350-801.premium.128q - DEMO

Number: 350-801
Passing Score: 800
Time Limit: 120 min

350-801

Implementing Cisco Collaboration Core Technologies

Exam A

QUESTION 1
Which two functionalities does Cisco Expressway provide in the Cisco Collaboration architecture? (Choose two.)

A. Survivable Remote Site Telephony functionality
B. customer interaction management services
C. secure firewall and NAT traversal for mobile or remote Cisco Jabber and TelePresence Video endpointsD. MGCP gateway registration
E. Secure business-to-business communications

Correct Answer: CE
Section: Infrastructure and Design
Explanation

Explanation/Reference:

QUESTION 2
An engineer must extend the corporate phone system to mobile users connecting through the internet with their own devices. One requirement is to keep that as simple as possible for end users. Which infrastructure element achieves these goals?

A. Cisco Express Mobility
B. Cisco Expressway-C and Expressway-E
C. Cisco Unified Border Element
D. Cisco Unified Instant Messaging and Presence

Correct Answer: C
Section: Infrastructure and Design
Explanation

Explanation/Reference:

QUESTION 3

A customer wants a video conference with five Cisco TelePresence IX5000 Series systems. Which media resource is necessary in the design to fully utilize the immersive functions?

A. Cisco PVDM4-128
B. software conference bridge on Cisco Unified Communications Manager
C. Cisco Webex Meetings Server
D. Cisco Meeting Server

Correct Answer: C
Section: Infrastructure and Design
Explanation

Explanation/Reference:

QUESTION 4
An engineer is designing a load balancing solution for two Cisco Unified Border Element routers. The first router (cube1.abc.com) takes 60% of the calls and the second router (cube2.abc.com) takes 40% of the calls. Assume all DNS A records have been created. Which two SRV records are needed for a load balanced solution? (Choose two.)

A. _sip._udp.abc.com 60 IN SRV 2 60 5060 cube1.abc.com
B. _sip._udp.abc.com 60 IN SRV 60 1 5060 cube1.abc.com
C. _sip._udp.abc.com 60 IN SRV 1 40 5060 cube2.abc.com
D. _sip._udp.abc.com 60 IN SRV 3 60 5060 cube2.abc.com
E. _sip._udp.abc.com 60 IN SRV 1 60 5060 cube1.abc.com

Correct Answer: CE

Section: Infrastructure and Design
Explanation

Explanation/Reference:

QUESTION 5 Which two functions are provided by Cisco Expressway Series?
(Choose two.)

A. interworking of SIP and H.323
B. endpoint registration
C. intercluster extension mobility
D. voice and video transcoding
E. voice and video conferencing

Correct Answer: AD
Section: Infrastructure and Design
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/expressway/config_guide/X8-11/Cisco-Meeting-Server-2-4-with-Cisco-Expressway-Deployment-Guide_X8-11-4.pdf

QUESTION 6
An incoming off-net call to a user fails. An engineer notices that the off-net call is G.711, but the phone accepts only G.729. Which media resource on a Cisco Unified Border Element and Cisco Unified Communications Manager must the engineer configure to manage the codec negotiation?

A. transcoder
B. CFB
C. MOH
D. MTP

Correct Answer: A
Section: Infrastructure and Design
Explanation

Explanation/Reference:

QUESTION 7
Which Cisco Unified Communications Manager service parameter should be enabled disconnect a multiparty call when the call initiator hangs up?

A. Drop Ad Hoc Conference
B. H.225 Block Setup Destination
C. Block OffNet To OffNet Transfer
D. Enterprise Feature Access Code for Conference

Correct Answer: A
Section: Infrastructure and Design
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmsys/CUCM_BK_SE5FCFB6_00_cucm-system-guide-100/CUCM_BK_SE5FCFB6_00_cucm-system-guide-100_chapter_011000.html#CUCM_TK_DFC66444_00

QUESTION 8
A network administrator deleted a user from the LDAP directory of a company. The end user shows as Inactive LDAP Synchronized User in Cisco Unified Communications Manager. Which step is next to remove this user from Cisco Unified Communications Manager?
A. Delete the user directly from Cisco Unified Communications Manager

B. Restart the Dirsync service after the user is deleted from LDAP directory.
C. Execute a manual sync to refresh the local database and delete the end user.
D. Wait 24 hours for the garbage collector to remove the user.

Correct Answer: B
Section: Infrastructure and Design
Explanation

Explanation/Reference:

QUESTION 9 A customer has Cisco Unity Connections that is integrated with LDAP. As a Unity Connection administrator, you have received a request to change the first name for VM user. Where must the change be performed?

A. Cisco Unity Connection
B. Cisco Unified Communications Manager end user
C. Active Directory
D. Cisco IM and Presence

Correct Answer: C
Section: Infrastructure and Design
Explanation

Explanation/Reference:

QUESTION 10 Which configuration step is necessary for a Cisco SIP phone to synchronize its time with a specific source?

A. Add a Phone NTP Reference to the Date/Time Group.
B. Assign the device to the correct region.
C. Change the Time Format from 24-hour to 12-hour.
D. Change the Time Zone from “America/Los_Angeles" to “Etc/GMT+8".

Correct Answer: A
Section: Infrastructure and Design
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0110.html

QUESTION 11
After an engineer runs the **utils ntp status** command on the Cisco Unified Communications Manager publisher, the stratum value is 16. Which issue can the Cisco Unified CM cluster experience?

A. Unified CM sends an NTPv4 packet.
B. Database replication is not synchronized on the Unified CM nodes.
C. The cluster loses access to port 124 at the firewall.
D. The date/time group on all phones defaults to the time zone of the engineer.

Correct Answer: B
Section: Infrastructure and Design
Explanation

Explanation/Reference:

QUESTION 12
When a new SIP phone is registered to Cisco Unified Communications Manager, it keeps failing and showing an “unprovisioned” error message in the phone display. Which problem is a possible cause of this issue?

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

Correct Answer: C
Section: Infrastructure and Design
Explanation

Explanation/Reference:

QUESTION 13 Which DHCP option must be set up for new phones to obtain the TFTP server IP address?

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

Correct Answer: C
Section: Infrastructure and Design
Explanation

Explanation/Reference:
Reference: https://blog.router-switch.com/2013/03/dhcp-option-150-dhcp-option-66/

QUESTION 14 Which two conditions must a user meet to provision a new device using the Self-Provisioning feature? (Choose two.)

A. The user must have a primary extension.
B. At least two DNs must be assigned to the user device.
C. The user must be part of "Standard CCM Super User".
D. The user must have the appropriate universal device template linked to the user profile.
E. The user must have at least one user device profile assigned.

Correct Answer: AD
Section: Infrastructure and Design
Explanation

Explanation/Reference:

QUESTION 15 How many DNS SRV entries can be defined in the SIP trunk destination address field in Cisco Unified Communications Manager?

A. 1
B. 8
C. 16
D. 4

Correct Answer: C
Section: Infrastructure and Design
Explanation

Explanation/Reference:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/CUCM_BK_SE5DAF88_00_cucm-system-configuration-guide-1151/CUCM_BK_SE5DAF88_00_cucm-system-configuration-guide1151_chapter_01110.html

QUESTION 16 On which Cisco Unified Communications Manager nodes can the TFTP service be enabled?

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

Correct Answer: C
Section: Infrastructure and Design
Explanation

Explanation/Reference:
Explanation:
You can configure the TFTP service on the first node or a subsequent node, but usually you should configure it on the first node. For small systems, the TFTP server can coexist with a Cisco Unified Communications Manager on the same server.

QUESTION 17 Which issue causes slips on a PRI?

A. incorrect clock source
B. incorrect encapsulation
C. incorrectly configured time zone
D. change in the line code

Correct Answer: A
Section: Infrastructure and Design
Explanation

Explanation/Reference:

QUESTION 18 An administrator recently upgraded a Cisco Webex DX80 through its web interface but discovered the next morning that the unit has received to its previous version. What must the administrator do to prevent this from happening again?

A. Assign a phone security profile with secure SIP.
B. Set the prepare cluster for rollback to pre-8.0 enterprise parameter to true.
C. Confirm the phone load name in the phone configuration.
D. Assign a universal device template to the phone.

Correct Answer: C
Section: Infrastructure and Design
Explanation

Explanation/Reference:

QUESTION 19
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 Software Upgrades page in CUCM OS Administration
B. The In-Room Control Editor on the webpage of the MX800
C. The phone configuration page in CUCM Administration
D. The SIP Trunk Security Profile page in CUCM Administration

Correct Answer: A
Section: Infrastructure and Design
Explanation

Explanation/Reference:

QUESTION 20
A presence redundancy group is deployed, and an engineer initiates a manual fallback. Which statement about Cisco Server Recovery Manager is true?

A. disconnects all users that had been failed over, and the users must log in again
B. disconnects all users that had been failed over
C. restarts critical services on the secondary node
D. restarts the Cisco Presence Engine

Correct Answer: B
Section: Infrastructure and Design
Explanation

Explanation/Reference:

QUESTION 21 Which packet delay is the maximum supported between Cisco Unified Communications Manager nodes for clustering over WAN deployments?

A. 150 ms round tip
B. 510 ms round trip
C. 40 ms round trip
D. 80 ms round trip

Correct Answer: D
Section: Infrastructure and Design
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/callpros.html

QUESTION 22 A user dials 9011841234567 to reach Vietnam. Which steps send the call to the PSTN provider as 011841234567? A.

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

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

in the Called Party Transformation Pattern Configuration section,
configure the Pattern as 9.011841234587
configure the Discard Digits as Predot 10-10-Dialing

in the Calling Party Transformation Patterns section,
configure the Pattern as a 9.011841234587
configure the Discard Digits as Predot

B.

C.

D.

Correct Answer: A
Section: Infrastructure and Design
Explanation

Explanation/Reference:
QUESTION 23 Where is the default for Maximum Session Bit Rate for a region configured?

A. Service Parameter Configuration
B. Enterprise Phone Configuration
C. Enterprise Parameters Configuration
D. Region Configuration

Correct Answer: A
Section: Infrastructure and Design
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_1_1/ccmcfg/CUCM_BK_A34970C5_00_admin-guide-91/CUCM_BK_A34970C5_00_admin-guide-91_chapter_0111.html

QUESTION 24 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 Video Output**
B. **xCommand Video Status**
C. **xConfiguration Video Output**
D. **xStatus HDMI Output**

Correct Answer: C
Section: Infrastructure and Design
Explanation

Explanation/Reference:

QUESTION 25
Which statement about Cisco Unified Communications Manager and Cisco IM and Presence backups is true?

A. Backups should be scheduled during off-peak hours to avoid system performance issues.
B. Backups are saved as .tar files and encrypted using the web administrator account.
C. Backups are saved as unencrypted.tar files.
D. Backups are not needed for subscriber Cisco Unified Communications Manager and Cisco IM and Presence servers.

Correct Answer: A
Section: Infrastructure and Design
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1_SU1/Administration/cucm_b_administration-guide-1151su1/cucm_b_administration-guide-1151su1_chapter_01010.html#CUCM_TK_S7FC26D5_00

QUESTION 26
What is a software-based media resource that is provided by the Cisco IP Voice Media Streaming Application?

A. video conference bridge
B. auto-attendant
C. transcoder
D. annunciator

Correct Answer: D
Section: Infrastructure and Design
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab09/clb09/media.html

QUESTION 27
When a user dials a number with a phone that is registered to the Cisco Unified Communications Manager, what is the default timeout before the number is sent?

A. 15 seconds
B. 5 seconds
C. 10 seconds
D. 3 seconds

Correct Answer: C
Section: Infrastructure and Design
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-callmanager/13920-call-routing.html

QUESTION 28
An engineer deploys a Cisco Expressway-E server for a customer who wants to utilize all features on the server. Which feature does the engineer configure on the Expressway-E?

A. H.323 endpoint registrations
B. VTC bridge
C. MRA
D. SIP gateway for PSTN providers

Correct Answer: C
Section: Infrastructure and Design
Explanation

Explanation/Reference:

QUESTION 29 Which SNMP service must be activated manually on the Cisco UCM after installation?

A. Host Resources Agent
B. Cisco CallManager SNMP
C. Connection SNMP Agent
D. SNMP Master Agent

Correct Answer: B
Section: Infrastructure and Design
Explanation

Explanation/Reference:
Explanation:
SNMP Master Agent serves as the primary service for the MIB interface. You must manually activate Cisco CallManager SNMP service; all other SNMP services should be running after installation.
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/service/9_0/admin/CUCM_BK_C136FE37_00_cisco-unified-serviceability-administration-90/CUCM_BK_C136FE37_00_cisco-unified-serviceability-administrationguide_chapter_0101.html

QUESTION 30 A company deploys centralized Cisco UCM architecture for a hub location and two remote sites.

▪ The company has only one ITSP connection at the hub location, and ITSP supports only G.711 calls.
▪ Remote site A has a 1-Gbps fiber connection to the hub location and calls to and from remote site A use G.711 codec. ▪
Remote site B has a 1-T1 connection to the hub location and calls to and from remote site B use G.729 codec.

Based on the provided guidance, a Cisco voice engineer must design media resource management for the customer. What is the method that needs to be followed?
A. configure the hardware transcoder on the site B router
B. configure the hardware transcoder on the site A router
C. configure the hardware transcoder on the hub location router
D. configure the software transcoder on Cisco UCM to support voice calls to and from both remote sites

Correct Answer: A
Section: Infrastructure and Design
Explanation

Explanation/Reference:

QUESTION 31 What are two key features of the Expressway series? (Choose two.)

A. IP to PSTN call connectivity
B. B2B calls
C. VPN connection toward the internal UC resources
D. SIP header modification
E. device registration over the Internet

Correct Answer: BE
Section: Infrastructure and Design
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/products/collateral/unified-communications/expressway-series/datasheet-c78-737605.html

QUESTION 32 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

Correct Answer: A
Section: Infrastructure and Design
Explanation

Explanation/Reference:
Reference:
https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/211393-Change-CUCM-Server-Definition-from-IP-Ad.html https://community.cisco.com/t5/ip-telephony-and-phones/cucm-10-5-host-name-and-dns-change/td-p/2996878

QUESTION 33
When configuring 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. Device Pool
D. Cisco UCM Group

Correct Answer: A
Section: Infrastructure and Design
Explanation

Explanation/Reference:
Explanation:
Cisco Unified SRST provides Cisco Unified CM with fallback support for Cisco Unified IP phones that are attached to a Cisco router on your local network. Cisco Unified SRST enables routers to provide call-handling support for Cisco Unified IP phones when they lose connection to remote primary, secondary, or tertiary Cisco Unified CM installations or when the WAN connection is down.

Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusrst/admin/sccp_sip_srst/configuration/guide/SCCP_and_SIP_SRST_Admin_Guide/srst_overview.html

QUESTION 34
Which version is used to provide encryption for SNMP management traffic in collaboration deployments?

A. SNMPv2c
B. SNMPv2C. SNMPv1
D. SNMPv3

Correct Answer: D
Section: Infrastructure and Design
Explanation

Explanation/Reference:
Reference:
https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/snmp/configuration/xe-16/snmp-xe-16-book/nm-snmp-encrypt-snmp-support.html

QUESTION 35 What is the validity period of the ITL Recovery certificate in Cisco UCM?

A. 1 year
B. 20 years
C. 5 years
D. 10 years

Correct Answer: B
Section: Infrastructure and Design
Explanation

Explanation/Reference:
Explanation:
The validity of ITLRecovery has been extended from 5 years to 20 years to ensure that the ITLRecovery certificate remains same for a longer period

Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/security/12_0_1/secugd/cucm_b_cucm-security-guide-1201/cucm_b_cucm-security-guide-1201_chapter_011.html

QUESTION 36 An engineer encounters third-party devices that do not support Cisco Discovery Protocol. What must be configured on the network to allow device discovery?

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

Correct Answer: C
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:

QUESTION 37
On which protocol and port combination does Cisco Prime Collaboration receive notifications (Traps and InformRequests) from several network devices in the Collaboration infrastructure for which it has requested notifications?
A. UDP 162
B. TCP 80
C. UDP 161
D. TCP 161

Correct Answer: A
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/net_mgmt/prime/collaboration/12-1/assurance/advanced/guide/cpco_b_cisco-prime-collaboration-assurance-guide-advanced-12-1/cpco_b_cisco-prime-collaboration-assurance-guideadvanced-12-1_chapter_01111.html

QUESTION 38 Which transport protocol does the application layer protocol SNMP use?

A. XML
B. UDP
C. SIP
D. HTTP

Correct Answer: B
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:
Reference: https://www.geeksforgeeks.org/simple-network-management-protocol-snmp/

QUESTION 39
Which protocol does Cisco Prime Collaboration Assurance use to poll the health status of different systems in the Collaboration environment?

A. SIP
B. SNMP
C. SCCP
D. SMTP

Correct Answer: B
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/products/collateral/cloud-systems-management/prime-collaboration/guide-c07-736946.html#_Toc446633083

QUESTION 40
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.711
B. G.722
C. G.723
D. G.729

Correct Answer: D
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/media.html

QUESTION 41

```
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:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
```

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

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

Correct Answer: D
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:

QUESTION 42

```
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=fmtp:101 0-15
  a=trafficclass:conversational.audio.immersive.aq:admitted
```

Refer to the exhibit. Endpoint A calls endpoint B. What is the only audio codec that can be used for the call?

A. Telephone-event/8000
B. G7221/16000
C. PCMA/8000
D. G722/8000

Correct Answer: B
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:

QUESTION 43

```
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
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0,mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1,mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=sendrecv
```

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

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

Correct Answer: DE
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:

QUESTION 44
An engineer wants to manually deploy a Cisco Webex DX80 video endpoint to an end user. Which type of provisioning can be configured on the endpoint?

A. CUBE
B. CMS
C. CUCM
D. Edge

Correct Answer: C
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/products/collateral/collaboration-endpoints/desktop-collaboration-experience-dx600-series/datasheet-c78-731879.html

QUESTION 45

```
v=0
o=Cisco-SIPUA 13439 0 IN IP4 10.10.10.10
s=SIP Call
b=AS:4064
t=0 0
m=audio 0 RTP/AVP 114 9 124 113 115 0 8 116 18 101
c=IN IP4 10.10.10.10
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114 maxplaybackrate=16000;sprop-maxcapturerate=16000;maxaveragebitrate=
64000;stereo=0;sprop-stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0,mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1,mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

Refer to the exhibit. A call is failing to establish between two SIP Devices. The called device answers with this SDP. Which SDP parameter causes this issue?

A. The payload for G.711ulaw must be 18.
B. The calling device did not offer a ptime value.
C. The media stream is set to sendonly.
D. The RTP port is set to 0.

Correct Answer: D
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:

QUESTION 46
A remote office has a less-than-optimal WAN connection and experiences packet loss, delay, and jitter. Which VoIP codec should be used in this situation?

A. G.711ulaw
B. iLBC
C. G.722.1
D. G.729A

Correct Answer: D

Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:
Reference: https://community.cisco.com/t5/collaboration-voice-and-video/summary-of-cucm-supported-codecs/ta-p/3162905

QUESTION 47

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

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

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

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

Correct Answer: D
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:

QUESTION 48 Which two types of device are supported by the Bulk Administration Tool? (Choose two.)

A. H.322
B. Cisco Unified IP phones (all models)
C. SIP trunks
D. H.225 trunks
E. music on hold servers

Correct Answer: AB
Section: Protocols, Codecs, and Endpoints

Explanation
Explanation/Reference:

QUESTION 49
You are adding regions in Cisco Unified Communications Manager. Which codec(s) are selected when a call is placed if you set up the max audio bit rate to use 8 kbps?

A. G.729
B. G.729 and G.711ulaw
C. G.711ulaw and G.711alaw
D. G.722

Correct Answer: A
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:

QUESTION 50
How can an administrator stop Cisco Unified Communications Manager from advertising the OPUS codec for recording enabled devices?

A. Route recorded calls through Cisco Unified Border Element because it does not support OPUS.
B. Go to the phone's configuration page and set "Advertise OPUS Codec" to be "false".
C. Integrate the Cisco Unified CM with 3 recording solution that does not support OPUS.
D. In CUCM Service Parameters set "Opus Codec Enabled" to "Enabled for all Devices Except Recording-Enabled Devices."

Correct Answer: D
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/211297-Configure-Opus-Support-on-Cisco-Unified.pdf

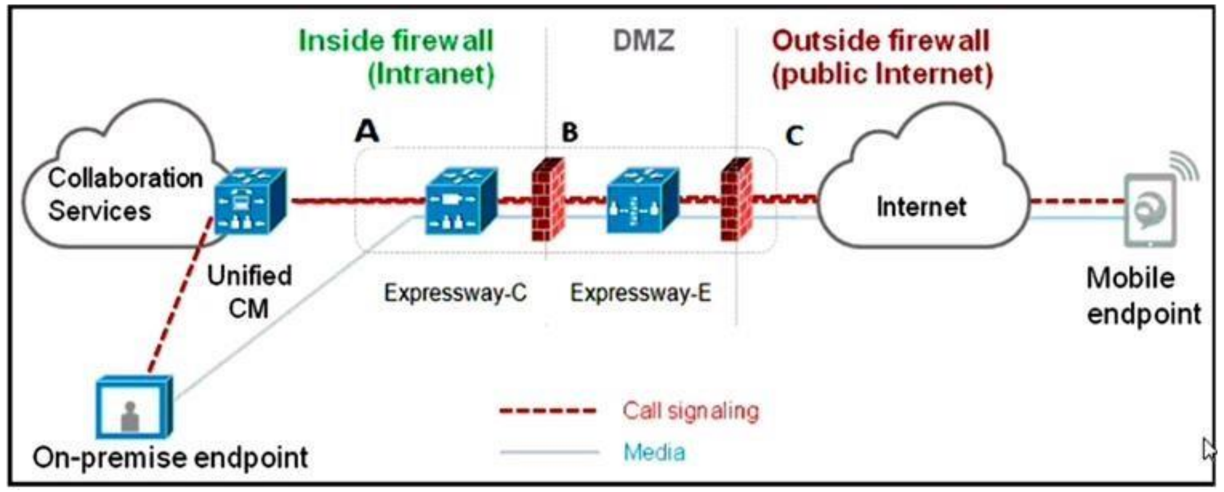
QUESTION 51
An engineer must manually provision a Cisco IP Phone 8845 using SIP. Which two fields must be configured for a successful provision? (Choose two.)

A. location
B. media resources group list
C. SIP profile
D. CSS
E. device security profile

Correct Answer: CE
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:

QUESTION 52



Refer to the exhibit. When making a call to a MRA client, what are the combinations of protocol on each of the different sections A-B-C?

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

Correct Answer: A
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:
Reference:
https://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/expressway/config_guide/X8-9/Mobile-Remote-Access-via-Expressway-Deployment-Guide-X8-9-1.pdf

QUESTION 53
An engineer is configuring a Cisco Unified Border Element to allow the video endpoints to negotiate without the Cisco Unified Border Element interfering in the process. What should the engineer configure on the Cisco Unified Border Element to support this process?

A. Configure codec transparent on the dial peers.
B. Configure a transcoder for video protocols.
C. Configure a hardcoded codec on the dial peers.
D. Configure pass-thru content sdp on the voice service.

Correct Answer: A
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:
Reference:
https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/cube-codec-basic.html

QUESTION 54
Due to service provider restriction, Cisco UCM cannot send video in the SDP. Which two options on Cisco UCM are configured to suppress video in the SDP in outgoing invites? (Choose two.)

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

Correct Answer: AD

Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:
Reference: https://community.cisco.com/t5/ip-telephony-and-phones/cucm-region-vs-location-video-bandwidth/td-p/1488224
https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/voi-audio-forced.html

QUESTION 55 A Cisco IP Phone 7841 that is registered to a Cisco UCM with default configuration receives a call setup message. Which codec is negotiated when the SDP offer includes this line of text?

m=audio 49181 RTP/AVP 0 8 97

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

Correct Answer: D
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:
Explanation:
Those are the codecs.

They are RTP payload types, and in preference order.

Eg 8=g711alaw, 0=g711ulaw, 18=g729, 97=this one is special, it's actually what they want to Mark RTP-NTP (RFC2833) DTMF relay as.

Reference: https://community.cisco.com/t5/ip-telephony-and-phones/sip-sdp-no-codec-specifed/td-p/3913990 https://en.m.wikipedia.org/wiki/RTP_payload_formats

QUESTION 56
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 transcoder on Cisco UCM
B. hardware transcoder on Cisco IOS Software
C. software MTP on Cisco IOS Software
D. software MTP on Cisco UCM
E. hardware MTP on Cisco IOS Software

Correct Answer: AD
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:

QUESTION 57
An engineer with ID0123456789 is designing a new dial plan for a customer that has offices in several countries on four continents around the world. This client also wants to integrate with a Microsoft Lync backend. Which dial plan type does the engineer recommend?

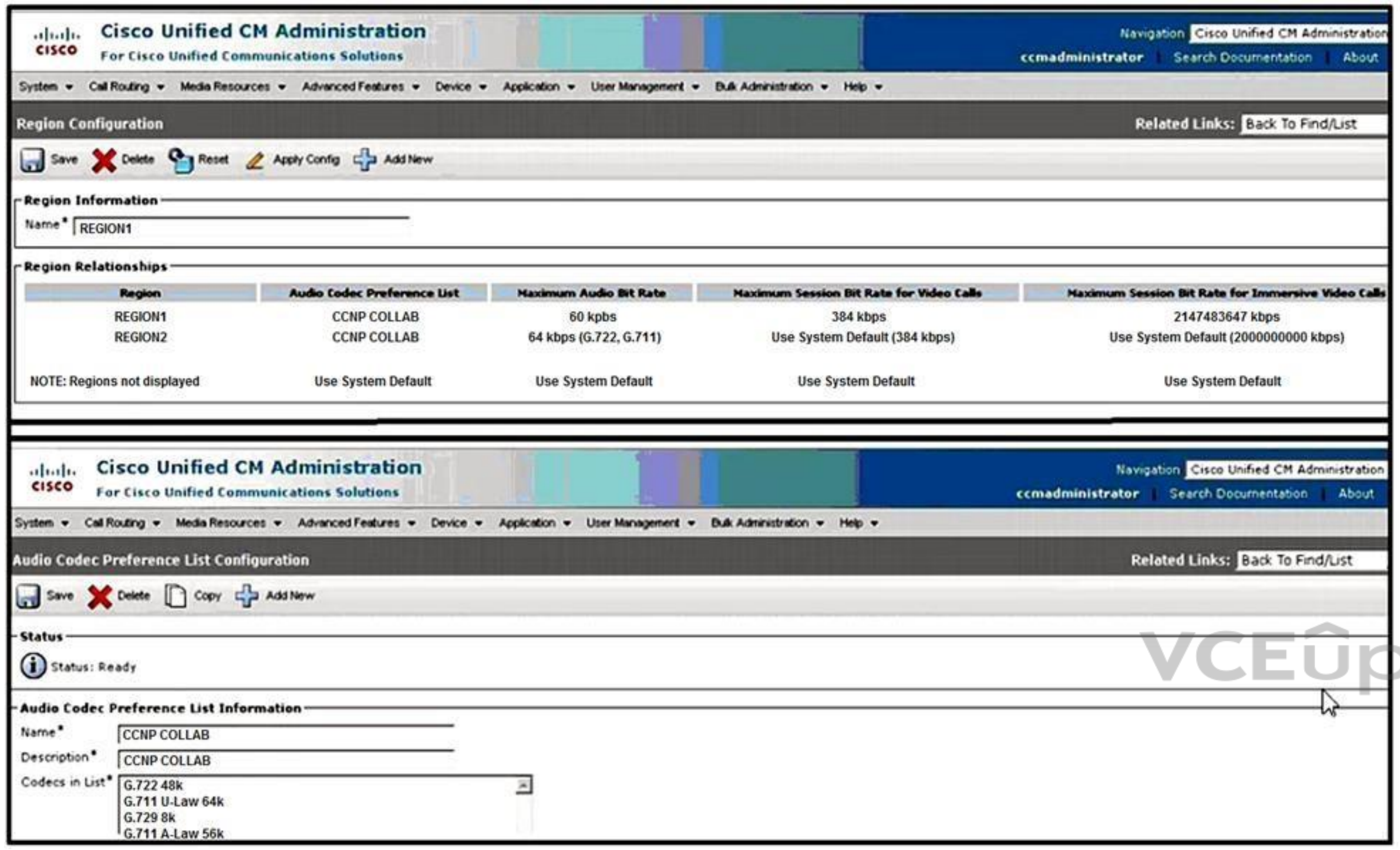
A. H.323
B. SIP URI
C. E.164
D. TEHO

Correct Answer: C

Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:
Reference: https://www.ciscolive.com/c/dam/r/ciscolive/us/docs/2018/pdf/BRKCOL-2610.pdf

QUESTION 58



Refer to the exhibit. An engineer is troubleshooting a codec negotiation issue where both endpoints that are involved in the call support the codecs listed in the exhibit. Which audio codec is selected if a call between two endpoints in Region1 is placed?

A. G.722
B. G.729
C. G.711a
D. G.711u

Correct Answer: A
Section: Protocols, Codecs, and Endpoints
Explanation

Explanation/Reference:

QUESTION 59
Which command is used in Cisco IOS XE TDM gateway to configure the voice T1/E1 controller to provide clocking?

A. **clock source line**
B. Cisco IOS XE TDM gateway T1/E1 controller
cannot provide clocking.

C. **clock source internal**
D. **clock source network**

Correct Answer: C
Section: Cisco IOS XE Gateway and Media Resources Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/routers/access/interfaces/NIM/software/configuration/guide/4gen-t1-e1-nim-guide.html

QUESTION 60 Which command in the MGCP gateway configuration defines the secondary Cisco Unified Communications Manager server?

A. **mgcpapp**
B. **ccm-manager fallback-mgcp**
C. **mgcp call-agent**
D. **ccm-manager redundant-host**

Correct Answer: B
Section: Cisco IOS XE Gateway and Media Resources Explanation

Explanation/Reference:

QUESTION 61
Which action is required if an engineer wants to have Cisco Unified Communications Manager control the configuration for an MGCP gateway?

A. Apply the ccm-manager configuration commands to the gateway.
B. Upload the custom configuration in the TFTP server in Cisco Unified CM.
C. From Cisco Unified CM > Device > Gateway > Add gateway, check the auto-configuration check box.
D. Configure the Cisco Unified CM's IP in voice service VoIP.

Correct Answer: C
Section: Cisco IOS XE Gateway and Media Resources Explanation

Explanation/Reference:

QUESTION 62
A user reports that when receiving an inbound call on their IP Phone from the PSTN they are unable to transfer this call to another PSTN number. What is the reason for this failures?

A. The IP phone is configured with the wrong region.
B. The incoming calling search space of the SIP trunk does not include the partition of the line in the IP phone.
C. The service parameter related to Offnet to Offnet Call Transfer is set to TRUE.
D. The gateway is configured with the wrong device pool.

Correct Answer: D
Section: Cisco IOS XE Gateway and Media Resources Explanation

Explanation/Reference:

QUESTION 63
A customer is deploying a SIP IOS gateway for a customer who requires that in-band DTMF relay is first priority and out-of-band DTMF relay is second priority.

Which IOS entry sets the required priority?
A. **dtmf-relay rtp-nte sip-notify**
B. **dtmf-relay cisco-rtp**
C. **sip-notify dtmf-relay rtp-nte**

D. **dtmf-relay sip-kpml cisco-rtp**

Correct Answer: A
Section: Cisco IOS XE Gateway and Media Resources Explanation

Explanation/Reference:

QUESTION 64 A DTMF mismatch is occurring between an MGCP gateway registered FXS port and a Cisco Unified Communications Manager SIP trunk. Which media resource can be leveraged to interwork this mismatch?

A. Conference Bridge
B. Trusted Relay Point
C. Media Termination Point
D. Annunciator

Correct Answer: C
Section: Cisco IOS XE Gateway and Media Resources Explanation

Explanation/Reference:

QUESTION 65

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

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

A. G.711ulaw
B. No codec is used (missing codec command).
C. G.711alaw
D. G.729r8

Correct Answer: A
Section: Cisco IOS XE Gateway and Media Resources Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/dialpeer/configuration/15-mt/vd-15-mt-book/vd-dp-cfg-examp.pdf

QUESTION 66
An engineer must configure an MGCP gateway and register it to Cisco Unified Communications Manager. Which prerequisite must be met before applying the gateway commands to enable MGCP?

A. The MGCP gateway and the Cisco Unified CM must be able to communicate aver ports 5060 and 5061.
B. Cisco Unified CM and the MGCP gateway must utilize the SIP OPTIONS ping feature to monitor status.
C. The MGCP gateway must have voice service VoIP configured.
D. The MGCP gateway and the Cisco Unified CM must be able to communicate over ports 2427, 2428, and 2727.

Correct Answer: D
Section: Cisco IOS XE Gateway and Media Resources
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/ios/voice/cminterop/configuration/guide/12_4t/vc_12_4t_book/vc_ucm_mgcp_gw.html **QUESTION**

67

```
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
```

Refer to the exhibit. Given this "debug isdn q921" output, what is the problem with the PRI?

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

Correct Answer: D
Section: Cisco IOS XE Gateway and Media Resources Explanation

Explanation/Reference:

QUESTION 68 An administrator is trying to change the default LINECODE for a voice ISDN T1 PRI. Which command makes this change?

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

Correct Answer: D
Section: Cisco IOS XE Gateway and Media Resources Explanation

Explanation/Reference:
Reference: https://www.cisco.com/en/US/docs/ios/dial/configuration/guide/dia_cfg_isdn_pri_external_docbase_0900e4b1806c752c_4container_external_docbase_0900e4b18216dd1b.html

QUESTION 69 Due to provider requirements, outgoing calls from the Enterprise to the PSTN must start with channel 1.

Which ISDN command changes the channel selection on IOS to meet this requirement?

A. **isdn bchan-number-order descending**
B. **isdn bchan-number-order ascending**
C. **isdn protocol-emulate network**
D. **isdn incoming-voice voice**

Correct Answer: B
Section: Cisco IOS XE Gateway and Media Resources Explanation

Explanation/Reference:

QUESTION 70

```
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
```

Refer to the exhibit. What is a possible cause of the PRI issue?

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

Correct Answer: D
Section: Cisco IOS XE Gateway and Media Resources Explanation

Explanation/Reference:

QUESTION 71


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

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

Refer to the exhibit. An engineer verifies the configuration of an MGCP gateway. The commands are already configured. Which command is necessary to enable MGCP?

A. **Device(config)# mgcp enable**
B. **Device(config)# ccm-manager enable**
C. **Device(config)# ccm-manager active**
D. **Device(config)# mgcp**

Correct Answer: D
Section: Cisco IOS XE Gateway and Media Resources Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-callmanager/42105-vg200-cfg.html

QUESTION 72

```
23031952: Apr  9 17:43:21.203 EDT: ISDN Se0/1/0:23 Q931: Applying typeplan for sw-type 0xD is 0x2 0x1, Calling num 4085556100
23031953: Apr  9 17:43:21.203 EDT: ISDN Se0/1/0:23 Q931: Sending SETUP callref = 0x12BE callID = 0xA3F5 switch = primary-ni interface = User
23031954: Apr  9 17:43:21.203 EDT: ISDN Se0/1/0:23 Q931: TX -> SETUP pd = 8 callref = 0x12BE
        Bearer Capability i = 0x8090A2
                Standard = CCITT
                Transfer Capability = Speech
                Transfer Mode = Circuit
                Transfer Rate = 64 kbit/s
        Channel ID i = 0xA98393
                Exclusive, Channel 19
        Progress Ind i = 0x8183 - Origination address is non-ISDN
        Calling Party Number i = 0x2181, '4085556100'
                Plan:ISDN, Type:National
        Called Party Number i = 0x91, '011443075552222'
                Plan:ISDN, Type:International
23031956: Apr  9 17:43:21.279 EDT: ISDN Se0/1/0:23 Q931: RX <- CALL_PROC pd = 8 callref = 0x92BE
        Channel ID i = 0xA98393
                Exclusive, Channed 19
23031957: Apr  9 17:43:21.283 EDT: ISDN Se0/1/0:23 Q931: RX <- PROGRESS pd = 8 callref = 0x92BE
        Cause i = 0x829F - Normal, unspecified
        Progress Ind i = 0x8488 - In-band info or appropriate now available
23031981: Apr  9 17:43:46.802 EDT: ISDN Se0/1/0:23 Q931: TX -> DISCONNECT pd = 8 callref = 0x12BE
        Cause i = 0x8090 - Normal call clearing
23031982: Apr  9 17:43:46.822 EDT: ISDN Se0/1/0:23 Q931: RX <- RELEASE pd = 8 callref = 0x92BE
23031983: Apr  9 17:43:46.822 EDT: ISDN Se0/1/0:23 Q931: TX -> RELEASE_COMP pd = 8 callref = 0x12BE
```

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

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

Correct Answer: A
Section: Cisco IOS XE Gateway and Media Resources Explanation

Explanation/Reference:

QUESTION 73

```
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
```

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

Correct Answer: A
Section: Cisco IOS XE Gateway and Media Resources Explanation

Explanation/Reference:

QUESTION 74
Which call flow matches traffic from a Mobile and Remote Access registered endpoint to central call control?

A. Endpoint > Expressway-E > Expressway-C > Cisco Unified CM
B. Endpoint > Expressway-E > Cisco Unified CM
C. Endpoint > Expressway-C > Cisco Unified CM
D. Endpoint > Expressway-C > Expressway-E > Cisco Unified CM

Correct Answer: A
Section: Call Control
Explanation

Explanation/Reference:

QUESTION 75
How does Cisco Unified Communications Manager perform a digit analysis on-hook versus off-hook for an outbound call from a Cisco IP phone that is registered to Cisco Unified CM?

A. On-hook, Unified CM performs a digit-by-digit analysis; off-hook, Unified CM considers all digits were dialed and does not wait for additional digits. B. On-hook, Unified CM considers all digits were dialed and does not wait for additional digits; off-hook, Unified CM performs a digit-by-digit analysis.
C. On-hook, by pressing the digits and entering "#" to process the call, Unified CM performs a digit-by-digit analysis; off-hook, Unified CM analyzes all digits as a string.
D. On-hook, no digit analysis is performed; off-hook, Unified CM requires the "**#*** to start the digit analysis.

Correct Answer: C
Section: Call Control
Explanation

Explanation/Reference:

QUESTION 76
How does an administrator make a Cisco IP phone display the last 10 digits of the calling number when the call is in the connected state, and also display the calling number in the E.164 format within call history on the phone?

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

Correct Answer: D
Section: Call Control
Explanation

Explanation/Reference:

QUESTION 77 Which endpoint feature is supported using Mobile and Remote Access through Cisco Expressway?

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

Correct Answer: C
Section: Call Control
Explanation

Explanation/Reference:

QUESTION 78 When a phone is registered over Mobile and Remote Access, where does it register?

A. Cisco Unified Presence Server
B. Expressway-E
C. Cisco Unified Communications Manager
D. Expressway-C

Correct Answer: B
Section: Call Control
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_b_system-configuration-guide-1201/cucm_b_system-configuration-guide-1201_chapter_01011010.html

QUESTION 79 Which description of the Mobile and Remote Access feature is true?

A. Collaboration Edge feature that enables remote individuals to perform international calls from Jabber with a VPN connection.
B. Collaboration Edge feature that enables remote individuals to access all enterprise collaboration services using a PC within the corporate environment.C. Collaboration Edge feature that enables remote individuals to access enterprise collaboration services via Jabber without the use of a VPN connection.
D. Collaboration Edge feature that enables remote individuals to access enterprise collaboration services via Jabber with the use of a VPN connection.

Correct Answer: C
Section: Call Control
Explanation

Explanation/Reference:

QUESTION 80

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

Refer to the exhibit. 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 contains “-”.
B. The translation rule is not matched because DNIS does not end with a “$”.
C. The translation rule is matched and the translated number is 02553431234.

D. The translation rule is matched and the translated number is 025553431234.

Correct Answer: A
Section: Call Control
Explanation

Explanation/Reference:

QUESTION 81 What field is configured to change the caller ID information on a SIP route pattern?

A. Route Partition
B. Called Party Transformation Mask
C. Calling Party Transformation Mask
D. Connected Line ID Presentation

Correct Answer: D
Section: Call Control
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0100111.pdf

QUESTION 82 An engineer configures Cisco Unified Communications Manager to prevent toll fraud. At which two points does the engineer block the pattern in Cisco Unified CM to complete this task? (Choose two.)

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

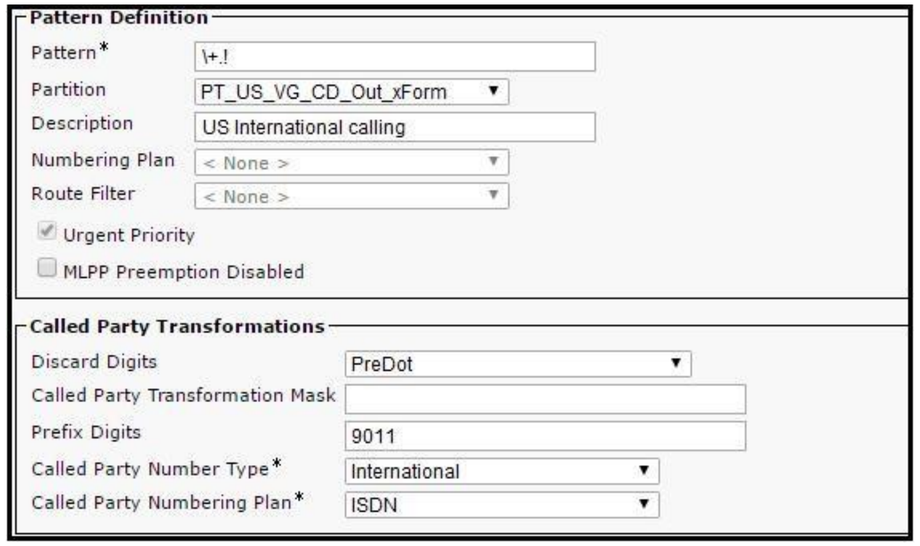
Correct Answer: CE
Section: Call Control
Explanation

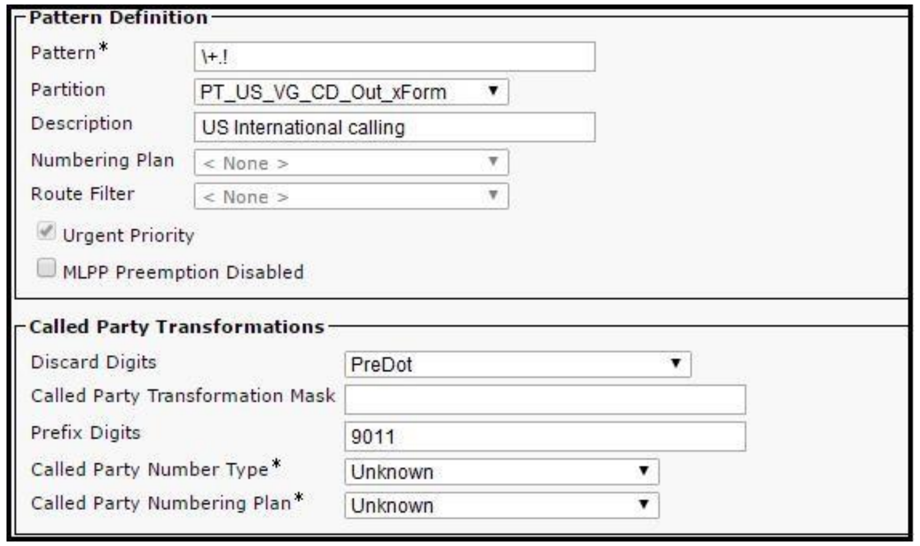
Explanation/Reference:

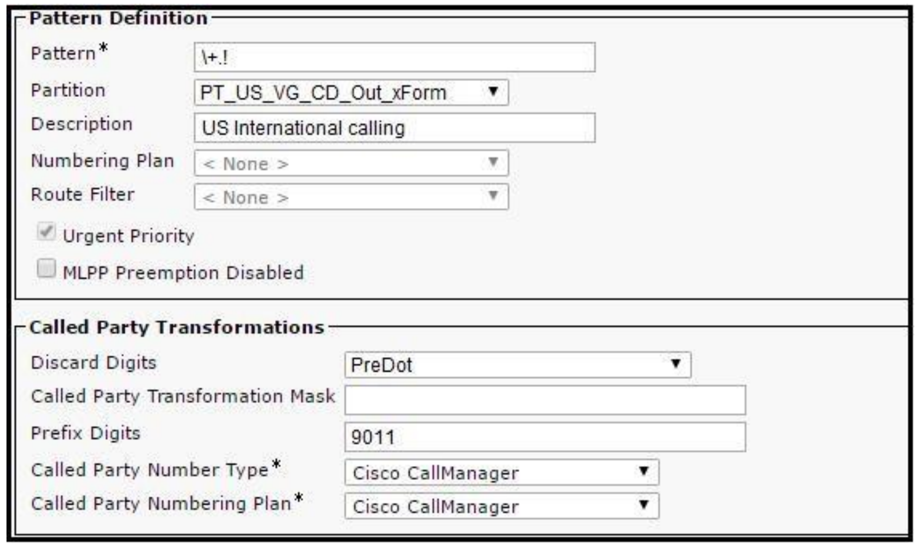
QUESTION 83
Given the H.323 gateway configuration and using Cisco best practices, how must the called party transformation pattern be configured to ensure that a proper ISDN type of number is set?

```
voice translation-rule 40
 rule 1 /3...$/ /408555&/
!
voice translation-profile INT
 translate calling 40
!
dial-peer voice 9011 pots
 translation-profile outgoing INT
 destination-pattern 9011T
 port 0/1/0:23
```

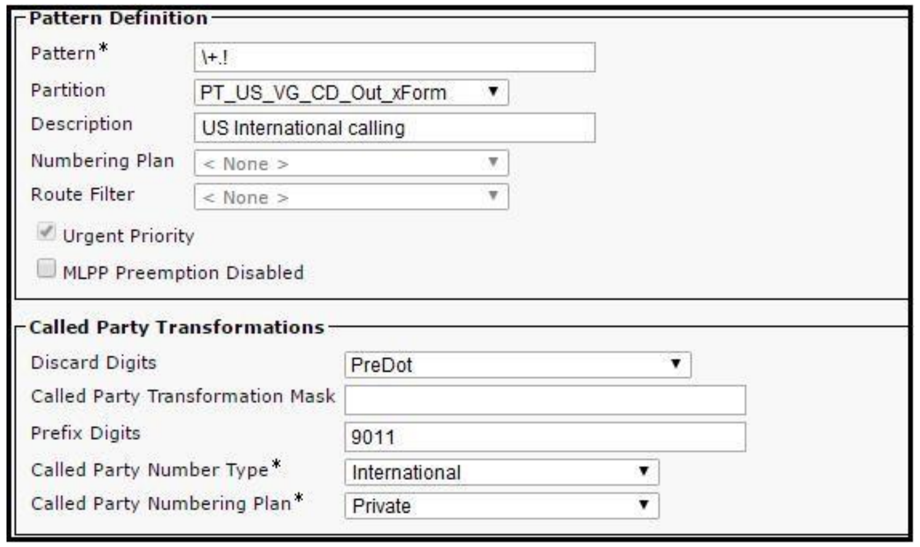
A.


Pattern Definition
Pattern* \+.!
Partition PT_US_VG_CD_Out_xForm
Description US International calling
Numbering Plan < None >
Route Filter < None >
Urgent Priority
MLPP Preemption Disabled
Called Party Transformations
Discard Digits PreDot
Called Party Transformation Mask
Prefix Digits 9011
Called Party Number Type* International
Called Party Numbering Plan* ISDN


Pattern Definition
Pattern* \+.!
Partition PT_US_VG_CD_Out_xForm
Description US International calling
Numbering Plan < None >
Route Filter < None >
Urgent Priority
MLPP Preemption Disabled
Called Party Transformations
Discard Digits PreDot
Called Party Transformation Mask
Prefix Digits 9011
Called Party Number Type* Unknown
Called Party Numbering Plan* Unknown


Pattern Definition
Pattern* \+.!
Partition PT_US_VG_CD_Out_xForm
Description US International calling
Numbering Plan < None >
Route Filter < None >
Urgent Priority
MLPP Preemption Disabled
Called Party Transformations
Discard Digits PreDot
Called Party Transformation Mask
Prefix Digits 9011
Called Party Number Type* Cisco CallManager
Called Party Numbering Plan* Cisco CallManager

B. C.

D. 

Correct Answer: C
Section: Call Control
Explanation

Explanation/Reference:

QUESTION 84
An engineer with ID012345678 must build an international dial plan in Cisco Unified Communications Manager. Which action should be taken when building a variable-length route pattern?

A. reduce the T302 timer to less than 4 seconds
B. configure single route pattern for international callsC. create a second route pattern followed by the # wildcard
D. set up all international route patterns to 0.!

Correct Answer: A
Section: Call Control
Explanation

Explanation/Reference:

QUESTION 85 An engineer configures local route 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. CSS
B. route pattern
C. device pool
D. route list

Correct Answer: D
Section: Call Control
Explanation

Explanation/Reference:

QUESTION 86
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

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 aacld**

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 codec mp4a-latm**

Correct Answer: B
Section: Call Control
Explanation

Explanation/Reference:

QUESTION 87 Which settings are needed to configure the SIP route pattern in Cisco Unified Communications Manager?

A. pattern usage, IPv6 pattern, and SIP trunk/Route list
B. pattern usage, IPv4 pattern, IPv6 pattern, and description
C. pattern usage, IPv4 pattern, and SIP trunk/Route list
D. SIP trunk/Route list, description, and IPv4 pattern

Correct Answer: C
Section: Call Control
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0100111.pdf

QUESTION 88 Which call routing pattern is used for phone numbers that are in the E.164 format?

A. \+.! Route Pattern
B. \+.! Translation Pattern
C. /+.! Route Pattern
D. \+1.[2-9]XX[2-9]XXXXXX Called Party Transformation Pattern

Correct Answer: D

Section: Call Control
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12/dialplan.html#pgfId-1591747

QUESTION 89
As a voice engineer, which two recommendations do you to make to your company to optimize Cisco Unified Communications Manager configuration to reduce the number of toll fraud incidents? (Choose two.)

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

Correct Answer: BE
Section: Call Control
Explanation

Explanation/Reference:

QUESTION 90 Which wildcard must an engineer configure to match a whole domain in SIP route patterns?

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

Correct Answer: D
Section: Call Control
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0100111.html

QUESTION 91
Multiple route patterns match a number. How does Cisco Unified Communications Managers determine which pattern to use?

A. the one that comes first in numerical order
B. the one with the longest match
C. the one with the closest match
D. the one that discards everything PreDot

Correct Answer: C
Section: Call Control
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-callmanager/13920-call-routing.html#bcr

QUESTION 92
Which action prevents toll fraud in Cisco Unified Communications Manager?
Implement toll fraud restriction in the Cisco IOS router. B.
Implement route patterns in Cisco Unified CM.
C. Allow off-net to off-net transfers.
D. Configure ad hoc conference restriction.

A.

Correct Answer: B
Section: Call Control
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-express/107626-cme-toll-fraud.html

QUESTION 93 A user forwards a corporate number to an international number. What are two methods to prevent this forwarded call? (Choose two.)

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

Correct Answer: BC
Section: Call Control
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmetrans.html

QUESTION 94
Calls are being delivered to the end user in the globalized format. Where does an engineer configure the calling number into a localized format?

A. route pattern
B. service parameters
C. IP phone
D. gateway

Correct Answer: C
Section: Call Control
Explanation

Explanation/Reference:

QUESTION 95 Which method is used to avoid toll fraud with Cisco Unified Communications Manager calls?

A. call policy service
B. TOLLFRAUD_APP
C. default zone access rules
D. class of service

Correct Answer: D
Section: Call Control
Explanation

Explanation/Reference:

QUESTION 96
Which two configuration elements are part of the Cisco Unified Communications Manager toll-fraud prevention? (Choose two.)

A. SIP trunk security profile
B. Calling Search Space

C. SUBSCRIBE Calling Search Space
D. feature control policy
E. partition

Correct Answer: BE
Section: Call Control
Explanation

Explanation/Reference:

QUESTION 97 How are E.164 called-party numbers normalized on a globalized call-routing environment in Cisco Unified Communications Manager?

A. Normalization is achieved by stripping or translating the called numbers to internally used directory numbers.
B. Normalization is achieved by setting up calling search spaces and partitions at the SIP trunks for PSTN connection.
C. Call ingress must be normalized before the call being routed.
D. Normalization is not required.

Correct Answer: A
Section: Call Control
Explanation

Explanation/Reference:

QUESTION 98 Which user group is targeted by MRA services?

A. production floor users who need wireless mobility in remote areas of the factory
B. mobile workers in a hot desk environment at HQ who log in every morning at possibly a different desk phone
C. on-the-go mobile workforce who connect to corporate phone services using their own mobile device
D. call center agents who dial out to remote customers

Correct Answer: C
Section: Call Control
Explanation

Explanation/Reference:

QUESTION 99 Which Cisco UCM feature is used to determine the maximum bit rate for a call between two video endpoints?

A. partitions
B. locations
C. regions
D. transformations

Correct Answer: C
Section: Call Control
Explanation

Explanation/Reference:

Explanation:
Regions provide capacity controls for Unified Communications Manager multi-site deployments where you may need to limit the bandwidth for certain calls. For example, you can use regions to limit the bandwidth for calls that are sent across a WAN link, while maintaining a higher bandwidth for internal calls. You can use regions to limit the bandwidth for audio and video calls by setting the maximum bitrate for intraregional or interregional calls to whatever the region(s) can provide.

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/11_5_1_SU1/cucm_b_system-configuration-guide-1151su1/cucm_b_system-configuration-guide-1151su1_chapter_0111.html

QUESTION 100
An engineer implements QoS in the enterprise network. Which command can be used to verify the correct 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 policy-map**
D. **show access-lists**

Correct Answer: C
Section: QoS
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/qos_classn/configuration/xe-16/qos-classn-xe-16-book/qos-classn-mrkg-ntwk-trfc-xe.html

QUESTION 101 What is a characteristic of video traffic that governs QoS requirements for video?

A. Video is typically variable bit rate.
B. Voice and video traffic are different, but they have the same QoS requirements.
C. Video is typically constant bit rate.
D. Voice and video traffic are the same, so they have the same QoS requirements.

Correct Answer: B
Section: QoS
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/cac.html

QUESTION 102 What causes poor voice quality and video pixelization in a video call?

A. The QoS is configured incorrectly.
B. A firewall is blocking the RTP ports.
C. Cisco Unified Communications Manager is configured to use G.711 instead of G.729.
D. 1 Gbps network ports are used instead of 100 Mbps network ports.

Correct Answer: A
Section: QoS
Explanation

Explanation/Reference:

QUESTION 103 What is a valid class included in the 8-Class QoS Strategy in a VoIP network?

A. Assured Forwarding
B. Broadcast Video
C. Multimedia Conferencing

D. Real-Time Interactive

Correct Answer: C
Section: QoS
Explanation

Explanation/Reference:
Reference: https://www.ciscopress.com/articles/article.asp?p=2756478&seqNum=8

QUESTION 104 Which issue can occur if QoS is not deployed on a Cisco Collaboration architecture across the WAN?

A. 403 Forbidden errors on SIP calls
B. excessive jitter
C. unexpected shut-down on Cisco Unified Communications Manager
D. packet fragmentation

Correct Answer: B
Section: QoS
Explanation

Explanation/Reference:

QUESTION 105 Which statement describes the outcome when the trust boundary is defined at the Cisco IP phone?

A. Packets or Ethernet frames are remarked at the access layer switch.
B. Packets or Ethernet frames are not remarked by the IP phone.
C. Packets or Ethernet frames are not remarked at the access layer switch.
D. Packets or Ethernet frames are remarked at the distribution layer switch.

Correct Answer: C
Section: QoS
Explanation

Explanation/Reference:
Reference: https://networklessons.com/quality-of-service/how-to-configure-qos-trust-boundary-on-cisco-switches

QUESTION 106 Why would we not include an end user's PC device in a QoS trust boundary?

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

Correct Answer: B
Section: QoS
Explanation

Explanation/Reference:

QUESTION 107
DRAG DROP

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

Select and Place:

Answer Area

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

Correct Answer:

Answer Area

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

Section: QoS
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/solutions/Enterprise/WAN_and_MAN/QoS_SRND/QoS-SRND-Book/QoSIntro.html

QUESTION 108 An engineer with troubleshoots poor voice quality on multiple calls. After looking at packet captures, the engineer notices high levels of jitter. Which two areas does the engineer check to prevent jitter? (Choose two.)

A. The network meets bandwidth requirements.
B. MTP is enabled on the SIP trunk to Cisco Unified Border Element.
C. Cisco UBE manages voice traffic, not data traffic.
D. All devices use wired connections instead of wireless connections.

E. Voice packets are classified and marked.

Correct Answer: AE
Section: QoS
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/support/docs/voice/voice-quality/20371-troubleshoot-qos-voice.html

QUESTION 109 Which configuration tells a switch port to send Cisco Discovery Protocol packets that configure an attached Cisco IP phone to trust tagged traffic that is received from a device that is connected to the access port on the Cisco IP phone?

A. **Router# configure terminal**
Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos trust extend
B. **Router# configure terminal**
Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos trust extend cos 3
C. **Router# configure terminal**
Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos trust extend cos 5
D. **Router# configure terminal**
Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos extend trust

Correct Answer: A
Section: QoS
Explanation

Explanation/Reference:

QUESTION 110
Which recommendation is the best practice for marking video and voice media in a Cisco Unified Communications network?

A. Voice Cos 5 (IP Precedence 6, PHB AF41, or DSCP 16)Video Cos 4 (IP Precedence 5, PHB EF, or DSCP 32)
B. Voice Cos 6 (IP Precedence 4, PHB AF41, or DSCP 24)Video Cos 5 (IP Precedence 4, PHB EF, or DSCP 34)
C. Voice Cos 5 (IP Precedence 2, PHB EF, or DSCP 48)
Video Cos 4 (IP Precedence 4, PHB AF41, or DSCP 46)
D. Voice Cos 5 (IP Precedence 5, PHB EF, or DSCP 46)
Video Cos 4 (IP Precedence 4, PHB AF41, or DSCP 34)

Correct Answer: D
Section: QoS
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/netstruc.html

QUESTION 111
How can an engineer determine location-based CAC bandwidth requirements for Cisco Unified Communications Manager?

A. Set the requirements in the service parameters.
B. Add the requirements for each audio and video codec and multiply how many calls must be supported.
C. Execute the Resource Reservation Protocol to return location-based requirements.
D. Calculate the number of calls against the license for Cisco Unified Border Element to determine calls per location.
Correct Answer: A

Section: QoS
Explanation

Explanation/Reference:

QUESTION 112 There is a saturated link that has traffic shaping configured. How is incoming traffic processed?

A. Traffic is compressed so that the traffic fits within the bandwidth of the link.
B. Excess traffic is queued, and then dropped after the timer expires.
C. Excess traffic is queued for later transmission.
D. Excess traffic is dropped.

Correct Answer: B
Section: QoS
Explanation

Explanation/Reference:
Reference:
https://www.cisco.com/c/en/us/support/docs/quality-of-service-qos/qos-policing/19645-policevsshape.html

QUESTION 113 Which field of a Real-Time Transport Protocol packet allows receiving devices to detect lost packets?

A. timestamp
B. CSRC (Contributing source ID)
C. sequence number
D. SSRC (Synchronization source identifier)

Correct Answer: C
Section: QoS
Explanation

Explanation/Reference:
Reference:
https://www.oreilly.com/library/view/packet-guide-to/9781449339661/ch04.html

QUESTION 114 Which DiffServ marking is the most likely to drop packets?

A. AF13
B. AF12C. AF11
D. AF32

Correct Answer: A
Section: QoS
Explanation

Explanation/Reference:
Reference:
https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/qos_dfsrv/configuration/15-mt/qos-dfsrv-15-mt-book/qos-dfsrv.html

QUESTION 115
A Cisco Unity Connection Administrator must set a voice mailbox so that it can be accessed from a secondary device. Which configuration on the voice mailbox makes this change?

A. Attempt Forward routing rule
B. Alternate Extensions

C. Alternate NamesD. Mobile User

Correct Answer: A
Section: Collaboration Applications
Explanation

Explanation/Reference:

QUESTION 116 Which Cisco Unified Communications Manager configuration is required for SIP MWI integrations?

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

Correct Answer: A
Section: Collaboration Applications
Explanation

Explanation/Reference:

QUESTION 117 Which configuration on Cisco Unified Communications Manager is required for SIP MWI to work?

A. The line partition must be inside the inbound CSS assigned to the CUC SIP trunk.
B. The line partition must be inside the rerouting CSS assigned to the Cisco Unity Connection SIP trunk.
C. Set the “Enable message waiting indicator” on the port group.
D. Assign a MWI extension on the mailbox.

Correct Answer: C
Section: Collaboration Applications
Explanation

Explanation/Reference:

QUESTION 118

```
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
         srv hostname   = expe.example.com
```

Refer to the exhibit. You deploy Mobile and Remote Access for Jabber and discover that Jabber for Windows does no register to Cisco Unified Communication Manager while outside of the office.

What is a cause of this issue?

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

Correct Answer: C
Section: Collaboration Applications
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/jabber/Windows/9_7/CJAB_BK_C606D8A9_00_cisco-jabber-dns-configuration-guide/CJAB_BK_C606D8A9_00_cisco-jabber-dns-configuration-guide_chapter_010.html

QUESTION 119 Which protocols does Cisco IM and Presence use to authenticate Jabber?

A. XMPP
B. SOAP
C. TCP
D. LDAP
E. QBE

Correct Answer: AB
Section: Collaboration Applications
Explanation

Explanation/Reference:

QUESTION 120
An engineer is configuring a BOT device for a Jabber user in Cisco Unified Communication Manager Which phone type must be selected?

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

Correct Answer: C
Section: Collaboration Applications
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/jabber/11_5/CJAB_BK_D00D8CBD_00_deployment-installation-guide-cisco-jabber115/CJAB_BK_D00D8CBD_00_deployment-installation-guide-ciscojabber115_chapter_01000.html

QUESTION 121 Which access control group is required on an end user to allow Jabber to do deskphone mode?

A. Allow Control of Device from CTI
B. Standard CTI Enabled
C. Standard CTI Allow Reception of SRTP Key Material
D. Standard CTI Secure Connection

Correct Answer: B
Section: Collaboration Applications
Explanation
Explanation/Reference:

QUESTION 122
Which two DNS records must be created to configure Service Discovery for on-premises Jabber? (Choose two.)

A. _cisco-uds._tls.cisco.com pointing to the IP address of Cisco Unified Communications Manager
B. _cuplogin._tcp.cisco.com pointing to a record of IM&P
C. _cuplogin._tls.cisco com pointing to the IP address of IM&P
D. _cisco-uds._tcp.cisco com painting to a record of Cisco Unified CM
E. _xmpp._tls.cisco.com pointing to a record of IM&P

Correct Answer: AB
Section: Collaboration Applications
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/jabber/Windows/9_7/CJAB_BK_C606D8A9_00_cisco-jabber-dns-configuration-guide/CJAB_BK_C606D8A9_00_cisco-jabber-dns-configuration-guide_chapter_010.html

QUESTION 123 What is the element of Cisco Collaboration infrastructure that allows Jabber clients outside of the network to register in Cisco Unified Communications Manager and use its resources?

A. Cisco IM and Presence node
B. Cisco Unified Border Element
C. Cisco Expressway
D. Cisco Prime Collaboration Provisioning server

Correct Answer: C
Section: Collaboration Applications
Explanation

Explanation/Reference:

QUESTION 124 Regarding SIP integrations with Cisco Unified Communications Manager, if the Cisco Unity Connection is configured to listen for incoming IPv4 and IPv6 traffic, how should the addressing mode be set up in the Cisco Unity Connection?

A. Set up is not required.
B. Set up for each group to use IPv4 and IPv6.
C. Set up media ports for each port group to use IPv4.
D. Set up IPv4 and IPv6 in Cisco Unified CM.

Correct Answer: B
Section: Collaboration Applications
Explanation

Explanation/Reference:

QUESTION 125 Which description of the function of call handlers in Cisco Unity Connection is true?

A. They answer calls, take messages, and provide menus of options.
B. They provide access to a corporate directory by playing an audio list that users and outside callers use to reach users and leave messages.
C. They collect information from callers by playing a series of questions and recording the answers.
D. They control outgoing calls by allowing you to specify the numbers that Cisco Unity Connection can dial to transfer calls, notify users of messages, and deliver faxes.

Correct Answer: A
Section: Collaboration Applications
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/10x/administration/guide/10xcucsagx/10xcucsag080.html

QUESTION 126 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 outage. Active/standby mode provides an always on standby node in the event of an outage, and it also provides load balancing.
B. Balanced mode provides user load balancing and user failover only for manually generated failovers. Active/standby mode provides an unconfigured standby node in the event of an outage, but it does not provide load balancing.
C. 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.
D. Balanced mode does not provide user load balancing, but it provides 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.

Correct Answer: C
Section: Collaboration Applications
Explanation

Explanation/Reference:

QUESTION 127
Which type of greeting in the Call Handler configuration in Cisco Unity Connection overwrites all other greetings?

A. supervisory
B. alternate
C. holiday
D. priority

Correct Answer: B
Section: Collaboration Applications
Explanation

Explanation/Reference:
Explanation:
Alternate
Can be used for a variety of special situations, such as vacations or a leave. An alternate greeting overrides all other greetings.
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/10x/administration/guide/10xcucsagx/10xcucsag080.html

QUESTION 128 Which protocol is used between Cisco Jabber clients for instant messaging and presence?

A. XMPP
B. P2P
C. SIP/SIMPLE
D. Jabber

Correct Answer: A
Section: Collaboration Applications
Explanation

Explanation/Reference:
Explanation:
An XMPP connection handles the presence information exchange and instant messaging operations for XMPP-based clients.

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/10_5_1/CUP0_BK_CE43108E_00_config-admin-guide-imp-105/CUP0_BK_CE43108E_00_config-admin-guide-imp-105_chapter_00.html