Cisco

Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

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

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

Answer: D

NEW QUESTION 2
Refer to the exhibit.

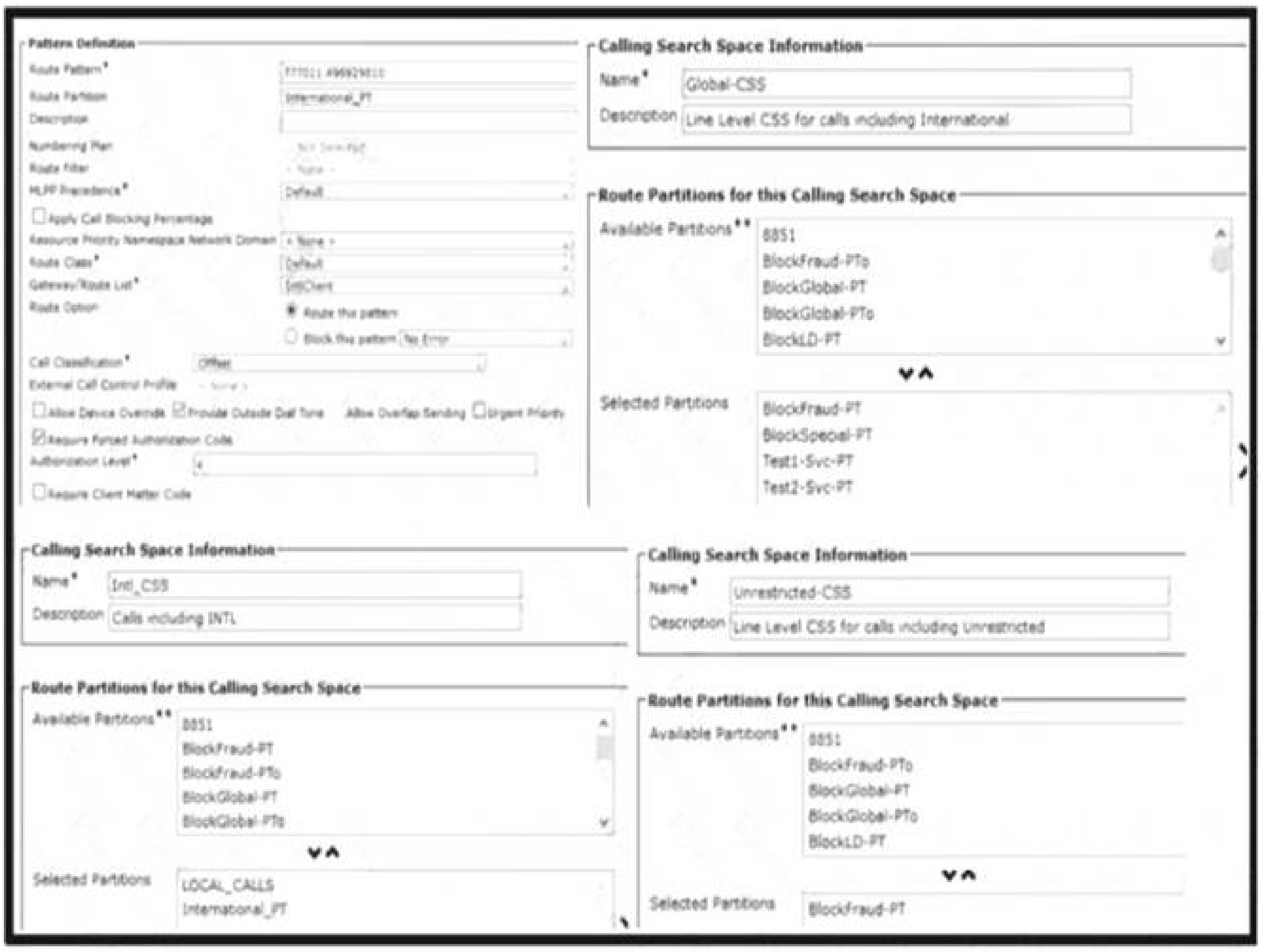
```
Gateway1#show sccp
SCCP Admin State: UP
Gateway Local Interface: Loopback0
        IPv4 Address: 192.168.12.1
        Port Number: 2000

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

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

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

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

Answer: BC

NEW QUESTION 3
Refer to the exhibit.



How must the +E.164 translation pattern be configured to reach international number 496929810?

Pattern= \+.496929810, CSS=Unrestricted-CSS, PreDot, Prefix=777011

Pattern= \+.777011496929810, CSS=Intl_CSS

Pattern= \+.011496929810, CSS=Global-CSS, PreDot, Prefix=777

Pattern= \+.496929810, CSS=Intl_CSS, PreDot, Prefix=777011

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

Answer: C

NEW QUESTION 4
What are two QoS requirements for VoIP traffic?

A. Voice traffic must be marked "to DSCP EF.
B. Loss must be no more man 1 percent.
C. Voice traffic must be marked to DSCP AF41.
D. One-way latency must be no more than 200 ms.
E. Average one-way jitter is greater than 50 ms.

Answer: AB

NEW QUESTION 5
How does an administrator make a Cisco IP phone display the last 10 digits of the calling number when the call is in the connected state, and also display the calling number in the E.164 format within call history on the phone?

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

Answer: A

NEW QUESTION 6
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 7
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 8
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 9
Callers from a branch report getting busy tones intermittetly when trying to reach colleagues in other office branches during peak hours. An engineer collects Cisco CallManager service traes to examine the situation. The traces show:

```
50805567.000 |07:35:39.676 |Sdl Sig |StationOoutputDisplayNotify |restart0
|StaatinD(1,100,63,6382) |StionCdpc(1,100,64,4725) |1,100,40,6.709919^*^*
|[R:N-H:0,L:0,V:0,Z:0,D:0] TimeOutValue=10 Status=x807 Unicode Status=Locale=1
50805567.001 |07:35:39.676 |AppInfo |StationD: (0006382) DisplayNotify
timeOutValue=10 notify='x807' content='Not Enough Bandwidth' ver=85720014.
```

What should be fixed to resolve the issue?

A. class of service configuration
B. region configuration
C. geolocation configuration
D. codec configuration

Answer: B

NEW QUESTION 10
Which dial plan function restricts calls that are made by a lobby phone to internal extensions only?

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

Answer: C

NEW QUESTION 10
Refer to the exhibit. Which two codec permutations should be transcoded by this dspfarm? (Choose two.)

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

Answer: CE

NEW QUESTION 11
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 14
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 16
What are two common attributes of XMPP XML stanzas? (Choase two.)

A. from
B. to
C. destination
D. version
E. Source

Answer: AB

NEW QUESTION 20
Refer to the exhibit.

NAME	TTL	CLASS	TYPE	Priority	Weight	Port	Target Address
_sip._tcp.sample.com	86400	IN	SRV	10	60	5060	server1.sample.com
_sip._tcp.sample.com	86400	IN	SRV	10	30	5060	server2.sample.com
_sip._tcp.sample.com	86400	IN	SRV	5	20	5060	server3.sample.com

An administrator must fix the SRV records to ensure that server1. sample.com is always contacted first from the three servers. Which solution should the engineer apply to resolve this issue?

A. Priority = 100, Weight = 90
B. Priority = 10, Weight = 5
C. Priority = 10, Weight = 10
D. Priority = 5, Weight = 70

Answer: D

NEW QUESTION 21
Which SNMP service must be activated manually on the Cisco Unified Communications Manager after installation?

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

Answer: A

NEW QUESTION 23
An administrator must configure the Local Route Group feature on Cisco UCM. Which step will enable this feature?

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

Answer: C

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

NEW QUESTION 25
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 27
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 29
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 31
Where in Cisco UCM is restrictions on audio bandwidth configured?

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

Answer: C

NEW QUESTION 32
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 37
On a Cisco Catalyst Switch, which command is required to send CDP packets on a switch port that configures a Cisco IP phone to transmit voice traffic in 802.10 frames, tagged with the voice VLAN ID 221?

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

Answer: A

NEW QUESTION 38
What are two features of Cisco Expressway that the customer gets if Expressway-C and Expressway-E are deployed?(Choose two.)

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

Answer: AD

NEW QUESTION 39
A company hosts a conference call with no local users. How does the administrator stop the conference from continuing?

A. modifies the Drop Ad Hoc Conference service parameter
B. modifies the Block OffNet to OffNet Transfer service parameter
C. removes the transcoder
D. changes the codecs that are supported on the conference resource

Answer: A

NEW QUESTION 43
What is the maximum number of servers that are in an IM and Presence presence redundancy group?

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

Answer: C

NEW QUESTION 47
How does traffic policing respond to violations?

A. Excess traffic is dropped.
B. Excess traffic is retransmitted.
C. All traffic is treated equally.
D. Excess traffic is queued.

Answer: A

NEW QUESTION 52
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 54
An engineer is notified that the Cisco TelePresence MX800 that is registered in Cisco Unified communications Manager shows an empty panel, and the Touch 10 shows a corresponding icon with no action when pressed. Where does the engineer go to remove the inactive custom panel?

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

Answer: D

NEW QUESTION 55
The chief officer at a company must reduce collaboration infrastructure costs by onbording all on-permises equipment to the cloud by using CISCO Webex Control Hub. Administrators need the ability to manage upgrades and set up hot desking for on-premises devices.
Which action must be taken before on boardng devices by using the Control Hub?

A. Configure tie Control Hub organization ID on the devices
B. Acquire a license for each device.
C. Allow HTTP traffic from each device to Control Hub.
D. Upgrade all the devices to software version CE9.15 or later

Answer: D

Explanation:
This is a prerequisite for using the Device Connector tool, which allows you to onboard and register several devices simultaneously to the Webex Control Hub1. The Device Connector tool creates a workspace, an activation code, and activates all of your devices in one go1. This way you don’t need to be physically present in the same room to activate the devices.
The other options are not required before onboarding devices by using the Control Hub:
- Configuring the Control Hub organization ID on the devices is not necessary, as the Device Connector tool will send the device information to your Webex organization and generate activation codes for them 1.
- Acquiring a license for each device is not necessary, as you can assign licenses to users and devices after they are registered to the Webex Control Hub2.
- Allowing HTTP traffic from each device to Control Hub is not necessary, as HTTPS connectivity is required for the Device Connector tool to communicate with the devices1.

NEW QUESTION 56
An engineer roust deploy the Cisco Wet*x app to a Windows Virtual Desktop Infrastructure environment that has a roaming database named spark roaming_store stored In a user's AppData\Roaming directory, Which two command line arguments must be used when running the installer? (Choose two.)

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

Answer: BE

Explanation:
The Cisco Webex app can be installed on a Windows Virtual Desktop Infrastructure (VDI) environment by using the following command-line arguments:
- ENABLEVDI=1 - This argument enables VDI mode for the Webex app.
- ROAMINGENABLED=1 - This argument enables roaming for the Webex app.

The ALLUSERS argument is not required when installing the Webex app on a VDI environment. The ENABLEVDI argument must be set to 1, and the ROAMINGENABLED argument must be set to 1.
The following is an example of the command that can be used to install the Webex app on a VDI environment:
Code snippet
msiexec /i WebexApp.msi ENABLEVDI=1 ROAMINGENABLED=1

NEW QUESTION 59
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 60
An engineer implements a new Cisco UCM based telephony system per these requirements.

•The local Ethernet bandwidth is sized based on the total bandwidth per call
•A G 736 codec is used.
•The bit rate is 64 kbps
•The codec sample interval is 10 ms
•The voice payload size is 160 bytes per 20 ms
What should the size of the Ethernet bandwidth be per call?

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

Answer: D

NEW QUESTION 62
What are two functions of Cisco Expressway in the Collaboration Edge? (Choose two.)

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

Answer: CE

NEW QUESTION 66
A customer is deploying a SIP IOS gateway for a customer who requires that in-band DTMF relay is first priority and out-of-band DTMF relay is second priority. Which 10$ entry sets the required priority?

A. dtmf-relay cisco-rtp
B. dtmf-relay sip-kpml cisco-rtp
C. sip-notify dtmf-relay rtp-nte
D. dtmf-relay rtp-nte sip-notify

Answer: D

NEW QUESTION 70
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 71
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 73
A customer enters no IP domain lookup on the Cisco IOS XE gateway to suppress the interpreting of invalid commands as hostnames Which two commands are needed to restore DNS SRV or A record resolutions? (Choose two.)

A. ip dhcp excluded-address
B. ip dhcp-sip
C. ip dhcp pool
D. transport preferred none
E. ip domain lookup

Answer: DE

NEW QUESTION 77
Users dial a 9 before a 10-digit phone number to make an off-net call All 11 digits are sent to the Cisco Unified Border Element before going out to the PSTN The PSTN provider accepts only 10 digits. Which configuration is needed on the Cisco Unified Border Element for calls to be successful?

A. voice translation-rule 1 rule 1 /^9/ //
B. voice translation-rule 1 rule 1 /^9(..........)/ //
C. voice translation-rule 1 rule 1 /^9.+/ //
D. voice translation-rule 1 rule 1 /^9........./ //

Answer: A

NEW QUESTION 78
SIP proxies have operations defined in RFC 3261 and supporting extensions. Though no IETF RFC completely defines how SBCs must function. SBCs evolved over the years.
Which two operations demonstrate the high-level differences between SBCs and SIP proxies? (Choose two.)

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

Answer: BD

NEW QUESTION 79
The security department will audit an IT department to ensure that the proper guidelines are being followed. The reports of the call detail records show unauthorized access to PSTN. Which two actions should an administrator check to prevent the unauthorized use of the telephony system? (Choose two.)

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

Answer: BE

NEW QUESTION 80
An engineer is configuring IP telephony. The network relies on DHCP to provide TFTP server addresses to the endpoints. Policy requires the endpoints to receive two server addresses. Which DHCP option must be configured?

A. 66

B. 143
C. 150
D. 166

Answer: C

NEW QUESTION 81
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 86
A customer asked to integrate Unity Connection with Cisco UCM using SIP protocol. Which two features must be enabled on SIP security profiles? (Choose two.)

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

Answer: CE

NEW QUESTION 89
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 93
Refer to the exhibit.

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

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

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

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

Answer: D

NEW QUESTION 97
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. media resources group list
B. CSS
C. location
D. device security profile
E. SIP profile

Answer: DE

NEW QUESTION 101
How many minutes does it take for automatic fallback to occur in a Presence Redundancy Group if the primary node lost a critical service?

A. 5 min
B. 10 min
C. 30 min
D. 60 min

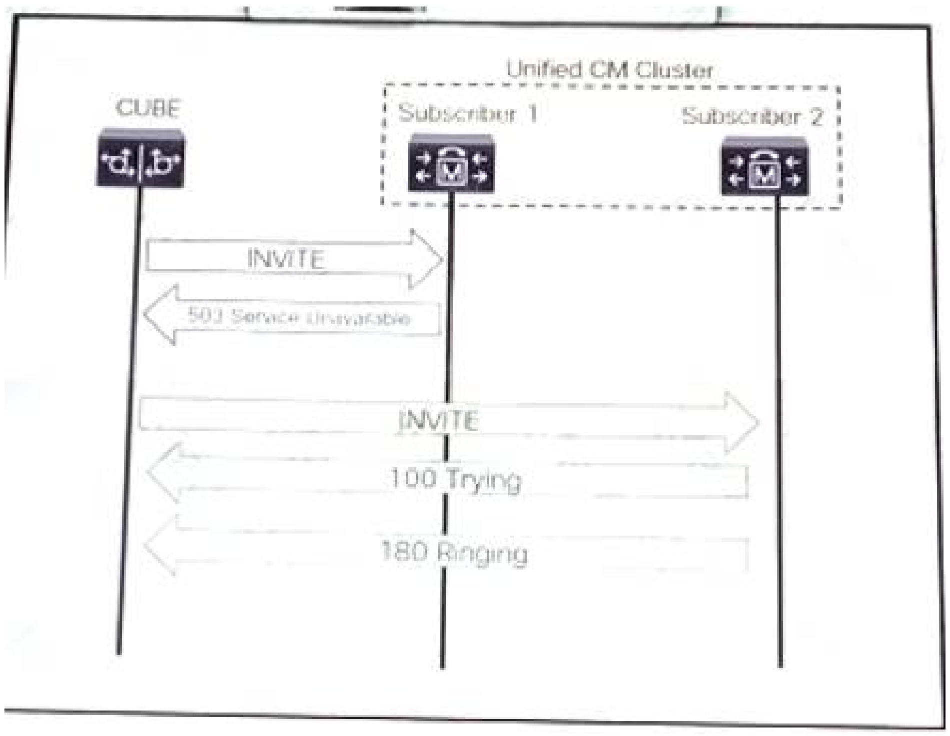
Answer: C

NEW QUESTION 102
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 105
Refer to the exhibit.



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

A. Verify that the correct calling search space is selected for the inbound Calls section
B. Verify that the run on all active United CM Nodes checkbox is enabled
C. Verify that the Significant Digits field for inbound Calls is set to All.
D. Verify that the PSTN Access checkbox is enabled.

Answer: B

NEW QUESTION 110
Refer to the exhibit.

```
    Bearer Capability i = 0x8090A2
        Standard = CCITT
        Transfer Capability = Speech
        Transfer Mode = Circuit
        Transfer Rate = 64 kbit/s
    Channel ID i = 0xA98388
        Exclusive, Channel 8
    Calling Party Number i = 0x2181, '5125551212'
        Plan: ISDN, Type:National
    Called Party Number i = 0xA1, '2145551212'
        Plan:ISDN, Type:National
Mar  1 02:35:37: ISDN Se0/1/1:23 Q931: RX <- CALL_PROC pd = 8  callref = 0x809A
    Channel ID i = 0xA98388
        Exclusive, Channel 8

interface Serial0/1/1:23
 description PRI Circuit to R1
 no ip address
 encapsulation hdlc
 isdn switch-type primary-ni
 isdn protocol-emulate network
 isdn incoming-voice voice
 no cdp enable
```

An engineer is troubleshooting why PSTN phones are not receiving the caller's name when called from a remote Cisco UCM site. An ISDN PRI connection is being used to reach the PSTN What must the administrator select to resolve the issue?

A. isdn supp-service name calling
B. isdn outgoing display-ie
C. isdn enable did
D. isdn send display le

Answer: B

NEW QUESTION 115
A collaboration engineer configures Global Dial Plan Replication for multiple Cisco UCM clusters. The local cluster acts as the hub cluster, and the remaining clusters act as spoke clusters Which service must the engineer configure on the local cluster'

A. Intercluster Lookup Service
B. Location Conveyance on intercluster SIP trunks
C. Intra-Cluster Communication Signaling
D. Mobility Cross Cluster

Answer: A

NEW QUESTION 119
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 120
Which Cisco Unified communications manager configuration is required for SIP MWI integration?

A. Select “Redirecting Diversion Header Delivery— Inbound’ on the SIP trunk
B. Enable “Accept presence subscription” on the SIP trunk security profile
C. Select “Redirecting Diversion Header Delivery – outbound” on the SIP trunk
D. Enable “Accept unsolicited notification” on the SIP Trunk security profile

Answer: D

NEW QUESTION 121
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 123

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 128
End users report bad video quality and voice choppiness on Cisco Collaboration endpoints. The engineer changed the device pool the users were in but did not correct the problem. Which action should be taken to troubleshoot this issue?

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

Answer: D

NEW QUESTION 131
Refer to the exhibit.

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

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

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

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

Answer: BD

NEW QUESTION 132
What is a capability of Cisco Expressway?

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

Answer: D

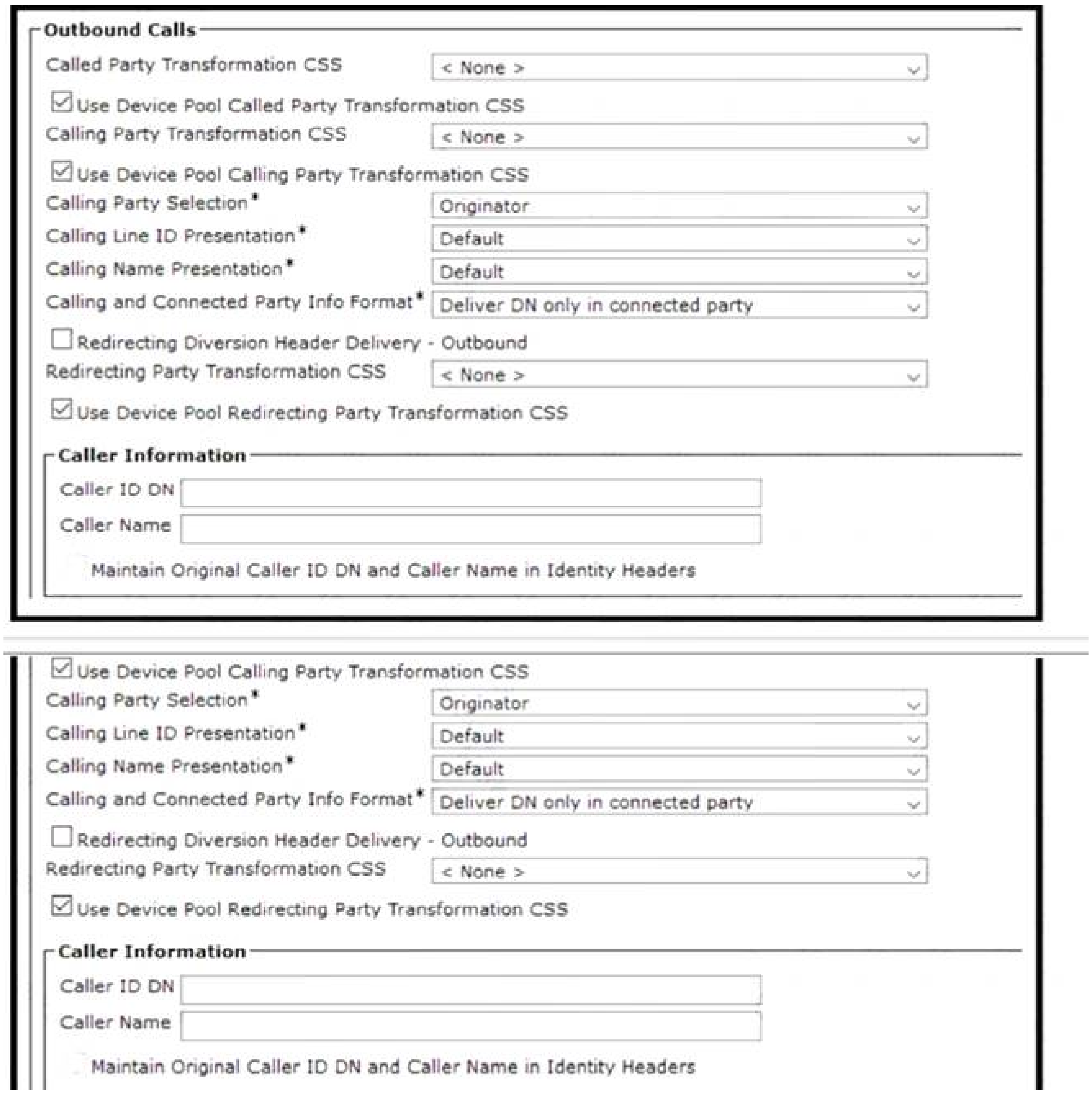
NEW QUESTION 136
Which two recommendations are made to optimize Cisco UCM configuration to reduce the number of toll fraud incidents in an organization? (Choose two.)

A. Classify all route patterns as on-net and prohibit on-net to on-net call transfers in Cisco UCM service parameters.

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

Answer: BE

NEW QUESTION 140
Refer to the exhibit.



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

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

Answer: B

NEW QUESTION 143
Which QoS marking is used when an administrator configures voice call signaling?

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

Answer: B

NEW QUESTION 144
An engineer troubleshoots outbound can failure on an ISDN-PRI circuit. The engineer ts suspecting the 'Incomplete Destination". Which debugs or commands are run in the voice gateway to troubleshoot the issue?

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

Answer: C

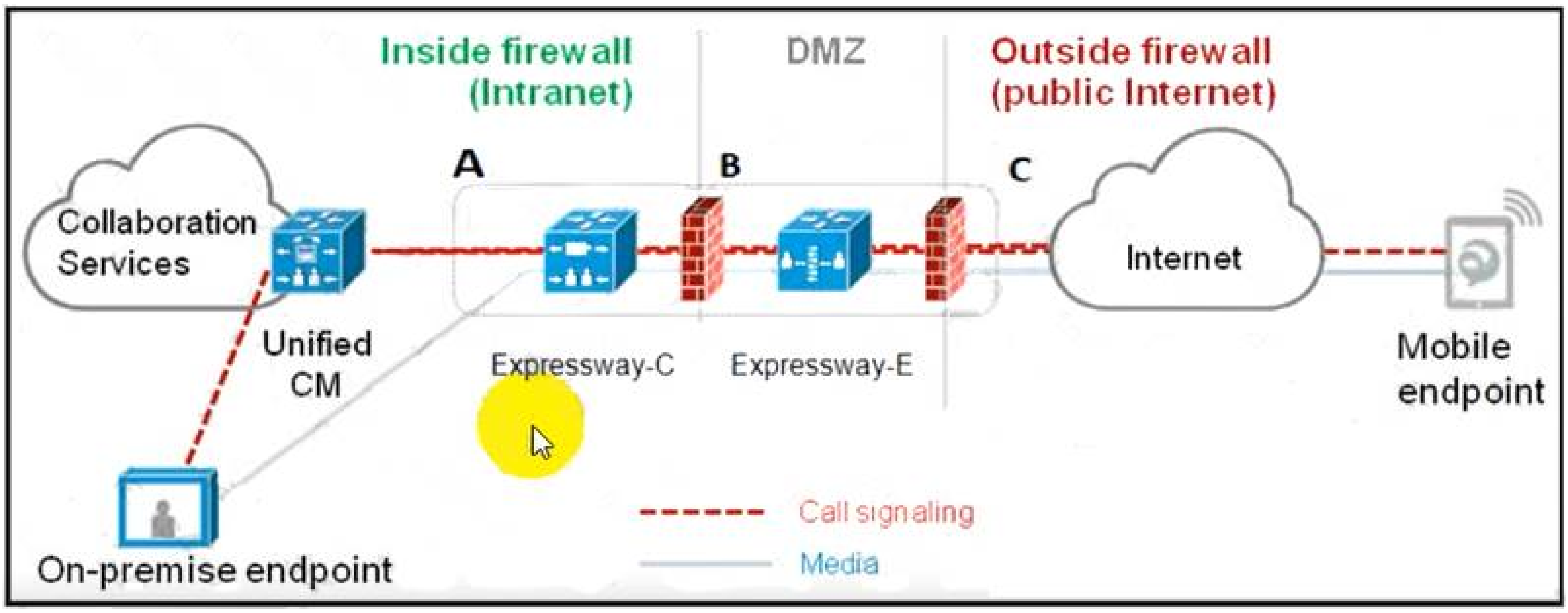
Explanation:
The engineer should run the following debugs or commands in the voice gateway to troubleshoot the issue: debug isdn q931 - This debug will show the ISDN Q.931 messages that are being exchanged between
the voice gateway and the ISDN switch. This can be used to identify the cause of the "Incomplete Destination" error.
show isdn status - This command will show the status of the ISDN PRI circuit. This can be used to verify that the circuit is up and running.
The other options are not correct. The debug isdn q921 command will show the ISDN Q.921 messages that are being exchanged between the voice gateway and the ISDN switch. This is not necessary for troubleshooting the issue. The term mon command will show the terminal monitor output. This is not necessary for troubleshooting the issue. The debug voip ecapi inout command will show the VoIP ECAP messages that are being exchanged between the voice gateway and the VoIP server. This is not necessary for troubleshooting the issue. The show controller ti command will show the status of the T1 controller. This is not necessary for troubleshooting the issue.

NEW QUESTION 148
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 150
Refer to the exhibit.



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

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

Answer: D

NEW QUESTION 154
A Cisco Unity Connection Administrator must set a voice mailbox so that it is accessed from a secondary device. Which configuration on the voice mailbox makes this change?

A. Attempt Forward routing rule
B. Mobile User
C. Alternate Extensions
D. Alternate Names

Answer: C

NEW QUESTION 155
An administrator must implement toll fraud prevention on Cisco UCM using these parameters:
• Enable Forced Authorization Code 112211.
• Set an authorization level of 3 for the route pattern 8005551212.
• Require no access code to dial 10-digit numbers. How must the route pattern be implemented?

A. Pattern = 1122113.8005551212
B. Pattern = 8005551212.1122113
C. Pattern = 8005xxxxxx
D. Pattern = 3.800xxxxxxx

Answer: A

Explanation:
To implement toll fraud prevention on Cisco UCM, an administrator can use the following parameters:

- Enable Forced Authorization Code 112211.
- Set an authorization level of 3 for the route pattern 8005551212.
- Require no access code to dial 10-digit numbers.

The route pattern must be implemented as follows: Pattern = 1122113.8005551212
This will require users to enter the authorization code 112211 followed by the number 8005551212 to dial this number. The authorization level of 3 will prevent users from transferring calls to this number.

NEW QUESTION 159
Which two DNS records must be created to configure Service Discovery for on-premises Jabber? (Choose two.)

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

Answer: BD

NEW QUESTION 160
Refer to the exhibit.

```
!
voice service voip
 ip address trusted list
  ipv4 192.168.100.101
  ipv4 192.168.101.0 255.255.255.128
!
dial-peer voice 1 voip
 destination-pattern +T
 session protocol sipv2
 session target ipv4:192.168.102.102
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
```

When a call is received on Cisco Unified Border Element. from which IP does it allow a connection?

A. 192.168.100.103
B. 192.168.102.102
C. 192.168.100.102
D. 192.168.101.201

Answer: B

NEW QUESTION 163
Why does Cisco UCM use DNS?

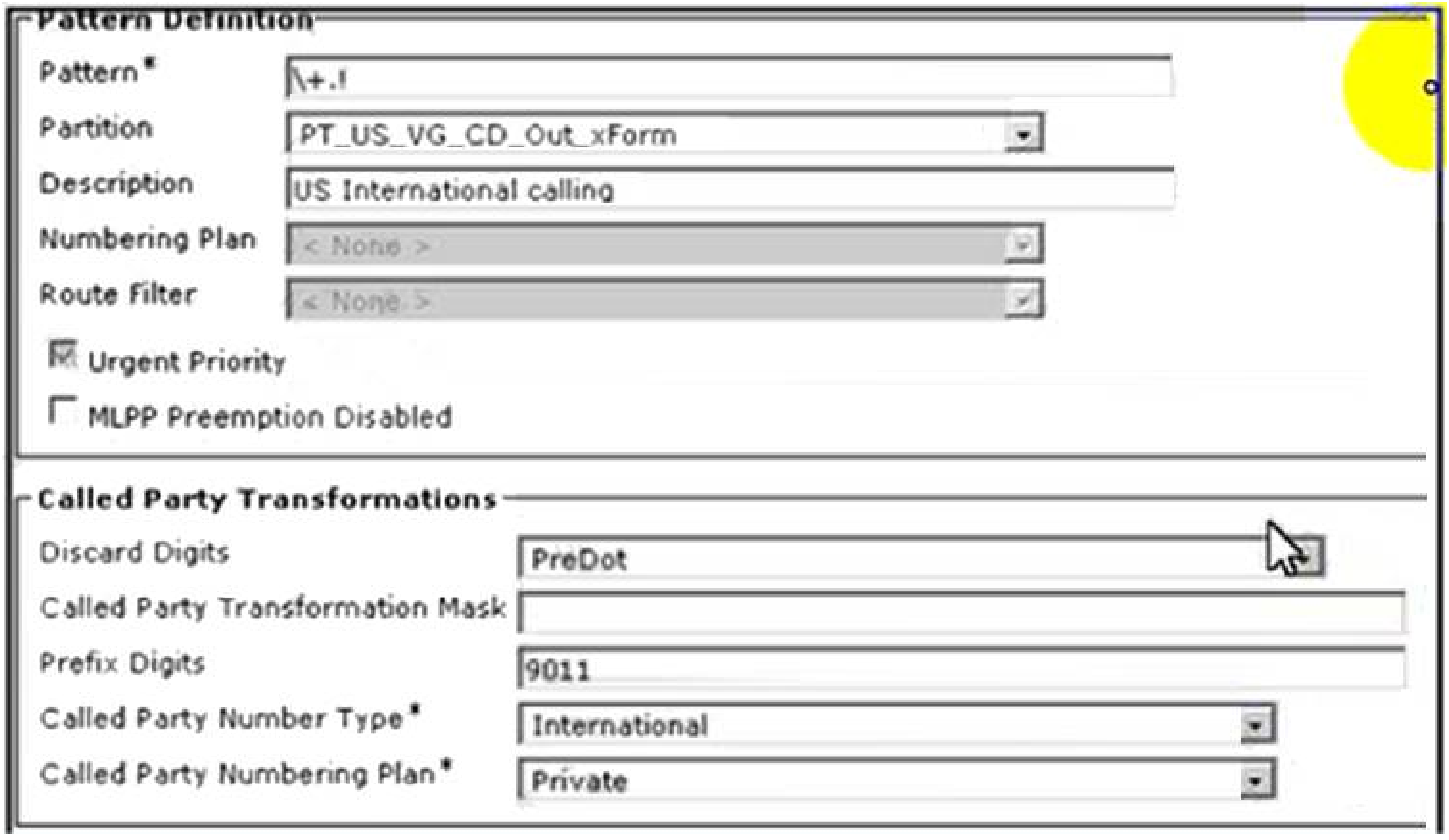
A. It provides certificate-based security for media
B. It resolves FQDN to IP address resolution for trunks
C. it connects endpoints to single sign-on services.
D. It provides SRV resolution to the endpoints registered

Answer: D

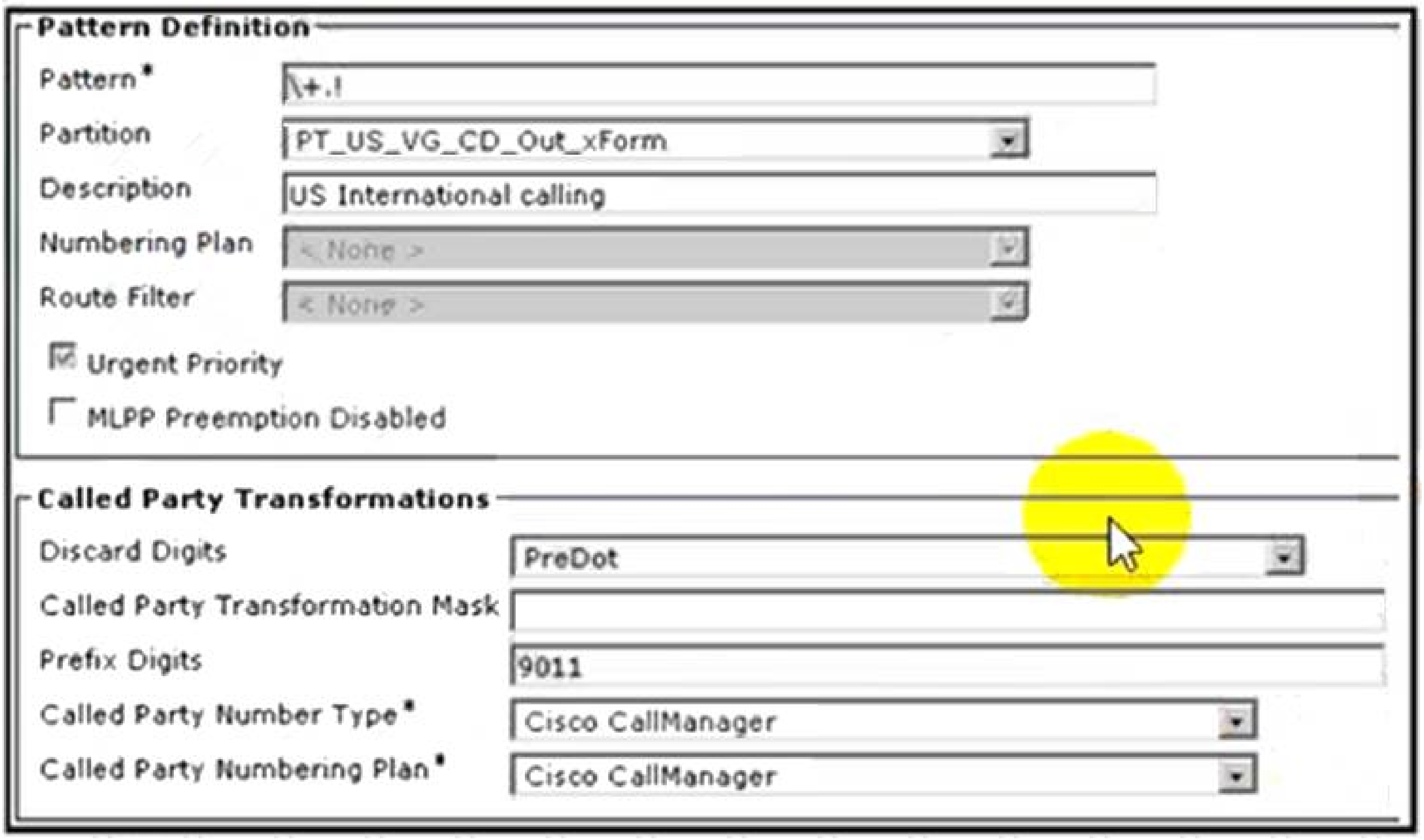
NEW QUESTION 164
Given this H.323 gateway configuration and using Cisco best practices, how must the called party transformation pattern be configured to ensure that a proper ISDN type of number is set?

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

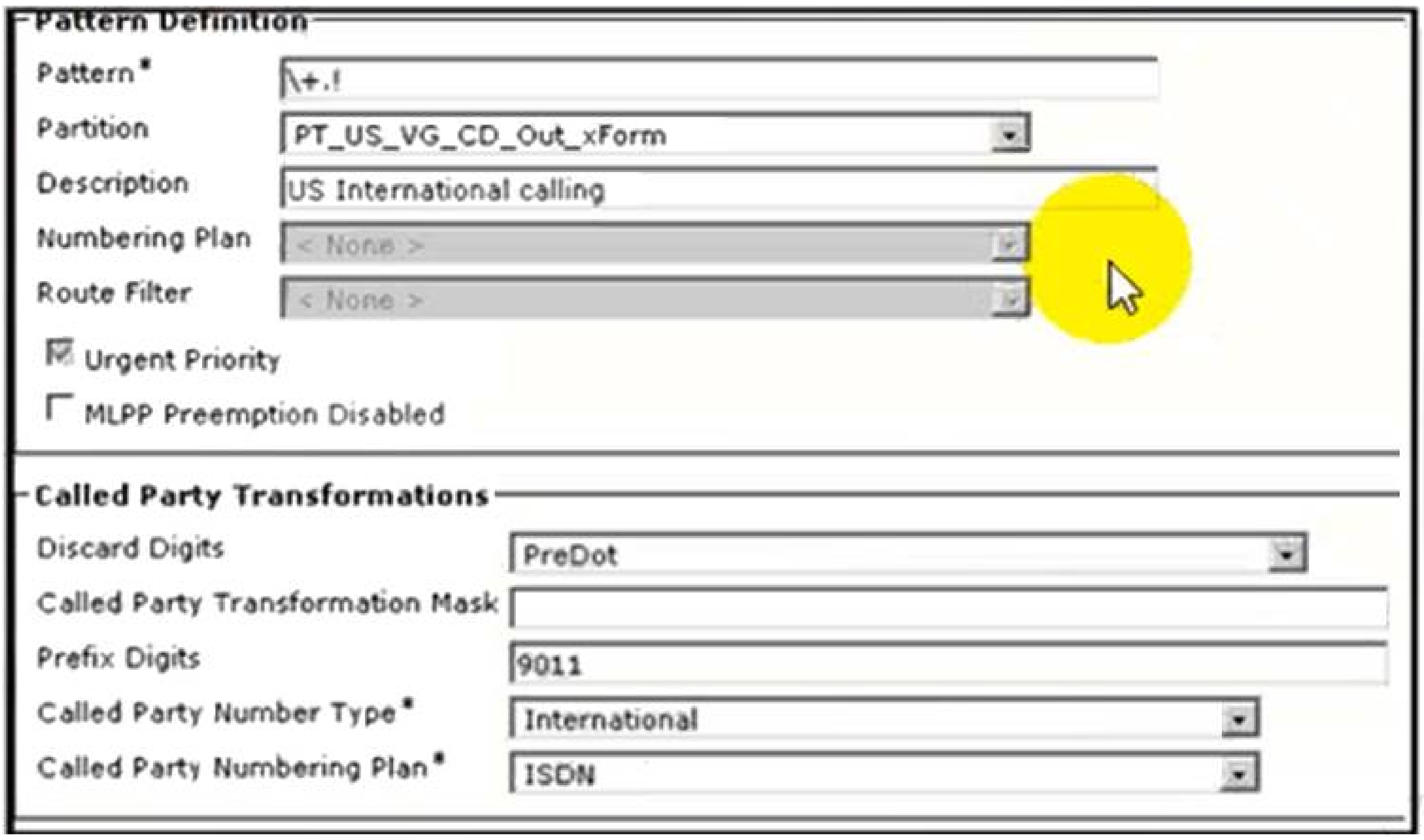
A.


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*
Private

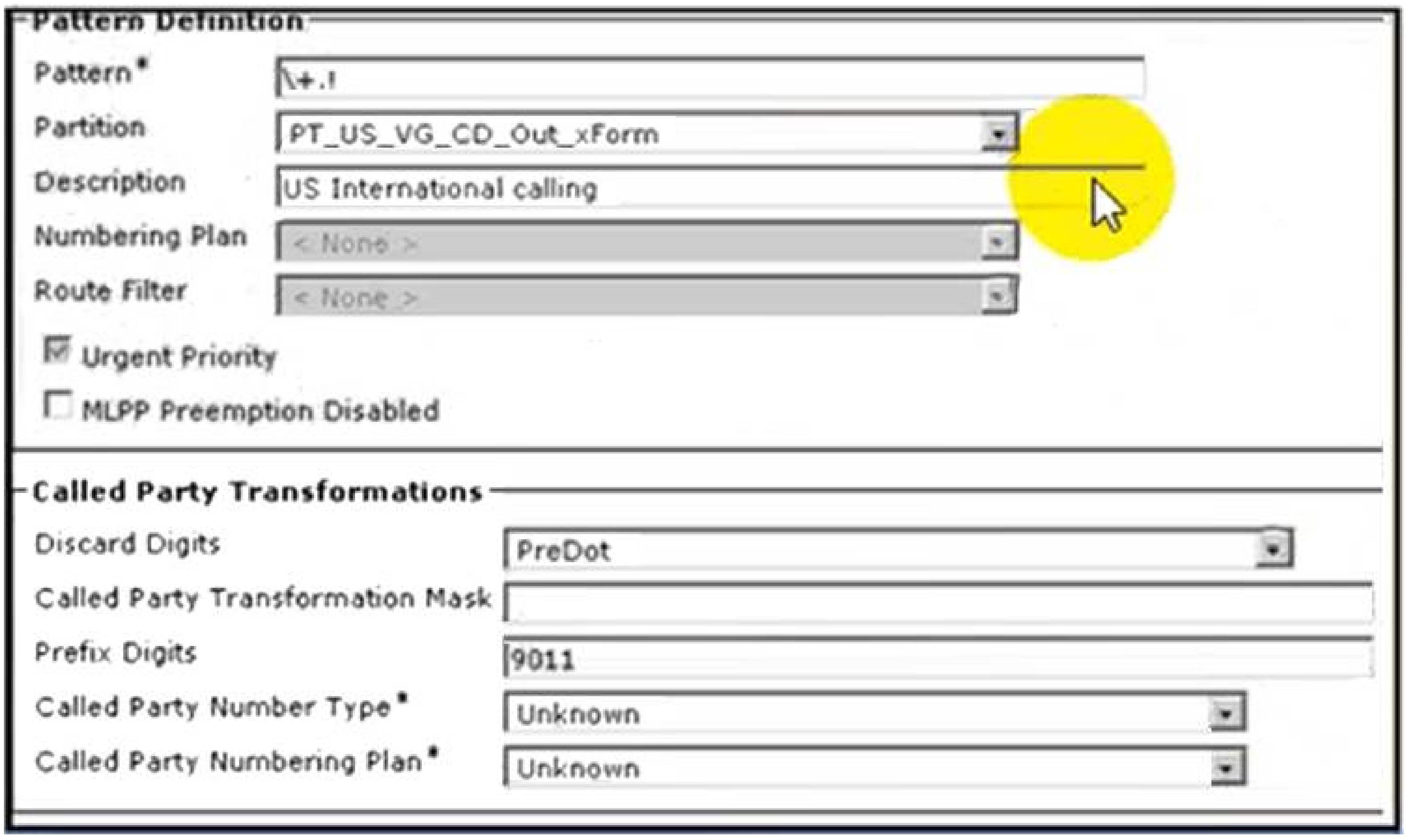
B.


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

C.


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

D.



Answer: C

NEW QUESTION 169
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 170
Refer to the exhibit.

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

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

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

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

Answer: C

NEW QUESTION 173
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 178
Refer to the exhibit.

```
Endpoint A:
m=audio 21796 RTP/AVP 108 9 104 105 101
 b=TIAS:64000
 a=extmap:14 http://protocols.cisco.com/timestamp#100us
 a=rtpmap:108 MP4A-LATM/90000
 a=fmtp:108 bitrate=64000;profile-level-id=24;object=23
 a=rtpmap:9 G722/8000
 a=rtpmap:104 G7221/16000
 a=fmtp:104 bitrate=32000
 a=rtpmap:105 G7221/16000
 a=fmtp:105 bitrate=24000
 a=rtpmap:101 telephone-event/8000
 a=fmtp:101 0-15
 a=trafficclass:conversational.audio.immersive.aq:admitted

Endpoint B:
m=audio 21796 RTP/AVP 105 0 8 18 101
 b=TIAS:64000
 a=extmap:14 http://protocols.cisco.com/timestamp#100us
 a=rtpmap:105 G7221/16000
 a=fmtp:105 bitrate=24000
 a=rtpmap:0 PCMU/8000
 a=rtpmap:8 PCMA/8000
 a=rtpmap:18 G729/8000
 a=fmtp:18 annexb=no
 a=rtpmap:101 telephone-event/8000
 a=rtpmap:101 telephone-event/8000
 a=fmtp:101 0-15
 a=trafficclass:conversational.audio.immersive.aq:admitted
```

Endpoint A calls endpoint B. What is the only audio codec that is used for the call?

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

Answer: C

NEW QUESTION 183
Why isn’t an end user s PC device in a QoS trust boundary included?

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

Answer: B

NEW QUESTION 187
An engineer is integrating Unity Connection with Cisco UCM. Which two actions must be configured so that recording and playback from the IP phones works at all times, including peak traffic hours? (Choose two.)

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

Answer: AC

NEW QUESTION 188
An engineer is deploying Webex app on Microsoft Windows computers. The engineer wants to ensure that the end users do not receive pop-IQ dialogues %'hen they start the application 'Much two actions ensure the end users are not promted to accept the end-user license (Choose two)

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

Answer: BC

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

- HKEY_LOCAL_MACHINE\Software\CiscoCollabHost\Eula Setting\Eula_disable

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

NEW QUESTION 190
Which call flow matches traffic from a Mobile and Remote Access registered endpoint to central call control?

A. Endpoint>Expressway-C>Expressway-E>Cisco UCM
B. Endpoint>Expressway-E>Expressway-C> Cisco UCM
C. Endpoint>Expressway-E> Cisco UCM
D. Endpoint>Expressway-C> Cisco UCM

Answer: A

Explanation:
The call flow for a Mobile and Remote Access registered endpoint to central call control is as follows:
- The endpoint registers with the Expressway-C.
- The Expressway-C forwards the registration request to the Expressway-E.
- The Expressway-E forwards the registration request to the Cisco UCM.
- The Cisco UCM registers the endpoint.

When the endpoint places a call, the call flow is as follows:
- The endpoint sends the call request to the Expressway-C.
- The Expressway-C forwards the call request to the Expressway-E.
- The Expressway-E forwards the call request to the Cisco UCM.
- The Cisco UCM places the call.

The Expressway-C and Expressway-E are used to provide secure access to the Cisco UCM for endpoints that are not located on the corporate network. The Expressway-C is located on the corporate network, and the Expressway-E is located in the DMZ.

NEW QUESTION 192
Which two parameters influence the total number of supported conference participants on a Cisco IOS XE router that has DSP modules? (Choose two.)

A. voice codec
B. session capacity of the PVDM module
C. number of protocol data units
D. software version Of the router
E. license types

Answer: AB

Explanation:
These are two parameters that influence the total number of supported conference participants on a Cisco IOS XE router that has DSP modules2. A voice codec is a software algorithm that compresses and decompresses voice signals for transmission over a network3. Different voice codecs have different bandwidth requirements and quality levels3. A PVDM module is a hardware component that provides digital signal processing (DSP) resources for voice applications such as conferencing and transcoding1. A PVDM module has a fixed session capacity, which is the maximum number of voice channels that it can support simultaneously1.

NEW QUESTION 193
Refer to the exhibit.

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

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

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

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

Answer: D

NEW QUESTION 195
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 199
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 201
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 202
In the cisco expressway solution, which two features does mobile and Remote access provide? (Choose two)

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

Answer: BE

NEW QUESTION 205
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 209
Which service on the Presence Server is responsible for maintaining the point-to-point chat connections between Jabber clients?

A. Cisco SIP Proxy
B. Cisco XCP Text Conference Manager
C. Cisco XCP Router
D. Cisco XCP XMPP Federation Manager

Answer: B

NEW QUESTION 214
An engineer must configure switch port 5/1 to send CDP packets to configure an attached Cisco IP phone to trust tagged traffic on it's access port. Which command is required to complete the configuration?

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

A. platform qos trust extend cos 3
B. platform qos trust extend
C. platform qos extend trust
D. platform qos trust extend cos 5

Answer: B

NEW QUESTION 216
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 218
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