Cisco

Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

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NEW QUESTION 1
Refer to the exhibit.

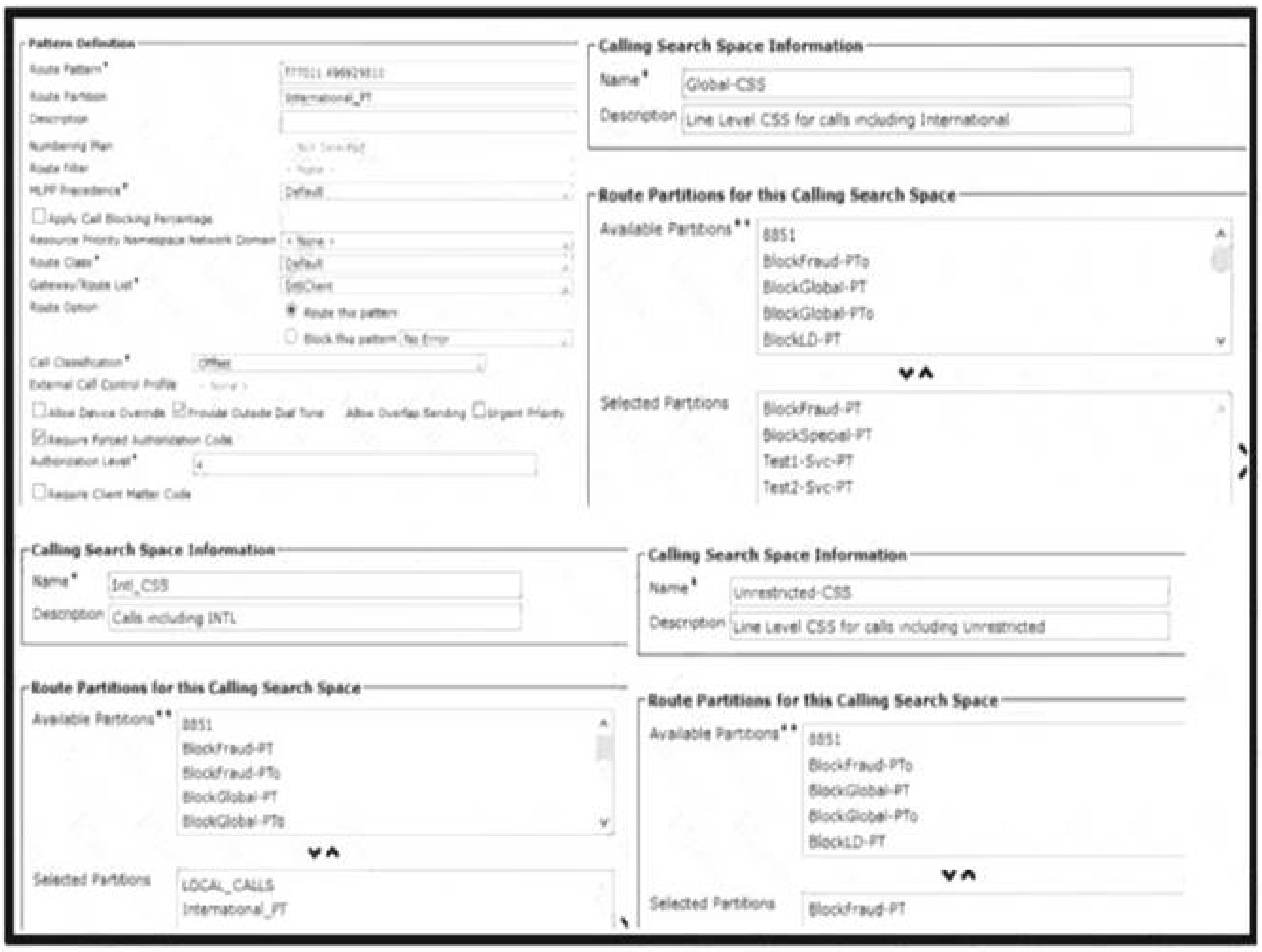
```
Gateway1#show sccp
SCCP Admin State: UP
Gateway Local Interface: Loopback0
        IPv4 Address: 192.168.12.1
        Port Number: 2000

Gateway1#
Gateway1#show ccm-manager
% Call Manager Application is not enabled
Gateway1#

Gateway1#show mgcp
MGCP Admin State DOWN, Oper State DOWN - Cause Code NONE
MGCP call-agent: none Initial protocol service is MGCP 0.1
MGCP validate call-agent source-ipaddr DISABLED
MGCP validate domain name DISABLED
MGCP block-newcalls DISABLED
MGCP send SGCP RSIP: forced/restart/graceful/disconnected DISABLED
```

A collaboration engineer adds an analog gateway to a Cisco UCM cluster. The engineer chooses MGCP over SCCP as the gateway protocol. Which two actions ensure that the gateway registers? (Choose two.)

A. Enter "no seep" on the gateway in configuration mode.
B. Enter "ccm-manager mgcp" on the gateway in configuration mode.
C. Enter "mgcp" on the gateway in configuration mode.
D. Enter "ccm-manager config" on the gateway in configuration mode.
E. Delete and re-add the gateway configuration in Cisco UCM.

Answer: BC

NEW QUESTION 2
Refer to the exhibit.



How must the +E.164 translation pattern be configured to reach international number 496929810?

Pattern= \+.496929810. CSS=Unrestricted-CSS. PreDot. Prefix=777011

Pattern= \+.777011496929810. CSS=Intl_CSS

Pattern= \+.011496929810. CSS=Global-CSS. PreDot. Prefix=777

Pattern= \+.496929810. CSS=Intl_CSS. PreDot. Prefix=777011

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

Answer: C

NEW QUESTION 3
What are two QoS requirements for VoIP traffic?

A. Voice traffic must be marked "to DSCP EF.
B. Loss must be no more man 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 4
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

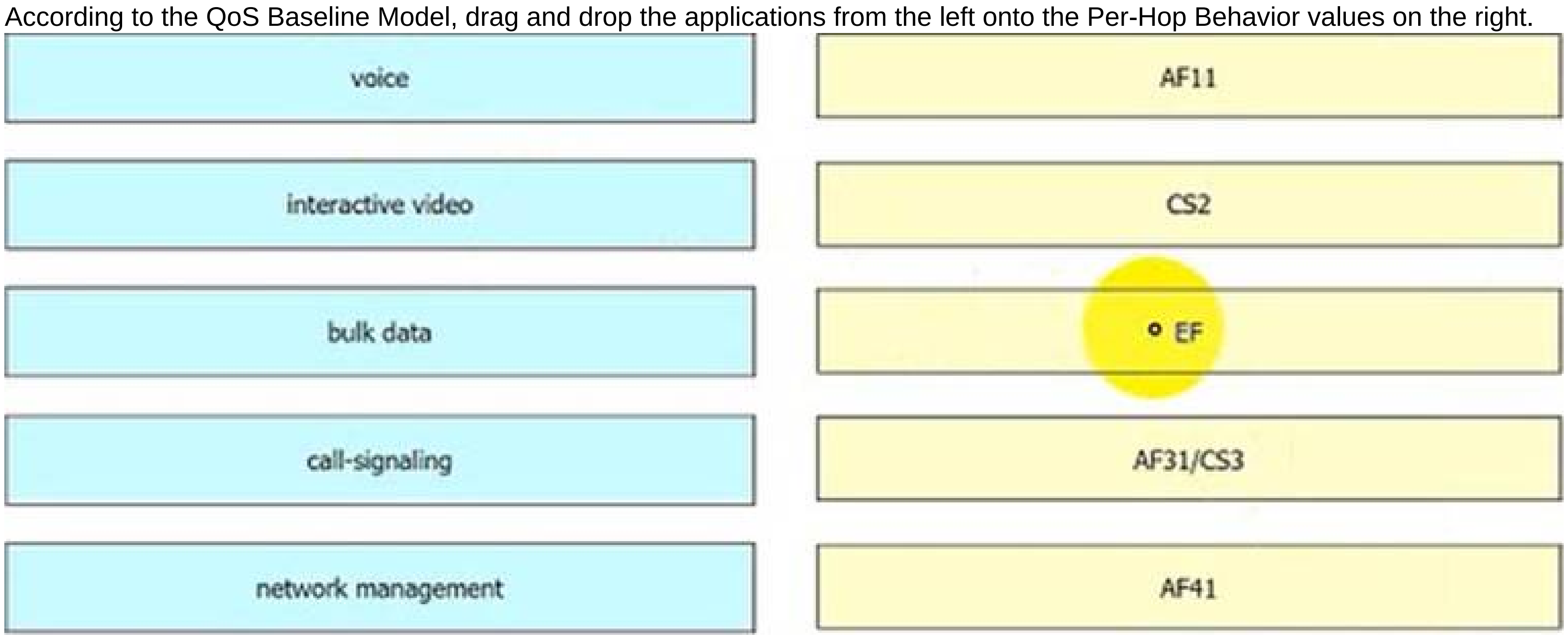
Answer: C

NEW QUESTION 5
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 6
According to the QoS Baseline Model, drag and drop the applications from the left onto the Per-Hop Behavior values on the right.



A. Mastered
B. Not Mastered

Answer: A

Explanation:

NEW QUESTION 7
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 8
Which two configuration elements are part of the Cisco UCM toll-fraud prevention?(Choose two.)

A. feature control policy
B. partition
C. SIP trunk security profile
D. SUBSCRIBE Calling Search Space
E. Calling Search Space

Answer: AE

Explanation:
The following are the configuration elements that are part of the Cisco UCM toll-fraud prevention:
- Feature control policy - This policy controls the features that are available to users. For example, you can use this policy to prevent users from making international calls.
- Calling Search Space - This space defines the numbers that users can call. For example, you can use this space to prevent users from calling premium-rate numbers.

NEW QUESTION 9
Which two protocols can be configured for the Cisco Unity Connection and Cisco UCM integration? (Choose two.)

A. 323
B. SIP
C. SCCP
D. MGCP
E. RTP

Answer: BC

Explanation:
The two protocols that can be configured for the Cisco Unity Connection and Cisco UCM integration are SIP and SCCP. SIP, or Session Initiation Protocol, is a signaling protocol used for initiating, maintaining, and terminating real-time sessions, including voice, video, and messaging applications.
SCCP, or Skinny Client Control Protocol, is a Cisco proprietary signaling protocol used for controlling Cisco IP phones.
H.323 is an older signaling protocol that is no longer widely used. MGCP, or Media Gateway Control Protocol, is a protocol used for controlling media gateways.
RTP, or Real-time Transport Protocol, is a protocol used for transporting real-time data, such as voice and video

NEW QUESTION 10
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 Gateway1. A Local Gateway is a supported session border controller that terminates the trunk on the premises2. A certificate-based trunk requires a certificate authority (CA) to issue and manage the certificates for both Webex Calling and the Local Gateway1.

NEW QUESTION 10
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 13
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 17
Callers from a branch report getting busy tones intermittetly when trying to reach colleagues in other office branches during peak hours. An engineer collects Cisco CallManager service traes to examine the situation. The traces show:

```
50805567.000 |07:35:39.676 |Sdl Sig |StationOoutputDisplayNotify |restart0
|StaatinD(1,100,63,6382) |StionCdpc(1,100,64,4725) |1,100,40,6.709919^*^*
|[R:N-H:0,L:0,V:0,Z:0,D:0] TimeOutValue=10 Status=x807 Unicode Status=Locale=1
50805567.001 |07:35:39.676 |AppInfo |StationD: (0006382) DisplayNotify
timeOutValue=10 notify='x807' content='Not Enough Bandwidth' ver=85720014.
```

What should be fixed to resolve the issue?

A. class of service configuration
B. region configuration
C. geolocation configuration
D. codec configuration

Answer: B

NEW QUESTION 21
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 22
Which command must be defined before an administrator changes the linecode value on an ISDN T1 PRI in slot 0/2 on an IOS-XE gateway?

A. isdn incoming-voice voice
B. pri-group timeslots 1-24
C. card type t1 0 2
D. voice-port 0/2/0:23

Answer: C

NEW QUESTION 27
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 31
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 32
Which two features of Cisco Prime Collaboration Assurance require advanced licensing? (Choose two.)

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

Answer: BC

NEW QUESTION 33
During the Cisco IP Phone registration process, the TFTP download (ails. What are two reasons (or this issue? (Choose two.)

A. The DNS server was not specified, which is needed to resolve the DHCP server IP address.
B. Option 100 string was not specified, or an incorrect Option 100 string was specified.
C. The Cisco IP Phone does not know the IP address of the TFTP server.
D. The Cisco IP Phone does not know the IP address of any of the Cisco UCM Subscriber nodes.
E. Option 150 string was not specified, or an incorrect Option 150 string was specified.

Answer: CE

NEW QUESTION 38
What is set when using COS to mark an Ethernet frame?

A. Ipp bits
B. IP ECN bits
C. DCSP bits
D. 802.1 p User Priority bits

Answer: D

Explanation:
When using COS to mark an Ethernet frame, the 802.1 p User Priority bits are set. These bits are used to indicate the priority of the frame. The higher the priority, the more likely the frame is to be transmitted first.

NEW QUESTION 43
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 g7llalaw preference 1 codec a7Hulaw preference 2
B. voice class codec 11j codec <?7iiulaw preferred codec g7iialaw
C. vice class codec 100 codec preference 1 g7llulaw 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 44
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 49
Where in Cisco UCM is restrictions on audio bandwidth configured?

A. location
B. partition
C. region
D. serviceability

Answer: C

NEW QUESTION 50
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-hoo
B. by pressing the digits and entering "#" to process the cal
C. UCM performs a digit-by-digit analysis; off-hoo
D. UCM analyzes all digits as a string.
E. On-hoo
F. no digit analysis is performed; off-hoo
G. UCM requires the '#" to start the digit analysis
H. On-hoo
I. UCM performs a digit-by-digit analysis; off-hoo
J. UCM considers all digits were dialed and does not wait for additional digits.
K. On-hoo
L. UCM considers all digits were dialed and does not wait for additional digits; off-hoo
M. UCM performs a digit-by-digit analysis.

Answer: D

NEW QUESTION 54
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 56
A customer wants to conduct B2B video calls with a partner using on-premises conferencing solution. Which two devices are needed to facilitate this request?

A. Expressway-C
B. Cisco Telepresence Management Suite
C. Expressway-E
D. MGCP gateway
E. Cisco Unified Border Element

Answer: AC

NEW QUESTION 61
Refer to the exhibit.

ip.addr==10.0.101.10

Time	Source	Destination	Info
18.683437	10.117.34.222	10.0.101.10	50310 → 5060 [SYN] Seq=0 Win=64240 Len=0 MSS=1460 WS=256 SACK
18.938881	10.117.34.222	10.0.101.10	50314 → 5060 [SYN] Seq=0 Win=64240 Len=0 MSS=1460 WS=256 SACK
21.686680	10.117.34.222	10.0.101.10	[TCP Retransmission] 50310 → 5060 [SYN] Seq=0 Win=64240 Len=0
21.941993	10.117.34.222	10.0.101.10	[TCP Retransmission] 50314 → 5060 [SYN] Seq=0 Win=64240 Len=0
27.687008	10.117.34.222	10.0.101.10	[TCP Retransmission] 50310 → 5060 [SYN] Seq=0 Win=64240 Len=0
27.942784	10.117.34.222	10.0.101.10	[TCP Retransmission] 50314 → 5060 [SYN] Seq=0 Win=64240 Len=0

An administrator is attempting to register a SIP phone to a Cisco UCM but the registration is failing. The IP address of the SIP Phone is 10.117.34.222 and the IP address of the Cisco UCM is 10.0.101.10. Pings from the SIP phone to the Cisco UCM are successful. What is the cause of this issue and how should it be resolved?

A. An NTP mismatch is preventing the connection of the TCP session between the SIP phone and the Cisco UC
B. The SIP phone and Cisco UCM must be set with identical NTP sources.
C. The certificates on the SIP phone are not trusted by the Cisco UC
D. The SIP phone must generate new certificates.
E. DNS lookup for the Cisco UCM FQDN is failin

F. The SIP phone must be reconfigured with the proper DNS server.
G. An network device is blocking TCP port 5060 from the SIP phone to the Cisco UC
H. This device must be reconfigured to allow traffic from the IP phone.

Answer: D

NEW QUESTION 64
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 68
According to QoS guidelines, what is the packet loss for streaming video?

A. Not more than 8%
B. Not more than 1%
C. Not more than 3%
D. Not more than 5%

Answer: B

NEW QUESTION 70
A collaboration engineer adds a voice gateway to Cisco UCM. The engineer creates a new gateway device in Cisco UCM. selects VG320 as the device type and selects MGCP as the protocol What must be done next to add the gateway to the Cisco UCM database?

A. Select the DTMF relay type for the gateway.
B. Select a device pool for the new gateway.
C. Add the FQDN or hostname of the device.
D. Configure the module in slot 0 of the new gateway.

Answer: C

NEW QUESTION 72
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 73
A company hosts a conference call with no local users. How does the administrator stop the conference from continuing?

A. modifies the Drop Ad Hoc Conference service parameter
B. modifies the Block OffNet to OffNet Transfer service parameter
C. removes the transcoder
D. changes the codecs that are supported on the conference resource

Answer: A

NEW QUESTION 78
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 81
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 Elemen
E. Cisco UCM, and Cisco Video Communication Server

Answer: B

NEW QUESTION 85
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 86
Which behavior occurs when Cisco UCM has a CallManager group that consists of two subscribers?

A. Endpoints attempt to register with the bottom subscriber in the list.
B. Endpoints attempt to register with the top 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: B

NEW QUESTION 90
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 92
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 97
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 99
The chief officer at a company must reduce collaboration infrastructure costs by onbording all on-permises 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 boardng 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 101
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 102
Refer to the exhibit.
https://i.postimg.cc/C57TkczG/image.png

```
v=0
o=Cisco-SIPUA 13439 0 IN IP4 10.10.10.10
s=SIP Call
b=AS:4064
t=0 0
m=audio 0 RTP/AVP 114 9 124 113 115 0 8 116 18 101
c=IN IP4 10.10.10.10
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114 maxplaybackrate=16000;sprop-
maxcapturerate=16000;maxaveragebitrate=64000;stereo=0;sprop-stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0;mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1;mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
```

A call is falling to establish between two SIP Devices The called device answers with these SOP Which SOP parameter causes issue?

A. The calling device did not offer a ptime value
B. The media stream is set to send only
C. The payload for G.711ulaw must be 18.

D. The RTP port is set to 0.

Answer: D

Explanation:
The RTP port is used to send and receive media packets during a call. If the RTP port is set to 0, the called device will not be able to send or receive media packets, and the call will fail.
The other options are not correct because:
- A. The calling device did not offer a ptime value: The ptime value is used to specify the amount of time between each media packet. If the calling device does not offer a ptime value, the called device will use the default value of 20 milliseconds.
- B. The media stream is set to sendonly: The media stream is set to sendonly when the called device is only able to send media packets, and not receive them. This is not a problem, and the call will still succeed.
- C. The payload for G.711ulaw must be 18: The payload for G.711ulaw is the type of media packet that is used. The payload must be set to 18 for G.711ulaw, but this is not a problem, and the call will still succeed.

NEW QUESTION 107
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 110
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 112
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 117
Which version is used to provide encryption for SNMP management traffic in collaboration deployments?

A. SNMPv1
B. SNMPv3
C. SNMPv2
D. SNMPv2c

Answer: B

NEW QUESTION 121
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 123
A Cisco UCM administrator sets up new route patterns to support phones in tour different locations, all with local gateways. The administrator wants to use the same route pattern for all four locations. How must the system be configured to achieve this goal?

A. Use CSS alternate routing rules.
B. Use standard local route groups.
C. Add a CSS to each local gateway.
D. Use transforms in the route groups.

Answer: B

NEW QUESTION 128
What is required for Cisco UCM to accept SIP calls with a URI in the format of 'sip:2001@cucmpub.cisco.com'?

A. Define Cluster Fully Qualified Domain Name under Servers in Cisco UCM.
B. Change the Destination Address to a Fully Qualified Domain Name on the SIP trunk.
C. Define Cluster Fully Qualified Domain Name in Enterprise Parameters.
D. Set the SIPS URI Handling to True in CallManager Service Parameters.

Answer: C

NEW QUESTION 132
Which certificate does the Disaster Recovery System in Cisco UCM use to encrypt its communications?

A. Cisco Tomcat
B. CAPF
C. Cisco CallManager
D. IPsec

Answer: D

NEW QUESTION 135
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 140
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 143
Refer to the exhibit.

```
rule 1 /^\(0[25]..\)\-\(...\)\-\(....$\)/ /\1\2\3/
```

The translation rule is configured on the voice gateway to translate DNIS. What is the outcome if the gateway receives 0255-343-1234 as DNIS?

A. The translation rule is not matched because DNIS does not end with a "$".
B. The translation rule is matched and the translated number is 02553431234.
C. The translation rule is matched and the translated number is 025553431234.
D. The translation rule is not matched because DNIS contains"-".

Answer: B

NEW QUESTION 146
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 151
SIP proxies have operations defined in RFC 3261 and supporting extensions. Though no lETF 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-D.lt 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 155
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 157
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 pag
D. The pattern must look like this:(*@company.com)
E. A regular route pattern with URI feature enable in the configuration pag
F. The pattern must look like this:MATCH(*@company.com)

Answer: C

NEW QUESTION 162
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 163
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 802.1q frames tagged with the voice VLAN ID 221?

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

Answer: D

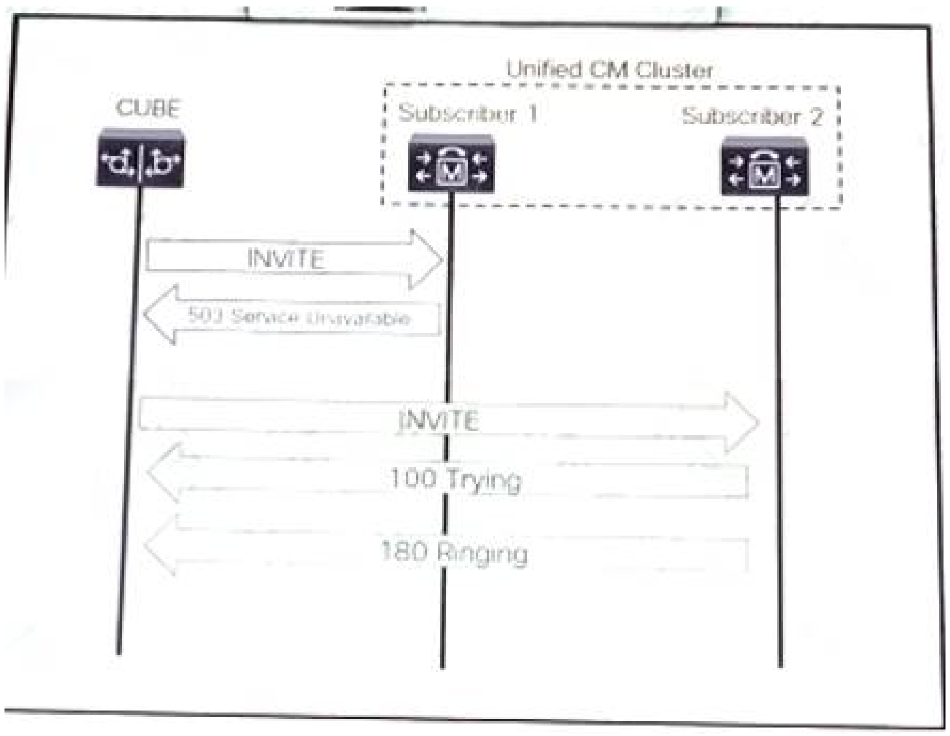
NEW QUESTION 164
What are two access management mechanisms in Cisco Webex Control Hub? (Choose two.)

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 165
Refer to the exhibit.



Cisco Unified element is attempting to establish a call with Subcribers1, but the call fails. Cisco Unified Border Element then retries the same call with Subcribers2, 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 168
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 173
An engineer configures local roule 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 175
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 176
Which characterstic of distributed class- based weighted fair queueing addresses jitter prevention?

A. It provides additional granularity by allowing a user to create classes
B. It minimizes jitter by implementing a priority queue for voice traffic
C. It uses a priority queue for voice traffic to avoid jitter.
D. It provides additional granularity by allowing a user to define custom class

Answer: B

NEW QUESTION 179
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 181
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 184
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 186
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 188
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 190
An end user at a remote site is trying to initiate an Ad Hoc conference call to an end user at the main site. The conference bridge is configured to support G.711. The remote user's phone only supports G.729. The remote end user receives an error message on the phone: "Cannot Complete Conference Call." What is the cause of the issue?

A. The remote phone does not have the conference feature assigned.
B. A software conference bridge is not assigned.
C. A Media Termination Point is missing.
D. The transcoder resource is missing.

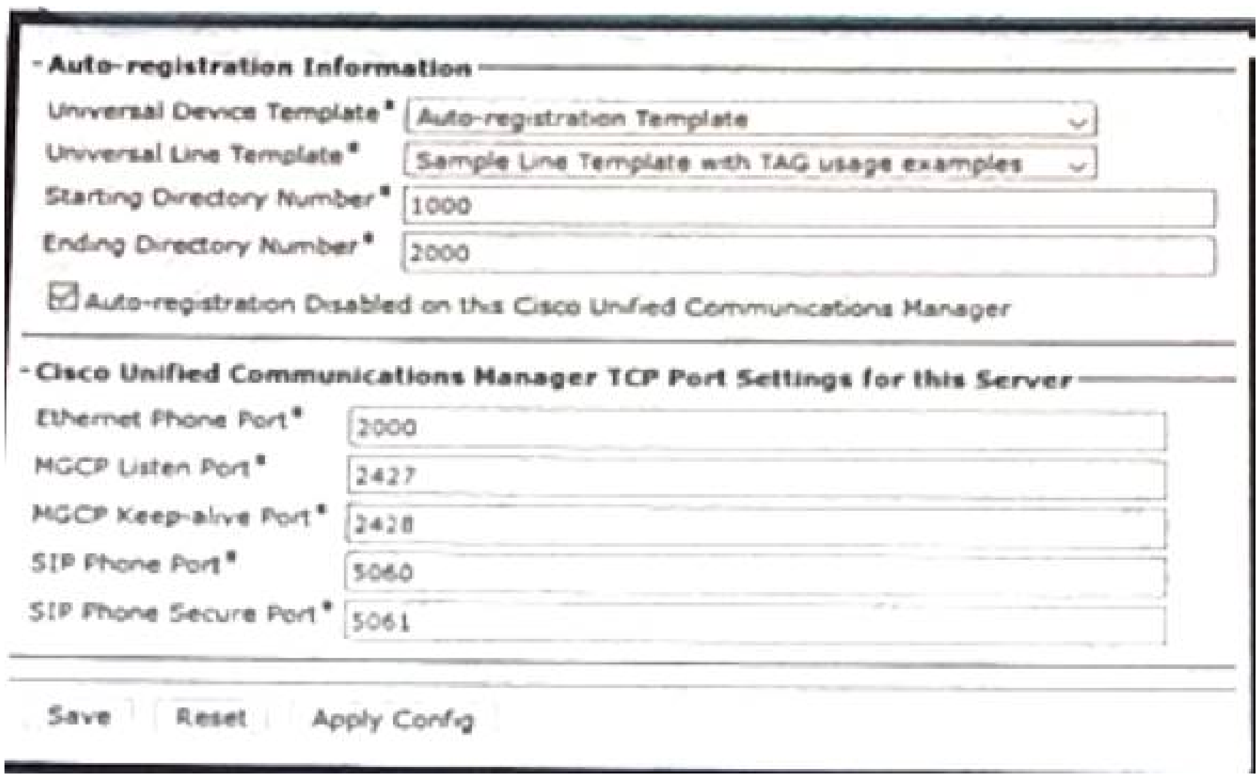
Answer: D

NEW QUESTION 192
Which two recommendations are made to optimize Cisco UCM configuration to reduce the number of toll fraud incidents in an organization? (Choose two.)

A. Classify all route patterns as on-net and prohibit on-net to on-net call transfers in Cisco UCM service parameters.
B. Classify all route patterns as on-net or off-net and prohibit off-net to off-net call transfers in Cisco UCM service parameters.
C. Classify all route patterns as off-net and prohibit off-net to off-net call transfers in Cisco UCM service parameters.
D. Inbound CSS on any gateway typically should have access to internal destinations and PSTN destinations.
E. Inbound CSS on any gateway typically should have access to internal destinations only and not PSTN destinations.

Answer: BE

NEW QUESTION 195
Refer to the exhibit.



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 197
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 202
Refer to the exhibit.

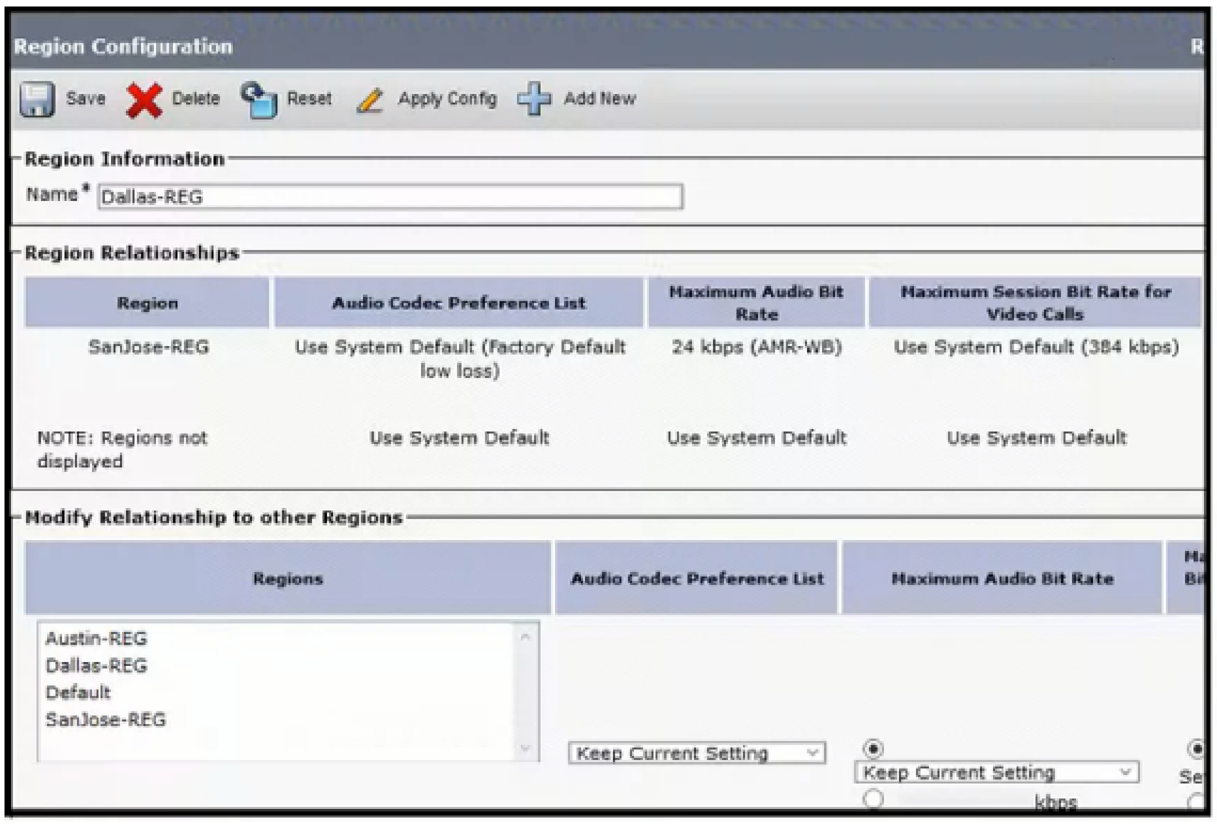
```
isdn switch-type primary-ni
controller t1 0/1/0
framing esf
linecode b8zs
pri-group timeslots 1-10
```

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 205
Refer to the exhibit.



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 209
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 213
Refer to the exhibit.

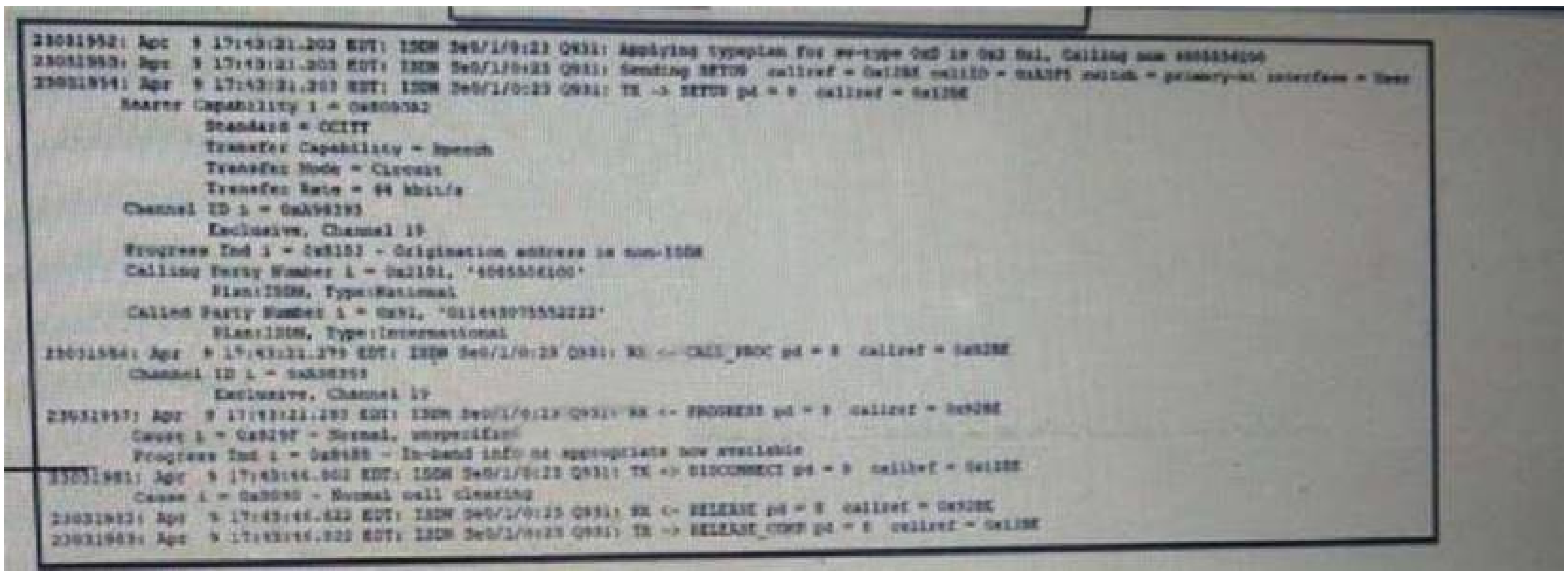
```
!
voice service voip
 ip address trusted list
  ipv4 192.168.100.101
  ipv4 192.168.101.0 255.255.255.128
!
dial-peer voice 1 voip
 destination-pattern +T
 session protocol sipv2
 session target ipv4:192.168.102.102
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
```

When a call is received on Cisco Unified Border Element. from which IP does it allow a connection?

A. 192.168.100.103
B. 192.168.102.102
C. 192.168.100.102
D. 192.168.101.201

Answer: B

NEW QUESTION 215
Refer to the exhibit.



A call to an international number has failed. Which action corrects this problem?

A. Assign a transcoder to the MRGL of the gateway.
B. Strip the leading 011 from the called party number
C. Add the bearer-cap speech command to the voice port.
D. Add the isdn switch-type primart-dms100 command to the serial interface.

Answer: B

NEW QUESTION 217
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 219
Where is urgent priority enabled to bypass the T302 timer?

A. route partition
B. transformation pattern
C. directory number
D. CTI port

Answer: C

Explanation:
Urgent priority is enabled on the directory number configuration page. This allows the call to be routed at once to the fully qualified DN without any necessity to wait for inter-digit-timeout. If the Urgent Priority checkbox is disabled and you have overlap patterns configured, then CUCM waits for the user to dial further digits.
The other options are incorrect because:
- Route partitions are used to group route patterns and route lists.
- Transformation patterns are used to convert dialed digits into a different format.
- CTI ports are used to connect Cisco Unified Communications Manager to third-party applications. https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callman

NEW QUESTION 220
The IP phones al 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(conlicj)# interface GigabitEthernet1/0/1
E. Switch(config># no lldp transmit

Answer: BC

NEW QUESTION 222
Which DSCP marking is represented as 101110 in an IP header?

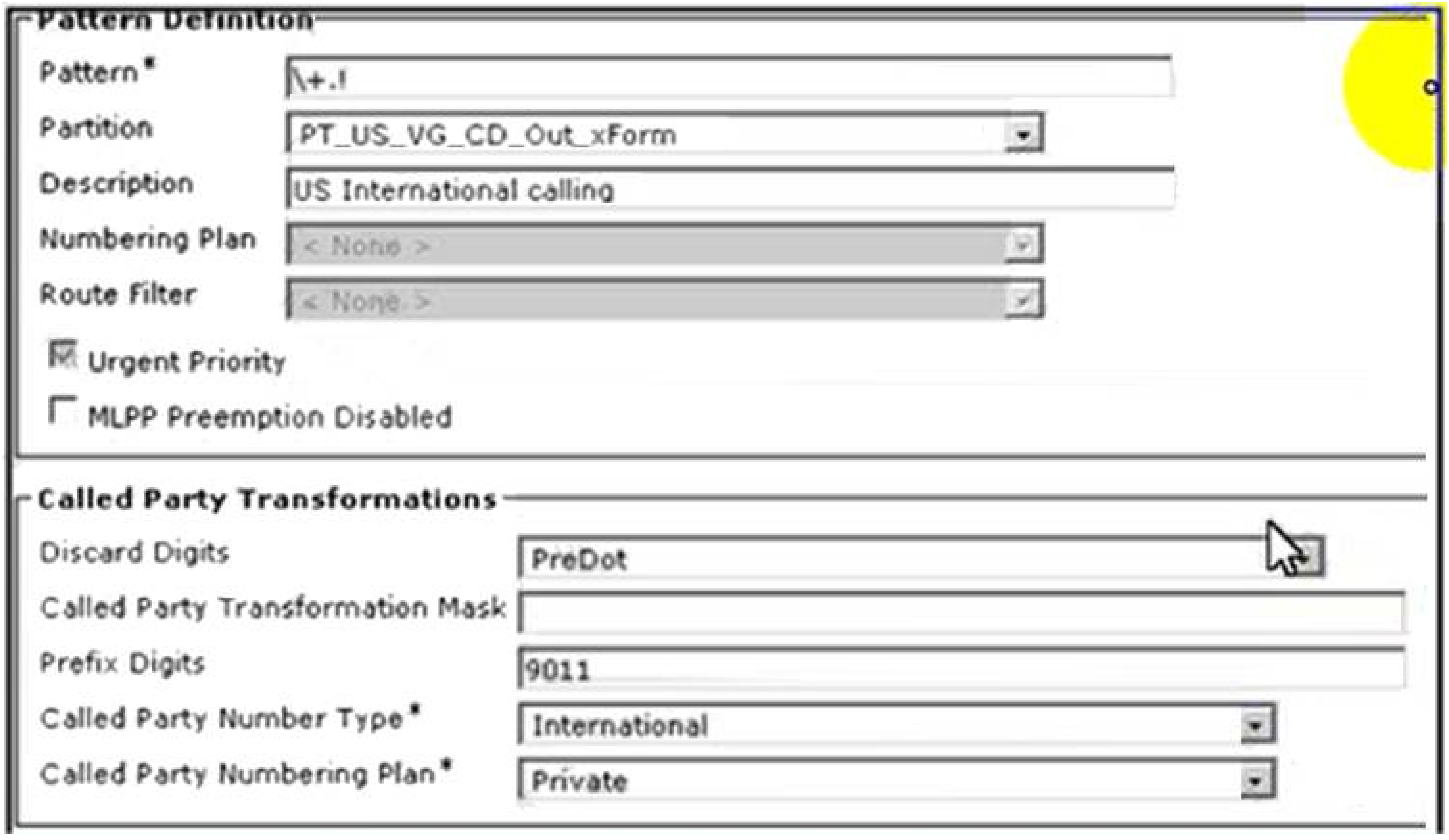
A. EF
B. CS3
C. AF41
D. AF31

Answer: A

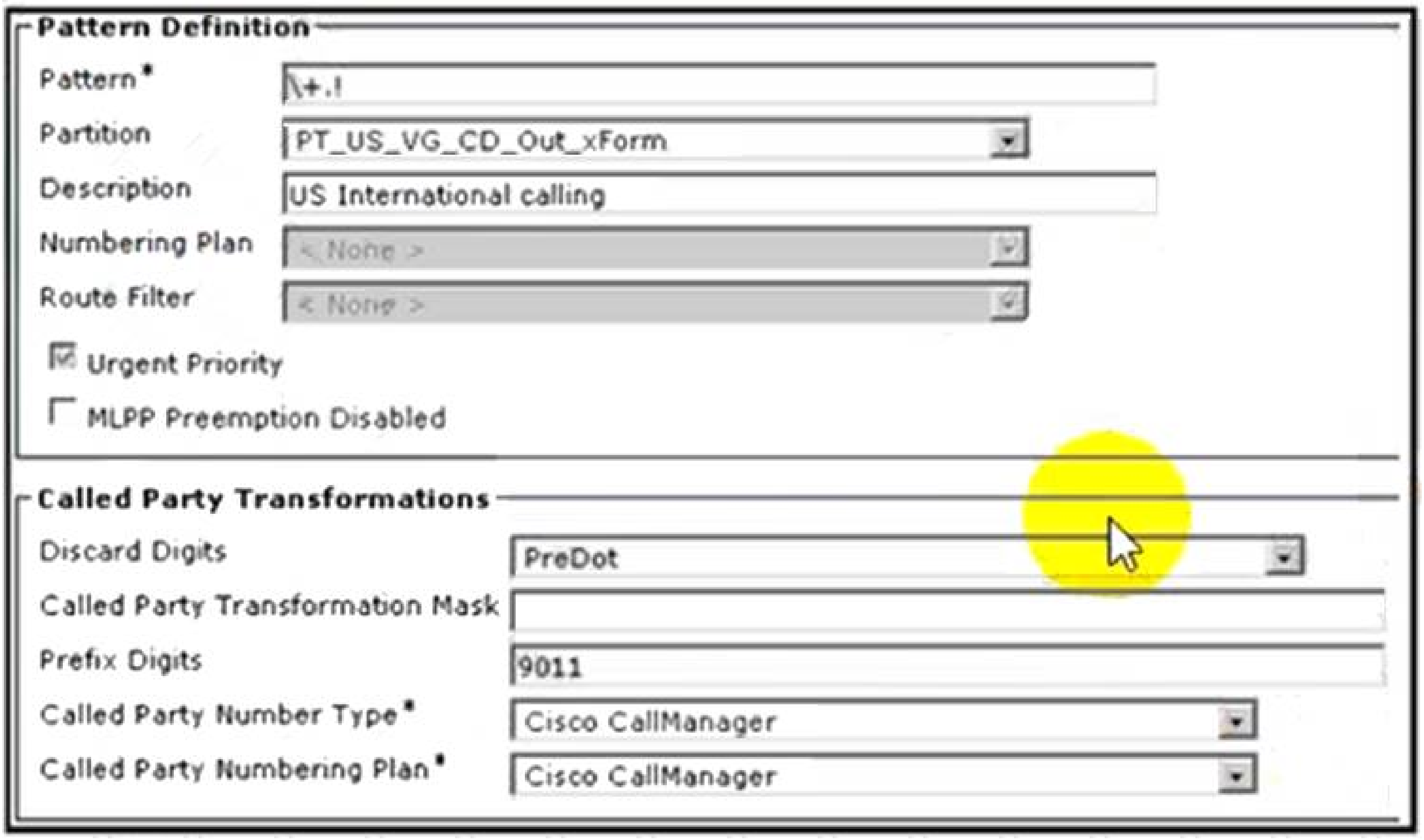
NEW QUESTION 227
Given this H.323 gateway configuration and using Cisco best practices, how must the called party transformation pattern be configured to ensure that a proper ISDN type of number is set?

```
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
```

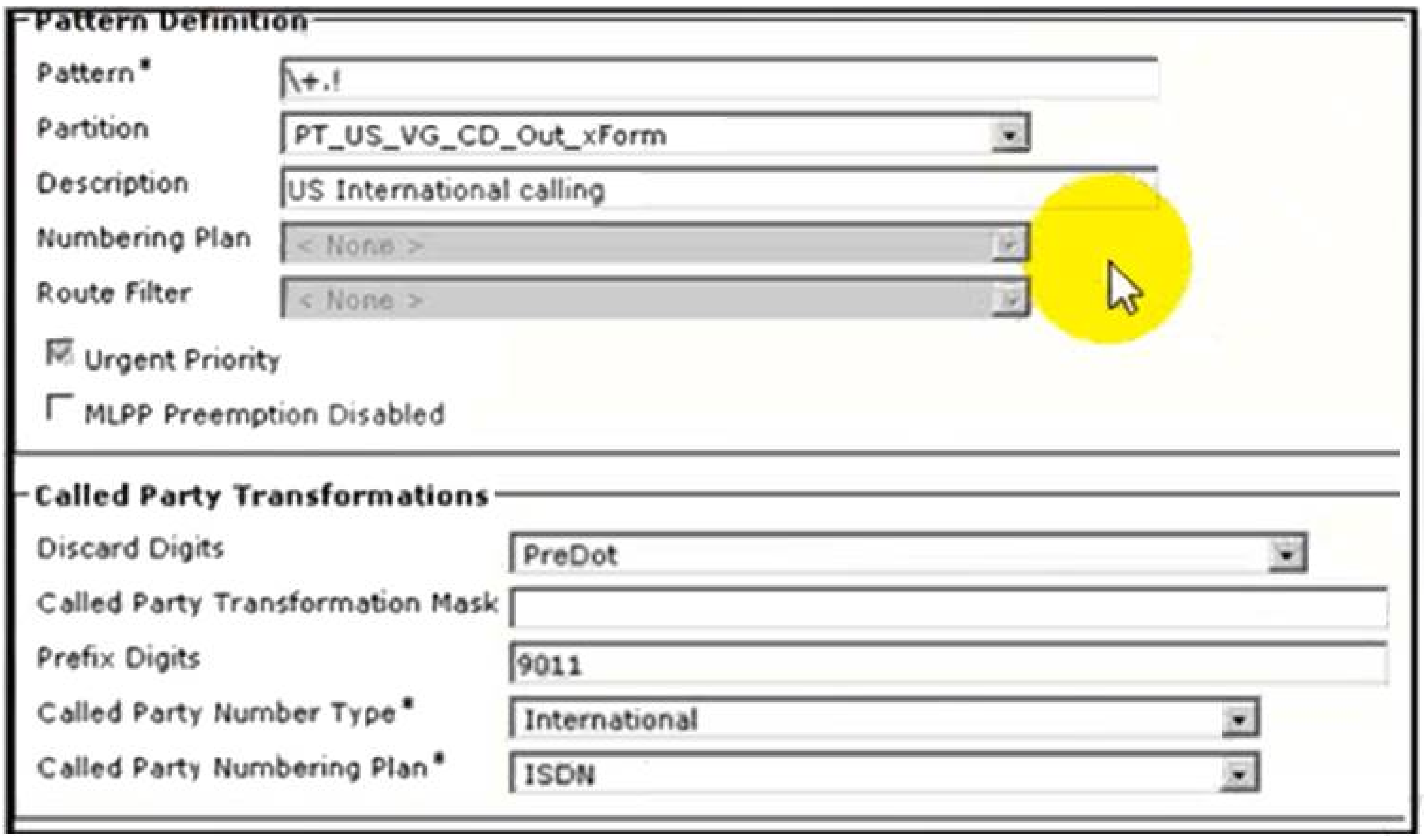
A.


Pattern Definition
Pattern*
\+.!
Partition
PT_US_VG_CD_Out_xForm
Description
US International calling
Numbering Plan
< None >
Route Filter
< None >
Urgent Priority
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

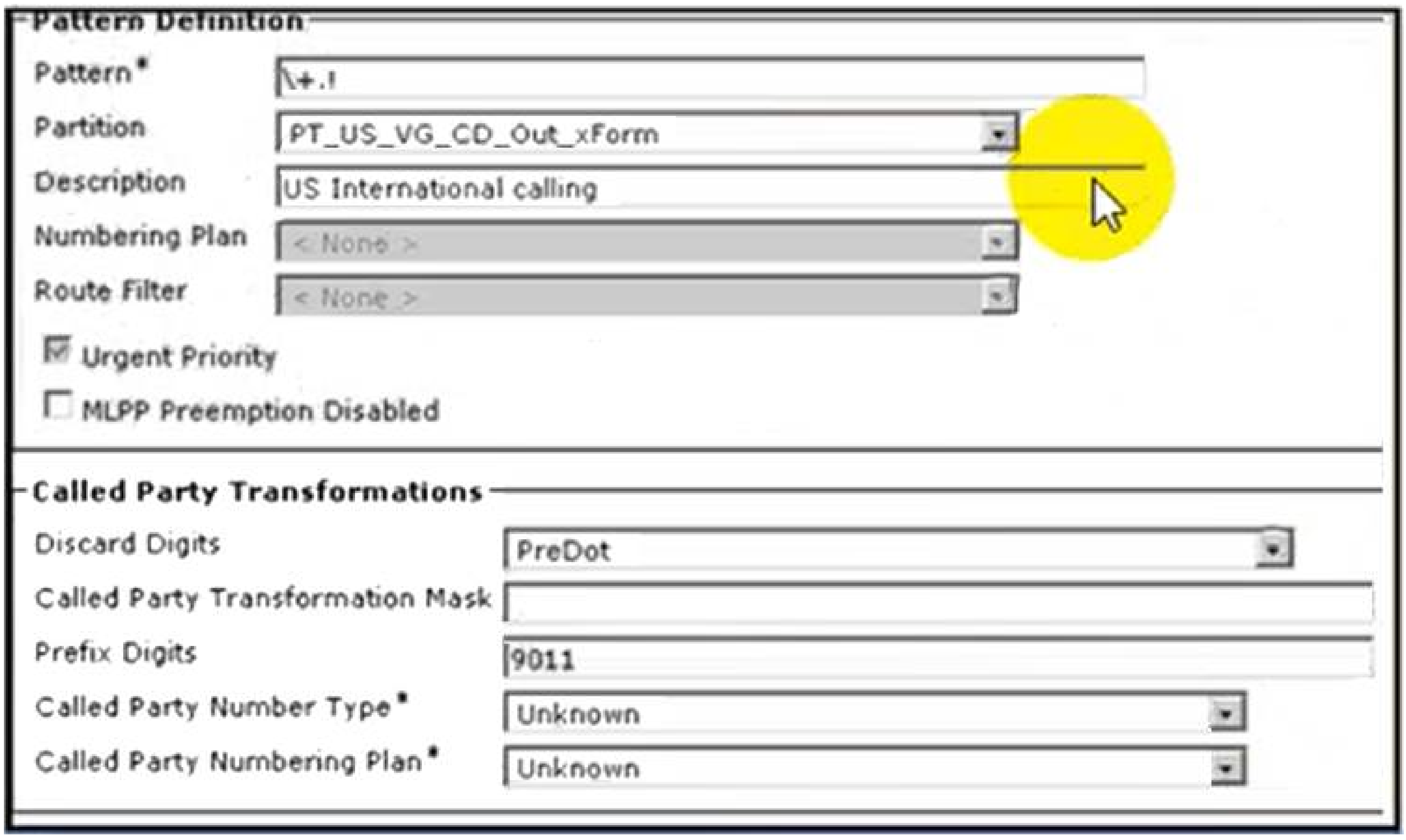
B.


Pattern Definition
Pattern*
\+.!
Partition
PT_US_VG_CD_Out_xForm
Description
US International calling
Numbering Plan
< None >
Route Filter
< None >
Urgent Priority
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 >
Urgent Priority
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.



Answer: C

NEW QUESTION 228
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.

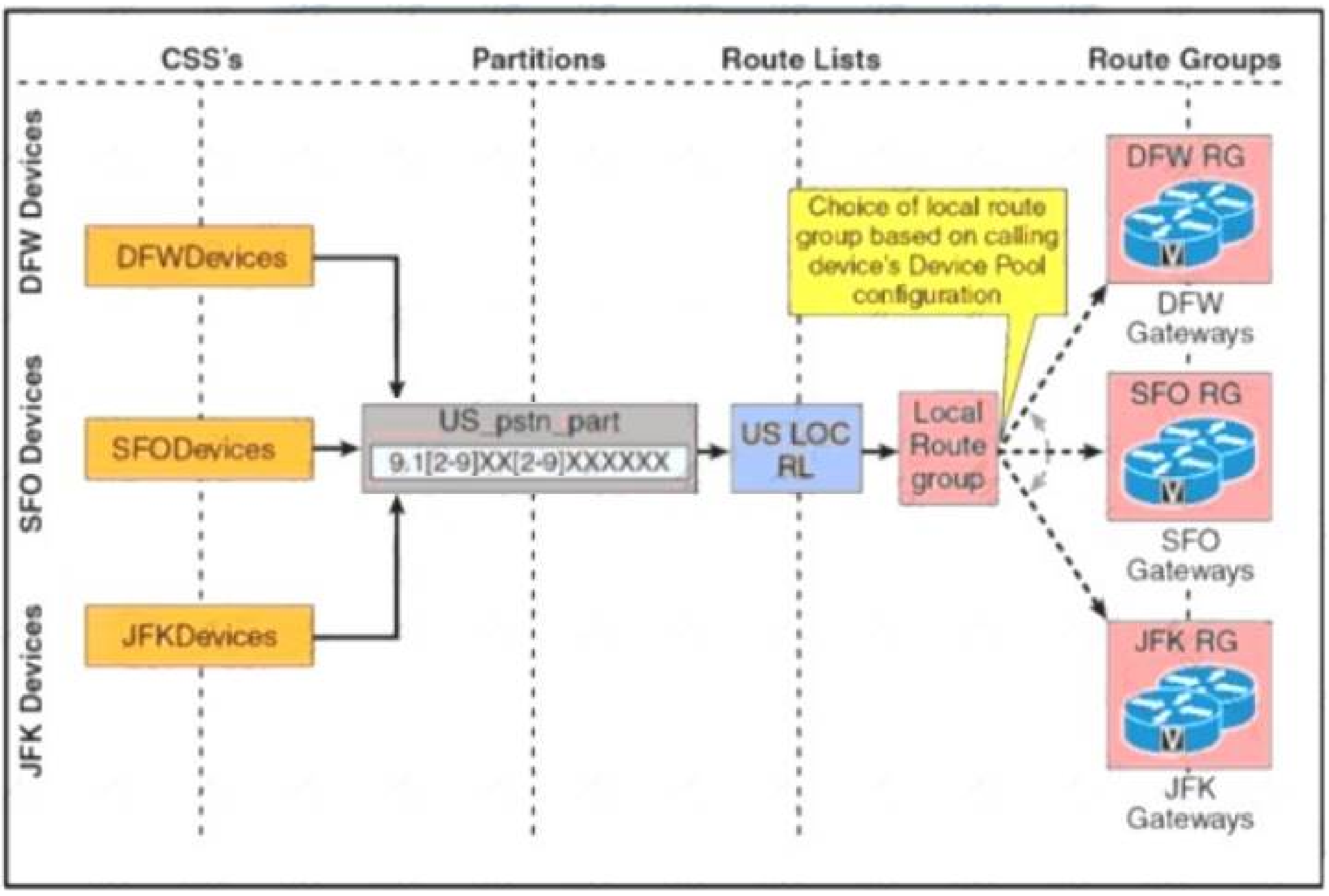
Answer: AE

NEW QUESTION 232
An administrator is configuring LDAP tor 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 235
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 Yor
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 whilestill keeping the number and resources assigned in San Francisco.
D. The call fails because the Standard Local Route Group is being used only it 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 238
What is a possible cause of the PRI issue?

```
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
```

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

Answer: D

NEW QUESTION 242
Refer to the exhibit.

```
Server: Cisco-SIPGateway/IOS-15.4.3.S4
CSeq: 101 OPTIONS
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,
INFO, REGISTER
Allow-Events: telephone-event
Accept: application/sdp
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Content-Type: application/sdp
Content-Length: 369

v=0
o=CiscoSystemsSIP-GW-UserAgent 6414 4717 IN IP4 10.8.140.23
s=SIP Call
c=IN IP4 10.8.140.23
t=0 0
m=audio 0 RTP/AVP 18 0 8 4 15
c=IN IP4 10.8.140.23
m=image 0 udptl t38
c=IN IP4 10.8.140.23
a=T38FaxVersion:0
a=T38MaxBitRate:9600
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:200
a=T38FaxMaxDatagram:320
a=T38FaxUdpEC:t38UDPRedundancy
```

A customer wants the SIP 200 OK shown to advertise codecs in the following order:

- G.729
- G.711u
- G.711a
- G.723
- G.728

After correcting the codec preferences. What should the audio payload show in the SIP Traces?

- m=audio 0 RTP/AVP 0 18 8 4 15
- m=audio 0 RTP/AVP 4 0 8 18 15
- m=audio 0 RTP/AVP 0 8 18 4 15
- m=audio 0 RTP/AVP 18 0 8 4 15

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

Answer: D

NEW QUESTION 244
A Cisco Telepresence SX80 suddenly has issues displaying main video to a display over HDMI. Which command can you use from the SX80 admin CLI to check the video output status to the monitor?

A. xStatus HDMI Output
B. xStatus video Output
C. xconfiguration video Output
D. xcommand video status

Answer: B

NEW QUESTION 245
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

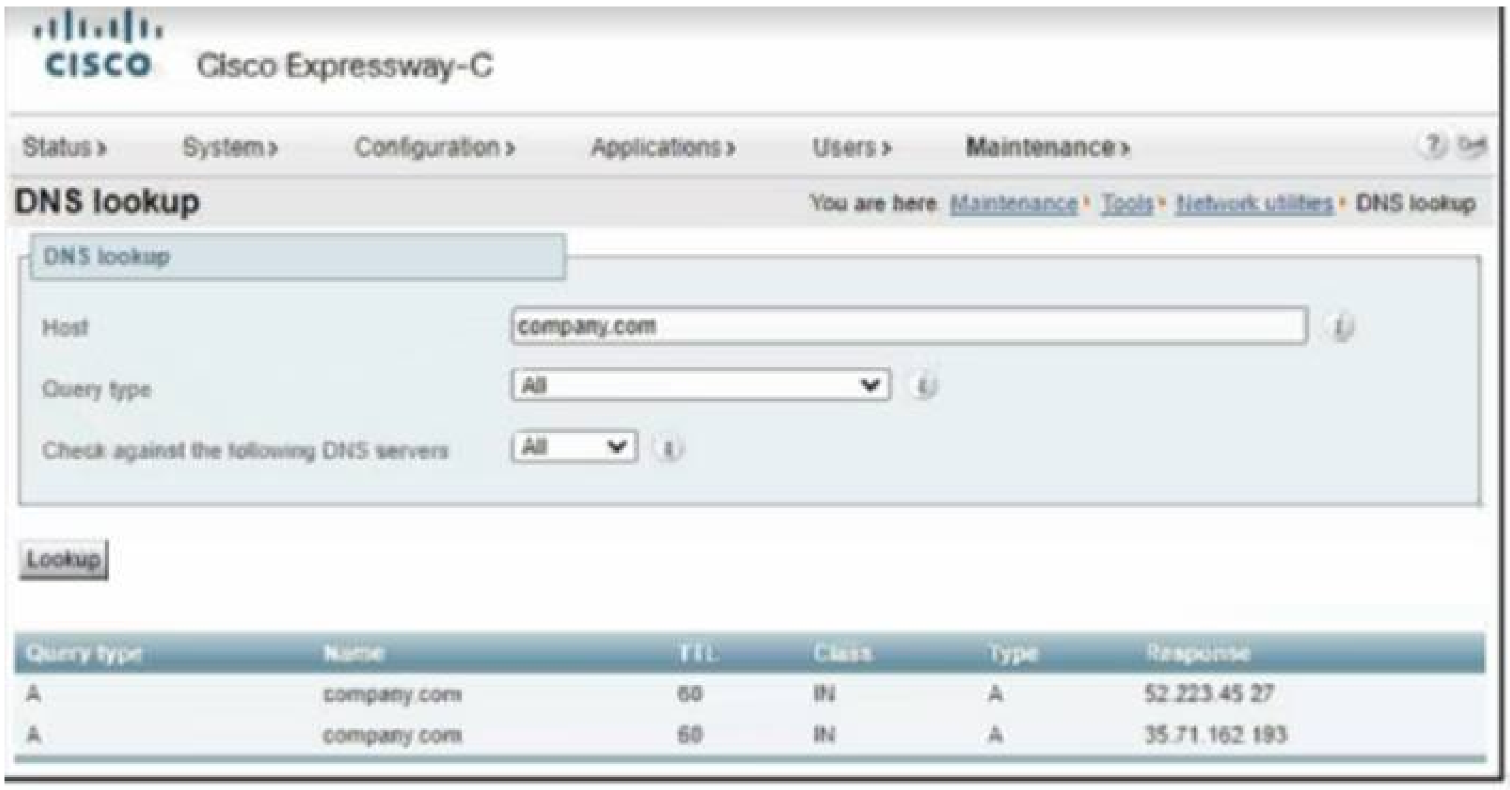
Answer: C

NEW QUESTION 247
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 250
Refer to exhibit.



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 tie What IS me 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 response for the company.com domain

Answer: B

NEW QUESTION 255
What is a capability of the call forwarding feature in a Cisco Webex dial plan?

A. device pool selection
B. Call Admission Control
C. business continuity
D. ringtone selection

Answer: C

Explanation:
Call forwarding is a feature that allows users to forward incoming calls to another number. This can be useful in a number of situations, such as when a user is not available to take a call, or when a user wants to forward calls to a different number during certain times of the day.
Call forwarding can be used to improve business continuity by ensuring that calls are always answered, even if the user is not available. For example, if a user is out of the office, they can forward their calls to their voicemail or to another employee. This ensures that customers and clients can always reach someone, even if the user is not available.

NEW QUESTION 260
Refer to the exhibit.

```
admin:utils ntp status
ntpd (pid 17428) is running...

     remote           refid        st t when poll reach   delay   offset   jitter
==============================================================================
*192.168.1.1     17.253.14.125     2 u   36   64   377    0.435    0.039    0.047
 192.168.1.2     .INIT.           16 u    -   64     0    0.000    0.000    0.000
```

A collaboration engineer adds a redundant NTP server to an existing Cisco Collaboration solution On the Cisco UCM OS Administration page, the new NTP server shows as "Not Accessible" Which action resolves this issue?

A. Restart NTPD on the Cisco UCM server.
B. Delete and re-add the new NTP server via the Cisco UCM command-line interface
C. Start the NTP service on the new NTP server
D. Configure the "reach" value as "377" for the new NTP server.

Answer: C

NEW QUESTION 263
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 265
What are the predefined call handlers in Cisco Unity Connection?

A. opening greeting, welcome, and default system
B. caller input, greetings, and transfer
C. greetings, operator, and closed
D. opening greeting, operator, and goodbye

Answer: D

NEW QUESTION 268
An engineer configures a Cisco Unified Border Element and must ensure that the codecs negotiated meet the ITSP requirements. The ITSP supports G.7llulaw 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 270
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 272
Which action prevents toll fraud in Cisco UCM?

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

Answer: D

NEW QUESTION 277
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 278
User A Calls user. The call gets connected, but the quality is bed. What are two reasons for this issue? (Choose two)

A. Incorrect partition
B. No region relationship
C. Network congestion
D. Incorrect QoS
E. Incompatible codec

Answer: CD

NEW QUESTION 279
Refer to the exhibit.

```
voice translation-rule 1
 rule 1 /^[2-9]......$/ /\0/ type any subscriber
 rule 2 /^1[2-9]..[2-9]......$/ /\0/ type any subscriber
```

What is the result of applying these two rules to a voice translation profile for use with an ISDN T1 PRI on a Cisco Voice Gateway?

A. The leading Plus is stripped from the numeric phone number.
B. The ISDN Plan is modified to the administrator's defined value.
C. Any zero is stripped from the numeric phone number.
D. The ISDN Type is modified to the administrator's defined value.

Answer: D

NEW QUESTION 284
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 289
Refer to the exhibit.

```
INVITE sip:4000@172.16.1.1:5061 SIP/2.0
Via: SIP/2.0/TLS 172.16.2.143:5061;branch=z9hG4bK8FD315E7
Remote-Party-ID: <sip:+14088335000@172.16.2.143>;party=calling;screen=no;privacy=off
From: <sip:+14088335000@172.27.2.143>;tag=7B42E5F6-9B8
To: <sip:4000@172.16.1.1>
Date: Tue, 06 Aug 2019 15:03:05 GMT
Call-ID: 4EA4363-B77111E9-8A4AFFCF-10B6D71B@172.16.2.143
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE:  1800
Cisco-Guid: 0082391505-3077640681-2319777743-0280418075
User-Agent: Cisco-SIPGateway/IOS-15.5.3.S4b
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1565089565
Contact: <sip:+14088335000@172.16.2.143:5061;transport=tls>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 68
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 416
v=0
o=CiscoSystemsSIP-GW-UserAgent 8486 8298 IN IP4 172.16.2.143
s=SIP Call
c=IN IP4 172.16.2.143
t=0 0
m=audio 44612 RTP/SAVP 0 101
c=IN IP4 172.16.2.143
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
```

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 291
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 296
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 300
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 Calling1. A TLS-based trunk uses Transport Layer Security (TLS) to secure the SIP signaling between the hybrid local gateway and Webex Calling1. A registration-based trunk uses SIP registration to authenticate the hybrid local gateway with Webex Calling and receive calls from the cloud1.

NEW QUESTION 301
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 305
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 309
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 313
A Cisco IP Phone 7841 that is registered to a Cisco Unified Communications Manager with default configuration receives a call setup message. Which codec is negotiated when the SDP offer includes this line of text?
M=audio 498181 RTP/AVP 0 8 97

A. G.711ulaw
B. iLBC
C. G.711alaw
D. G.722

Answer: A

Explanation:
The SDP offer includes the following line of text: M=audio 498181 RTP/AVP 0 8 97
This line of text indicates that the following codecs are available:
- 0: G.711ulaw
- 8: G.711alaw
- 97: iLBC

The Cisco IP Phone 7841 is registered to a Cisco Unified Communications Manager with default configuration. This means that the phone will negotiate the G.711ulaw codec.
The G.711ulaw codec is a standard codec that is used for voice communication. It is a low-bandwidth codec that provides good quality.
The iLBC codec is a newer codec that is designed for use in low-bandwidth environments. It provides good quality, but it is not as widely supported as the G.711ulaw codec.
The G.722 codec is a high-quality codec that is used for voice communication. It provides excellent quality, but it requires more bandwidth than the G.711ulaw codec.

NEW QUESTION 315
An administrator would like to set several Cisco Jabber configuration parameters to only apply to mobile clients (iOS and Android). How does the administrator accomplish this with Cisco Jabber 12.9 and Cisco UCM 12.5?

A. Assign the desired configuration file to "Mobile" Jabber Client Configuration in the Service Profile.
B. Upload the jabber-config.enc file to TFTP
C. Create a user profile in Jabber Policies.
D. Deploy jabber-config-user.xml on iOS and Android devices.

Answer: A

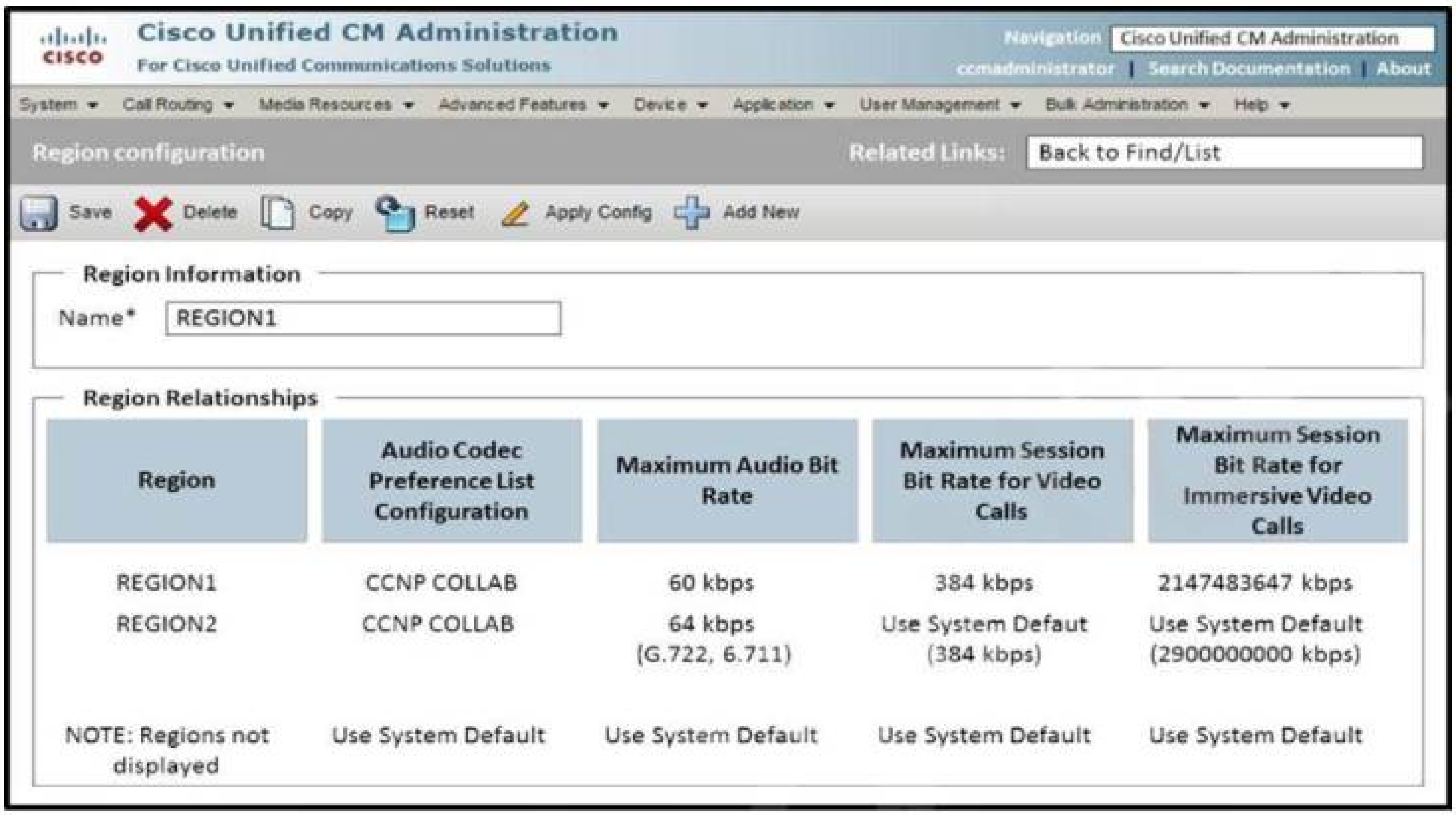
NEW QUESTION 318
An engineer must configure switch port 5/1 to send CDP packets to configure an attached Cisco IP phone to trust tagged traffic on it's access port. Which command is required to complete the configuration?

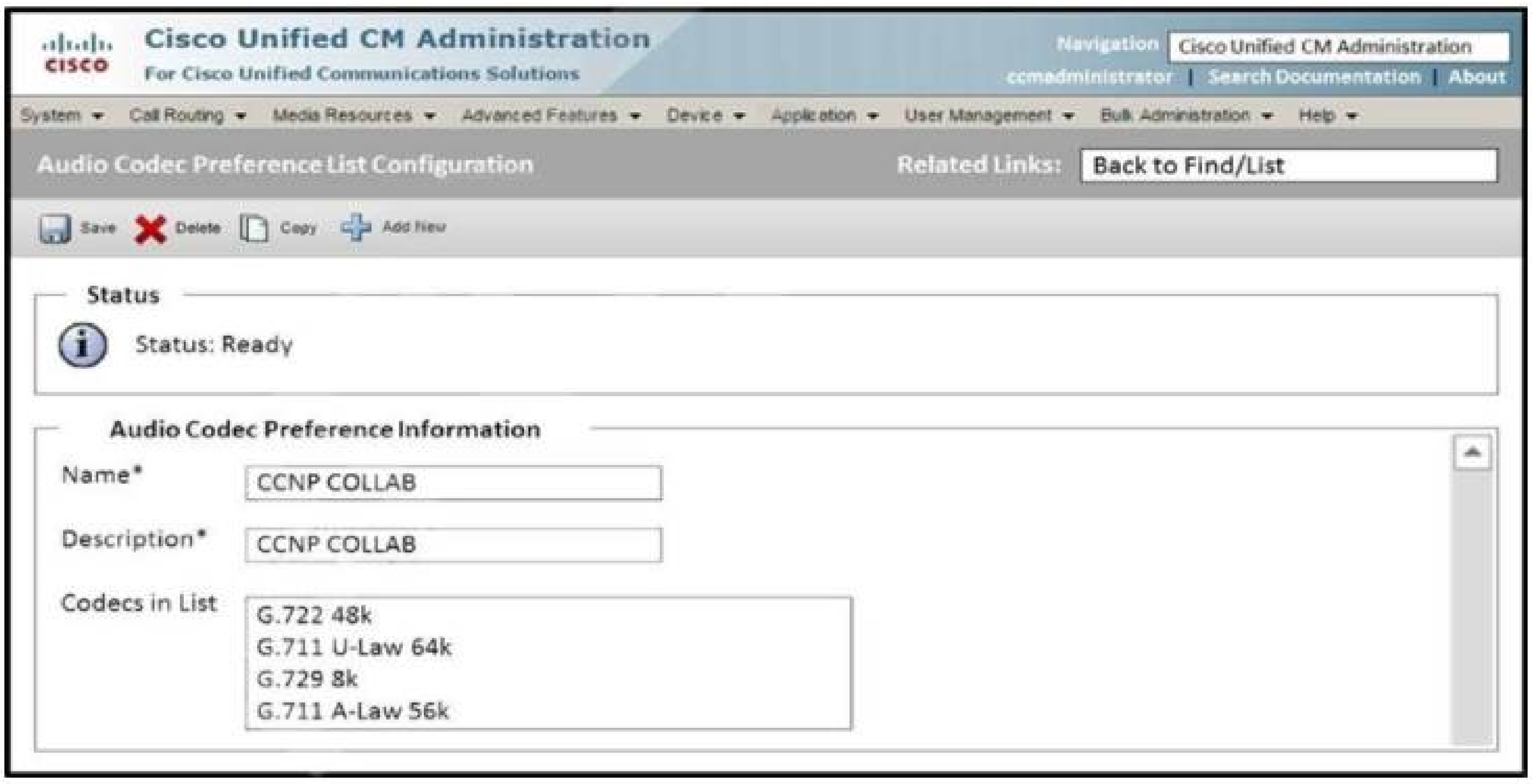
```
Router# configure terminal
Router(config)# interface gigabitethernet 5/1
Router config-if)# description Cube E41.228-0097
```

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 323
Refer to the exhibit.





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 Regio1l 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 324
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