

Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

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NEW QUESTION 1

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 2

When a remote office location is set up with limited bandwidth resources, which codec would allow the most voice calls with the limited bandwidth?

- A. G.722
- B. G.711
- C. G.729
- D. G.723

Answer: C

NEW QUESTION 3

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 4

Which two technical reasons make QoS a necessity in a video deployment? (Choose Two)

- A. Low response time between endpoints
- B. Provisioned bandwidth of the link
- C. Variable bit rate of the video stream
- D. Bursly behavior of video traffic

Answer: CD

NEW QUESTION 5

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 6

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

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

Answer: CE

NEW QUESTION 7

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 8

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 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

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 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 15

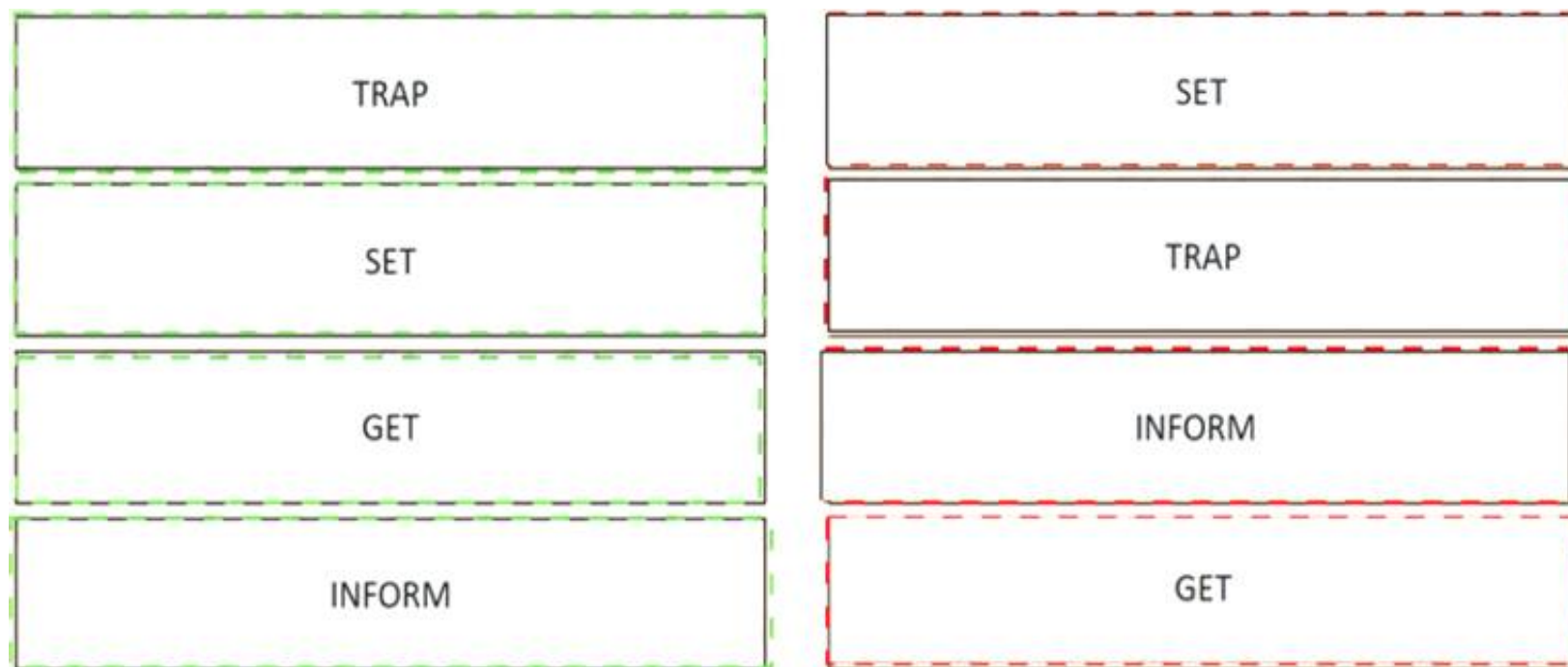
Drag and drop the SNMPv3 message types from the left onto the corresponding definitions on the right.

TRAP	messages used to modify a value of an object variable
SET	unreliable messages that alert the SNMP manager to a condition on the network
GET	reliable messages that alert the SNMP manager to a condition on the network
INFORM	messages used to retrieve an object instance

- A. Mastered
- B. Not Mastered

Answer: A

Explanation:



NEW QUESTION 16

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 21

Refer to the exhibit.

```
Voice class codec 1
codec preference 1 g711alaw
codec preference 2 g711ulaw
codec preference 3 g729r8

dial-peer voice 13 voip
description incoming dialpeer from ITSP
incoming called-number .
voice-class codec 1

dial-peer voice 19 voip
description outgoing dialpeer to CUCM
destination-pattern T
session protocol sipv2
session-target ipv4.3.3.3
voice-class codec 1

Incoming SDP from ITSP

v=0
o=sip.test@2.2.2.2 1 16 IN IP4 2.2.2.2
s=sip.test@2.2.2.2
c=IN IP4 2.2.2.2
t=0 0
m=audio 5000 RTP/AVP 18 0
a=rtpmap:0 PCMU/8000/1
a=rtpmap:18 G729/8000/1
```

Which outgoing m-line SDP is sent to Cisco UCM after matching the appropriate dial peers via Cisco Unified Border Element?

- A. m=audio 16550 RTP/AVP 8 0 18
a=rtpmap:0 PCMU/8000/1
a=rtpmap:8 PCMA/8000/1
a=rtpmap:18 G729/8000/1
- B. m=audio 16550 RTP/AVP 18 0
a=rtpmap:0 PCMU/8000/1
a=rtpmap:18 G729/8000/1
- C. m=audio 16550 RTP/AVP 18 0
a=rtpmap:8 PCMA/8000/1
a=rtpmap:0 PCMU/8000/1
a=rtpmap:18 G729/8000/1
- D.

```
m=audio 16550 RTP/AVP 0 8 18
a=rtpmap:0 PCMU/8000/1
a=rtpmap:8 PCMA/8000/1
a=rtpmap:18 G729/8000/1
```

Answer: B

NEW QUESTION 24

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

NEW QUESTION 27

Refer to the exhibit.

```
isdn switch-type primary-ni
controller t1 0/1/0
framing esf
linecode b8zs
pri-group timeslots 1-10
```

An engineer configures ISDN on a voice gateway. The provider confirms that the PRI is configured with 10 channels the engineer ordered and is working from the provider side, but the engineer cannot get a B-channel to carry voice. The rest of the configuration for the serial interface and voice network is functioning correctly. Which actions must be taken to carry voice?

- A. The engineer must activate the controller card on the voice gateway before configuring the device.
- B. The engineer used a T1 interface but must use an E1 interface.
- C. The pri-group timeslots command must be 0-9 for the 10 channels because all values on a router start with 0.
- D. The engineer must manually revert the order of using the channels.

Answer: A

NEW QUESTION 30

Refer to the exhibit.

```
voice translation-rule 1
rule 1 /^[2-9].....$/ /\0/ type any subscriber
rule 2 /^1[2-9]..[2-9].....$/ /\0/ type any subscriber
```

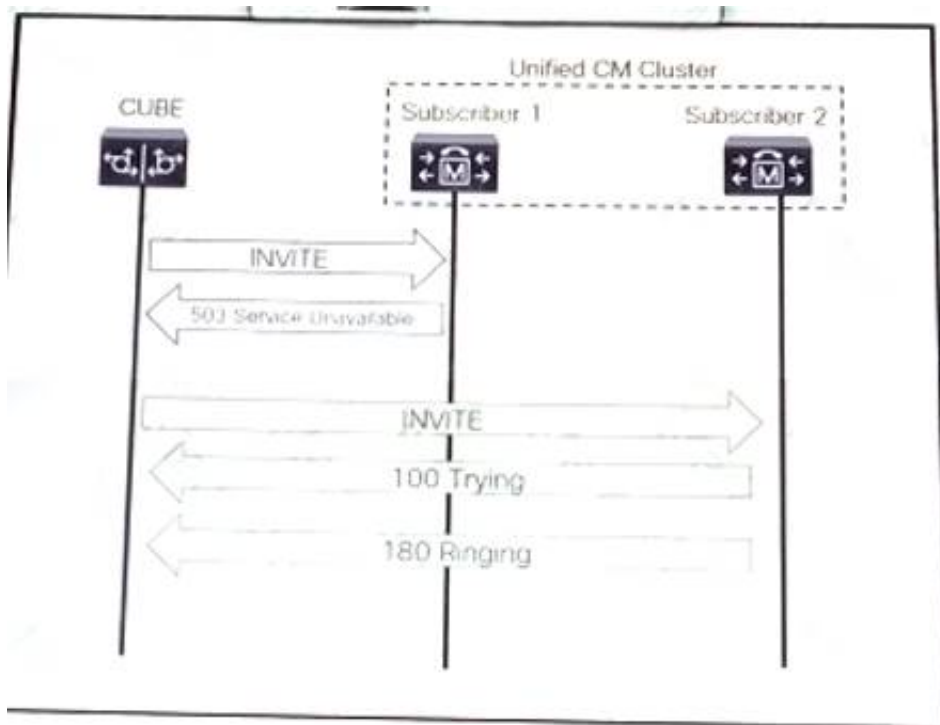
What is the result of applying these two rules to a voice translation profile for use with an ISDN T1 PRI on a Cisco Voice Gateway?

- A. The leading Plus is stripped from the numeric phone number.
- B. The ISDN Plan is modified to the administrator's defined value.
- C. Any zero is stripped from the numeric phone number.
- D. The ISDN Type is modified to the administrator's defined value.

Answer: D

NEW QUESTION 33

Refer to the exhibit.



Cisco Unified element is attempting to establish a call with Subscribers1, but the call fails. Cisco Unified Border Element then retries the same call with Subscribers2, and the call proceeds normally.

Which action resolves the issue?

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

Answer: A

NEW QUESTION 38

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 39

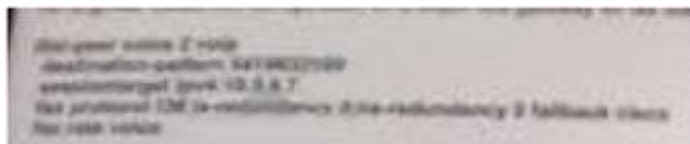
Which behavior occurs when Cisco UCM has a CallManager group that consists of two subscribers?

- 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 44

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



Which command is required to complete the configuration?

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

Answer: D

NEW QUESTION 45

Refer to the exhibit.

```

INVITE sip:4000@172.16.1.1:5061 SIP/2.0
Via: SIP/2.0/TLS 172.16.2.143:5061;branch=z9hG4bK8FD31SE7
Remote-Party-ID: <sip:+14088335000@172.16.2.143>;party=calling;screen=no;privacy=off
From: <sip:+14088335000@172.27.2.143>;tag=7B42E5F6-9B8
To: <sip:4000@172.16.1.1>
Date: Tue, 06 Aug 2019 15:03:05 GMT
Call-ID: 4EA4363-B77111E9-8A4AFFCF-10B6D71B@172.16.2.143
Supported: 100rel,timer,resource-priority,replaces,sdp-angat
Min-SE: 1800
Cisco-Guid: 0082391505-3077640681-2319777743-0280418075
User-Agent: Cisco-SIPGateway/IOS-15.5.3.S4b
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1565089565
Contact: <sip:+14088335000@172.16.2.143:5061;transport=tls>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 68
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 416
v=0
o=CiscoSystemsSIP-GW-UserAgent 8486 8298 IN IP4 172.16.2.143
s=SIP Call
c=IN IP4 172.16.2.143
t=0 0
m=audio 44612 RTP/SAVP 0 101
c=IN IP4 172.16.2.143
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
aptime:20

```

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

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

Answer: A

NEW QUESTION 48

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 PVD4-128

Answer: C

NEW QUESTION 52

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 56

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)

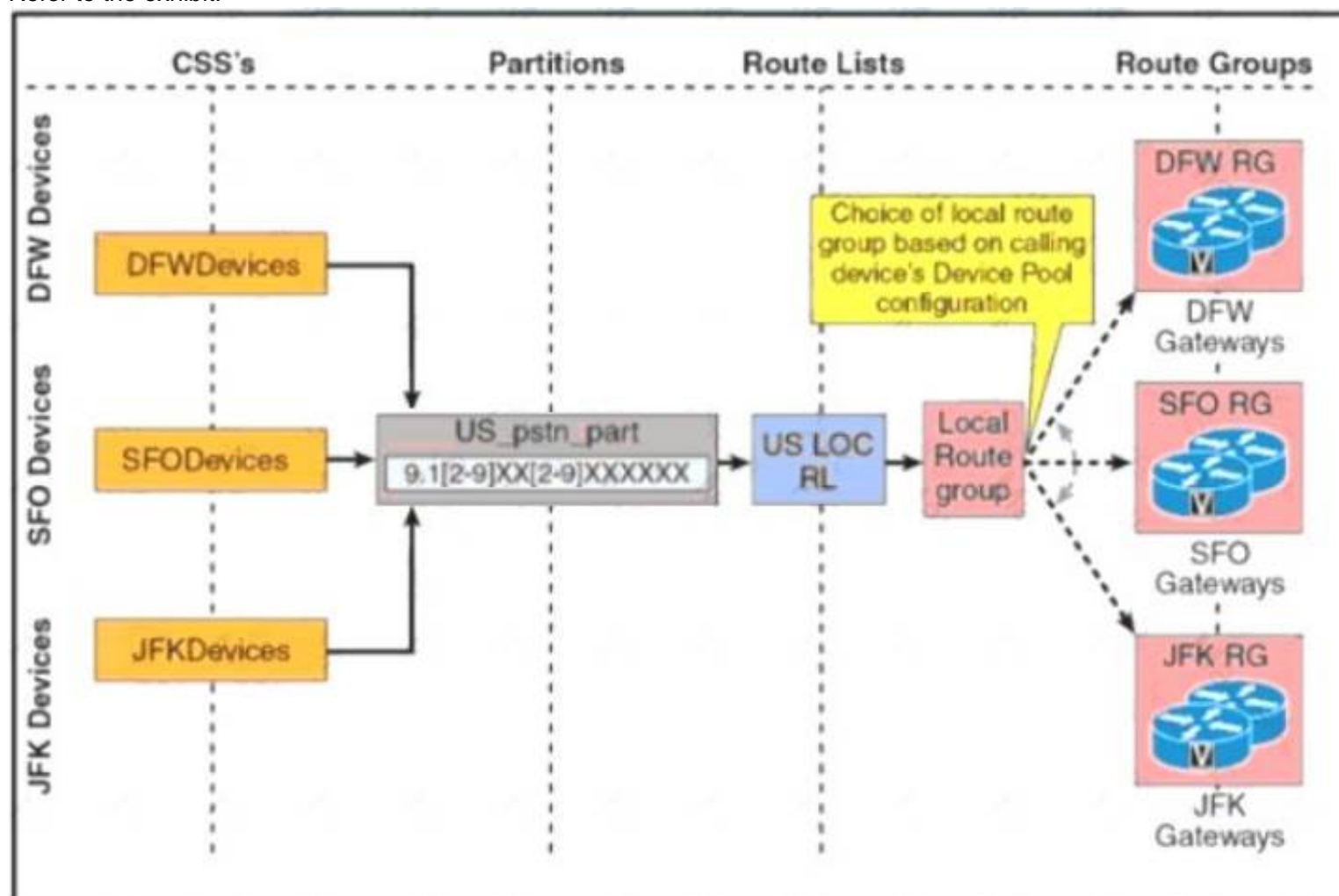
```
dial-peer voice 1 voip
translation-profile outgoing Strip9
max-conn 5
destination-pattern 91[2-9].[2-9]...$
session protocol sipv2
session target ipv4:142.45.10.1
dtmf-relay rtp-nte sip-notify sip-kpml
```

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

Answer: B

NEW QUESTION 59

Refer to the exhibit.



A user takes a phone from San Francisco to New York for a short reassignment. The phone was set up to use the San Francisco device pool, and device mobility is enabled on the Cisco UCM. The user makes a call that matches a route pattern in a route list that contains the Standard Local Route Group. To where does the call retreat?

- A. The call fails because device mobility is turned on, and the phone is not configured in New York
- B. The engineer must configure which sites the device should be roaming to.

- C. The call egresses in San Francisco because the user uses device mobility and is allowed to roam while still keeping the number and resources assigned in San Francisco.
- D. The call fails because the Standard Local Route Group is being used only if no configuration is set for the device pools.
- E. The call egresses in New York because the device automatically is assigned a New York device pool and uses the local gateway.

Answer: B

NEW QUESTION 63

To provide high-quality voice and take advantage of the full voice feature set, which two access layer switches provide support?

- 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 802.1Q trunking and 802.1p 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 required 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 65

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 67

Refer to the exhibit. An engineer is configuring 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 69

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

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

Answer: A

NEW QUESTION 70

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 75

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 77

Refer to the exhibit.

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

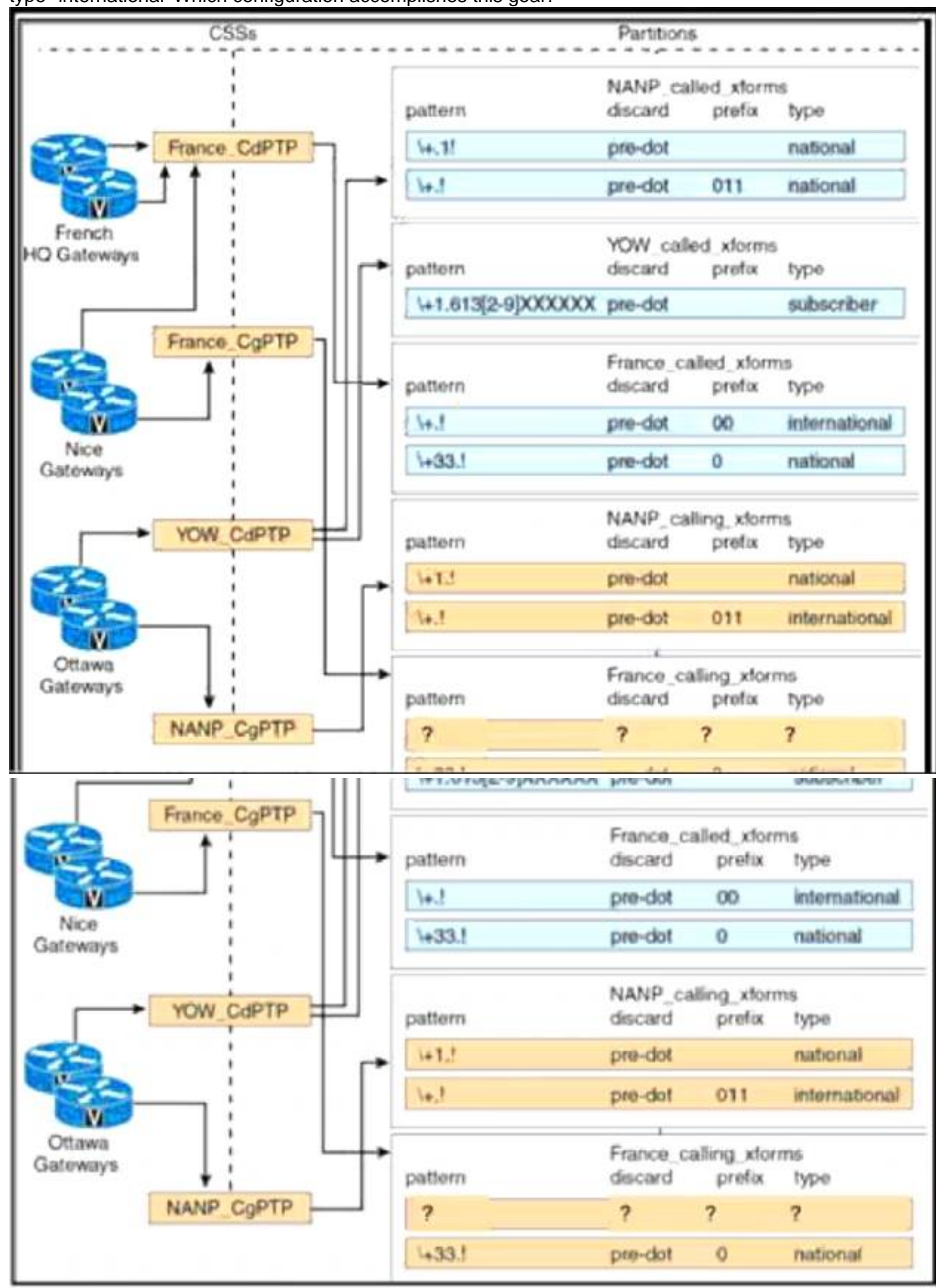
An administrator must fix the SRV records to ensure that server1. sample.com is always contacted first from the three servers. Which solution should the engineer apply to resolve this issue?

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

Answer: D

NEW QUESTION 82

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. \+1! pre-dot 00 international
- D. 613XXXXXXX none +011 internationa

Answer: C

NEW QUESTION 84

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 85

Which DHCP option must be set up for 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 88

What are the last two bits of a DS field in DiffServe Byte used for?

- A. INC
- B. AFxy
- C. ECN
- D. RMI

Answer: C

NEW QUESTION 90

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 94

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 95

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 99

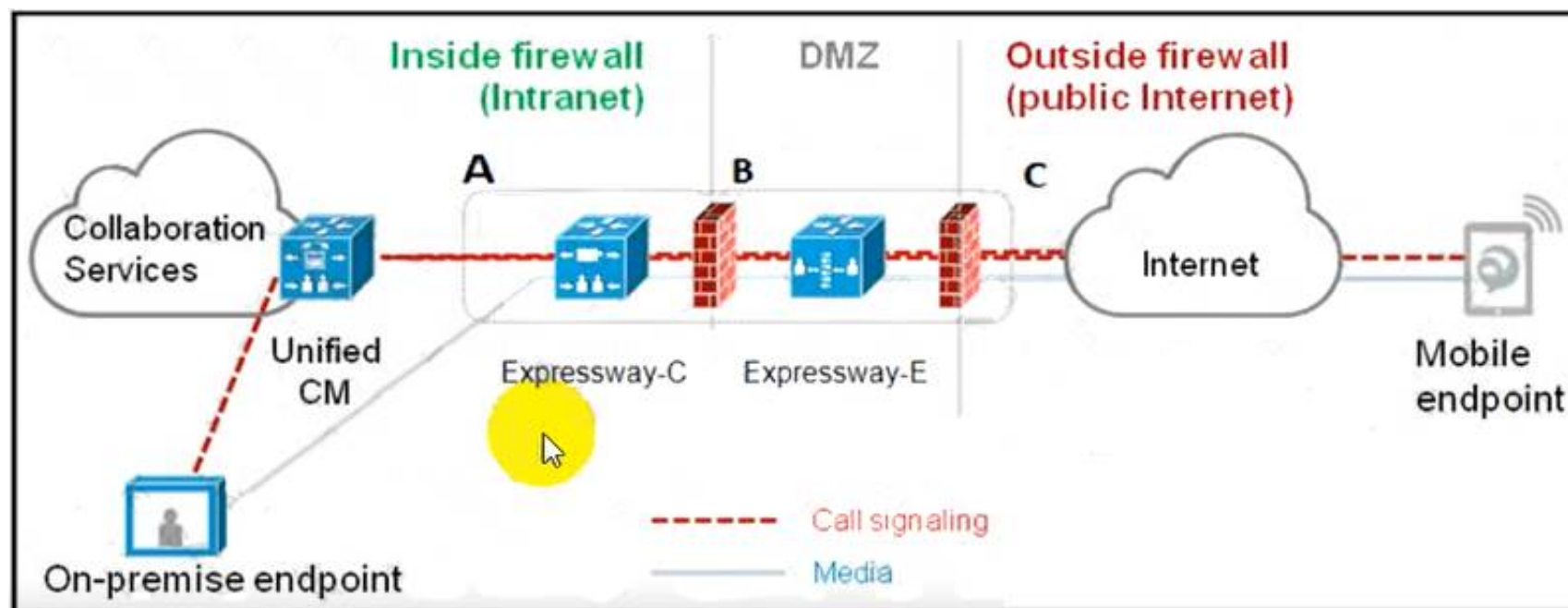
An engineer must configure a SIP route pattern using domain routing with destination +13135551212. The domain ciscocm1.jupiter.com resolves to 192.168.1.3. How must the IPV4 Pattern be configured?

- 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 104

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 108

An administrator would like to set several Cisco Jabber configuration parameters to only apply to mobile clients (iOS and Android). How does the administrator accomplish this with Cisco Jabber 12.9 and Cisco UCM 12.5?

- A. Assign the desired configuration file to "Mobile" Jabber Client Configuration in the Service Profile.
- B. Upload the jabber-config.enc file to TFTP
- C. Create a user profile in Jabber Policies.
- D. Deploy jabber-config-user.xml on iOS and Android devices.

Answer: A

NEW QUESTION 109

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

- A. SNMPv1
- B. SNMPv3
- C. SNMPv2
- D. SNMPv2c

Answer: B

NEW QUESTION 110

Which command must be defined before an administrator changes the linecode value on an ISDN T1 PRI in slot 0/2 on an IOS-XE gateway?

- A. isdn incoming-voice voice
- B. pri-group timeslots 1-24
- C. card type t1 0 2
- D. voice-port 0/2/0:23

Answer: C

NEW QUESTION 111

An engineer configures local route group names to simplify a dial plan. Where does the engineer set the route groups according to the local route group names that are configured?

- A. route list
- B. device pool
- C. CSS
- D. route pattern

Answer: B

NEW QUESTION 113

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 115

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 118

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 119

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 120

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 125

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

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

Answer: A

NEW QUESTION 127

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 128

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 131

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 134

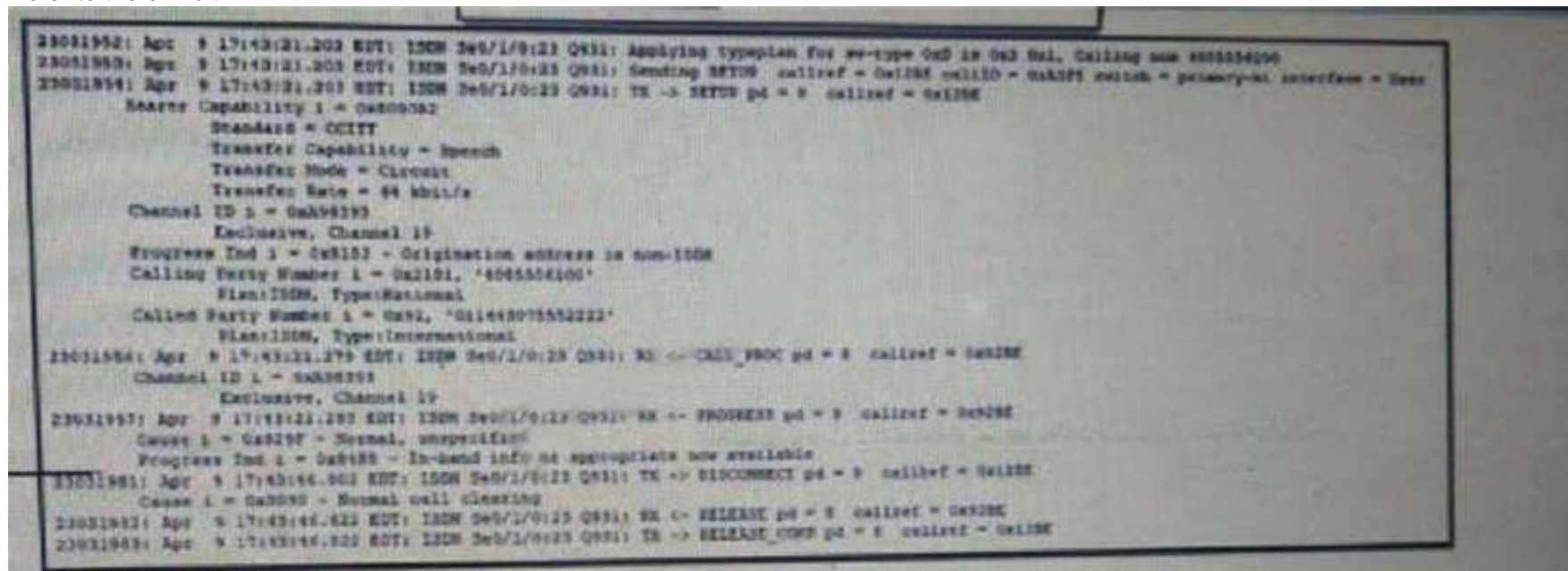
What describes the outcome when the trust boundary is defined at the Cisco IP phone?

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

Answer: B

NEW QUESTION 135

Refer to the exhibit.



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

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

Answer: B

NEW QUESTION 139

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 142

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