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.

```
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 3
What is an advantage of using Cisco Webex Control HuB?

A. enables the provisioning, administration, and management of Webex services and Webex Hybrid Services
B. brings Video, audio, and web communication together to meet the collaboration needs of the modern workplace
C. provides streamlined communication and collaboration for a hybrid workforce
D. offers easy contact management, centralized administration, and centralized configuration management

Answer: A

Explanation:
Cisco Webex Control Hub is a cloud-based management platform that enables you to provision, administer, and manage Webex services and Webex Hybrid Services. It provides a single pane of glass for managing all of your Webex services, including Webex Meetings, Webex Teams, and Webex Calling.
Webex Control Hub offers a number of features and benefits, including:
- A single pane of glass for managing all of your Webex services
- Centralized user management
- Simplified provisioning and administration
- Real-time analytics and reporting
- Enhanced security and compliance

Webex Control Hub is a powerful tool that can help you manage your Webex services more effectively. It is easy to use and provides a number of features and benefits that can help you improve your productivity and efficiency.

NEW QUESTION 4
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 5
Which option must be used when configuring the Local Gateway for a Cisco Webex Calling trunk?

A. local authentication
B. certificate-based
C. mutual TLS
D. Auth-based

Answer: B

Explanation:
A certificate-based trunk is a type of trunk that uses certificates to authenticate the connection between Webex Calling and the Local 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 6
Which information is needed to restore the backup of a Cisco UCM publisher successfully?

A. the TFTP server details
B. the application credentials for Cisco UCM
C. the security password for Cisco UCM
D. the FTP server details

Answer: C

NEW QUESTION 7
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
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 9
An engineer is configuring a phone system CISCO UCM and wants to activate TFTP service. The engineer selects the serviceability page for configuration. Which nodes configurable for TFTP?

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

Answer: C

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

NEW QUESTION 10
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 10
A remote office has a less-than-optimal WAN connection and experiences packet loss, delay and jitter. Which VoIP codec is used in this situation?

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

Answer: B

NEW QUESTION 14
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 17
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 19
When configure Cisco UCM,which configuration enables phones to automatically reregister to a Cisco UCM publisher 'when the connection to the subscriber is lost?

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

Answer: A

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

NEW QUESTION 23
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 25
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 30
Which type of message must an administrator configure in the SIP Trunk Security Profile for a Message Waiting Indicator light to work with a SIP integration between Cisco UCM and Cisco Unity Connection?

A. Unsolicited NOTIFY
B. 200 ok
C. SIP Register
D. TCP port 5060

Answer: A

NEW QUESTION 33
How does Cisco UCM perform a digit analysis on-hook versus off-hook for an outbound call from a Cisco IP phone that is registered to Cisco UCM?

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

Answer: D

NEW QUESTION 35
Which attribute contains an XMPP stanza?

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

Answer: A

NEW QUESTION 36
A customer wants to conduct B2B video calls with a partner using on-premises conferencing solution. Which two devices are needed to facilitate this request?

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

Answer: AC

NEW QUESTION 37
Refer to the exhibit.

ip.addr==10.0.101.10

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

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

A. An NTP mismatch is preventing the connection of the TCP session between the SIP phone and the Cisco UC
B. The SIP phone and Cisco UCM must be set with identical NTP sources.
C. The certificates on the SIP phone are not trusted by the Cisco UC
D. The SIP phone must generate new certificates.
E. DNS lookup for the Cisco UCM FQDN is failin
F. The SIP phone must be reconfigured with the proper DNS server.
G. An network device is blocking TCP port 5060 from the SIP phone to the Cisco UC
H. This device must be reconfigured to allow traffic from the IP phone.

Answer: D

NEW QUESTION 41
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 44
According to QoS guidelines, what is the packet loss for streaming video?

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

Answer: B

NEW QUESTION 45
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 47
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

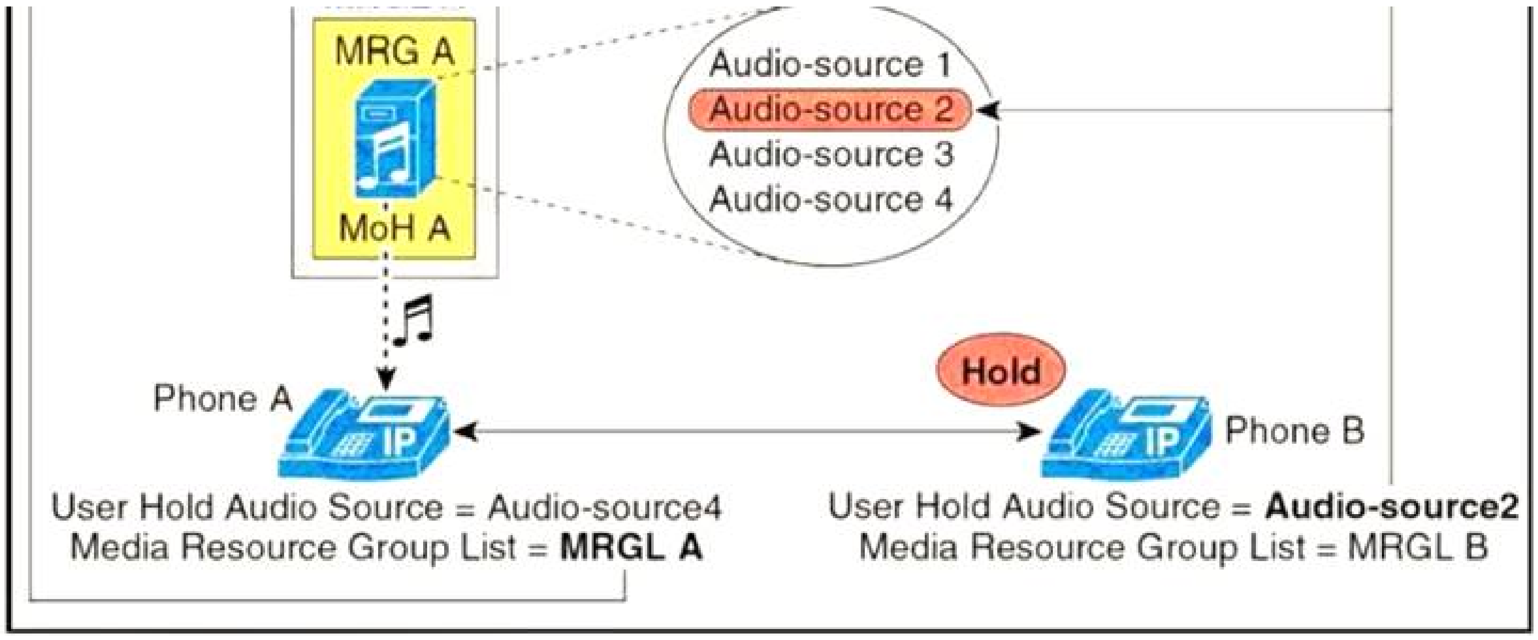
D. intercluster extension mobility
E. endpoint registration

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 49
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 51
Endpoint A is attempting to call endpoint B. Endpoint A only supports G.711ulaw with a packetization rate of 20 ms, and endpoint B supports packetization rate of 30 ms for G.711ulaw. Which two media resources are allocated to normalize packetization rates through transrating? (Choose two.)

A. software MTP on Cisco IOS Software
B. software MTP on Cisco UCM
C. software transcoder on Cisco UCM
D. hardware transcoder on Cisco IOS Software
E. hardware MTP on Cisco IOS Software

Answer: BE

NEW QUESTION 53
To provide high-quality voice and take advantage of the full voice feature set, which two access layer switches provide support?

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 802, 1Q trunking and 802.1p 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 required 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 58
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 60
What are two Cisco UCM location bandwidths that are deducted when G 729 and G.711 codecs are used? (Choose two.)

A. If a call uses G.729. Cisco UCM subtracts 16k.
B. If a call uses G.711, Cisco UCM subtracts 64k
C. If a call uses G.711, Cisco UCM subtracts 80k
D. If a call uses G.729. Cisco UCM subtracts 24k.
E. If a call uses G.729. Cisco UCM subtracts 40k

Answer: CD

NEW QUESTION 63
What are two reasons that AF41 is marked for the audio and video channels of a video call? (Choose two.)

A. to prioritize video over other high -priority traffic classes
B. to give video calls a higher priority than AP41 in the QoS policy
C. to allow high-definition quality calls over low-speed links
D. to preserve lip synchronization between the audio and video channels
E. to provide separate classes for audio calls and video calls

Answer: DE

NEW QUESTION 68
Which external DNS SRV record must be present for Mobile and Remote Access?

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

Answer: B

NEW QUESTION 71
Which configuration on Cisco UCM is required for SIP MWl to work?

A. Assign an MWl 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 74
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 77
A SIP phone has been configured in the system with MAC address 0030.96D2.D5CB. The phone retrieves the configuration file from the Cisco UCM. Which naming format is the file that is downloaded?

A. SIP003096D2D5CB.cnf.xml
B. SEP003096D2D5CB.cnf.xml
C. SEP003096D2D5CB.cnf
D. SIP003096D2D5CB.cnf

Answer: B

NEW QUESTION 80
How are network devices monitored in a collaboration network?

A. The Cisco Discovery Protocol table is shared among devices.
B. Ping Sweep reports "unmanaged' state devices.
C. System logs are collected in a Cisco Prime Collaboration Server.
D. Simple Network Managed Protocol is enabled on each device to poll specific values periodically.

Answer: C

NEW QUESTION 84
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 86
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 89
A Cisco UCM administrator sets up new route patterns to support phones in tour different locations, all with local gateways. The administrator wants to use the same route pattern for all four locations. How must the system be configured to achieve this goal?

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

Answer: B

NEW QUESTION 90
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 94
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 97
Which certificate does the Disaster Recovery System in Cisco UCM use to encrypt its communications?

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

Answer: D

NEW QUESTION 102
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 105
Refer to the exhibit.

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

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

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

Answer: B

NEW QUESTION 106
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 110
Which actions required for a firewall configuration on a Mobile and Remote Access through Cisco Expressway deployment?

A. The traversal zone on Expressway-c points to Expressway-e through the peer address field on the traversal zone, which specifies the Expressway-e server addres
B. For dual NIC deployments, set the Expressway-e address using an FQDN that resolves the IP address of the internal interface
C. The external firewall must allow these inbound connections to Expressway: SIP: TCP 5061; HTTPS: TCP 8443; XMPP TCP 5222; media: UDP 36002 to 59999
D. Do not use a shared address for Expressway-e and Expressway-c, as the firewallcannot distinguish between the
E. If static NAT for IP addressing on Expressway-e is used, ensure that any NAT operation on expressway-c does not resolve the same traffic IP addres
F. Shared NAT IS not supported
G. The internal firewall must allow these inbound and outbound connections between expressway - c and Expressway-e :sip;HTTPS(tunneled over SSH between C and E.TCP 2222: TCP 7001: Traversal Media: UDP 2776 to 2777(or 36000 to 36011 for large VM/appliance);XMPP:TCP 7400

Answer: B

NEW QUESTION 115
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 117
What is the traffic classification for voice and video conferencing?

A. Voice is classified as CoS 4, and video conferencing is CoS 5.
B. Voice and video conferencing are both classified is CoS 3.
C. Voice is classified as CoS 5, and video conferencing is CoS 4.
D. Video conferencing is classified as CoS 1, and voice is CoS 2.

Answer: B

NEW QUESTION 119
What are two access management mechanisms in Cisco Webex Control Hub? (Choose two.)

A. multifactor authentication
B. Active Directory synchronization
C. attribute-based access control
D. single sign-on with Google
E. Client ID/Client Secret

Answer: AB

Explanation:
The correct answers are A and B.
The two access management mechanisms in Cisco Webex Control Hub are multifactor authentication and Active Directory synchronization.
Multifactor authentication is a security measure that requires users to provide two or more pieces of evidence to verify their identity. This can include something they know, such as a password, and something they have, such as a security token.
Active Directory synchronization is a process that allows Cisco Webex Control Hub to automatically synchronize user accounts from an Active Directory domain. This can simplify user management and provide users with single sign-on access to Cisco Webex Control Hub and other applications.

NEW QUESTION 120
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 124
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 127
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

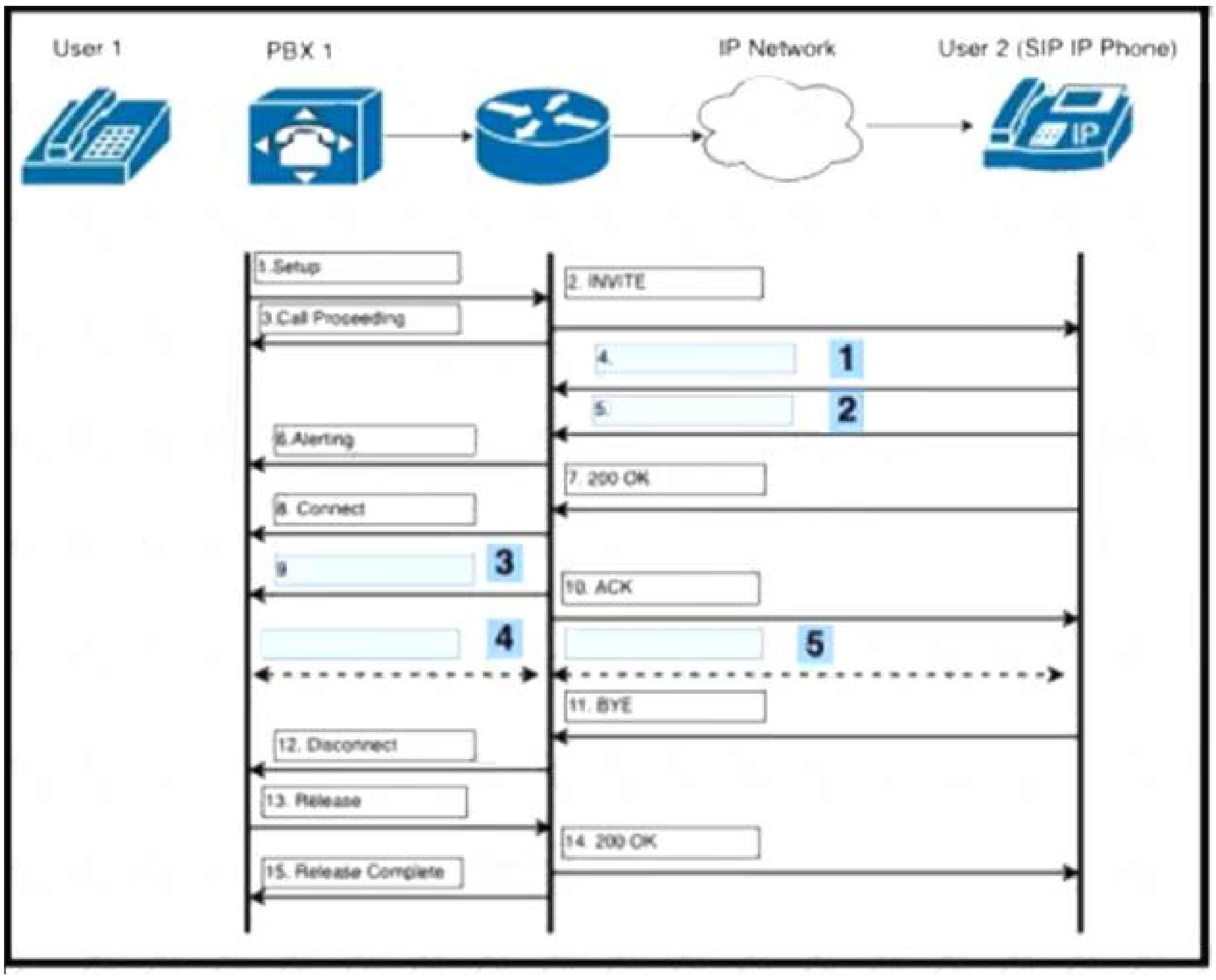
Answer: C

NEW QUESTION 131
An administrator executes the debug isdn q931 command while debugging a failed call. After a test call is placed, the logs return a disconnect cause code of 1. What is the cause of this problem?

A. The media resource is unavailable.
B. The destination number rejects the call.
C. The destination number is busy.
D. The dialed number is not assigned to an endpoint.

Answer: D

NEW QUESTION 133
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 134
A collaboration engineer troubleshoots issues with a Cisco IP Phone 7800 Series. The IPv4 address of the phone is reachable via ICMP and HTTP, and the phone is registered to Cisco UCM However the engineer cannot reach the CU of the phone Which two actions in Cisco UCM resolve the issue? (Choose two)

A. Enable SSH Access under Product Specific Configuration Layout in Cisco UCM
B. Disable Web Access under Product Specific Configuration Layout in Cisco UCM
C. Set a username and password under Secure Shell information in Cisco UCM
D. Enable Settings Access under Product Specific Configuration Layout in Cisco UCM
E. Enable FIPS Mode under Product Specific Configuration Layout in Cisco UCM

Answer: AB

NEW QUESTION 139
Exhibit.

```
admin:utils ntp status
ntpd (pid 14550) is running...

     remote          refid          st t   when   poll   reach   delay
offset   jitter
=================================================================
*192.168.1.1    17.253.14.125      2 u   39    64     3    0.456   -0.236
0.116
*192.168.1.2    17.253.14.125      2 u   38    64     3    0.817   -0.695
0.395
```

Refer the exhibit. A collaboration engineer needs to replace the original, single NTP server that was configured during the inital install of a Cisco UCM server. What is the first step to accomplish this task?

A. Restart the NTP service on Cisco UCM
B. Delete the original NTP server from Cisco UCM
C. Stop the NTP service on Cisco UCM
D. Enable NTP authentication for the new NTP server on Cisco UCM

Answer: B

NEW QUESTION 140
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 143
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 144
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 147
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 150
A company has an excessive number of call transfers to local and long-distance PSTN from Cisco Unity Connection voicemail. Which action in the Cisco Unity Connection restriction table resolves this issue?

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

Answer: A

NEW QUESTION 152
An engineer troubleshoots a Cisco Jabber login problem on a Windows PC. The login fails with the error message "Cannot find your services automatically. Click advanced settings to set up manually " Which action should the engineer take first?

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

Answer: C

NEW QUESTION 157
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 162
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 164
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 166
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 171
What is the function of the Cisco Unity Connection Call Handler?

A. routes calls to a user based on caller input
B. queues calls
C. allows customized scripts for IVR capabilities
D. searches a list of extensions until the call is answered

Answer: A

Explanation:
A Cisco Unity Connection Call Handler is a software application that answers calls, plays greetings, and routes calls to users based on caller input. Call handlers can be used to create automated attendants, voice menus, and other interactive voice response (IVR) applications.
Call handlers are created and managed using the Cisco Unity Connection Administration interface. When creating a call handler, you can specify a variety of settings, including the greeting that is played, the caller input options that are available, and the destination that calls are routed to.
Call handlers are a powerful tool that can be used to create a variety of IVR applications. By using call handlers, you can improve the efficiency of your organization's communications and provide a better experience for your callers.
Here are some additional tips for using call handlers:

- Use call handlers to create automated attendants that can answer calls and route them to the appropriate person or department.
- Use call handlers to create voice menus that can provide callers with information or options.
- Use call handlers to create interactive voice response (IVR) applications that can collect information from callers and process their requests.

NEW QUESTION 174
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 177
What is the default TCP port for SIP OAuth mode in Cisco UCM?

A. 5011

B. 3174
C. 8443
D. 5090

Answer: D

Explanation:
The Cisco Unified Communications Manager (CUCM) uses SIP Phone OAuth Port (5090) to listen for SIP line registration from Jabber OnPremise devices over TLS. However, CUCM uses SIP Mobile Remote Access Port (default 5091) to listen for SIP line registrations from Jabber over Expressway through mTLS. Both of these ports are configurable.

NEW QUESTION 178
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 181
Which value should be changed when each Cisco UCM node does not allow for more than 5000 phones to be registered?

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

Answer: C

NEW QUESTION 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 186
What is a possible cause of the PRI issue?

```
ISDN Serial1:23 interface
        dsl 1, interface ISDN Switchtype =
primary-5ess
    Layer 1 Status:
        ACTIVE
    Layer 2 Status:
        TEI = 0, Ces = 1, SAPI = 0, State =
TEI_ASSIGNED
    Layer 3 Status:
        0 Active Layer 3 Call(s)
    Activated dsl 1 CCBs = 0
    The Free Channel Mask:  0x807FFFFF
    Total Allocated ISDN CCBs = 5
```

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

Answer: D

NEW QUESTION 188
Refer to the exhibit.

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

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

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

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

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

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

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

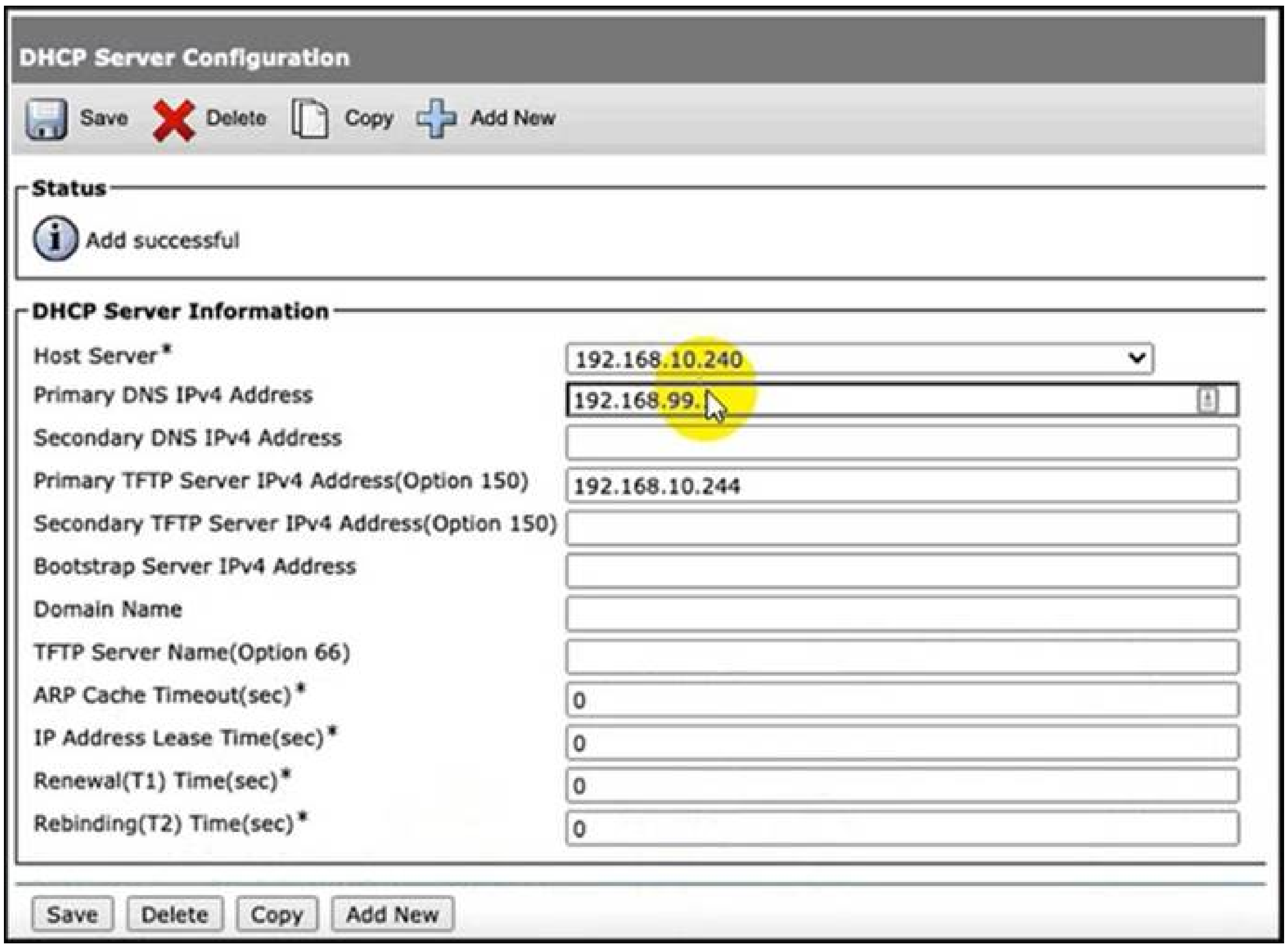
Answer: D

NEW QUESTION 191
A Cisco Telepresence SX80 suddenly has issues displaying main video to a display over HDMI. Which command can you use from the SX80 admin CLI to check the video output status to the monitor?

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

Answer: B

NEW QUESTION 192
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 196
What are the predefined call handlers in Cisco Unity Connection?

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

Answer: D

NEW QUESTION 201
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 204
User A Calls user. The call gets connected, but the quality is bed. What are two reasons for this issue? (Choose two)

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

Answer: CD

NEW QUESTION 209
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 213
Refer to the exhibit.

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

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

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

Answer: D

NEW QUESTION 216
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 219
Refer to the exhibit.

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

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

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

Answer: A

NEW QUESTION 220
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 225
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 230
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 235
......