Cisco

Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

NEW QUESTION 1
What happens when a Cisco IP phone loses connectivity to the duster during an active call?

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

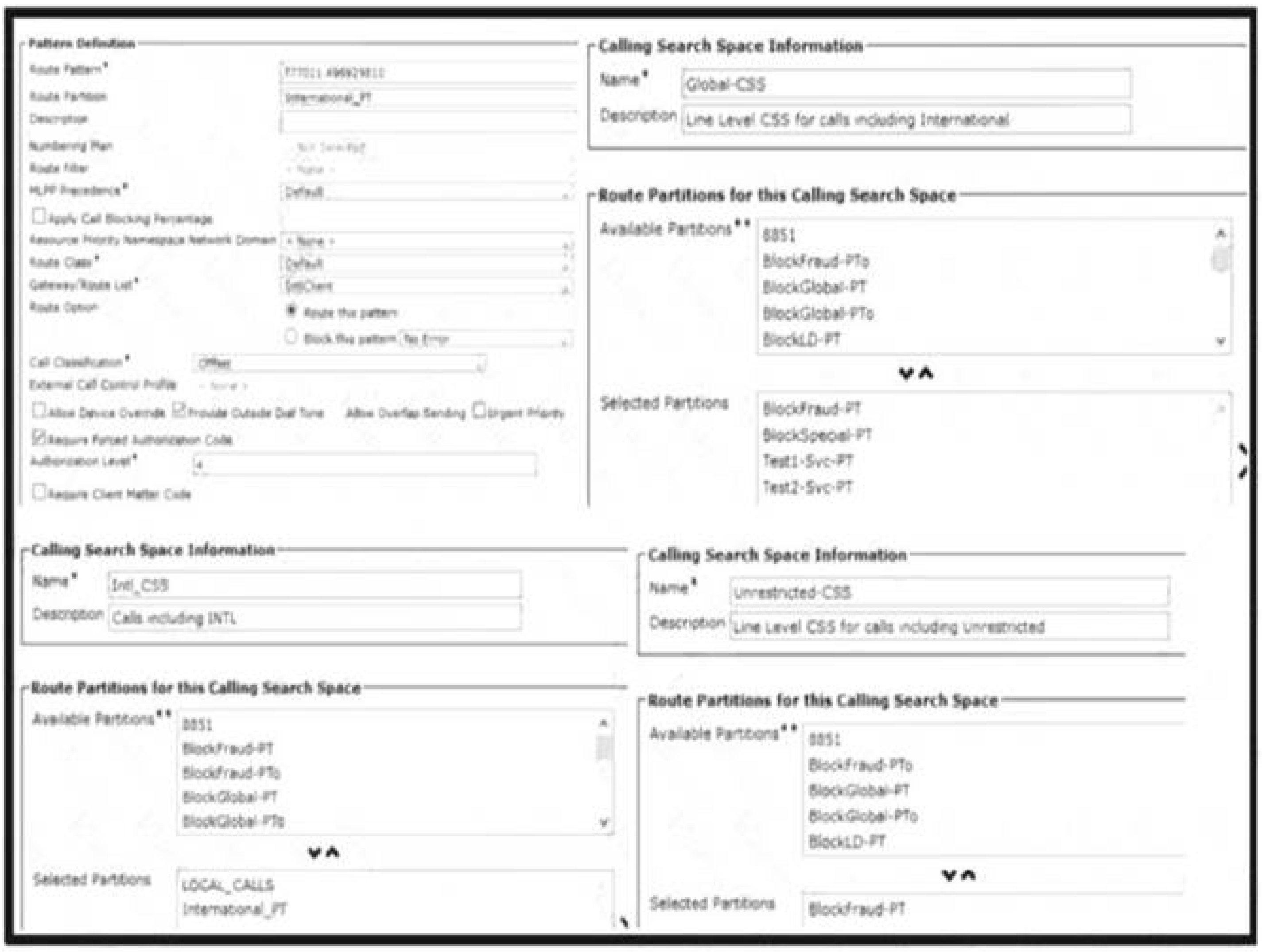
Answer: D

NEW QUESTION 2
What is an indicator of network congestion in VoIP communications?

A. jitter increase due to variable delay
B. discards in the interface of routers and switches
C. video loss due to video frame corruption
D. gaps in the voice due to packet loss

Answer: A

NEW QUESTION 3
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 4
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 5
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 6
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 7
Refer to the exhibit.

```
Sent:
INVITE sip:2004@192.168.100.100:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.100.200:5060;branch=z9hG4bKF1FED
From: "7000" <sip:7000@192.168.100.200>;tag=43CDE-1A22
To: <sip:2004@192.168.100.100>
Call-ID: 26BCA00-4C4E11EA-80169514-B1C46126@192.168.100.200
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE:  1800
User-Agent: Cisco-SIPGateway/IOS-16.9.5
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Contact: <sip:7000@192.168.100.200:5060>
Expires: 180
Max-Forwards: 68
P-Asserted-Identity: "7000" <sip:7000@192.168.100.200>
Session-Expires:  1800
Content-Type: application/sdp
Content-Length: 254

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

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

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

Handler

Answer: B

NEW QUESTION 8
Refer to the exhibit.

```
controller t1 0/0/1
 pri-group timeslots 1-24
 clock source line
 linecode b8zs
 framing esf
```

An administrator must replace the T1 card with an E1 card. What is the correct configuration if the administrator was asked to configure 12 time slots?

A.
```
controller e1 0/0/1
 pri-group timeslots 1-12
 clock source network
 linecode hdb3
 framing crc4
```

B.
```
controller e1 0/0/1
 pri-group timeslots 1-11, 12
 clock source line
 linecode hdb3
 framing crc4
```

C.
```
controller e1 0/0/1
 pri-group timeslots 1-12
 clock source line
 linecode hdb3
 framing crc4
```

D.
```
controller e1 0/0/1
 pri-group timeslots 1-12
 clock source line
 linecode crc4
 framing hd3
```

Answer: C

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 11
An engineer must configure a SIP route pattern using domain routing with destination +13135551212. The domain ciscocm1.jupiter.com resolves to 192.168.1.3. How must the IPV4 Pattern be configured?

A. +13135551212@192.168.1.3
B. ciscocm1.jupiter.com
C. \+13135551212@192.168.1.3
D. 192.168.1.3

Answer: B

NEW QUESTION 16
Which dial plan function restricts calls that are made by a lobby phone to internal extensions only?

A. manipulation of dialed destination
B. path selection
C. calling privileges
D. endpoint addressing

Answer: C

NEW QUESTION 21
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 25
Which configuration concept allows for high-availability on IM and Presence services in a UC environment?

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

Answer: D

NEW QUESTION 27
An engineer is configuring a phone system CISCO UCM and wants to activate TFTP service. The engineer selects the serviceability page for configuration. Which nodes configurable for TFTP?

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

Answer: C

Explanation:
TFTP is a network protocol that is used to transfer files between devices. It is often used to transfer firmware and configuration files to network devices. In order to use TFTP, the device must have a TFTP server configured.
In Cisco UCM, the TFTP server is configured on the serviceability page. The TFTP server can be configured on any node that has Cisco UCM service enabled. The TFTP server cannot be configured on nodes that do not have Cisco UCM service enabled.

NEW QUESTION 31
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 36
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 38
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 42
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 43
An administrator must configure the Local Route Group feature on Cisco UCM. Which step will enable this feature?

A. For each route group, check the box for the Local Route Group feature.
B. For each route pattern, select the Local Route Group as the destination.
C. For each device pool, configure a route group to use as a Local Route Group for that device pool
D. For each route list, configure a route group to use as a Local Route Group.

Answer: C

Explanation:
The Local Route Group feature allows you to use a route group as the destination for calls that are placed from a device pool. The route group that you use as the destination for calls from a device pool is called the Local Route Group for that device pool.
To configure the Local Route Group feature, you must first create a route group. You can then configure the Local Route Group feature for a device pool by selecting the route group that you want to use as the Local Route Group for that device pool.

NEW QUESTION 46
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 47
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 50
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 52
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 56
Where in Cisco UCM is restrictions on audio bandwidth configured?

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

Answer: C

NEW QUESTION 60
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 65
Which type of input is required when configuring a third-party SIP phone?

A. digest user
B. manufacturer
C. serial number350-801 2023-4
D. authorization code

Answer: A

NEW QUESTION 66
What should be used to detect common issues on a Cisco IOS XE-based Local Gateway and generate an email?

A. Real-Time Monitoring Tool
B. diagnostic signatures
C. syslog

D. SNMP

Answer: B

Explanation:
Diagnostic signatures are a feature that proactively detects commonly observed issues in the IOS XE-based Local Gateway and generates email, syslog, or terminal message notification of the event12. You can also install the diagnostic signatures to automate diagnostics data collection and transfer collected data to the Cisco TAC case to accelerate resolution time12.

NEW QUESTION 70
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 75
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 80
An engineer configures a new phone in Cisco UCM The phone boots and gets IP when it connects to the network, however the phone fails to register With CISCO UCM, The engineer observes that the phone has a status Rejected In CISCO UCM
What must be verified first 'Mien troubleshooting the issue?

A. whether auto-registration is enabled in Cisco UCM
B. whether the Initial Trust List and Certificate Trust List files on the phone are correct
C. whether the phone ts in the correct VLAN
D. whether the phone's MAC address is correct In Cisco UCM

Answer: A

Explanation:
This is the first thing that must be verified when troubleshooting the issue of phone status showing rejected in Cisco UCM1. Auto-registration allows new phones to register with Cisco UCM without manual configuration 1. If auto-registration is disabled, the phone will not be able to register and will show a rejected statu1s.
The other options are not the first things that must be verified when troubleshooting the issue:
- B. whether the Initial Trust List and Certificate Trust List files on the phone are correct is not the first thing to verify, but it may be a possible cause of the issue if the phone has an ITL file from another cluster that prevents it from registering with Cisco UCM2. To resolve this issue, the ITL file needs to be deleted from the phone or exchanged between the clusters2.
- C. whether the phone is in the correct VLAN is not the first thing to verify, but it may be a possible cause of the issue if the phone is not in the same VLAN as the Cisco UCM server or cannot reach it due to network issues3. To resolve this issue, the network connectivity and VLAN configuration need to be checked and fixed3.
- D. whether the phone's MAC address is correct in Cisco UCM is not the first thing to verify, but it may be a possible cause of the issue if the phone's MAC address does not match the one configured in Cisco UCM. To resolve this issue, the MAC address needs to be corrected and updated in Cisco UCM.

NEW QUESTION 83
Which location must be assigned to the SIP trunk to replicate enhanced location information via a SIP trunk?

A. phantom
B. replica
C. hub_none
D. shadow

Answer: D

NEW QUESTION 86
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 89
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 94
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 98
What is the major difference between the two possible Cisco IM and Presence high-availability modes?

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

Answer: C

Explanation:
Balanced mode provides user load balancing and user failover in the event of an outage. Active/standby mode provides an always on standby node in the event of an outage, but it does not provide load balancing.
Here is a more detailed explanation of the two modes:
- Balanced mode: In balanced mode, the IM and Presence Service nodes are configured to work together to provide high availability. The nodes are configured in a redundancy group, and the system automatically balances the load of users across the nodes in the group. If one of the nodes fails, the system automatically fails over the users to the other nodes in the group.
- Active/standby mode: In active/standby mode, one of the IM and Presence Service nodes is designated as the active node, and the other nodes are designated as standby nodes. The active node handles all of the user traffic, and the standby nodes are only used if the active node fails. If the active node fails, the system automatically fails over to one of the standby nodes.

NEW QUESTION 99
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 101

What must be configured on a Cisco Unity Connection voice mailbox to access the mailbox from a secondary device?

A. mobile user
B. alternate names
C. alternate extensions
D. Attempt Forward routing rule

Answer: C

Explanation:
To access a Cisco Unity Connection voice mailbox from a secondary device, you must configure an alternate extension for the mailbox. This is a phone number that is different from the mailbox's primary extension. When you call the alternate extension, you will be prompted to enter the mailbox's PIN. Once you have entered the PIN, you will be able to access the mailbox just as you would if you were calling from the primary device.

NEW QUESTION 105
What is the maximum number of servers that are in an IM and Presence presence redundancy group?

A. 10
B. 6
C. 2
D. 4

Answer: C

NEW QUESTION 108
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 109
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 111
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 112
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 117
A user forwards a corporate number to an international number. What are two methods to prevent this forwarded call? (Choose two.)

A. Configure a Forced Authorization Code on the international route pattern.
B. Block international dial patterns in the SIP trunk CSS.
C. Set Call Forward All CSS to restrict international dial patterns.
D. Set the Call Classification to OnNet for the international route pattern.
E. Check Route Next Hop By Calling Party Number on the international route pattern.

Answer: AC

NEW QUESTION 120
Refer to the exhibit.

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

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

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

Answer: B

NEW QUESTION 121
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 123
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 126
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 127
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 128
An engineer implements a new Cisco UCM based telephony system per these requirements.
•The local Ethernet bandwidth is sized based on the total bandwidth per call
•A G 736 codec is used.
•The bit rate is 64 kbps
•The codec sample interval is 10 ms
•The voice payload size is 160 bytes per 20 ms
What should the size of the Ethernet bandwidth be per call?

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

Answer: D

NEW QUESTION 133
Where in Cisco UCM are codec negotiations configured for endpoints?

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

Answer: C

NEW QUESTION 135
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 136
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 139
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 144
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 147
Which two protocols are proxied over an Expressway-E/C pair when a Mobile and Remote Access login including phone services is performed? (Choose two.)

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

Answer: AC

NEW QUESTION 151
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 153
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 156
A company deploys centralized cisco ucm architecture for a hub location and two remote sites.
*The company has only one ITSP connection at the hub connection, and ITSP supports only G.711 calls
*Remote site A has a 1-Gbps fiber connection to the hub connection and calls to and from remote side A use G.711 codec
*Remote site B has a 1 T1 connection to the hub location and calls to and from remote site B use G.729 codec Based on the provided guidance, a Cisco voice engineer must design media resource management for the
customer What is the method that needs to be followed?

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

Answer: C

NEW QUESTION 161
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 164
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 169
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 173
The security department will audit an IT department to ensure that the proper guidelines are being followed. The reports of the call detail records show unauthorized access to PSTN. Which two actions should an administrator check to prevent the unauthorized use of the telephony system? (Choose two.)

A. Ensure that ad hoc conference calls are dropped if an external user is add.
B. Call forward settings (ALL/Busy/No Answer) are restricted to internal extensions in the network
C. Add an additional firewall between the Cisco UCM server and the Expressway Core server.
D. For extension mobility, logged-out CSS is restricted to internal extensions and emergencies.
E. Forced authorization code is used to recognize a dialing extension and authorize an international call.

Answer: BE

NEW QUESTION 175
A customer asked to integrate Unity Connection with Cisco UCM using SIP protocol. Which two features must be enabled on SIP security profiles? (Choose two.)

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

Answer: CE

NEW QUESTION 179
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 184
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 186
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 189
Refer to the exhibit.

```
hostname GATEWAY
ccm-manager config
ccm-manager config server 192.168.1.100
ccm-manager mgcp

mgcp call-agent CCMSub1.domain.com 2427 service-type mgcp version 0.1
```

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 193
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 198
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 200
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 201
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 203
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 208
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 209
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 214
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

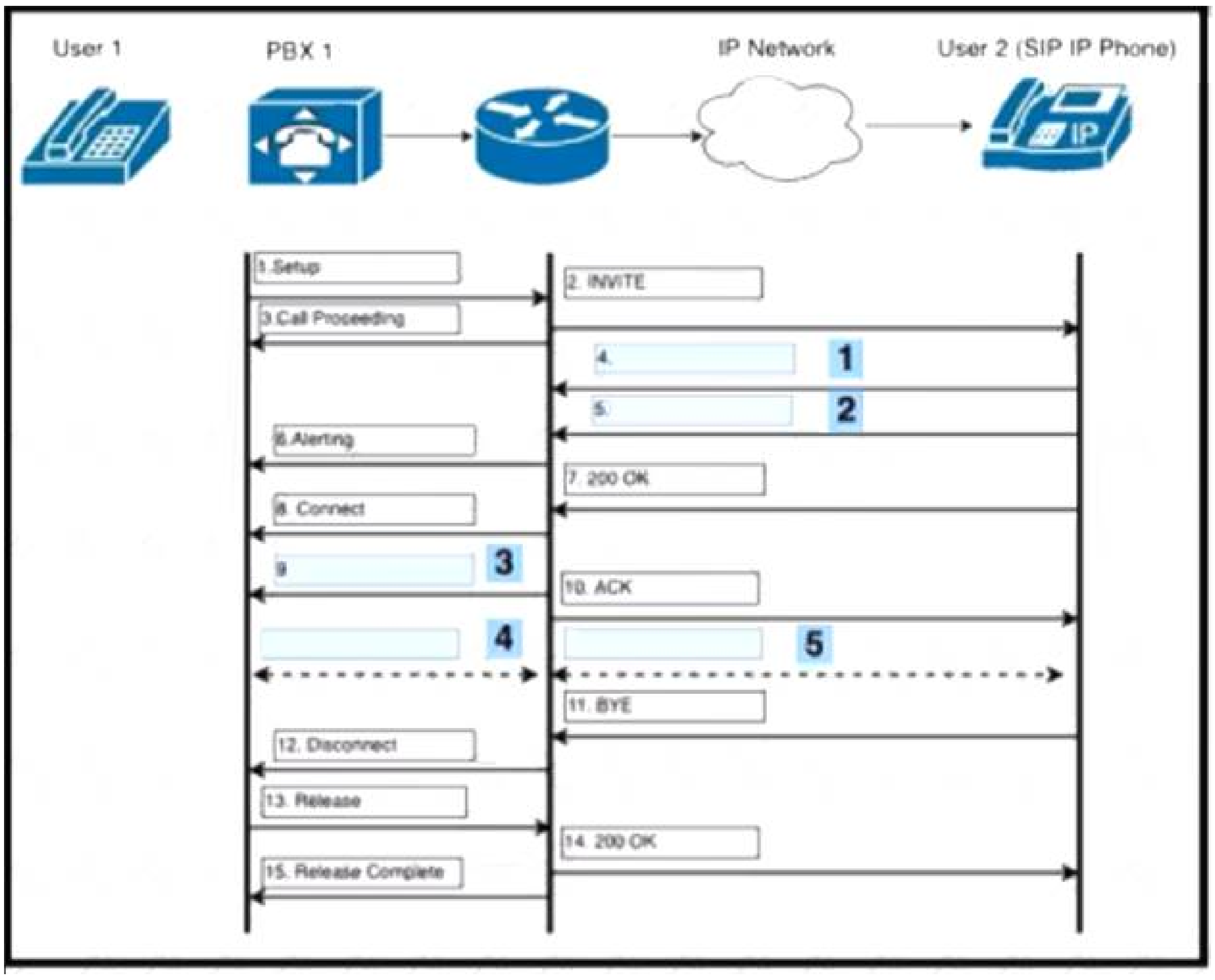
Answer: C

NEW QUESTION 215
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 220
Refer to the exhibit.



https://i.postimg.cc/wMYy0Fhm/image.png
Drag and drop the flow step labels from the left into the correct order on the right to establish this call flow:
• User 1 calls user 2.
• User 2 answers the call
• user 2 disconnects the call

two-way voice path

two-way RTP channel

100 Trying

Connect ACK

180 Ringing

A. Mastered
B. Not Mastered

Answer: A

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

NEW QUESTION 225
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 230
An engineer encounters third-party devices that do not support Cisco Discovery Protocol. What must be configured on the network to allow device discovery?

A. LLDP
B. TFTP
C. LACP
D. SNMP

Answer: A

Explanation:
LLDP (Link Layer Discovery Protocol) is a vendor-neutral network discovery protocol that is used to discover the topology of a network. LLDP is similar to CDP (Cisco Discovery Protocol), but it is not proprietary to Cisco. LLDP is supported by a wide range of network devices, including switches, routers, and firewalls.
To configure LLDP on a network, you must enable LLDP on the devices that you want to discover. You can then use a network management tool, such as Cisco Network Assistant, to view the topology of the network.
The other options are incorrect. TFTP (Trivial File Transfer Protocol) is a network protocol that is used to transfer files between devices. LACP (Link Aggregation Control Protocol) is a network protocol that is used to aggregate multiple network links into a single logical link. SNMP (Simple Network Management Protocol) is a network protocol that is used to manage network devices.

NEW QUESTION 231
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 233
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 234
What is a capability of Cisco Expressway?

A. It functions as an analog telephony adapter.
B. It has remote endpoint enrollment with Certificate Authority Proxy Function.
C. It gives directory access for remote users via Cisco Directory Integration.
D. It provides access to on-premises Cisco Unified Communications infrastructure for remote endpoints.

Answer: D

NEW QUESTION 236
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

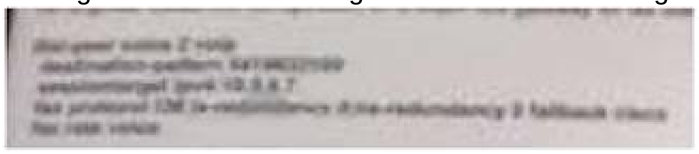
NEW QUESTION 239

Which Cisco UCM configuration is required for SIP MWI integrations?

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

Answer: C

NEW QUESTION 242
An engineer builds the configuration on a Cisco IOS gateway for the dial-peers:



Which command is required to complete the configuration?

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

Answer: D

NEW QUESTION 245
Where is Directory Connector hosted in a Cisco Webex Hybrid Services deployment?

A. on a server in the Webex Data Center
B. on a dedicated on-premises server
C. on a Cisco Expressway-C connector host server
D. on an on-premises Microsoft Active Directory server

Answer: B

Explanation:
The Cisco Directory Connector is a software application that is installed on a dedicated on-premises server. It synchronizes user identities between the on-premises directory and the Cisco Webex cloud.

NEW QUESTION 246
An administrator needs to help a remote employee make a free call to an international destination. The administrator calls the employee, then conferences in the international party. The administrator drops the call, and the employee and the international party continue their conversation. Which action prevents this type of toll fraud in the Cisco UCM?

A. Set service parameter 'Advanced Ad Hoc Conference" to FALSE.
B. Set service parameter "Drop Ad Hoc Conference" to "When Conference Controller leaves."
C. Set service parameter "Advanced Ad Hoc Conference" to 2.
D. Set service parameter "Drop Ad Hoc Conference" to "Do not allow outside parties."

Answer: B

NEW QUESTION 248
An administrator is developing an 8-class QoS baseline model. The CS3 standards-based marking recommendation is used for which type of class?

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

Answer: A

NEW QUESTION 250
Which IP Precedence value is used to classify a call signalling packet?

A. 6
B. 5
C. 4
D. 3

Answer: D

NEW QUESTION 255
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 260
Refer to the exhibit.

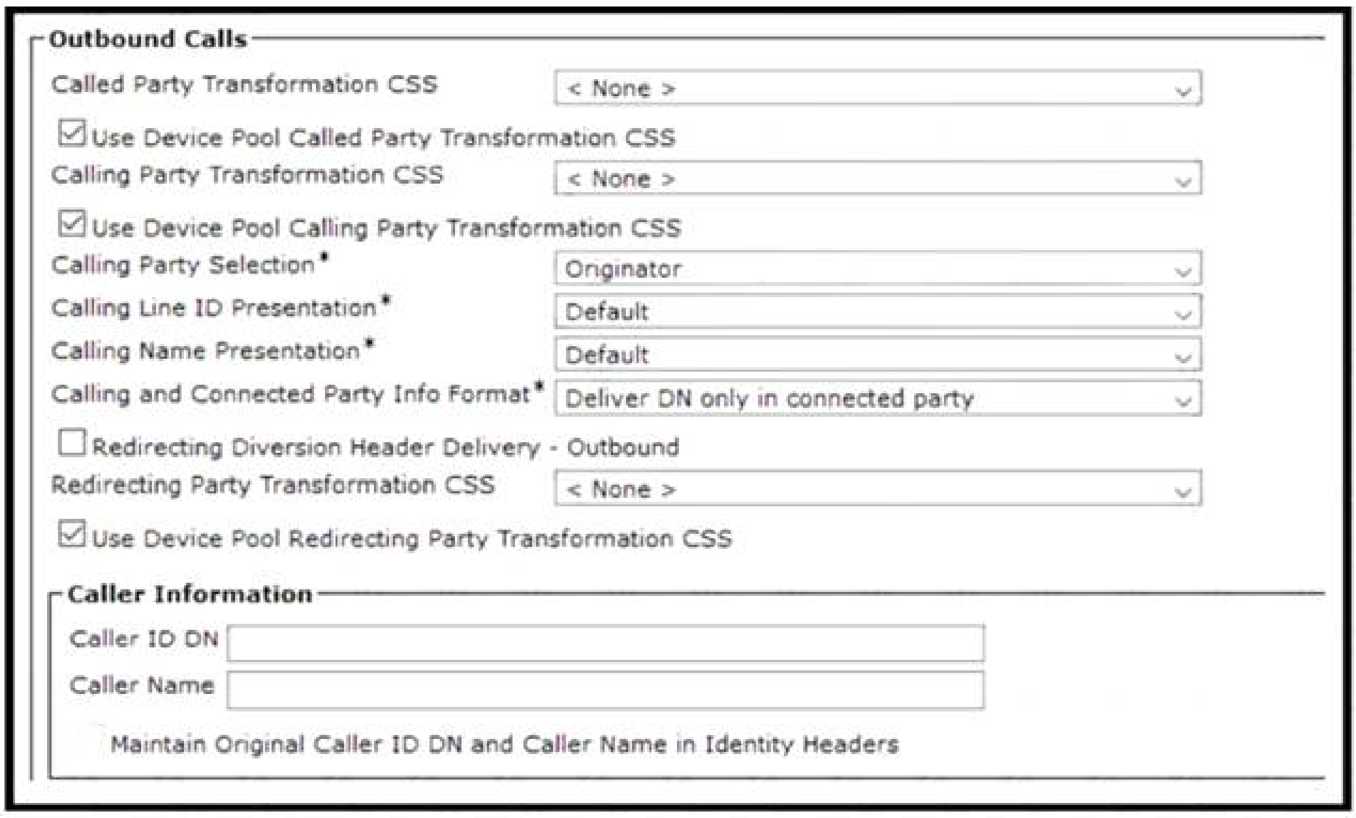
```
isdn switch-type primary-ni
controller t1 0/1/0
framing esf
linecode b8zs
pri-group timeslots 1-10
```

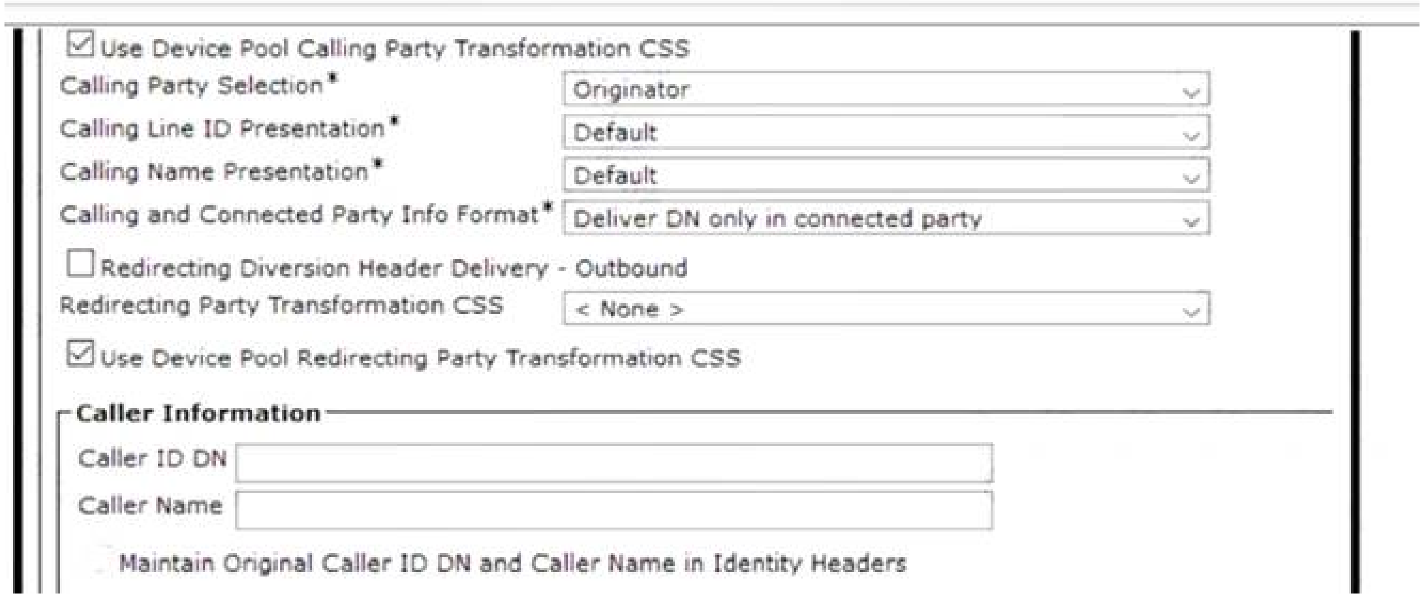
An engineer configures ISDN on a voice gateway. The provider confirms that the PRI is configured with 10 channels the engineer ordered and is working from the provider side, but the engineer cannot get a B-channel to carry voice. The rest of the configuration for the serial interface and voice network 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 262
Refer to the exhibit.





Unanswered calls do not reach the voicemail associated with the phones Instead, callers receive the default greeting Which action fixes the configuration?

A. Reboot Cisco Unity Connection.
B. Check the box "Redirecting Diversion Header Delivery - Outbound", then reset the trunk.
C. Check the box 'Redirecting Diversion Header Delivery - Outbound".
D. Review the conversation manager logs on Cisco Unity Connection.

Answer: B

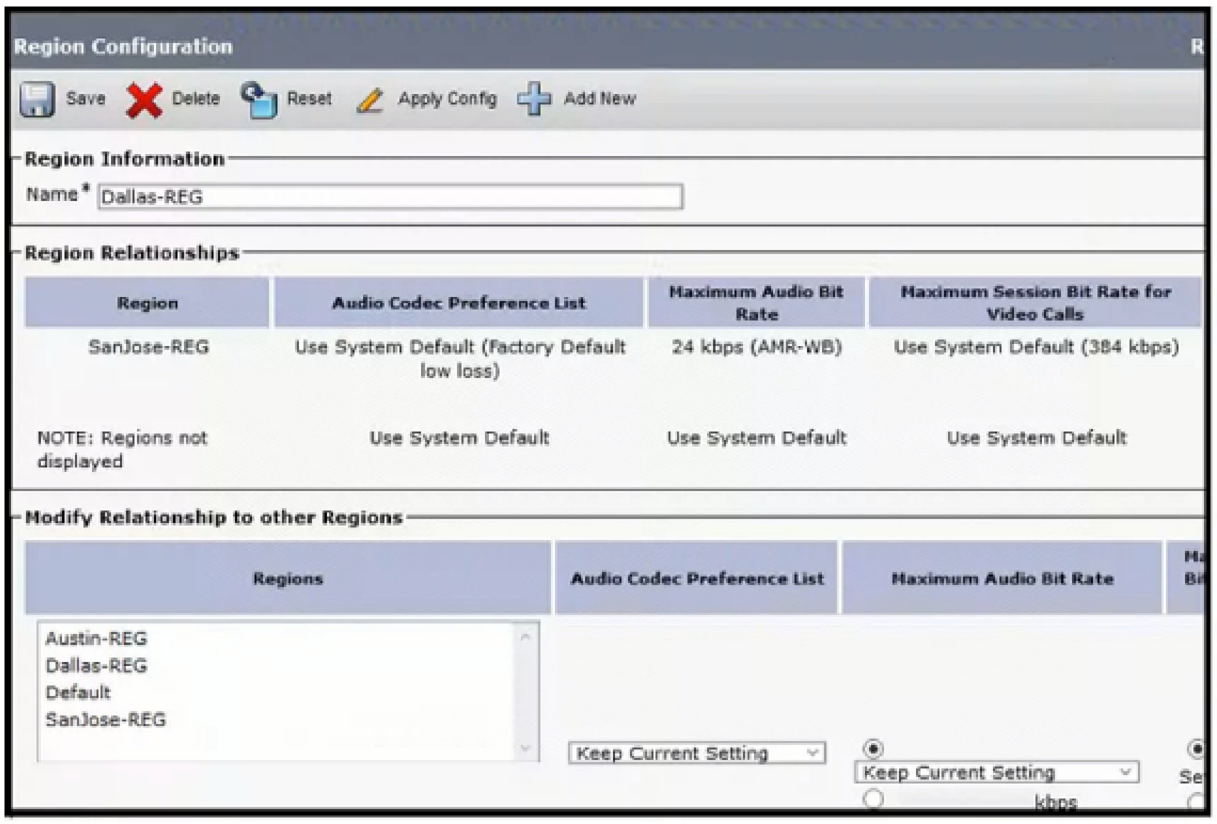
NEW QUESTION 265
An engineer troubleshoots a Cisco Jabber login problem on a Windows PC. The login fails with the error message "Cannot find your services automatically. Click advanced settings to set up manually " Which action should the engineer take first?

A. Verify whether the cup-xmpp certificates are valid.

B. Verify the username and password and try again.
C. Perform a manual DNS lookup of SRV record _cisco-uds._tcp.domain.com.
D. Perform a manual DNS lookup of SRV record _collab-edge._tls.domain.com.

Answer: C

NEW QUESTION 269
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 274
Which service must be enabled when LDAP on cisco UCM is used?

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

Answer: C

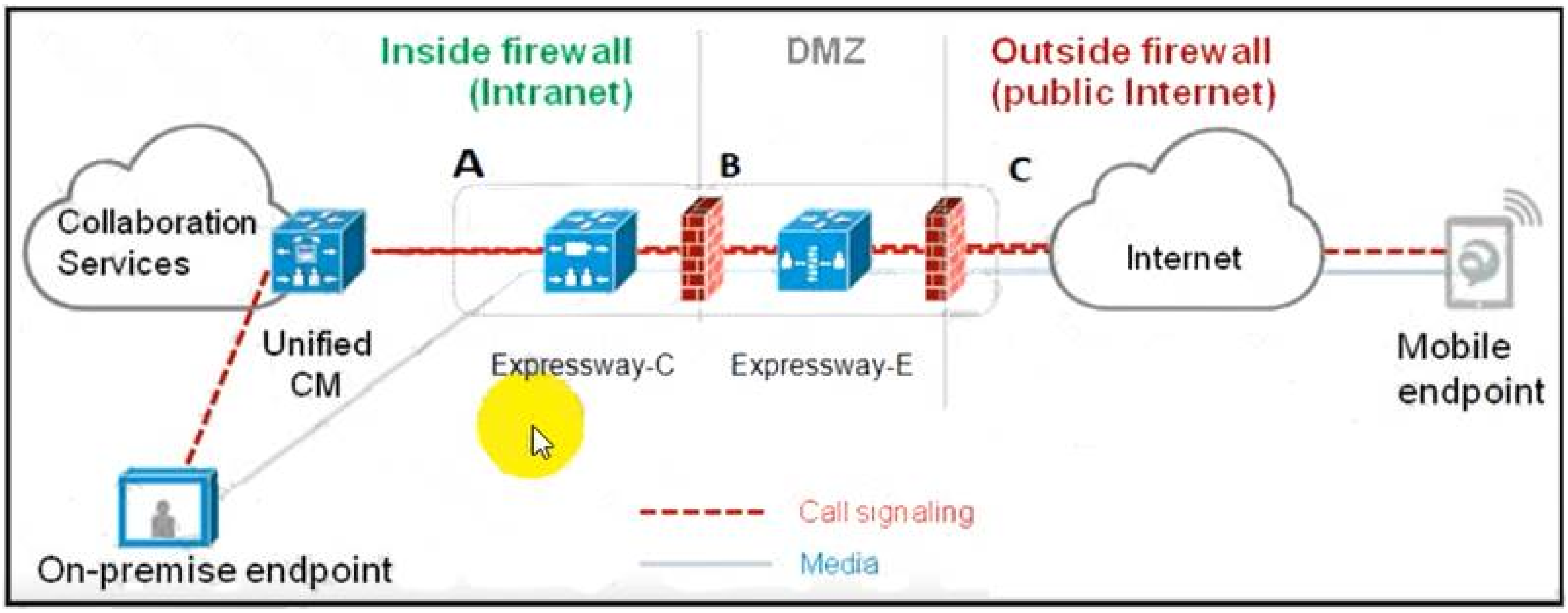
NEW QUESTION 277
Which two steps should be taken to provision a phone after the Self-Provisioning feature was configured for end users? (Choose two.)

A. Ask the Cisco UCM administrator to associate the phone to an end user.
B. Plug the phone into the network.

C. Dial the hunt pilot extension and associate the phone to an end user
D. Dial the self-provisioning IVR extension and associate the phone to an end user.
E. Enter settings menu on the phone and press * ,*,# (star, star, pound).

Answer: BD

NEW QUESTION 280
Refer to the exhibit.



When making a call to a Mobile and Remote Access client, what are the combinations of protocol on each of the different sections A-B-C?

A. IP TCP/TLS (A) + SIP TCP/TLS (B) + SIP TLS (C)
B. SIP TCP/TLS (A) + SIP TCP/TLS (B) + SIP TCP/TLS (C)
C. SIP TLS (A) + SIP TLS (B) + SIP TLS (C)
D. SIP TCP/TLS (A) + SIP TLS (B) + SIP TLS (C)

Answer: D

NEW QUESTION 284
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 287
An administrator troubleshoots call flows and suspects that there are issues with the dial plan. Which tool enables a quick analysis of the dial plan and provides call flows of dialled digits?

A. Cisco Dial Plan Analyzer
B. Dial Plan Analyzer
C. Digit Analysis Analyzer
D. Dialed Number Analyzer

Answer: D

NEW QUESTION 288
Which two DNS records must be created to configure Service Discovery for on-premises Jabber? (Choose two.)

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

Answer: BD

NEW QUESTION 293
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 297
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 300
An engineer configures Cisco UCM to prevent toll fraud. At which two points does the engineer block the pattern in Cisco UCM to complete this task? (Choose two.)

A. partition
B. route partem
C. translation pattern
D. CSS
E. route group

Answer: AD

NEW QUESTION 303
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 307
Which value should be changed when each Cisco UCM node does not allow for more than 5000 phones to be registered?

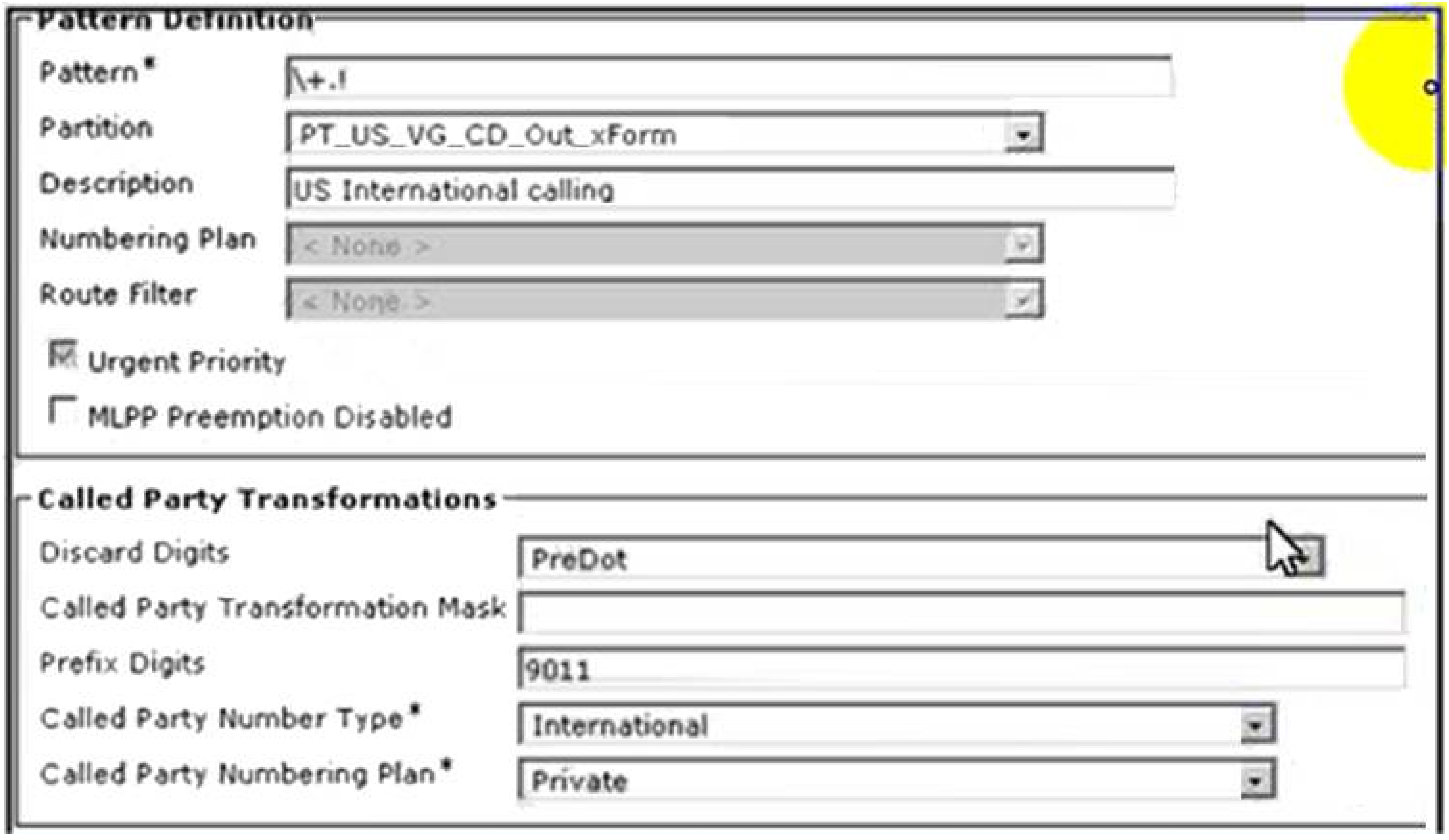
A. Maximum Number of Registered and Unregistered Devices service parameter on each node
B. Minimum Number of Phones service parameter on each node
C. Maximum Number of Registered Devices service parameter on each node
D. Maximum Number of Phones service parameter on the Publisher

Answer: C

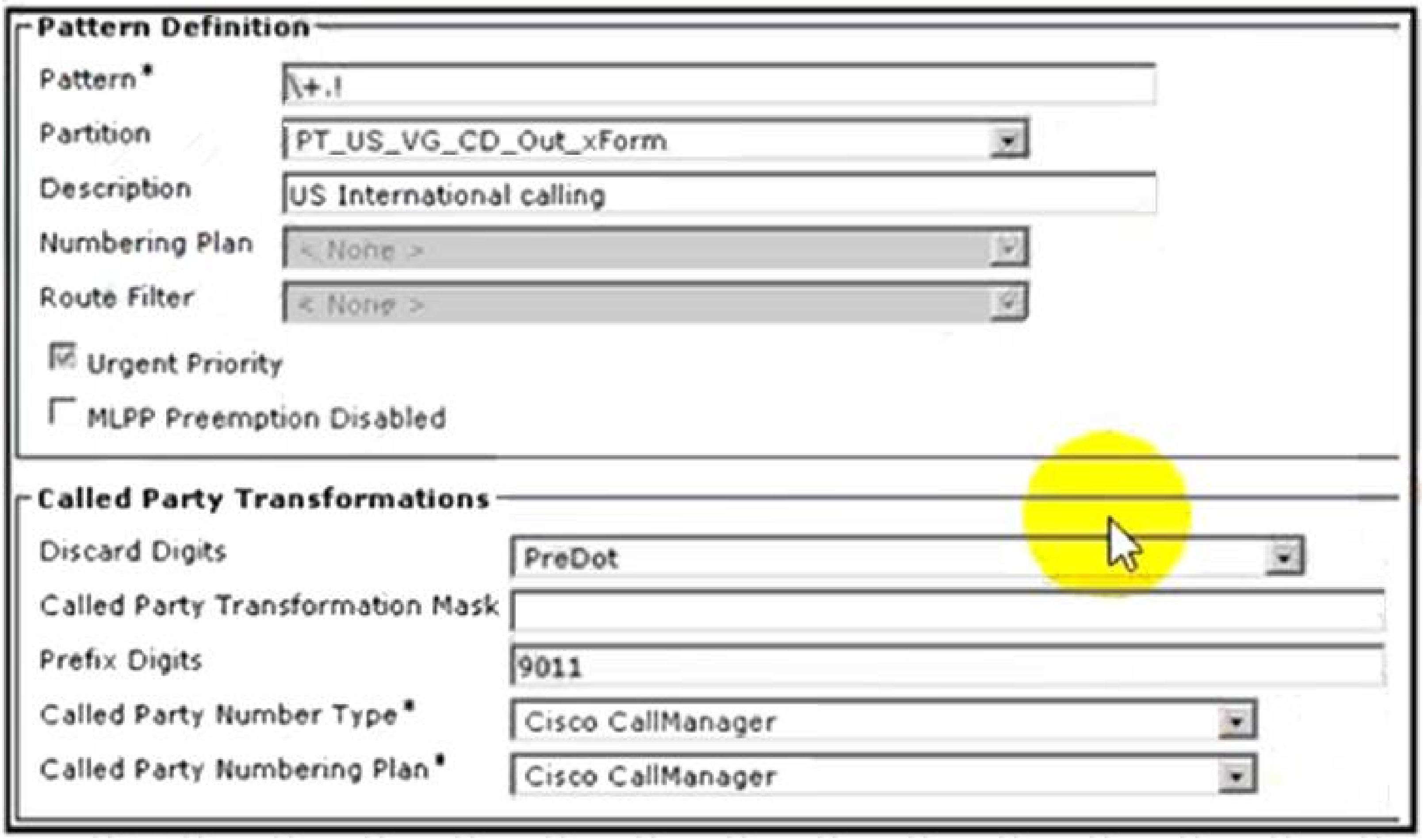
NEW QUESTION 309
Given this H.323 gateway configuration and using Cisco best practices, how must the called party transformation pattern be configured to ensure that a proper ISDN type of number is set?

```
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
```

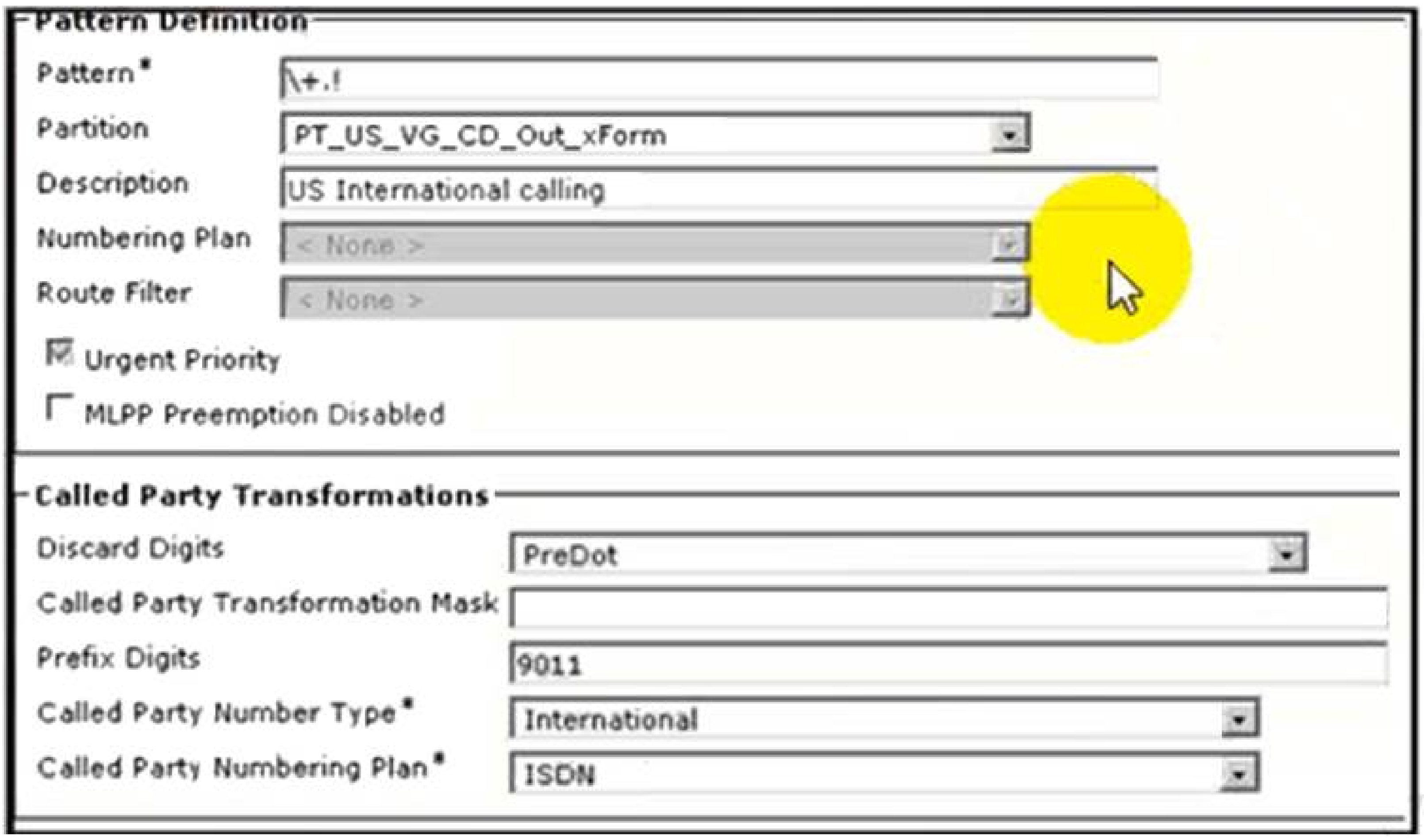
A.



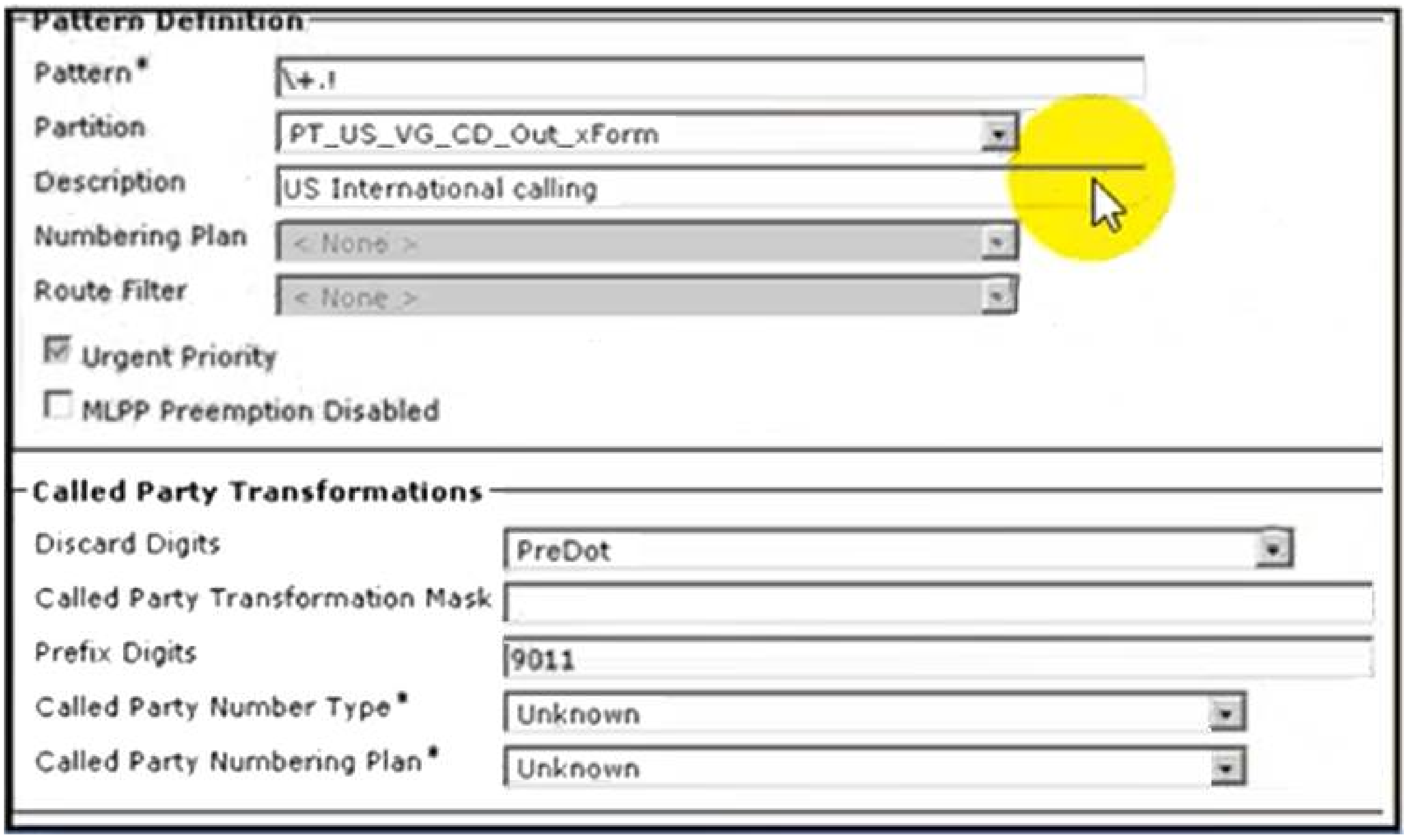
B.



C.



D.



Answer: C

NEW QUESTION 313
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 314
Refer to the exhibit.

```
Voice class codec 1
codec preference 1 g711alaw
codec preference 2 g711ulaw
codec preference 3 g729r8

dial-peer voice 13 voip
description incoming dialpeer from ITSP
incoming called-number .
voice-class codec 1

dial-peer voice 19 voip
description outgoing dialppeer to CUCM
destination-pattern .T
session protocol sipv2
session-target ipv4:3.3.3.3
voice-class codec 1

Incoming SDP from ITSP

v=0
o=sip:test@2.2.2.2 1 16 IN IP4 2.2.2.2
s=sip:test@2.2.2.2
c=IN IP4 2.2.2.2
t=0 0
m=audio 5000 RTP/AVP 18 0
a=rtpmap:0 PCMU/8000/1
a=rtpmap:18 G729/8000/1
```

Which outgoing m-line SDP is sent to Cisco UCM after matching the appropriate dial peers via Cisco Unified Border Element?

A.
```
m=audio 16550 RTP/AVP 8 0 18
a=rtpmap:0 PCMU/8000/1
a=rtpmap:8 PCMA/8000/1
a=rtpmap:18 G729/8000/1
```

B.
```
m=audio 16550 RTP/AVP 18 0
a=rtpmap:0 PCMU/8000/1
a=rtpmap:18 G729/8000/1
```

C.
```
m=audio 16550 RTP/AVP 18 0
a=rtpmap:8 PCMA/8000/1
a=rtpmap:0 PCMU/8000/1
a=rtpmap:18 G729/8000/1
```

D.
```
m=audio 16550 RTP/AVP 0 8 18
a=rtpmap:0 PCMU/8000/1
a=rtpmap:8 PCMA/8000/1
a=rtpmap:18 G729/8000/1
```

Answer: B

NEW QUESTION 315
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 319
Refer to the exhibit.

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

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

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

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

Answer: C

NEW QUESTION 324
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 325
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 328
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 329
An engineer is integrating Unity Connection with Cisco UCM. Which two actions must be configured so that recording and playback from the IP phones works at all times, including peak traffic hours? (Choose two.)

A. Increase the number of voice ports.
B. If it's a Unity Connection Cluster, ensure that replication is fine and not in split-brain mode.
C. The phone system to which the phones are registered in Unity Connection has the Default Trap Switch check box enabled.
D. Add dedicated dial-out ports with the allow trap connections setting selected.
E. Ensure that you have set up SIP Digest Authentication on the SIP trunk security profile.

Answer: AC

NEW QUESTION 332
Which behavior occurs when Cisco UCM has a Call Manager group that consists of two subscribers?

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

Answer: A

NEW QUESTION 334
An engineer is deploying Webex app on Microsoft Windows computers. The engineer wants to ensure that the end users do not receive pop-IQ dialogues %'hen they start the application 'Much two actions ensure the end users are not promted to accept the end-user license (Choose two)

A. Set the DELETEUSERDATA=r installation argument
B. Set the "HKEY_LOCAL_MACHINE,.Software'«WOW6432Node .CiscoCollabHost -Eula_disable
C. Set the "HKEY_LCX^M._MACHINE .SoftwareCiscoCollabHo$t€ula Setting registry Eula_disable
D. Set the DEFAULT^'THEMEs-Dark"" installation argument
E. Set the "/quiet installation argument

Answer: BC

Explanation:
The correct answers are B and C.
To ensure that end users are not prompted to accept the end-user license agreement (EULA) when they start the Webex app, the engineer must set the following two registry keys:
- HKEY_LOCAL_MACHINE\Software\WOW6432Node\CiscoCollabHost\Eula_disable

- HKEY_LOCAL_MACHINE\Software\CiscoCollabHost\Eula Setting\Eula_disable

Setting these registry keys will disable the EULA prompt for all users who start the Webex app.
The other options are not valid actions to ensure that end users are not prompted to accept the EULA.

NEW QUESTION 339
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 342
What are the last two bits of a DS field in DiffServe Byte used for?

A. INC
B. AFxy
C. ECN
D. RMI

Answer: C

NEW QUESTION 343
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 348
When a new SIP phone is registered to Cisco Unified communications Manager, it keeps failing and showing an “unprovisioned” error message in the phone display. Which problem is a possible cause of this issue?

A. Auto-registration is disabled on the Cisco Unified Communications Manager nodes and the phone device does not have a DN configured.
B. The DHCP settings are incorrectly and the phone does not have an alternate TFTP defined.
C. The phone cannot download and install the latest firmware.
D. The DN assigned to the phone is already in use by another SIP phone.
E. The DN configuration for this phone is shared with SCCP phone, which is not supported.

Answer: B

NEW QUESTION 350
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 352
What happens to voice packets from a Cisco 8845 IP phone in the QoS trust boundary?

A. The voice packets are not trusted, and the access layer switch reclassifies the packets.
B. The voice packets are classified by the phone, and the classification is accepted
C. The voice and access layer switch negotiate the classification of packets
D. Cisco UCM determines how the voice packers are classified.

Answer: B

NEW QUESTION 356
An engineer is configuring a BOT device for a Jabber user in Cisco Unified Communication Manager. Which phone type must be selected?

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

Answer: A

NEW QUESTION 358
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 363
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 366
In the cisco expressway solution, which two features does mobile and Remote access provide? (Choose two)

A. VPN-based enterprise access for a subset of Cisco Unified IP Phone models
B. secure reverse proxy firewall traversal connectivity
C. the ability 10 register third-party SIP or H 323 devices on Cisco UCM without requiring VPN
D. the ability of Cisco IP Phones to access the enterprise through VPN connection
E. the ability for remote users and their devices to access and consume enterprise collaboration applications and services

Answer: BE

NEW QUESTION 368
Users want their mobile phones to be able to access their cisco unity connection mailboxes with only having to enter their voicemail pin at the login prompt calling pilot number where should an engineer configure this feature?

A. transfer rules
B. message settings
C. alternate extensions
D. greetings

Answer: C

NEW QUESTION 371

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 372
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 375
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 378
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 379
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