Free Cisco 300-815 CLACC Practice Questions

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Question 1

Question Type: MultipleChoice

An administrator is troubleshooting call failures on an H.323 gateway via the CLI. To see signaling for media and call setup, which two debugs must the administrator turn on? (Choose two).

Options:

A- debug H.245 asn1
B- debug H.323 message
C- debug H.225 asn1
D- debug H.225 media
E- debug H.323 asn1

Answer:

A, C

Question 2

Question Type: MultipleChoice

Refer to the exhibit.

```
02904115.001 |09:08:07.093 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.1.1.102 on
port 50244 index 22157
[166156,NET]
ACK sip:1315932e-ff29-4203-23e3-ef940216638e@10.1.1.102:50244;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.1.1.5:5060;branch=z9hG4bKb4fa1fa89d7e
From: <sip:1001@10.1.1.5>;tag=93016~bb788fb3-a1ef-4d03-a96d-651038e22050-28377951
To: <sip:1000@cucm1251.cisco.lab>;tag=7001b5dab46425b45ec6648a-25dc4f91
Date: Wed, 28 Jul 2021 13:08:07 GMT
Call-ID: cee27300-1ed10da7-b403-3251300e@10.1.1.5
User-Agent: Cisco-CUCM12.5
Max-Forwards: 70
CSeq: 103 ACK
Allow-Events: presence
Session-ID: 36ed016300105000a0002834a2824611;remote=49f3b76a00105000a0007001b5dab464
Content-Type: application/sdp
Content-Length: 412

v=0
o=CiscoSystemsCCM-SIP 93016 3 IN IP4 10.1.1.5
s=SIP Call
c=IN IP4 10.1.1.5
t=0 0
m=audio 4000 RTP/AVP 0
a=X-cisco-media:umoh
b=TIAS:64000
a=ptime:20
a=rtpmap:0 PCMU/8000
a=inactive
```

Refer to the exhibit. This message is sent to the device being placed on hold for the Music On Hold audio setup. The held party reports receiving dead air rather than music when the call was put on hold. The software Music On Hold server on Cisco UCM is used in this scenario. Assume that the audio leg between the Music On Hold server and the held device uses G.711, and the relevant region relationship is configured for 64 kbps. What is the cause of the issue?

Options:

A- The bandwidth configured for this region relationship is too low and must be increased to 96 kbps or higher.
B- The device that is placed on hold does not support G.711, and a transcoder could not be allocated for the call.
C- Cisco UCM is sending a=inactive to the held device.
D- The Music On Hold server does not support G.711 and a transcoder could not be allocated for the call.

Answer:

C

Question 3

Question Type: MultipleChoice

Due to a shortage of physical interfaces on a device the administrator requires that a loopback for RTP is used. Which command is required when using a loopback interface for RTP?

Options:

A- voice-class sip resources priority mode passthrough
B- voice-class sip bind control source-interface Loopback0
C- voice-class sip early-offer forced.
D- voice-class sip bind media source-interface Loopback0

Answer:

D

Question 4

Question Type: MultipleChoice

An engineer needs to deploy Cisco Extension Mobility. The engineer already activated the required services, configured the Extension Mobility Phone Service, and created an Extension Mobility device profile that is associated with end users. Which two additional steps are required to complete the configuration? (Choose two.)

Options:

A- Subscribe IP phones to the Extension Mobility Phone Service.
B- Consolidate Cisco UCM certificates by using Bulk Certificate Management.
C- Subscribe directory numbers to the Extension Mobility Phone Service.
D- Subscribe the device profile to the Extension Mobility Phone Service.
E- Define the default Device Template on IP phones when users log out.

Answer:

D, E

Question 5

Question Type: MultipleChoice

Which configuration must an administrator perform to display Translation Pattern operations in Cisco Unified Communications Manager SDL traces?

Options:

A- Enable the Detailed Call Analysis option under Enterprise Parameters for Unified CM.
B- Set up the Digit Analysis Complexity in Service Parameters for Cisco Unified CM to TranslationAndAlternatePatternAnalysis.
C- Check the Translation Patterns Analysis check box in Micro Traces on the Cisco Unified CM Serviceability page.
D- By default, the Translation Patterns operations are printed in SDL traces, so no additional configuration is necessary.

Answer:

D

Question 6

Question Type: MultipleChoice

Calls to a particular extension are not routing to voicemail. The user reaches the voicemail system from the handset by pressing the Messages button Which configuration parameter causes this problem?

Options:

A- The voicemail pilot number for call forwarding is missing from the ephone-dn
B- The voicemail pilot number is missing from the call handling on Cisco Unity Express
C- The voicemail pilot number is missing from the telephony service configuration on Cisco UCME
D- The voicemail pilot number for call forwarding is missing from the ephone

Answer:

A

Question 7

Question Type: MultipleChoice

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.)

Options:

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 third-party gateway does not support supplementary features, so Media Termination Point (MTP) must be inserted.
D- The Media Resource Group List of the H.323 gateway contains only transcoders and conference bridges but no MTP.

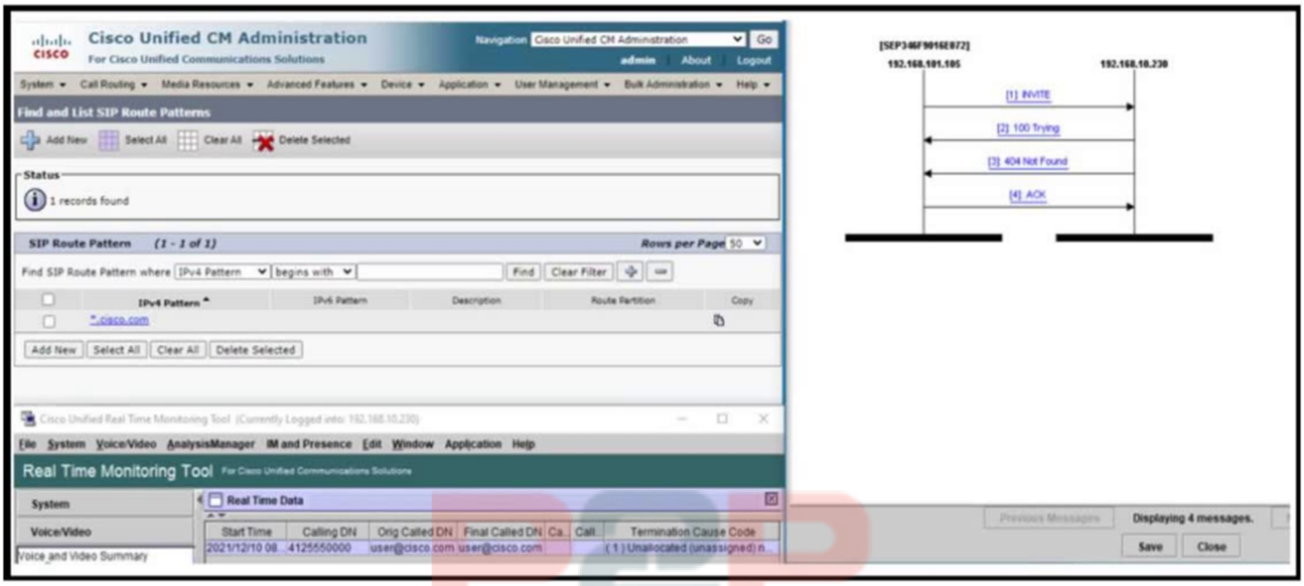
Answer:

C, D, D

Question 8

Question Type: MultipleChoice

Refer to the exhibit.



Refer to the exhibit. A collaboration engineer is troubleshooting a Cisco UCM issue where users report that calls placed to the domain cisco.com are failing and calls to sip.cisco.com are working. Which action resolves the issue?

Options:

A- Change the ''Route Partition'' setting on the *.cisco.com route pattern.
B- Add a SIP route pattern for *.*.
C- Select ''Block Pattern'' on the *.cisco.com route pattern.
D- Delete the *.cisco.com route pattern.

Answer:

B

Question 9

Question Type: MultipleChoice

An administrator is asked to configure egress call routing by applying globalization and localization on Cisco UCM. How should this be accomplished?

Options:

A- Localize the calling and called numbers to PSTN format and globalize the calling and called numbers in the gateway.
B- Globalize the calling and called numbers to PSTN format and localize the calling number in the gateway.

C- Localize the calling and called numbers to E. 164 format and globalize the called number in the gateway.
D- Globalize the calling and called numbers to E. 164 format and localize the called number in the gateway.

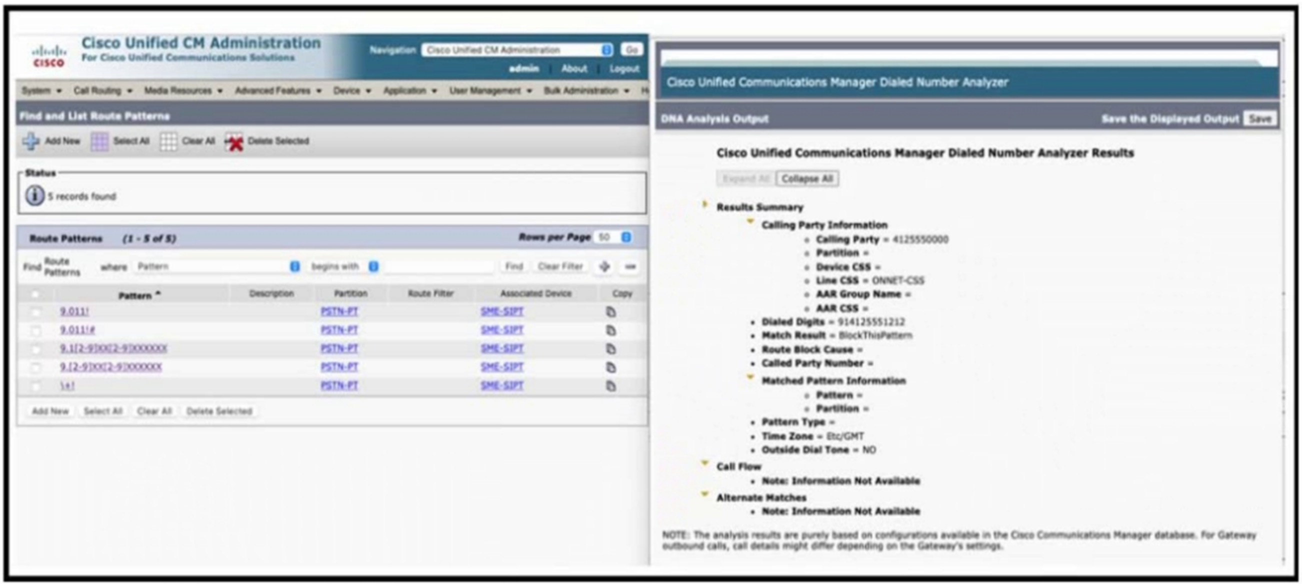
Answer:

D

Question 10

Question Type: MultipleChoice

Refer to the exhibit.



Refer to the exhibit. A collaboration engineer is troubleshooting an issue where a user of a Cisco UCM IP phone reports failed calls when trying to dial out to the PSTN. Which action resolves the issue?

Options:

A- Assign a calling search space to the line or the device that has access to the route pattern.
B- Deselect ''Block this pattern'' on the ''Route Option'' setting of the route pattern.
C- Select the ''Urgent Priority'' setting on the route pattern.
D- Instruct the user to not dial a ''1'' before their local area code.

Answer:

A

Question 11

Question Type: MultipleChoice

An administrator is configuring a new deployment using Cisco Unified CME. The SCCP phones register without any issues, but SIP phones are not registering. Assume that all other configuration is valid. Which code allows SIP phones to register to Cisco UCME?

Options:

A- voice service voip
allow-connections sip to h323
B- voice service voip
sip
bind media source-interface Vlan100
C- voice service voip
sip
bind control source-interface Vlan100
D- voice service voip
sip
registrar server expires max 600 min 60

Answer:

C

Question 12

Question Type: MultipleChoice

A user requests a feature to send an active call to the mobile phone number on the physical phone. As an administrator, what should be configured in the Cisco UCM to accomplish this?

Options:

A- A Remote Destination Profile having the same extension as the Mobile phone's number adds the physical phone's Directory Number to the RDP as a Remote Destination. Add the softkey 'Mobility' to the physical phone's softkey template.
B- A Remote Destination Profile having the same extension as the physical phone's Directory Number, add the mobile phone number to the RDP as a Remote Destination. Add the softkey 'Mobility' to the physical phone's softkey template.
C- A Remote Destination Profile having the same extension as the Mobile phone's number adds the physical phone's Directory Number to the RDP as a Remote Destination. Add the softkey

'Join' to the physical phone's softkey template
D- A Remote Destination Profile having the same extension as the physical phone's Directory Number, add the mobile phone number to the RDP as a Remote Destination. Add the softkey 'Join' to the physical phone's softkey template.

Answer:

B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