

Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

<https://www.2passeasy.com/dumps/350-801/>



NEW QUESTION 1

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.Ip 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 2

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 3

An engineer is configuring Cisco Jabber for Windows and must implement desk phone control mode for some of the users. Which access control group must be configured for those users to enable this functionality?

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

Answer: A

NEW QUESTION 4

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 5

An engineer is going to redesign a network, and while looking at the QoS configuration, the engineer sees that a portion of the network is marked with AF42. Which type of traffic is marked with this tag?

- A. signaling
- B. voice
- C. video conference
- D. streaming video

Answer: A

NEW QUESTION 6

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 7

A Company s employees have been complaining that they have been unable to select options on the internal IVR of the help desk. IT support has been given Cisco UCM traces and below is the snippet of the SDP of the INVITE packet.

```
m=audio 25268 RTP/AVP 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=ptime:20
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

How is this issue resolved?

- A. Configure CODEC for G.729.
- B. Configure DTMF for KPML.
- C. Configure CODEC for G.722.
- D. Configure DTMF for RFC 2833.

Answer: B

NEW QUESTION 8

Which DiffServe PHB preserves backward compatibility with any IP precedence scheme?

- A. expedited forwarding
- B. class selector
- C. assured forwarding
- D. default

Answer: B

NEW QUESTION 9

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 10

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 10

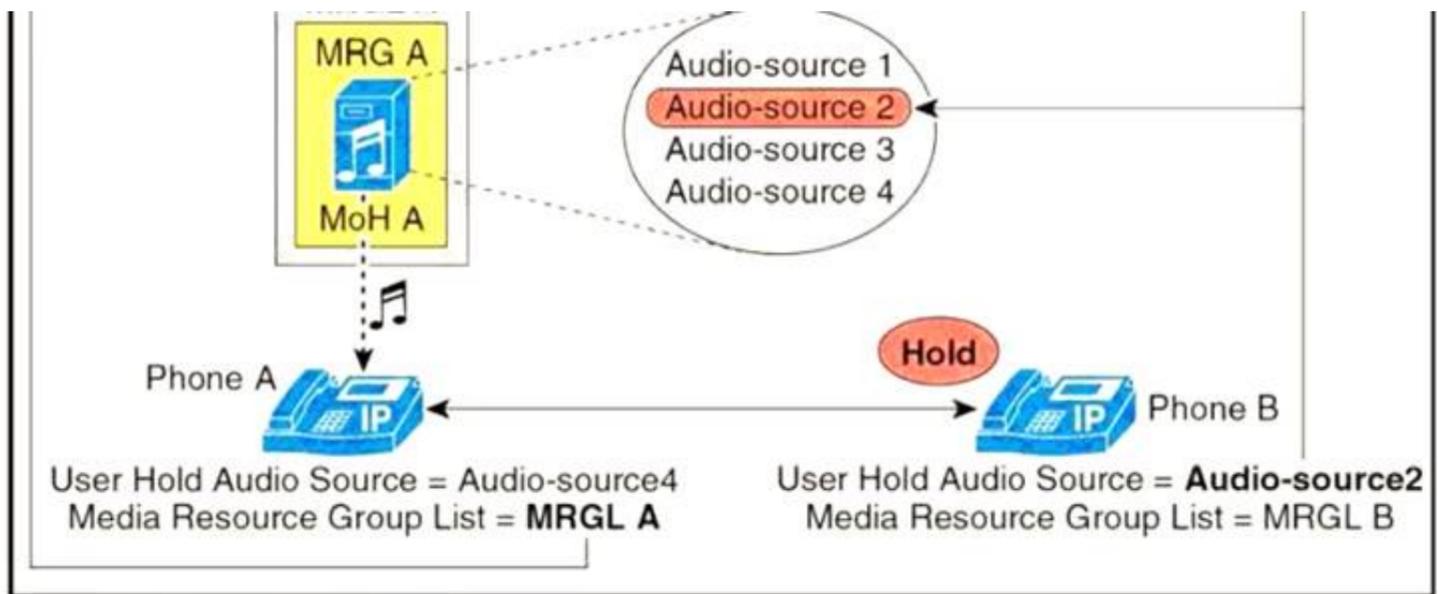
What is the traffic classification for voice and video conferencing?

- 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 11

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 13

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 16

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 21

Refer to the exhibit.

Outbound Calls

Called Party Transformation CSS

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

Use Device Pool Calling Party Transformation CSS

Calling Party Selection*

Calling Line ID Presentation*

Calling Name Presentation*

Calling and Connected Party Info Format*

Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS

Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

Maintain Original Caller ID DN and Caller Name in Identity Headers

Use Device Pool Calling Party Transformation CSS

Calling Party Selection*

Calling Line ID Presentation*

Calling Name Presentation*

Calling and Connected Party Info Format*

Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS

Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

Maintain Original Caller ID DN and Caller Name in Identity Headers

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 22

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 24

Refer to the exhibit.

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 26

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 27

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 32

Refer to the exhibit.

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

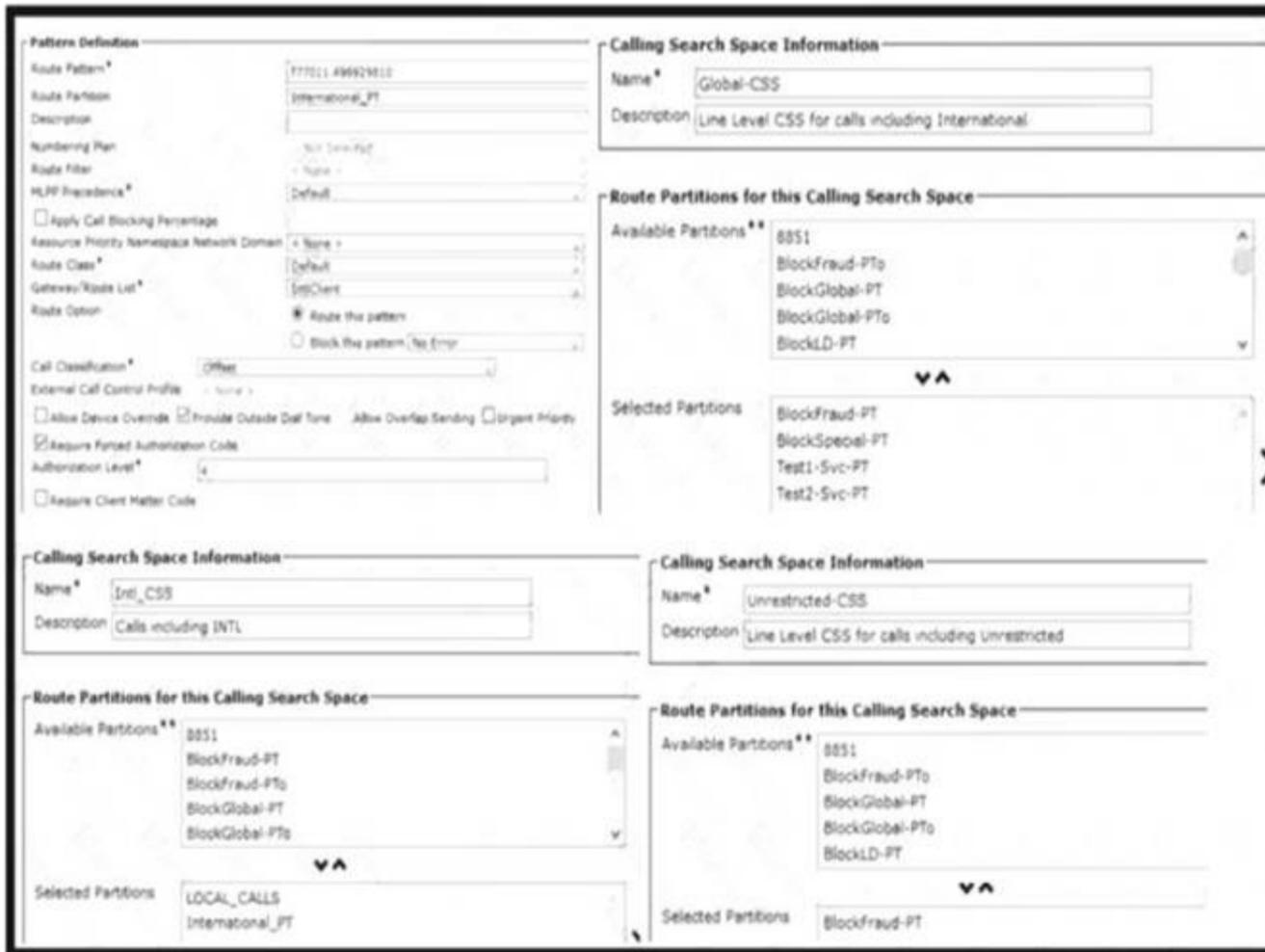
An engineer configures a VoIP dial peer on a Cisco gateway. Which codec is used?

- A. G711alaw
- B. No codec is used (missing codec command)
- C. G.711ulaw
- D. G729r8

Answer: D

NEW QUESTION 34

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: D

NEW QUESTION 35

An engineer is asked to implement on-net/off-net call classification in Cisco UCM. Which two components are required to implement this configuration? (Choose two.)

- A. CTI route point
- B. SIP route patterns
- C. route group
- D. route pattern
- E. SIP trunk

Answer: DE

NEW QUESTION 38

An administrator uses the Cisco Unified Real-Time Monitoring Tool to investigate recent calls on a Cisco UCM cluster. The SIP trace for an on-net, direct-media call shows two 180 Ringing and two 11 BYE messages. Why are there multiples of each message type in the trace?

- A. The source phone sends a 180 Ringing signal to the Cisco UC
- B. which sends a 180 Ringing signal to the destination phon
- C. The same process applies to 11 BYE messages.
- D. The source phone must signal to the destination phone that it is ringing, and the destination phone signals back with a 180 Ringing messag
- E. The same process applies to 11 BYE messages.
- F. The calls have an MTP in the call path due to different codec suppor
- G. The calls are subsequently split into two call legs.
- H. The destination phone signals back to the Cisco UCM that it is ringing, and the Cisco UCM signals back to the source phone.

Answer: A

NEW QUESTION 42

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

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

Answer: AB

NEW QUESTION 45

During the Cisco IP Phone registration process, the TFTP download fails. What are two reasons for 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 47

Cisco UCM delays routing of a call during digit analysis with an overlapping dial plan. How long is the default wait time?

- A. 5 seconds
- B. 10 seconds
- C. 15 seconds
- D. 20 seconds

Answer: C

NEW QUESTION 48

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 50

Users want their mobile phones to be able to access their Cisco Unity connection mailboxes with only having to enter their voicemail pin at the login prompt. Calling pilot number where should an engineer configure this feature?

- A. transfer rules
- B. message settings
- C. alternate extensions
- D. greetings

Answer: C

NEW QUESTION 54

When a call is delivered to a gateway, the calling and called party number must be adapted to the PSTN service requirements of the trunk group. If a call is destined locally, the + sign and the explicit country code must be replaced with a national prefix. For the same city or region, the local area code must be replaced by a local prefix as applicable. Assuming that a Cisco UCM has a SIP trunk to a New York gateway (area code 917), which two combinations of solutions localize the calling and called party for a New York phone user? (Choose two.)

- A. Configure two calling party transformation patterns:
`\+1917.XXXXXXX, strip pre-dot, numbering type: subscriber`
`\+1.!, strip pre-dot, numbering type: national`
- B. Configure the gateway to translate called numbers and apply it to the dial peer. Combine it with a translation profile for calling number:

```

!
voice translation-rule 1
 rule 1 /^1917/ //
 rule 2 /^[+]1917/ //
!
voice translation-profile strip+1
 translate called 1
!

```

C. Configure the gateway to translate the calling number and apply it to the dial peer. Combine it with a translation profile for called nu

```
!
voice translation-rule 1
rule 1 /^1917/ //
rule 2 /^[+]1917/ //
!
voice translation-profile strip+1
translate calling 1
!
```

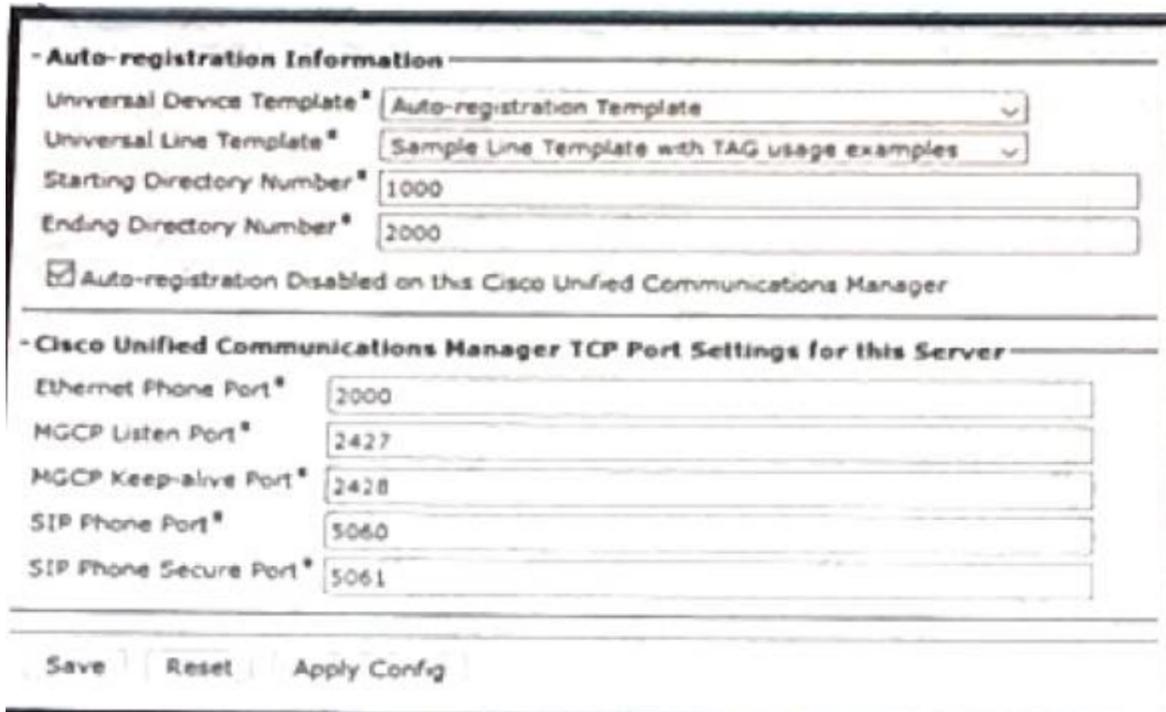
D. Configure two called party transformation patterns:
 \+1917.XXXXXXX, strip pre-dot, numbering type: subscriber
 \+1.!, strip pre-dot, numbering type: national

E. Configure two calling party transformation patterns:
 \+1917.CCCCCC, strip pre-dot, numbering type: subscriber
 \+!, strip pre-dot, numbering type: national

Answer: BC

NEW QUESTION 56

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 58

Refer to the exhibit.

```
voice class codec 20
codec preference 1 g722-64
codec preference 2 ilbc mod 30
!
dial-peer voice 200 voip
destination-pattern ^408555...$
session target ipv4:10.2.3.4
incoming called-number 9T
dtmf-relay h245-alphanumeric rtp-nte
no vad
!
```

An administrator configured a codec preference list with 0,122 and ILBC codecs. Which change must the administrator make in the dial-peer section of the configuration to use this list?

- A. add voice-codecs 20
- B. add session codec 20
- C. add codec preference 20
- D. add voice-class codec 20

Answer: D

NEW QUESTION 60

Refer to the exhibit.

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

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

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

Answer: B

NEW QUESTION 62

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

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

Answer: CD

NEW QUESTION 65

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 70

Which behavior occurs when Cisco UCM has a CallManager group that consists of two subscribers?

- A. Endpoints attempt to register with the bottom subscriber in the list.
- B. Endpoints attempt to register with the top 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: B

NEW QUESTION 75

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 78

An engineer builds the configuration on a Cisco IOS gateway for the dial-peers:

```
dial-peer voice 2 voip
  destination-pattern 5419822100
  session-targeted yes 10.5.5.7
  no protocol 100 is-redundancy 2 fallback class
  no 100 10000
```

Which command is required to complete the configuration?

- A. Codec g726r32
- B. Codec g729cr81
- C. Codec g723ar63
- D. Codec g711ulaw

Answer: D

NEW QUESTION 82

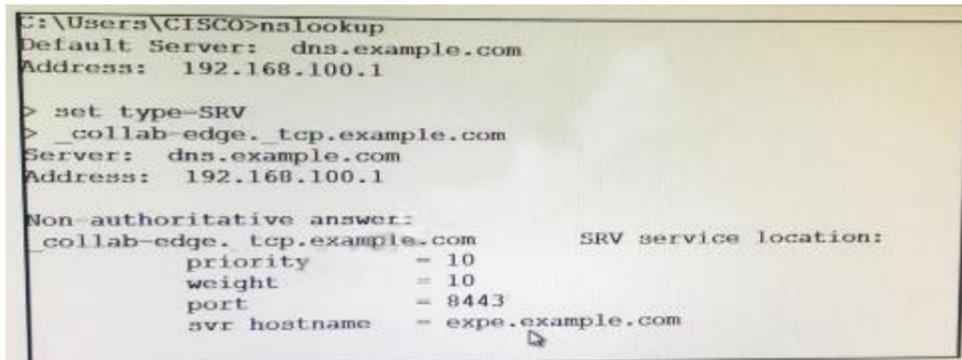
An administrator works with an ISDN PRI that is connected to a third-party PBX. The ISDN link does not come up, and the administrator finds that the third-party PBX uses the OSIG signaling method. Which command enables the Cisco IOS Gateway to use QSIG signaling on the ISDN link?

- A. isdn incoming-voice voice
- B. isdn switch-type basic-ni
- C. isdn switch-type basic-qsig
- D. isdn switch-type primary-qsig

Answer: D

NEW QUESTION 85

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 90

A customer wants a video conference with five Cisco Telepresence 1X5000 Series systems. Which media resource is necessary in the design to fully utilize the immersive functions?

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

Answer: C

NEW QUESTION 93

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 95

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 98

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 100

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 103

What is a characteristic of a SIP endpoint configured in Cisco UCM with 'Use Trusted Relay Point' set to 'On'?

- A. It creates a trust relationship with the called party.
- B. It enables the Use Trusted Relay Point setting from the associated common device configuration.
- C. It enables Cisco UCM to insert an MTP or transcoder designated as a TRP.
- D. If TRP is allocated and MTP is also required for the endpoint
- E. calls fail.

Answer: C

NEW QUESTION 107

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 108

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 112

An administrator is configuring LDAP for Cisco UCM with Active Directory integration. A customer has requested to use "ipphone" instead of "telephoneNumber" as the phone number attribute. Where does the administrator specify this attribute mapping in Cisco UCM?

- A. LDAP Custom Filter
- B. LDAP Directory user fields
- C. LDAP Directory custom user fields
- D. LDAP Authentication

Answer: B

NEW QUESTION 116

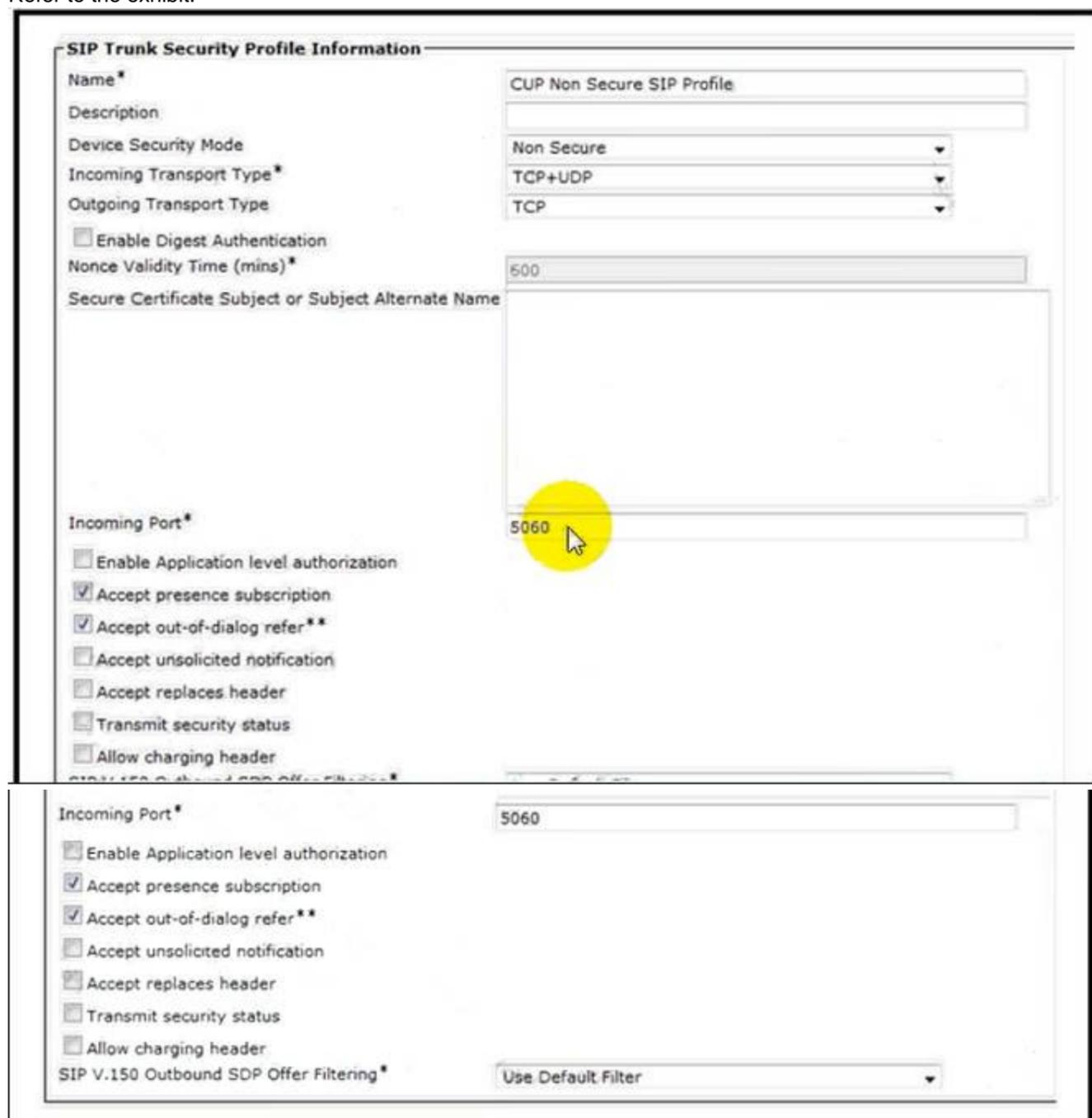
An administrator is in the process of moving Cisco Unity Connection mailboxes between mailbox stores. The administrator notices that some mailboxes have active Message Waiting Indicators. What happens to these mailboxes when they are moved?

- 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 118

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 122

Refer to the exhibit.

```
Via: SIP/2.0/TCP
10.10.10.2:5060;branch=a8bH5bK7954A198F
From:
<sip:012345678@10.10.10.2>;tag=8D79AF62-DB2
To: <sip:90123456@10.10.4.14>;
tag=811681~ffa80926-5fac-4cc5-b802-
2dbde74ae7w2-
v=0
o=CiscoSystemsCCM-SIP 811681 1 IN IP4
10.10.4.14
s=SIP Call
c=IN IP4 10.5.4.3
t=0 0
m=audio 27839 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

Which Codec is negotiated?

- A. G.729
- B. ILBC
- C. G.711ulaw
- D. G.728

Answer: C

NEW QUESTION 124

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 126

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 127

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-maxcapture=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

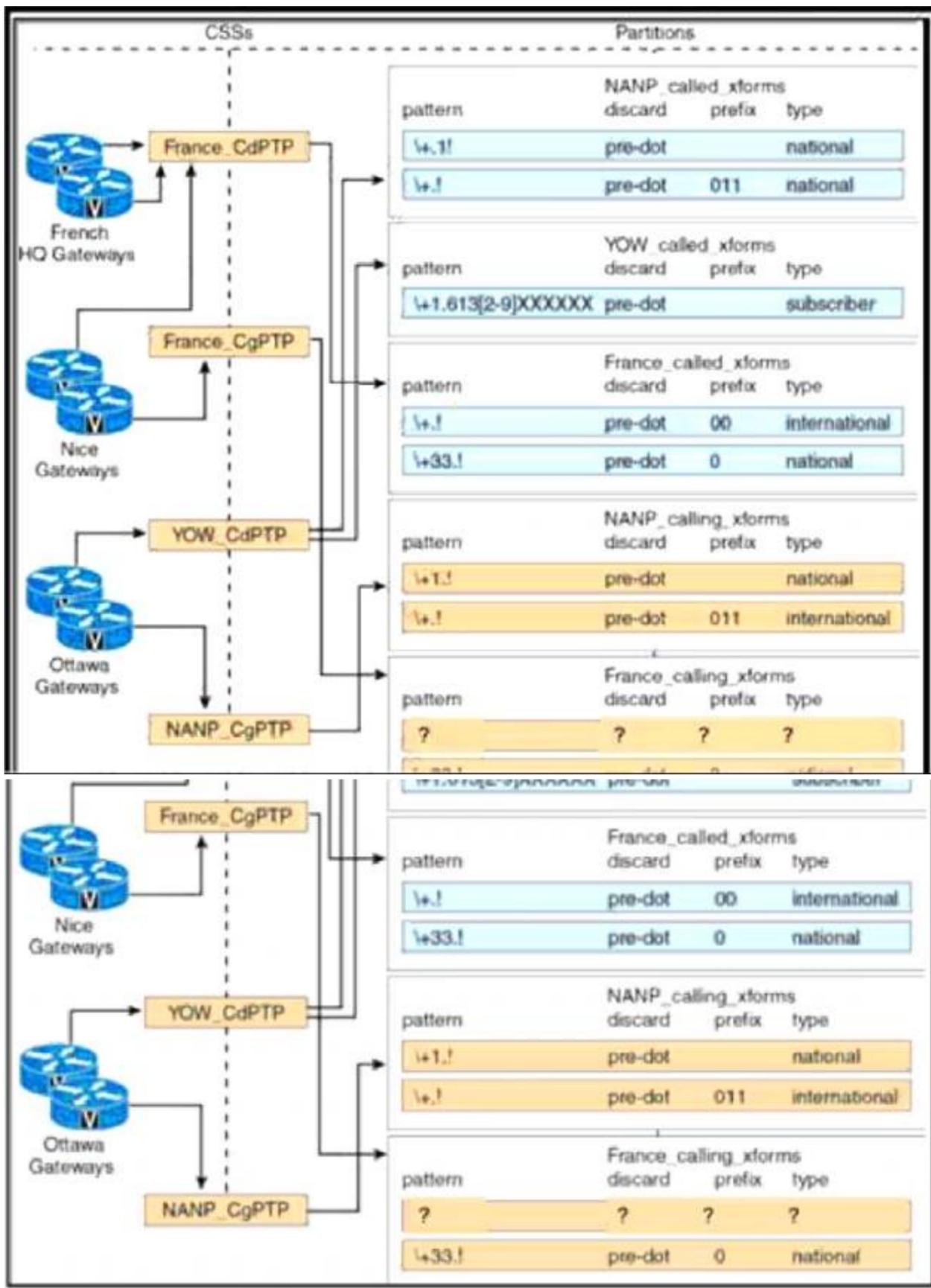
How are network devices monitored in a collaboration network?

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

Answer: C

NEW QUESTION 133

Refer to the exhibit A call from +1 613 555 1234 that is sent out through the Nice Gateways should result in a calling party of 001 613 555 1234 with the numbering type "international" Which configuration accomplishes this goal?



- A. \+.001! pre-dot 1 international
- B. \+1.1 none pre-dot 001 international
- C. \+! pre-dot 00 international
- D. 613XXXXXXX none +011 international

Answer: C

NEW QUESTION 137

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 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 142

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 147

Which command in the MGCP gateway configuration defines the secondary Cisco UCM server?

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

Answer: A

NEW QUESTION 152

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 153

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

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

- A. Mastered
- B. Not Mastered

Answer: A

Explanation:

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

NEW QUESTION 156

Which attribute contains an XMPP stanza?

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

Answer: A

NEW QUESTION 161

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 164

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 167

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 172

An engineer implements QoS in the enterprise network. Which command is used to verify 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 173

Refer to the exhibit.

```
controller t1 0/0/1
pri-group timeslots 1-24
clock source line
linecode b8zs
framing esf
```

An administrator must replace the T1 card with an E1 card. What is the correct configuration if the administrator was asked to configure 12 time slots?

- A.

```
controller e1 0/0/1
pri-group timeslots 1-12
clock source network
linecode hdb3
framing crc4
```
- B.

```
controller e1 0/0/1
pri-group timeslots 1-11, 12
clock source line
linecode hdb3
framing crc4
```
- C.

```
controller e1 0/0/1
pri-group timeslots 1-12
clock source line
linecode hdb3
framing crc4
```

D. controller e1 0/0/1
 pri-group timeslots 1-12
 clock source line
 linecode crc4
 framing hd3

Answer: C

NEW QUESTION 178

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 182

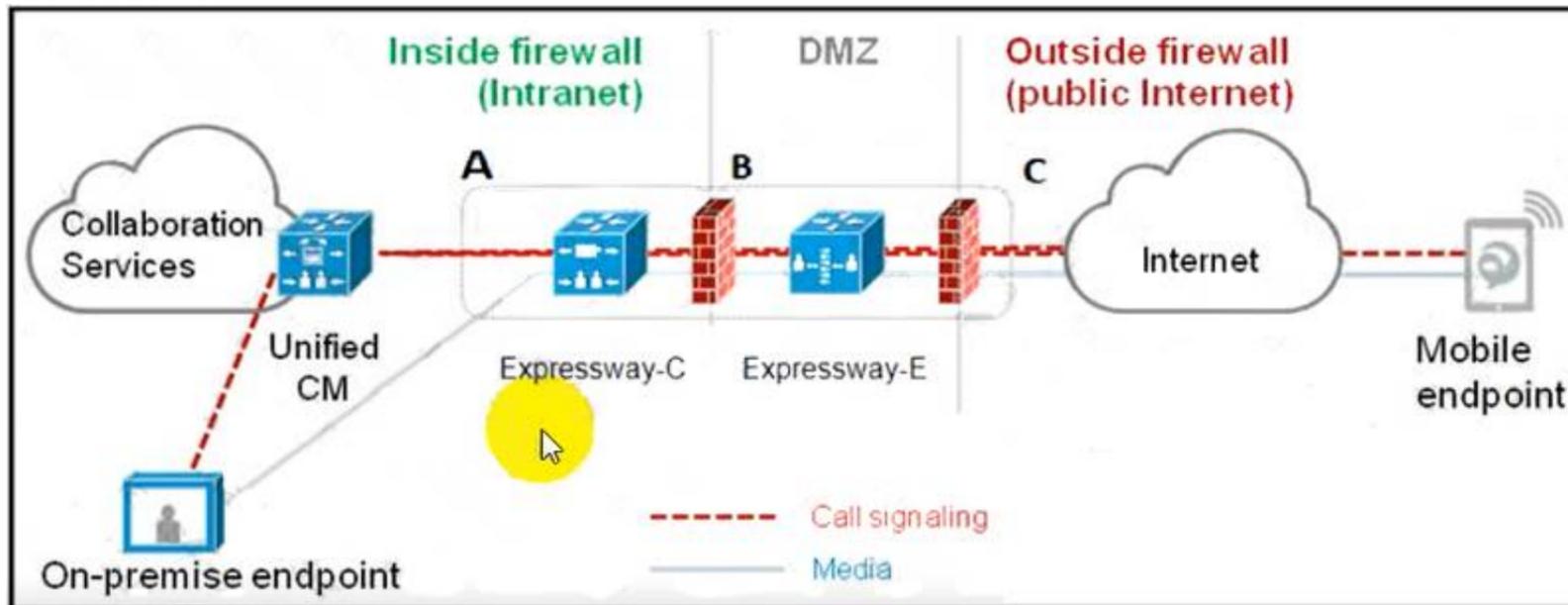
Which action prevents toll fraud in Cisco UCM?

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

Answer: D

NEW QUESTION 183

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 184

What happens to voice packets from a Cisco 8845 IP phone in the QoS trust boundary?

- A. The voice packets are not trusted, and the access layer switch reclassifies the packets.
- B. The voice packets are classified by the phone, and the classification is accepted
- C. The voice and access layer switch negotiate the classification of packets
- D. Cisco UCM determines how the voice packers are classified.

Answer: B

NEW QUESTION 185

Refer to the exhibit.

```

Server: Cisco-ServiceGateway/1.0.0.0
CSeq: 101 OPTIONS
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,
INFO, REGISTER
Allow-Events: telephone-event
Accept: application/sdp
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Content-Type: application/sdp
Content-Length: 369

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

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

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

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

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

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

Answer: D

NEW QUESTION 187

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 192

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 XXXXXXXXXXXX.
- 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 195

An administrator configures international calling on a Cisco UCM cluster and wants to minimize the number of route patterns that are needed. Which route pattern enables the administrator to match variable-length numbers?

- A. 9.011#
- B. 9.011@
- C. 9.011!
- D. 9.011*

Answer: C

NEW QUESTION 198

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 >
<input checked="" type="checkbox"/> Urgent Priority <input type="checkbox"/> 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 >
<input checked="" type="checkbox"/> Urgent Priority <input type="checkbox"/> 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

Description

Numbering Plan

Route Filter

Urgent Priority

MLPP Preemption Disabled

Called Party Transformations

Discard Digits

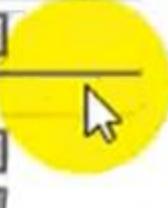
Called Party Transformation Mask

Prefix Digits

Called Party Number Type*

Called Party Numbering Plan*

D. **Pattern Definition**

Pattern* 

Partition

Description

Numbering Plan

Route Filter

Urgent Priority

MLPP Preemption Disabled

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type*

Called Party Numbering Plan*

Answer: C

NEW QUESTION 203

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 205

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 209

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 214

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 218

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 221

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

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

Answer: D

NEW QUESTION 222

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 224

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

- A. EF
- B. CS3
- C. AF41

D. AF31

Answer: A

NEW QUESTION 226

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 227

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

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

Answer: AD

NEW QUESTION 230

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 233

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 234

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. C:\Users\wk\Desktop\mudassar\Untitled.jpg

```
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. C:\Users\wk\Desktop\mudassar\Untitled.jpg

```
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. C:\Users\wk\Desktop\mudassar\Untitled.jpg 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 239

What is the purpose of Mobile and Remote Access (MRA) in the Cisco UCM architecture?

- A. MRA is used to access Webex cloud services only if authenticated with on-premises LDAP service.
- B. MRA is used to make secure PSTN calls by Cisco UCM only while on-premises authentication.
- C. MRA is used to make B2B calls through Expressway registration.
- D. MRA is used to access the collaboration services offered by Cisco UCM from off-premises network connections

Answer: D

NEW QUESTION 242

.....

THANKS FOR TRYING THE DEMO OF OUR PRODUCT

Visit Our Site to Purchase the Full Set of Actual 350-801 Exam Questions With Answers.

We Also Provide Practice Exam Software That Simulates Real Exam Environment And Has Many Self-Assessment Features. Order the 350-801 Product From:

<https://www.2passeasy.com/dumps/350-801/>

Money Back Guarantee

350-801 Practice Exam Features:

- * 350-801 Questions and Answers Updated Frequently
- * 350-801 Practice Questions Verified by Expert Senior Certified Staff
- * 350-801 Most Realistic Questions that Guarantee you a Pass on Your FirstTry
- * 350-801 Practice Test Questions in Multiple Choice Formats and Updatesfor 1 Year