300-815 PDF Pass Leader, 300-815 Latest Real Test [Q66-Q87

300-815 PDF Pass Leader, 300-815 Latest Real Test
Valid 300-815 Test Answers & 300-815 Exam PDF

About the Exam

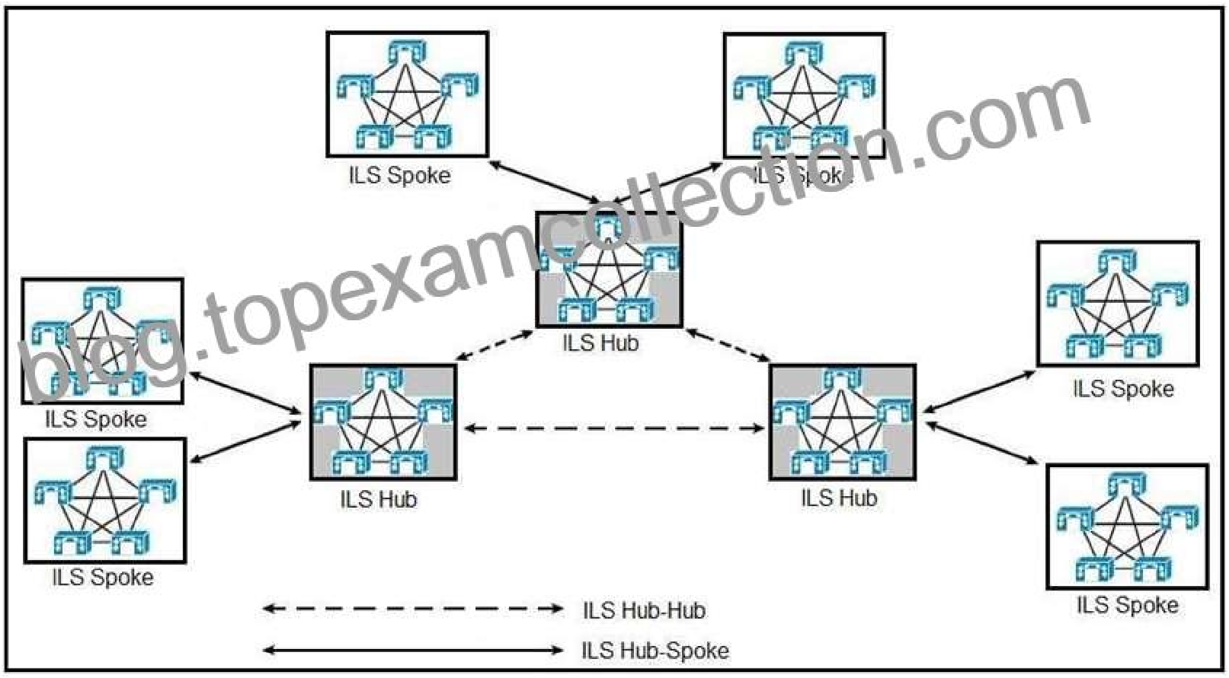
To be eligible, you have to be 18 years or older. If your age group falls between 13-17, you can take part in the test provided your parents or legal guardians support you. However, you're not eligible if you're below 13 years of age.
The exam can be taken in English and requires one to pay a registration fee of $300. The questions are expected to be provided in different forms, including MSQs, fill-in-the-blanks, testlets, and others. The time limit for this test is 1,5 hours.
300-815 is administered by Pearson VUE and can be taken at its testing centers. You can also opt for online format, and any location is fine for taking the test as long as it's quiet and private. In both cases, you will be under close supervision throughout the exam duration. The online format involves high-level supervision through the OnVUE software, so you should have it along with a webcam. Remember to provide an ID that is government-issued. If you decide to go for 300-815 at the center, you should have two valid and self-signed identification documents.

QUESTION 66

When a third-party SIP Phone System is dialed inbound across a Cisco Unified Border Element, DTMF is failing. The third-party

vendor accepts only out-of-band DTMF. Which configuration should be added to the outgoing dial peer to resolve this issue?
* dtmf-relay h245-signal
* dtmf-relay rtp-nte
* dtmf-relay cisco-rtp
* dtmf-relay sip-kpml

QUESTION 67

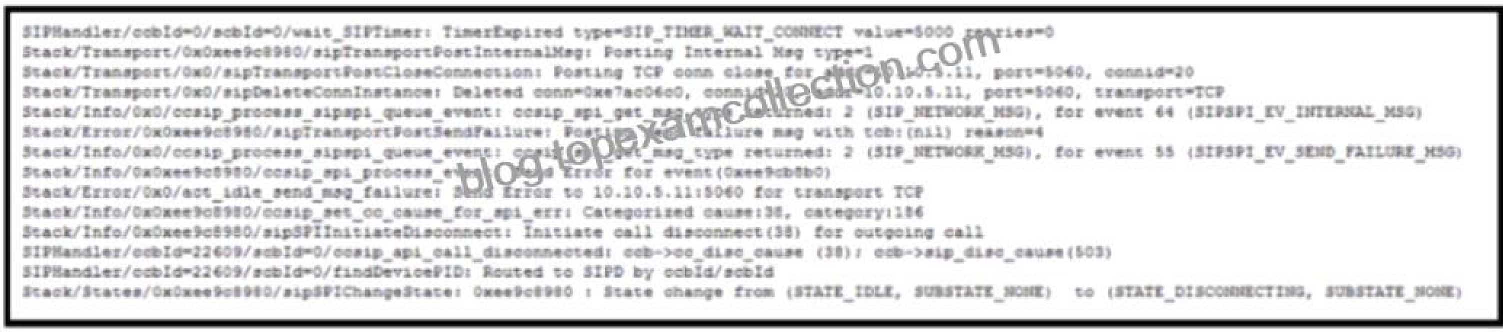


Refer to the exhibit. How many maximum hops can an ILS updarte traverse?
* 3
* 6
* 9
* 12

Section: Cisco Unified CM Call Control Features

QUESTION 68

Refer to the exhibit.



```
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=0xe7ac06c0, connid=20, addr=10.10.5.11, port=5060, transport=TCP
Stack/Info/0x0/ccsip_process_sipspi_queue_event: ccsip_spi_get_msg_type returned: 2 (SIP_NETWORK_MSG), for event 64 (SIPSPI_EV_INTERNAL_MSG)
Stack/Error/0x0xee9c8980/sipTransportPostSendFailure: Posting send failure msg with tcb:(nil) reason=4
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_cc_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_DISCONNECTING, SUBSTATE_NONE)
```

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?
* The administrator needs to disable 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 enable OPTIONS pings on the SIP trunks for both clusters.
* The administrator must associate the route pattern with a calling search space the device can dial.

QUESTION 69

An administrator is configuring a cluster for ILS and wants to limit the amount of entities that Cisco Unified Communications Manager can write to the database for data that is learned through ILS. Which service parameter is used to adjust this limit?
* ILS Max Number of Learned Objects in Database
* ILS Active Learned Object Upper Limit
* Global Data Service Parameter Limit
* Imported Dial Plan Replication Database Object Lower Limit

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_5_1SU1/systemConfig/cucm_b_system-configuration-guide-1251su1/cucm_b_system-configuration-guide-1251su1_restructured_chapter_0100011.html#CUCM_TK_I7C708C2_00

QUESTION 70

Refer to the exhibit.

```
!
dial-peer voice 1 voip
 description to ITSP
 destination-pattern 555.....
 session target ipv4:209.110.110.1
 incoming called-number .
 codec g711ulaw
!
!
```

An engineer configures Cisco Unified Border Element to connect the enterprise VoIP network with a SIP telephony provider. Calls are not working in either direction. What must be configured in the dial peer 1 to fix the issue?
* answer-address 555 ……..
* codec g729
* session-protocol sipv2
* incoming called number 555…….

QUESTION 71

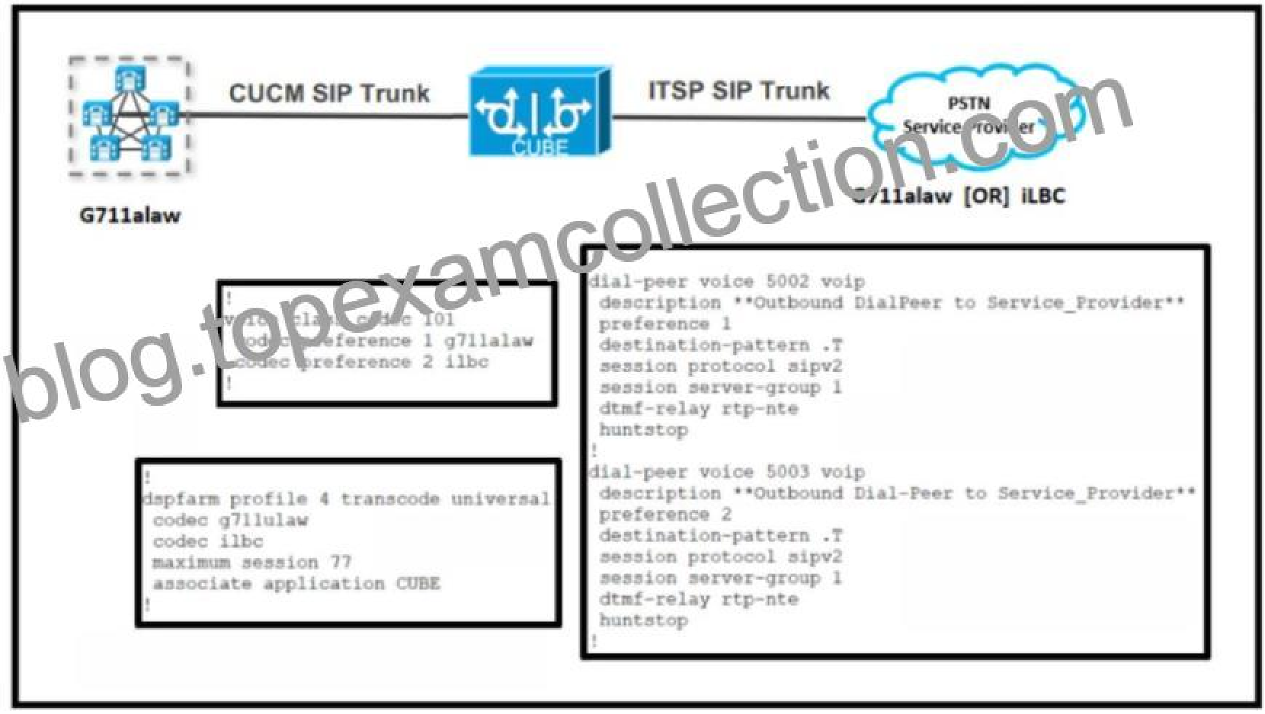
End users at a new site report being unable to hear the remote party when calling or being called by users at headquarters. Calls to and from the PSTN work as expected. To investigate the SIP signaling to troubleshoot the problem, which field can provide a hint for troubleshooting?
* Contact: header of the 200 OK response
* Allow: header if the 200 OK response
* o= line of SDP content
* c= line of SDP content

QUESTION 72

A single site reports that when they dial select numbers, the call connects, but they do not get audio. The administrator finds that the calls are not routing out of the normal gateway but out of another site’s gateway due to a TEHO configuration. What is the next step to diagnose and solve the issue?
* Verify that IP routing is correct between the gateway and the IP phone.
* Verify that the route pattern is not blocking calls to the destination number.
* Verify that the dial peer of the gateway has the correct destination pattern configured.
* Verify that the route pattern has the correct calling-party transformation mask

QUESTION 73

Refer to the exhibit.



Outbound calls to the service provider cause intermittent errors due to a codec mismatch. The internal network sends early offer SDP that contains only G.711 A-law. The service provider reports that some destinations support only G.711 A-law while others support only iLBC. The service provider also allows only 20 active calls at a time Which configuration allows successful media negotiation for all calls using outbound dial peers 5002 and 5003?

```
dial-peer voice 5002 voip
codec g711alaw ilbc
!
dial-peer voice 5003 voip
codec g711alaw ilbc

dial-peer voice 5002 voip
voice-class codec 101 offer-all
!
dial-peer voice 5003 voip
voice-class codec 101 offer-all

dial-peer voice 5002 voip
codec g711alaw
!
dial-peer voice 5003 voip
codec ilbc

dial-peer voice 5002 voip
voice-class codec 101
!
dial-peer voice 5003 voip
voice-class codec 101
```

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

QUESTION 74

If all patterns below are configured in Cisco Unified Communications Manager which would be used when dialing the pattern “123”?
* 12!
* 12X (urgent priority set)
* 1XX (urgent Priority Set)
* 12[2-5]

QUESTION 75

What is a component of Cisco Unified Mobility?
* Unified IVR
* Mobile Connect
* Smart Client Support
* Single Number Connect
Section: Mobility

Explanation/Reference:

QUESTION 76

Signal number reach call phone that not answered are leaving voicemails on the cell phone rather the corporate mailbox. Which two options will resolve this issue? (Choose two.)
* Check the Enable Extend and Connect checkbox
* Check the Enable Unified Mobility features checkbox
* Decrease the T302 timer
* Decrease the T301 timer Decrease the Answer Too Late timer

QUESTION 77

```
voice translation-rule 84
        rule 1 /^\ ([2-9]..[2-9]……$\)/ /\2/
```

Refer to the exhibit. Users report that outbound PSTN calls from phones registered to Cisco Unified Communications Manager are not completing. The local service provider in North America has a requirement to receive calls in 10-digit format. The Cisco Unified CM sends the calls to the Cisco Unified Border Element router in a globalized E.164 format. There is an outbound dial peer on Cisco Unified Border Element configured to send the calls to the provider. The dial peer has a voice translation profile applied in the correct direction but an incorrect voice translation rule applied, which is shown in the exhibit. Which rule modified DNIS in the format that the provider is expecting?

* rule 1 //+([1].*)/ /0111/
* rule 1/+1([2-9]..[2-9]……$)/ /1/
* rule 1 /([2-9]..[2-9]……$)/ /1/
* rule 1 /+1([2-9]..[2-9]……$)/ //

Section: Cisco Unified Border Element

Explanation/Reference:

QUESTION 78

Refer to the exhibit.

```
!

dial-peer voice 1 voip
 description to ITSP
 destination-pattern 555…..
 session target ipv4:209.110.110.1
 incoming called-number .
 codec g711ulaw
!
!
```

An engineer configures Cisco Unified Border Element to connect the enterprise VoIP network with a SIP telephony provider. Calls are not working in either direction. What must be configured in the dial peer 1 to fix the issue?

* answer-address 555 ……..
* codec g729
* session-protocol sipv2
* incoming called number 555…….

QUESTION 79

Route Patterns (1–5 of 5)

Find Route Patterns where Pattern begins with | Find | Clear Filter

	Pattern	Description	Partition	Route Filter	Associated Device
☐	41XXXX	To AMER Cluster	Global-Internal		2-AMER-RL
☐	55XX	Rendezvous meetings	Global-Internal		Rendezvous-Conductor
☐	9.0XXXXXXXXX	Local PSTN	Global-Internal		LocalDevice RL
☐	9.911	Emergency PSTN	Global-Internal		LocalDevice RL
☐	9.91[1-9]!	Emergency PSTN	Global-Internal		LocalDevice RL

Refer to the exhibit. Users report that when they dial the emergency number 9911 from any internal phone, it takes a long time to connect with the emergency operator. Which action resolves this issue?
* Adjust the service parameter T302 timet to the desired value.
* Adjust the service parameter T204 timer to the desired value.
* Check the Urgent Priority check box under 9.911 pattern.
* Point the emergency pattern directly to the PSTN gateway.
Section: Call Control and Dial Planning

QUESTION 80

Refer to the exhibit.

An administrator is troubleshooting a situation where a call placed from a phone registered to Cisco Unified Communications Manager does not complete. The administrator wants to use the Dialed Number Analyzer on Cisco Unified CM to check which translation pattern the call is matching. However, when logging in to Cisco Unified Serviceability there is no option for Dialed Number Analyzer under the tool menu. Which two steps must be performed to resolve this issue? (Choose two.)

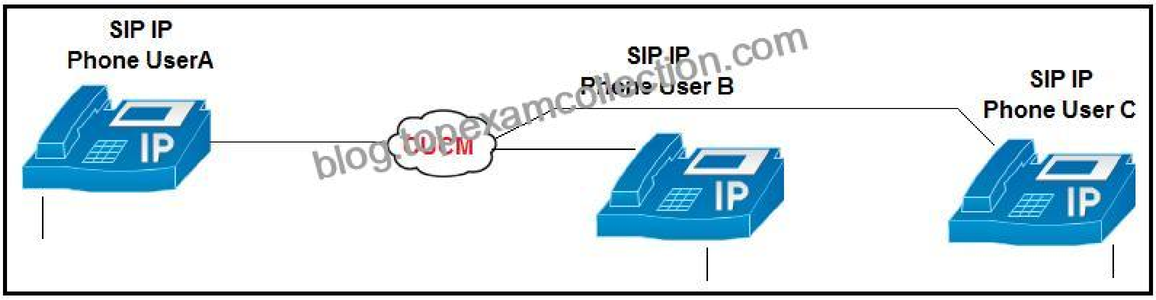
* Restart the subscriber
* Activate the Cisco Extended Functions service.
* Activate the Cisco CallManager service.
* Activate the Cisco Dialed Number Analyzer service.
* Activate the Cisco Dialed Number Analyzer Server service.

QUESTION 81

An engineer must configure call queuing under a Hunt Pilot. After the engineer receives the audio file that will be played to callers during queuing, which two steps should be taken to complete the configuration? (Choose two.)
* Assign the uploaded audio file to the hunting Line Group member’s “User Hold MOH Audio Source
* Assign the uploaded audio file to the hunting Line Group member’s “Network Hold MOH Audio Source”.
* Upload the audio file in “TFTP File Management” via OS Administration GUI
* Assign the uploaded audio file to “Network Hold MOH Source & Announcements” under Hunt Pilot’s Queuing section.
* Upload the audio file in “MOH Audio File Management” via CM Administration GUI

QUESTION 82

Refer to the exhibit.



In an active SIP call between phone user A and phone user B, phone A initiates a call transfer to phone user C.

Which two scenarios are correct? (Choose two.)
* Phone_A sends a SIP-REFER message to the Cisco Unified Communications Manager with Phone_C information in the Refer-To section.
* Phone_B sends a SIP-REFER message to the Cisco Unified CM with Phone_C information in the Refer-To section.
* As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the MOH and the MOH audio is chosen from Phone_B User Hold MOH Audio Source settings.
* As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the music on hold and the MOH audio is chosen from Phone_A Network Hold MOH Audio Source settings.
* As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the MOH and the MOH audio is chosen from Phone_A User Hold MOH Audio Source settings.

QUESTION 83

A company has an SRST gateway running an IOS XE image. The company plans to enable the IPv6 addressing companywide. To enable the IPv6 in a unified SRST gateway to support SIP phones, what are two supported supplementary features for an IPv6 fallback scenario? (Choose two.)

* three-way conference
* secure SIP lines
* T.38 fax relay
* transcoding
* SIP trunk

QUESTION 84

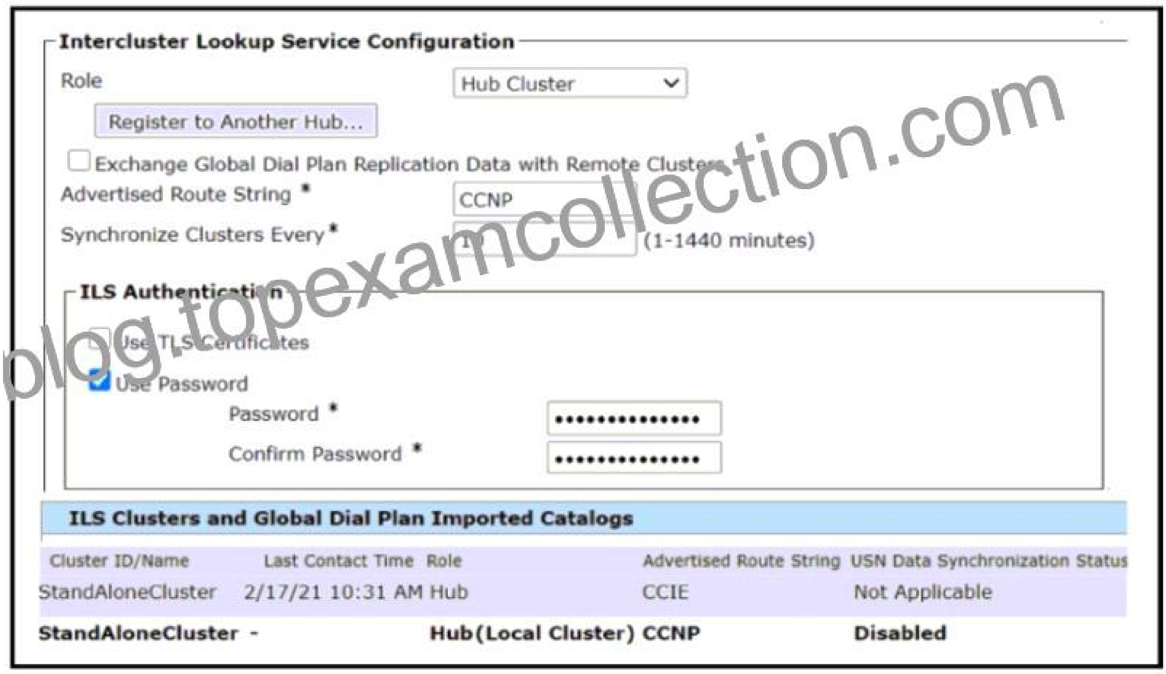
An engineer is troubleshooting local ringback on a Cisco SIP gateway The gateway is not ignoring the SIP 180 response with SDP from the service provider, and the far end device is sending the 180 with SDP to play ringback from the IP address specified m the SDP Which configuration change must be made on the gateway to resolve the issue?
* Router(conf-voi-serv)# dlisable-early-media 180
* Router(conftg-sip-ua)# disable-early-media 180
* Router(con(-voi-serv)# no disable-early-media 180
* Router(config-sip-ua)# no disable-early-media 180

QUESTION 85

Which two descriptions of the Standard Local Route Group deployment are true? (Choose two.)
* can be associated under the route group
* can be associated only under the route list
* chooses the route group that is configured under the device pool of the calling-party device
* chooses the route group that is configured under the device pool of the called-party device
* can be assigned directly to the route pattern
Section: Call Control and Dial Planning

QUESTION 86

Refer to the exhibit.

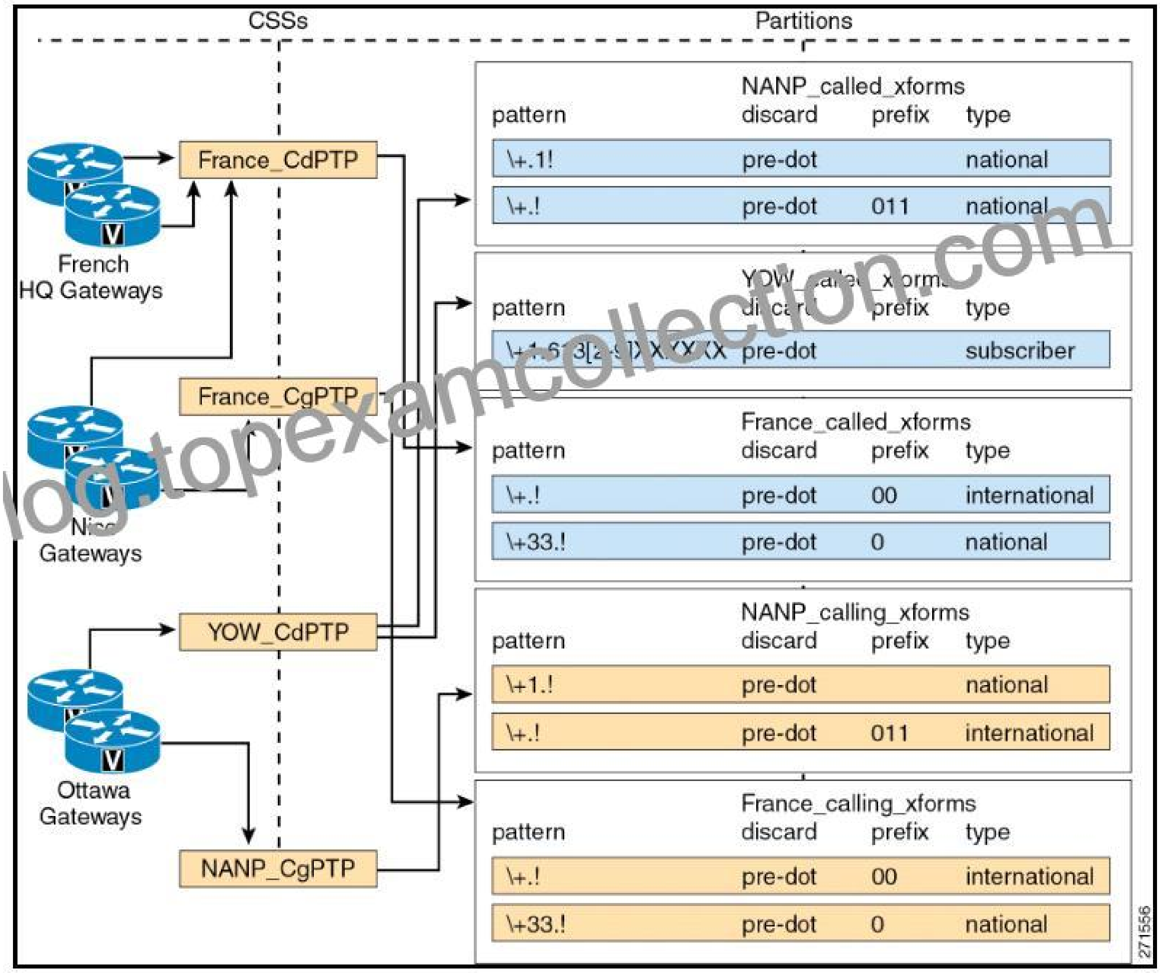


ILS has been configured between two hubs using this configuration. The hubs appear to register successfully, but ILS is not functioning as expected. Which configuration step is missing?

* A password has never been set for ILS.
* Use TLS Certificates must be selected.
* Trust certificates for ILS have not been installed on the clusters
* The Cluster IDs have not been set to unique values

QUESTION 87

Refer to the exhibit.



Within the North American Numbering Plan, gateways located in Ottawa, Canada and marked as “YOW” are assigned to the Calling Party Transformation CSS NANP_CgPTP, which contains partition NANP_calling_xforms. What is the calling-party number and the numbering type if the calling user +1613-555-1234 dials the number?

* calling number 613-555-1234 and numbering type “subscriber”
* calling number 011-1-613-555-1234 and numbering type “subscriber”
* calling number 011613-555-1234 and numbering type “international”
* calling number 613-555-1234 and numbering type “national”

300-815 Dumps Ensure Your Passing: https://www.topexamcollection.com/300-815-vce-collection.html]