Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

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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
Refer to the exhibit.

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

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

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

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

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

Answer: BC

NEW QUESTION 3
Refer to the exhibit.
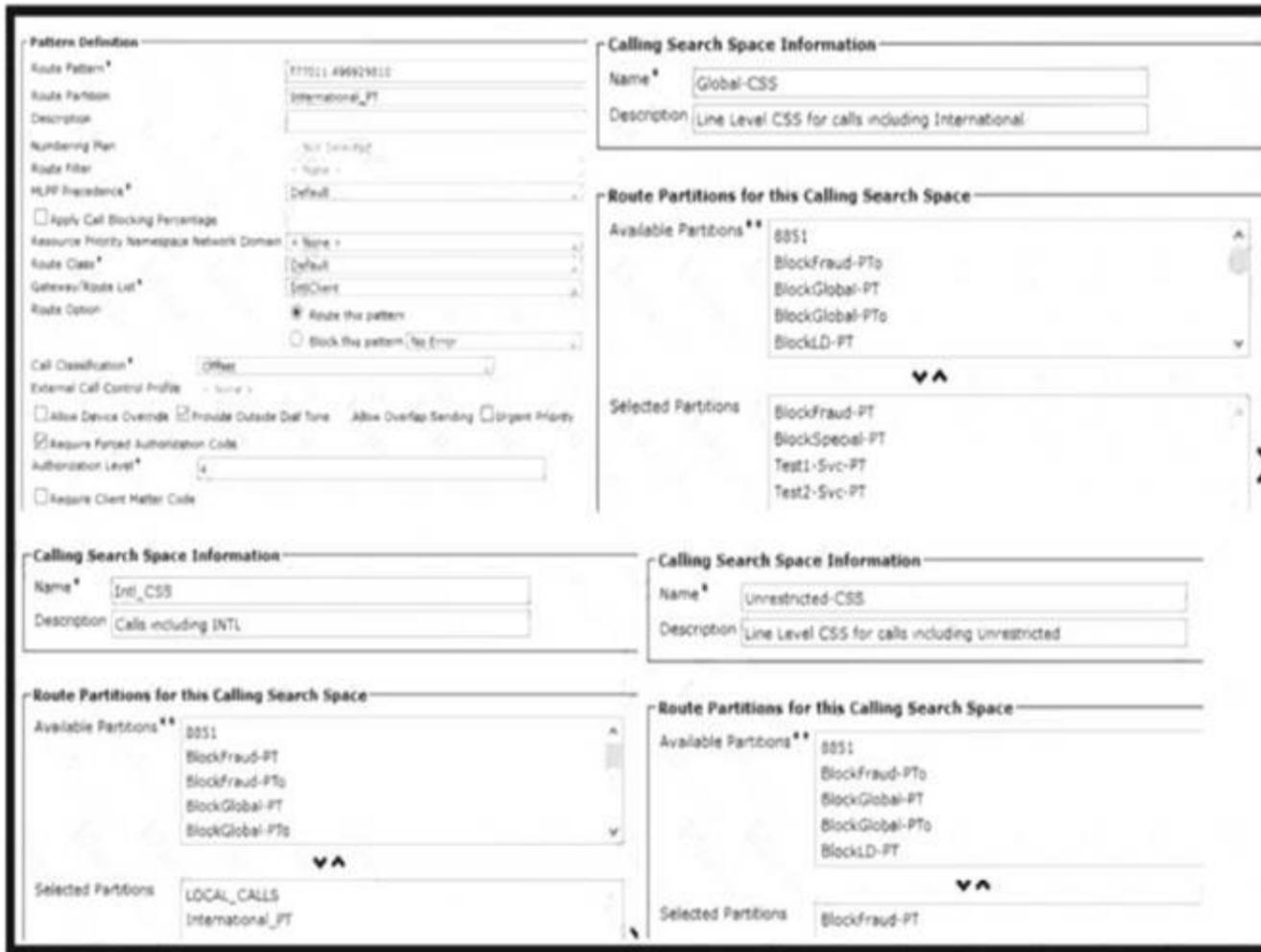
How must the +E.164 translation pattern be configured to reach international number 496929810?

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

A. Mastered
B. Not Mastered

Answer: A

Explanation:

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

NEW QUESTION 5
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 6
Which two configuration elements are part of the Cisco UCM toll-fraud prevention?(Choose two.)

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

Answer: AE

Explanation:
The following are the configuration elements that are part of the Cisco UCM toll-fraud prevention:

➤ Feature control policy - This policy controls the features that are available to users. For example, you can use this policy to prevent users from making international calls.

➤ Calling Search Space - This space defines the numbers that users can call. For example, you can use this space to prevent users from calling premium-rate numbers.

NEW QUESTION 7
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 8
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 9
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
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 10
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 15
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 18
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 23
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 26
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 31
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 33
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 36
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 41
When designing the capacity for a Cisco UCM 12.x cluster, an engineer must decide which VMware template will be used for each node. What is the lowest number of users supported in a template and the highest number of users in a template?

A. 750 and 15.000 users
B. 750 and 10.000 users
C. 500 and 10.000 users
D. 1000 and 10.000 users

Answer: D

NEW QUESTION 46
Where in Cisco UCM is restrictions on audio bandwidth configured?

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

Answer: C

NEW QUESTION 51
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 53
Which Webex Calling construct is used to organize calling features within a physical site?

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

Answer: B

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

NEW QUESTION 55
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 57
An administrator is trying to change the default LINECODE for a voice ISDN T1 PRI. Which command makes this change?

A. linecode ami
B. linecode b8zs
C. linecode hdb3
D. linecode esf

Answer: A

NEW QUESTION 62
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 64
Refer to the exhibit.

```
dial-peer voice 10 voip
         destination-pattern 1…
         session target ipv4:10.1.1.1
         no vad
```

An engineer configures a VoIP dial peer on a Cisco gateway. Which codec is used?

A. G711alaw
B. No codec is used (missing codec command)
C. G.711ulaw
D. G729r8

Answer: D

NEW QUESTION 67
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 72
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 74
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 77
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 80
Refer to the exhibit
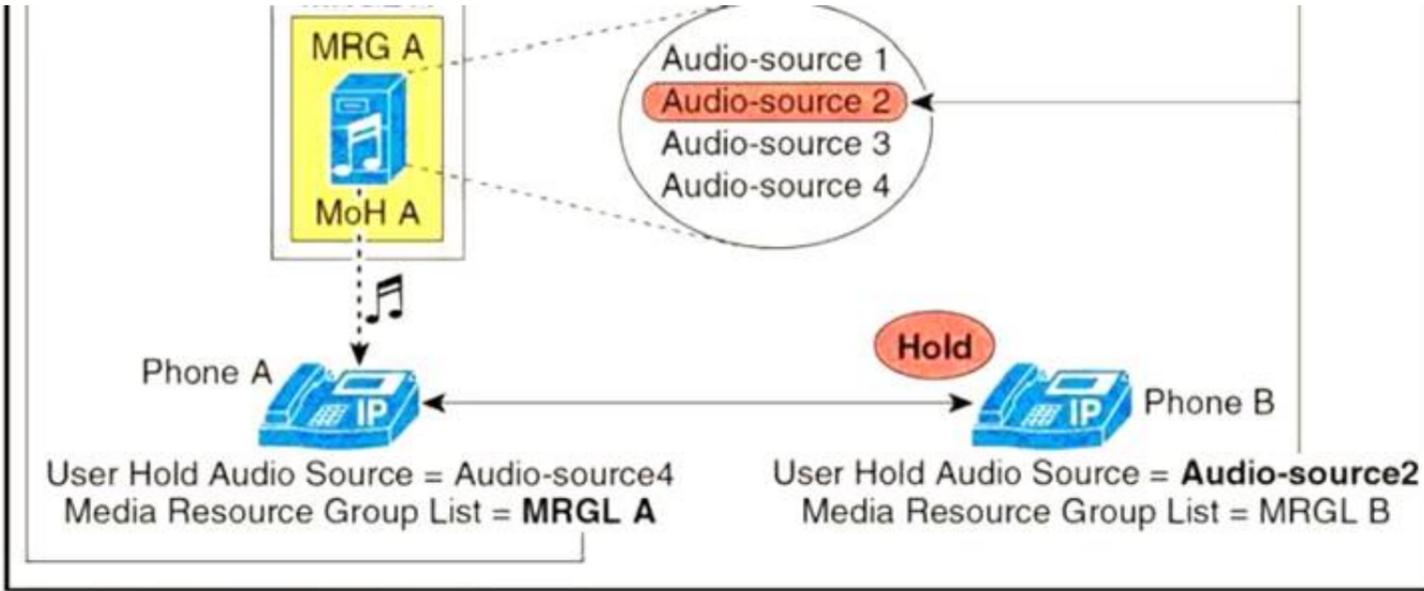
There is a call flow between Phone A and Phone B Phone B (holder) places Phone A (holder) on hold Which MRGL and Audio Source are played to Phone A?

A. MRGL A and Audio Source 4
B. MRGL B and Audio Source 4
C. MRGL A and Audio Source 2
D. MRGL B and Audio Source 2

Answer: C

NEW QUESTION 83
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 86
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 91
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 96
An engineer is notified that the Cisco TelePresence MX800 that is registered in Cisco Unified communications Manager shows an empty panel, and the Touch 10 shows a corresponding icon with no action when pressed. Where does the engineer go to remove the inactive custom panel?

A. The phone configuration page in CUCM Administration
B. The SIP Trunk security profile page in CUCM Administration
C. The software Upgrades page in CUCM OS Administration
D. The In-Room control Editor on the webpage of the MX800

Answer: D

NEW QUESTION 100
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 101
An engineer roust deploy the Cisco Wet*x app to a Windows Virtual Desktop Infrastructure environment that has a roaming database named spark roaming_store stored In a user's AppData\Roaming directory, Which two command line arguments must be used when running the installer? (Choose two.)

A. ALLUSERS=0
B. ENABLEVDI=1
C. ALLUSERS=1
D. ENABLEVDI=2
E. ROAMINGENABLED=1

Answer: BE

Explanation:
The Cisco Webex app can be installed on a Windows Virtual Desktop Infrastructure (VDI) environment by using the following command-line arguments:
≫ ENABLEVDI=1 - This argument enables VDI mode for the Webex app.
≫ ROAMINGENABLED=1 - This argument enables roaming for the Webex app.
The ALLUSERS argument is not required when installing the Webex app on a VDI environment. The ENABLEVDI argument must be set to 1, and the ROAMINGENABLED argument must be set to 1.
The following is an example of the command that can be used to install the Webex app on a VDI environment:
Code snippet
msiexec /i WebexApp.msi ENABLEVDI=1 ROAMINGENABLED=1

NEW QUESTION 105
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 108
An engineer wants to manually deploy a CISCO Webex DX80 Video endpoint to a remote user. Which type of provisioning is configured on the endpoint?

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

Answer: D

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

➤ Automatically, using the Cisco Unified Communications Manager (CUCM) or Cisco Video Communication Server (VCS)

➤ Manually, using the Edge provisioning mode

The Edge provisioning mode is used when the endpoint is not connected to the CUCM or VCS. In this mode, the endpoint is configured with the necessary settings, such as the IP address, SIP/H.323 parameters, and time and date.
The Cisco Unified Border Element (Cisco UBE) is a network element that provides security and call control for IP telephony networks. The Cisco Unity Connection is a unified messaging system that provides voicemail, email, and fax services. The Cisco Meeting Server is a video conferencing system that provides high-quality video and audio conferencing.

NEW QUESTION 111
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 114
Which DiffServe PHB preserves backward compatibility with any IP precedence scheme?

A. expedited forwarding
B. class selector
C. assured forwarding
D. default

Answer: B

NEW QUESTION 116
SIP proxies have operations defined in RFC 3261 and supporting extensions. Though no IETF RFC completely defines how SBCs must function. SBCs evolved over the years.
Which two operations demonstrate the high-level differences between SBCs and SIP proxies? (Choose two.)

A. Stateful proxies are context-aware and can terminate communication sessions by themselves
B. SIP proxies add a Via header and optionally a Record-Route header, and the rest of the headers are left untouched
C. SBCs can modify headers such as To, From, Contact, and Call-D.It can introduce new headers into the SIP message
D. SBCs are capable of interworking completely different protocols to set up, modify, and tear down communication session
E. It includes SIP, H.323,and MGCP protocols
F. SIP proxies are SDP-aware and can change the SDP bodies

Answer: BD

NEW QUESTION 117
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 119
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 123
On a cisco catalyst switch which command is required to send CDP packets on a switch port that configures a cisco IP phone to transmit voice traffic in 802.1q frames tagged with the voice VLAN ID 221?

A. Device(config-if)# switchport access vlan 221

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

Answer: D

NEW QUESTION 126
Refer to the exhibit.
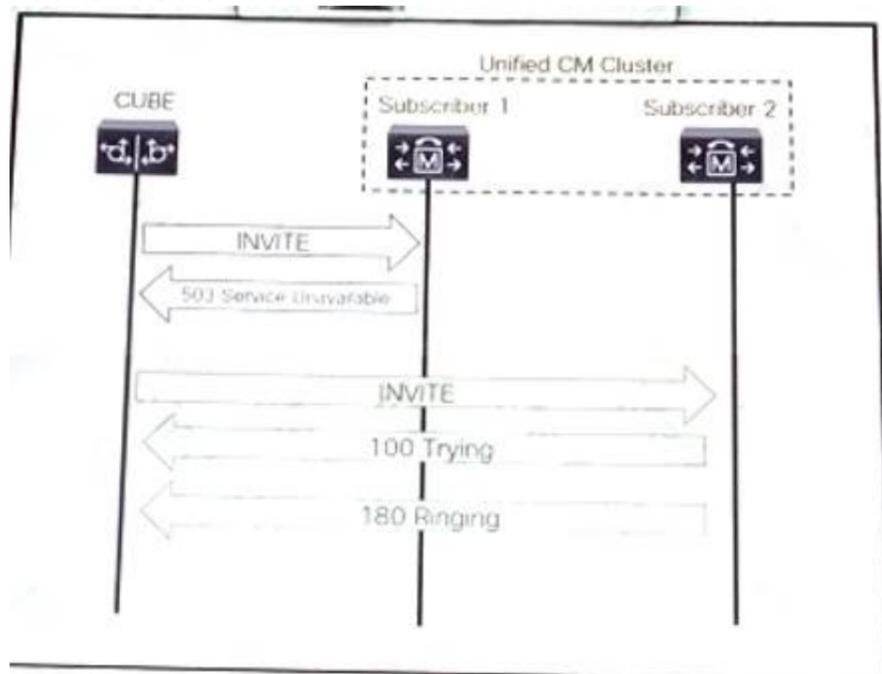
Cisco Unified element is attempting to establish a call with Subcribers1, but the call fails. Cisco Unified Border Element then retries the same call with Subcribers2, and the call proceeds normally.
Which action resolves the issue?

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 131
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 135
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 139
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 142
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 145
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 150
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 152
Refer to the exhibit.
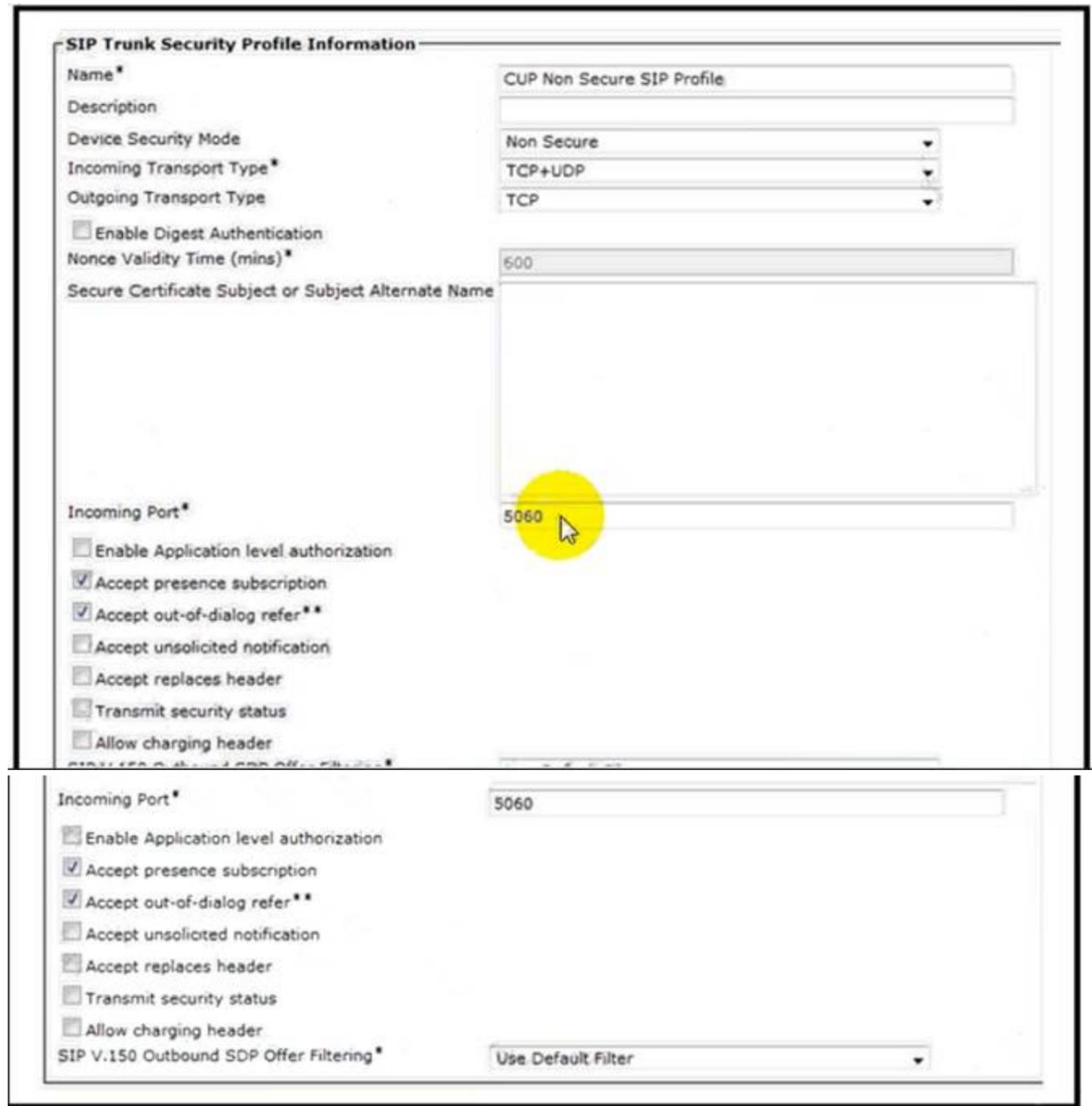
A collaboration engineer is configuring the Cisco UCM IM and Presence Service. Which two steps complete the configuration of the SIP trunk security profile? (Choose two.)

A. Check the box to enable application-level authorization.
B. Check the box to allow charging header.
C. Check the box to accept unsolicited notification.
D. Check the box to transmit security status.
E. Check the box to accept replaces header.

Answer: CE

NEW QUESTION 157
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 158
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 160
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 161
Refer to the exhibit.
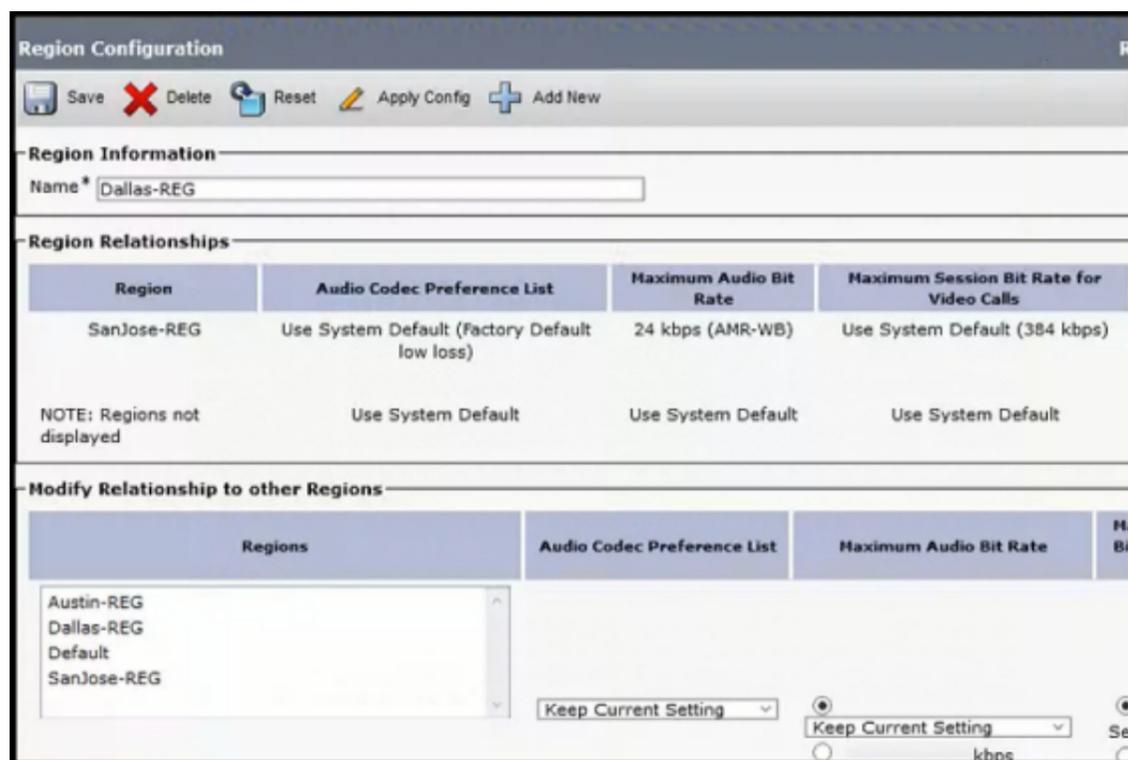
Which codec should an engineer select for a call made between "Dallas-REG' & "Austin-REG'?

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 166
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 170
Cisco UCM delays routing of a call during digit analysis with an overlapping dial plan. How long is the default wait time?

A. 5 seconds
B. 10 seconds
C. 15 seconds
D. 20 seconds

Answer: C

NEW QUESTION 175
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 177
When a remote office location is set up with limited bandwidth resources, which codec would allow the most voice calls with the limited bandwidth?

A. G.722
B. G.711
C. G.729
D. G.723

Answer: C

NEW QUESTION 182
When setting a new primary DNS server in the Cisco UCM CLI what is required for the change to take affect?

A. restart of CallManager service
B. restart of DirSync service
C. restart of the network service
D. restart of TFTP service

Answer: C

NEW QUESTION 183
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 188
Refer to the exhibit.
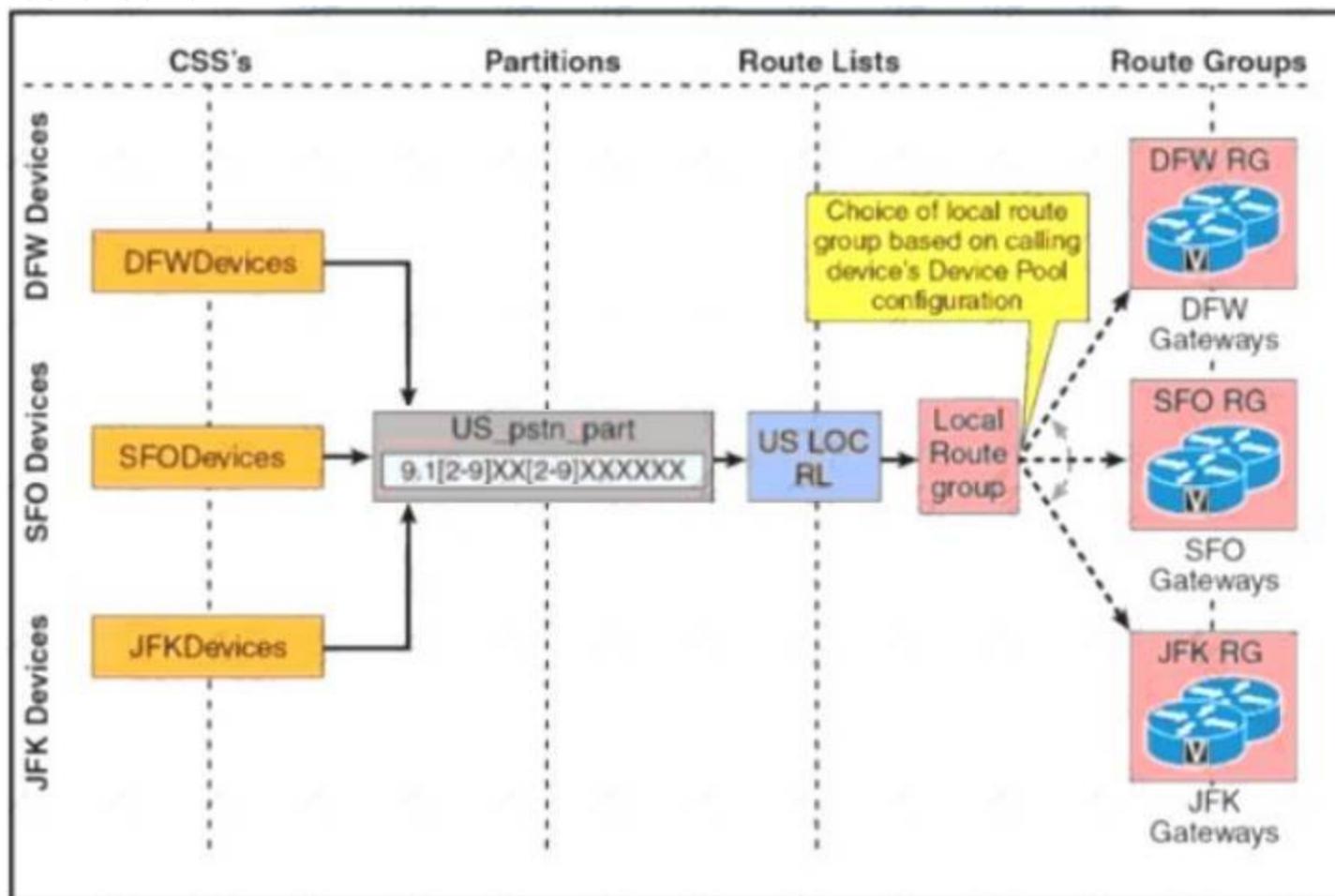
A user takes a phone from San Francisco to New York for a short reassignment. The phone was set up to use the San Francisco device pool, and device mobility is enabled on the Cisco UCM. The user makes a call that matches a route pattern in a route list that contains the Standard Local Route Group. To 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 192
Refer to the exhibit.

```
Server: Cisco-SIPGateway/IOS-15.4.3.54
CSeq: 101 OPTIONS
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,
INFO, REGISTER
Allow-Events: telephone-event
Accept: application/sdp
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Content-Type: application/sdp
Content-Length: 369

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

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

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

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

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

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

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

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

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

Answer: D

NEW QUESTION 194
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 198
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 200
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 203
Refer to the exhibit.

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 205
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 206
Refer to the exhibit.

```
┌─ SIP Trunk Security Profile Information ──────────────────────────────┐
│                                                                        │
│  Name*                                      │ Secure SIP Trunk Profile                        │
│  Description                                │ Non Secure SIP Trunk Profile authenticated by null string │
│  Device Security Mode                       │ Encrypted                                    ▼  │
│  Incoming Transport Type*                   │ TLS                                          ▼  │
│  Outgoing Transport Type                    │ TLS                                          ▼  │
│  ☐ Enable Digest Authentication             │                                                  │
│  Nonce Validity Time (mins)*                │ 600                                              │
│  Secure Certificate Subject or Subject Alternate Name  │ |                                       │
│                                             │                                                  │
│                                             │                                                  │
│                                             │                                                  │
│                                             │                                                  │
│  Incoming Port*                             │ 5061                                             │
└────────────────────────────────────────────────────────────────────────┘
```

An administrator configures a secure SIP trunk on Cisco UCM.
Which value is needed in the secure certificate subject or subject alternate name field to accomplish this task?

A. The fully qualified domain name of the remote device that is configured on the SIP trunk.
B. The common name of the Cisco UCM CallManager certificate.
C. The full qualified domain name of all Cisco UCM nodes that run the CallManager service.
D. The common name of the remote device certificates.

Answer: B

NEW QUESTION 211
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 212
Refer to the exhibit.

```
🖥 hqcucmpub.pkinane.com - PuTTY

login as: admin
admin@hqcucmpub.pkinane.com's password:
Command Line Interface is starting up, please wait ...

    Welcome to the Platform Command Line Interface

VMware Installation:
        2 vCPU: Intel(R) Xeon(R) CPU E5-2699 v3 @ 2.30GHz
        Disk 1: 110GB, Partitions aligned
        8192 Mbytes RAM
        WARNING: DNS unreachable

admin:
```

An administrator accesses the terminal of a Cisco UCM and starts a packet capture. Which two commands must the administrator use on Cisco UCM to generate DNS traffic? (Choose two.)

A. utils ntp status
B. show cdp neighbor
C. show version active
D. utils diagnose test
E. autils diagnose module validate Network

Answer: DE

NEW QUESTION 216
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 218
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 222
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 224
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 227
Which two types of trunks can be used when configuring a hybrid Local Gateway for Cisco Webex Calling? (Choose Two.)

A. TLS-based
B. certificate-based
C. registration-based
D. authentication-based
E. OAuth-based

Answer: AC

Explanation:
These are the two types of trunks that can be used when configuring a hybrid local gateway for Cisco Webex Calling1. A TLS-based trunk uses Transport Layer Security (TLS) to secure the SIP signaling between the hybrid local gateway and Webex Calling1. A registration-based trunk uses SIP registration to authenticate the hybrid local gateway with Webex Calling and receive calls from the cloud1.

NEW QUESTION 230
An administrator needs to create a partial PRI consisting of the first seven timeslots available. Which configuration snippet configures the ISDN E1 PRI for this task?

A.
```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#interface Serial0/0/0:15
2900(config-controller)#pri-group timeslots 1-7
```

B.
```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#controller e1 0/0/0
2900(config-controller)#pri-timeslots 1-7
```

C.
```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#controller e1 0/0/0
2900(config-controller)#pri-group timeslots 1-7
```

D.
```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#pri-group timeslots 1-7
```

Answer: C

NEW QUESTION 234
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 239
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 241
......

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