

# Cisco

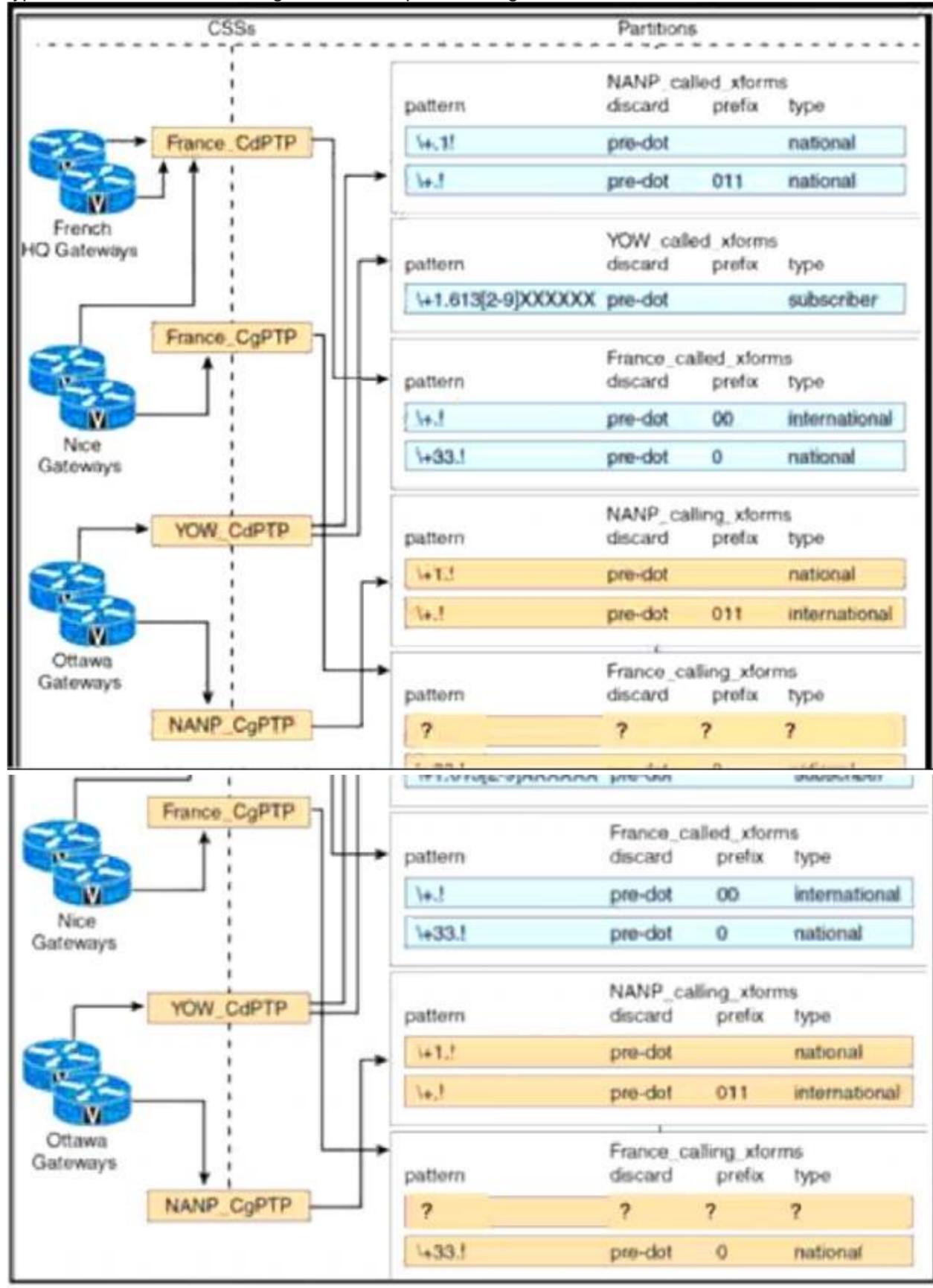
## Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies



**NEW QUESTION 1**

Refer to the exhibit A call from +1 613 555 1234 that is sent out through the Nice Gateways should result in a calling party of 001 613 555 1234 with the numbering type "international" Which configuration accomplishes this goal?



- A. \+.001! pre-dot 1 international
- B. \+1.1 none pre-dot 001 international
- C. \+! pre-dot 00 international
- D. 613XXXXXX none +011 internationa

**Answer: C**

**NEW QUESTION 2**

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 XXXXXXXXXX.
- 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 3**

Refer to the exhibit.

```
C:\Users\CISCO>nslookup
Default Server: dns.example.com
Address: 192.168.100.1

> set type=SRV
> _collab-edge._tcp.example.com
Server: dns.example.com
Address: 192.168.100.1

Non-authoritative answer:
_collab-edge._tcp.example.com      SRV service location:
  priority      = 10
  weight       = 10
  port        = 8443
  svr hostname = expe.example.com
```

You deploy Mobile and Remote Access for Jabber and discover that Jabber for Windows does not register to Cisco Unified Communications Manager while outside of the office. What is a cause of this issue?

- A. The DNS record should be created for \_cisco-uds.\_tcp example.com.
- B. The DNS record should be changed from \_collab-edge.\_tls example.com.
- C. The DNS record type should be changed from SRV to A.
- D. Server 4.2.2.2 is not a valid DNS server.

**Answer: B**

**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**

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 6**

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 7**

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 8**

Refer to the exhibit. Which two codec permutations should be transcoded by this dspfarm? (Choose two.)

- A. iLBC to G.711ulaw
- B. G.728br8 to G.711alaw
- C. G.729r8 to G.711ulaw
- D. G.722 to G.729r8
- E. G.729ar8 to G.711alaw

**Answer: CE**

**NEW QUESTION 9**

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 10**

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 10**

What are two common attributes of XMPP XML stanzas? (Choose two.)

- A. from
- B. to
- C. destination
- D. version
- E. Source

**Answer: AB**

**NEW QUESTION 15**

Refer to the exhibit.

NAME	TTL	CLASS	TYPE	Priority	Weight	Port	Target Address
_sip._tcp.sample.com	86400	IN	SRV	10	60	5060	server1.sample.com
_sip._tcp.sample.com	86400	IN	SRV	10	30	5060	server2.sample.com
_sip._tcp.sample.com	86400	IN	SRV	5	20	5060	server3.sample.com

An administrator must fix the SRV records to ensure that server1. sample.com is always contacted first from the three servers. Which solution should the engineer apply to resolve this issue?

- A. Priority = 100, Weight = 90
- B. Priority = 10, Weight = 5
- C. Priority = 10, Weight = 10
- D. Priority = 5, Weight = 70

**Answer: D**

**NEW QUESTION 16**

If a phone needs to register with cucm1.cisco.com, which network service assists with the phone registration process?

- A. SNMP
- B. ICMP
- C. SMTP
- D. DNS

**Answer: D**

**Explanation:**

According to the Cisco Community website<sup>1</sup>, the phone uses DNS to resolve the hostname of the CUCM server (cucm1.cisco.com) to its IP address. DNS is a network service that translates domain names into IP addresses.

#### NEW QUESTION 18

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 21

When configure Cisco UCM, which configuration enables phones to automatically reregister to a Cisco UCM publisher when the connection to the subscriber is lost?

- A. SRST
- B. Route Group
- C. Cisco UCM
- D. Device Pool

**Answer:** A

#### Explanation:

SRST, or Survivable Remote Site Telephony, is a feature that allows Cisco IP phones to continue to function even when the connection to the Cisco UCM publisher is lost. When SRST is configured, the phones will automatically reregister to the publisher when the connection is restored. Route groups are used to route calls to different destinations based on the caller's phone number or other criteria. Cisco UCM is the call management system that controls the IP phones. Device pools are used to group phones together for administrative purposes.

#### NEW QUESTION 24

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 26

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 28

Which attribute contains an XMPP stanza?

- A. iq
- B. message
- C. type
- D. presence

**Answer:** A

#### NEW QUESTION 32

Which type of input is required when configuring a third-party SIP phone?

- A. digest user
- B. manufacturer
- C. serial number 350-801 2023-4
- D. authorization code

**Answer:** A

#### NEW QUESTION 34

Which Webex Calling construct is used to organize calling features within a physical site?

- A. client settings
- B. locations
- C. service settings
- D. call routing

**Answer:** B

**Explanation:**

A location is a physical site that contains users, devices, and resources. Locations are used to organize calling features within a physical site. For example, you can create a location for each of your offices and then assign users, devices, and resources to that location. This will allow you to manage calling features for each office separately.

**NEW QUESTION 39**

An engineer is going to redesign a network, and while looking at the QoS configuration, the engineer sees that a portion of the network is marked with AF42. Which type of traffic is marked with this tag?

- A. signaling
- B. voice
- C. video conference
- D. streaming video

**Answer:** D

**NEW QUESTION 42**

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 47**

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 52**

A collaboration engineer must configure Cisco Unified Border Element to support up to five concurrent outbound calls across an Ethernet Link with a bandwidth of 160 kb to the Internet Telephony service provider. Which set of commands allows the engineer to complete the task without compromising voice quality?

A)

```
dial-peer voice 1 voip
translation-profile outgoing Strip9
max-conn 5
destination-pattern 91[2-9].[2-9]...$
session protocol sipv2
session target ipv4:142.45.10.1
dtmf-relay rtp-nte sip-notify sip-kpml
codec aacld
```

B)

```
dial-peer voice 1 voip
translation-profile outgoing Strip9
max-conn 5
destination-pattern 91[2-9].[2-9]...$
session protocol sipv2
session target ipv4:142.45.10.1
dtmf-relay rtp-nte sip-notify sip-kpml
codec ilbc mode 20
```

C)

```
dial-peer voice 1 voip
translation-profile outgoing Strip9
max-conn 5
destination-pattern 91[2-9].[2-9]...$
session protocol sipv2
session target ipv4:142.45.10.1
dtmf-relay rtp-nte sip-notify sip-kpml
codec mp4a-latm
```

D)

```
dial-peer voice 1 voip
translation-profile outgoing Strip9
max-conn 5
destination-pattern 91[2-9].[2-9]...$
session protocol sipv2
session target ipv4:142.45.10.1
dtmf-relay rtp-nte sip-notify sip-kpml
```

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

Answer: B

**NEW QUESTION 54**

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 58**

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 60**

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 65**

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 70**

What is the purpose of Mobile and Remote Access (MRA) in the Cisco UCM architecture?

- A. MRA is used to access Webex cloud services only if authenticated with on-premises LDAP service.
- B. MRA is used to make secure PSTN calls by Cisco UCM only while on-premises authentication.
- C. MRA is used to make B2B calls through Expressway registration.
- D. MRA is used to access the collaboration services offered by Cisco UCM from off-premises network connections

**Answer: D**

**NEW QUESTION 72**

Which external DNS SRV record must be present for Mobile and Remote Access?

- A. \_cisco-uds.Jcp.example.com
- B. \_collab-edge.\_tls.example.com
- C. \_collab-edge.\_tcp.example.com
- D. \_cisco-uds.\_tls.example.com

**Answer: B**

**NEW QUESTION 73**

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 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 79**

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 83

A customer enters no IP domain lookup on the Cisco IOS XE gateway to suppress the interpreting of invalid commands as hostnames Which two commands are needed to restore DNS SRV or A record resolutions? (Choose two.)

- A. ip dhcp excluded-address
- B. ip dhcp-sip
- C. ip dhcp pool
- D. transport preferred none
- E. ip domain lookup

Answer: DE

#### NEW QUESTION 86

Users dial a 9 before a 10-digit phone number to make an off-net call All 11 digits are sent to the Cisco Unified Border Element before going out to the PSTN The PSTN provider accepts only 10 digits. Which configuration is needed on the Cisco Unified Border Element for calls to be successful?

- A. voice translation-rule 1 rule 1 /^9/ //
- B. voice translation-rule 1 rule 1 /^9(.....)/ //
- C. voice translation-rule 1 rule 1 /^9.+/ //
- D. voice translation-rule 1 rule 1 /^9...../ //

Answer: A

#### NEW QUESTION 89

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 93**

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 94**

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 96**

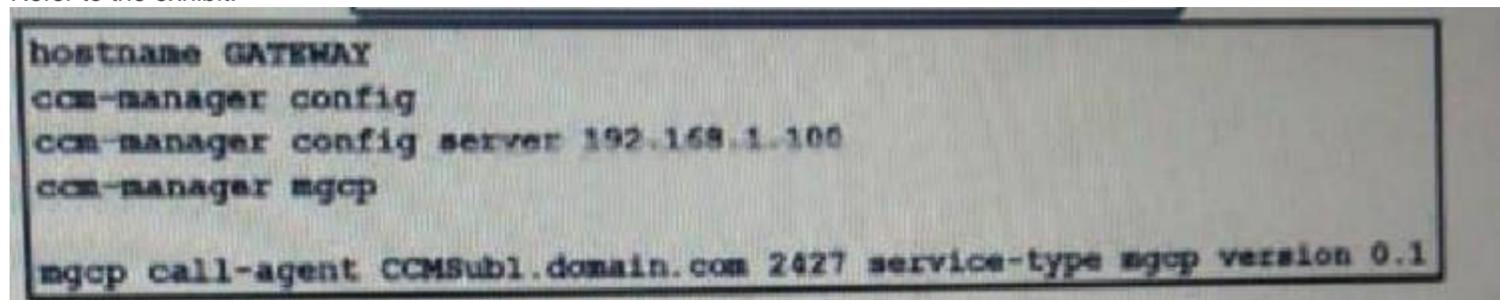
In which location does an administrator look to determine which subscriber the phone registers to if loses registration with the current Cisco UCM subscriber?

- 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 99**

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 104**

An engineer roust configure DTMF relay on a Cisco Unified Border Element by using RFC2833 as the preferred relay mechanism and KPML as the next preferred relay mechanism. The engineer logs in to the CUBE and enters the dial-peer configuration level. Which command should be run at dial-peer configuration level?

- A. dtmf-relay sip-kvmi rtp-nte
- B. dtmf- relay rtp-nte sip-kpml

- C. dtmf-relay sip-kgml rtp-inband
- D. dtmf-relay rtp-inband sip-kvmi

**Answer: B**

**Explanation:**

The dtmf-relay command is used to configure DTMF relay on a Cisco Unified Border Element. The rtp-nre option specifies that RFC2833 is the preferred relay mechanism, and the sip-kgml option specifies that KPML is the next preferred relay mechanism.

**NEW QUESTION 105**

What are two characteristics of jitter in voice and video over IP communications? (Choose two.)

- A. The packets arrive with frame errors.
- B. The packets arrive at varying time intervals.
- C. The packets arrive out of sequence.
- D. The packets never arrive due to tail drop.
- E. The packets arrive at uniform time intervals.

**Answer: BC**

**NEW QUESTION 110**

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 111**

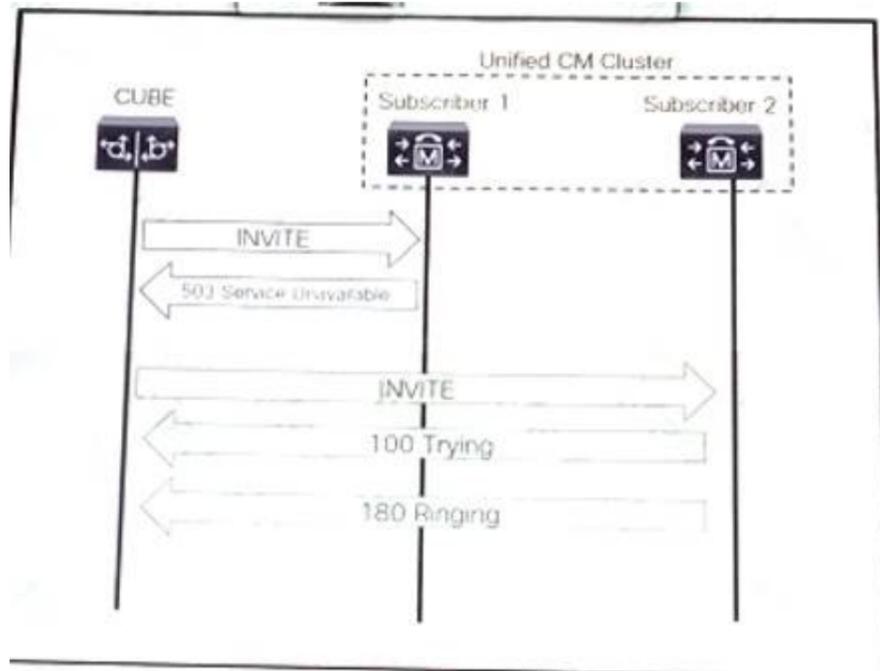
Which task is required when configuring self-provisioning for an end user in Cisco UCM?

- 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 113**

Refer to the exhibit.



Cisco Unified element is attempting to establish a call with Subscribers1, but the call fails. Cisco Unified Border Element then retries the same call with Subscribers2, and the call proceeds normally. Which action resolves the issue?

- A. Verify that the correct calling search space is selected for the inbound Calls section
- B. Verify that the run on all active United CM Nodes checkbox is enabled
- C. Verify that the Significant Digits field for inbound Calls is set to All.
- D. Verify that the PSTN Access checkbox is enabled.

**Answer: B**

**NEW QUESTION 115**

Refer to the exhibit.

```

Bearer Capability i = 0x8090A2
  Standard = CCITT
  Transfer Capability = Speech
  Transfer Mode = Circuit
  Transfer Rate = 64 kbit/s
Channel ID i = 0xA98388
  Exclusive, Channel 8
Calling Party Number i = 0x2181, '5125551212'
  Plan: ISDN, Type: National
Called Party Number i = 0xA1, '2145551212'
  Plan: ISDN, Type: National
Mar 1 02:35:37: ISDN Se0/1/1:23 Q931: RX <- CALL_PROC pd = 8 callref = 0x809A
Channel ID i = 0xA98388
  Exclusive, Channel 8

interface Serial0/1/1:23
description PRI Circuit to R1
no ip address
encapsulation hdlc
isdn switch-type primary-ni
isdn protocol-emulate network
isdn incoming-voice voice
no cdp enable
  
```

An engineer is troubleshooting why PSTN phones are not receiving the caller's name when called from a remote Cisco UCM site. An ISDN PRI connection is being used to reach the PSTN. What must the administrator select to resolve the issue?

- A. isdn supp-service name calling
- B. isdn outgoing display-ie
- C. isdn enable did
- D. isdn send display le

**Answer: B**

**NEW QUESTION 120**

What are two key features of the Expressway series? (Choose two.)

- A. VPN connection toward the internal UC resources
- B. SIP header modification
- C. B2B calls
- D. device registration over the Internet
- E. IP to PSTN call connectivity

**Answer: CD**

**NEW QUESTION 123**

Refer to the exhibit.

```

ROUTER-1(config)# policy-map LLQ_POLICY
ROUTER-1(config-pmap)# class VOICE
ROUTER-1(config-pmap-c)# bandwidth 170
ROUTER-1(config-pmap-c)# exit
ROUTER-1(config-pmap)# class VIDEO
ROUTER-1(config-pmap-c)# bandwidth remaining percent 30
ROUTER-1(config-pmap-c)# exit
ROUTER-1(config-pmap)# exit
  
```

An engineer must modify the existing QoS policy-map statement to implement LLQ for voice traffic. Which change must the engineer make in the configuration?

- A. bandwidth 170 to reserve 170
- B. bandwidth 170 to LL1 170
- C. bandwidth 170 to priority 170
- D. bandwidth 170 to percent 170

**Answer: C**

**NEW QUESTION 127**

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 129

End users report bad video quality and voice choppiness on Cisco Collaboration endpoints. The engineer changed the device pool the users were in but did not correct the problem. Which action should be taken to troubleshoot this issue?

- A. Use direct IP address calls between two endpoints to troubleshoot call quality issues.
- B. Restart the Cisco Location Bandwidth Manager service on the Cisco UCM publisher.
- C. Check for duplex/speed mismatches between the network port settings of the system and network switch.
- D. Set the service parameter Use Video Bandwidth Pool for Immersive Video Calls to "false".

Answer: D

#### NEW QUESTION 134

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 136

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 141

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 146

What is a reason for using a Diffserv value of AF41 for video traffic?

- A. Video traffic cannot tolerate any packet loss and has a latency of 150 milliseconds
- B. Video traffic can tolerate up to 10% packet loss and latency of 10 seconds
- C. Video traffic can tolerate up to 5% packet loss and latency of 5 seconds
- D. Video traffic can tolerate a packet loss of up to 1% and latency of 150 milliseconds

Answer: D

#### NEW QUESTION 148

A company has an excessive number of call transfers to local and long-distance PSTN from Cisco Unity Connection voicemail. Which action in the Cisco Unity Connection restriction table resolves this issue?

- A. Block PSTN patterns on Default Transfe
- B. Default Outdia
- C. and Default System Transfer.
- D. Implement password complexity on voicemail boxes to prevent accounts from being compromised.
- E. Create a custom restriction table ?????????? and block it.
- F. Create a custom restriction table \*\*\*\*\*and block it.

Answer: A

#### NEW QUESTION 151

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 156**

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 157**

An engineer must enable onboarding of on-premises devices by using activation to a Cisco UCM server. The engineer activated the CSCO Device Activation Service and set the default registration method to use the codes. Which action completes the configuration?

- A. Set Enable Activation Code enterprise parameter to True
- B. Manually provision new phones that have an activation code requirement
- C. Create a Bulk Administration Tool provisioning template.
- D. Generate 16-digit codes by using the Bulk Administration Tool

**Answer:** A

**Explanation:**

The engineer must set the Enable Activation Code enterprise parameter to True. This will enable the use of activation codes for onboarding on-premises devices to a Cisco UCM server. The other options are not necessary to complete the configuration.

Here are the steps to complete the configuration:

- > Log in to the Cisco Unified Communications Manager (CUCM) Administration interface.
- > Go to System > Enterprise Parameters.
- > Set the Enable Activation Code enterprise parameter to True.
- > Click Save.

The activation code onboarding feature is now enabled. You can use it to onboard new phones to the CUCM server.

**NEW QUESTION 159**

A Cisco Unity Connection Administrator must set a voice mailbox so that it is accessed from a secondary device. Which configuration on the voice mailbox makes this change?

- A. Attempt Forward routing rule
- B. Mobile User
- C. Alternate Extensions
- D. Alternate Names

**Answer:** C

**NEW QUESTION 160**

An engineer is configuring a Cisco Unified Border Element to allow the video endpoints to negotiate without the Cisco Unified Border Element interfering in the process. What should the engineer configure on the Cisco Unified Border Element to support this process?

- A. Configure path-thru content sdp on the voice service.
- B. Configure a hardcoded codec on the dial peers.
- C. Configure a transcoder for video protocols.
- D. Configure codec transparent on the dial peers.

**Answer:** D

**NEW QUESTION 163**

What is the default TCP port for SIP OAuth mode in Cisco UCM?

- A. 5011
- B. 3174
- C. 8443
- D. 5090

**Answer:** D

**Explanation:**

The Cisco Unified Communications Manager (CUCM) uses SIP Phone OAuth Port (5090) to listen for SIP line registration from Jabber OnPremise devices over

TLS. However, CUCM uses SIP Mobile Remote Access Port (default 5091) to listen for SIP line registrations from Jabber over Expressway through mTLS. Both of these ports are configurable.

**NEW QUESTION 165**

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 169**

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 170**

An administrator installed a Cisco Unified IP 8831 Conference Phone that is failing to register. Which two actions should be taken to troubleshoot the problem? (Choose two.)

- A. Verify that the switch port of the phone is enabled.
- B. Verify that the RJ-11 cable is plugged into the PC port.
- C. Disable HSRP on the access layer switch.
- D. Check the RJ-65 cable.
- E. Verify that the phone's network can access the option 150 server.

Answer: AE

**NEW QUESTION 174**

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 179**

Refer to the Exhibit.

```
dspfarm profile 1 mtp
codec g711ulaw
maximum sessions software 50
associate application SCCP
```

Which command is required to allow this media resource to handle Video Media streams?

- A. maximum sessions hardware 50
- B. video codec h264
- C. codec pass-through
- D. associate application Cisco unified border element

Answer: C

**NEW QUESTION 181**

Refer to the exhibit.

```
Endpoint A:
m=audio 21796 RTP/AVP 108 9 104 105 101
b=TIAS:64000
a=extmap:14 http://protocols.cisco.com/timestamp#100us
a=rtpmap:108 MP4A-LATM/90000
a=fmtp:108 bitrate=64000;profile-level-id=24;object=23
a=rtpmap:9 G722/8000
a=rtpmap:104 G7221/16000
a=fmtp:104 bitrate=32000
a=rtpmap:105 G7221/16000
a=fmtp:105 bitrate=24000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=trafficclass:conversational.audio.immersive.aq:admitted

Endpoint B:
m=audio 21796 RTP/AVP 105 0 8 18 101
b=TIAS:64000
a=extmap:14 http://protocols.cisco.com/timestamp#100us
a=rtpmap:105 G7221/16000
a=fmtp:105 bitrate=24000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=trafficclass:conversational.audio.immersive.aq:admitted
```

Endpoint A calls endpoint B. What is the only audio codec that is used for the call?

- A. G722/8000

- B. Telephone-event/8000
- C. G7221/16000
- D. PCMA/8000

Answer: C

**NEW QUESTION 182**

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 187**

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 190**

Which two parameters influence the total number of supported conference participants on a Cisco IOS XE router that has DSP modules? (Choose two.)

- A. voice codec
- B. session capacity of the PVDM module
- C. number of protocol data units
- D. software version Of the router
- E. license types

Answer: AB

**Explanation:**

These are two parameters that influence the total number of supported conference participants on a Cisco IOS XE router that has DSP modules. A voice codec is a software algorithm that compresses and decompresses voice signals for transmission over a network. Different voice codecs have different bandwidth requirements and quality levels. A PVDM module is a hardware component that provides digital signal processing (DSP) resources for voice applications such as conferencing and transcoding. A PVDM module has a fixed session capacity, which is the maximum number of voice channels that it can support simultaneously.

**NEW QUESTION 194**

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:1, s:on, t:10, 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:1, s:on, t:10, 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:1, s:on, t:10, 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:1, s:on, t:10, 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:1, s:on, t:10, 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:1, s:on, t:10, 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:1, s:on, t:10, 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 195**

Refer to the exhibit.



This INVITE is sent to an endpoint that only supports G.729. What must be done for this call to succeed?

- A. Add a transcoder that supports G.711ulaw and G.729.
- B. Nothing; both sides support G.729.
- C. Add a media termination point that supports G.711ulaw and G.729.
- D. Nothing both sides support payload type 101.

**Answer:** A

#### **NEW QUESTION 203**

An engineer is deploying the Webex app on a Microsoft Windows computer that has multiple user accounts. Which CLI argument allows the engineer to install the application as a "per machine" installation?

- A. ACCEPT\_EULA
- B. INSTALL\_ROOT
- C. ALLUSERS
- D. FORCELOCKDOWN

**Answer:** C

#### **Explanation:**

The ALLUSERS property is a command-line argument that can be used to install the Webex app as a "per machine" installation. This means that the app will be installed for all users on the computer, regardless of which user account is currently logged in.

The ACCEPT\_EULA property is a command-line argument that can be used to accept the end-user license agreement (EULA) for the Webex app. The INSTALL\_ROOT property is a command-line argument that can be used to specify the installation directory for the Webex app. The FORCELOCKDOWN property is a command-line argument that can be used to prevent users from uninstalling the Webex app.

#### **NEW QUESTION 207**

Which two devices are supported by the Flexible DSCP Marking and Video Promotion feature? (Choose two.)

- A. MGCP devices
- B. SCCP devices
- C. pass-through MTPs
- D. H.323 trunks
- E. DX80

**Answer:** BC

#### **NEW QUESTION 209**

A customer wants a video conference with five Cisco Telepresence 1X5000 Series systems. Which media resource is necessary in the design to fully utilize the immersive functions?

- A. Cisco Webex Meetings Server
- B. software conference bridge on Cisco UCM
- C. Cisco Meeting Server
- D. Cisco PVDM4-128

**Answer:** C

#### **NEW QUESTION 212**

A collaboration engineer is configuring the QoS trust boundary for Cisco UCM voice and video conferencing. Which two trust boundary configurations are valid? (choose two)

- A. QoS trust boundaries include all the devices directly attached to the access switch ports
- B. QoS trust boundaries can be extended to Jabber running on a PC
- C. QoS trust boundaries exclude Jabber softphone running on a PC
- D. QoS trust boundaries can be extended to voice and video devices if the connected PCs are included
- E. QoS trust boundaries can be extended to voice and video devices exclusively

**Answer:** CD

#### **NEW QUESTION 213**

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 216**

How is bandwidth allocated to traffic flows in a flow-based WFQ solution?

- A. All the bandwidth is divided based on the QoS marking of the packets.
- B. Each type of traffic flow has equal bandwidth.

- C. Bandwidth is divided among traffic flow
- D. Voice has priority.
- E. Voice has priority and the other types of traffic share the remaining bandwidth.

**Answer:** D

**Explanation:**

In a flow-based WFQ solution, bandwidth is allocated to traffic flows based on the following criteria:

- The priority of the traffic flow
- The amount of bandwidth that is available
- The number of traffic flows that are competing for bandwidth

Voice traffic is typically given a higher priority than other types of traffic, such as data traffic. This is because voice traffic is more sensitive to latency and jitter than data traffic.

When there is not enough bandwidth to accommodate all of the traffic flows, the WFQ algorithm will prioritize the traffic flows based on their priority. The traffic flows with the highest priority will be given the most bandwidth, and the traffic flows with the lowest priority will be given the least bandwidth.

If there is still not enough bandwidth to accommodate all of the traffic flows, the WFQ algorithm will start to drop packets. The packets that are dropped will be the packets from the traffic flows with the lowest priority.

**NEW QUESTION 217**

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