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Exam : **300-070**

Title : Implementing Cisco IP
Telephony & Video, Part 1
v1.0

Vendor : Cisco

Version : DEMO

NO.1 The Cisco IP Phone of your company operator is on extension 7000. However, users normally dial 0 to get the operator. Which action sends calls that were made to 0 to the operator?

- A. Create a translation pattern for 0 and a called party transform mask of 7000.
- B. Create the IP phone overlay feature so the operator's line has 0 and 7000 assigned to it.
- C. Create a route pattern of 0 and use a called party transform mask of 7XXX.
- D. Create a translation pattern for 0 and a calling party transform mask of 7000.
- E. Create a translation pattern for 7000 and a called party transform mask of 0.

Answer: A

NO.2 You run a dial plan analysis from the Cisco Unified Communications Manager platform. Upon successful completion, you download this file to your local computer.

What is the naming convention of this file?

- A. dnaOutput_<dialeddigits>_<callingparty>.xml
- B. dnaOutput_<dialeddigits>_<callingparty>.xml
- C. dnaOutput_<callingparty >_< dialeddigits >.xml
- D. dnaOutput_<dialeddigits>_<callingparty>.csv

Answer: C

NO.3 Which two conferencing endpoints support the Multi Site video conferencing feature? (choose two)

- A. SX20
- B. SX80
- C. SX90
- D. SX10

Answer: A,B

NO.4 IP Voice Media Stream Application service should be enabled on which server?

- A. Must enabled on dedicated server
- B. Enable on the server which TFTP also need to be enable
- C. Enable on gatekeeper
- D. Only on one node

Answer: B

NO.5 What is the correct procedure for implementing call coverage?

- A. configure partitions, directory numbers, hunt pilot, hunt list, line groups
- B. configure directory numbers, partitions, hunt pilot, hunt list, line groups
- C. configure partitions, directory numbers, line groups, hunt lists, hunt pilot
- D. configure partitions, line groups, hunt lists, hunt pilot, directory numbers

Answer: C

NO.6 Which two options can a route pattern be assigned? (Choose two.)

- A. a gateway that is assigned to a route group
- B. a layer 3 switch

- C. a route list with one or more route groups configured
- D. a route list with no route groups configured
- E. a gateway

Answer: D,E

NO.7 When creating a route pattern, which two of these options should you avoid? (Choose two)

- A. Wildcards.
- B. Special characters and escape sequences, such as @ and \+.
- C. Spaces.
- D. Lowercase characters, such as a, b, c
- E. Numbers.
- F. Uppercase characters, such as A, B, C.

Answer: C,D

NO.8 What is a route list?

- A. an ordered list of gateways and trunks
- B. a list of route patterns
- C. a list of locations
- D. an ordered list of regions
- E. an ordered list of route groups

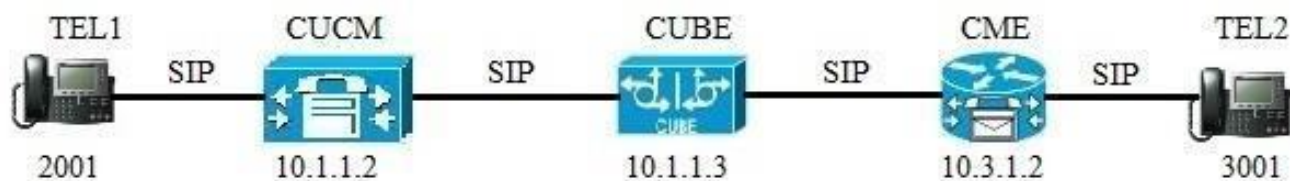
Answer: E

NO.9 Refer to the exhibit. An engineer is working on a ticket that reports TEL1 cannot call TEL2. These dial peers are configured to route the call between the endpoints.

```
dial-peer voice 2001 voip
destination-pattern 2001
session protocol sipv2
session target ipv4:10.1.1.2
incoming called-number .
codec g722-64
video codec h264
dtmf-relay rtp-nte sip-notify sip-kpml
no vad
dial-peer voice 3001 voip
destination-pattern 3001
session protocol sipv2
session target ipv4:10.3.1.2
incoming called-number .
codec g722-64
video codec h264
dtmf-relay rtp-nte sip-notify sip-kpml
no vad
```

While troubleshooting, the engineer discovers that the Cisco Unified Border Element is giving the error "SIP/2.0 403 Forbidden".

Which two Cisco Unified Border Element configurations resolve this issue? (Choose two.)



A. voice service voip
mode border-element
ip address trusted list
ipv4 10.1.1.0 255.255.255.0
ipv4 10.2.1.0 255.255.255.0
allow-connections sip to sip
sip
bind control source-interface loopback0
bind media source-interface loopback0

B. voice service voip
mode border-element
ip address trusted list
allow-connections sip to sip
sip
bind control source-interface loopback0
bind media source-interface loopback0

C. voice service voip
mode border-element
allow-connections sip to sip
sip
bind control source-interface loopback0
bind media source-interface loopback0

D. voice service voip
mode border-element
ip address trusted list
ipv4 10.1.1.0 255.255 255.0
ipv4 10.3.1.0 255.255.255.0
allow-connections sip to sip
sip
bind control source-interface loopback0
bind media source-interface loopback0

E. voice service voip
no ip address trusted authenticate
mode border-element
allow-connections sip to sip
sip
bind control source-interface loopback0
bind media source-interface loopback0

Answer: D,E

NO.10 Refer to the exhibit. Which option is the most highly recommended option to install Cisco TelePresence Conductor on the virtual machine?

- A. Thick Provision Lazy Zeroed
- B. Thick Provision Eager Zeroed
- C. Thick Provision
- D. Thin Provision

Answer: A

Explanation:

http://www.cisco.com/c/dam/en/us/td/docs/telepresence/infrastructure/conductor/install_guide/TelePresence-Conductor-Virtual-Machine-Install-Guide-XC3-0.pdf (page 8)

NO.11 Which two CLI commands allow you to verify the correct VoIP dial-peer matching? (Choose two)

- A. debug voip dialpeer inout
- B. show dialplan number
- C. debug voip ccapi service
- D. debug ccsip messages
- E. debug dialpeer match

Answer: A,B

NO.12 Which DSCP classification is defined in RFC 2474?

- A. DSCP CS2.
- B. DSCP AF21.
- C. DSCP CS3.
- D. DSCP AF11.
- E. DSCP 0.
- F. DSCP EF.
- G. DSCP CS1.
- H. DSCP CS6.

Answer: C

NO.13 What are two characteristics of a Cisco Unified Communications Manager Express dial plan? (Choose two.)

- A. voice translation rules
- B. dial peers
- C. partitions and calling search spaces
- D. route patterns
- E. COR

Answer: B,E

NO.14 In which scenario can a software conference bridge from Cisco Unified Communications Manager be used for an audio conference?

- A. when all participants use the G.711 codec
- B. when there are only three participants
- C. when only SCCP phones are used
- D. when all participants use both the G.711 and G.729 codecs

Answer: A

NO.15 Which three options are components of Media Resources Architecture? (Choose three)

- A. Music on hold.
- B. Route group
- C. Media Resource Manager.
- D. Route list.
- E. Media Resource Group List.
- F. Media Resource Group.

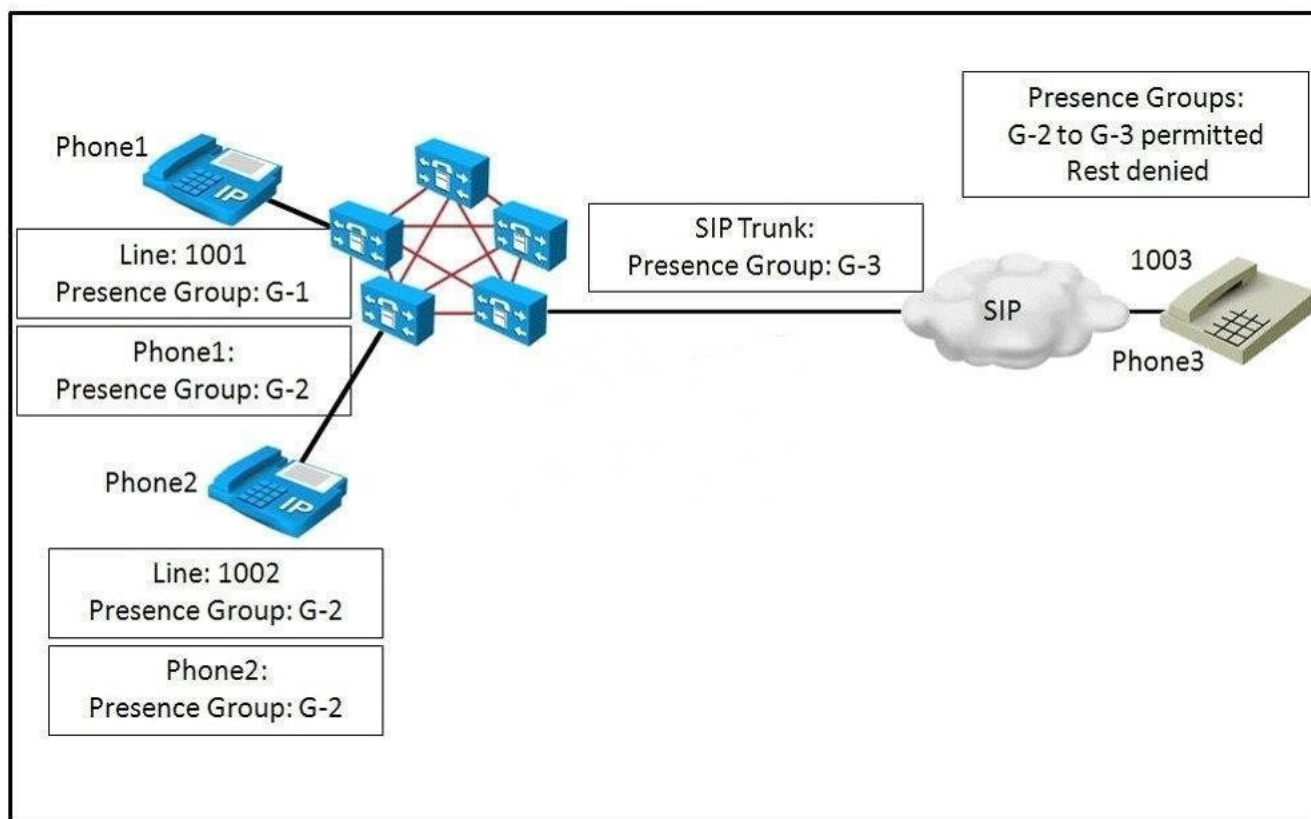
Answer: C,E,F

NO.16 Which three options are valid digit discard instructions?

- A. Trailing#.
- B. PostAt.
- C. PreDot.
- D. PostDot.
- E. PreAt.
- F. Leading#.

Answer: A,C,E

NO.17 Refer to the Exhibit. Which two of these statements about call lists enabled for Cisco Unified Presence are true? (Choose two)



- A. 1001 can watch 1002
- B. 1001 cannot watch 1003
- C. 1003 can watch 1001
- D. 1002 can watch 1003
- E. 1002 can watch 1001

Answer: A,D

NO.18 Which lists features that are supported by Cisco Unified Border Element?

- A. colocation of VRF with Cisco Unified Communications Management Express and Cisco Unified Border Element
- B. CVP survivability TCL support with high availability
- C. Cisco "x-cisco-tip" headers
- D. low complexity codec for the iSAC codec

Answer: B

NO.19 Which three Cisco Unified Communications Manager settings can be used to support IP phone services? (Choose three.)

- A. Common Phone Profile
- B. Enterprise Parameters
- C. IP Phone Service Configuration window
- D. Service Parameters
- E. Device Defaults
- F. Phone Configuration window

Answer: A,B,F

NO.20 An engineer is configuring media resources and a language annunciator.
Which component is required?

- A. transcoder
- B. annunciator
- C. software conference bridge
- D. music on hold

Answer: B

NO.21 Refer to the exhibit. A Network Administrator needs to create a SIP route pattern to communicate with Monterrey's Gateway.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System > Call Routing > Media Resources > Advanced Features > Dev

IP Route Pattern Configuration

Save Delete Copy Add New

Status
Add successful

Pattern Definition

Pattern Usage	Enrutamiento de dirección IP
IPv4 Pattern*	10.200.6.3
IPv6 Pattern	
Description	Monterrey
Route Partition	Interna
SIP Trunk/Route List*	RL-Monterrey

Block Pattern

Calling Party Transformations

Use Calling Party's External Phone Mask

Calling Party Transformation Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation* Default

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System > Call Routing > Media Resources > Advanced Features > Dev

Gateway Configuration

Save Delete Copy Reset Apply Config

Status
Status: Ready

Device Information

Product	H.323 Gateway
Device Protocol	H.225
Registration	Unknown
IP Address	10.200.6.3
Device is not trusted	
Device Name*	10.200.6.3
Description	10.200.6.3 Me
Device Pool*	DP-Monterrey
Common Device Configuration	< None >
Call Classification*	OffNet
Media Resource Group List	MRGL-Monter
Packet Capture Mode*	None
Packet Capture Duration	0

The Administrator does not understand why this is not working properly. Which two statements explain the issue and resolve the problem? (Choose two.)

- A. The Administrator needs to set up a SIP trunk between Monterrey's Gateway and CUCM.
- B. The Administrator should type the reload command into Monterrey's Gateway.
- C. The issue is caused by incorrect protocol integration between Monterrey's Gateway and CUCM.
- D. The issue is caused by a failure of communication between the Monterrey's Gateway and CUCM.

Answer: A,C

NO.22 Route lists determine the order of preference for route group usage. If a route list is configured, what is the minimum number of route groups that you must configure?

- A. 4
- B. 7
- C. 2
- D. 5
- E. 3
- F. 1
- G. 6
- H. 8

Answer: F

NO.23 Refer to the exhibit an engineer with ID341888740 is troubleshooting this video conference issue.

- A video call between a cisco 9971 in region 1 and another cisco 8961 in region 1 works

- As soon as the cisco 9971 in region1 conferences in a cisco 8945 in region 2, the region1 endpoint cannot see the region2 endpoint video

Which cause of this issue is true?

Region Relationships Table:

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
REGION1	CCNP COLLAB	60 kbps	384 kbps	2147483647 kbps
REGION2	CCNP COLLAB	64 kbps (G.722, G.711)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

Audio Codec Preference List Information:

Name*: CCNP COLLAB
 Description*: CCNP COLLAB
 Codecs in List*: G.722 48k, G.711 U-Law 64 k, G.729 8k, G.711 A-Law 56k

- A. Cisco 8945 doesnot have a camera connected
- B. maximum audio bit rate should be increased
- C. maximum session bit rate for immersive video calls is too low
- D. maximum session bit rate for video calls is too low

Answer: D

NO.24 Which three characteristics of a video stream require the implementation of QoS? (Choose three.)

- A. high packet rate
- B. small packets
- C. variable packet rate
- D. constant bandwidth
- E. high MTU

Answer: A,C,E

NO.25 Which two options are valid SIP URI formats? (Choose two.)

- A. tel:98765@cisco.com
- B. phone: 98765cisco.com
- C. user.jsmith@cisco.com
- D. Usipbob.cisco.com
- E. sip: 12345@10.10.10.105060; user= phone

Answer: A,E