350-801 Dumps

Implementing and Operating Cisco Collaboration Core Technologies

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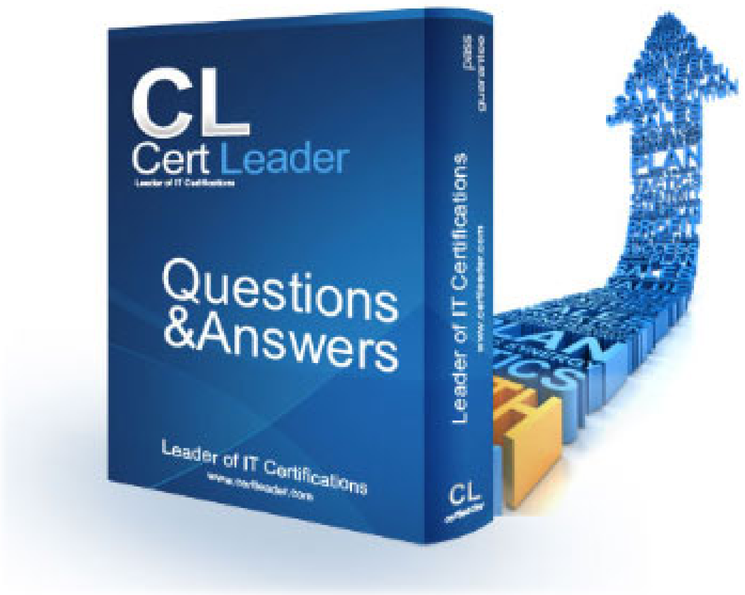


NEW QUESTION 1
Which call routing pattern is used for phone numbers that are in the E.164 format?

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

Answer: B

NEW QUESTION 2
A company deploys centralized cisco ucm architecture for a hub location and two remote sites.
*The company has only one ITSP connection at the hub connection, and ITSP supports only G.711 calls
*Remote site A has a 1-Gbps fiber connection to the hub connection and calls to and from remote side A use G.711 codec
*Remote site B has a 1 T1 connection to the hub location and calls to and from remote site B use G.729 codec Based on the provided guidance, a Cisco voice engineer must design media resource management for the customer What is the method that needs to be followed?

A. configure the hardware transcoder on the site B router
B. configure the hardware transcoder on the site A router
C. configure the hardware transcoder on the hub location router
D. configure the software transcoder on Cisco UCM to support voice calls to and from both remote sites

Answer: C

NEW QUESTION 3
A customer wants to conduct B2B video calls with a partner using on-premises conferencing solution. Which two devices are needed to facilitate this request?

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

Answer: AC

NEW QUESTION 4
Which external DNS SRV record must be present for Mobile and Remote Access?

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

Answer: B

NEW QUESTION 5
A company has an excessive number of call transfers to local and long-distance PSTN from Cisco Unity Connection voicemail. Which action in the Cisco Unity Connection restriction table resolves this issue?

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

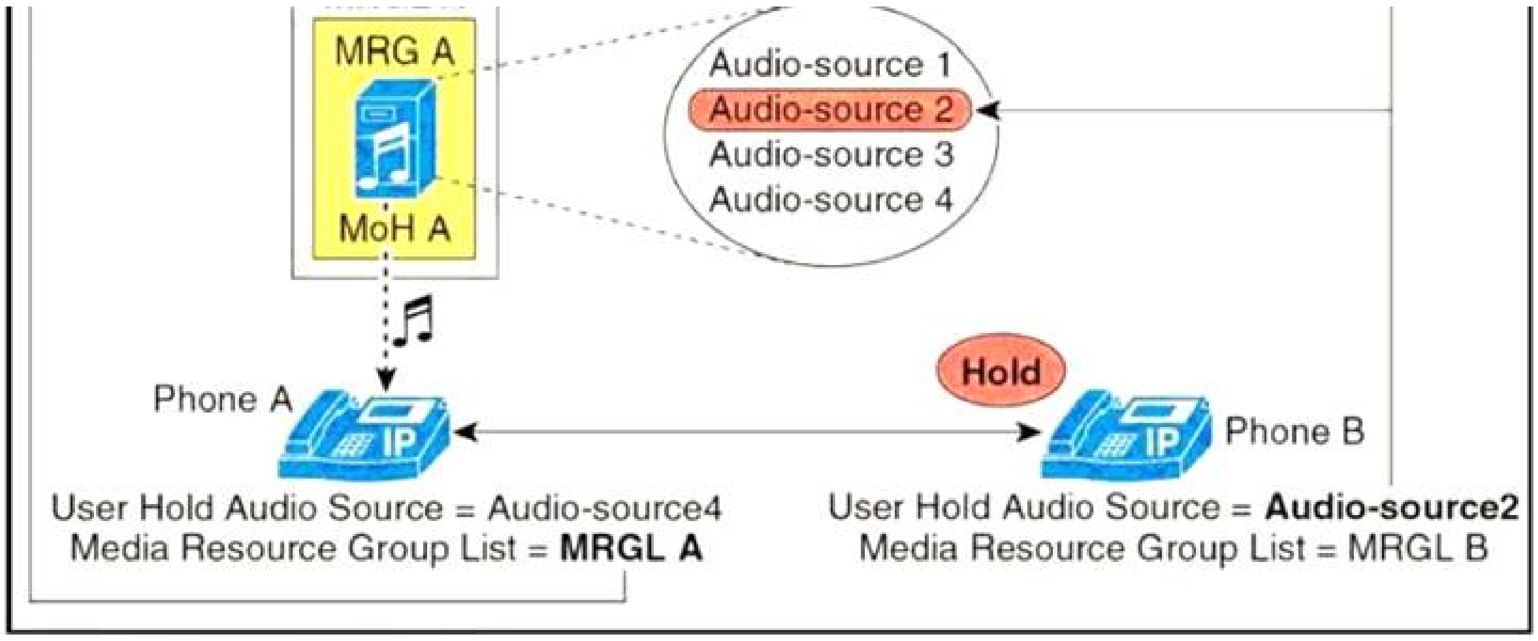
Answer: A

NEW QUESTION 6
What is the traffic classification for voice and video conferencing?

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

Answer: B

NEW QUESTION 7
Refer to the exhibit



There is a call flow between Phone A and Phone B Phone B (holder) places Phone A (holder) on hold Which MRGL and Audio Source are played to Phone A?

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

Answer: C

NEW QUESTION 8
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 9
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 10
What are two key features of the Expressway series? (Choose two.)

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

Answer: CD

NEW QUESTION 10
On a cisco catalyst switch which command is required to send CDP packets on a switch port that configures a cisco IP phone to transmit voice traffic in 802.1q frames tagged with the voice VLAN ID 221?

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

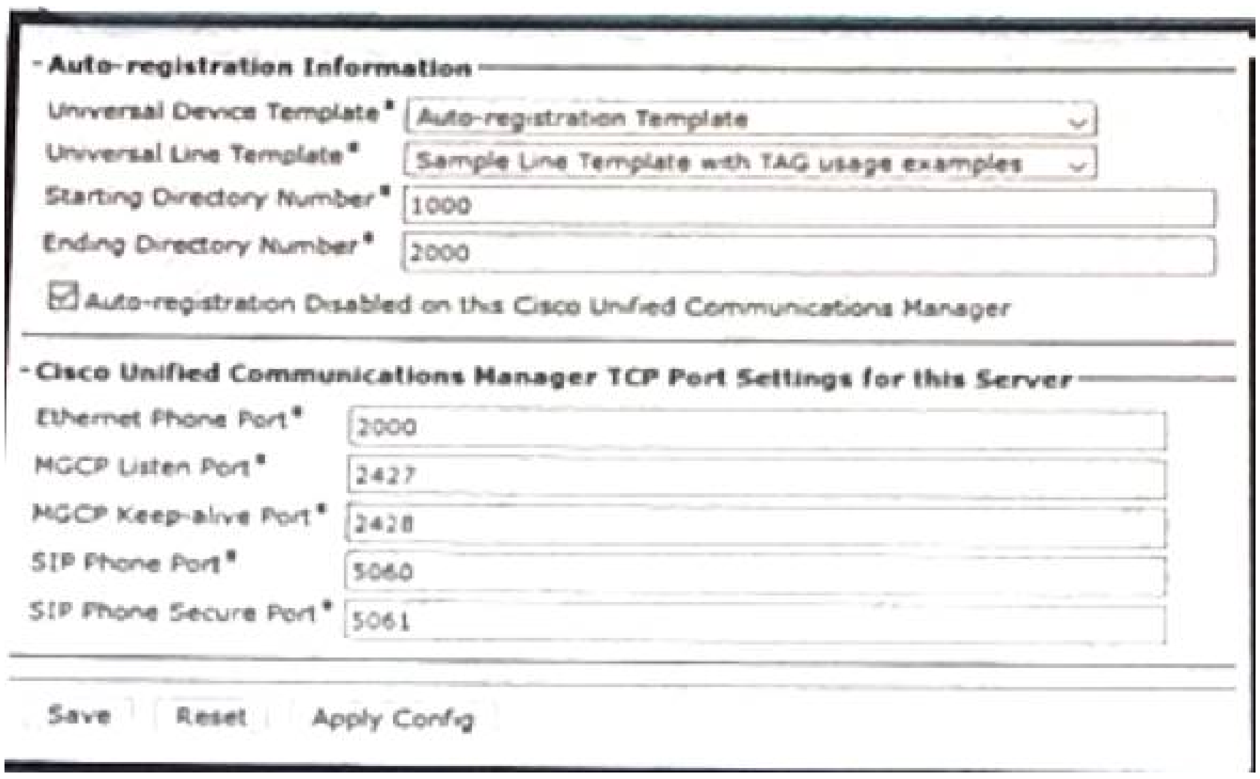
Answer: D

NEW QUESTION 11
A network administrator deleted a user from the LDAP directory of a company. The end user shows as Inactive LDAP Synchronized User in Cisco UCM. Which step is next to remove this user from Cisco UCM?

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

Answer: B

NEW QUESTION 14
Refer to the exhibit.



Which action must an engineer take to implement self-provisioning on a primary communications manager server?

A. Select a different Universal Line Template.
B. Change the SIP Phone Secure Port.
C. Uncheck the auto-registration Disabled checkbox.
D. Select a different Universal Device Template.

Answer: C

NEW QUESTION 18
Which two steps should be taken to provision a phone after the Self-Provisioning feature was configured for end users? (Choose two.)

A. Ask the Cisco UCM administrator to associate the phone to an end user.
B. Plug the phone into the network.
C. Dial the hunt pilot extension and associate the phone to an end user
D. Dial the self-provisioning IVR extension and associate the phone to an end user.
E. Enter settings menu on the phone and press * ,*,# (star, star, pound).

Answer: BD

NEW QUESTION 19
Which two protocols are proxied over an Expressway-E/C pair when a Mobile and Remote Access login including phone services is performed? (Choose two.)

A. HTTPS
B. H.323
C. SIP
D. SCCP
E. SRTP

Answer: AC

NEW QUESTION 22
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 24
Which QoS marking is used when an administrator configures voice call signaling?

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

Answer: B

NEW QUESTION 27
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 31
What are the predefined call handlers in Cisco Unity Connection?

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

Answer: D

NEW QUESTION 32
What is a description of the DiffServ model used for implementing QoS?

A. AF41 has higher drop precedence than AF42. which has higher drop precedence than AF43.
B. Voice and video calls are marked with different DSCP values and placed in different queues.
C. AF43 has higher drop precedence than AF42 but lower drop precedence than AF41.
D. RTP traffic from voice and video calls is marked EF and placed in the same queue.

Answer: A

NEW QUESTION 37
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

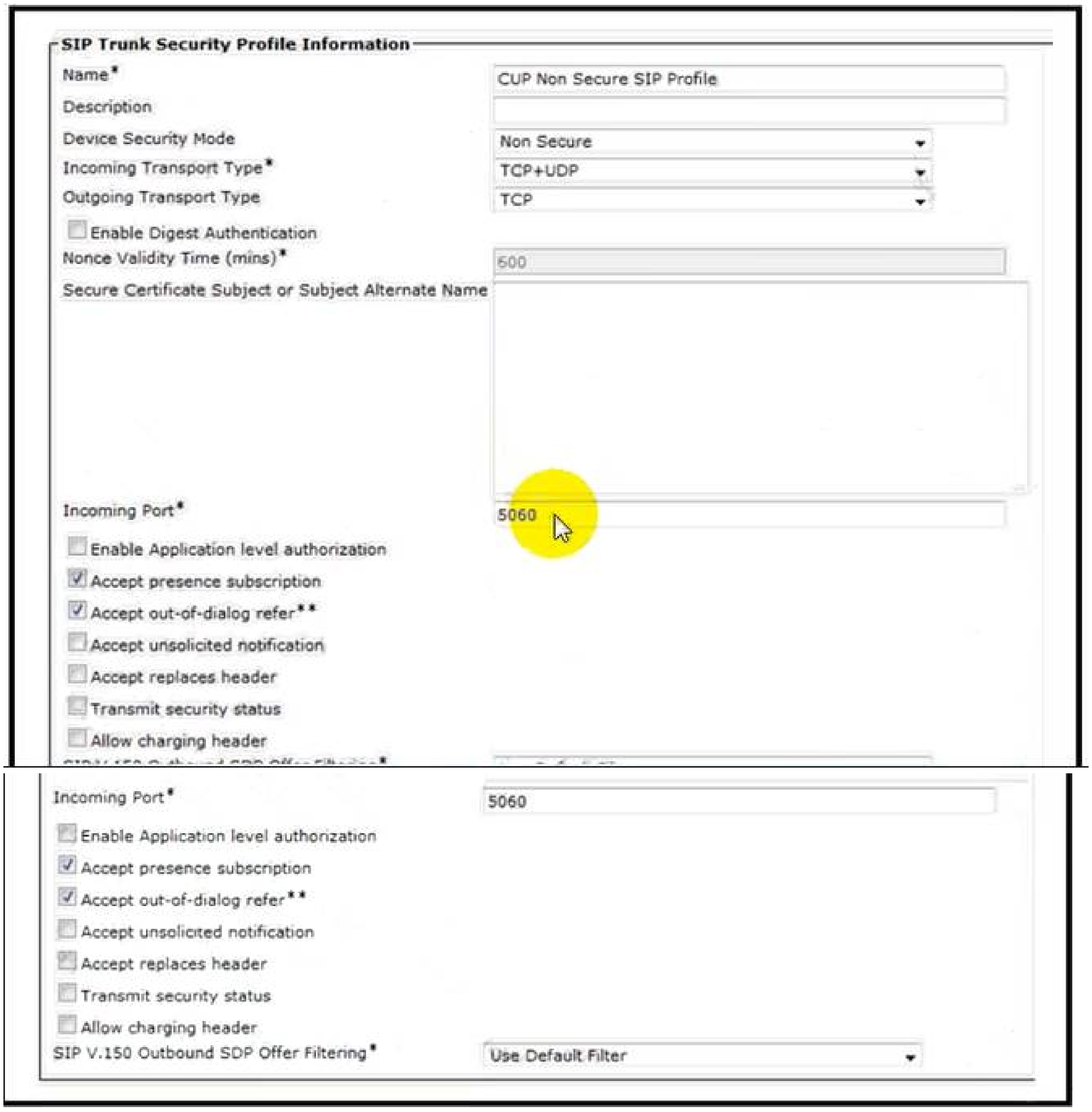
Answer: D

NEW QUESTION 38
An administrator is in the process of moving Cisco Unity Connection mailboxes between mailbox stores. The administrator notices that some mailboxes have active Message Waiting Indicators. What happens to these mailboxes when they are moved?

A. The move will fail if MWI status is active.
B. The MWI status is retained after a mailbox is moved from one store to another.
C. If the source and target mailbox store are not disabled, MWI status is not retained.
D. Moving the mailboxes from one store to another fails if MWI is turned on.

Answer: B

NEW QUESTION 42
Refer to the exhibit.



A collaboration engineer is configuring the Cisco UCM IM and Presence Service. Which two steps complete the configuration of the SIP trunk security profile? (Choose two.)

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

Answer: CE

NEW QUESTION 46
An engineer deploys a Cisco Expressway-E server for a customer who wants to utilize all features on the server. Which feature does the engineer configure on the Expressway-E?

A. H.323 endpoint registrations
B. Mobile and Remote Access
C. SIP gateway for PSTN providers
D. VTC bridge

Answer: B

NEW QUESTION 51
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 56
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 61
Endpoint A is attempting to call endpoint B. Endpoint A only supports G.711ulaw with a packetization rate of 20 ms, and endpoint B supports packetization rate of 30 ms for G.711ulaw. Which two media resources are allocated to normalize packetization rates through transrating? (Choose two.)

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

Answer: BE

NEW QUESTION 62
A collaboration engineer adds a voice gateway to Cisco UCM. The engineer creates a new gateway device in Cisco UCM. selects VG320 as the device type and selects MGCP as the protocol What must be done next to add the gateway to the Cisco UCM database?

A. Select the DTMF relay type for the gateway.
B. Select a device pool for the new gateway.
C. Add the FQDN or hostname of the device.
D. Configure the module in slot 0 of the new gateway.

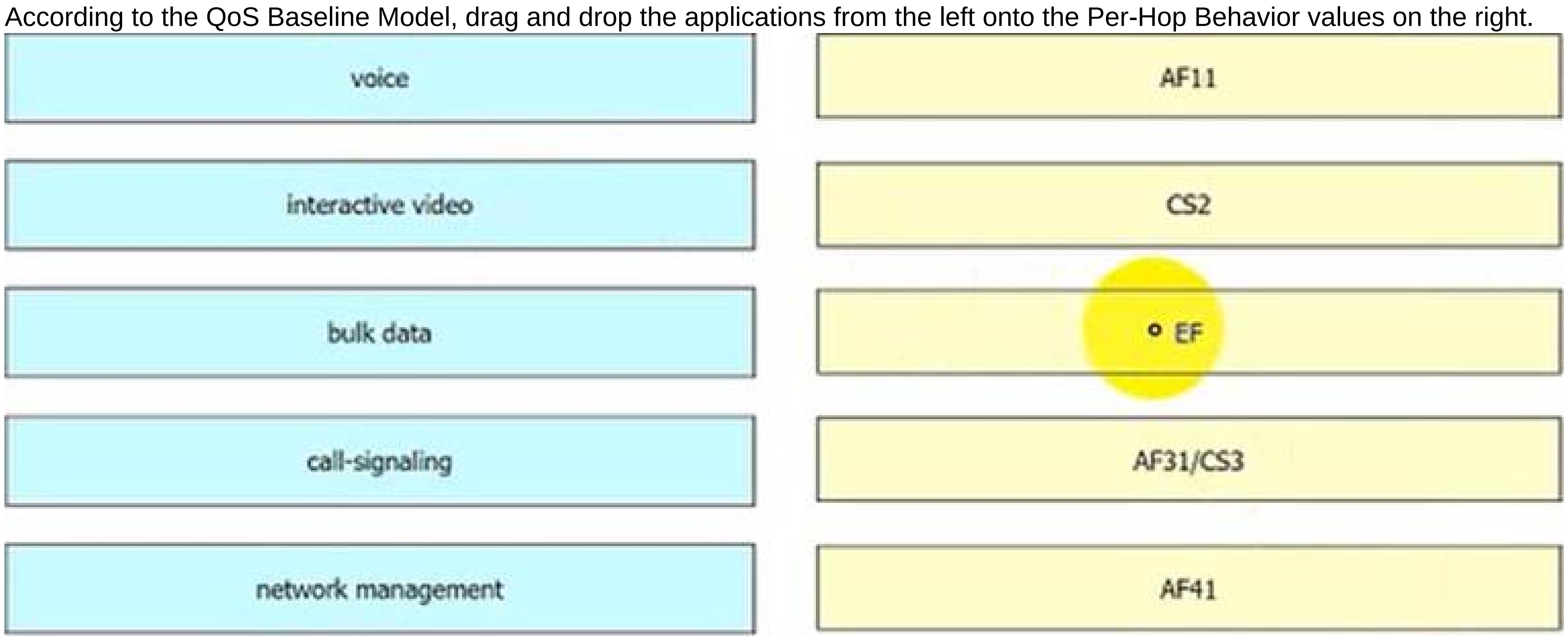
Answer: C

NEW QUESTION 65
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 66
According to the QoS Baseline Model, drag and drop the applications from the left onto the Per-Hop Behavior values on the right.



A. Mastered
B. Not Mastered

Answer: A

Explanation:

NEW QUESTION 70
A Cisco UCM administrator sets up new route patterns to support phones in tour different locations, all with local gateways. The administrator wants to use the same route pattern for all four locations. How must the system be configured to achieve this goal?

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

Answer: B

NEW QUESTION 71
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

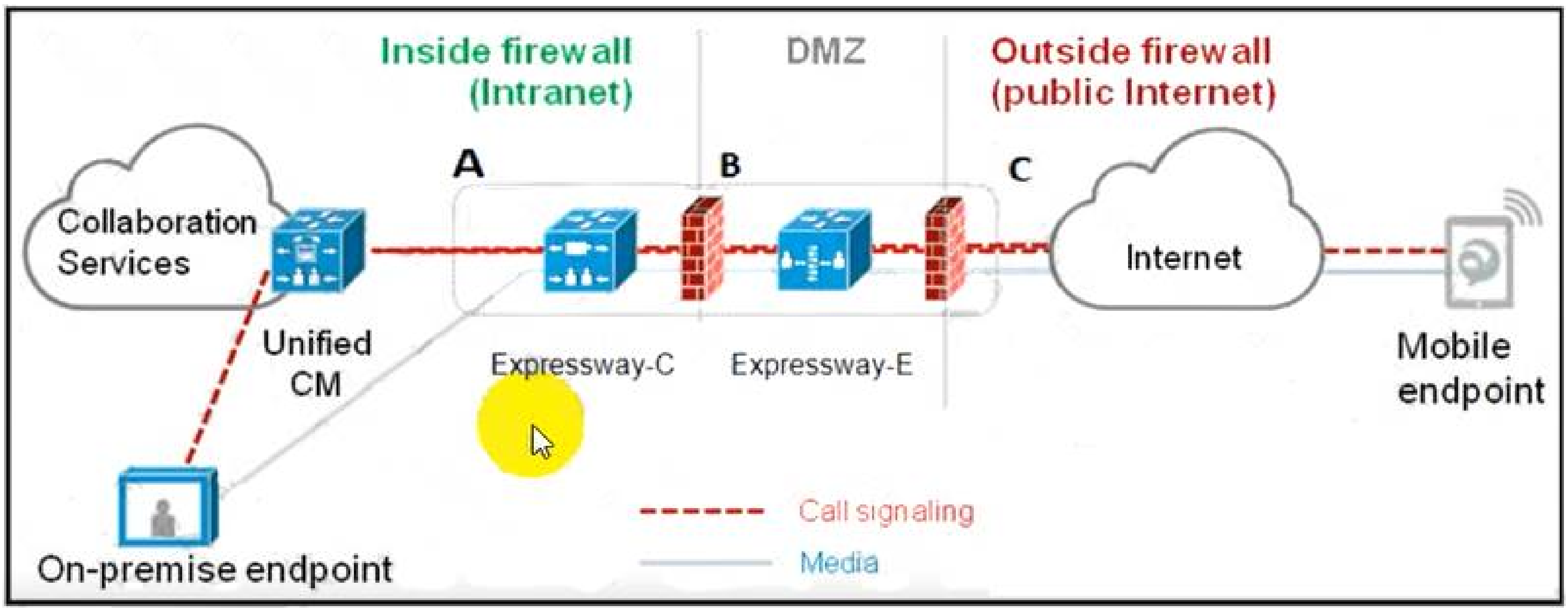
Answer: A

NEW QUESTION 74
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 78
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 79
A user dials 9011841234567 to reach Vietnam. Which steps send the call to the PSTN provider as 011841234567?

A)

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

B)

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

C)

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

D)

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

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

Answer: A

NEW QUESTION 81
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 84
Which two steps are required for bulk configuration transactions on the Cisco UCM database utilizing BAT? (Choose two.)

A. A data file in Abstract Syntax Notation One format must be uploaded to Cisco UCM
B. A server template must be created in Cisco UCM
C. A data file in comma-separated values format must be uploaded to Cisco UCM
D. A data file in Extensible Markup Language format must be uploaded to Cisco UCM
E. A device template must be created in Cisco UCM

Answer: CE

NEW QUESTION 86
Refer to the exhibit.

```
admin:utils ntp status
ntpd (pid 17428) is running...

     remote           refid      st t when poll reach   delay   offset  jitter
==============================================================================
*192.168.1.1     17.253.14.125    2 u   36   64   377    0.435   0.039   0.047
 192.168.1.2     .INIT.          16 u    -   64     0    0.000   0.000   0.000
```

A collaboration engineer adds a redundant NTP server to an existing Cisco Collaboration solution On the Cisco UCM OS Administration page, the new NTP server shows as "Not Accessible" Which action resolves this issue?

A. Restart NTPD on the Cisco UCM server.
B. Delete and re-add the new NTP server via the Cisco UCM command-line interface
C. Start the NTP service on the new NTP server
D. Configure the "reach" value as "377" for the new NTP server.

Answer: C

NEW QUESTION 89
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 94
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?

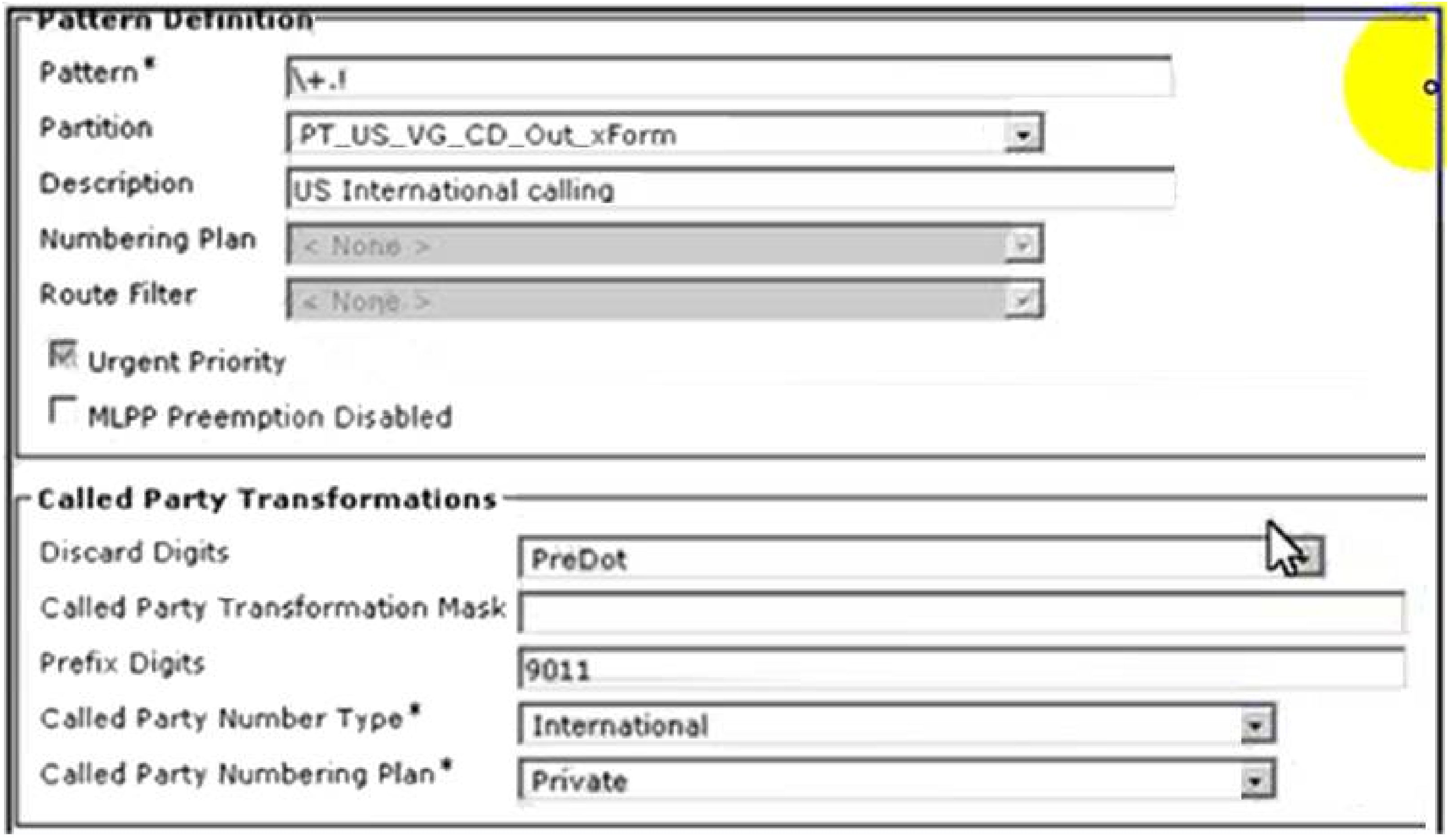
A. Change the service parameter Apply Transformations On Remote Number to True.
B. Configure a translation pattern that has a Calling Party Transform Mask of XXXXXXXXXX.
C. On the inbound SIP trunk, change Significant Digits to 10.
D. Configure a calling party transformation pattern that keeps only the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

Answer: A

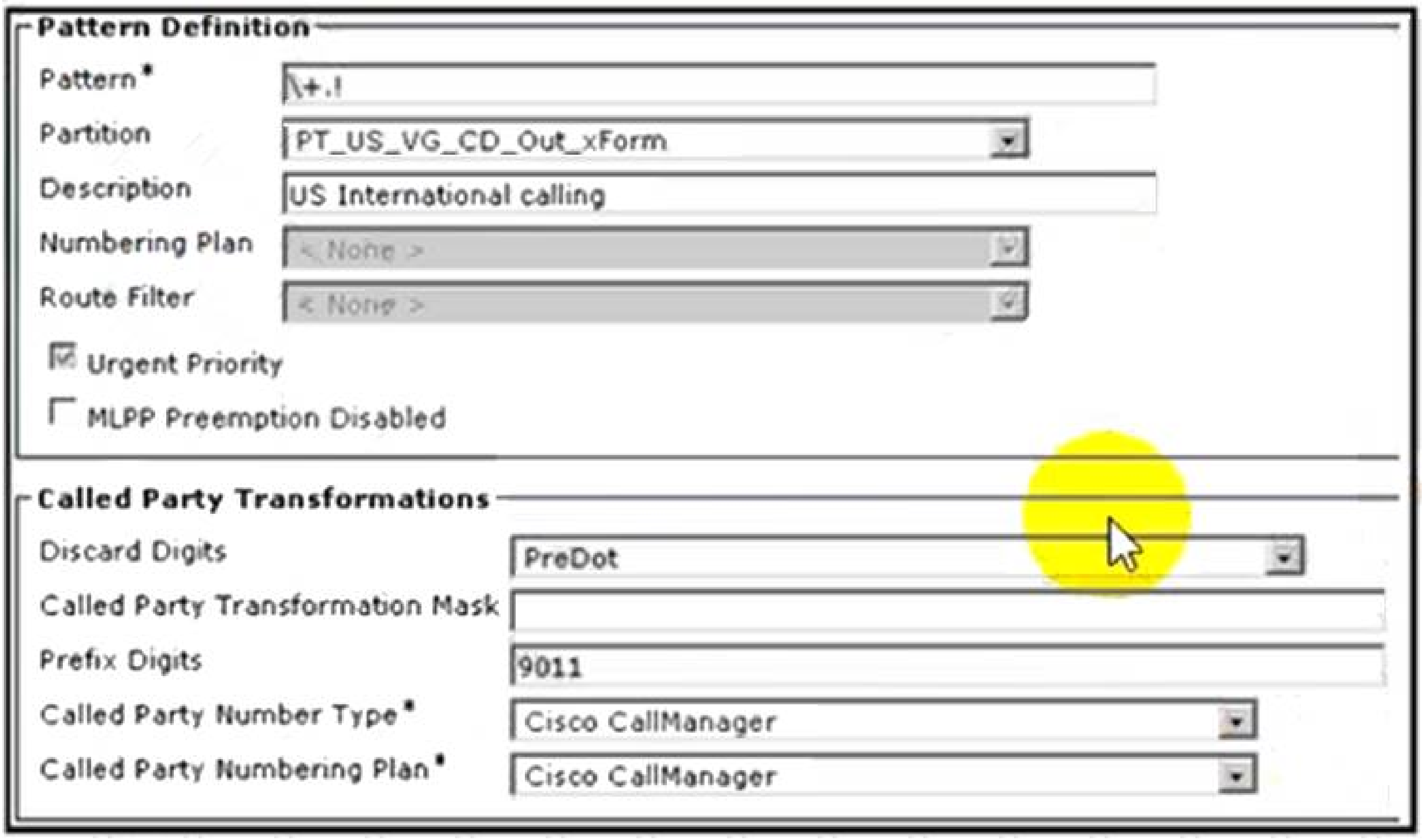
NEW QUESTION 99
Given this H.323 gateway configuration and using Cisco best practices, how must the called party transformation pattern be configured to ensure that a proper ISDN type of number is set?

```
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
```

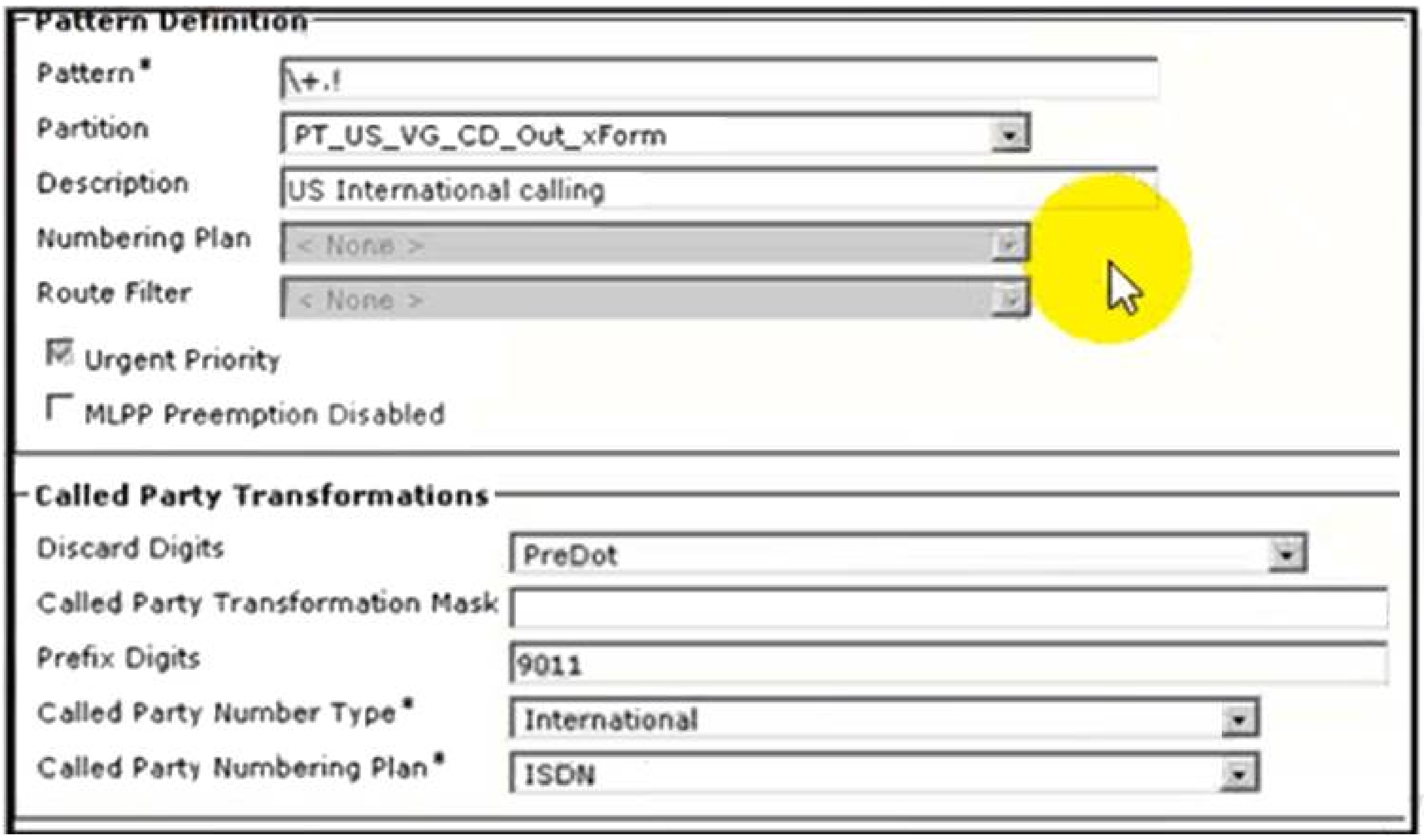
A.


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

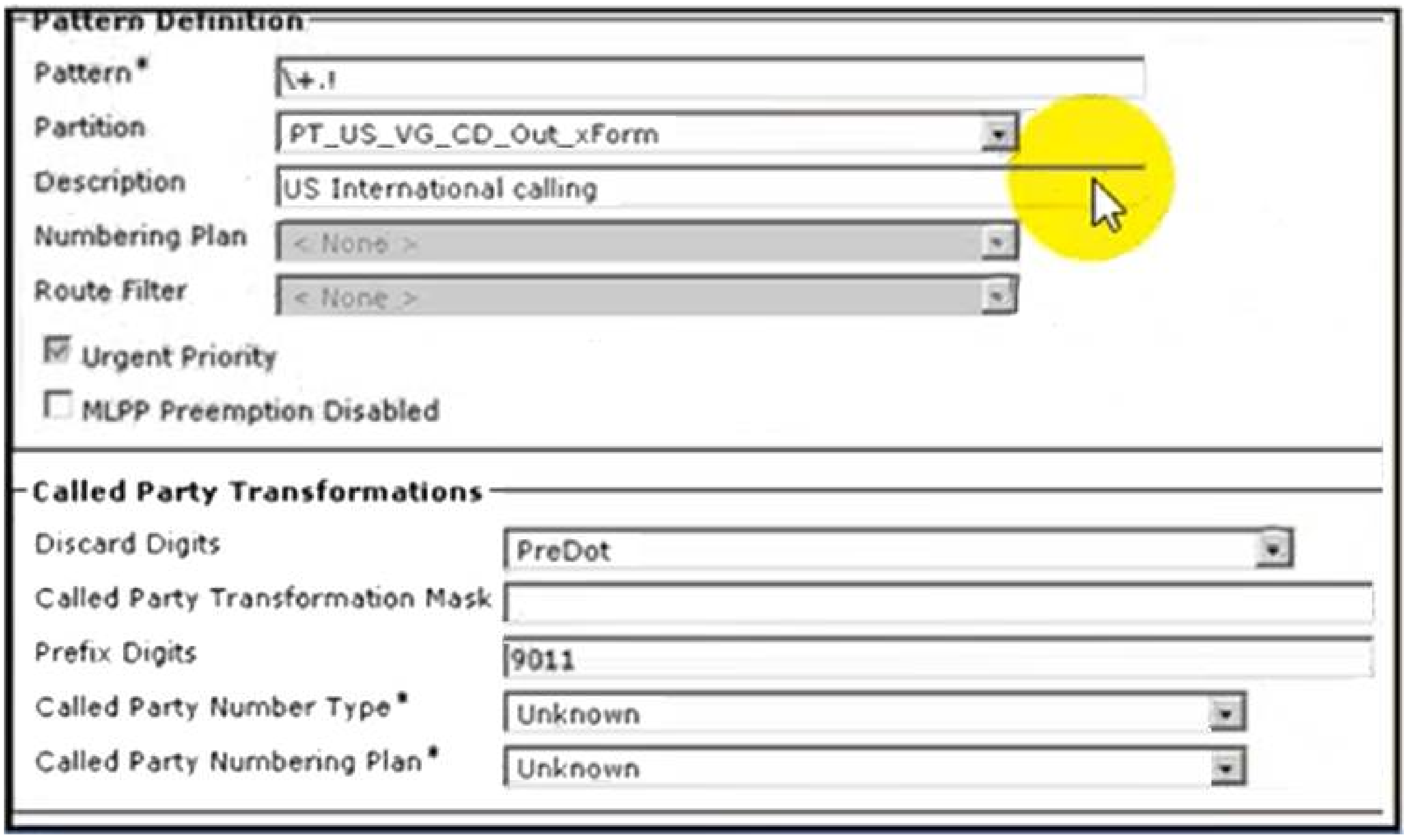
B.


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 103
When a new SIP phone is registered to Cisco Unified communications Manager, it keeps failing and showing an “unprovisioned” error message in the phone display. Which problem is a possible cause of this issue?

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

Answer: A

NEW QUESTION 106
An engineer is configuring a Cisco Unified Border Element to allow the video endpoints to negotiate without the Cisco Unified Border Element interfering in the process. What should the engineer configure on the Cisco Unified Border Element to support this process?

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

Answer: D

NEW QUESTION 109
User A Calls user. The call gets connected, but the quality is bed. What are two reasons for this issue? (Choose two)

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

Answer: CD

NEW QUESTION 110
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 113

Which DSCP marking is represented as 101110 in an IP header?

A. EF
B. CS3
C. AF41
D. AF31

Answer: A

NEW QUESTION 118
What is a characteristic of video traffic that governs QoS requirements for video?

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

Answer: D

NEW QUESTION 119
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 121
......

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