300-815[Q&As]

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

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QUESTION 1

A customer routes PSTN calls to ITSP through a SIP trunk on Cisco UCM that forwards and receives calls to and from ITSP. ITSP is set to send an E.164 number when the customer\\'s extension is four digits. Which action should be taken to route the incoming calls to four-digit extensions?

A. Configure a voice translation profile to map the E.164 number to four digits and assign it to the incoming dial-peer on Cisco Unified Border Element.

B. Set the Significant Digits to 8 on the SIP trunk.

C. Set the Significant Digits to 4 on the SIP trunk.

D. Configure a voice translation rule to map the E.164 number to four digits and assign it to the incoming dial-peer on Cisco Unified Border Element.

Correct Answer: C

QUESTION 2

An engineer has temporarily disabled toll fraud prevention for SIP line calls on a Cisco CME12.6x and must enforce security and toll fraud prevention for the SIP line side on Cisco Unified CME. Which configuration must be used to start this process?

A. voice service voip enable ip address trust list

B. voice service voip ip address trusted list

C. voice service voip ip address trusted authenticate

D. voice service voip enable ip address trust authentication

Correct Answer: B

Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmetoll.html

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QUESTION 3

```
!
dial-peer voice 101 voip
  description Inbound to CUCM
  destination-pattern 1…
  session target ipv4: 10.1.1.1
  session protocol sipv2
  codec g711ulaw
  no vad
!
```

Refer to the exhibit. When setting up a new connection to Cisco UCM, the engineer must use out-of-band DTMF. Which configuration meets this requirement?

A. dtmf-relay h245-alphanumeric

B. dtmf-relay sip-kpml

C. dtmf-relay cisco-rtp

D. dtmf-relay cisco-rtp

Correct Answer: B

QUESTION 4

An administrator deployed a third-party H.323 gateway in a voice environment, but users report call failures when using features like call hold or call transfer. What are two reasons that these features fail? (Choose two.)

A. The CSS of the transfer initiating line does not contain the partition of the supplementary feature extension (DirectTransfer or MoH Number).

B. The MTP that is configured for use within the H.323 gateway configuration is configured as a trusted source, but the third-party gateway does not trust the signing root CA certificate of the MTP certificate.

C. The MTP does not support the negotiated codec, and media renegotiating during the call is not supported.

D. The Media Resource Group List of the H.323 gateway contains only transcoders and conference bridges but no

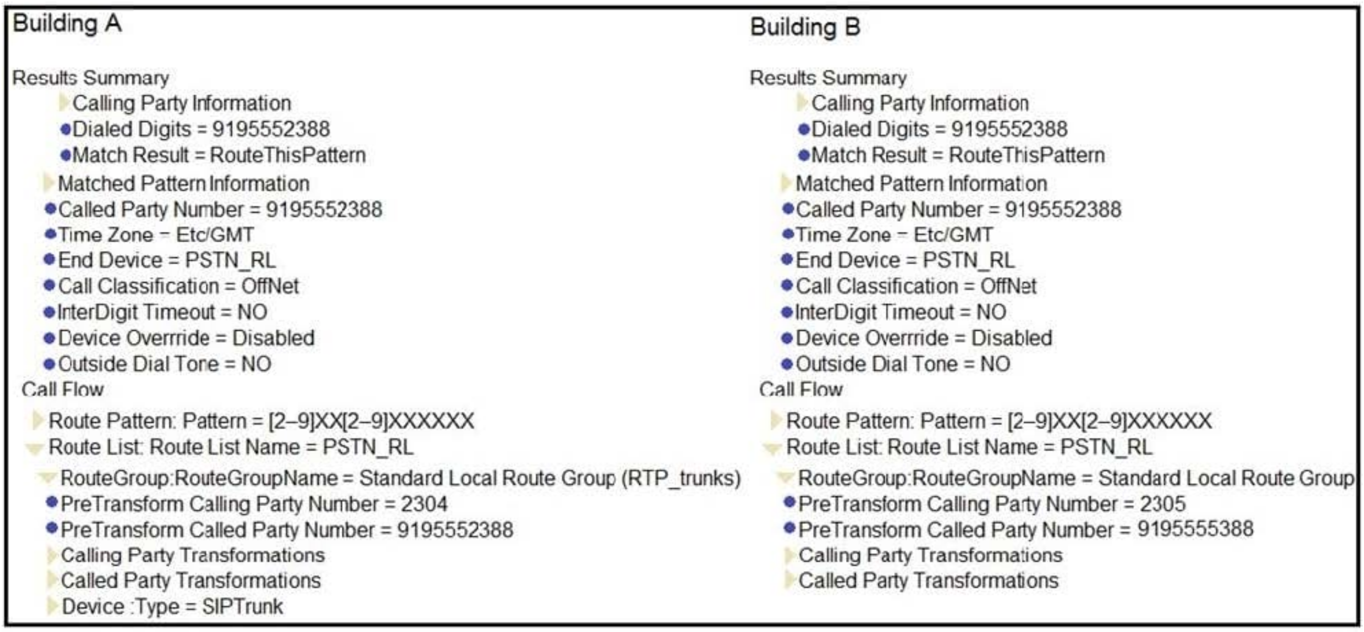
MTP.

E. The third-party gateway does not support supplementary features, so Media Termination Point (MTP) must be inserted.

Correct Answer: CD

QUESTION 5

Refer to the exhibit.



Building A

Results Summary
Calling Party Information
Dialed Digits = 9195552388
Match Result = RouteThisPattern
Matched Pattern Information
Called Party Number = 9195552388
Time Zone = Etc/GMT
End Device = PSTN_RL
Call Classification = OffNet
InterDigit Timeout = NO
Device Override = Disabled
Outside Dial Tone = NO
Call Flow
Route Pattern: Pattern = [2–9]XX[2–9]XXXXXX
Route List: Route List Name = PSTN_RL
RouteGroup:RouteGroupName = Standard Local Route Group (RTP_trunks)
PreTransform Calling Party Number = 2304
PreTransform Called Party Number = 9195552388
Calling Party Transformations
Called Party Transformations
Device :Type = SIPTrunk

Building B

Results Summary
Calling Party Information
Dialed Digits = 9195552388
Match Result = RouteThisPattern
Matched Pattern Information
Called Party Number = 9195552388
Time Zone = Etc/GMT
End Device = PSTN_RL
Call Classification = OffNet
InterDigit Timeout = NO
Device Override = Disabled
Outside Dial Tone = NO
Call Flow
Route Pattern: Pattern = [2–9]XX[2–9]XXXXXX
Route List: Route List Name = PSTN_RL
RouteGroup:RouteGroupName = Standard Local Route Group
PreTransform Calling Party Number = 2305
PreTransform Called Party Number = 9195555388
Calling Party Transformations
Called Party Transformations

A standard local route group is configured for long-distance calls. Calls from building A succeed, but calls from building B fail. On the system, each building has its own device pool. The DNA tool is used to test the configuration. How is this issue resolved?

A. Change the partition of the route pattern.

B. Add a sip trunk inside route group Standard Local Route Group.

C. Modify the route pattern to add a prefix of 91.

D. Add a local route group on the device pool configuration.

Correct Answer: B