

## Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

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#### NEW QUESTION 1

What happens when a Cisco IP phone loses connectivity to the duster during an active call?

- A. The call continues to be active, but features like transfer or hold do not work.
- B. The call continues and all features work.
- C. The call drops immediately.
- D. The call drops after missing two keepalives from Cisco UCM.

**Answer:** D

#### NEW QUESTION 2

A Cisco voice gateway is configured to use a sip-kpml DTMF relay in global settings. A new SIP dial peer is configured for a third-party application that only supports an in-band DTMF relay. Which commands must an engineer run on the dial peer?

- A. dtmf-relay sip-info
- B. dtmf-relay sip-notify
- C. dtmf-relay rtp-net
- D. no dtmf-relay sip-kpml

**Answer:** C

#### NEW QUESTION 3

How does an administrator make a Cisco IP phone display the last 10 digits of the calling number when the call is in the connected state, and also display the calling number in the E.164 format within call history on the phone?

- A. Change the service parameter Apply Transformations On Remote Number to True.
- B. Configure a translation pattern that has a Calling Party Transform Mask of XXXXXXXXXXXX.
- C. On the inbound SIP trunk, change Significant Digits to 10.
- D. Configure a calling party transformation pattern that keeps only the last 10 digits.

**Answer:** A

#### NEW QUESTION 4

What is an advantage of using Cisco Webex Control HuB?

- A. enables the provisioning, administration, and management of Webex services and Webex Hybrid Services
- B. brings Video, audio, and web communication together to meet the collaboration needs of the modern workplace
- C. provides streamlined communication and collaboration for a hybrid workforce
- D. offers easy contact management, centralized administration, and centralized configuration management

**Answer:** A

#### Explanation:

Cisco Webex Control Hub is a cloud-based management platform that enables you to provision, administer, and manage Webex services and Webex Hybrid Services. It provides a single pane of glass for managing all of your Webex services, including Webex Meetings, Webex Teams, and Webex Calling.

Webex Control Hub offers a number of features and benefits, including:

- A single pane of glass for managing all of your Webex services
- Centralized user management
- Simplified provisioning and administration
- Real-time analytics and reporting
- Enhanced security and compliance

Webex Control Hub is a powerful tool that can help you manage your Webex services more effectively. It is easy to use and provides a number of features and benefits that can help you improve your productivity and efficiency.

#### NEW QUESTION 5

Refer to the exhibit.

```
Sent:
INVITE sip:2004@192.168.100.100:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.100.200:5060;branch=z9hG4bKFlFED
From: "7000" <sip:7000@192.168.100.200>;tag=43CDE-1A22
To: <sip:2004@192.168.100.100>
Call-ID: 26BCA00-4C4E11EA-80169514-B1C46126@192.168.100.200
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
User-Agent: Cisco-SIPGateway/IOS-16.9.5
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Contact: <sip:7000@192.168.100.200:5060>
Expires: 180
Max-Forwards: 68
P-Asserted-Identity: "7000" <sip:7000@192.168.100.200>
Session-Expires: 1800
Content-Type: application/sdp
Content-Length: 254

v=0
o=CiscoSystemsSIP-GW-UserAgent 5871 9974 IN IP4 192.168.100.200
s=SIP Call
c=IN IP4 192.168.100.200
t=0 0
m=audio 8002 RTP/SAVP 0
c=IN IP4 192.168.100.200
a=rtpmap:0 PCMU/8000
a=ptime:20
```

Calls to Cisco Unity Connection are failing across Cisco Unified Border Element when callers try to select a menu prompt Why is this happening and how is it fixed?

- A. Cisco Unity Connection is configured on G.729 onl
- B. Cisco Unity Connection must be reconfigured to support iLBC.
- C. Cisco Unified Border Element is not sending any support for DTM
- D. OTMF configuration must be added to the appropriate dial peer.
- E. Cisco Unified Border Element is sending the incorrect media IP address
- F. The IP address of the loopback interface must be advertised in the SDP
- G. The Cisco Unity Connection Call Handler is configured for a "Release to Switch" transfer type Transfers must be disabled for the Cisco Unity Connection Call Handler

**Answer: B**

#### NEW QUESTION 6

How are E.164 called-party numbers normalized on a globalized call-routing environment in Cisco UCM?

- A. Call ingress must be normalized before the call being routed.
- B. Normalization is not required.
- C. Normalization is achieved by stripping or translating the called numbers to internally used directory numbers.
- D. Normalization is achieved by setting up calling search spaces and partitions at the SIP trunks for PSTN connection.

**Answer: C**

#### NEW QUESTION 7

Which option must be used when configuring the Local Gateway for a Cisco Webex Calling trunk?

- A. local authentication
- B. certificate-based
- C. mutual TLS
- D. Auth-based

**Answer: B**

#### Explanation:

A certificate-based trunk is a type of trunk that uses certificates to authenticate the connection between Webex Calling and the Local Gateway<sup>1</sup>. A Local Gateway is a supported session border controller that terminates the trunk on the premises<sup>2</sup>. A certificate-based trunk requires a certificate authority (CA) to issue and manage the certificates for both Webex Calling and the Local Gateway<sup>1</sup>.

#### NEW QUESTION 8

Which information is needed to restore the backup of a Cisco UCM publisher successfully?

- A. the TFTP server details
- B. the application credentials for Cisco UCM
- C. the security password for Cisco UCM
- D. the FTP server details

**Answer: C**

#### NEW QUESTION 9

An engineer configures a SIP trunk for MWI between a Cisco UCM cluster and Cisco Unity Connection. The Cisco UCM cluster fails to receive the SIP notify messages. Which two SIP trunk settings resolve this issue? (Choose two.)

- A. accept out-of-dialog refer
- B. accept out-of-band notification
- C. transmit security status
- D. allow changing header
- E. accept unsolicited notification

**Answer:** AE

#### NEW QUESTION 10

Which Cisco IM and Presence service handles failover and state changes in the cluster?

- A. XCP Sync Agent
- B. Cisco Server Recovery Manager
- C. Cisco XCP Connection Manager
- D. XCP router

**Answer:** B

#### NEW QUESTION 10

Which configuration concept allows for high-availability on IM and Presence services in a UC environment?

- A. IM and Presence subclusters (configured on Cisco UCM)
- B. Presence Redundancy Groups (configured on Cisco Unified IM and Presence)
- C. IM and Presence subclusters (configured on Cisco Unified IM and Presence)
- D. Presence Redundancy Groups (configured on Cisco UCM)

**Answer:** D

#### NEW QUESTION 13

An engineer is configuring a phone system CISCO UCM and wants to activate TFTP service. The engineer selects the serviceability page for configuration. Which nodes configurable for TFTP?

- A. any two nodes
- B. any node
- C. only nodes that have Cisco UCM service enabled
- D. any subscriber nodes

**Answer:** C

#### Explanation:

TFTP is a network protocol that is used to transfer files between devices. It is often used to transfer firmware and configuration files to network devices. In order to use TFTP, the device must have a TFTP server configured.

In Cisco UCM, the TFTP server is configured on the serviceability page. The TFTP server can be configured on any node that has Cisco UCM service enabled. The TFTP server cannot be configured on nodes that do not have Cisco UCM service enabled.

#### NEW QUESTION 17

An engineer is configuring Cisco Jabber for Windows and must implement desk phone control mode for some of the users. Which access control group must be configured for those users to enable this functionality?

- A. Allow Control of Device from CTI
- B. Standard CTI Secure Connection
- C. Standard CTI Enabled
- D. Standard CTI Allow Reception of SRTP Key Material

**Answer:** C

#### NEW QUESTION 18

An administrator is designing a new Cisco UCM for a company with many departments and firm structure on their communications policies. The administrator must make sure that these communication policies are reflected in the phone system setup. Certain departments cannot be accessed directly, even if they have dedicated DID numbers. Some phones, especially public phones, must not be able to dial international numbers Which type of function is configured to control which device is allowed to call another device in Cisco UCM?

- A. partitions and calling search spaces
- B. calling patterns and route patterns
- C. regions and device pools
- D. links and pipes

**Answer:** A

#### NEW QUESTION 23

A remote office has a less-than-optimal WAN connection and experiences packet loss, delay and jitter. Which VoIP codec is used in this situation?

- A. G722.1
- B. iLBC
- C. G.711alaw
- D. G.729A

**Answer:** B

#### NEW QUESTION 25

Which command in the MGCP gateway configuration defines the secondary Cisco UCM server?

- A. ccm-manager redundant-host
- B. ccm-manager fallback-mgcp
- C. mgcpapp
- D. mgcp call-agent

**Answer:** A

#### NEW QUESTION 28

Which two features of Cisco Prime Collaboration Assurance require advanced licensing? (Choose two.)

- A. real time alarm browse
- B. multicluster support
- C. call quality monitoring
- D. email notifications
- E. customizable events

**Answer:** BC

#### NEW QUESTION 29

Which wildcard must an engineer configure to match a whole domain in SIP route patterns?

- A. \*
- B. @
- C. !
- D. .

**Answer:** A

#### Explanation:

The asterisk (\*) wildcard is used to match any sequence of characters, including an empty sequence. Therefore, it can be used to match any domain name in a SIP Route Pattern.

The other options are not correct because:

- C. !: The ! symbol is used to negate a character class.
- D. .: The . symbol is used to match any single character.

#### NEW QUESTION 30

An engineer must configure codec on a Cisco Unified Border Element to prefer the G.711 ulaw and use G.711 codec as the next The engineer logs In to the CUBE, enters the dial.peer configuration level, and runs the voice-class codec 100 command. Which set of commands completes the configuration?

- A. voice class codec 100 codec g711alaw preference 1 codec a7Hulaw preference 2
- B. voice class codec 11j codec <?7iulaw preferred codec g7iialaw
- C. vice class codec 100 codec preference 1 g711ulaw codec preference 2 o711alaw
- D. voice class codec ::: codec g711ulaw g711alaw

**Answer:** C

#### Explanation:

The following commands are used to configure the codec on a Cisco Unified Border Element to prefer the G.711 ulaw and use G.711 alaw as the next codec:

Code snippet

```
voice class codec 100
```

```
codec preference 1 g711ulaw codec preference 2 g711alaw
```

The voice class codec 100 command creates a new voice class with the ID of 100. The codec preference 1 g711ulaw command sets the preference for the G.711 ulaw codec to 1. The codec preference 2 g711alaw command sets the preference for the G.711 alaw codec to 2.

#### NEW QUESTION 32

When designing the capacity for a Cisco UCM 12.x cluster, an engineer must decide which VMware template will be used for each node. What is the lowest number of users supported in a template and the highest number of users in a template?

- A. 750 and 15.000 users
- B. 750 and 10.000 users
- C. 500 and 10.000 users
- D. 1000 and 10.000 users

**Answer:** D

#### NEW QUESTION 36

Which endpoint feature is supported using Mobile and Remote Access through Cisco Expressway?

- A. SSO
- B. H.323 registration proxy to Cisco Unified Communications Manager
- C. MGCP gateway registration



D. SRST

**Answer:** A

#### NEW QUESTION 37

How does Cisco UCM perform a digit analysis on-hook versus off-hook for an outbound call from a Cisco IP phone that is registered to Cisco UCM?

- A. On-hook
- B. by pressing the digits and entering "#" to process the call
- C. UCM performs a digit-by-digit analysis; off-hook
- D. UCM analyzes all digits as a string.
- E. On-hook
- F. no digit analysis is performed; off-hook
- G. UCM requires the '#' to start the digit analysis
- H. On-hook
- I. UCM performs a digit-by-digit analysis; off-hook
- J. UCM considers all digits were dialed and does not wait for additional digits.
- K. On-hook
- L. UCM considers all digits were dialed and does not wait for additional digits; off-hook
- M. UCM performs a digit-by-digit analysis.

**Answer:** D

#### NEW QUESTION 40

Refer to the exhibit.

```
Via: SIP/2.0/TCP
10.10.10.2:5060;branch=a8bH5bK7954A198F
From:
<sip:012345678@10.10.10.2>;tag=8D79AF62-DB2
To: <sip:90123456@10.10.4.14>;
tag=811681~ffa80926-5fac-4cc5-b802-
2dbde74ae7w2-
v=0
o=CiscoSystemsCCM-SIP 811681 1 IN IP4
10.10.4.14
s=SIP Call
c=IN IP4 10.5.4.3
t=0 0
m=audio 27839 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

Which Codec is negotiated?

- A. G.729
- B. ILBC
- C. G.711ulaw
- D. G.728

**Answer:** C

#### NEW QUESTION 44

An administrator is trying to change the default LINECODE for a voice ISDN T1 PRI. Which command makes this change?

- A. linecode ami
- B. linecode b8zs
- C. linecode hdb3
- D. linecode esf

**Answer:** A

#### NEW QUESTION 45

Which location must be assigned to the SIP trunk to replicate enhanced location information via a SIP trunk?

- A. phantom
- B. replica
- C. hub\_none
- D. shadow

**Answer:** D

#### NEW QUESTION 49

Refer to the exhibit.

```
dial-peer voice 10 voip
  destination-pattern 1...
  session target ipv4:10.1.1.1
  no vad
```

An engineer configures a VoIP dial peer on a Cisco gateway. Which codec is used?

- A. G711alaw
- B. No codec is used (missing codec command)
- C. G.711ulaw
- D. G729r8

**Answer: D**

#### NEW QUESTION 53

Which two functions are provided by Cisco Expressway Series? (Choose two.)

- A. voice and video transcoding
- B. voice and video conferencing
- C. interworking of SIP and H.323
- D. intercluster extension mobility
- E. endpoint registration

**Answer: AC**

#### Explanation:

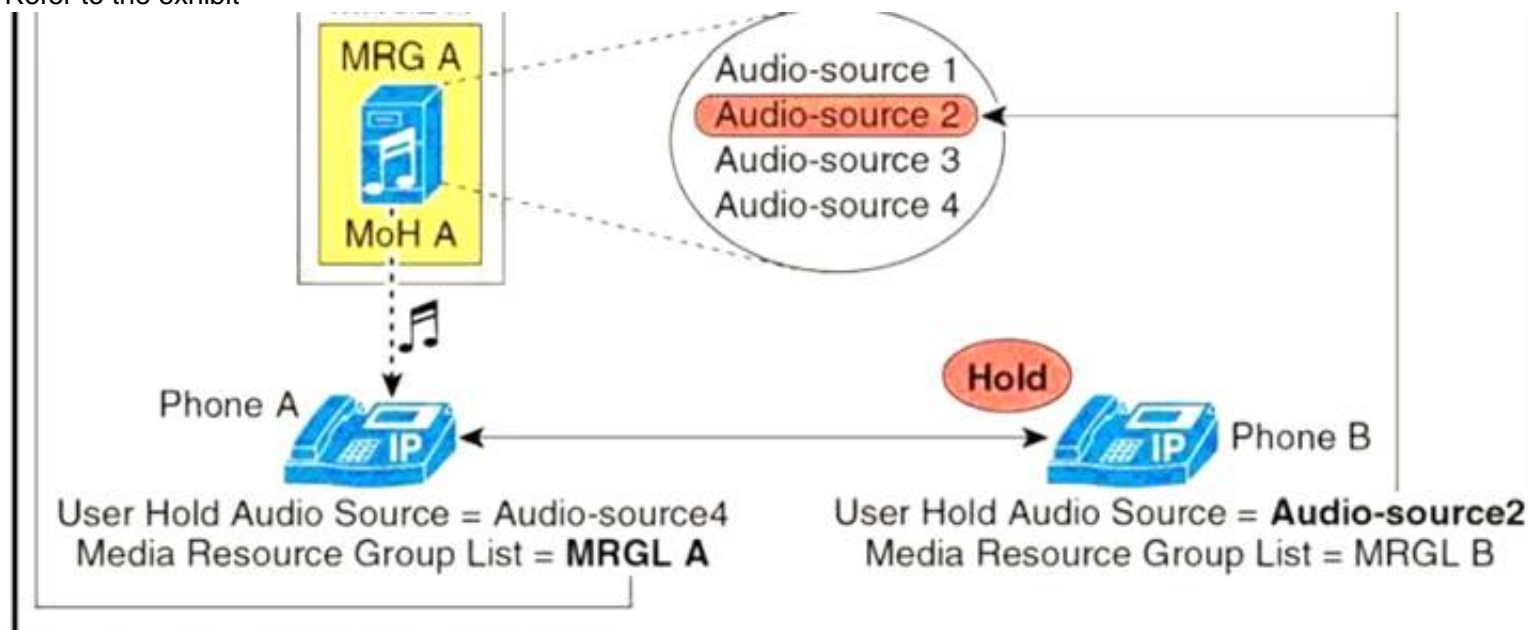
The Cisco Expressway Series provides the following functions:

- > Voice and video transcoding
- > Interworking of SIP and H.323
- > Firewall traversal
- > Session border controller (SBC) functionality
- > Endpoint registration
- > Call admission control (CAC)
- > Quality of service (QoS)
- > Security

The Cisco Expressway Series does not provide voice and video conferencing or intercluster extension mobility.

#### NEW QUESTION 54

Refer to the exhibit



There is a call flow between Phone A and Phone B Phone B (holder) places Phone A (holder) on hold Which MRGL and Audio Source are played to Phone A?

- A. MRGL A and Audio Source 4
- B. MRGL B and Audio Source 4
- C. MRGL A and Audio Source 2
- D. MRGL B and Audio Source 2

**Answer: C**

#### NEW QUESTION 56

An engineer with ID012345678 must build an international dial plan in Cisco UCM. Which action is taken when building a variable-length route pattern?

- A. configure single route pattern for international calls
- B. set up all international route patterns to 0.!
- C. reduce the T302 timer to less than 4 seconds
- D. create a second route pattern followed by the # wildcard

**Answer: D**

**Explanation:**

When building a variable-length route pattern, you need to create a second route pattern followed by the # wildcard. This will allow the user to indicate the end of the number by dialing #. For example, if you want to create a route pattern for international calls, you would create a route pattern like this: 9.011!#

This route pattern will match any number that starts with 9.011, followed by any number of digits, and then ends with #.

The other options are incorrect because:

- Configuring a single route pattern for international calls will not allow the user to indicate the end of the number.
- Reducing the T302 timer to less than 4 seconds will not allow the user to indicate the end of the number.

**NEW QUESTION 60**

Endpoint A is attempting to call endpoint B. Endpoint A only supports G.711ulaw with a packetization rate of 20 ms, and endpoint B supports packetization rate of 30 ms for G.711ulaw. Which two media resources are allocated to normalize packetization rates through transrating? (Choose two.)

- A. software MTP on Cisco IOS Software
- B. software MTP on Cisco UCM
- C. software transcoder on Cisco UCM
- D. hardware transcoder on Cisco IOS Software
- E. hardware MTP on Cisco IOS Software

**Answer:** BE

**NEW QUESTION 65**

A company wants to provide remote user with access to its premises Cisco collaboration features. Which components are required to enable cisco mobile and remote access for the users?

- A. Cisco Unified Border Element, Cisco IM and Presence Server, and Cisco Video Communication Server
- B. Cisco Expressway-E Cisco Expressway-C and Cisco UCM
- C. Cisco Expressway-E, Cisco IM and Presence Server, and Cisco Video Communication Server
- D. Cisco Unified Border Element
- E. Cisco UCM, and Cisco Video Communication Server

**Answer:** B

**NEW QUESTION 68**

An administrator has been asked to implement toll fraud prevention in Cisco UCM Which tool is used to begin this process?

- A. Cisco UCM class of service
- B. Cisco Unified Mobility
- C. Cisco UCM Access Control Group restrictions
- D. Cisco Unified Real-Time Monitoring Tool

**Answer:** A

**NEW QUESTION 73**

To provide high-quality voice and take advantage of the full voice feature set, which two access layer switches provide support?

- A. Use multiple egress queues to provide priority queuing of RTP voice packet streams and the ability to classify or reclassify traffic and establish a network trust boundary.
- B. Use 802, 1Q trunking and 802.1p for proper treatment of Layer 2 CoS packet marking on ports with phones connected.
- C. Implement IP RTP header compression on the serial Interface to reduce the bandwidth required per voice call on point-to-point links.
- D. Deploy RSVP to improve VoIP QoS only where it can have a positive impact on quality and functionality where there is limited bandwidth and frequent network congestion.
- E. Map audio and video streams of video calls (AF41 and AF42) to a class-based queue with weighted random early detection.

**Answer:** AB

**NEW QUESTION 78**

What are two Cisco UCM location bandwidths that are deducted when G 729 and G.711 codecs are used? (Choose two.)

- A. If a call uses G.729. Cisco UCM subtracts 16k.
- B. If a call uses G.711, Cisco UCM subtracts 64k
- C. If a call uses G.711, Cisco UCM subtracts 80k
- D. If a call uses G.729. Cisco UCM subtracts 24k.
- E. If a call uses G.729. Cisco UCM subtracts 40k

**Answer:** CD

**NEW QUESTION 79**

Refer to the exhibit.

```
000193: Dec 5 14:35:31.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:32.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:33.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:34.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
```

Given this "debug isdn q921" output, what is the problem with the PRI?



- A. Nothing, the PRI is sending keepalives.
- B. Layer 2 is down on the controller.
- C. PRI does not have an IP address configured on the interface.
- D. Layer 1 is down on the controller.

**Answer:** B

#### NEW QUESTION 83

Refer to the exhibit.

```
ISDN Serial1:23 interface
    dsl 1, interface ISDN Switchtype =
primary-5ess
    Layer 1 Status:
        ACTIVE
    Layer 2 Status:
        TEI = 0, Ces = 1, SAPI = 0, State =
TEI_ASSIGNED
    Layer 3 Status:
        0 Active Layer 3 Call(s)
    Activated dsl 1 CCBs = 0
    The Free Channel Mask: 0x807FFFFF
    Total Allocated ISDN CCBs = 5
```

What causes the PRI issue?

- A. The controller shut down
- B. The cable is unplugged
- C. The framing is configured incorrectly
- D. The clock source is incorrect.

**Answer:** B

#### Explanation:

The show controller t1 command shows that the T1 interface is up but the line protocol is down. This indicates that the physical layer is working but the data link layer is not. The most likely cause of this is that the cable is unplugged.

#### NEW QUESTION 88

A user dials 9011841234567 to reach Vietnam. Which steps send the call to the PSTN provider as 011841234567?

A)

in the Called Party Transformation Pattern Configuration section,  
 configure the Pattern as 9.011841234567  
 configure the Discard Digits as Predot

B)

in the Calling Party Transformation Patterns section,  
 configure the Pattern as 9.011841234567  
 configure the Discard Digits as Predot 10.10-Dialing

C)

in the Calling Party Transformation Patterns section,  
 configure the Pattern as 9.011841234567  
 configure the Discard Digits as Predot

D)

in the Called Party Transformation Pattern Configuration section,  
 configure the Pattern as 9.011841234567  
 configure the Discard Digits as Predot 10.10-Dialing

- A. Option A
- B. Option B
- C. Option C
- D. Option D

**Answer:** A

#### NEW QUESTION 93

What are two reasons that AF41 is marked for the audio and video channels of a video call? (Choose two.)

- A. to prioritize video over other high -priority traffic classes
- B. to give video calls a higher priority than AP41 in the QoS policy
- C. to allow high-definition quality calls over low-speed links
- D. to preserve lip synchronization between the audio and video channels
- E. to provide separate classes for audio calls and video calls

**Answer:** DE

#### NEW QUESTION 98

An administrator uses the Cisco Unified Real-Time Monitoring Tool to investigate recent calls on a Cisco UCM cluster. The SIP trace for an on-net. direct-media call shows two 180 Ringing and two 11 BYE messages. Why are there multiples of each message type in the trace?

- A. The source phone sends a 180 Ringing signal to the Cisco UC
- B. which sends a 180 Ringing signal to the destination phon
- C. The same process applies to 11 BYE messages.
- D. The source phone must signal to the destination phone that it is ringing, and the destination phone signals back with a 180 Ringing messag
- E. The same process applies to 11 BYE messages.
- F. The calls have an MTP in the call path due to different codec suppor
- G. The calls are subsequently split into two call legs.
- H. The destination phone signals back to the Cisco UCM that it is ringing, and the Cisco UCM signals back to the source phone.

**Answer:** A

#### NEW QUESTION 101

Which configuration on Cisco UCM is required for SIP MWI to work?

- A. Assign an MWI extension on the mailbox.
- B. The line partition must be inside the inbound CSS assigned to the CUC SIP trunk.
- C. The line partition must be inside the rerouting CSS assigned to the Cisco Unity Connection SIP trunk.
- D. Set the "Enable message waiting indicator" on the port group.

**Answer:** B

#### Explanation:

The line partition must be inside the inbound CSS assigned to the CUC SIP trunk. This ensures that the SIP MWI messages are sent to the correct destination. The other options are incorrect because:

- Assigning an MWI extension on the mailbox is not required for SIP MWI to work.
- The line partition does not need to be inside the rerouting CSS assigned to the Cisco Unity Connection SIP trunk.
- Setting the "Enable message waiting indicator" on the port group is not required for SIP MWI to work.

#### NEW QUESTION 106

Why would we not include an end user's PC device in a QoS trust boundary?

- A. The end user could incorrectly tag their traffic to advertise their PC as a default gateway.
- B. The end user could incorrectly tag their traffic to bypass firewalls.
- C. There is no reason not to include an end user's PC device in a QoS trust boundary.
- D. The end user may incorrectly tag their traffic to be prioritized over other network traffic.

**Answer:** D

#### NEW QUESTION 108

An engineer is notified that the Cisco TelePresence MX800 that is registered in Cisco Unified communications Manager shows an empty panel, and the Touch 10 shows a corresponding icon with no action when pressed. Where does the engineer go to remove the inactive custom panel?

- A. The phone configuration page in CUCM Administration
- B. The SIP Trunk security profile page in CUCM Administration
- C. The software Upgrades page in CUCM OS Administration
- D. The In-Room control Editor on the webpage of the MX800

**Answer:** D

#### NEW QUESTION 110

Which field is configured to change the caller ID information on a SIP route pattern?

- A. Route Partition
- B. Calling Party Transformation Mask
- C. Called Party Transformation Mask
- D. Connected Line ID Presentation

**Answer:** B

#### NEW QUESTION 114

A SIP phone has been configured in the system with MAC address 0030.96D2.D5CB. The phone retrieves the configuration file from the Cisco UCM. Which naming format is the file that is downloaded?

- A. SIP003096D2D5CB.cnf.xml

- B. SEP003096D2D5CB.cnf.xml
- C. SEP003096D2D5CB.cnf
- D. SIP003096D2D5CB.cnf

**Answer:** B

#### NEW QUESTION 117

An engineer implements a new Cisco UCM based telephony system per these requirements.

- The local Ethernet bandwidth is sized based on the total bandwidth per call
- A G 736 codec is used.
- The bit rate is 64 kbps
- The codec sample interval is 10 ms
- The voice payload size is 160 bytes per 20 ms

What should the size of the Ethernet bandwidth be per call?

- A. 31.2 kbps
- B. 38.4 kbps
- C. 55.2 kbps
- D. 87.2 kbps

**Answer:** D

#### NEW QUESTION 118

How are network devices monitored in a collaboration network?

- A. The Cisco Discovery Protocol table is shared among devices.
- B. Ping Sweep reports "unmanaged" state devices.
- C. System logs are collected in a Cisco Prime Collaboration Server.
- D. Simple Network Managed Protocol is enabled on each device to poll specific values periodically.

**Answer:** C

#### NEW QUESTION 123

Where in Cisco UCM are codec negotiations configured for endpoints?

- A. under device profiles in device settings
- B. in in-service parameters
- C. under regions using preference lists
- D. in enterprise parameters

**Answer:** C

#### NEW QUESTION 126

An administrator is in the process of moving Cisco Unity Connection mailboxes between mailbox stores. The administrator notices that some mailboxes have active Message Waiting Indicators. What happens to these mailboxes when they are moved?

- A. The move will fail if MWI status is active.
- B. The MWI status is retained after a mailbox is moved from one store to another.
- C. If the source and target mailbox store are not disabled, MWI status is not retained.
- D. Moving the mailboxes from one store to another fails if MWI is turned on.

**Answer:** B

#### NEW QUESTION 128

Which two protocols are proxied over an Expressway-E/C pair when a Mobile and Remote Access login including phone services is performed? (Choose two.)

- A. HTTPS
- B. H.323
- C. SIP
- D. SCCP
- E. SRTP

**Answer:** AC

#### NEW QUESTION 131

Refer to the exhibit.

```
05:50:14.102: ISDN BR0/1/1 Q921: User TX -> IDREQ ri=21653 ai=127
05:50:14.134: ISDN BR0/1/1 Q921: User RX <- SABMEp sapi=0 tei=0
05:50:14.150: ISDN BR0/1/1 Q921: User TX -> IDREQ ri=19004 ai=127
05:50:14.165: ISDN BR0/1/1 Q921 User RX <- SABMEp sapi=0 tei=0
```

A customer submits this debug output, captured on a Cisco IOS router. Assuming that an MGCP gateway is configured with an ISDN BRI interface, which BRI changes resolve the issue?

A.

```
interface BRI0/1/0
no ip address
isdn switch-type basic-net3
isdn point-to-multipoint-setup
isdn incoming-voice voice
isdn send-alerting
isdn static-tei 0
```

B. 

```
interface BRI0/1/1
no ip address
isdn switch-type basic-net3
isdn point-to-multipoint-setup
isdn incoming-voice voice
isdn send-alerting
isdn static-tei 0
```

C. 

```
interface BRI0/1/1
no ip address
isdn switch-type basic-net3
isdn point-to-point-setup
isdn incoming-voice voice
isdn send-alerting
isdn static-tei 0
```

D. 

```
interface BRI0/1/1
no ip address
isdn switch-type basic-net3
isdn incoming-voice voice
isdn send-alerting
isdn static-tei 0
```

**Answer:** C

#### NEW QUESTION 136

Which DiffServe PHB preserves backward compatibility with any IP precedence scheme?

- A. expedited forwarding
- B. class selector
- C. assured forwarding
- D. default

**Answer:** B

#### NEW QUESTION 141

An administrator is asked to implement toll fraud prevention in Cisco UCM, specifically to restrict off-net to off-net call transfers. How is this implemented?

- A. Enforce ad-hoc conference restrictions.
- B. Set the appropriate service parameter.
- C. Implement time-of-day routing.
- D. Use the correct route filters.

**Answer:** B

#### Explanation:

To restrict off-net to off-net call transfers, an administrator can set the "Block Offnet to Offnet Transfer" service parameter to "On". This will prevent users from transferring calls from one external number to another external number.

The other options are not correct because:

- A. Enforce ad-hoc conference restrictions: This will prevent users from creating ad-hoc conferences, but it will not prevent them from transferring calls.
- C. Implement time-of-day routing: This will allow calls to be routed to different destinations based on the time of day, but it will not prevent users from transferring calls.
- D. Use the correct route filters: This will allow calls to be filtered based on the destination, but it will not prevent users from transferring calls.

#### NEW QUESTION 144

A company deploys centralized cisco ucm architecture for a hub location and two remote sites.

\*The company has only one ITSP connection at the hub connection, and ITSP supports only G.711 calls

\*Remote site A has a 1-Gbps fiber connection to the hub connection and calls to and from remote side A use G.711 codec

\*Remote site B has a 1 T1 connection to the hub location and calls to and from remote site B use G.729 codec Based on the provided guidance, a Cisco voice



engineer must design media resource management for the customer What is the method that needs to be followed?

- A. configure the hardware transcoder on the site B router
- B. configure the hardware transcoder on the site A router
- C. configure the hardware transcoder on the hub location router
- D. configure the software transcoder on Cisco UCM to support voice calls to and from both remote sites

**Answer:** C

#### NEW QUESTION 145

SIP proxies have operations defined in RFC 3261 and supporting extensions. Though no IETF RFC completely defines how SBCs must function. SBCs evolved over the years.

Which two operations demonstrate the high-level differences between SBCs and SIP proxies? (Choose two.)

- A. Stateful proxies are context-aware and can terminate communication sessions by themselves
- B. SIP proxies add a Via header and optionally a Record-Route header, and the rest of the headers are left untouched
- C. SBCs can modify headers such as To, From, Contact, and Call-ID. It can introduce new headers into the SIP message
- D. SBCs are capable of interworking completely different protocols to set up, modify, and tear down communication session
- E. It includes SIP, H.323, and MGCP protocols
- F. SIP proxies are SDP-aware and can change the SDP bodies

**Answer:** BD

#### NEW QUESTION 150

An engineer is configuring IP telephony. The network relies on DHCP to provide TFTP server addresses to the endpoints. Policy requires the endpoints to receive two server addresses. Which DHCP option must be configured?

- A. 66
- B. 143
- C. 150
- D. 166

**Answer:** C

#### NEW QUESTION 154

What is a capability of a Cisco IOS XE media resource?

- A. It provides a hardware conferencing solution.
- B. It provides call forwarding capabilities.
- C. It provides redundancy for voice calls.
- D. It provides a voice packet optimization solution.

**Answer:** A

#### Explanation:

A Cisco IOS XE media resource provides a hardware conferencing solution. It can be used to mix multiple media streams, such as audio and video, into a single stream that can be sent to all participants in a conference call. This is done using a digital signal processor (DSP), which is a specialized processor that is designed to handle the processing of digital signals, such as audio and video.

#### NEW QUESTION 158

A customer asked to integrate Unity Connection with Cisco UCM using SIP protocol. Which two features must be enabled on SIP security profiles? (Choose two.)

- A. accept presence subscription
- B. allow changing header
- C. accept unsolicited notification
- D. enable application-level authorization
- E. accept replaces header

**Answer:** CE

#### NEW QUESTION 160

An administrator must make a pattern to route calls to two different destinations john.doe@company.com and jane.doe@company.com Which type of patterns are needed in the Cisco UCM. and what must the pattern look like?

- A. A SIP route pattern that looks like the \*@company.com
- B. A SIP route pattern that looks like this company.com
- C. A regular route pattern with URI feature enable in the configuration page
- D. The pattern must look like this: (\*@company.com)
- E. A regular route pattern with URI feature enable in the configuration page
- F. The pattern must look like this: MATCH(\*@company.com)

**Answer:** C

#### NEW QUESTION 162

In which location does an administrator look to determine which subscriber the phone registers to if loses registration with the current Cisco UCM subscriber?

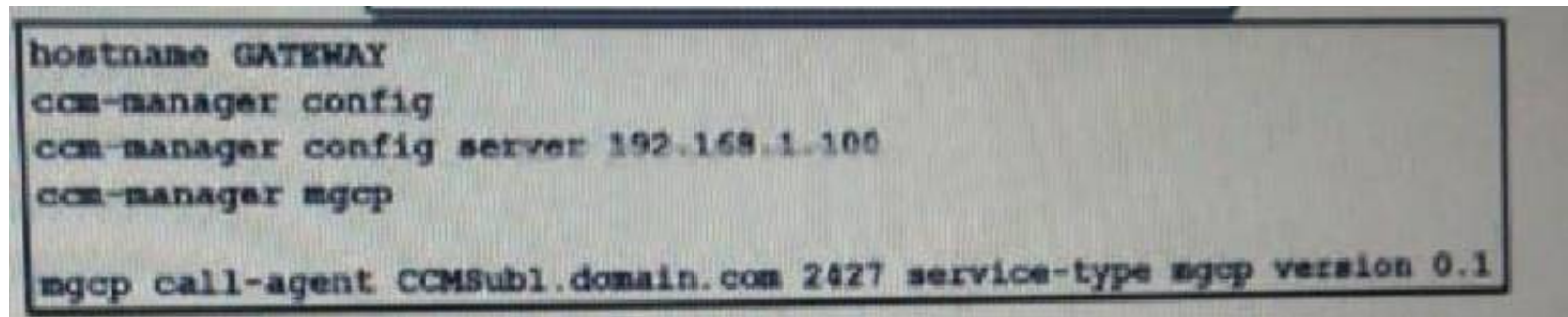


- A. On Cisco UCM Administration Page Device > Phone > Phone Configuration page
- B. On Cisco UCM Administrator Page server > Cisco UCM
- C. On Cisco UCM Administrator page system > Device Pool > Cisco UCM group
- D. On Cisco UCM Administrator page system > Enterprise Parameters

**Answer:** C

#### NEW QUESTION 167

Refer to the exhibit.



An engineer verifies the configured of an MGCP gateway. The commands are already configured. Which command is necessary to enable MGCP?

- A. Device(config)# mgcp enable
- B. Device(config)# ccm-manager enable
- C. Device (config) # com-manager active
- D. Device (config)# mgcp

**Answer:** D

#### NEW QUESTION 172

Which task is required when configuring self-provisioning for an end user in Cisco UCM?

- A. Enable Auto-Registration.
- B. Associate the end user to the Standard CCM Super Users group
- C. Associate the end user to a SIP Profile.
- D. Disable Auto-Registration.

**Answer:** A

#### NEW QUESTION 175

What is the traffic classification for voice and video conferencing?

- A. Voice is classified as CoS 4, and video conferencing is CoS 5.
- B. Voice and video conferencing are both classified is CoS 3.
- C. Voice is classified as CoS 5, and video conferencing is CoS 4.
- D. Video conferencing is classified as CoS 1, and voice is CoS 2.

**Answer:** B

#### NEW QUESTION 176

What are two access management mechanisms in Cisco Webex Control Hub? (Choose two.)

- A. multifactor authentication
- B. Active Directory synchronization
- C. attribute-based access control
- D. single sign-on with Google
- E. Client ID/Client Secret

**Answer:** AB

#### Explanation:

The correct answers are A and B.

The two access management mechanisms in Cisco Webex Control Hub are multifactor authentication and Active Directory synchronization.

Multifactor authentication is a security measure that requires users to provide two or more pieces of evidence to verify their identity. This can include something they know, such as a password, and something they have, such as a security token.

Active Directory synchronization is a process that allows Cisco Webex Control Hub to automatically synchronize user accounts from an Active Directory domain. This can simplify user management and provide users with single sign-on access to Cisco Webex Control Hub and other applications.

#### NEW QUESTION 178

An engineer configures local route group names to simplify a dial plan. Where does the engineer set the route groups according to the local route group names that are configured?

- A. route list
- B. device pool
- C. CSS
- D. route pattern

**Answer:** B

#### NEW QUESTION 182

Which Cisco Unified communications manager configuration is required for SIP MWI integration?

- A. Select "Redirecting Diversion Header Delivery— Inbound" on the SIP trunk
- B. Enable "Accept presence subscription" on the SIP trunk security profile
- C. Select "Redirecting Diversion Header Delivery – outbound" on the SIP trunk
- D. Enable "Accept unsolicited notification" on the SIP Trunk security profile

**Answer:** D

#### NEW QUESTION 184

Drag and drop the SNMPv3 message types from the left onto the corresponding definitions on the right.

TRAP	messages used to modify a value of an object variable
SET	unreliable messages that alert the SNMP manager to a condition on the network
GET	reliable messages that alert the SNMP manager to a condition on the network
INFORM	messages used to retrieve an object instance

- A. Mastered
- B. Not Mastered

**Answer:** A

#### Explanation:

Table Description automatically generated

#### NEW QUESTION 186

What is a characteristic of video traffic that governs QoS requirements for video?

- A. Video is typically constant bit rate.
- B. Voice and video are the same, so they have the same QoS requirements.
- C. Voice and video traffic are different, but they have the same QoS requirements.
- D. Video is typically variable bit rate.

**Answer:** D

#### NEW QUESTION 189

An administrator executes the debug isdn q931 command while debugging a failed call. After a test call is placed, the logs return a disconnect cause code of 1. What is the cause of this problem?

- A. The media resource is unavailable.
- B. The destination number rejects the call.
- C. The destination number is busy.
- D. The dialed number is not assigned to an endpoint.

**Answer:** D

#### NEW QUESTION 191

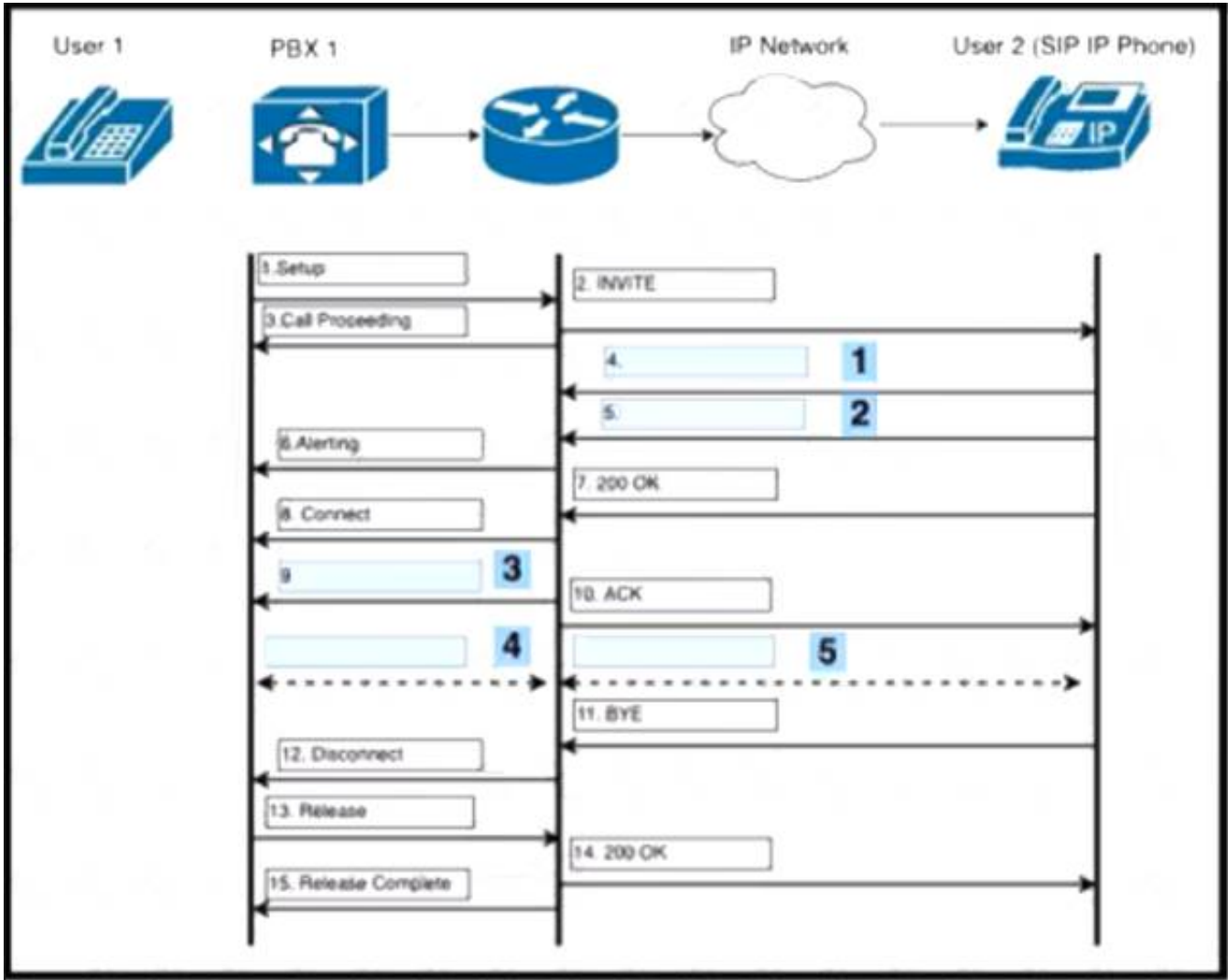
What is a characteristic of a SIP endpoint configured in Cisco UCM with 'Use Trusted Relay Point' set to "On"?

- A. It creates a trust relationship with the called party.
- B. It enables the Use Trusted Relay Point setting from the associated common device configuration.
- C. It enables Cisco UCM to insert an MTP or transcoder designated as a TRP.
- D. If TRP is allocated and MTP is also required for the endpoint
- E. calls fail.

**Answer:** C

#### NEW QUESTION 194

Refer to the exhibit.



https://i.postimg.cc/wMYy0Fhm/image.png

Drag and drop the flow step labels from the left into the correct order on the right to establish this call flow:

- User 1 calls user 2.
- User 2 answers the call
- user 2 disconnects the call

two-way voice path

two-way RTP channel

100 Trying

Connect ACK

180 Ringing

- A. Mastered
- B. Not Mastered

Answer: A

Explanation:

- \* 1. 100 Trying
- \* 2. 180 Ringing
- \* 3. two-way voice path
- \* 4. Connect ACK
- \* 5. two-way RTP channel

NEW QUESTION 196

A collaboration engineer troubleshoots issues with a Cisco IP Phone 7800 Series. The IPv4 address of the phone is reachable via ICMP and HTTP, and the phone is registered to Cisco UCM However the engineer cannot reach the CU of the phone Which two actions in Cisco UCM resolve the issue? (Choose two)

- A. Enable SSH Access under Product Specific Configuration Layout in Cisco UCM
- B. Disable Web Access under Product Specific Configuration Layout in Cisco UCM
- C. Set a username and password under Secure Shell information in Cisco UCM
- D. Enable Settings Access under Product Specific Configuration Layout in Cisco UCM
- E. Enable FIPS Mode under Product Specific Configuration Layout in Cisco UCM

Answer: AB

#### NEW QUESTION 197

Exhibit.

```
admin:utils ntp status
ntpd (pid 14550) is running...
```

```

      remote           refid      st t   when   poll   reach   delay
offset jitter
=====
*192.168.1.1    17.253.14.125      2 u   39    64     3    0.456   -0.236
0.116
*192.168.1.2    17.253.14.125      2 u   38    64     3    0.817   -0.695
0.395
```

Refer the exhibit. A collaboration engineer needs to replace the original, single NTP server that was configured during the initial install of a Cisco UCM server. What is the first step to accomplish this task?

- A. Restart the NTP service on Cisco UCM
- B. Delete the original NTP server from Cisco UCM
- C. Stop the NTP service on Cisco UCM
- D. Enable NTP authentication for the new NTP server on Cisco UCM

**Answer: B**

#### NEW QUESTION 200

A customer reports that the Cisco UCM toll-fraud prevention does not work correctly, and the customer is receiving charges for unapproved international calls as a result. Which two configuration changes resolve the issues? (Choose two.)

- A. Mark patterns as off-net or on-net.
- B. Modify the Block OffNet to OffNet Transfer service parameter.
- C. Disable call forwarding on the phone.
- D. Use Cisco Unified Border Element to debug the calls.
- E. Make the calls route through a firewall.

**Answer: AB**

#### NEW QUESTION 201

Which action prevent toll fraud in Cisco Unified Communication Manager?

- A. Configure ad hoc conference restriction
- B. Implement toll fraud restriction in the Cisco IOS router
- C. Allow off-net to off-net transfer
- D. Implement route patterns in Cisco Unified CM

**Answer: A**

#### NEW QUESTION 202

An administrator works with an ISDN PRI that is connected to a third-party PBX. The ISDN link does not come up, and the administrator finds that the third-party PBX uses the OSIG signaling method. Which command enables the Cisco IOS Gateway to use QSIG signaling on the ISDN link?

- A. isdn incoming-voice voice
- B. isdn switch-type basic-ni
- C. isdn switch-type basic-qsig
- D. isdn switch-type primary-qsig

**Answer: D**

#### NEW QUESTION 204

Refer to the exhibit.



```
INVITE sip:1@10.10.10.219;user=phone SIP/2.0
Via: SIP/2.0/TCP 10.10.10.84:50083;branch=z9hG4bK471df613
From: "1234 - My Phone" <sip:1234@10.10.10.219>;tag=381c1aba7a78002c558eda31-12b8af63
To: <sip:1@10.10.10.219>
Call-ID: 381c1aba-7a78000d-4ca6894a-41dd3e0f@10.10.10.84
Max-Forwards: 70
CSeq: 101 INVITE
Contact: <sip:1234@10.10.10.84:50083;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Allow-Events: kpml,dialog
Content-Type: application/sdp
Content-Length: 658

v=0
o=Cisco-SIPUA 26529 0 IN IP4 10.10.10.84
s=SIP Call
b=AS:4064
t=0 0
m=audio 32136 RTP/AVP 114 9 124 113 115 0 8 116 18
c=IN IP4 10.10.10.84
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114
maxplaybackrate=16000;sprop-maxcapture=16000;maxaveragebitrate=64000;stereo=0;sprop-
stereo=0;usedtx=0
```

When a UC Administrator is troubleshooting DTMF negotiated by this SIP INVITE, which two messages are examined next to further troubleshoot the issue? (Choose two.)

- A. REGISTER
- B. SUBSCRIBE
- C. PRACK
- D. NOTIFY
- E. UPDATE

Answer: BD

#### NEW QUESTION 209

Which Cisco UCM configuration is required for SIP MWI integrations?

- A. Enable "Accept presence subscription" on the SIP Trunk Security Profile.
- B. Select "Redirecting Diversion Header Delivery - Outbound" on the SIP trunk.
- C. Enable "Accept unsolicited notification" on the SIP Trunk Security Profile.
- D. Select "Redirecting Diversion Header Delivery - Inbound" on the SIP trunk.

Answer: C

#### NEW QUESTION 214

Refer to the exhibit.

The screenshot shows the configuration page for a Cisco Unified Communications Manager server. It is divided into two main sections: "Auto-registration Information" and "Cisco Unified Communications Manager TCP Port Settings for this Server".

**Auto-registration Information:**

- Universal Device Template: Auto-registration Template
- Universal Line Template: Sample Line Template with TAG usage examples
- Starting Directory Number: 1000
- Ending Directory Number: 2000
- ☒ Auto-registration Disabled on this Cisco Unified Communications Manager

**Cisco Unified Communications Manager TCP Port Settings for this Server:**

- Ethernet Phone Port: 2000
- MGCP Listen Port: 2427
- MGCP Keep-alive Port: 2428
- SIP Phone Port: 5060
- SIP Phone Secure Port: 5061

At the bottom, there are buttons for "Save", "Reset", and "Apply Config".

Which action must an engineer take to implement self-provisioning on a primary communications manager server?

- A. Select a different Universal Line Template.
- B. Change the SIP Phone Secure Port.
- C. Uncheck the auto-registration Disabled checkbox.

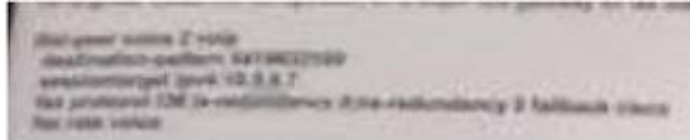


D. Select a different Universal Device Template.

Answer: C

#### NEW QUESTION 219

An engineer builds the configuration on a Cisco IOS gateway for the dial-peers:



Which command is required to complete the configuration?

- A. Codec g726r32
- B. Codec g729cr81
- C. Codec g723ar63
- D. Codec g711ulaw

Answer: D

#### NEW QUESTION 220

An administrator is developing an 8-class QoS baseline model. The CS3 standards-based marking recommendation is used for which type of class?

- A. Scavenger
- B. best effort
- C. voice
- D. call signaling

Answer: A

#### NEW QUESTION 224

Refer to the exhibit.

The screenshot shows the 'SIP Trunk Security Profile Information' configuration page. The 'Name' field is 'CUP Non Secure SIP Profile'. The 'Device Security Mode' is 'Non Secure'. The 'Incoming Transport Type' is 'TCP+UDP'. The 'Outgoing Transport Type' is 'TCP'. The 'Enable Digest Authentication' checkbox is unchecked. The 'Nonce Validity Time (mins)' is '600'. The 'Secure Certificate Subject or Subject Alternate Name' field is empty. The 'Incoming Port' is '5060'. The 'Enable Application level authorization' checkbox is unchecked. The 'Accept presence subscription' checkbox is checked. The 'Accept out-of-dialog refer' checkbox is checked. The 'Accept unsolicited notification' checkbox is unchecked. The 'Accept replaces header' checkbox is unchecked. The 'Transmit security status' checkbox is unchecked. The 'Allow charging header' checkbox is unchecked. The 'SIP V.150 Outbound SDP Offer Filtering' dropdown is set to 'Use Default Filter'.

A collaboration engineer is configuring the Cisco UCM IM and Presence Service. Which two steps complete the configuration of the SIP trunk security profile? (Choose two.)

- A. Check the box to enable application-level authorization.
- B. Check the box to allow charging header.
- C. Check the box to accept unsolicited notification.

- D. Check the box to transmit security status.
- E. Check the box to accept replaces header.

**Answer:** CE

#### NEW QUESTION 225

What is a description of the DiffServ model used for implementing QoS?

- A. AF41 has higher drop precedence than AF42. which has higher drop precedence than AF43.
- B. Voice and video calls are marked with different DSCP values and placed in different queues.
- C. AF43 has higher drop precedence than AF42 but lower drop precedence than AF41.
- D. RTP traffic from voice and video calls is marked EF and placed in the same queue.

**Answer:** A

#### NEW QUESTION 228

Refer to the exhibit.

```
isdn switch-type primary-ni
controller t1 0/1/0
framing esf
linecode b8zs
pri-group timeslots 1-10
```

An engineer configures ISDN on a voice gateway. The provider confirms that the PRI is configured with 10 channels the engineer ordered and is working from the provider side, but the engineer cannot get a B-channel to carry voice. The rest of the configuration for the serial interface and voice network is functioning correctly. Which actions must be taken to carry voice?

- A. The engineer must activate the controller card on the voice gateway before configuring the device.
- B. The engineer used a T1 interface but must use an E1 interface.
- C. The pri-group timeslots command must be 0-9 for the 10 channels because all values on a router start with 0.
- D. The engineer must manually revert the order of using the channels.

**Answer:** A

#### NEW QUESTION 233

An engineer troubleshoots outbound can failure on an ISDN-PRI circuit. The engineer ts suspecting the 'Incomplete Destination'. Which debugs or commands are run in the voice gateway to troubleshoot the issue?

- A. debug isdn q921term mon
- B. debug voip ecapi inout show controller ti
- C. debug isdn q931 show isdn status
- D. debug isdn q921 debug voip ecapi inout

**Answer:** C

#### Explanation:

The engineer should run the following debugs or commands in the voice gateway to troubleshoot the issue: ➤ debug isdn q931 - This debug will show the ISDN Q.931 messages that are being exchanged between the voice gateway and the ISDN switch. This can be used to identify the cause of the "Incomplete Destination" error.

➤ show isdn status - This command will show the status of the ISDN PRI circuit. This can be used to verify that the circuit is up and running.

The other options are not correct. The debug isdn q921 command will show the ISDN Q.921 messages that are being exchanged between the voice gateway and the ISDN switch. This is not necessary for troubleshooting the issue. The term mon command will show the terminal monitor output. This is not necessary for troubleshooting the issue. The debug voip ecapi inout command will show the VoIP ECAP messages that are being exchanged between the voice gateway and the VoIP server. This is not necessary for troubleshooting the issue. The show controller ti command will show the status of the T1 controller. This is not necessary for troubleshooting the issue.

#### NEW QUESTION 235

An engineer troubleshoots a Cisco Jabber login problem on a Windows PC. The login fails with the error message "Cannot find your services automatically. Click advanced settings to set up manually " Which action should the engineer take first?

- A. Verify whether the cup-xmpp certificates are valid.
- B. Verify the username and password and try again.
- C. Perform a manual DNS lookup of SRV record \_cisco-uds.\_tcp.domain.com.
- D. Perform a manual DNS lookup of SRV record \_collab-edge.\_tls.domain.com.

**Answer:** C

#### NEW QUESTION 239

Which service must be enabled when LDAP on cisco UCM is used?

- A. Cisco AXL Web Service
- B. Cisco CallManager SNMP Service
- C. Cisco DirSync
- D. Cisco Bulk Provisioning Service

**Answer:** C

#### NEW QUESTION 244

Cisco UCM delays routing of a call during digit analysis with an overlapping dial plan. How long is the default wait time?

- A. 5 seconds
- B. 10 seconds
- C. 15 seconds
- D. 20 seconds

**Answer:** C

#### NEW QUESTION 247

What is the purpose of a hybrid Local Gateway?

- A. to handle calls between Webex Calling and Cisco Calling Plans
- B. to handle calls between Webex Calling and Cloud Connected PSTN
- C. to handle calls between Cisco IJCM and Webex Calling
- D. to handle calls between the Public Switched Telephone Network and Webex Calling

**Answer:** D

#### Explanation:

A hybrid local gateway handles calls between the Public Switched Telephone Network (PSTN) and Webex Calling. It is commonly deployed on the customer's premises but can also be hosted by a partner. The local gateway registers with Webex Calling and handles all calls between the PSTN and Webex Calling. It gives customers the flexibility to bring their own service provider or continue using their existing provider for a smooth and effective transition to the cloud.

#### NEW QUESTION 250

Which two technical reasons make QoS a necessity in a video deployment? (Choose Two)

- A. Low response time between endpoints
- B. Provisioned bandwidth of the link
- C. Variable bit rate of the video stream
- D. Bursly behavior of video traffic

**Answer:** CD

#### NEW QUESTION 252

What is the function of the Cisco Unity Connection Call Handler?

- A. routes calls to a user based on caller input
- B. queues calls
- C. allows customized scripts for IVR capabilities
- D. searches a list of extensions until the call is answered

**Answer:** A

#### Explanation:

A Cisco Unity Connection Call Handler is a software application that answers calls, plays greetings, and routes calls to users based on caller input. Call handlers can be used to create automated attendants, voice menus, and other interactive voice response (IVR) applications.

Call handlers are created and managed using the Cisco Unity Connection Administration interface. When creating a call handler, you can specify a variety of settings, including the greeting that is played, the caller input options that are available, and the destination that calls are routed to.

Call handlers are a powerful tool that can be used to create a variety of IVR applications. By using call handlers, you can improve the efficiency of your organization's communications and provide a better experience for your callers.

Here are some additional tips for using call handlers:

- Use call handlers to create automated attendants that can answer calls and route them to the appropriate person or department.
- Use call handlers to create voice menus that can provide callers with information or options.
- Use call handlers to create interactive voice response (IVR) applications that can collect information from callers and process their requests.

#### NEW QUESTION 253

An administrator configures Cisco UCM to use UDP for SIP signaling and finds that an endpoint cannot make calls. Which action resolves this issue?

- A. Change the common phone profile.
- B. Change the SIP dial rules.
- C. Change the SIP profile.
- D. Change the phone security profile.

**Answer:** D

#### NEW QUESTION 255

The IP phones at a customer site do not pick an IP address from the DHCP. An engineer must temporarily disable LLDP on all ports of the switch to leave only CDP. Which two commands accomplish this task? (Choose two.)

- A. Switch# copy running-config startup-config
- B. Switch(config)# no lldp run
- C. Switch# configure terminal
- D. Switch(config)# interface GigabitEthernet1/0/1
- E. Switch(config)# no lldp transmit

Answer: BC

### NEW QUESTION 258

When setting a new primary DNS server in the Cisco UCM CLI what is required for the change to take affect?

- A. restart of CallManager service
- B. restart of DirSync service
- C. restart of the network service
- D. restart of TFTP service

Answer: C

### NEW QUESTION 263

Given this H.323 gateway configuration and using Cisco best practices, how must the called party transformation pattern be configured to ensure that a proper ISDN type of number is set?

```
voice translation-rule 40
 rule 1 /3...$/ /408555&/
!
voice translation-profile INT
 translate calling 40
!
dial-peer voice 9011 pots
 translation-profile outgoing INT
 destination-pattern 9011T
 port 0/1/0:23
```

A.

Pattern Definition	
Pattern *	\+.
Partition	PT_US_VG_CD_Out_xForm
Description	US International calling
Numbering Plan	< None >
Route Filter	< None >
<input checked="" type="checkbox"/> Urgent Priority <input type="checkbox"/> MLPP Preemption Disabled	

Called Party Transformations	
Discard Digits	PreDot
Called Party Transformation Mask	
Prefix Digits	9011
Called Party Number Type *	International
Called Party Numbering Plan *	Private

B.

Pattern Definition	
Pattern *	\+.
Partition	PT_US_VG_CD_Out_xForm
Description	US International calling
Numbering Plan	< None >
Route Filter	< None >
<input checked="" type="checkbox"/> Urgent Priority <input type="checkbox"/> MLPP Preemption Disabled	

Called Party Transformations	
Discard Digits	PreDot
Called Party Transformation Mask	
Prefix Digits	9011
Called Party Number Type *	Cisco CallManager
Called Party Numbering Plan *	Cisco CallManager



C.

Pattern Definition	
Pattern *	\+.
Partition	PT_US_VG_CD_Out_xForm
Description	US International calling
Numbering Plan	< None >
Route Filter	< None >
<input checked="" type="checkbox"/> Urgent Priority <input type="checkbox"/> MLPP Preemption Disabled	

Called Party Transformations	
Discard Digits	PreDot
Called Party Transformation Mask	
Prefix Digits	9011
Called Party Number Type *	International
Called Party Numbering Plan *	ISDN

D.

Pattern Definition	
Pattern *	\+.
Partition	PT_US_VG_CD_Out_xForm
Description	US International calling
Numbering Plan	< None >
Route Filter	< None >
<input checked="" type="checkbox"/> Urgent Priority <input type="checkbox"/> MLPP Preemption Disabled	

Called Party Transformations	
Discard Digits	PreDot
Called Party Transformation Mask	
Prefix Digits	9011
Called Party Number Type *	Unknown
Called Party Numbering Plan *	Unknown

Answer: C

#### NEW QUESTION 268

An administrator is configuring LDAP for Cisco UCM with Active Directory integration. A customer has requested to use "ipphone" instead of "telephoneNumber" as the phone number attribute. Where does the administrator specify this attribute mapping in Cisco UCM?

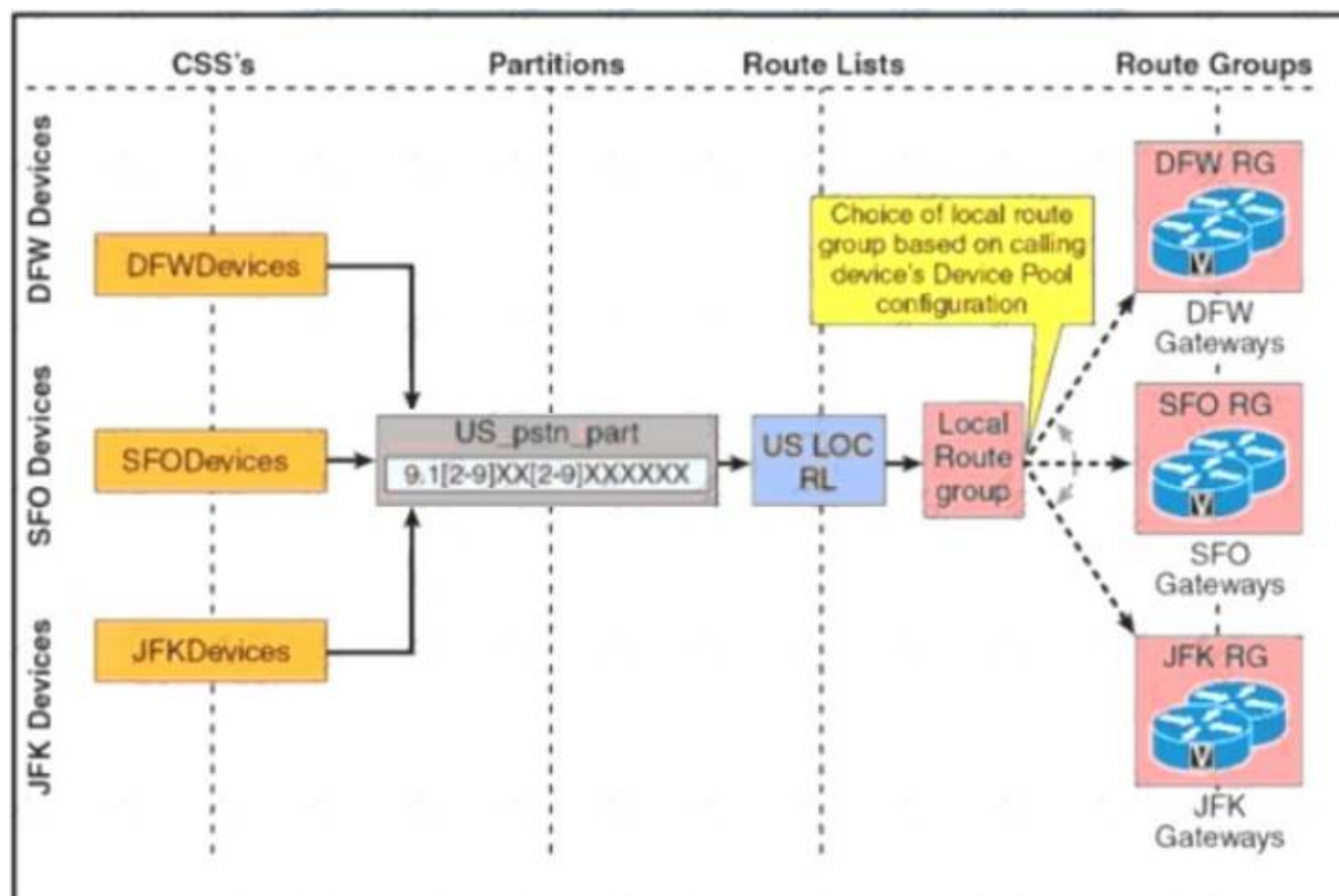
- A. LDAP Custom Filter
- B. LDAP Directory user fields
- C. LDAP Directory custom user fields
- D. LDAP Authentication

Answer: B

#### NEW QUESTION 273

Refer to the exhibit.





A user takes a phone from San Francisco to New York for a short reassignment. The phone was set up to use the San Francisco device pool, and device mobility is enabled on the Cisco UCM. The user makes a call that matches a route pattern in a route list that contains the Standard Local Route Group. To where does the call retreat?

- A. The call fails because device mobility is turned on, and the phone is not configured in New York
- B. The engineer must configure which sites the device should be roaming to.
- C. The call egresses in San Francisco because the user uses device mobility and is allowed to roam while still keeping the number and resources assigned in San Francisco.
- D. The call fails because the Standard Local Route Group is being used only if no configuration is set for the device pools.
- E. The call egresses in New York because the device automatically is assigned a New York device pool and uses the local gateway.

Answer: B

#### NEW QUESTION 277

Refer to the exhibit.

```
INVITE sip:2002@10.10.10.10:5060 SIP/2.0
[..truncated..]
v=0
o=UAC 6107 7816 IN IP4 10.10.10.11
s=SIP Call
c=IN IP4 10.10.10.11
t=0 0
m=audio 8190 RTP/AVP 18 110
c=IN IP4 10.10.10.11
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:110 telephone-event/8000
a=fmtp:110 0-16
a=ptime:20

SIP/2.0 200 OK
[..truncated..]
v=0
o=UAS 4692 9609 IN IP4 10.10.10.10
s=SIP Call
c=IN IP4 10.10.10.10
t=0 0
m=audio 8056 RTP/AVP 18
c=IN IP4 10.10.10.10
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
```

The SDP offer/answer has been completed successfully but there is no DTMF when users press keys. What is the cause of the issue?

- A. Payload type 110 was negotiated rather than type 101.
- B. DTMF was negotiated property in these messages.
- C. DTMF was not negotiated on the call.
- D. G.729 rather than G.711ulaw was negotiated.

**Answer:** C

#### NEW QUESTION 282

Refer to exhibit.

The screenshot shows the Cisco Expressway-C web interface. The 'DNS lookup' tab is selected. The 'Host' field contains 'company.com', the 'Query type' is set to 'All', and the 'Check against the following DNS servers' is set to 'All'. A 'Lookup' button is visible. Below the form, a table displays the results of the DNS lookup:

Query type	Name	TTL	Class	Type	Response
A	company.com	60	IN	A	52.223.45.27
A	company.com	60	IN	A	35.71.162.193

A company recently deployed CISCO Jabber Users log in to Jabber by using their email address in a domain named company.com. The users report that they cannot register their telephony services when working from unless they use a VPN. An engineer runs DNS lookup tool in Cisco Expressway-C to troubleshoot the issue. What is the cause of the issue?

- A. The company.com domain must be resolved only in Expressway-E
- B. There is a missing SRV record for the company.com domain.
- C. The TTL value for the company.com is too short.
- D. There must be only one response for the company.com domain

**Answer:** B

#### NEW QUESTION 287

Why isn't an end user's PC device in a QoS trust boundary included?

- A. The end user could incorrectly tag their traffic to bypass firewalls.
- B. The end user may incorrectly tag their traffic to be prioritized over other network traffic.
- C. There is no reason not to include an end user's PC device in a QoS trust boundary.
- D. The end user could incorrectly tag their traffic to advertise their PC as a default gateway.

**Answer:** B

#### NEW QUESTION 290

Which behavior occurs when Cisco UCM has a Call Manager group that consists of two subscribers?

- A. Endpoints attempt to register with the top subscriber in the list.
- B. Endpoints attempt to register with the bottom subscriber in the list.
- C. Endpoints attempt to register with both subscribers in a load-balanced method.
- D. If a subscriber is rebooted, endpoints deregister until the rebooted system is back in service.

**Answer:** A

#### NEW QUESTION 293

Which call flow matches traffic from a Mobile and Remote Access registered endpoint to central call control?

- A. Endpoint>Expressway-C>Expressway-E>Cisco UCM
- B. Endpoint>Expressway-E>Expressway-C> Cisco UCM
- C. Endpoint>Expressway-E> Cisco UCM
- D. Endpoint>Expressway-C> Cisco UCM

**Answer:** A

#### Explanation:

The call flow for a Mobile and Remote Access registered endpoint to central call control is as follows:

- > The endpoint registers with the Expressway-C.
- > The Expressway-C forwards the registration request to the Expressway-E.
- > The Expressway-E forwards the registration request to the Cisco UCM.

- The Cisco UCM registers the endpoint.
- When the endpoint places a call, the call flow is as follows:
- The endpoint sends the call request to the Expressway-C.
  - The Expressway-C forwards the call request to the Expressway-E.
  - The Expressway-E forwards the call request to the Cisco UCM.
  - The Cisco UCM places the call.

The Expressway-C and Expressway-E are used to provide secure access to the Cisco UCM for endpoints that are not located on the corporate network. The Expressway-C is located on the corporate network, and the Expressway-E is located in the DMZ.

#### NEW QUESTION 298

An engineer configures a Cisco Unified Border Element and must ensure that the codecs negotiated meet the ITSP requirements. The ITSP supports G.711ulaw and G.729 for audio and H.264 for video. The preferred voice codec is G.711. Which configuration meets this requirement?

A.

```
voice class codec 10
  codec preference 1 g711ulaw
  codec preference 2 g729r8

dial-peer voice 101 voip
  session protocol sipv2
  destination e164-pattern-map 1
  voice-class codec 10
```

B.

```
voice class codec 10
  codec preference 1 g729r8
  codec preference 2 g711ulaw
  video codec h264

dial-peer voice 101 voip
  session protocol sipv2
  destination e164-pattern-map 1
  voice-class codec 10
```

C.

```
voice class codec 10
  codec preference 1 g711ulaw
  codec preference 2 g729r8
  video codec h264

dial-peer voice 101 voip
  session protocol sipv2
  destination e164-pattern-map 1
  voice-class codec 100
```

D.



```
voice class codec 10
  codec preference 1 g711ulaw
  codec preference 2 g729r8
  video codec h264

dial-peer voice 101 voip
  session protocol sipv2
  destination e164-pattern-map 1
  voice-class codec 10
```

Answer: D

#### NEW QUESTION 299

What is required when deploying co-resident VMs by using Cisco UCM?

- A. Provide a guaranteed bandwidth of 10 Mbps.
- B. Deploy the VMs to a server running Cisco UCM.
- C. Avoid hardware oversubscription.
- D. Ensure that applications will perform QoS.

Answer: C

#### Explanation:

When deploying co-resident VMs by using Cisco UCM, it is important to avoid hardware oversubscription. This means that you should not assign more resources to the VMs than the physical hardware can provide. For example, if you have a server with 16 CPU cores, you should not assign more than 16 CPU cores to the VMs.

If you oversubscribe the hardware, the VMs will not be able to get the resources they need to run properly. This can lead to performance problems and even outages.

To avoid hardware oversubscription, you should carefully plan your VM deployments. You should also monitor the performance of the VMs to make sure that they are not overusing the resources.

Here are some additional tips for deploying co-resident VMs by using Cisco UCM: ➤ Use a virtualization platform that supports Cisco UCM.

- Make sure that the VMs have the correct operating system and software installed.
- Configure the VMs to use the correct network settings.
- Monitor the performance of the VMs to make sure that they are running properly.

#### NEW QUESTION 300

Refer to the exhibit.

```
000142: *Apr 23 19:41:49.050: MGCP Packet received from 192.168.100.100:2427--->
AUEP 4 AALN/S0/SU0/0@VG320.cisco.local MGCP 0.1
F: X, A, I
<---

000143: *Apr 23 19:41:49.050: MGCP Packet sent to 192.168.100.101:2427--->
200 4
I:
X: 2
L: p:10-20, a:PCMU:PCMA:G.nx64, b:64, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-220, a:G.729:G.729a:G.729b, b:8, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-110, a:G.726-16:G.728, b:16, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-70, a:G.726-24, b:24, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-50, a:G.726-32, b:32, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:30-270, a:G.723.1-H:G.723:G.723.1a-H, b:6, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:30-330, a:G.723.1-L:G.723.1a-L, b:5, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
M: sendonly, recvonly, sendrecv, inactive, loopback, contest, data, netwloop, netwtest
<---
```

What is the registration state of the analog port in this debug output?

- A. The analog port failed to register to Cisco UCM with an error code 200.
- B. The MGCP Gateway is not communicating with the Cisco UCM.
- C. The analog port is currently shut down.
- D. The analog port is registered to Cisco UCM.

Answer: D

#### NEW QUESTION 305

Which call routing pattern is used for phone numbers that are in the E.164 format?

- A. /+! Route Pattern
- B. \+! Route pattern
- C. \+! Translation Pattern
- D. \+1.[2-9]XX[2-9]XXXXXXX called Party Transformation Pattern

**Answer:** B

**NEW QUESTION 307**

Which Cisco unity Connection handler plays a greeting at announces the option to dial a user extension by default?

- A. the operator call handler
- B. the Interview handler
- C. the Goodbye call handler
- D. the Directory handler

**Answer:** A

**NEW QUESTION 310**

An administrator configures international calling on a Cisco UCM cluster and wants to minimize the number of route patterns that are needed. Which route pattern enables the administrator to match variable-length numbers?

- A. 9.011#
- B. 9.011@
- C. 9.011!
- D. 9.011\*

**Answer:** C

**NEW QUESTION 312**

In the cisco expressway solution, which two features does mobile and Remote access provide? (Choose two)

- A. VPN-based enterprise access for a subset of Cisco Unified IP Phone models
- B. secure reverse proxy firewall traversal connectivity
- C. the ability to register third-party SIP or H.323 devices on Cisco UCM without requiring VPN
- D. the ability of Cisco IP Phones to access the enterprise through VPN connection
- E. the ability for remote users and their devices to access and consume enterprise collaboration applications and services

**Answer:** BE

**NEW QUESTION 313**

Which two types of trunks can be used when configuring a hybrid Local Gateway for Cisco Webex Calling? (Choose Two.)

- A. TLS-based
- B. certificate-based
- C. registration-based
- D. authentication-based
- E. OAuth-based

**Answer:** AC

**Explanation:**

These are the two types of trunks that can be used when configuring a hybrid local gateway for Cisco Webex Calling. A TLS-based trunk uses Transport Layer Security (TLS) to secure the SIP signaling between the hybrid local gateway and Webex Calling. A registration-based trunk uses SIP registration to authenticate the hybrid local gateway with Webex Calling and receive calls from the cloud.

**NEW QUESTION 315**

An engineer deploys a Cisco Expressway-E server for a customer who wants to utilize all features on the server. Which feature does the engineer configure on the Expressway-E?

- A. H.323 endpoint registrations
- B. Mobile and Remote Access
- C. SIP gateway for PSTN providers
- D. VTC bridge

**Answer:** A

**NEW QUESTION 318**

An administrator needs to create a partial PRI consisting of the first seven timeslots available. Which configuration snippet configures the ISDN E1 PRI for this task?

- A. 

```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#interface Serial0/0/0:15
2900(config-controller)#pri-group timeslots 1-7
```

B.



```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#controller e1 0/0/0
2900(config-controller)#pri-timeslots 1-7
```

- C. 

```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#controller e1 0/0/0
2900(config-controller)#pri-group timeslots 1-7
```
- D. 

```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#pri-group timeslots 1-7
```

**Answer:** C

#### NEW QUESTION 321

Which two steps are required for bulk configuration transactions on the Cisco UCM database utilizing BAT? (Choose two.)

- A. A data file in Abstract Syntax Notation One format must be uploaded to Cisco UCM
- B. A server template must be created in Cisco UCM
- C. A data file in comma-separated values format must be uploaded to Cisco UCM
- D. A data file in Extensible Markup Language format must be uploaded to Cisco UCM
- E. A device template must be created in Cisco UCM

**Answer:** CE

#### NEW QUESTION 322

Which service on the Presence Server is responsible for maintaining the point-to-point chat connections between Jabber clients?

- A. Cisco SIP Proxy
- B. Cisco XCP Text Conference Manager
- C. Cisco XCP Router
- D. Cisco XCP XMPP Federation Manager

**Answer:** B

#### NEW QUESTION 324

An administrator installs a new Cisco TelePresence video endpoint and receives this error: "AOR is not permitted by Allow/Deny list. Which action should be taken to resolve this problem?

- A. Reboot the VCS server and attempt reregistration.
- B. Change the SIP trunk configuration.
- C. Correct the restriction policy settings.
- D. Upload a new policy in VCS.

**Answer:** C

#### Explanation:

The error message "AOR is not permitted by Allow/Deny list" indicates that the endpoint is not allowed to register with the VCS server because it is not on the Allow List or it is on the Deny List. To resolve this problem, you must correct the restriction policy settings.

#### NEW QUESTION 329

An engineer must configure switch port 5/1 to send CDP packets to configure an attached Cisco IP phone to trust tagged traffic on it's access port. Which command is required to complete the configuration?

```
Router# configure terminal
Router(config)# interface gigabitethernet 5/1
Router config-if# description Cube E41.228-0097
```

- A. platform qos trust extend cos 3
- B. platform qos trust extend
- C. platform qos extend trust
- D. platform qos trust extend cos 5

**Answer:** B

#### NEW QUESTION 334

Refer to the exhibit.

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Region configuration Related Links: [Back to Find/List](#)

Save Delete Copy Reset Apply Config Add New

**Region Information**

Name\* REGION1

**Region Relationships**

Region	Audio Codec Preference List Configuration	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, 6.711)	Use System Default (384 kbps)	Use System Default (2900000000 kbps)
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

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Save Delete Copy Add Item

**Status**

Status: Ready

**Audio Codec Preference Information**

Name\* CCNP COLLAB

Description\* CCNP COLLAB

Codecs in List

- G.722 48k
- G.711 U-Law 64k
- G.729 8k
- G.711 A-Law 56k

An engineer is troubleshooting this video conference issue:

\*A video call between a Cisco 9971 in Region1 and another Cisco 9971 in Region1 works.

\*As soon as the Cisco 9971 in Region1 conferences in a Cisco 8945 in Region2. the Region1 endpoint cannot see the Region2 endpoint video.  
What is the cause of this issue?

- A. Maximum Audio Bit Rate must be increased.
- B. Maximum Session Bit Rate for Immersive Video Calls is too low.
- C. Maximum Session Bit Rate for Video Calls is too low.
- D. Cisco 8945 does not have a camera connected.

Answer: C

## NEW QUESTION 335

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