pattern with a calling search space the device can dial.
* The administrator needs to enable OPTIONS pings on the SIP trunks for both clusters.
* The administrator must allow connectivity so that TCP connections do not fail between the nodes.
* The administrator needs to disable OPTIONS pings on the SIP trunks for both clusters.

Q56. How does an engineer globalize routing for ingress calls coming from the PSTN to internal DNs?
* At the PSTN gateway, put the calling number in PSTN format and the called number in DN format.
* At Cisco Unified CM, put the calling number in E.164 format and the called number in PSTN format.
* At the PSTN gateway, put the calling number in E.164 format and the called number in localized (DN) format.
* At Cisco Unified Communications Manager, put the calling number in E.164 format and the called number in E.164 format.