Cisco

Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

About Exambible

Your Partner of IT Exam

Found in 1998

Exambible is a company specialized on providing high quality IT exam practice study materials, especially Cisco CCNA, CCDA, CCNP, CCIE, Checkpoint CCSE, CompTIA A+, Network+ certification practice exams and so on. We guarantee that the candidates will not only pass any IT exam at the first attempt but also get profound understanding about the certificates they have got. There are so many alike companies in this industry, however, Exambible has its unique advantages that other companies could not achieve.

Our Advances

* 99.9% Uptime

    All examinations will be up to date.

* 24/7 Quality Support

    We will provide service round the clock.

* 100% Pass Rate

    Our guarantee that you will pass the exam.

* Unique Gurantee

    If you do not pass the exam at the first time, we will not only arrange FULL REFUND for you, but also provide you another exam of your claim, ABSOLUTELY FREE!

NEW QUESTION 1
What are two QoS requirements for VoIP traffic?

A. Voice traffic must be marked "to DSCP EF.
B. Loss must be no more 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 2
How does an administrator make a Cisco IP phone display the last 10 digits of the calling number when the call is in the connected state, and also display the calling number in the E.164 format within call history on the phone?

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

Answer: A

NEW QUESTION 3
According to the QoS Baseline Model, drag and drop the applications from the left onto the Per-Hop Behavior values on the right.

voice	AF11
interactive video	CS2
bulk data	° EF
call-signaling	AF31/CS3
network management	AF41

A. Mastered
B. Not Mastered

Answer: A

Explanation:

voice	interactive video
interactive video	network management
bulk data	voice
call-signaling	call-signaling
network management	bulk data

NEW QUESTION 4
Refer to the exhibit.

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

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

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

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

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

Answer: B

NEW QUESTION 5
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 6
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 7
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 8
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 9
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 10
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 10
Refer to the exhibit.

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

An administrator configured a codec preference list with 0,122 and ILBC codecs. Which change must the administrator make in the dial-peer section of the configuration to use this list?

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

Answer: D

NEW QUESTION 13
Refer to the exhibit. Which two codec permutations should be transcoded by this dspfarm? (Choose two.)

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

Answer: CE

NEW QUESTION 14
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 18
An administrator is designing a new Cisco UCM for a company with many departments and firm structure on their communications policies. The administrator must make sure that these communication policies are reflected in the phone system setup. Certain departments cannot be accessed directly, even if they have dedicated DID numbers. Some phones, especially public phones, must not be able to dial international numbers Which type of function is configured to control which device is allowed to call another device in Cisco UCM?

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

Answer: A

NEW QUESTION 23
A remote office has a less-than-optimal WAN connection and experiences packet loss, delay and jitter. Which VoIP codec is used in this situation?

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

Answer: B

NEW QUESTION 27
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 31
If a phone needs to register with cucm1.cisco.com, which network service assists with the phone registration process?

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

Answer: D

Explanation:
According to the Cisco Community website1, the phone uses DNS to resolve the hostname of the CUCM server (cucm1.cisco.com) to its IP address. DNS is a network service that translates domain names into IP addresses.

NEW QUESTION 34
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 36
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 37
Which two access layer switches provide support to provide high-quality voice and take advantage of the full voice feature set. To provide high-quality voice and take advantage of the full voice feature set, which two access layer switches provide support? Choose two

A. Use multiple egress queues to provide priority queuing of RTP voice packet streams and the ability to classify or reclassify traffic and establish a network trust boundary.
B. Use 808.IQ trunking and 802.lp for proper treatment of Layer 2 CoS packet marking on ports with phones connected.
C. Implement IP RTP header compression on the serial interface to reduce the bandwidth requlreq per voice call on point-to-point links.
D. Deploy RSVP to improve VoIP QoS only where it can have a positive Impact on quality and functionality where there Is limited bandwidth and frequent network congestion.
E. Map audio and video streams of video calls (AF41 and AF42) to a class-based queue with weighted random early detection.

Answer: AB

NEW QUESTION 40
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 42
Which wildcard must an engineer configure to match a whole domain in SIP route patterns?

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

Answer: A

Explanation:
The asterisk (*) wildcard is used to match any sequence of characters, including an empty sequence. Therefore, it can be used to match any domain name in a SIP Route Pattern.
The other options are not correct because:

> C. !: The ! symbol is used to negate a character class.

> D. .: The . symbol is used to match any single character.

NEW QUESTION 46
A high-speed network is often configured with a five-class QoS model. Which classes are used in the model?

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

Answer: A

NEW QUESTION 48
Where in Cisco UCM is restrictions on audio bandwidth configured?

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

Answer: C

NEW QUESTION 53
Which endpoint feature is supported using Mobile and Remote Access through Cisco Expressway?

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

Answer: A

NEW QUESTION 56
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 61
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 64
An employee of company ABC just quit. The IT administrator deleted the employee's user id from the active directory at 10 a m. on March 4th The nightly sync occurs at 10 p.m. daily. The IT administrator wants to troubleshoot and find a way to delete the user id as soon as possible How is this issue resolved?

A. Wart until 10 pm on March 4th when the user is automatically removed from Cisco UCM.
B. Wait until 10 pm on March 5th when the user is automatically removed from Cisco UCM.
C. Wait until 3 15 a.
D. on March 6th for garbage collection to remove the user from Cisco UCM.
E. Wait until 315am on March 5th for garbage collection to remove the user from Cisco UCM.

Answer: C

NEW QUESTION 66
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 71
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 75

An administrator is configuring a new Cisco UCM with PSTN capabilities. Due to bandwidth constraints, audio compression is used on the codec. DTMF must work as expected because the customer is calling many call centers where the users must select options in the call. Where is DTMF out-of-band in a CCM 12.5 with SIP-based gateway configured?

A. in the DTMF setting under SIP profile on the Cisco Unified Border Element
B. in the dial peer on the Cisco IOS router
C. in regions on the Cisco UCM where the appropriate codec to use is set
D. in DTMF settings in the audio codec preference list under regions in the Cisco UCM

Answer: B

NEW QUESTION 80
What are two features of Cisco Expressway that the customer gets if Expressway-C and Expressway-E are deployed?(Choose two.)

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

Answer: AD

NEW QUESTION 84
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 89
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 93
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 95
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 100
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 103
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 108
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 111
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 113
What is the purpose of Mobile and Remote Access (MRA) in the Cisco UCM architecture?

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

Answer: D

NEW QUESTION 115
A user dials 9011841234567 to reach Vietnam. Which steps send the call to the PSTN provider as 011841234567?

A)

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

B)

    in the Calling Party Transformation Patterns section,
        configure the Pattern as 9.011841234567
        configure the Discard Digits as Predot 10-10-Dialing

C)

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

D)

    in the Called Party Transformation Pattern Configuration section,
        configure the Pattern as 9.011841234567
        configure the Discard Digits as Predot 10-10-Dialing

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

Answer: A

NEW QUESTION 118
An administrator uses the Cisco Unified Real-Time Monitoring Tool to investigate recent calls on a Cisco UCM cluster. The SIP trace for an on-net. direct-media call shows two 180 Ringing and two 11 BYE messages. Why are there multiples of each message type in the trace?

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

Answer: A

NEW QUESTION 122
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 124
Due to service provider restriction. Cisco UCM cannot send video in the SDR Which two options on Cisco UCM are configured to suppress video in the SDP in

outgoing invites? (Choose two.)

A. Add the audio forced command to voice service VoIP on the Cisco Unified Border Element.
B. Check the Retry Video Call as Audio on the SIP trunk.
C. Set Video Bandwidth in the Region settings to 0.
D. Change the Video Capabilities dropdown on the endpoint to Disabled.
E. Check the Send send-receive SDP in mid-call INVITE check box on the SIP trunk SIP profile.

Answer: CD

NEW QUESTION 128
An engineer implements QoS in the enterprise network. Which command is used to verity the classification and marking on a Cisco IOS switch?

A. show class-map interface GigabitEthernet 1/0/1
B. show policy-map interface GigabitEthernet 1/0/1
C. show access-lists
D. show policy-map

Answer: B

NEW QUESTION 130
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 135
An administrator configures the voicemail feature in a Cisco collaboration deployment. The user mailboxes must be configured when the Cisco Unity Connection server is configured. Which action accomplishes this task?

A. Configure a SIP integration with Cisco UCM to sync users.

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

Answer: C

NEW QUESTION 139
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 141
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 145
What are two functions of Cisco Expressway in the Collaboration Edge? (Choose two.)

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

Answer: CE

NEW QUESTION 148
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 150
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 153
Refer to the exhibit.

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

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

A.

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

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

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

Answer: C

NEW QUESTION 157
A company wants to provide remote users with access to its on-premises Cisco collaboration features. Which components are required to enable Cisco Mobile and Remote Access for the users?

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

Answer: C

NEW QUESTION 158
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 162
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 167
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 171
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 175
An engineer is configuring IP telephony. The network relies on DHCP to provide TFTP server addresses to the endpoints. Policy requires the endpoints to receive two server addresses. Which DHCP option must be configured?

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

Answer: C

NEW QUESTION 177
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 178
When multiple potential patterns are present, which two things are considered when Cisco UCM selects a destination pattern? (Choose two.)

A. The pattern matches the shortest explicit prefix.
B. The pattern does not match the dialed string.
C. The pattern represents the smallest number of endpoints.
D. The pattern matches the dialed string.
E. The pattern represents the largest number of endpoints.

Answer: AD

NEW QUESTION 183
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.
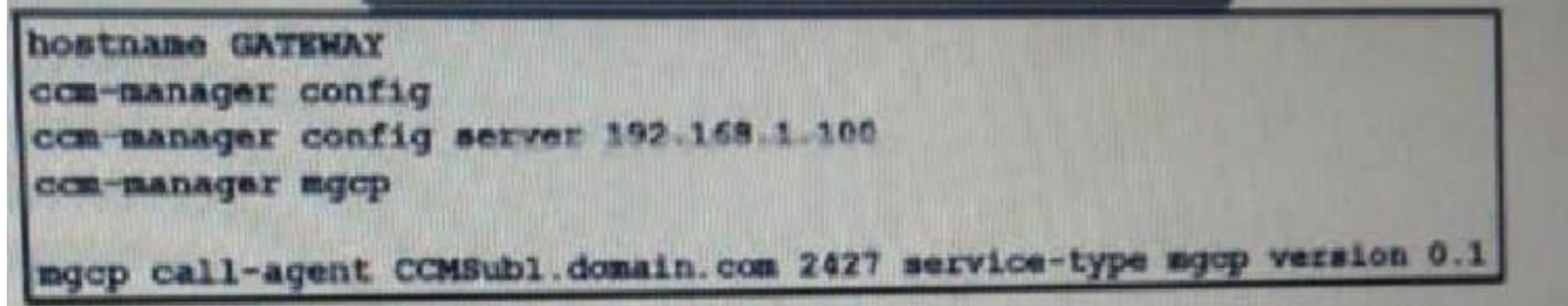
NEW QUESTION 188
Refer to the exhibit.


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

A. dtmf-relay sip-kvmi rtp-nte
B. dtmf- relay rtp-nte sip-kpml
C. dtmf-relay sip-kgml rtp-inband
D. dtmf-relay rtp-inband sip-kvmi

Answer: B

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

NEW QUESTION 196
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 197
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
Refer to the exhibit.
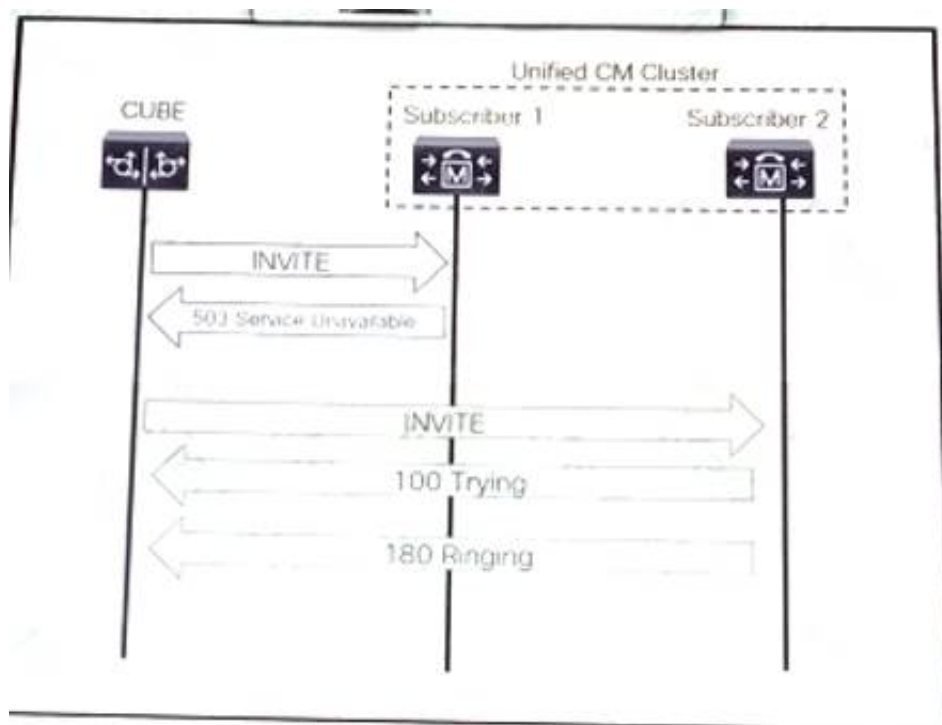
Cisco Unified element is attempting to establish a call with Subcribers1, but the call fails. Cisco Unified Border Element then retries the same call with Subcribers2, and the call proceeds normally.
Which action resolves the issue?

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 203
Refer to the exhibit.

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 206
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 208
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 211
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 212
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 216
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 218
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.
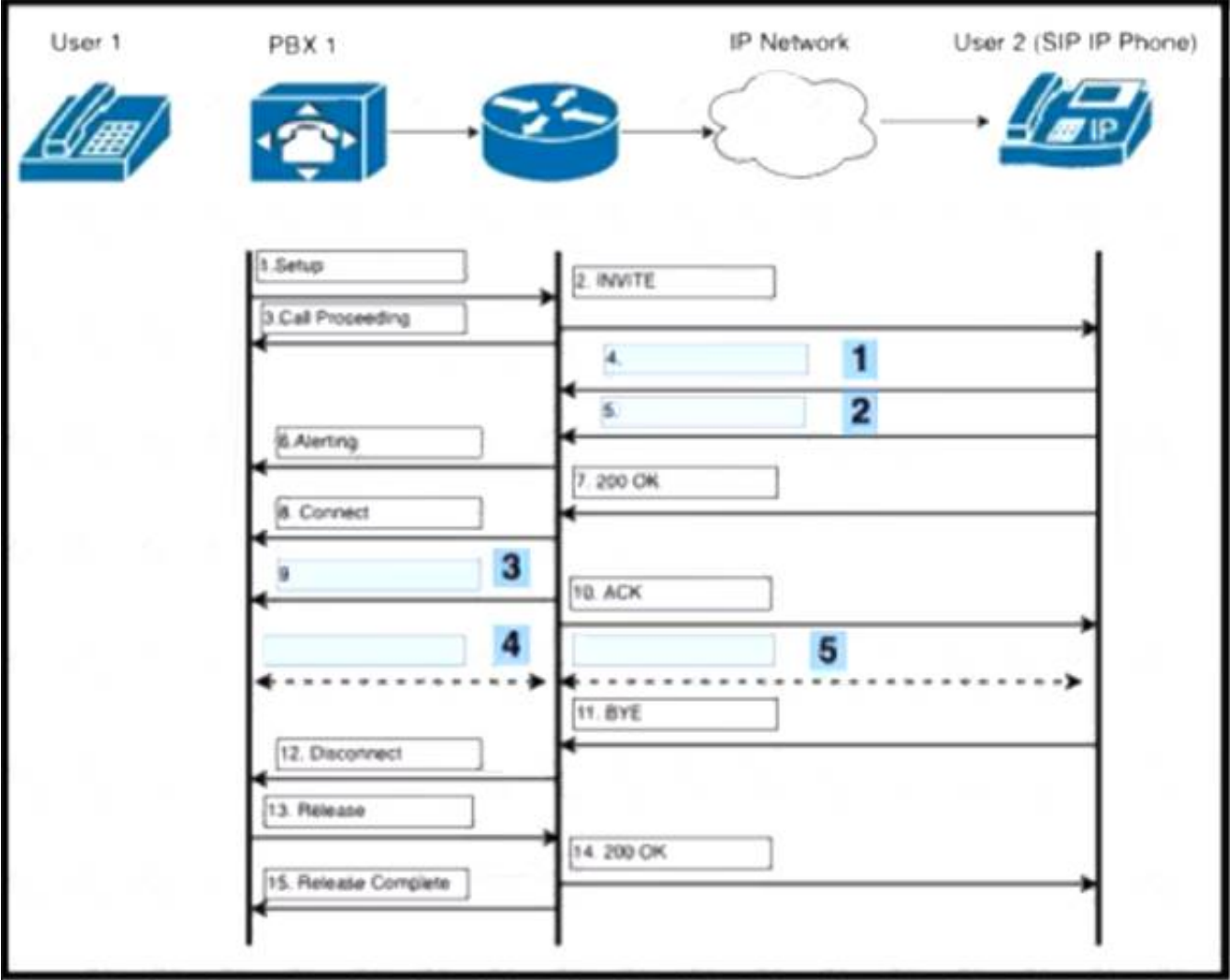
Answer: D

NEW QUESTION 221
End users report bad video quality and voice choppiness on Cisco Collaboration endpoints. The engineer changed the device pool the users were in but did not correct the problem. Which action should be taken to troubleshoot this issue?

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

Answer: D

NEW QUESTION 222

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 223
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 226
Refer to the exhibit.

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

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

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

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

Answer: BD

NEW QUESTION 227
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 231
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.
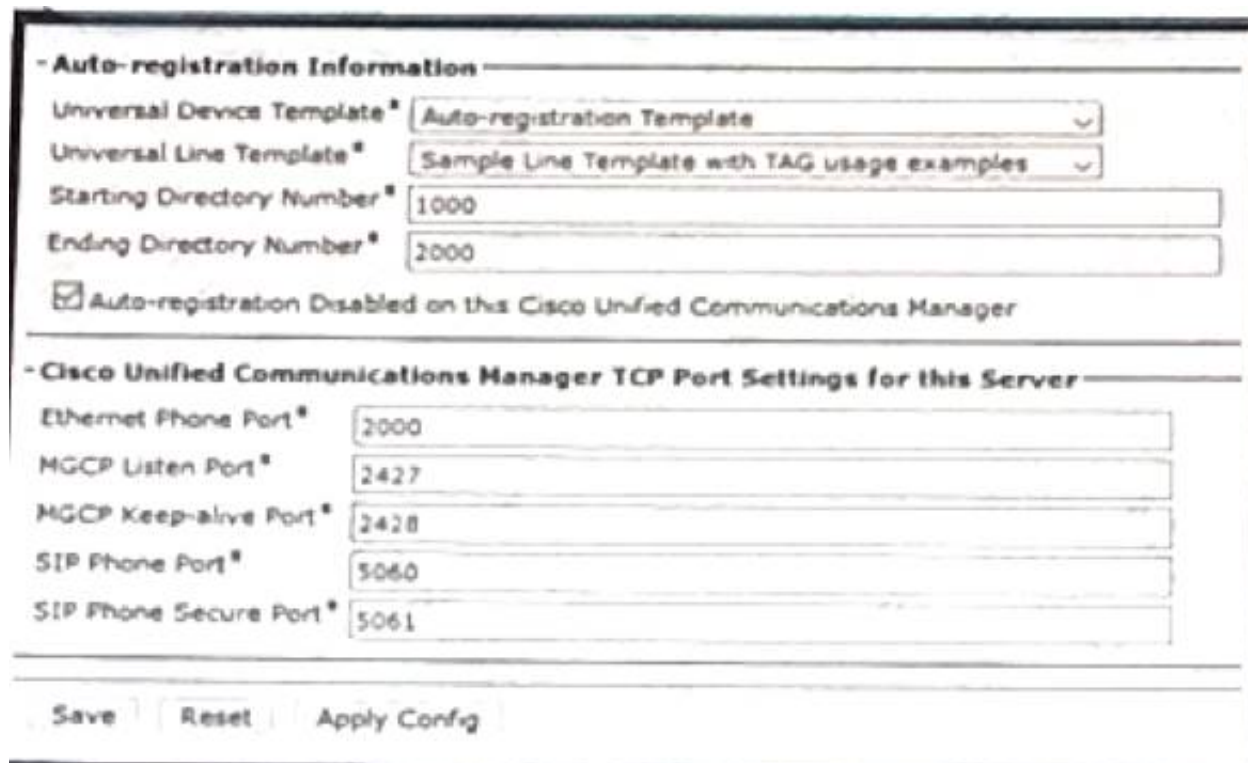
Answer: BE

NEW QUESTION 232
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.

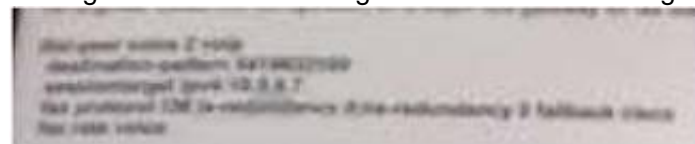**Answer:** C

NEW QUESTION 233
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 237
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 238
A Cisco UCM administrator wants to enable the Self-Provisioning feature for end users. Which two prerequisites must be met first? (Choose two.)

A. End users must have a secondary extension.
B. Cisco Extended Functions service must be running
C. End users must belong to Standard CCM Admin Users group, the Standard CCM End Users group, and the Standard CCM Self-Provisioning group.
D. End users must have a primary extension.
E. End users must be associated to a user profile or feature group template that includes a universal line template and universal device template.

Answer: DE

NEW QUESTION 242
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 244
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 247
Which IP Precedence value is used to classify a call signalling packet?

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

Answer: D

NEW QUESTION 251
Refer to the exhibit.

```
isdn switch-type primary-ni
controller t1 0/1/0
framing esf
linecode b8zs
pri-group timeslots 1-10
```
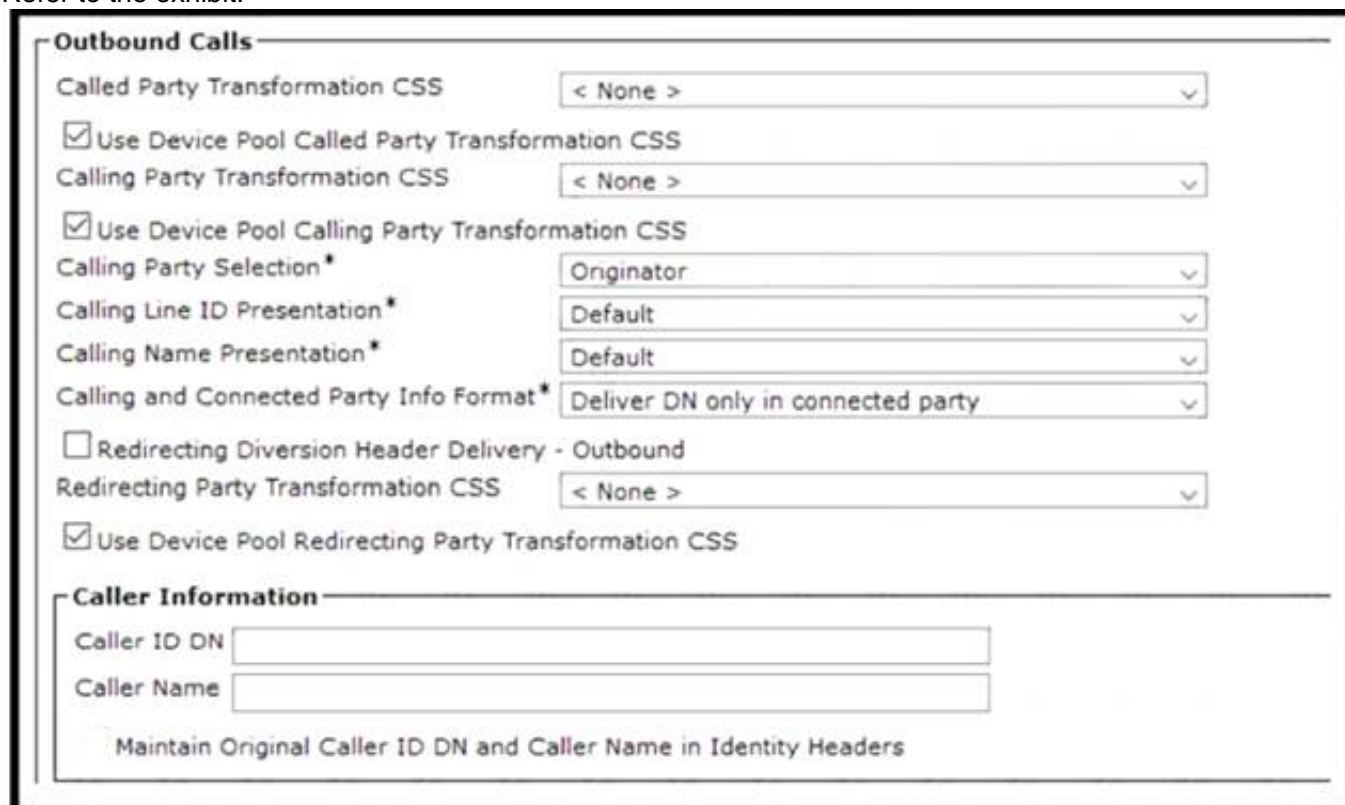
An engineer configures ISDN on a voice gateway. The provider confirms that the PRI is configured with 10 channels the engineer ordered and is working from the provider side, but the engineer cannot get a B-channel to carry voice. The rest of the configuration for the serial interface and voice network 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.
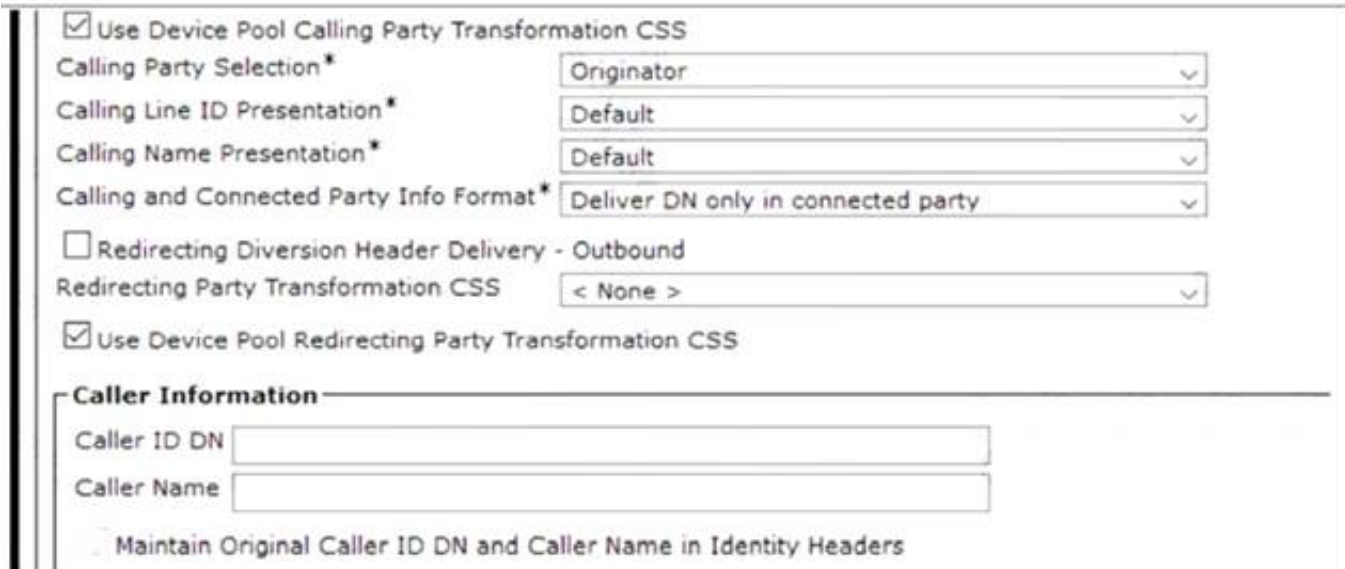D. The engineer must manually revert the order of using the channels.

Answer: A

NEW QUESTION 252
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 257
Which QoS marking is used when an administrator configures voice call signaling?

A. AF41
B. CS3
C. CS4
D. EF

Answer: B

NEW QUESTION 261
An engineer troubleshoots outbound can failure on an ISDN-PRI circuit. The engineer ts suspecting the 'Incomplete Destination". Which debugs or commands are run in the voice gateway to troubleshoot the issue?

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

Answer: C

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

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

NEW QUESTION 263
A Company s employees have been complaining that they have been unable to select options on the internal IVR of the help desk. IT support has been given Cisco UCM traces and below is the snippet of the SDP of the INVITE packet.

```
m=audio 25268 RTP/AVP 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=ptime:20
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

How is this issue resolved?

A. Configure CODEC for G.729.
B. Configure DTMF for KPML.
C. Configure CODEC for G.722.
D. Configure DTMF for RFC 2833.

Answer: B

NEW QUESTION 265
What is the purpose of a hybrid Local Gateway?

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

Answer: D

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

NEW QUESTION 266
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 267
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 268
Refer to the exhibit.
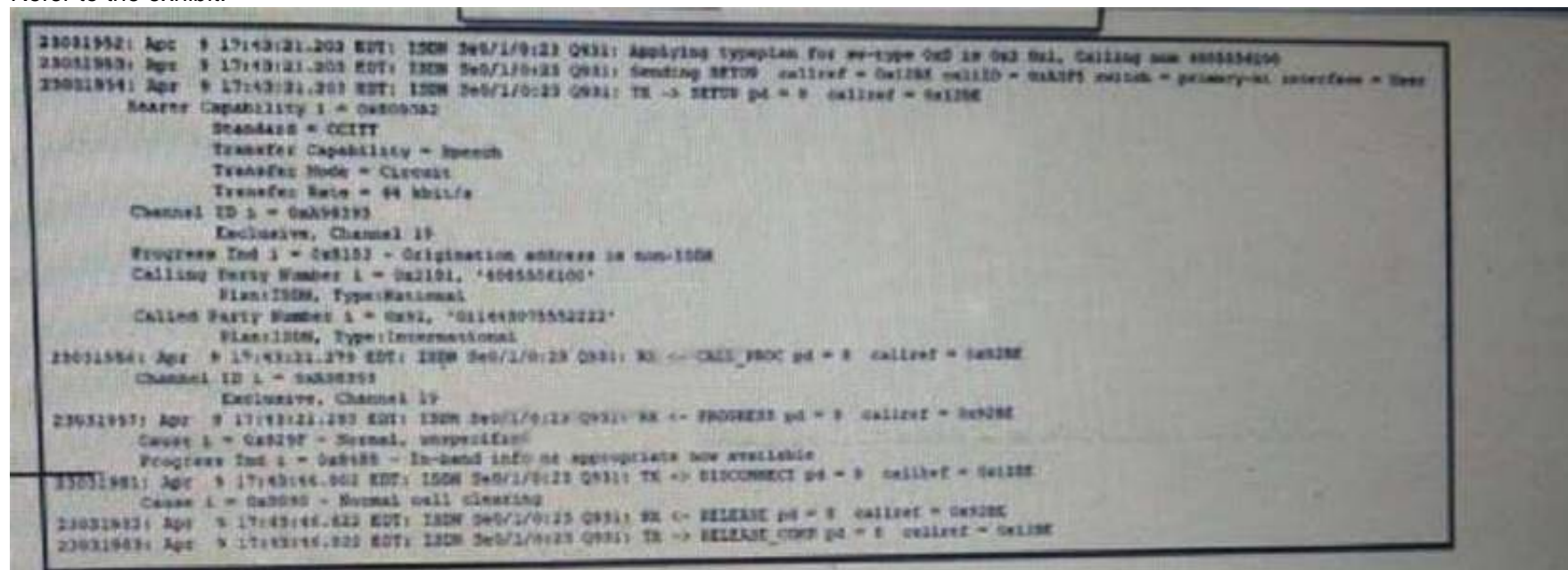
A call to an international number has failed. Which action corrects this problem?

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 271
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 274
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 275
An administrator configures Cisco UCM to use UDP for SIP signaling and finds that an endpoint cannot make calls. Which action resolves this issue?

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

Answer: D

NEW QUESTION 277
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 278
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 279
Why does Cisco UCM use DNS?

A. It provides certificate-based security for media
B. It resolves FQDN to IP address resolution for trunks
C. it connects endpoints to single sign-on services.
D. It provides SRV resolution to the endpoints registered
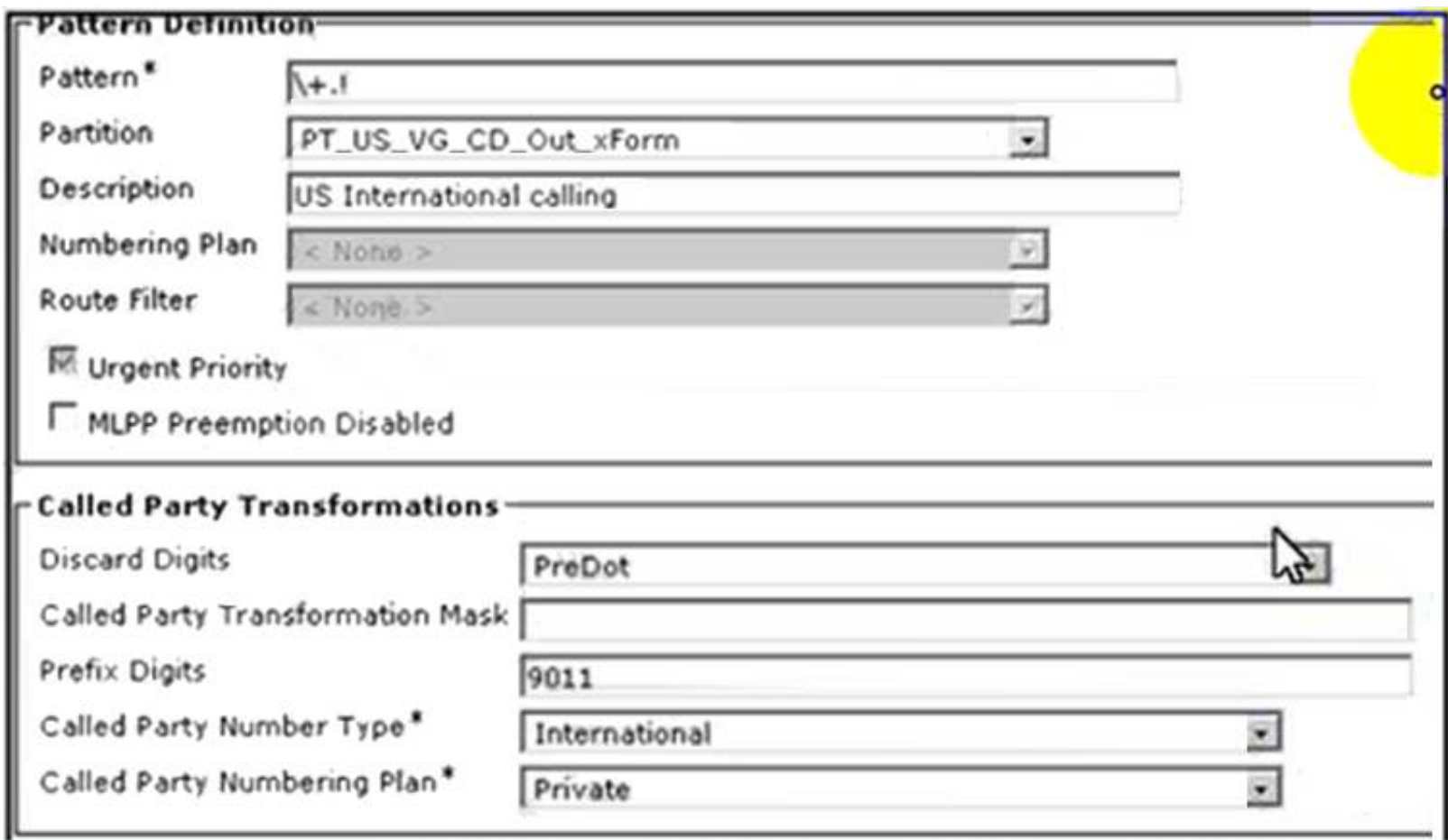
Answer: D

NEW QUESTION 283
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.

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. **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*	Unknown
Called Party Numbering Plan*	Unknown

Answer: C

NEW QUESTION 285
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 286
Refer to the exhibit.
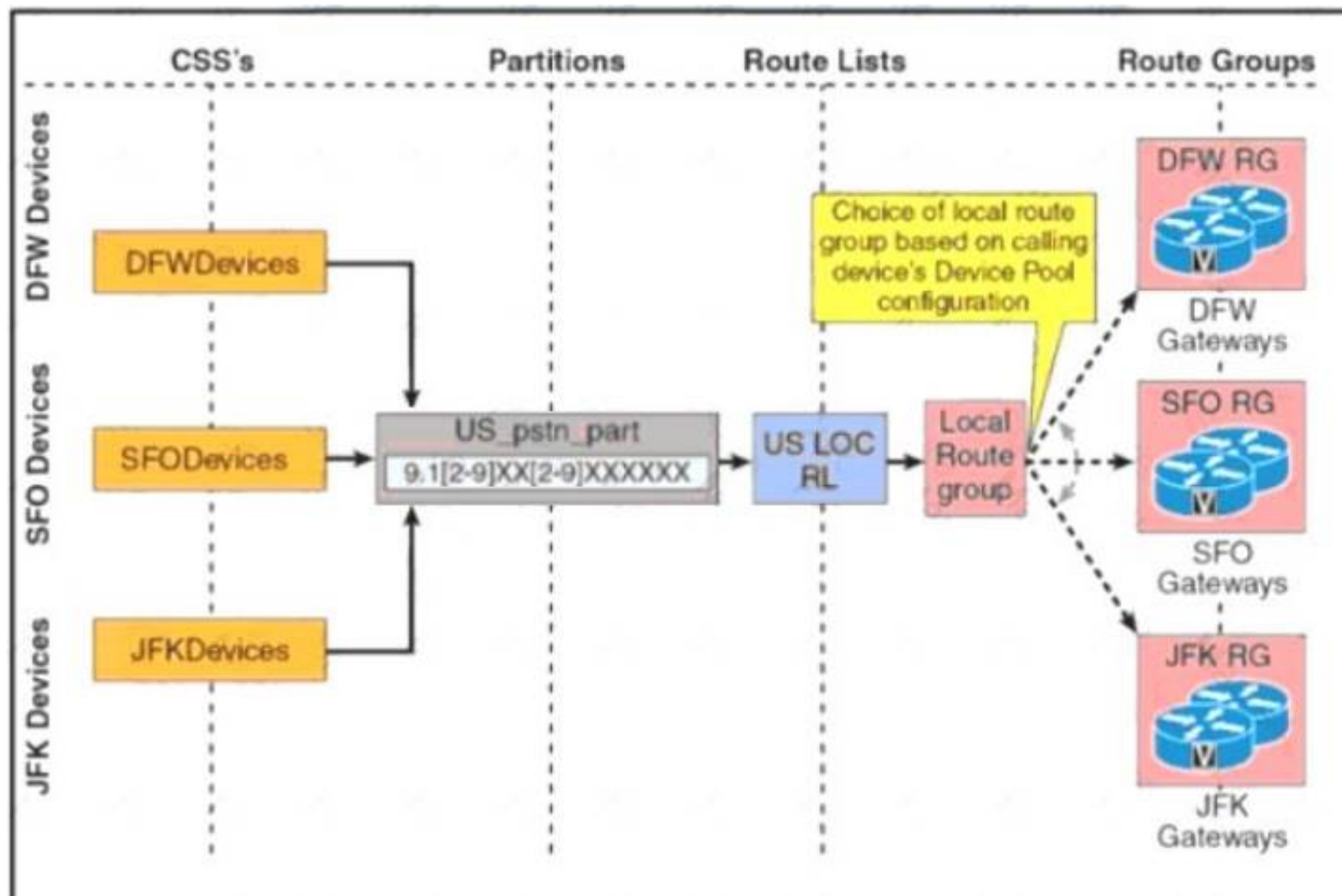
A user takes a phone from San Francisco to New York for a short reassignment. The phone was set up to use the San Francisco device pool, and device mobility is enabled on the Cisco UCM. The user makes a call that matches a route pattern in a route list that contains the Standard Local Route Group. To 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 291
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 293
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 298
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 300
Why isn't an end user s PC device in a QoS trust boundary included?

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

Answer: B

NEW QUESTION 302
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 304
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
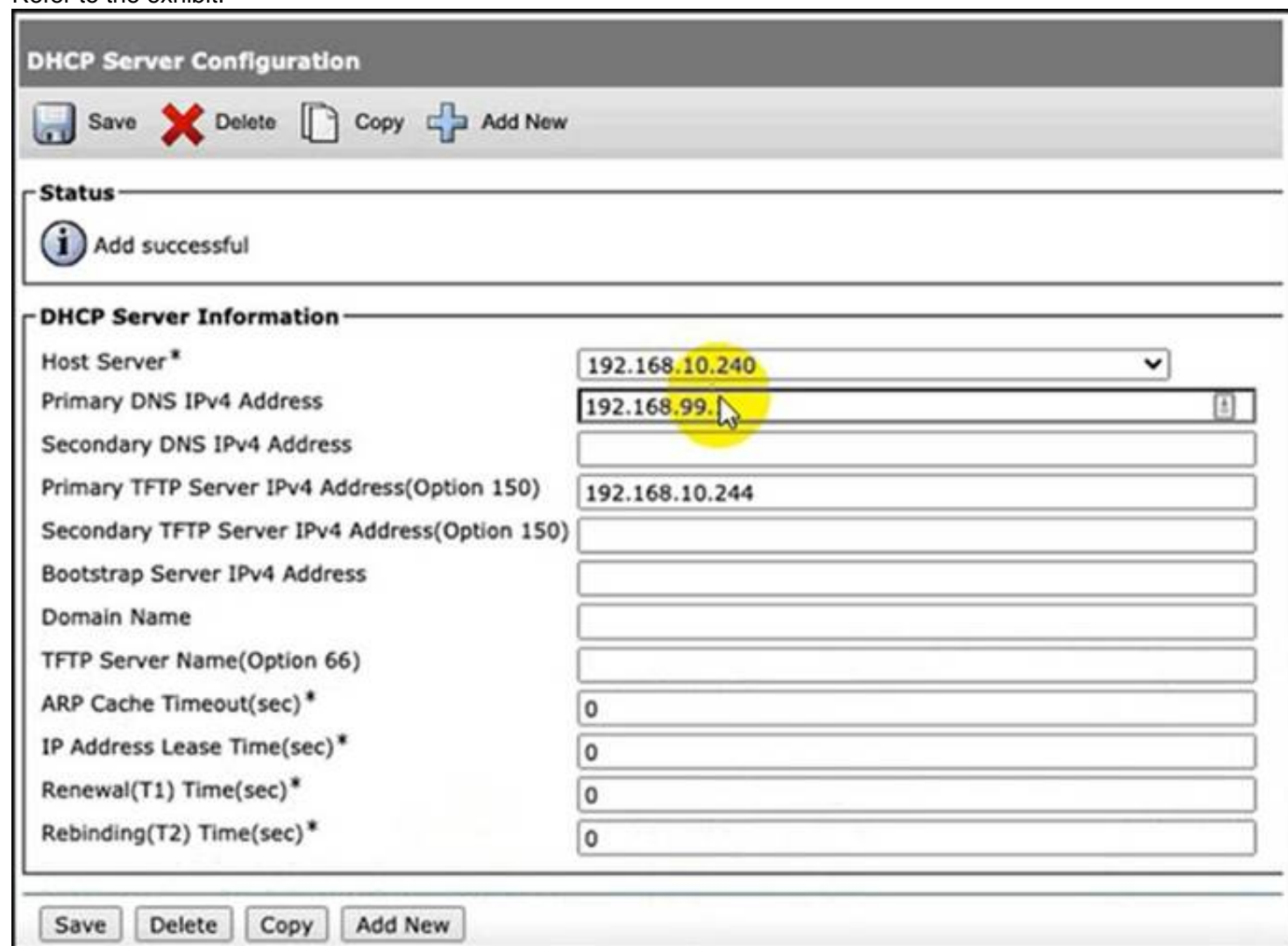
Explanation:
Call forwarding is a feature that allows users to forward incoming calls to another number. This can be useful in a number of situations, such as when a user is not available to take a call, or when a user wants to forward calls to a different number during certain times of the day.
Call 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 309

Refer to the exhibit.

DHCP Server Configuration

Save   Delete   Copy   Add New

Status
(i) Add successful

DHCP Server Information

Host Server*	192.168.10.240
Primary DNS IPv4 Address	192.168.99.
Secondary DNS IPv4 Address	
Primary TFTP Server IPv4 Address(Option 150)	192.168.10.244
Secondary TFTP Server IPv4 Address(Option 150)	
Bootstrap Server IPv4 Address	
Domain Name	
TFTP Server Name(Option 66)	
ARP Cache Timeout(sec)*	0
IP Address Lease Time(sec)*	0
Renewal(T1) Time(sec)*	0
Rebinding(T2) Time(sec)*	0

Save   Delete   Copy   Add New

A collaboration engineer configures Cisco UCM to act as a DHCP server. What must be done next to configure the DHCP server?

A. Restart the Cisco DHCP Monitor Service under Cisco Unified Serviceability
B. Add the new DHCP server to the primary DNS server
C. Restart the TFTP service under Cisco Unified Serviceability.
D. Add a DHCP subnet to the DHCP server under Cisco UCM Administration.

Answer: D

NEW QUESTION 310

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 313

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 316
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 317
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 319
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 324
Refer to the exhibit.

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

000143: *Apr 23 19:41:49.050: MGCP Packet sent to 192.168.100.101:2427--->
200 4
I:
X: 2
L: p:10-20, a:PCMU;PCMA;G.nX64, b:64, e:on, gc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
L: p:10-220, a:G.729;G.729a;G.729b, b:8, e:on, gc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
L: p:10-110, a:G.726-16;G.728, b:16, e:on, gc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
L: p:10-70, a:G.726-24, b:24, e:on, gc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
L: p:10-50, a:G.726-32, b:32, e:on, gc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
L: p:30-270, a:G.723.1-H;G.723;G.723.1a-H, b:6, e:on, gc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
L: p:30-330, a:G.723.1-L;G.723.1a-L, b:5, e:on, gc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
M: sendonly, recvonly, sendrecv, inactive, loopback, conttest, data, netwloop, netwtest
<---
```

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

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

Answer: D

NEW QUESTION 329
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 330
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]XXXXXX called Party Transformation Pattern

Answer: B

NEW QUESTION 335
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 338
Which DHCP option must be set up tor new phones to obtain the TFTP server IP address?

A. option 66
B. option 15
C. option 6
D. option 120

Answer: A

NEW QUESTION 339
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 343
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 345
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 348
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 351

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 352

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 353

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 357

After an engineer implements the FAC and CMC features together, users report that calls take almost one minute to complete and that they occasionally hear the reorder tone. Which two actions address this issue?(Choose two)

A. Adjust the T302 timer from the default of 15 seconds to 5 seconds to shorten the interdigit timer
B. Change the code if the problem persists
C. Advise the user to hang up and try again
D. Do not wait for the tones immediately dial the FAC and CMC
E. Advise the user to press the"#"button after dialing the FAC and CMC codes

Answer: AB

NEW QUESTION 359

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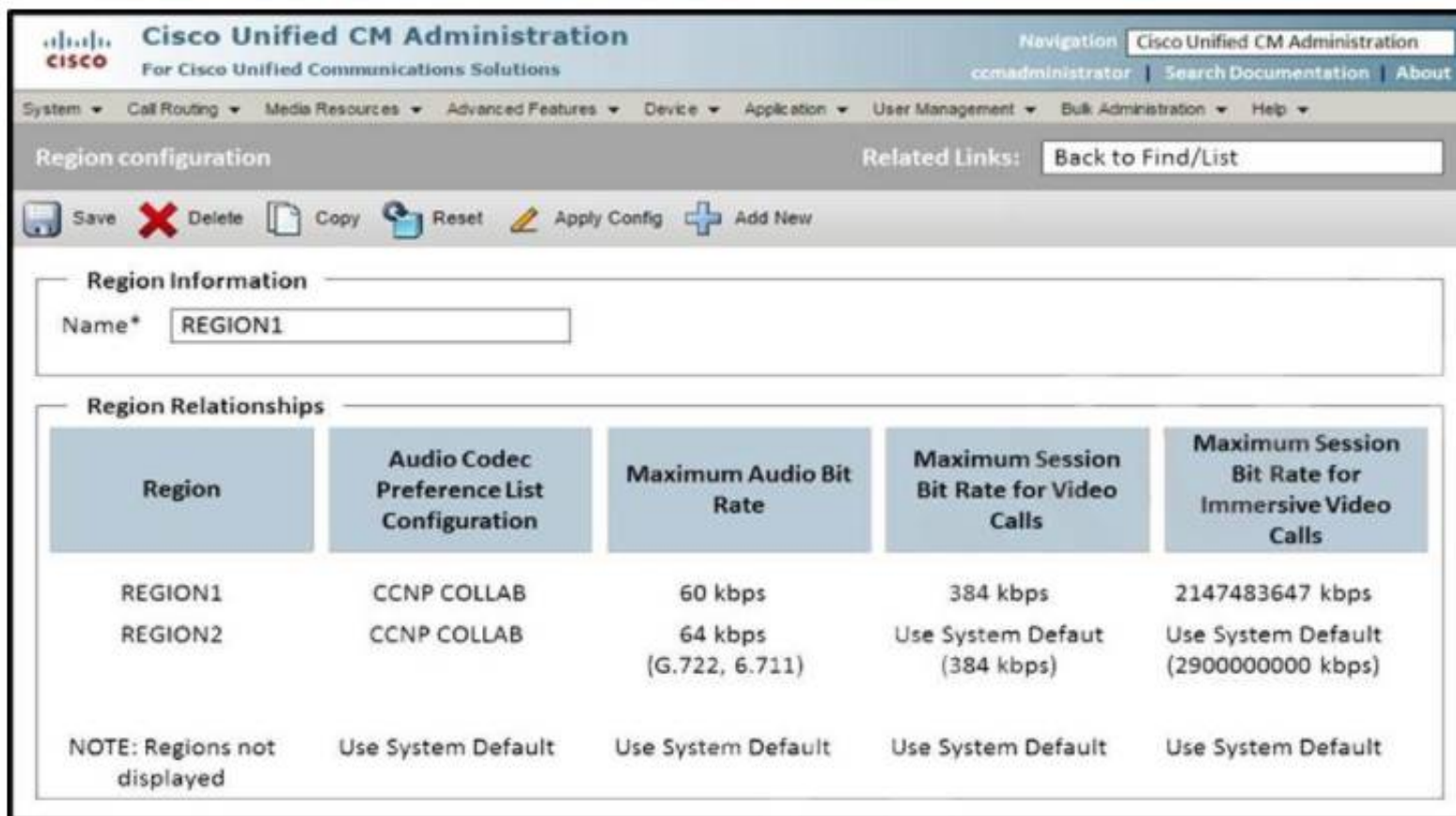
```
Router# configure terminal
Router(config)# interface gigabitethernet 5/1
Router config-if)# description Cube E41.228-0097
```

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 362
Refer to the exhibit.

| Cisco Unified CM Administration | | | Navigation | Cisco Unified CM Administration |

For Cisco Unified Communications Solutions — ccmadministrator | Search Documentation | About

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Region configuration Related Links: Back to Find/List

💾 Save ❌ Delete 📋 Copy 🔄 Reset ✏️ Apply Config ➕ Add New

Region Information

Name* REGION1

Region Relationships

Region	Audio Codec Preference List Configuration	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
REGION1	CCNP COLLAB	60 kbps	384 kbps	2147483647 kbps
REGION2	CCNP COLLAB	64 kbps (G.722, 6.711)	Use System Defaut (384 kbps)	Use System Default (2900000000 kbps)
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

| Cisco Unified CM Administration | | | Navigation | Cisco Unified CM Administration |

For Cisco Unified Communications Solutions — ccmadministrator | Search Documentation | About

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Audio Codec Preference List Configuration Related Links: Back to Find/List

💾 Save ❌ Delete 📋 Copy ➕ Add New

Status

ℹ️ Status: Ready

Audio Codec Preference Information

Name* CCNP COLLAB

Description* CCNP COLLAB

Codecs in List
```
G.722 48k
G.711 U-Law 64k
G.729 8k
G.711 A-Law 56k
```

An engineer is troubleshooting this video conference issue:
*A video call between a Cisco 9971 in Region1 and another Cisco 9971 in Region1 works.
*As soon as the Cisco 9971 in Region1 conferences in a Cisco 8945 in Region2. the 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 366
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 371
......

Relate Links

100% Pass Your 350-801 Exam with Exambible Prep Materials

https://www.exambible.com/350-801-exam/

Contact us

We are proud of our high-quality customer service, which serves you around the clock 24/7.

Viste - https://www.exambible.com/