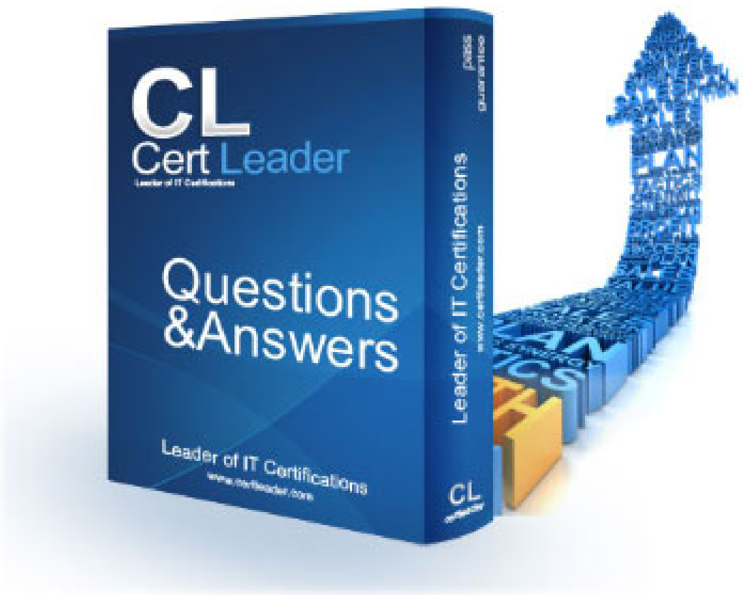
350-801 Dumps

Implementing and Operating Cisco Collaboration Core Technologies

https://www.certleader.com/350-801-dumps.html



NEW QUESTION 1
What is an indicator of network congestion in VoIP communications?

A. jitter increase due to variable delay
B. discards in the interface of routers and switches
C. video loss due to video frame corruption
D. gaps in the voice due to packet loss

Answer: A

NEW QUESTION 2
How does an administrator make a Cisco IP phone display the last 10 digits of the calling number when the call is in the connected state, and also display the calling number in the E.164 format within call history on the phone?

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

Answer: A

NEW QUESTION 3
Refer to the exhibit.

```
Sent:
INVITE sip:2004@192.168.100.100:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.100.200:5060;branch=z9hG4bKF1FED
From: "7000" <sip:7000@192.168.100.200>;tag=43CDE-1A22
To: <sip:2004@192.168.100.100>
Call-ID: 26BCA00-4C4E11EA-80169514-B1C46126@192.168.100.200
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE:  1800
User-Agent: Cisco-SIPGateway/IOS-16.9.5
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Contact: <sip:7000@192.168.100.200:5060>
Expires: 180
Max-Forwards: 68
P-Asserted-Identity: "7000" <sip:7000@192.168.100.200>
Session-Expires:  1800
Content-Type: application/sdp
Content-Length: 254

v=0
o=CiscoSystemsSIP-GW-UserAgent 5871 9974 IN IP4 192.168.100.200
s=SIP Call
c=IN IP4 192.168.100.200
t=0 0
m=audio 8002 RTP/SAVP 0
c=IN IP4 192.168.100.200
a=rtpmap:0 PCMU/8000
a=ptime:20
```

Calls to Cisco Unity Connection are failing across Cisco Unified Border Element when callers try to select a menu prompt Why is this happening and how is it fixed?

A. Cisco Unity Connection is configured on G.729 onl
B. Cisco Unity Connection must be reconfigured to support iLBC.
C. Cisco Unified Border Element is not sending any support for DTM
D. OTMF configuration must be added to the appropriate dial peer.
E. Cisco Unified Border Element is sending the incorrect media IP addres
F. The IP address of the loopback interface must be advertised in the SDP
G. The Cisco Unity Connection Call Handler is configured for a "Release to Switch" transfer type Transfers must be disabled for the Cisco Unity Connection Call Handler

Answer: B

NEW QUESTION 4
How are E.164 called-party numbers normalized on a globalized call-routing environment in Cisco UCM?

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

Answer: C

NEW QUESTION 5
An engineer configures a SIP trunk for MWI between a Cisco UCM cluster and Cisco Unity Connection. The Cisco UCM cluster fails to receive the SIP notify messages. Which two SIP trunk settings resolve this issue? (Choose two.)

A. accept out-of-dialog refer
B. accept out-of-band notification
C. transmit security status
D. allow changing header
E. accept unsolicited notification

Answer: AE

NEW QUESTION 6
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 7
An engineer must configure a SIP route pattern using domain routing with destination +13135551212. The domain ciscocm1.jupiter.com resolves to 192.168.1.3. How must the IPV4 Pattern be configured?

A. +13135551212@192.168.1.3
B. ciscocm1.jupiter.com
C. \+13135551212@192.168.1.3
D. 192.168.1.3

Answer: B

NEW QUESTION 8
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 9
An engineer is configuring a phone system CISCO UCM and wants to activate TFTP service. The engineer selects the serviceability page for configuration. Which nodes configurable for TFTP?

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

Answer: C

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

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

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

Answer: B

NEW QUESTION 10
If a phone needs to register with cucm1.cisco.com, which network service assists with the phone registration process?

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

Answer: D

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

NEW QUESTION 11
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 16
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 21
What is set when using COS to mark an Ethernet frame?

A. Ipp bits
B. IP ECN bits
C. DCSP bits
D. 802.1 p User Priority bits

Answer: D

Explanation:
When using COS to mark an Ethernet frame, the 802.1 p User Priority bits are set. These bits are used to indicate the priority of the frame. The higher the priority, the more likely the frame is to be transmitted first.

NEW QUESTION 26
Where in Cisco UCM is restrictions on audio bandwidth configured?

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

Answer: C

NEW QUESTION 27
Which endpoint feature is supported using Mobile and Remote Access through Cisco Expressway?

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

Answer: A

NEW QUESTION 32

Which attribute contains an XMPP stanza?

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

Answer: A

NEW 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. UDP161
B. TCP 161
C. UDP 162
D. TCP 80

Answer: C

NEW QUESTION 40
An employee of company ABC just quit. The IT administrator deleted the employee’s user id from the active directory at 10 a m. on March 4th The nightly sync occurs at 10 p.m. daily. The IT administrator wants to troubleshoot and find a way to delete the user id as soon as possible How is this issue resolved?

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

Answer: C

NEW QUESTION 43
Refer to the exhibit.

ip.addr==10.0.101.10

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

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

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

Answer: D

NEW QUESTION 44
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 46
A collaboration engineer must configure Cisco Unified Border Element to support up to five concurrent outbound calls across an Ethernet Link with a bandwidth of 160 kb to the Internet Telephony service provider. Which set of commands allows the engineer to complete the task without compromising voice quality?

A)

```
dial-peer voice 1 voip
 translation-profile outgoing Strip9
 max-conn 5
 destination-pattern 91[2-9]..[2-9]......$
 session protocol sipv2
 session target ipv4:142.45.10.1
 dtmf-relay rtp-nte sip-notify sip-kpml
 codec aacld
```

B)

```
dial-peer voice 1 voip
 translation-profile outgoing Strip9
 max-conn 5
 destination-pattern 91[2-9]..[2-9]......$
 session protocol sipv2
 session target ipv4:142.45.10.1
 dtmf-relay rtp-nte sip-notify sip-kpml
 codec ilbc mode 20
```

C)

```
dial-peer voice 1 voip
 translation-profile outgoing Strip9
 max-conn 5
 destination-pattern 91[2-9]..[2-9]......$
 session protocol sipv2
 session target ipv4:142.45.10.1
 dtmf-relay rtp-nte sip-notify sip-kpml
 codec mp4a-latm
```

D)

```
dial-peer voice 1 voip
 translation-profile outgoing Strip9
 max-conn 5
 destination-pattern 91[2-9]..[2-9]......$
 session protocol sipv2
 session target ipv4:142.45.10.1
 dtmf-relay rtp-nte sip-notify sip-kpml
```

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

Answer: B

NEW QUESTION 50
What is the major difference between the two possible Cisco IM and Presence high-availability modes?

A. Balanced mode provides user load balancing and user failover in the event of an outag
B. Active/standby mode provides an always on standbynode in the event of an outage, and it also provides load balancing.
C. Balanced mode provides user load balancing and user failover only for manually generated failovers.Active/standby mode provides anunconfigured standby node in the event of an outage, but it does not provide load balancing.
D. Balanced mode provides user load balancing and user failover in the event of an outag
E. Active/standby mode provides an always on standbynode in the event of an outage, but it does not provide load balancing.
F. Balanced mode does not provide user load balancing, but it provides user failover in the event of an outag
G. Active/standby mode provides analways on standby node in the event of an outage, but it does not provide load balancing.

Answer: C

Explanation:
Balanced mode provides user load balancing and user failover in the event of an outage. Active/standby mode provides an always on standby node in the event of an outage, but it does not provide load balancing.
Here is a more detailed explanation of the two modes:

- Balanced mode: In balanced mode, the IM and Presence Service nodes are configured to work together to provide high availability. The nodes are configured in a redundancy group, and the system automatically balances the load of users across the nodes in the group. If one of the nodes fails, the system automatically fails over the users to the other nodes in the group.
- Active/standby mode: In active/standby mode, one of the IM and Presence Service nodes is designated as the active node, and the other nodes are designated as standby nodes. The active node handles all of the user traffic, and the standby nodes are only used if the active node fails. If the active node fails, the system automatically fails over to one of the standby nodes.

NEW QUESTION 55
Which two functions are provided by Cisco Expressway Series? (Choose two.)

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

Answer: AC

Explanation:
The Cisco Expressway Series provides the following functions:
- Voice and video transcoding
- Interworking of SIP and H.323
- Firewall traversal
- Session border controller (SBC) functionality
- Endpoint registration
- Call admission control (CAC)
- Quality of service (QoS)
- Security

The Cisco Expressway Series does not provide voice and video conferencing or intercluster extension mobility.

NEW QUESTION 59
An engineer with ID012345678 must build an international dial plan in Cisco UCM. Which action is taken when building a variable-length route pattern?

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

Answer: D

Explanation:
When building a variable-length route pattern, you need to create a second route pattern followed by the # wildcard. This will allow the user to indicate the end of the number by dialing #. For example, if you want to create a route pattern for international calls, you would create a route pattern like this: 9.011!#
This route pattern will match any number that starts with 9.011, followed by any number of digits, and then ends with #.
The other options are incorrect because:
- Configuring a single route pattern for international calls will not allow the user to indicate the end of the number.
- Reducing the T302 timer to less than 4 seconds will not allow the user to indicate the end of the number.

NEW QUESTION 64
An administrator has been asked to implement toll fraud prevention in Cisco UCM Which tool is used to begin this process?

A. Cisco UCM class of service
B. Cisco Unified Mobility
C. Cisco UCM Access Control Group restrictions
D. Cisco Unified Real-Time Monitoring Tool

Answer: A

NEW QUESTION 66
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 69
Refer to the exhibit.

```
000193: Dec  5 14:35:31.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec  5 14:35:32.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec  5 14:35:33.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec  5 14:35:34.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
```

Given this "debug isdn q921" output, what is the problem with the PRI?

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

Answer: B

NEW QUESTION 73
Which field is configured to change the caller ID information on a SIP route pattern?

A. Route Partition
B. Calling Party Transformation Mask
C. Called Party Transformation Mask

D. Connected Line ID Presentation

Answer: B

NEW QUESTION 75
An engineer wants to manually deploy a CISCO Webex DX80 Video endpoint to a remote user. Which type of provisioning is configured on the endpoint?

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

Answer: D

Explanation:
The Cisco Webex DX80 Video endpoint can be provisioned in two ways:
- Automatically, using the Cisco Unified Communications Manager (CUCM) or Cisco Video Communication Server (VCS)
- Manually, using the Edge provisioning mode

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

NEW QUESTION 80
Where in Cisco UCM are codec negotiations configured for endpoints?

A. under device profiles in device settings
B. in in-service parameters
C. under regions using preference lists
D. in enterprise parameters

Answer: C

NEW QUESTION 81
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 82
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 87
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 91
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 93
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 97
When multiple potential patterns are present, which two things are considered when Cisco UCM selects a destination pattern? (Choose two.)

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

Answer: AD

NEW QUESTION 99
Which Webex Calling dial plan settings restrict a user from placing a particular outbound call type?

A. Block
B. Transfer to Number
C. Reject
D. Restrict

Answer: D

Explanation:
The Restrict setting in the Webex Calling dial plan prevents users from placing certain types of outbound calls. For example, you can use the Restrict setting to prevent users from making international calls or calls to premium-rate numbers.
The Block setting in the Webex Calling dial plan prevents users from placing any outbound calls. The Transfer to Number setting in the Webex Calling dial plan transfers all outbound calls to a specified number. The Reject setting in the Webex Calling dial plan rejects all outbound calls.
Here is a table summarizing the different dial plan settings and their effects:
Dial Plan Setting Effect
Block
Prevents users from placing any outbound calls. Transfer to Number
Transfers all outbound calls to a specified number. Reject
Rejects all outbound calls. Restrict
Prevents users from placing certain types of outbound calls.

NEW QUESTION 104
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 108
Refer to the exhibit.

```
ROUTER-1(config)# policy-map LLQ_POLICY
ROUTER-1(config-pmap)# class VOICE
ROUTER-1(config-pmap-c)# bandwidth 170
ROUTER-1(config-pmap-c)# exit
ROUTER-1(config-pmap)# class VIDEO
ROUTER-1(config-pmap-c)# bandwidth remaining percent 30
ROUTER-1(config-pmap-c)# exit
ROUTER-1(config-pmap)# exit
```

An engineer must modify the existing QoS policy-map statement to implement LLQ for voice traffic. Which change must the engineer make in the configuration?

A. bandwidth 170 to reserve 170
B. bandwidth 170 to LL1 170
C. bandwidth 170 to priority 170

D. bandwidth 170 to percent 170

Answer: C

NEW QUESTION 109
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. LLDP
B. TFTP
C. LACP
D. SNMP

Answer: A

Explanation:
LLDP (Link Layer Discovery Protocol) is a vendor-neutral network discovery protocol that is used to discover the topology of a network. LLDP is similar to CDP (Cisco Discovery Protocol), but it is not proprietary to Cisco. LLDP is supported by a wide range of network devices, including switches, routers, and firewalls.
To configure LLDP on a network, you must enable LLDP on the devices that you want to discover. You can then use a network management tool, such as Cisco Network Assistant, to view the topology of the network.
The other options are incorrect. TFTP (Trivial File Transfer Protocol) is a network protocol that is used to transfer files between devices. LACP (Link Aggregation Control Protocol) is a network protocol that is used to aggregate multiple network links into a single logical link. SNMP (Simple Network Management Protocol) is a network protocol that is used to manage network devices.

NEW QUESTION 113
Which action prevent toll fraud in Cisco Unified Communication Manager?

A. Configure ad hoc conference restriction
B. Implement toll fraud restriction in the Cisco IOS router
C. Allow off-net to off-net transfer
D. Implement route patterns in Cisco Unified CM

Answer: A

NEW QUESTION 115
Which Cisco UCM configuration is required for SIP MWI integrations?

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

Answer: C

NEW QUESTION 116
A Cisco UCM administrator wants to enable the Self-Provisioning feature for end users. Which two prerequisites must be met first? (Choose two.)

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

Answer: DE

NEW QUESTION 118
An administrator needs to help a remote employee make a free call to an international destination. The administrator calls the employee, then conferences in the international party. The administrator drops the call, and the employee and the international party continue their conversation. Which action prevents this type of toll fraud in the Cisco UCM?

A. Set service parameter 'Advanced Ad Hoc Conference" to FALSE.
B. Set service parameter "Drop Ad Hoc Conference" to "When Conference Controller leaves."
C. Set service parameter "Advanced Ad Hoc Conference" to 2.
D. Set service parameter "Drop Ad Hoc Conference" to "Do not allow outside parties."

Answer: B

NEW QUESTION 119
An administrator is developing an 8-class QoS baseline model. The CS3 standards-based marking recommendation is used for which type of class?

A. Scavenger
B. best effort
C. voice
D. call signaling

Answer: A

NEW QUESTION 122
What is a reason for using a Diffserv value of AF41 for video traffic?

A. Video traffic cannot tolerate any packet loss and has a latency of 150 miliseconds
B. Video traffic can tolerate up to 10% packet loss and latency of 10 seconds
C. Video trafic can tolerate up to 5% packet loss and latency of 5 seconds
D. Video traffic can tolerate a packet loss of up to 1% and latency of 150 milliseconds

Answer: D

NEW QUESTION 125
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 126
What is the purpose of a hybrid Local Gateway?

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

Answer: D

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

NEW QUESTION 129
An administrator troubleshoots call flows and suspects that there are issues with the dial plan. Which tool enables a quick analysis of the dial plan and provides call flows of dialled digits?

A. Cisco Dial Plan Analyzer
B. Dial Plan Analyzer
C. Digit Analysis Analyzer
D. Dialed Number Analyzer

Answer: D

NEW QUESTION 133
Refer to the exhibit. An engineer is confining class of control for a user in Cisco UCM. Which change will ensure that the user is unable to call 2143?

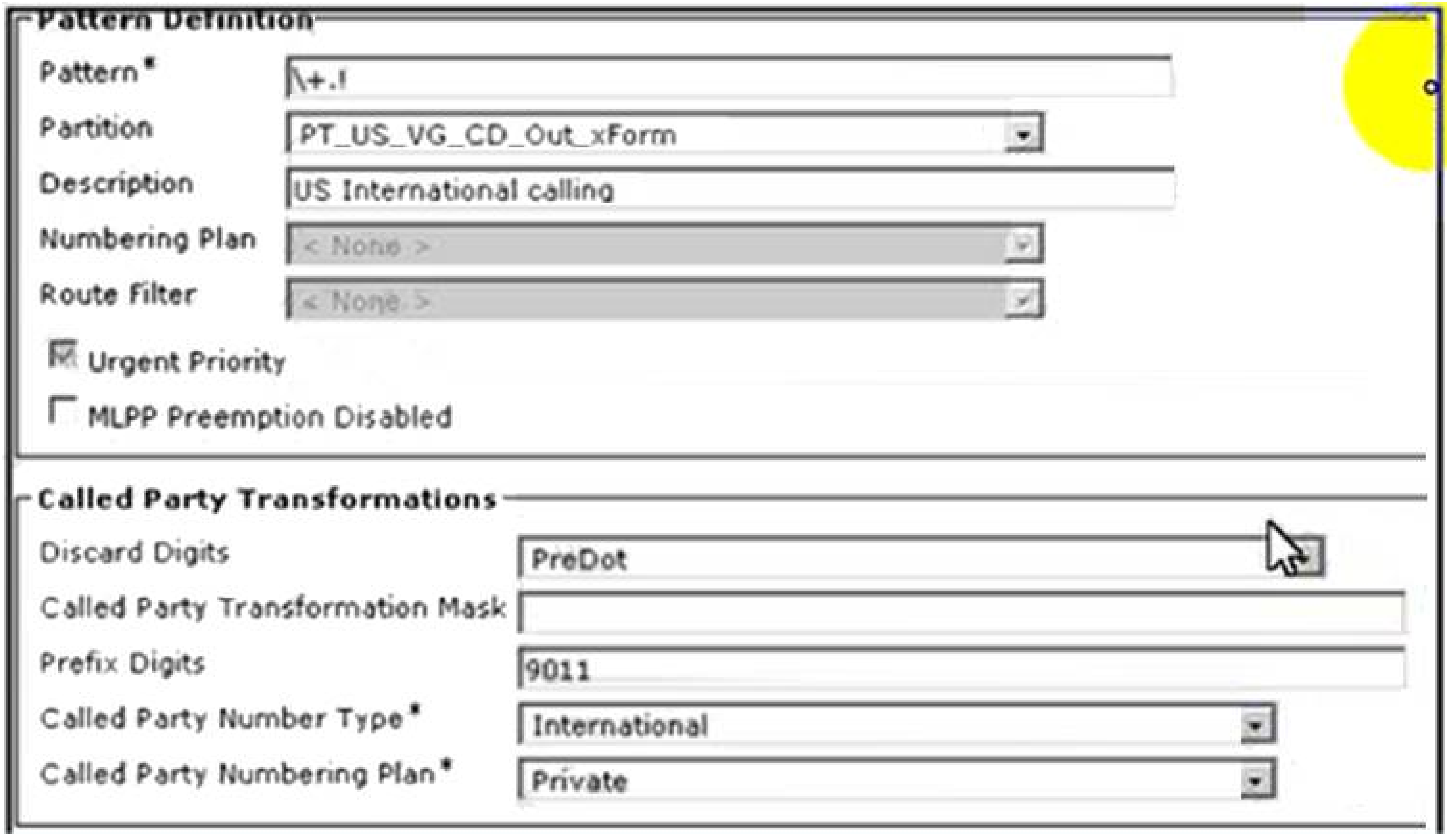
A. Change line partition to Partition_A
B. Change line CSS to only contain Partition_B
C. Set the user's line CSS to <None>
D. Set the user's device CSS to <None>

Answer: D

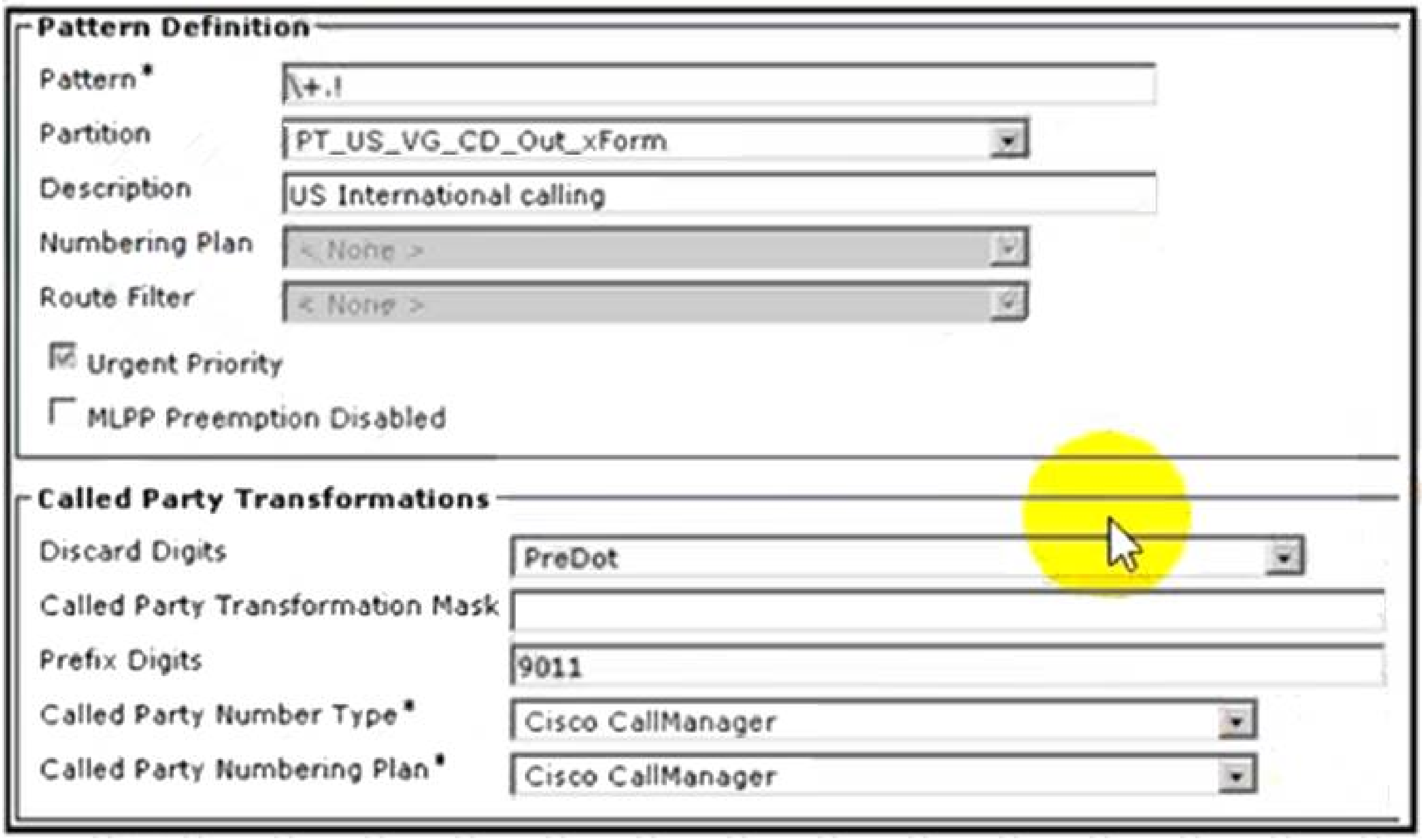
NEW QUESTION 135
Given this H.323 gateway configuration and using Cisco best practices, how must the called party transformation pattern be configured to ensure that a proper ISDN type of number is set?

```
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
```

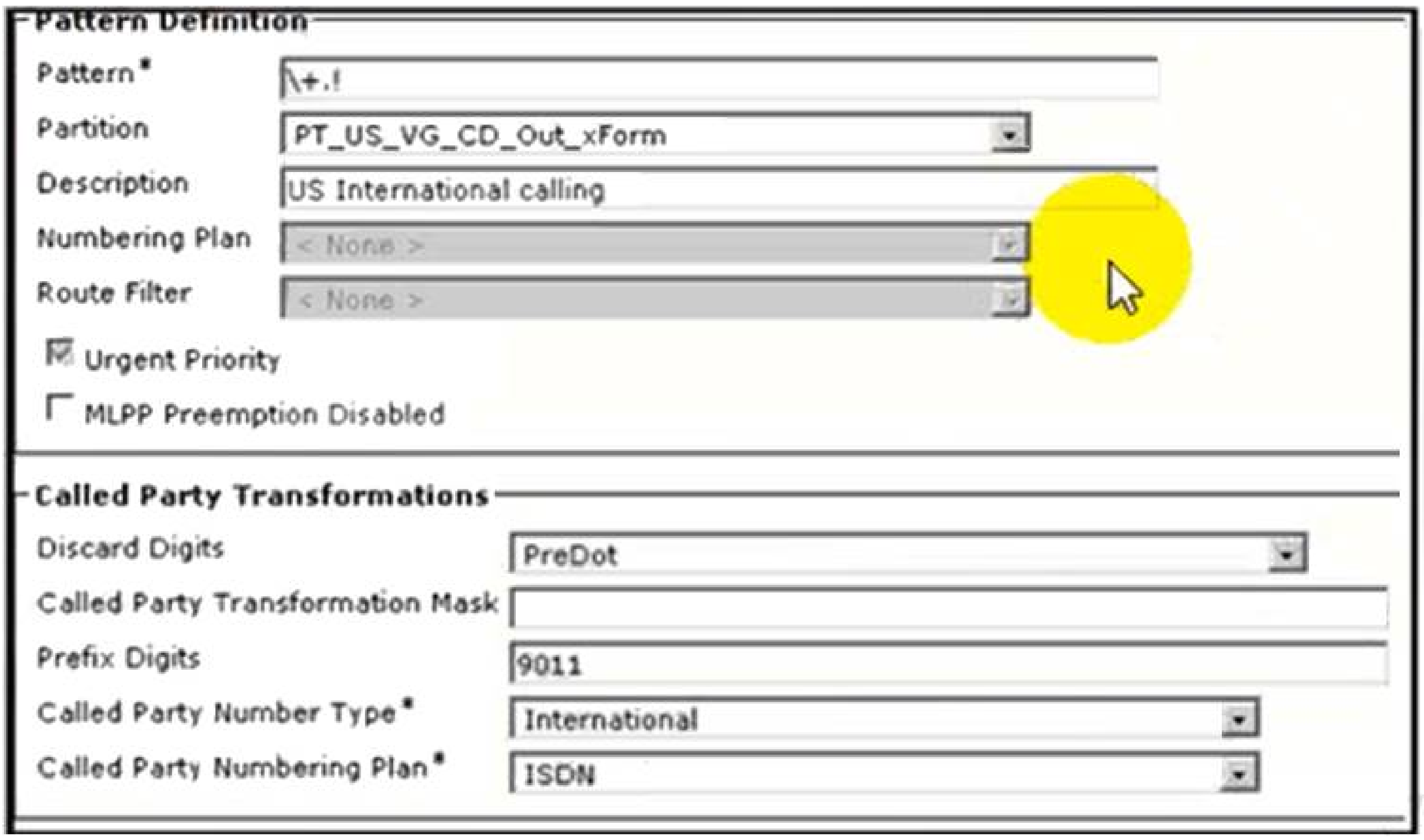
A.


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

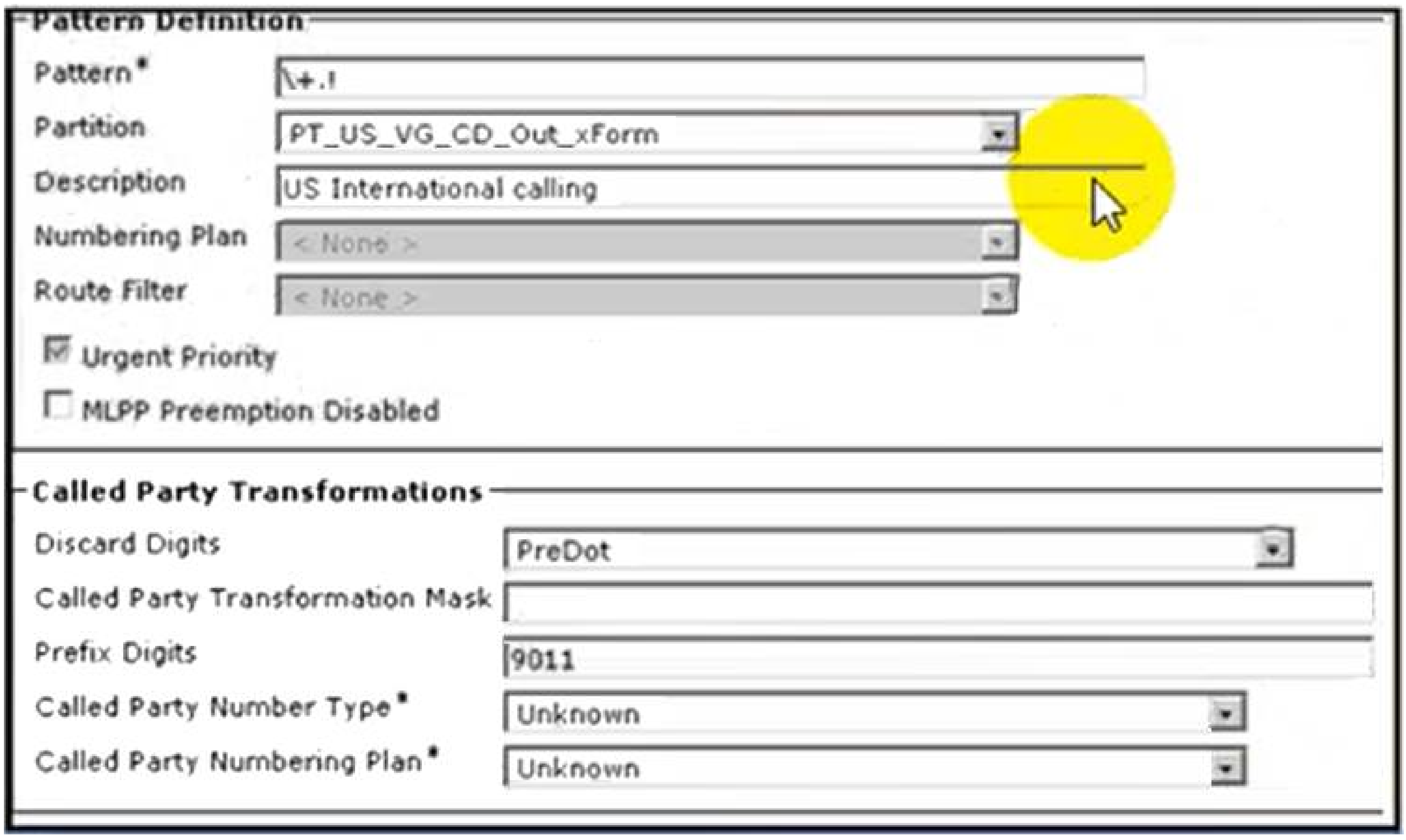
B.


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 138
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 143
Which behavior occurs when Cisco UCM has a Call Manager group that consists of two subscribers?

A. Endpoints attempt to register with the top subscriber in the list.
B. Endpoints attempt to register with (he bottom subscriber in the list.
C. Endpoints attempt to register with both subscribers in a load-balanced method.
D. If a subscriber is rebooted, endpoints deregister until the rebooted system is back in service.

Answer: A

NEW QUESTION 144
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 147

A collaboration engineer is configuring the QoS trust boundary for Cisco UCM voice and video conferencing. Which two trust boundary configurations are valid? (choose two)

A. QoS trust boundaries include all the devices directly attached to the access switch ports
B. QoS trust boundaries can be extended to Jabber running on a PC
C. QoS trust boundaries exclude Jabber softphone running on a PC
D. QoS trust boundaries can be extended to voice and video devices if the connected PCs are included
E. QoS trust boundaries can be extended to voice and video devices exclusively

Answer: CD

NEW QUESTION 148
An administrator needs to create a partial PRI consisting of the first seven timeslots available. Which configuration snippet configures the ISDN E1 PRI for this task?

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

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

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

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

Answer: C

NEW QUESTION 149
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 154
A Cisco IP Phone 7841 that is registered to a Cisco Unified Communications Manager with default configuration receives a call setup message. Which codec is negotiated when the SDP offer includes this line of text?
M=audio 498181 RTP/AVP 0 8 97

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

Answer: A

Explanation:
The SDP offer includes the following line of text: M=audio 498181 RTP/AVP 0 8 97
This line of text indicates that the following codecs are available:
- 0: G.711ulaw
- 8: G.711alaw
- 97: iLBC

The Cisco IP Phone 7841 is registered to a Cisco Unified Communications Manager with default configuration. This means that the phone will negotiate the G.711ulaw codec.
The G.711ulaw codec is a standard codec that is used for voice communication. It is a low-bandwidth codec that provides good quality.
The iLBC codec is a newer codec that is designed for use in low-bandwidth environments. It provides good quality, but it is not as widely supported as the

G.711ulaw codec.
The G.722 codec is a high-quality codec that is used for voice communication. It provides excellent quality, but it requires more bandwidth than the G.711ulaw codec.

NEW QUESTION 156
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 158
......

Thank You for Trying Our Product

* 100% Pass or Money Back

All our products come with a 90-day Money Back Guarantee.

* One year free update

You can enjoy free update one year. 24x7 online support.

* Trusted by Millions

We currently serve more than 30,000,000 customers.

* Shop Securely

All transactions are protected by VeriSign!

100% Pass Your 350-801 Exam with Our Prep Materials Via below:

https://www.certleader.com/350-801-dumps.html