

Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

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NEW QUESTION 1

What are two QoS requirements for VoIP traffic?

- A. Voice traffic must be marked to DSCP EF.
- B. Loss must be no more than 1 percent.
- C. Voice traffic must be marked to DSCP AF41.
- D. One-way latency must be no more than 200 ms.
- E. Average one-way jitter is greater than 50 ms.

Answer: AB

NEW QUESTION 2

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 3

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 4

Refer to the exhibit.

```
voice class codec 20
codec preference 1 g722-64
codec preference 2 ilbc mod 30
!
dial-peer voice 200 voip
destination-pattern ^408555...$
session target ipv4:10.2.3.4
incoming called-number 9T
dtmf-relay h245-alphanumeric rtp-nte|
no vad
!
```

An administrator configured a codec preference list with G.722 and ILBC codecs. Which change must the administrator make in the dial-peer section of the configuration to use this list?

- A. add voice-codecs 20
- B. add session codec 20
- C. add codec preference 20
- D. add voice-class codec 20

Answer: D

NEW QUESTION 5

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 6

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. G.722.1

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

Answer: B

NEW QUESTION 7

Which SNMP service must be activated manually on the Cisco Unified Communications Manager after installation?

- A. Cisco CallManager SNMP
- B. SNMP Master Agent
- C. Connection SNMP Agent
- D. Host Resources Agent

Answer: A

NEW QUESTION 8

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 9

A high-speed network is often configured with a five-class QoS model. Which classes are used in the model?

- A. real-time, call-signaling, critical data, best-effort, and scavenger
- B. real-time, signaling, critical data, best-effort and drop-class
- C. call-signaling, real-time, critical data, best-effort, and drop-class
- D. voice, video, signaling, critical data, and best-effort

Answer: A

NEW QUESTION 10

Which type of message must an administrator configure in the SIP Trunk Security Profile for a Message Waiting Indicator light to work with a SIP integration between Cisco UCM and Cisco Unity Connection?

- A. Unsolicited NOTIFY
- B. 200 ok
- C. SIP Register
- D. TCP port 5060

Answer: A

NEW QUESTION 10

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 13

On a Cisco Catalyst Switch, which command is required to send CDP packets on a switch port that configures a Cisco IP phone to transmit voice traffic in 802.10 frames, tagged with the voice VLAN ID 221?

- A. Device(config-if)# switchport voice vlan 221
- B. Device(config-if)# switchport vlan voice 221
- C. Device(config-if)# switchport access vlan 221
- D. Device(config-if)# switchport trunk allowed vlan 221

Answer: A

NEW QUESTION 18

An employee of company ABC just quit. The IT administrator deleted the employee's user id from the active directory at 10 a.m. on March 4th. The nightly sync occurs at 10 p.m. daily. The IT administrator wants to troubleshoot and find a way to delete the user id as soon as possible. How is this issue resolved?

- A. Wait until 10 pm on March 4th when the user is automatically removed from Cisco UCM.
- B. Wait until 10 pm on March 5th when the user is automatically removed from Cisco UCM.
- C. Wait until 3:15 a.m.
- D. On March 6th for garbage collection to remove the user from Cisco UCM.
- E. Wait until 3:15 a.m. on March 5th for garbage collection to remove the user from Cisco UCM.

Answer: C

NEW QUESTION 20

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 25

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 28

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 31

What are two features of Cisco Expressway that the customer gets if Expressway-C and Expressway-E are deployed? (Choose two.)

- A. highly secure free-traversal technology to extend organizational reach.
- B. additional visibility of the edge traffic in an organization.
- C. complete endpoint registration and monitoring capabilities for devices that are local and remote.
- D. session-based access to comprehensive collaboration for remote workers, without the need for a separate VPN client.
- E. utilization and adoption metrics of all remotely connected devices.

Answer: AD

NEW QUESTION 33

What is the major difference between the two possible Cisco IM and Presence high-availability modes?

- A. Balanced mode provides user load balancing and user failover in the event of an outag
- B. Active/standby mode provides an always on standbynode in the event of an outage, and it also provides load balancing.
- C. Balanced mode provides user load balancing and user failover only for manually generated failovers.Active/standby mode provides anunconfigured standby node in the event of an outage, but it does not provide load balancing.
- D. Balanced mode provides user load balancing and user failover in the event of an outag
- E. Active/standby mode provides an always on standbynode in the event of an outage, but it does not provide load balancing.
- F. Balanced mode does not provide user load balancing, but it provides user failover in the event of an outag
- G. Active/standby mode provides analways on standby node in the event of an outage, but it does not provide load balancing.

Answer: C

Explanation:

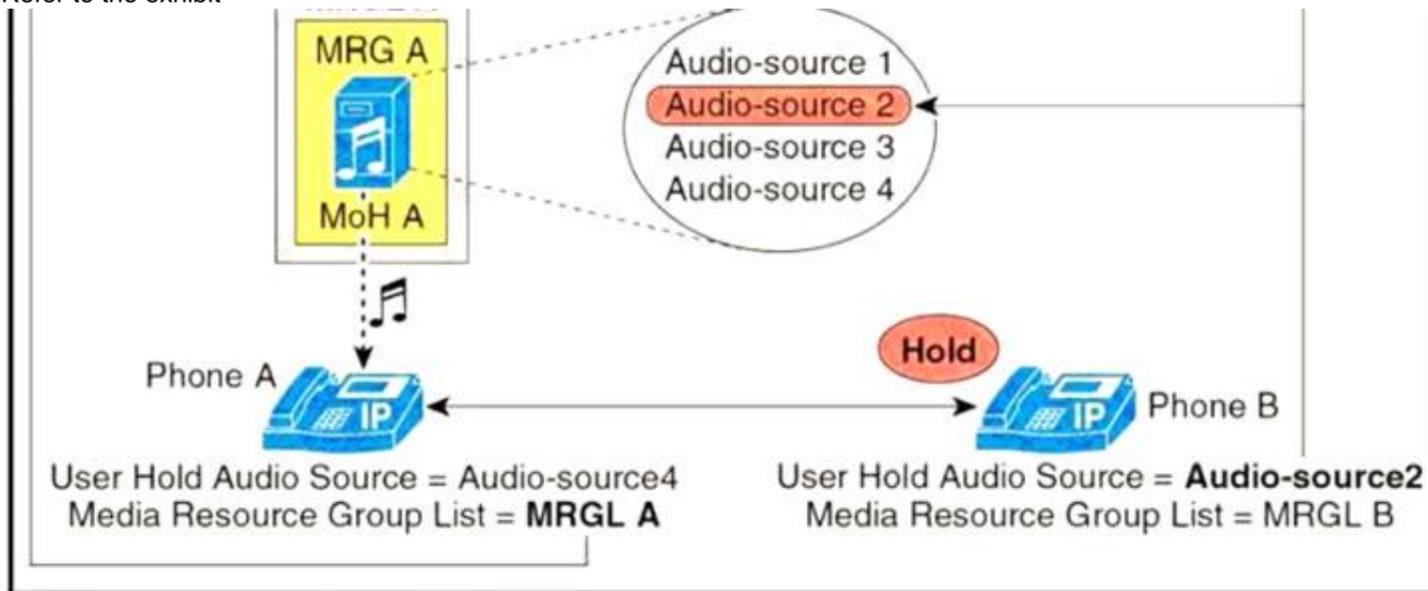
Balanced mode provides user load balancing and user failover in the event of an outage. Active/standby mode provides an always on standby node in the event of an outage, but it does not provide load balancing.

Here is a more detailed explanation of the two modes:

- > **Balanced mode:** In balanced mode, the IM and Presence Service nodes are configured to work together to provide high availability. The nodes are configured in a redundancy group, and the system automatically balances the load of users across the nodes in the group. If one of the nodes fails, the system automatically fails over the users to the other nodes in the group.
- > **Active/standby mode:** In active/standby mode, one of the IM and Presence Service nodes is designated as the active node, and the other nodes are designated as standby nodes. The active node handles all of the user traffic, and the standby nodes are only used if the active node fails. If the active node fails, the system automatically fails over to one of the standby nodes.

NEW QUESTION 37

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 42

How does traffic policing respond to violations?

- A. Excess traffic is dropped.
- B. Excess traffic is retransmitted.
- C. All traffic is treated equally.
- D. Excess traffic is queued.

Answer: A

NEW QUESTION 46

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 50

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 52

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 55

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 support
- 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 57

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 60

The chief officer at a company must reduce collaboration infrastructure costs by onboarding all on-premises equipment to the cloud by using CISCO Webex Control Hub. Administrators need the ability to manage upgrades and set up hot desking for on-premises devices.

Which action must be taken before on boarding devices by using the Control Hub?

- A. Configure tie Control Hub organization ID on the devices
- B. Acquire a license for each device.
- C. Allow HTTP traffic from each device to Control Hub.
- D. Upgrade all the devices to software version CE9.15 or later

Answer: D

Explanation:

This is a prerequisite for using the Device Connector tool, which allows you to onboard and register several devices simultaneously to the Webex Control Hub1. The Device Connector tool creates a workspace, an activation code, and activates all of your devices in one go1. This way you don't need to be physically present in the same room to activate the devices.

The other options are not required before onboarding devices by using the Control Hub:

- > Configuring the Control Hub organization ID on the devices is not necessary, as the Device Connector tool will send the device information to your Webex organization and generate activation codes for them 1.
- > Acquiring a license for each device is not necessary, as you can assign licenses to users and devices after they are registered to the Webex Control Hub2.
- > Allowing HTTP traffic from each device to Control Hub is not necessary, as HTTPS connectivity is required for the Device Connector tool to communicate with the devices1.

NEW QUESTION 64

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 69

An engineer wants to manually deploy a CISCO Webex DX80 Video endpoint to a remote user. Which type of provisioning is configured on the endpoint?

- A. Cisco Unified Border Element
- B. Cisco Unity Connection
- C. Cisco Meeting Server
- D. Edge

Answer: D

Explanation:

The Cisco Webex DX80 Video endpoint can be provisioned in two ways:

- > Automatically, using the Cisco Unified Communications Manager (CUCM) or Cisco Video Communication Server (VCS)
- > Manually, using the Edge provisioning mode

The Edge provisioning mode is used when the endpoint is not connected to the CUCM or VCS. In this mode, the endpoint is configured with the necessary settings, such as the IP address, SIP/H.323 parameters, and time and date.

The Cisco Unified Border Element (Cisco UBE) is a network element that provides security and call control for IP telephony networks. The Cisco Unity Connection is a unified messaging system that provides voicemail, email, and fax services. The Cisco Meeting Server is a video conferencing system that provides high-quality video and audio conferencing.

NEW QUESTION 74

An administrator configures the voicemail feature in a Cisco collaboration deployment. The user mailboxes must be configured when the Cisco Unity Connection server is configured. Which action accomplishes this task?

- A. Configure a SIP integration with Cisco UCM to sync users.
- B. Configure an SCCP integration with Cisco UCM.
- C. Configure an AXL server to access the Cisco UCM users.
- D. Configure an active directory to sync the users who will have a voicemail box.

Answer: C

NEW QUESTION 77

What are two functions of Cisco Expressway in the Collaboration Edge? (Choose two.)

- A. Expressway-C provides encryption (or Mobile and Remote Access but not (or business-to-business communications.
- B. The Expressway-C and Expressway-E pair can enable connectivity from the corporate network to the PSTN via a T1/E1 trunk.
- C. The Expressway-C and Expressway-E pair can interconnect H.323-to-SIP calls for voice.
- D. Expressway-E provides a VPN entry point for Cisco IP phones with a Cisco AnyConnect client using authentication based on certificates.
- E. Expressway-E provides a perimeter network that separates the enterprise network from the Internet.

Answer: CE

NEW QUESTION 78

A customer is deploying a SIP IOS gateway for a customer who requires that in-band DTMF relay is first priority and out-of-band DTMF relay is second priority. Which 10\$ entry sets the required priority?

- A. dtmf-relay cisco-rtp
- B. dtmf-relay sip-kpml cisco-rtp
- C. sip-notify dtmf-relay rtp-nte
- D. dtmf-relay rtp-nte sip-notify

Answer: D

NEW QUESTION 80

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 85

A network administrator deleted a user from the LDAP directory of a company. The end user shows as Inactive LDAP Synchronized User in Cisco UCM. Which step is next to remove this user from Cisco UCM?

- A. Delete the user directly from Cisco UCM.
- B. Wait 24 hours for the garbage collector to remove the user.
- C. Restart the Dirsync service after the user is deleted from LDAP directory.
- D. Execute a manual sync to refresh the local database and delete the end user.

Answer: B

NEW QUESTION 86

A company wants to provide remote users with access to its on-premises Cisco collaboration features. Which components are required to enable Cisco Mobile and Remote Access for the users?

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

Answer: C

NEW QUESTION 91

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 95

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 99

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 102

Which Webex Calling dial plan settings restrict a user from placing a particular outbound call type?

- A. Block
- B. Transfer to Number
- C. Reject
- D. Restrict

Answer: D

Explanation:

The Restrict setting in the Webex Calling dial plan prevents users from placing certain types of outbound calls. For example, you can use the Restrict setting to prevent users from making international calls or calls to premium-rate numbers.

The Block setting in the Webex Calling dial plan prevents users from placing any outbound calls. The Transfer to Number setting in the Webex Calling dial plan transfers all outbound calls to a specified number. The Reject setting in the Webex Calling dial plan rejects all outbound calls.

Here is a table summarizing the different dial plan settings and their effects:

Dial Plan Setting Effect

Block

Prevents users from placing any outbound calls. Transfer to Number

Transfers all outbound calls to a specified number. Reject

Rejects all outbound calls. Restrict

Prevents users from placing certain types of outbound calls.

NEW QUESTION 104

An engineer must manually provision a Cisco IP Phone 8845 using SIP. Which two fields must be configured for a successful provision? (Choose two.)

- A. media resources group list
- B. CSS
- C. location
- D. device security profile
- E. SIP profile

Answer: DE

NEW QUESTION 105

How many minutes does it take for automatic fallback to occur in a Presence Redundancy Group if the primary node lost a critical service?

- A. 5 min
- B. 10 min
- C. 30 min
- D. 60 min

Answer: C

NEW QUESTION 108

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 110

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 111

A collaboration engineer configures Global Dial Plan Replication for multiple Cisco UCM clusters. The local cluster acts as the hub cluster, and the remaining clusters act as spoke clusters Which service must the engineer configure on the local cluster'

- A. Intercluster Lookup Service
- B. Location Conveyance on intercluster SIP trunks
- C. Intra-Cluster Communication Signaling
- D. Mobility Cross Cluster

Answer: A

NEW QUESTION 116

Which DSCP value and PHB equivalent are the default for audio calls?

- A. 48 and EF
- B. 34 and AF41
- C. 32 and AF41
- D. 32 and CS4

Answer: A

NEW QUESTION 120

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 123

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 125

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 126

When a call is delivered to a gateway, the calling and called party number must be adapted to the PSTN service requirements of the trunk group. If a call is destined locally, the + sign and the explicit country code must be replaced with a national prefix. For the same city or region, the local area code must be replaced by a local prefix as applicable. Assuming that a Cisco UCM has a SIP trunk to a New York gateway (area code 917). which two combinations of solutions localize the calling and called party for a New York phone user? (Choose two.)

- A. Configure two calling party transformation patterns:
 \+1917.XXXXXXX, strip pre-dot, numbering type: subscriber
 \+1.!, strip pre-dot, numbering type: national
- B. Configure the gateway to translate called numbers and apply it to the dial peer. Combine it with a translation profile for calling number:

```

!
voice translation-rule 1
rule 1 /^1917/ //
rule 2 /^[+]1917/ //
!
voice translation-profile strip+1
translate called 1
!
            
```
- C. Configure the gateway to translate the calling number and apply it to the dial peer. Combine it with a translation profile for called number:

```

!
voice translation-rule 1
rule 1 /^1917/ //
rule 2 /^[+]1917/ //
!
voice translation-profile strip+1
translate calling 1
!
            
```
- D. Configure two called party transformation patterns:
 \+1917.XXXXXXX, strip pre-dot, numbering type: subscriber
 \+1.!, strip pre-dot, numbering type: national
- E. Configure two calling party transformation patterns:
 \+1917.CCCCCC, strip pre-dot, numbering type: subscriber
 \+!, strip pre-dot, numbering type: national

Answer: BC

NEW QUESTION 128

Refer to the exhibit.

The screenshot shows the Cisco Region Configuration interface. Under "Region Information", the "Name" is "Dallas-REG". Under "Region Relationships", there is a table with columns: Region, Audio Codec Preference List, Maximum Audio Bit Rate, and Maximum Session Bit Rate for Video Calls. The row for "SanJose-REG" shows "Use System Default (Factory Default low loss)", "24 kbps (AMR-WB)", and "Use System Default (384 kbps)". Below this, a note states "NOTE: Regions not displayed" with "Use System Default" for all three columns. Under "Modify Relationship to other Regions", a dropdown menu shows "Austin-REG", "Dallas-REG", "Default", and "SanJose-REG". Below the dropdown, there are two "Keep Current Setting" dropdown menus and radio buttons for "Keep Current Setting" and "Se" (likely "Set").

Which codec should an engineer select for a call made between "Dallas-REG" & "Austin-REG"?

- A. MP4A-LATM
- B. G.711
- C. OPUS
- D. G.729

Answer: D

Explanation:

The codec preference list for the "Dallas-REG" region is "Factory Default low loss". This list includes the following codecs in order of preference:

- > G.729
- > G.711
- > OPUS
- > MP4A-LATM

The codec preference list for the "Austin-REG" region is "Factory Default low loss". This list includes the following codecs in order of preference:

- > G.729
- > G.711
- > OPUS
- > MP4A-LATM

Since both regions have the same codec preference list, the codec that will be used for a call made between "Dallas-REG" and "Austin-REG" is G.729.

G.729 is a narrowband speech codec that was developed by the ITU-T in 1988. It is a low-bitrate codec that provides good quality speech at a bitrate of 8 kbps.

G.729 is widely used in VoIP applications and is the default codec for many VoIP systems.

G.711 is a wideband speech codec that was developed by the ITU-T in 1972. It is a high-bitrate codec that provides excellent quality speech at a bitrate of 64 kbps. G.711 is not as widely used as G.729 due to its high bitrate requirements.

OPUS is a lossy audio codec that was developed by the IETF in 2012. It is a low-bitrate codec that provides good quality speech at a bitrate of 6 kbps. OPUS is widely used in VoIP applications and is the default codec for many VoIP systems.

MP4A-LATM is a lossy audio codec that was developed by the IETF in 1999. It is a high-bitrate codec that provides excellent quality speech at a bitrate of 24 kbps. MP4A-LATM is not as widely used as G.729 or OPUS due to its high bitrate requirements.

NEW QUESTION 132

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 135

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 138

Which two DNS records must be created to configure Service Discovery for on-premises Jabber? (Choose two.)

- A. _cisco-uds._tls.cisco.com pointing to the IP address of Cisco UCM
- B. _cuplogin_tcp.cisco.com pointing to a record of IM and Presence
- C. _cuplogin._tls.cisco.com pointing to the IP address of IM and Presence
- D. _cisco-uds.tcp.cisco.com pointing to a record of Cisco UCM
- E. _xmpp.tls.cisco.com pointing to a record of IM and Presence

Answer: BD

NEW QUESTION 142

Refer to the exhibit. An engineer is confining class of control for a user in Cisco UCM. Which change will ensure that the user is unable to call 2143?

- A. Change line partition to Partition_A
- B. Change line CSS to only contain Partition_B
- C. Set the user's line CSS to <None>
- D. Set the user's device CSS to <None>

Answer: D

NEW QUESTION 146

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. Bursty behavior of video traffic

Answer: CD

NEW QUESTION 148

Which value should be changed when each Cisco UCM node does not allow for more than 5000 phones to be registered?

- A. Maximum Number of Registered and Unregistered Devices service parameter on each node
- B. Minimum Number of Phones service parameter on each node
- C. Maximum Number of Registered Devices service parameter on each node
- D. Maximum Number of Phones service parameter on the Publisher

Answer: C

NEW QUESTION 149

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 153

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

Description

Numbering Plan

Route Filter

Urgent Priority

MLPP Preemption Disabled

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type *

Called Party Numbering Plan *

B.

Pattern Definition

Pattern*

Partition

Description

Numbering Plan

Route Filter

Urgent Priority

MLPP Preemption Disabled

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type*

Called Party Numbering Plan*

C. **Pattern Definition**

Pattern*

Partition

Description

Numbering Plan

Route Filter

Urgent Priority

MLPP Preemption Disabled

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type*

Called Party Numbering Plan*

D. **Pattern Definition**

Pattern*

Partition

Description

Numbering Plan

Route Filter

Urgent Priority

MLPP Preemption Disabled

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

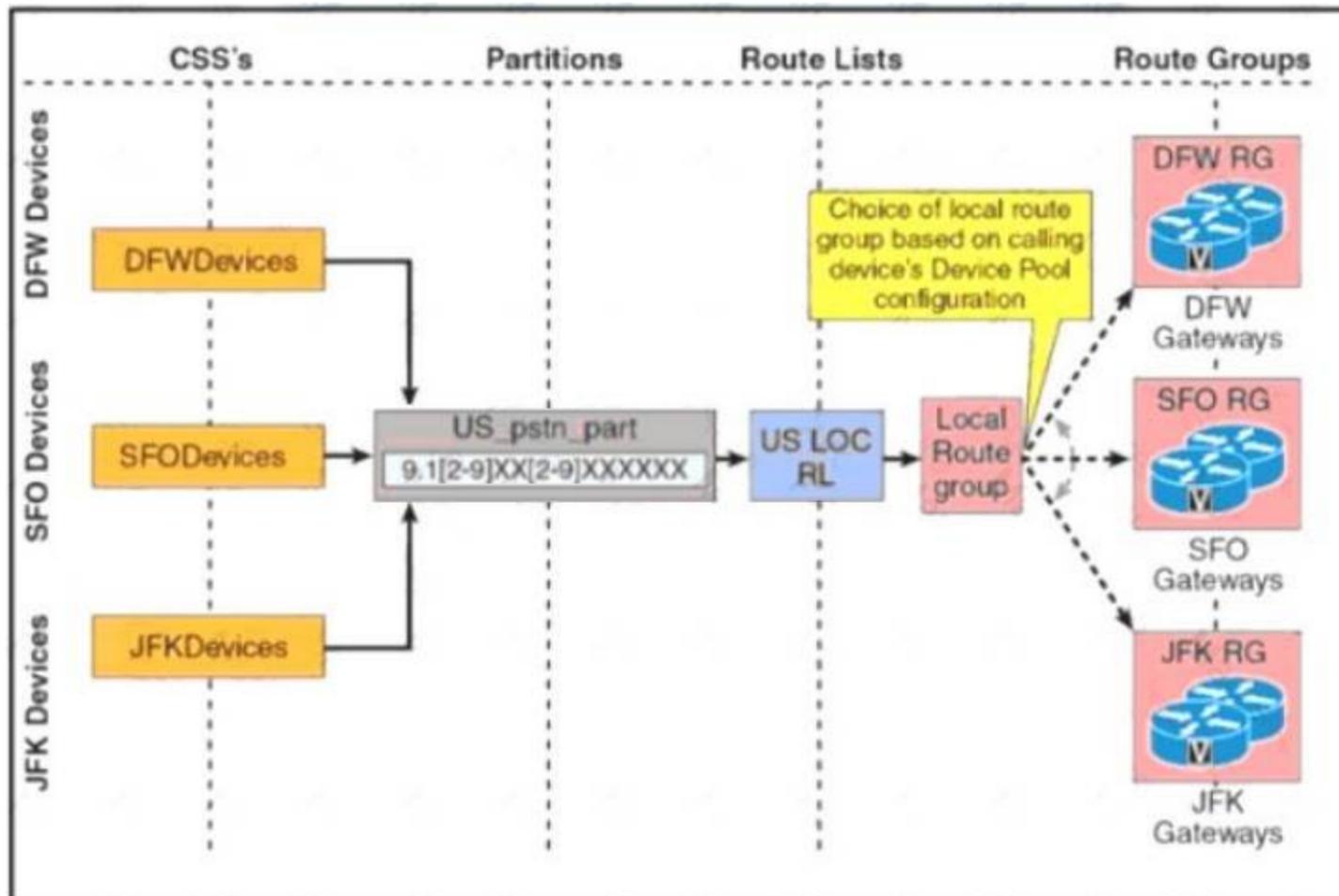
Called Party Number Type*

Called Party Numbering Plan*

Answer: C

NEW QUESTION 155

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 159

An engineer is asked to implement on-net/off-net call classification in Cisco UCM. Which two components are required to implement this configuration? (Choose two.)

- A. CTI route point
- B. SIP route patterns
- C. route group
- D. route pattern
- E. SIP trunk

Answer: DE

NEW QUESTION 162

An engineer is deploying Webex app on Microsoft Windows computers. The engineer wants to ensure that the end users do not receive pop-IQ dialogues when they start the application. Which two actions ensure the end users are not prompted to accept the end-user license (Choose two)

- A. Set the DELETEUSERDATA=r installation argument
- B. Set the "HKEY_LOCAL_MACHINE\Software\WOW6432Node\CiscoCollabHost -Eula_disable
- C. Set the "HKEY_LOCAL_MACHINE\Software\CiscoCollabHost\Eula Setting registry Eula_disable
- D. Set the DEFAULT^THEMES-Dark" installation argument
- E. Set the "/quiet installation argument

Answer: BC

Explanation:

The correct answers are B and C.

To ensure that end users are not prompted to accept the end-user license agreement (EULA) when they start the Webex app, the engineer must set the following two registry keys:

- > HKEY_LOCAL_MACHINE\Software\WOW6432Node\CiscoCollabHost\Eula_disable
- > HKEY_LOCAL_MACHINE\Software\CiscoCollabHost\Eula Setting\Eula_disable

Setting these registry keys will disable the EULA prompt for all users who start the Webex app.

The other options are not valid actions to ensure that end users are not prompted to accept the EULA.

NEW QUESTION 164

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 166**

An engineer is configuring a BOT device for a Jabber user in Cisco Unified Communication Manager. Which phone type must be selected?

- A. Cisco Dual Mode for Android
- B. Cisco Unified Client Services Framework
- C. Cisco Dual Mode for iPhone
- D. third-party SIP device

Answer: A

NEW QUESTION 170

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 175

Refer to the exhibit.

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

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 178

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