- **Vendor:** Cisco
- **Exam Code:** 300-815
- **Exam Name:** Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)
- **New Updated Questions from Braindump2go (Updated in Sep/2020)**

QUESTION 45
Which two statements are correct with respect to the Client Matter Code setting in the route pattern configuration? (Choose two.)

A. The Client Matter Code feature does not support overlap sending because the Cisco Unified CM cannot determine when to prompt the user for the code.
B. If you check the Allow Overlap Sending check box, the Require Client Matter Code check box becomes disabled.
C. If you check the Allow Overlap Sending check box, you can also check the Require Client Matter Code check box.
D. The Client Matter Code feature does support overlap sending because the Cisco Unified Communications Manager can determine when to prompt the user for the code.
E. The Client Matter Code has the option to configure Authorization Level such as in the Forced Authorization Code.

Answer: AB
Explanation:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmfeat/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100_chapter_010000.pdf

QUESTION 46
A network engineer designs a new dial plan and wants to block a certain range of numbers (8135100 through 8135105). What is the most specific route pattern that can be configured to block only the numbers in this range?

A. 813510[012345]
B. 813510[12345]
C. 813510[^0-5]
D. 81XXXXX

Answer: A

QUESTION 47
Which three CLI commands are used when configuring H.323 call survivability for all calls? (Choose three.)

A. voice service voip
B. telephony-service
C. h323
D. call preserve

E. call-router h323-annexg
F. transfer-system

Answer: ACD

QUESTION 48
Which command displays the detailed configuration of all the Cisco Unified IP phones, voice ports, and dial peers of the Cisco Unified SRST router?

A. show call-manager-fallback all
B. show dial-peer voice summary
C. show ephone summary
D. show voice port summary

Answer: A

QUESTION 49
Which two descriptions of the Standard Local Route Group deployment are true? (Choose two.)

A. can be associated under the route group
B. can be associated only under the route list
C. chooses the route group that is configured under the device pool of the calling-party device
D. chooses the route group that is configured under the device pool of the called-party device
E. can be assigned directly to the route pattern

Answer: BD

QUESTION 50
Refer to the exhibit. 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?

```
!

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

A. answer-address 555 ....
B. codec g729
C. session-protocol sipv2
D. incoming called number 555.......

Answer: D

QUESTION 51
After configuring a Cisco CallManager Express with Cisco Unity Express, inbound calls from the PSTN SIP trunk

receive a ring tone for 20 seconds and then a busy signal instead of voicemail. Which configuration fixes this problem?

A. Router(config)# voice service voip
Router(conf-voi-serv)#allow-connections h323 to h323
B. Router(config)#dial-peer voice 2 voip
Router(config-dial-peer)#no vad
C. Router(config)# voice service voip
Router(conf-voi-serv)#allow-connections voice-mail mod
D. Router(config)# voice service voip
Router(conf-voi-serv)#no supplementary-service sip moved-temporarily

Answer: A
Explanation:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusrst/admin/sccp_sip_srst/configuration/guide/SCCP_and_SIP_SRST_Admin_Guide/ srst_call_handling.html

QUESTION 52
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.)

A. Incoming Port in SIP Information section of the SIP Trunk configuration.
B. Incoming Port in Security Information of the SIP Profile configuration.
C. Destination Port in SIP Information section of the SIP Trunk configuration
D. Incoming Port in SIP Trunk Security Profile configuration
E. Destination Port in SIP Trunk Security Profile configuration

Answer: CD

QUESTION 53
In Cisco Unified Communications Manager, which tool do you use to check SIP traces?

A. MTP
B. CCSIP
C. RTMT
D. OS Administration Page

Answer: C

QUESTION 54
If all patterns below are configured in Cisco Unified Communications Manager which would be used when dialing the pattern "123"?

A. 12!
B. 12X (urgent priority set)
C. 1XX (urgent Priority Set)
D. 12[2-5]

Answer: B

QUESTION 55
Which configuration must an administrator perform to display Translation Pattern operations in Cisco Unified Communications Manager SDL traces?

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: A
Explanation:
https://community.cisco.com/t5/collaboration-voice-and-video/taking-sip-call-trace-on-cisco-unified-cm-using-rtmt/ta-p/3161200