Question: 1

Refer to the exhibit.

IfMMr==ljAltUO

Time	Source	Destination
18.683437	10.117.34.222	10.0.101.10
18.938881	10.117.34.222	10.0.101.10
21.686680	10.117.34.222	10.0.101.10
21.941993	10.117.34.222	10.0.101.10
27.687008	10.117.34.222	10.0.101.10
27.942784	10.117.34.222	10.0.101.10

Info

50310 -. 5060 [SYN] Seq-0 Win=64240 Len=0 M$S=1460 WS=256 SACK 50314 -> 5060 [SYN] Seq=0 Win=64240 Len=0 HSS=1460 WS-256 SACK [TCP Retransmission] 50310 + 5060 [SYN] Seq-0 Win-64240 Len=€ [TCP Retransmission] 50314 ■» 5060 [SYN] Seq-0 Win-64240 Len-t [TCP Retransmission] 50310 ■» 5060 [SYN] Seq-0 Win-64240 Len-C [TCP Retransmission] 50314 -. 5060 [SYN] Seq=0 Win-64240 Len=(

Refer to the exhibit. 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 UCM. The SIP phone and Cisco UCM must be set with identical NTP sources.

B. The certificates on the SIP phone are not trusted by the Cisco UCM. The SIP phone must generate new certificates.

C. DNS lookup for the Cisco UCM FQDN is failing. The SIP phone must be reconfigured with the proper DNS server.

D. An network device is blocking TCP port 5060 from the SIP phone to the Cisco UCM. This device must be reconfigured to allow traffic from the IP phone.

Answer: D

Explanation:

Question: 2

Refer to exhibit.

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

Refer to the exhibit. 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

Explanation:

Question: 3

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 1+1917.XXXXXXX, strip pre-dot, numbering type: subscriber VdJ, 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 numbers I voice translation-rule 1 rule 1 /*1917/« rule2/*[*]1917/// !
voice translation-profile strip+1 translate called 1 !

C.

Configure the gateway to translate the calltng number and apply it to the dial peer Combine it with a translation profile for called numbers

voice translation-rule 1
rule 1 /^1917///
rule 2 /*[*]1917/// t
voice translation-profile strip* 1
translate calling 1 t

D.

Configure two called party transformation patterns:
U1917.XXXXXXX. strip pre-dot, numbering type: subscriber
1+1.1, strip pre-dot, numbering type: national

E.

configure two calling party transformation patterns:
U1917.CCCCCC, strip pre-dot, numbering type: subscriber UI, strip pre-dot, numbering type: national

Answer: B, C

Explanation:

Question: 4

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-conlroller)#pri-group timeslots 1-7

B.

config t
2900(config)#isdn switch-type primary-ni
2900(COnfig)#controller e1 0/0/0
2900(conf ig-control ler >#pri-ti m es lots 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

Explanation:

Question: 5

A customer reports that the Cisco UCM toll-fraud prevention does not work correctly, and the customer is receiving charges for unapproved international calls as a result. Which two configuration changes resolve the issues? (Choose two.)

A. Mark patterns as off-net or on-net.

B. Modify the Block OffNet to OffNet Transfer service parameter.

C. Disable call forwarding on the phone.

D. Use Cisco Unified Border Element to debug the calls.

E. Make the calls route through a firewall.

Answer: A, B

Explanation:

Question: 6

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
sFrtpmap® PCM uwo
a=ripmap:1£ G729/8000
a=ptime?20

a=rtpmapJ01 telepho ne-event/8000
a=fmtp;101 CM 5

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

Explanation:

Question: 7

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

A. Attempt Forward routing rule

B. Mobile User

C. Alternate Extensions

D. Alternate Names

Answer: C

Explanation:

Question: 8

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

Explanation:

Question: 9

Which configuration concept allows for high-availability on IM and Presence services in a UC environment?

A. IM and Presence subclusters (configured on Cisco UCM)

B. Presence Redundancy Groups (configured on Cisco Unified IM and Presence)

C. IM and Presence subclusters (configured on Cisco Unified IM and Presence)

D. Presence Redundancy Groups (configured on Cisco UCM)

Answer: D

Explanation:

Question: 10

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: C, D

Explanation:

Question: 11

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

Explanation:

Question: 12

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

Explanation:

Question: 13

Refer to exhibit.

```
Gatewaylfshow seep SCCP Admin State: UP Gateway Local Interface: Loopback0 IPv4
Address: 192.168.12.1 Port Number: 2000


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


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

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

A. Enter "no seep" on the gateway in configuration mode.

B. Enter "ccm-manager mgcp" on the gateway in configuration mode.

C. Enter "mgcp" on the gateway in configuration mode.

D. Enter "ccm-manager config" on the gateway in configuration mode.

E. Delete and re-add the gateway configuration in Cisco UCM.

Answer: B, C

Explanation:

Question: 14

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

```
Hauter^ configure terminal
Rouier(con(ig)# interface gigsbilethemet 5/1
Rdutej tQTfig-if)# description Cube E41.228*009 7
```

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

Answer: B

Explanation:

Question: 15

The IP phones al a customer site do not pick an IP address from the DHCP An engineer must temporarily disable LLDP on all ports of the switch to leave only CDP. Which two commands accomplish this task? (Choose two.)

A. Switch# copy running-config startup-config

B. Switch(config)# no lldp run

C. Switch# configure terminal

D. Switch(conlicj)# interface GigabitEthernet1/0/1

E. Switch(config># no lldp transmit

Answer: B, C

Explanation:

Question: 16

Which location must be assigned to the SIP trunk to replicate enhanced location information via a SIP trunk?

A. phantom

B. replica

C. hub_none

D. shadow

Answer: D

Explanation:

Question: 17

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

Explanation:

Question: 18

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 UCM. which sends a 180 Ringing signal to the destination phone. The same process applies to 11 BYE messages.

B. The source phone must signal to the destination phone that it is ringing, and the destination phone signals back with a 180 Ringing message. The same process applies to 11 BYE messages.

C. The calls have an MTP in the call path due to different codec support. The calls are subsequently split into two call legs.

D. 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

Explanation:

Question: 19

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

Explanation:

Question: 20

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: B, E

Explanation:

Question: 21

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: C, E

Explanation:

Question: 22

An engineer configures Cisco UCM to prevent toll fraud. At which two points does the engineer block the pattern in Cisco UCM to complete this task? (Choose two.)

A. partition

B. route partem

C. translation pattern

D. CSS

E. route group

Answer: A, D

Explanation:

Question: 23

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: A, C

Explanation:

Question: 24

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

Explanation:

Question: 25

A user forwards a corporate number to an international number. What are two methods to prevent this forwarded call? (Choose two.)

A. Configure a Forced Authorization Code on the international route pattern.

B. Block international dial patterns in the SIP trunk CSS.

C. Set Call Forward All CSS to restrict international dial patterns.

D. Set the Call Classification to OnNet for the international route pattern.

E. Check Route Next Hop By Calling Party Number on the international route pattern.

Answer: A, C

Explanation:

Question: 26

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

Explanation:

Question: 27

Refer to exhibit.

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

Refer to the exhibit. 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

Explanation:

Question: 28

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

Explanation:

Question: 29

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

A. manipulation of dialed destination

B. path selection

C. calling privileges

D. endpoint addressing

Answer: C

Explanation:

Question: 30

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 Transfer. Default Outdial. and Default System Transfer.

B. Implement password complexity on voicemail boxes to prevent accounts from being compromised.

C. Create a custom restriction table ??????????? and block it.

D. Create a custom restriction table **********and block it.

Answer: A

Explanation:

Question: 31

Which Cisco UCM configuration is required for SIP MWI integrations?

A. Enable "Accept presence subscription" on the SIP Trunk Security Profile.

B. Select "Redirecting Diversion Header Delivery - Outbound" on the SIP trunk.

C. Enable "Accept unsolicited notification" on the SIP Trunk Security Profile.

D. Select "Redirecting Diversion Header Delivery - Inbound" on the SIP trunk.

Answer: C

Explanation:

Question: 32

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: B, D

Explanation:

Question: 33

An administrator installed a Cisco Unified IP 8831 Conference Phone that is failing to register. Which two actions should be taken to troubleshoot the problem? (Choose two.)

A. Verify that the switch port of the phone is enabled.

B. Verify that the RJ-11 cable is plugged into the PC port.

C. Disable HSRP on the access layer switch.

D. Check the RJ-65 cable.

E. Verify that the phone's network can access the option 150 server.

Answer: A, E

Explanation:

Question: 34

Refer to 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
```

Refer to the exhibit. A customer submits this debug output, captured on a Cisco IOS router. Assuming that an MGCP gateway is configured with an ISDN BRI interface, which BRI changes resolve the issue?

A.

interface BRI0/1/0 no ip address isdn switch-type basic-net3 isdn point-to-multipoint-setup isdn incoming-voice voice isdn send-alerting isdn static-tei 0

B.

interface BRIO/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 BRIO/1/1 no ip address isdn switch-type basic-net3 isdn point-to-point-setup isdn incoming-voice voice isdn send-alerting isdn static-tei 0

D.

interface BRIO/1/1 no ip address isdn switch-type basic-net3 isdn incoming-voice v^ice isdn send-alerting isdn static-tei 0

Answer: C

Explanation:

Question: 35

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

Explanation:

Question: 36

Refer to exhibit.

```
Sent:
INVITE sip:2004HM. 168.100.100:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.100.200:5060;branch=z9hG4bKFlFED
From: "7000" <sip:70008192.168.100.200>;tag-43CDE-lA22
To: <sip:200«192.168.100.100>
Call-ID: 26BCA00-4C4E11EA-80169514-B1C461268192.168.100.200
Supported: lOOrel,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
User-Agent: Cisco-SI?Gateway/IOS-l6.9.5
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, FRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER cseq: 101
INVITE
Contact: <sip:70008192.168.100.200:5060>
Expires: 180
Max-Forwards: 68
P-Assexted-Identity: "7000" <sip:70009192.168.100.200>
Session-Expires: 1800
Content-Type: application/sdp
Content-Length: 254

v=0
o=CiscoSystemsSIP-GK-UserAgent 5871 9974 IN IF4 192.168.100.200
s=SIP Call
c-IN IF4 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
```

Refer to the exhibit. 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 only. Cisco Unity Connection must be reconfigured to support iLBC.

B. Cisco Unified Border Element is not sending any support for DTMF. OTMF configuration must be added to the appropriate dial peer.

C. Cisco Unified Border Element is sending the incorrect media IP address. The IP address of the loopback interface must be advertised in the SDP

D. 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

Explanation:

Question: 37

An end user at a remote site is trying to initiate an Ad Hoc conference call to an end user at the main site. The conference bridge is configured to support G.711. The remote user's phone only supports G.729. The remote end user receives an error message on the phone: "Cannot Complete Conference Call." What is the cause of the issue?

A. The remote phone does not have the conference feature assigned.

B. A software conference bridge is not assigned.

C. A Media Termination Point is missing.

D. The transcoder resource is missing.

Answer: D

Explanation:

Question: 38

An engineer configures a Cisco Unified Border Element and must ensure that the codecs negotiated meet the ITSP requirements. The ITSP supports G.7llulaw and G.729 for audio and H.264 for video. The preferred voice codec is G.711. Which configuration meets this requirement?

A.

```
voice class codec 10
 codec preference 1 g711ulaw
 codec preference 2 g729r8

dial-peer voice 101 voip session protocol sipv2 destination el64-pattern-aap 1
 voice-class codec 10
```

B.

```
 volco class codec 10
  codec preference 1 g729r8 codec preference 2 g711ulaw video codec h264

 dial-peer voice 101 voip session protocol sipv2 destination el 64-pattern-map 1
  voice-class codec 10
```

C.

```
voice class codec 10 codec preference 1 g711ulaw codec preference 2 g729r8 video
```

```
 codec h264

dial-peer voice 101 voip session protocol sipv2 destination el64-pattern-nap 1

 voice-class codec 100
```

D.

```
 voice class codec 10 codec preference 1 g711ulaw
       codec preference 2 g729r8
        video codec h264

     dial-peer voice 101 voip session protocol sipv2 destination el64-pattern-
     map 1 voice-class codec 10
```

Answer: D

Explanation:

Question: 39

Refer to 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 $000
aMintp:101 0-15
a^strafFtcclass:convers^tionaLaudio.iniiuersive.aq:adnutted

Refer to the exhibit. 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

Explanation:

Question: 40

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

A. Cisco SIP Proxy

B. Cisco XCP Text Conference Manager

C. Cisco XCP Router

D. Cisco XCP XMPP Federation Manager

Answer: B

Explanation:

Question: 41

Which information is needed to restore the backup of a Cisco UCM publisher successfully?

A. the TFTP server details

B. the application credentials for Cisco UCM

C. the security password for Cisco UCM

D. the FTP server details

Answer: C

Explanation:

Question: 42

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

A. AF41

B. CS3

C. CS4

D. EF

Answer: B

Explanation:

Question: 43

Which value should be changed when each Cisco UCM node does not allow for more than 5000 phones to be registered?

A. Maximum Number of Registered and Unregistered Devices service parameter on each node

B. Minimum Number of Phones service parameter on each node

C. Maximum Number of Registered Devices service parameter on each node

D. Maximum Number of Phones service parameter on the Publisher

Answer: C

Explanation:

Question: 44

An administrator is configuring LDAP tor 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

Explanation:

Question: 45

Refer to exhibit.

DHCP Server Configuration

Save Dolese Copy C^a Add Now

- Status

^^ Add successful

Host Server *	192.168 10.240
Pnmary DNS IPv4 Address	192.168.99. I\ 0
Secondary DNS IPv4 Address	
Primary TFTP Server IPv4 Address(Option 150)	192.168.10.244
Secondary TFTP Server IPv4 Address!Option ISO)	
Bootstrap Server IPv4 Address	
Domain Name	
TFTP Server Namc(Option 66)	
ARP Cache Timeout!sec) *	
IP Address Lease Time!sec)*	0
Renewal(TI) Time!sec)'	o
RebInding(T2) TIme(sec)'	0
TFTP Server Name(Option 66)	
ARP Cache Timeout!sec)'	2
IP Address Lease Time(sec)'	0
Renewal(TI) Time(sec)*	0
Rebinding(T2) TIme{sec)*	0

Save Delete Copy | Add New

Refer to the exhibit. A collaboration engineer configures Cisco UCM to act as a DHCP server. What must be done next to configure the DHCP server?

A. Restart the Cisco DHCP Monitor Service under Cisco Unified Serviceability

B. Add the new DHCP server to the primary DNS server

C. Restart the TFTP service under Cisco Unified Serviceability.

D. Add a DHCP subnet to the DHCP server under Cisco UCM Administration.

Answer: D

Explanation:

Question: 46

Refer to exhibit.

INVITE sip:l@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=381claba7a78002c558eda31-12b8af63 To: <sip:1@10.10.10.219*
Call-ID: 381claba-7a78000d-4ca6894a-41dd3e0f@10.10.10.84 Max-Forwards: 70 CSeq: 101 INVITE
Contact: <sip:1234010.10.10.84:50083;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO Allow-Events: kpml,dialog Content-Type: application/sdp
Content-Length: 658

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

Refer to the exhibit. 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: B, D

Explanation:

Question: 47

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

A. The end user could incorrectly tag their traffic to bypass firewalls.

B. The end user may incorrectly tag their traffic to be prioritized over other network traffic.

C. There is no reason not to include an end user's PC device in a QoS trust boundary.

D. The end user could incorrectly tag their traffic to advertise their PC as a default gateway.

Answer: B

Explanation:

Question: 48

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

Explanation:

Question: 49

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

Explanation:

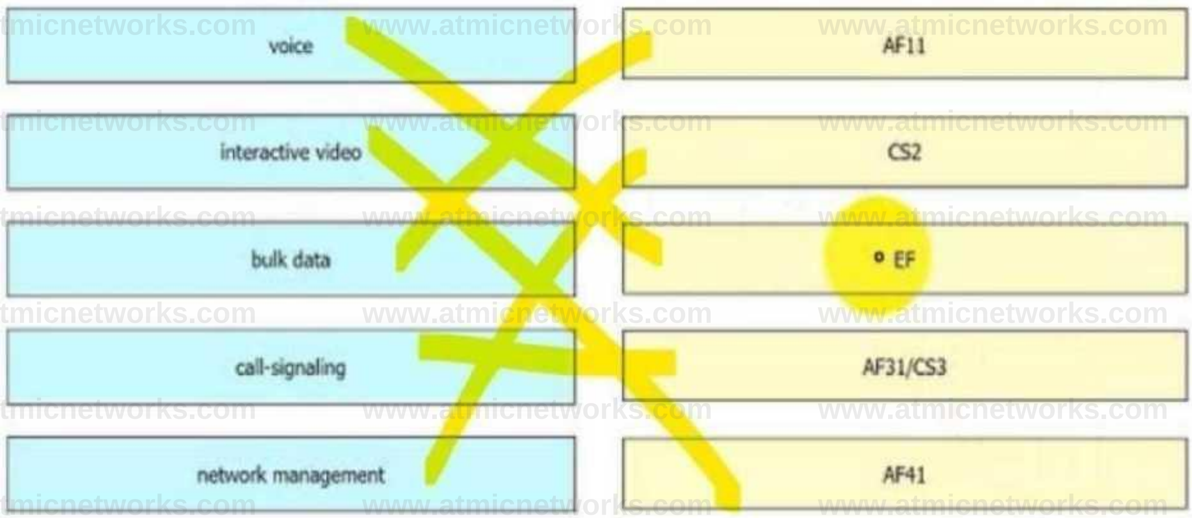
Question: 50

DRAG DROP

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

Answer:

Explanation:



Question: 51

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

Explanation:

Question: 52

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

Explanation:

Question: 53

Which certificate does the Disaster Recovery System in Cisco UCM use to encrypt its communications?

A. Cisco Tomcat

B. CAPF

C. Cisco CallManager

D. IPsec

Answer: D

Explanation:

Question: 54

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-hook. by pressing the digits and entering "#" to process the call. UCM performs a digit-by-digit analysis; off-hook. UCM analyzes all digits as a string.

B. On-hook. no digit analysis is performed; off-hook. UCM requires the '#" to start the digit analysis

C. On-hook. UCM performs a digit-by-digit analysis; off-hook. UCM considers all digits were dialed and does not wait for additional digits.

D. On-hook. UCM considers all digits were dialed and does not wait for additional digits; off-hook. UCM performs a digit-by-digit analysis.

Answer: D

Explanation:

Question: 55

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

Refer to the exhibit. 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

Explanation:

Question: 56

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: A

Explanation:

Question: 57

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

8. ciscocm1.jupiter.com

C. \+13135551212@192.168.1.3

D. 192.168.1.3

Answer: B

Explanation:

Question: 58

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".

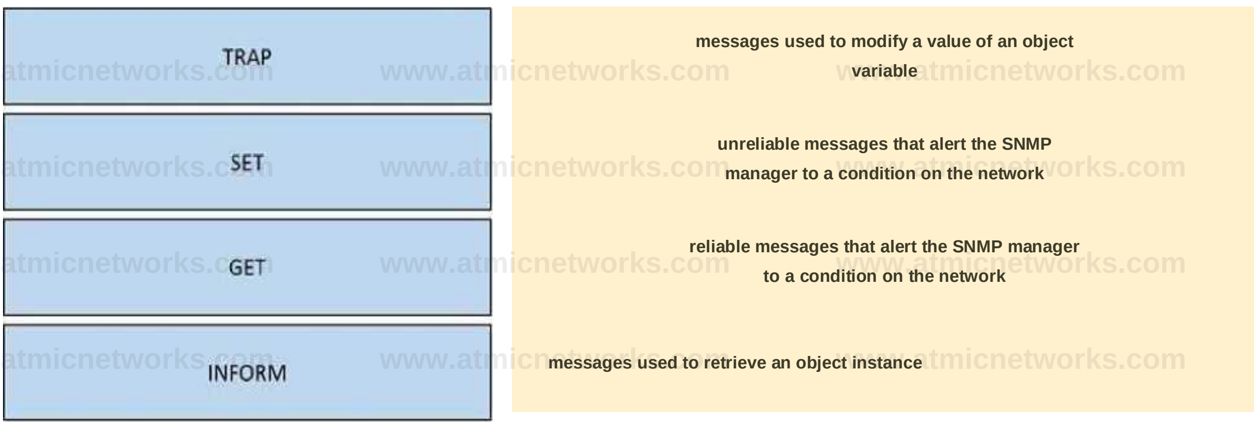
Answer: D

Explanation:

Question: 59

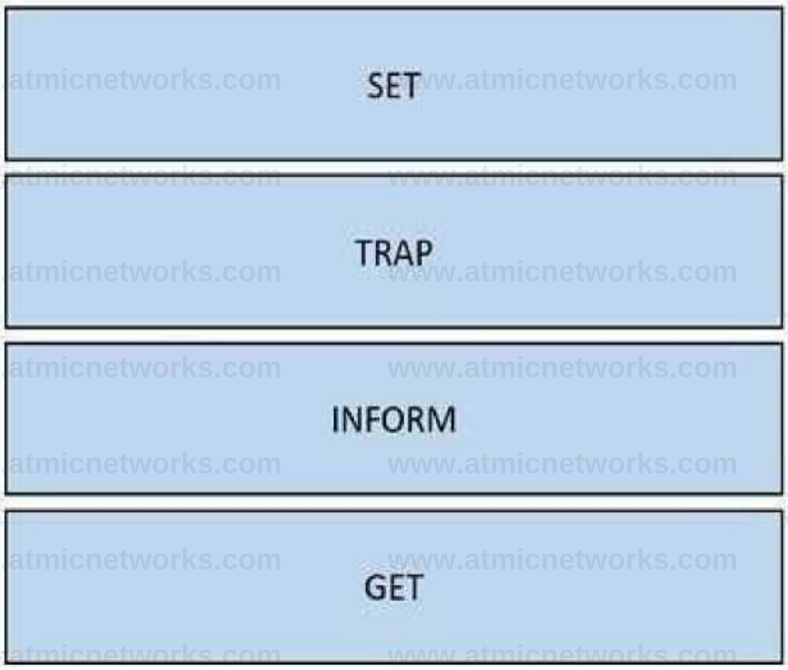
DRAG DROP

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



Answer:

Explanation:



Question: 60

An administrator configures the voicemail feature in a Cisco collaboration deployment. The user mailboxes must be configured when the Cisco Unity Connection server is configured. Which action accomplishes this task?

A. Configure a SIP integration with Cisco UCM to sync users.

B. Configure an SCCP integration with Cisco UCM.

C. Configure an AXL server to access the Cisco UCM users.

D. Configure an active directory to sync the users who will have a voicemail box.

Answer: C

Explanation:

Question: 61

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

Explanation:

Question: 62

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

Explanation:

Question: 63

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

Explanation:

Question: 64

An administrator is trying to change the default LINECODE for a voice ISDN T1 PRI. Which command makes this change?

A. linecode ami

B. linecode b8zs

C. linecode hdb3

D. linecode esf

Answer: A

Explanation:

Question: 65

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

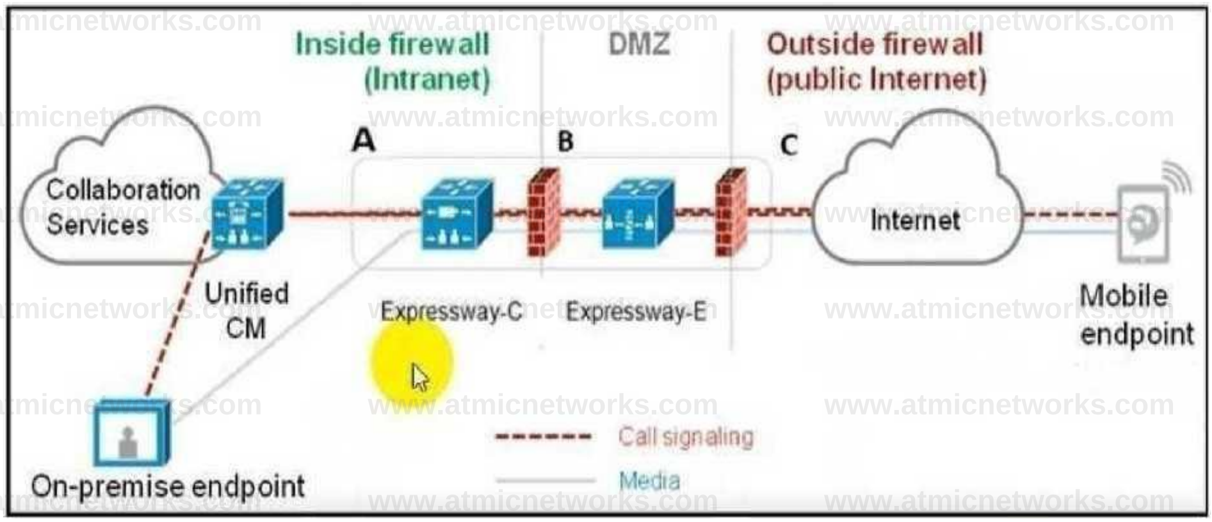
D. Connected Line ID Presentation

Answer: B

Explanation:

Question: 66

Refer to exhibit.



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

Explanation:

Question: 67

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

Explanation:

Question: 68

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: B, E

Explanation:

Question: 69

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: C, D

Explanation:

Question: 70

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

Explanation:

Question: 71

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

Explanation:

Question: 72

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

Explanation:

Question: 73

Refer to exhibit.

```
Voice class codec 1
codec preference 1 g7Haiaw
codec preference 2 g711ulaw
codec preference 3 g729i8

dial-peer voice 13 voip
description incoming dalpeer from ITSP
Incoming called-numPer
voce-dess codec 1

tfias-peer voice 19 voip
description outgoes darppeer io CUCM
destination-pattern T
session protocol Sipv2
session target ipv4 3 3 3 3
voce-class codec 1

Incoming SOP from ITSP

v-0
O-5IP tesr@2 2 2 21 16 IN IP4 2 2 2 2
s=sip.test^2 22 2
C-INIP4 22 2 2
MO
m=audio 5000 RTP/AVP18 0
a=rtpmap 0 PCMUf8000/l
a-rtpmap 18 G729J8O00/1
```

Refer to the exhibit. 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 18G729/8000/1

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 0 818
a=rtpmap:0 PCMU/8000/1
a=rtpmap 8 PCMA/8000/1 a=rtpmap:18 G729/8000/1

Answer: B

Explanation:

Question: 74

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

Explanation:

Question: 75

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

Explanation:

Question: 76

Refer to exhibit.

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

Refer to the exhibit. 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

Explanation:

Question: 77

When designing the capacity for a Cisco UCM 12.x cluster, an engineer must decide which VMware template will be used for each node. What is the lowest number of users supported in a template and the highest number of users in a template?

A. 750 and 15.000 users

B. 750 and 10.000 users

C. 500 and 10.000 users

D. 1000 and 10.000 users

Answer: D

Explanation:

Question: 78

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

Explanation:

Question: 79

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

Explanation:

Question: 80

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

Explanation:

Question: 81

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

A. Set service parameter 'Advanced Ad Hoc Conference" to FALSE.

B. Set service parameter "Drop Ad Hoc Conference" to "When Conference Controller leaves."

C. Set service parameter "Advanced Ad Hoc Conference" to 2.

D. Set service parameter "Drop Ad Hoc Conference" to "Do not allow outside parties."

Answer: B

Explanation:

Question: 82

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

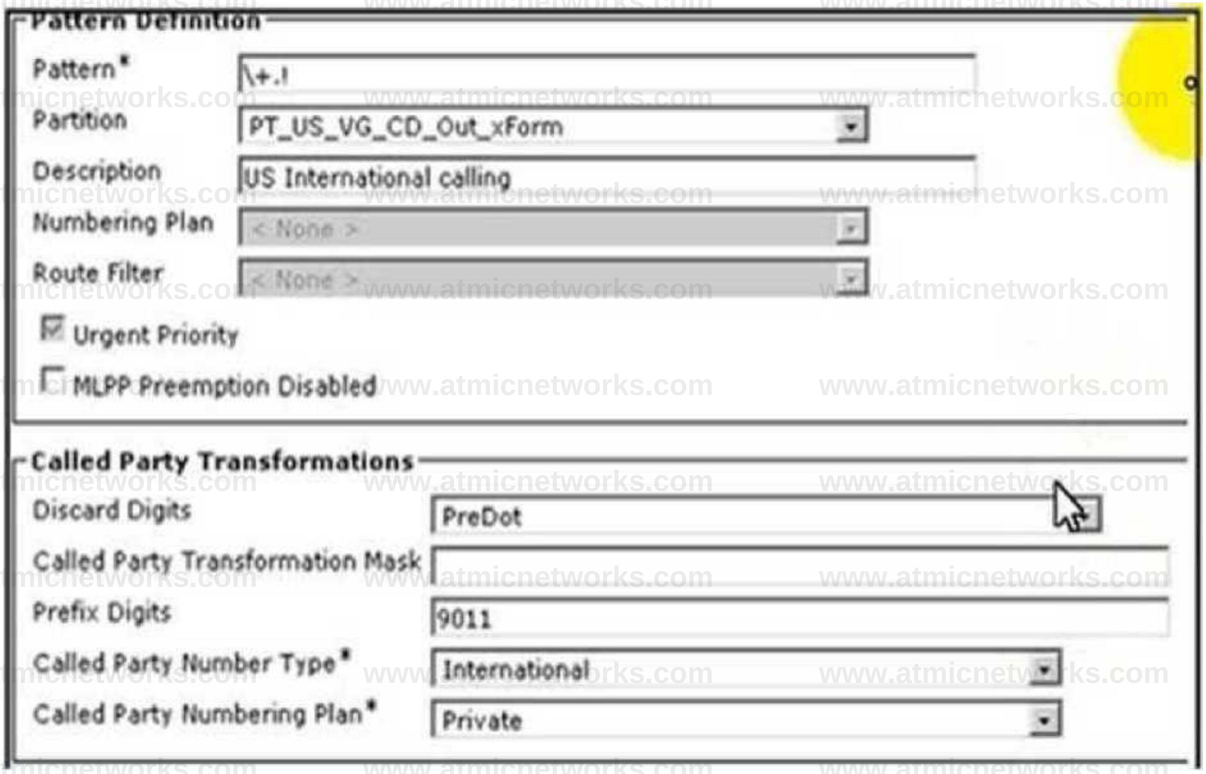
Explanation:

Question: 83

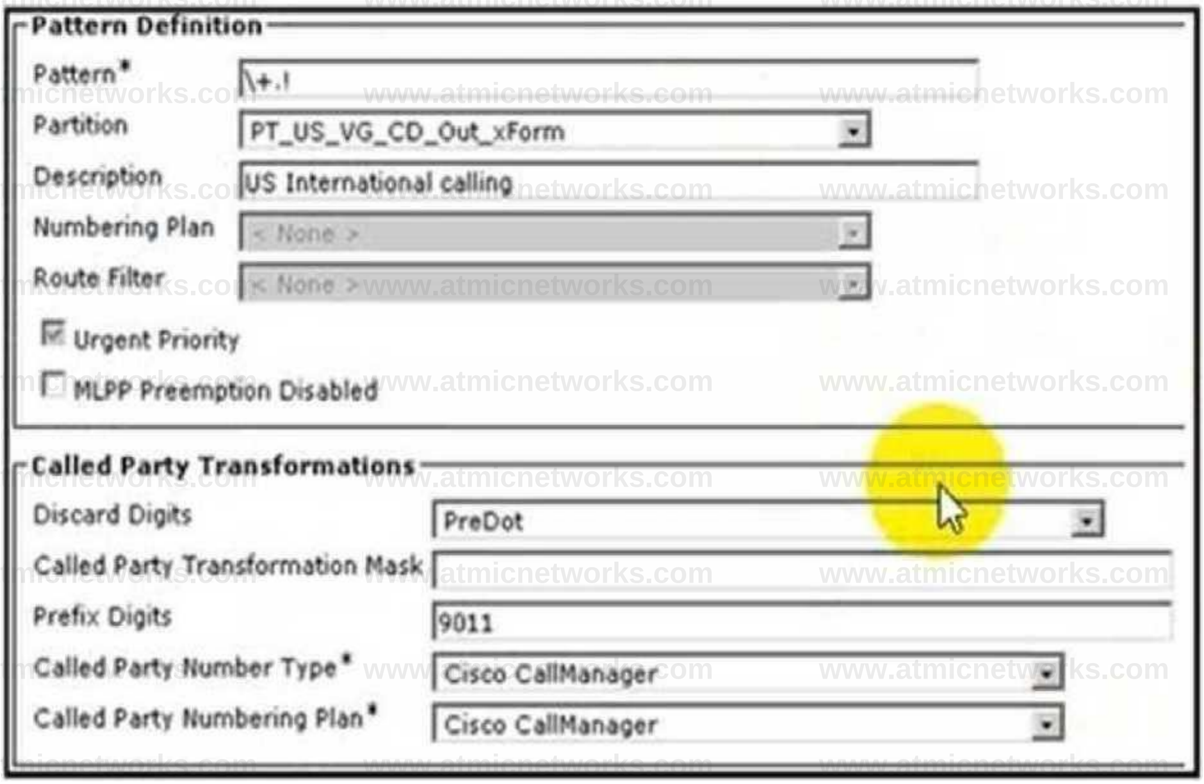
Given this H.323 gateway configuration and using Cisco best practices, how must the called party transformation pattern be configured to ensure that a proper ISDN type of number is set?

```
voice translation-rule 40
 rule 1 /3...S/ /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
```

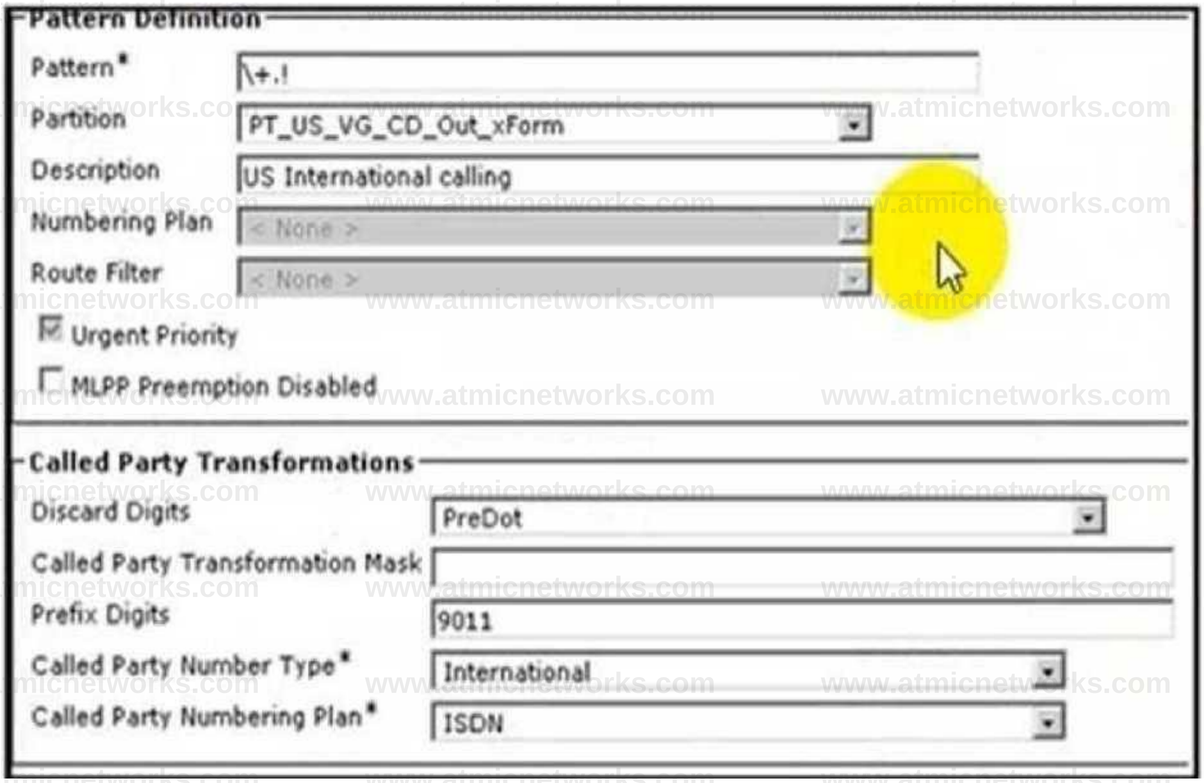
A.


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

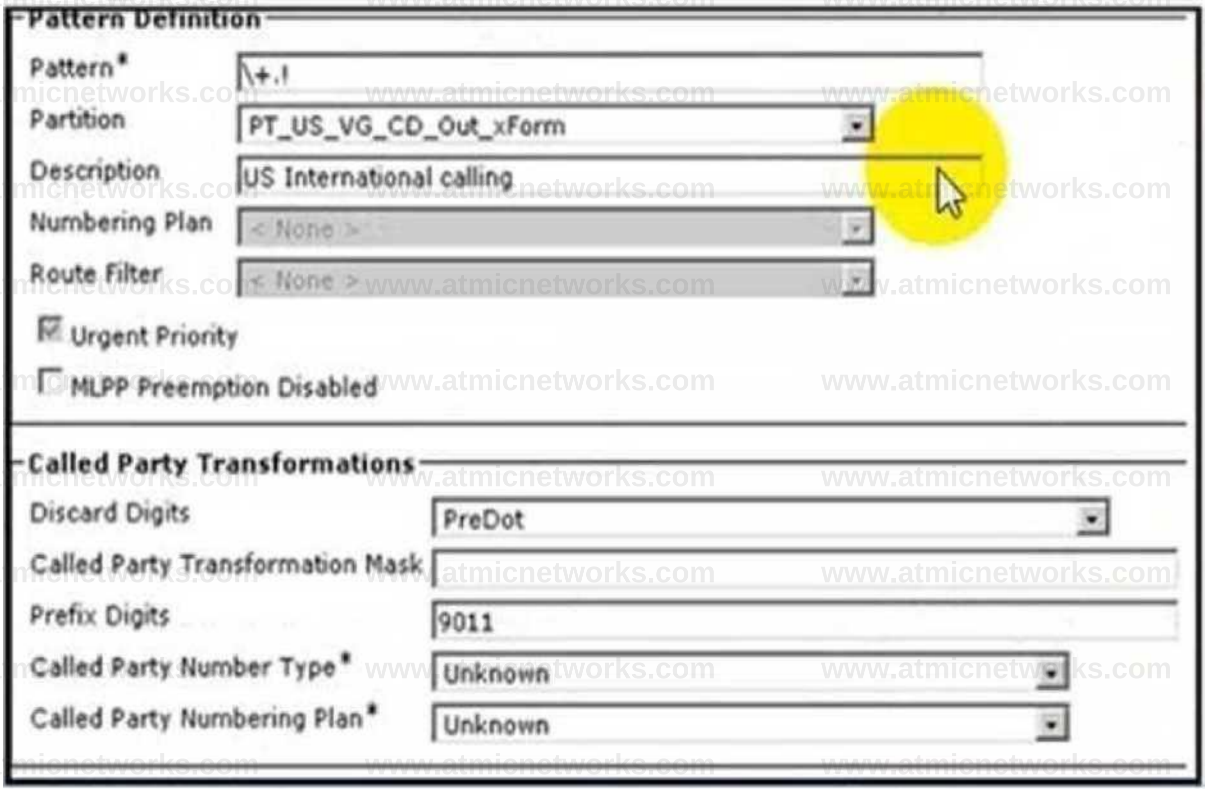
B.


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

Explanation:

Question: 84

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: C, E

Explanation:

Question: 85

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: D, E

Explanation:

Question: 86

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: D, E

Explanation:

Question: 87

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

Explanation:

Question: 88

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.

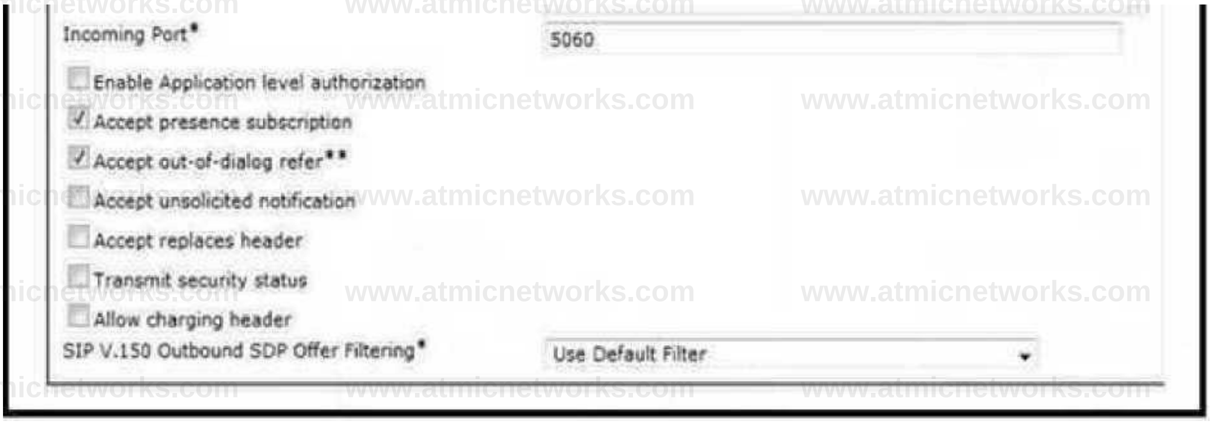
E. The pattern represents the largest number of endpoints.

Answer: A, D

Explanation:

Question: 89

Refer to exhibit.



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: C, E

Explanation:

Question: 90

Refer to exhibit.

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

Refer to the exhibit. 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

Explanation:

Question: 91

During the Cisco IP Phone registration process, the TFTP download (ails. What are two reasons (or this issue? (Choose two.)

A. The DNS server was not specified, which is needed to resolve the DHCP server IP address.

B. Option 100 string was not specified, or an incorrect Option 100 string was specified.

C. The Cisco IP Phone does not know the IP address of the TFTP server.

D. The Cisco IP Phone does not know the IP address of any of the Cisco UCM Subscriber nodes.

E. Option 150 string was not specified, or an incorrect Option 150 string was specified.

Answer: C, E

Explanation:

Question: 92

Refer to exhibit.

000193: Dec 5 14:35:31.054:ISDN SeO/l/O:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:32.054: ISDN Se0/l/0:15 Q9 21: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:33.054: ISDN Se0/l/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:34.054: ISDN Se0/l/0:15 Q921: User TX -> SABMEp sapi=0 tei=0

Refer to the exhibit. 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

Explanation:

Question: 93

What is required for Cisco UCM to accept SIP calls with a URI in the format of 'sip:2001@cucmpub.cisco.com'?

A. Define Cluster Fully Qualified Domain Name under Servers in Cisco UCM.

B. Change the Destination Address to a Fully Qualified Domain Name on the SIP trunk.

C. Define Cluster Fully Qualified Domain Name in Enterprise Parameters.

D. Set the SIPS URI Handling to True in CallManager Service Parameters.

Answer: C

Explanation:

Question: 94

Refer to exhibit.

```
INVITE sip:4000@172.16.1.1:5061 SIP/2.0
Via: SIP/2.0/TLS 172.16.2.143:5061;branch=z9hG4bK8FD315E7
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-anat
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:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
```

Refer to the exhibit. 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

Explanation:

Question: 95

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. calls fail.

Answer: C

Explanation:

Question: 96

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.

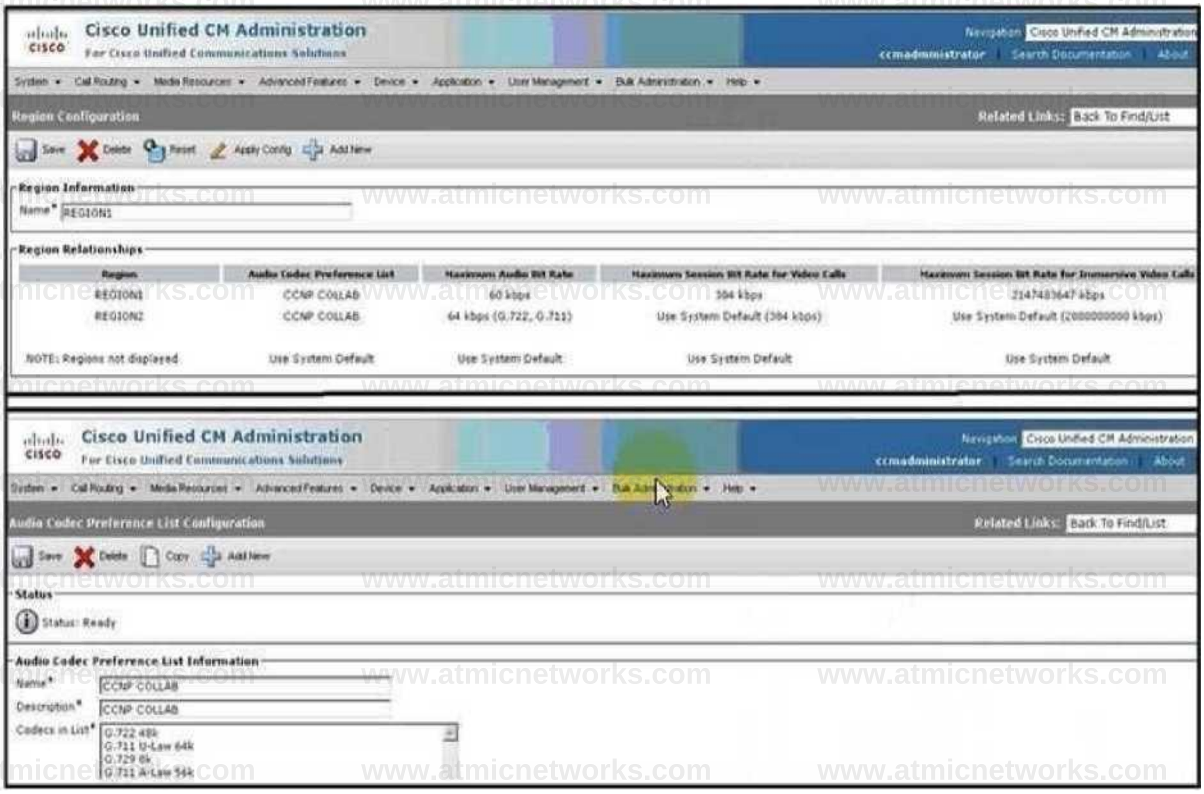
D. Deploy jabber-config-user.xml on iOS and Android devices.

Answer: A

Explanation:

Question: 97

Refer to exhibit.



Refer to the exhibit. An engineer is troubleshooting this video conference issue:

*A video call between a Cisco 9971 in Region1 and another Cisco 9971 in Region1 works.

*As soon as the Cisco 9971 in Region1 conferences in a Cisco 8945 in Region2. the Regio1l endpoint cannot see the Region2 endpoint video.

What is the cause of this issue?

A. Maximum Audio Bit Rate must be increased.

B. Maximum Session Bit Rate for Immersive Video Calls is too low.

C. Maximum Session Bit Rate for Video Calls is too low.

D. Cisco 8945 does not have a camera connected.

Answer: C

Explanation:

Question: 98

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

Explanation:

Question: 99

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

Explanation:

Question: 100

Refer to exhibit.

```
000142: 'Apr 22 19:41:49.050: MGCP packet received trots 192.160.100.100:2427—> AVE? 4 AA1A730/3U0/08VG320.cisco, local
MGCP 0.1 F: X, A, X

000143: 'Apr 23 19:41:49.050: MGCP Packet sent to 192.163.100.101:2427—> 200 4 1: X: 2
1:    p:10-20, a:KMU:PCHA:G.nXc4, b:64, e:on, yc:1, s:on, t:10, ::?, nt:I», v:T;G:D:L.'H.-R:ATM:SST: PRE
1:    p:10-220, a:G.729;G.729a;G.729b, b:6, e:on. oc:1, s:on. t:10, r:g, nt:ix, v:T:G:b;t,H;R:ATM,SS7,PRE
2:    p:10-110, a:G.726-16;G.728, b:le, e:on, nc:1, s:on, t:10, r:g, nt:Ui, V:T;G;D:L:H;R;ATK:S3T;PRE
3:    p:10-70, a:G.726-24, b:24, e:on, ?c:l,        s:on,   t:10,    r:g, nt:IK,    v:T;G;O;L;H,R;AiM;SSX,PR£
L: p:10-SO, a:G.726-32, b:32. e:on, gc:1.           s:on,   t:10,    r:g. nt:IN,    v:-;G:D;I.:H;F:ATM;33T;FRE
4:    p:30-270, a:G.723.1-H:G.723:G.723.1a-H. b:6,      e:on,    gc:1, s:on,    t:10, r:g, nt:IK, v;T:G:D:L:H:R;ATM;3ST:PRE
5:    p:30-330, a:G.723.1-t;G.723.la-L, b:5,        e:on,  gc:1,    s:on, t:10,    r:?, nt:IN, v:T;G;D.i.,H:R;AXKS5I,PRE
M: sendonly, recvonly, sendrecv, inactive, loopback, contrast, data, netwloop. netwtest
```

Refer to the exhibit. What is the registration state of the analog port in this debug output?

A. The analog port failed to register to Cisco UCM with an error code 200.

B. The MGCP Gateway is not communicating with the Cisco UCM.

C. The analog port is currently shut down.

D. The analog port is registered to Cisco UCM.

Answer: D

Explanation:

Question: 101

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

A. Device(config-if)# switchport voice vlan 221

B. Device(config-if)# switchport vlan voice 221

C. Device(config-if)# switchport access vlan 221

D. Device(config-if)# switchport trunk allowed vlan 221

Answer: A

Explanation:

Question: 102

An engineer implements QoS in the enterprise network. Which command is used to verity the classification and marking on a Cisco IOS switch?

A. show class-map interface GigabitEthernet 1/0/1

B. show policy-map interface GigabitEthernet 1/0/1

C. show access-lists

D. show policy-map

Answer: B

Explanation:

Question: 103

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

Explanation:

Question: 104

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

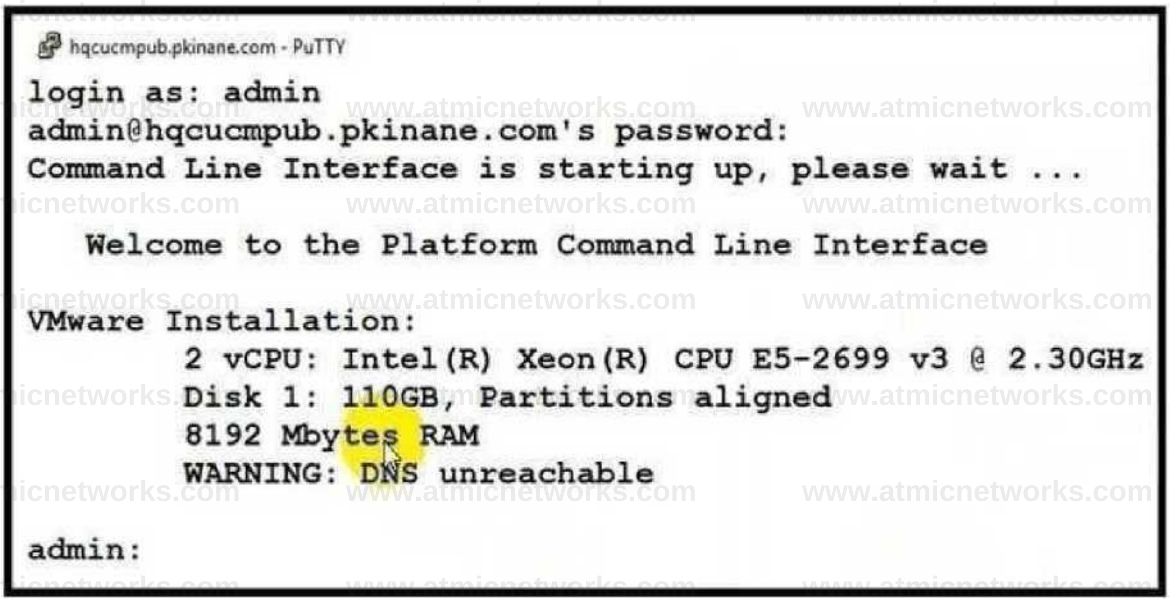
D. gaps in the voice due to packet loss

Answer: A

Explanation:

Question: 105

Refer to exhibit.



Refer to the exhibit. An administrator accesses the terminal of a Cisco UCM and starts a packet capture. Which two commands must the administrator use on Cisco UCM to generate DNS traffic? (Choose two.)

A. utils ntp status

B. show cdp neighbor

C. show version active

D. utils diagnose test

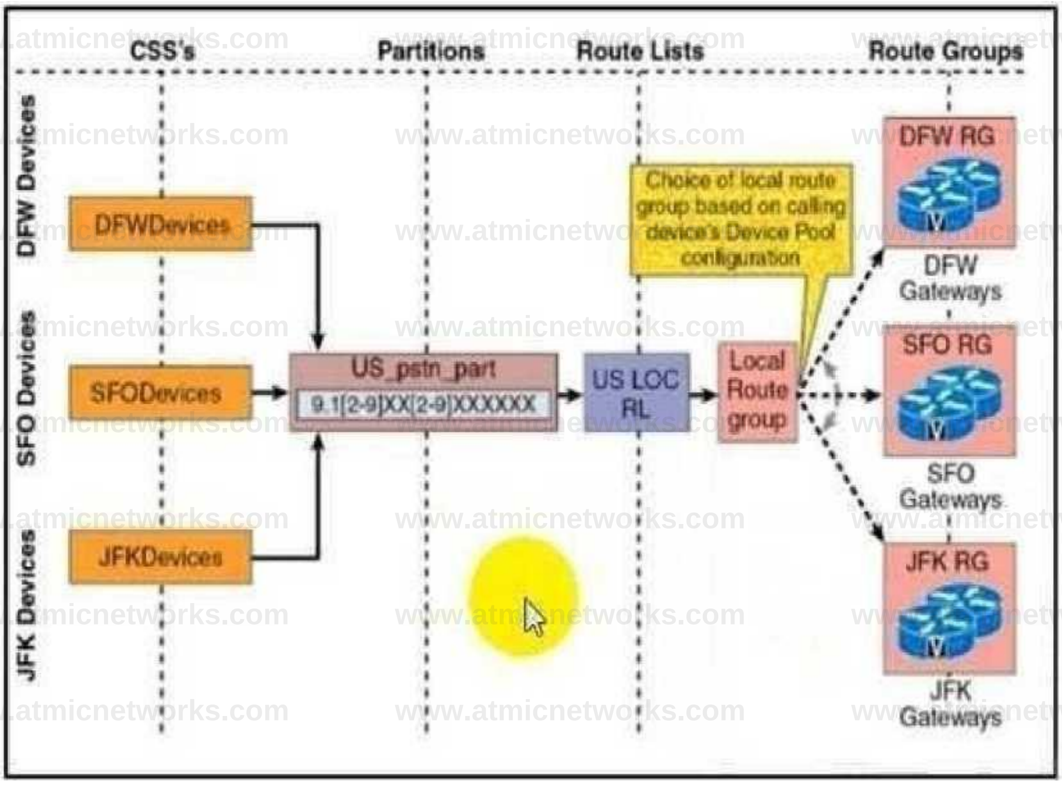
E. autils diagnose module validate Network

Answer: D, E

Explanation:

Question: 106

Refer to exhibit.



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. The engineer must configure which sites the device should be roaming to.

B. 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.

C. The call fails because the Standard Local Route Group is being used only it no configuration is set for the device pools.

D. The call egresses in New York because the device automatically is assigned a New York device pool and uses the local gateway.

Answer: B

Explanation:

Question: 107

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

A. Scavenger

B. best effort

C. voice

D. call signaling

Answer: A

Explanation:

Question: 108

Which Cisco Unified communications manager configuration is required for SIP MWI integration?

A. Select “Redirecting Diversion Header Delivery— Inbound’ on the SIP trunk

B. Enable “Accept presence subscription” on the SIP trunk security profile

C. Select “Redirecting Diversion Header Delivery – outbound” on the SIP trunk

D. Enable “Accept unsolicited notification” on the SIP Trunk security profile

Answer: D

Explanation:

Question: 109

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

Explanation:

Question: 110

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

A. expedited forwarding

B. class selector

C. assured forwarding

D. default

Answer: B

Explanation:

Question: 111

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

Explanation:

Question: 112

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

Explanation:

Question: 113

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

Explanation:

Question: 114

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

A.10

B. 6

C. 2

D. 4

Answer: C

Explanation:

Question: 115

Refer to 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
!
```

Refer to the exhibit. 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

Explanation:

Question: 116

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

A. modifies the Drop Ad Hoc Conference service parameter

B. modifies the Block OffNet to OffNet Transfer service parameter

C. removes the transcoder

D. changes the codecs that are supported on the conference resource

Answer: A

Explanation:

Question: 117

When setting a new primary DNS server in the Cisco UCM CLI what is required for the change to take affect?

A. restart of CallManager service

B. restart of DirSync service

C. restart of the network service

D. restart of TFTP service

Answer: C

Explanation:

Question: 118

Refer to exhibit.

```
admin;utils ntp status
 ntpd (pid 17426) is running..•

      reacts              refid
                                      st t when poll  reach     delay  offset  jitter

*192.165.1.1        17.253.14.125      2 u 36    64   377      0.435   0.039   0.047
  192.169.1.2      -IKIT.            16 u - 64         0      0.000   0.000   0.000
```

Refer to the exhibit. 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

Explanation:

Question: 119

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

Explanation:

Question: 120

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

Explanation:

Question: 121

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: A, B

Explanation:

Question: 122

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

A. Wart until 10 pm on March 4th when the user is automatically removed from Cisco UCM.

B. Wait until 10 pm on March 5th when the user is automatically removed from Cisco UCM.

C. Wait until 3 15 a.m. on March 6th for garbage collection to remove the user from Cisco UCM.

D. Wait until 315am on March 5th for garbage collection to remove the user from Cisco UCM.

Answer: C

Explanation:

Question: 123

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: A, B

Explanation:

Question: 124

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 locationrouter

D. configure the software transcoder on Cisco UCM to support voice calls to and from both remote sites

Answer: C

Explanation:

Question: 125

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: A, B

Explanation:

Question: 126

Which two features of Cisco Prime Collaboration Assurance require advanced licensing? (Choose two.)

A. real lime alarm browse

B. multicluster support

C. call quality monitoring

D. email notifications

E. customizable events

Answer: B, C

Explanation:

Question: 127

In the cisco expressway solution, which two features does mobile and Remote access provide? (Choose two)

A. VPN-based enterprise access for a subset of Cisco Unified IP Phone models

B. secure reverse proxy firewall traversal connectivity

C. the ability 10 register third-party SIP or H 323 devices on Cisco UCM without requiring VPN

D. the ability of Cisco IP Phones to access the enterprise through VPN connection

E. the ability for remote users and their devices to access and consume enterprise collaboration applications and services

Answer: B, E

Explanation:

Question: 128

A company wants to provide remote user with access to its premises Cisco collaboration features. Which components are required to enable cisco mobile and remote access for the users?

A. Cisco Unified Border Element, Cisco IM and Presence Server, and Cisco Video Communication Server

B. Cisco Expressway-E Cisco Expressway-C and Cisco UCM

C. Cisco Expressway-E, Cisco IM and Presence Server, and Cisco Video Communication Server

D. Cisco Unified Border Element. Cisco UCM, and Cisco Video Communication Server

Answer: B

Explanation:

Question: 129

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: C, E

Explanation:

Question: 130

User's phone has:

Line level CSS which contains Partition_A
Device level CSS which contains Partition_B

Configured Route Patterns:

2XXX in Partition_A set to block call
21XX in Partition_B set to route call

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

A. Change line partition to Partition_A

B. Change line CSS to only contain Partition_B

C. Set the user's line CSS to <None>

D. Set the user's device CSS to <None>

Answer: D

Explanation:

Question: 131

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

Explanation:

Question: 132

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: A, C

Explanation:

Question: 133

According to QoS guidelines, what is the packet loss for streaming video?

A. Not more than 8%

B. Not more than 1%

C. Not more than 3%

D. Not more than 5%

Answer: B

Explanation:

Question: 134

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

Explanation:

Question: 135

Which IP Precedence value is used to classify a call signalling packet?

A. 6

B. 5

C. 4

D. 3

Answer: D

Explanation:

Question: 136

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

Explanation:

Question: 137

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: C, D

Explanation:

Question: 138

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

A. G722.1

B. iLBC

C. G.711alaw

D. G.729A

Answer: B

Explanation:

Question: 139

In which location does an administrator look to determine which subscriber the phone registers to if loses registration with the current Cisco UCM subscriber?

A. On Cisco UCM Administration Page Device > Phone > Phone Configuration page

B. On Cisco UCM Administrator Page server > Cisco UCM

C. On Cisco UCM Administrator page system > Device Pool > Cisco UCM group

D. On Cisco UCM Administrator page system > Enterprise Parameters

Answer: C

Explanation:

Question: 140

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

A. The call continues to be active, but features like transfer or hold do not work.

B. The call continues and all features work.

C. The call drops immediately.

D. The call drops after missing two keepalives from Cisco UCM.

Answer: D

Explanation:

Question: 141

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

A. EF

B. CS3

C. AF41

D. AF31

Answer: A

Explanation:

Question: 142

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.

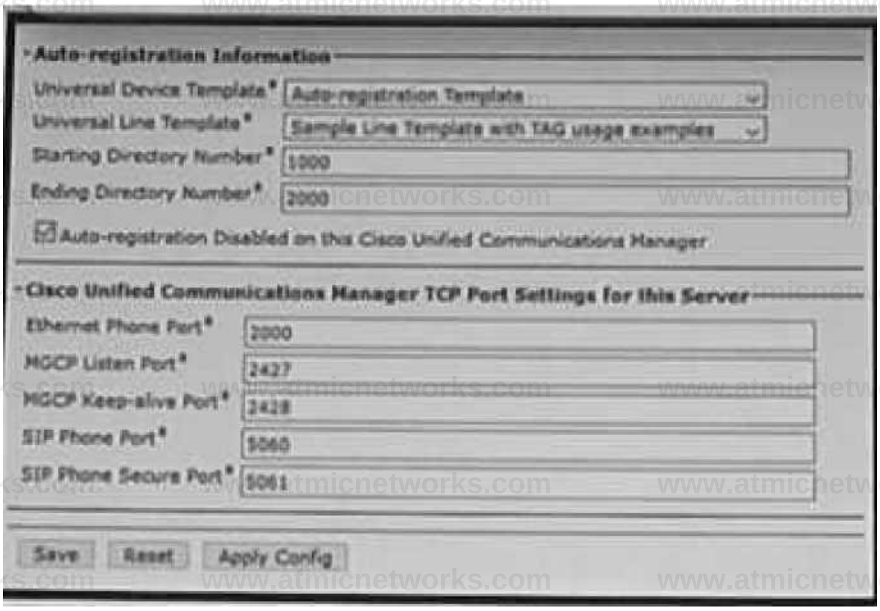
D. Video conferencing is classified as CoS 1, and voice is CoS 2.

Answer: B

Explanation:

Question: 143

Refer to exhibit.



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

Explanation:

Question: 144

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: B, D

Explanation:

Question: 145

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: D, E

Explanation:

Question: 146

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: C

Explanation:

Question: 147

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: A, E

Explanation:

Question: 148

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

A. Increase the number of voice ports.

B. If it's a Unity Connection Cluster, ensure that replication is fine and not in split-brain mode.

C. The phone system to which the phones are registered in Unity Connection has the Default Trap Switch check box enabled.

D. Add dedicated dial-out ports with the allow trap connections setting selected.

E. Ensure that you have set up SIP Digest Authentication on the SIP trunk security profile.

Answer: A, C

Explanation:

Question: 149

Refer to exhibit.

controller Ll 0/0/1

pri-group timeslots Ui dock source line

'inecode bBzs framing esf

Refer to the exhibit. 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 el 0/0/1 pn-group timeslots 1-11.12 clock source tine linecode hdb3

framing crc4

C.

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

D.

controller e1 0/0/1
prigroup timeslots 1*12 clock source line
linecode crc4
framing hd3

Answer: C

Explanation:

Question: 150

A company wants to provide remote users with access to its on-premises Cisco collaboration features. Which components are required to enable Cisco Mobile and Remote Access for the users?

A. Cisco Expressway-E, Cisco IM and Presence Server, and Cisco Video Communication Server

B. Cisco Unified Border Element, Cisco IM and Presence Server and Cisco Video Communication Server

C. Cisco Expressway-E, Cisco Expressway-C, and Cisco UCM

D. Cisco Unified Border Element, Cisco UCM, and Cisco Video Communication Server

Answer: C

Explanation:

Question: 151

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: D

Explanation:

Question: 152

What are two Cisco UCM location bandwidths that are deducted when G 729 and G.711 codecs are used? (Choose two.)

A. If a calluses G.729. Cisco UCM subtracts 16k.

B. If a calluses G.711, Cisco UCM subtracts 64k

C. If a calluses G.711, Cisco UCM subtracts 80k

D. If a calluses G.729. Cisco UCM subtracts 24k.

E. If a calluses G.729. Cisco UCM subtracts 40k

Answer: C, D

Explanation:

Question: 153

Refer to exhibit.

```
Beater Capability I = 0x8090A2 Standard * CCITT Transfer Capability ■ Speech
Transfer Mode = Circuit Transfer Rate = 64 kbit/s Channel ID i = 0xA98388
        Exclusive, Channel 8
Calling Party Number I = 0x2181, '5125551212' Plan: ISDN, Type:National
Called Party Number I» OxAl, '2145551212
        Plan:ISDN, Type National
Mar 102:35:37: ISDN S«O/1/1:23 Q931 RX <• CALl.PROC pd > 8 callref = 0x809A
Channel ID I = 0xA98388
        Exclusive, Channel 8

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

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

A. isdn supp-service name calling

B. isdn outgoing display-ie

C. isdn enable did

D. isdn send display le

Answer: B

Explanation:

Question: 154

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: D, E

Explanation:

Question: 155

An administrator must make a pattern to route calls to two different destinations john doe@company com and jane doe@company com Which type of patterns are needed in the Cisco UCM. and what must the pattern look like?

A. A SIP route pattern that looks like the *@company.com

B. A SIP route pattern that looks like this company.com

C. A regular route pattern with URI feature enable in the configuration page. The pattern must look like this:(*@company.com)

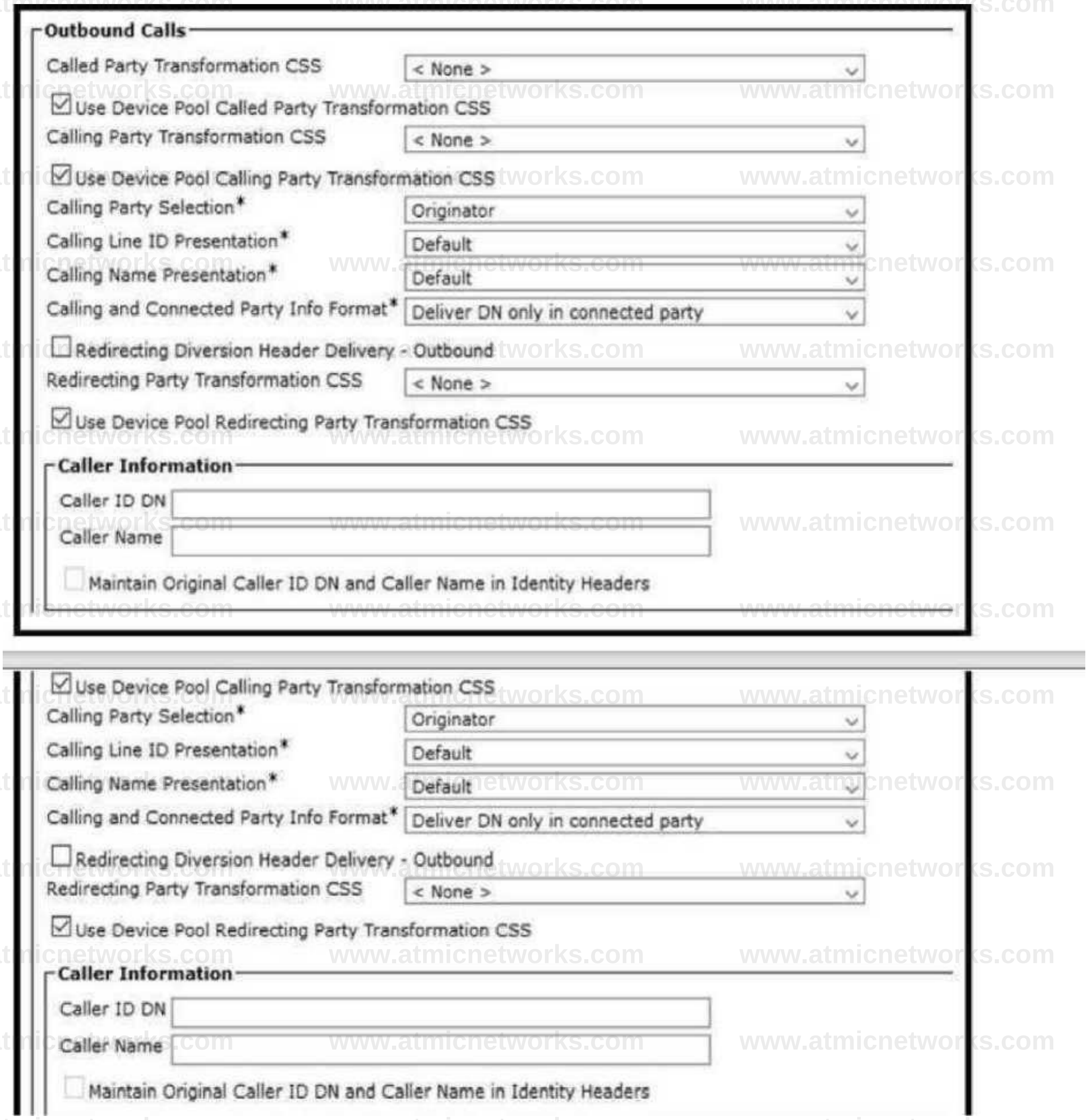
D. A regular route pattern with URI feature enable in the configuration page. The pattern must look like this:MATCH(*@company.com)

Answer: C

Explanation:

Question: 156

Refer to exhibit.



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

A. Reboot Cisco Unity Connection.

B. Check the box "Redirecting Diversion Header Delivery - Outbound", then reset the trunk.

C. Check the box 'Redirecting Diversion Header Delivery - Outbound".

D. Review the conversation manager logs on Cisco Unity Connection.

Answer: B

Explanation:

Question: 157

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

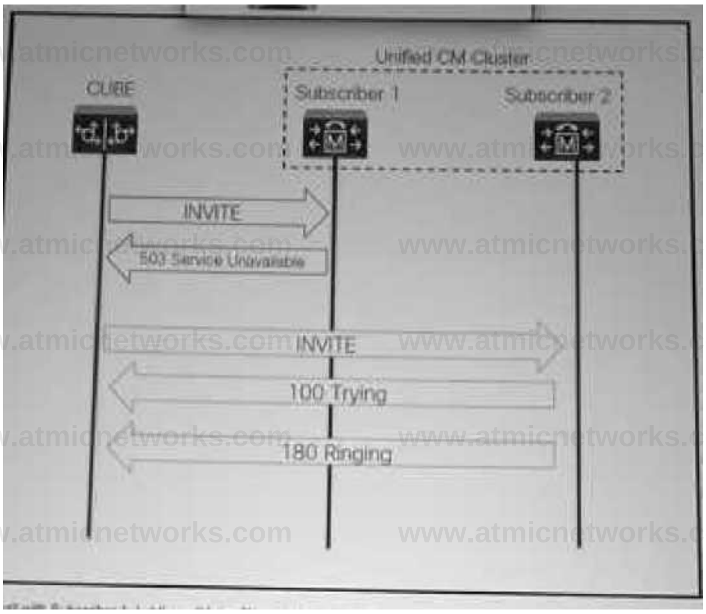
D. xcommand video status

Answer: B

Explanation:

Question: 158

Refer to exhibit.



Refer to the exhibit. Cisco Unified element is attempting to establish a call with Subcribers1, but the call fails. Cisco Unified Border Element then retries the same call with Subcribers2, and the call proceeds normally.

Which action resolves the issue?

A. Verify that the correct calling search space is selected for the inbound Calls section

B. Verify that the run on all active United CM Nodes checkbox is enabled

C. Verify that the Significant Digits field for inbound Calls is set to All.

D. Verify that the PSTN Access checkbox is enabled.

Answer: B

Explanation:

Question: 159

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

A. 66

B. 143

C. 150

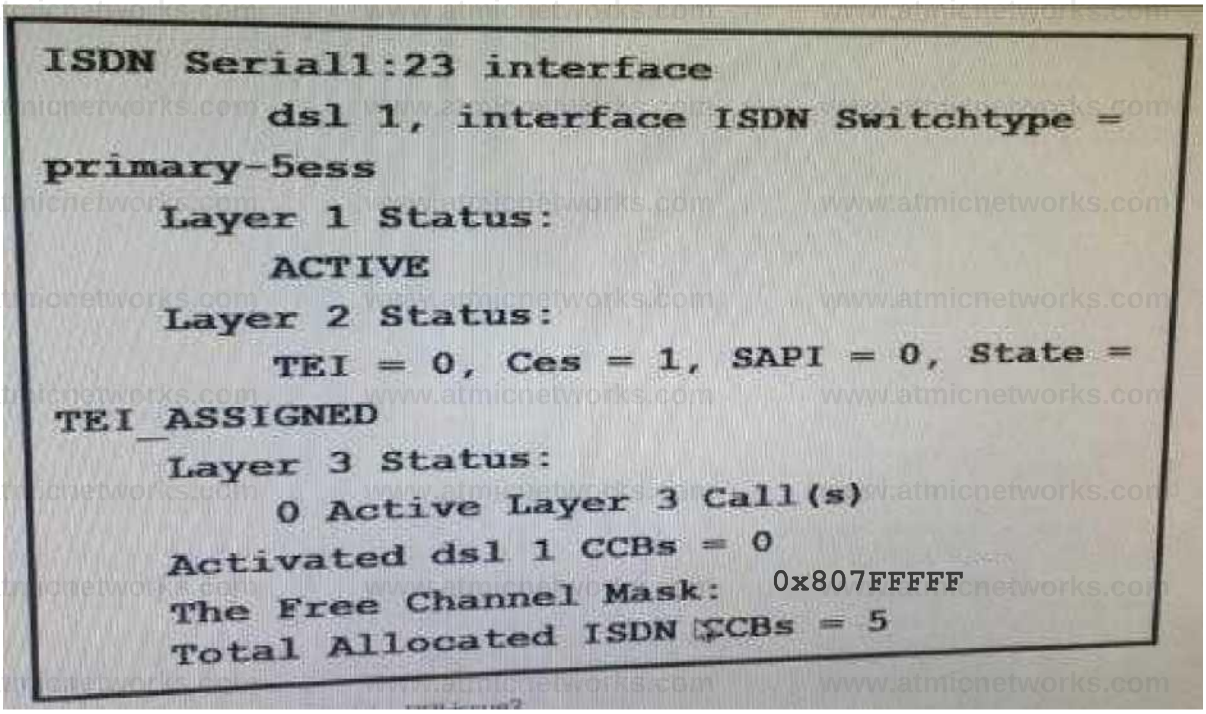
D. 166

Answer: C

Explanation:

Question: 160

Refer to exhibit.



```
ISDN Serial1:23 interface
        dsl 1, interface ISDN Switchtype =
primary-5ess
    Layer 1 Status:
        ACTIVE
    Layer 2 Status:
        TEI = 0, Ces = 1, SAPI = 0, State =
TEI_ASSIGNED
    Layer 3 Status:
        0 Active Layer 3 Call(s)
    Activated dsl 1 CCBs = 0
    The Free Channel Mask:   0x807FFFFF
    Total Allocated ISDN CCBs = 5
```

What is a possible cause of the PRI issue?

A. The cable is unplugged.

B. The controller shut down.

C. The clock source is incorrect.

D. The framing is configured incorrectly.

Answer: D

Explanation:

Question: 161

Refer to the exhibit.

```
Via: SIP/2.0/TCP
10.10.10.2:5060;branch=a8bH5bK7954A198F From:
<sip:012345678610.10.10.2>;tag=8D79AF62-DB2 To:
<sip:9O123456O1O.lO.4.14>;
tag=811681-ffa80926-5fac-4cc5-b802- 2dbde74ae7w2-

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
auptime: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

Explanation:

Question: 162

Refer to 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..1 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

Refer to the exhibit. 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

Explanation:

Question: 163

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

Explanation:

Question: 164

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

Explanation:

Question: 165

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

A. Use multiple egress queues to provide priority queuing of RTP voice packet streams and the ability to classify or reclassify traffic and establish a network trust boundary.

B. Use 808.IQ trunking and 802.lp for proper treatment of Layer 2 CoS packet marking on ports with phones connected.

C. Implement IP RTP header compression on the serial interface to reduce the bandwidth requlreq per voice call on point-to-point links.

D. Deploy RSVP to improve VoIP QoS only where it can have a positive Impact on quality and functionality where there Is limited bandwidth and frequent network congestion.

E. Map audio and video streams of video calls (AF41 and AF42) to a class-based queue with weighted random early detection.

Answer: A, B

Explanation:

Question: 166

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

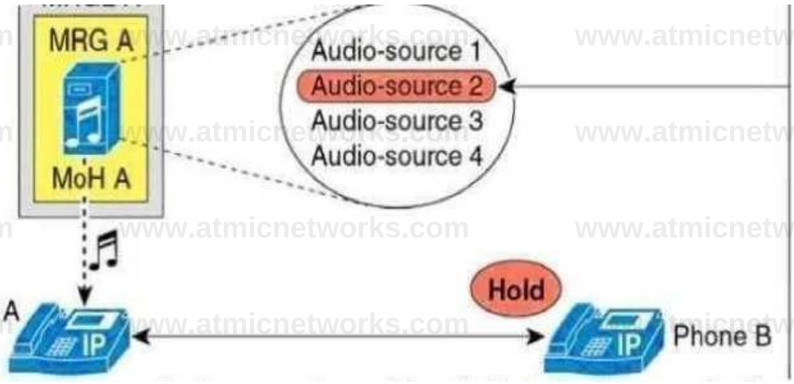
D. The In-Room control Editor on the webpage of the MX800

Answer: D

Explanation:

Question: 167

Refer to exhibit.



User Hold Audio Source = Audio-source4 User Hold Audio Source = Audio-source2 Media Resource Group List = MRGL A Media Resource Group List = MRGL B

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

Explanation:

Question: 168

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

Explanation:

Question: 169

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

Explanation:

Question: 170

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

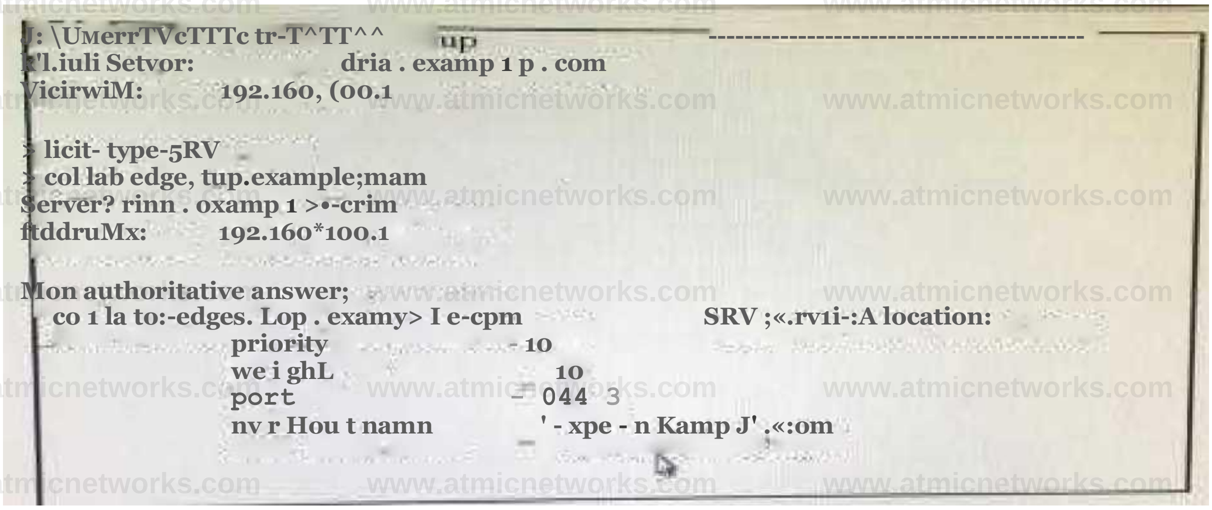
D. greetings

Answer: C

Explanation:

Question: 171

Refer to the exhibit.



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

Explanation:

Question: 172

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: B

Explanation:

Question: 173

Which service must be enabled when LDAP on cisco UCM is used?

A. Cisco AXL Web Service

B. Cisco CallManager SNMP Service

C. Cisco DirSync

D. Cisco Bulk Provisioning Service

Answer: C

Explanation:

Question: 174

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

Explanation:

Question: 175

A user dials 9011841234567 to reach Vietnam. Which steps send the call to the PSTN provider as 011841234567?

A)

m the CalU^ Party 11 anrthwnp^ipn1 'jdi^ri l»i rfun J aiMf i twtSwm, < R.u ılHJUi L■ lltii f 'dtliTii ■as
W 0Hfi417iWifi/
i IHlhcjUfl, ttlR Lh'^-riid UUptTl J3-l^r-firA

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 ... configuration
configure the Pattern ...
... Predot 10

A. Option A

B. Option B

C. Option C

D. Option D

Answer: A

Explanation:

Question: 176

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

A. SSO

B. H.323 registration proxy to Cisco Unified Communications Manager

C. MGCP gateway registration

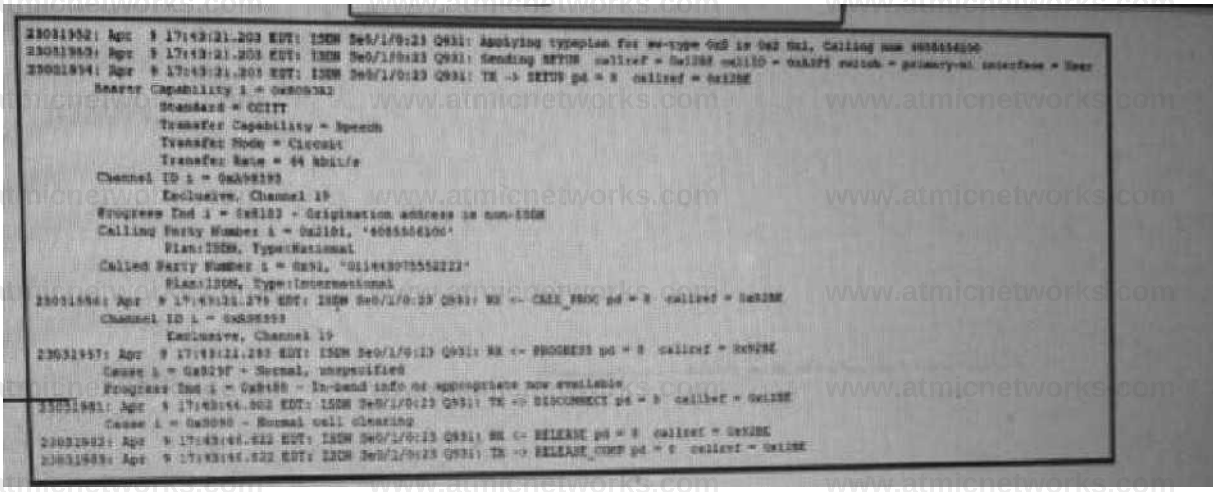
D. SRST

Answer: A

Explanation:

Question: 177

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

Explanation:

Question: 178

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

Explanation:

Question: 179

Which task is required when configuring self-provisioning for an end user in Cisco UCM?

A. Enable Auto-Registration.

B. Associate the end user to the Standard CCM Super Users group

C. Associate the end user to a SIP Profile.

D. Disable Auto-Registration.

Answer: A

Explanation:

Question: 180

Which command must be defined before an administrator changes the line code 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

Explanation:

Question: 181

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

Explanation:

Question: 182

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

Explanation:

Question: 183

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: D, E

Explanation:

Question: 184

Which attribute contains an XMPP stanza?

A. iq

B. message

C. type

D. presence

Answer: A

Explanation:

Question: 185

Refer to the Exhibit.

```
dspfarm profile 1 mtp|
    codec q711ulaw
    maximum sessions software 50 associate application SCCP
```

Which command is required to allow this media resource to handle Video Media streams?

A. maximum sessions hardware 50

B. video codec h264

C. codec pass-through

D. associate application Cisco unified border element

Answer: C

Explanation:

Question: 186

A Cisco IP Phone 7841 that is registered to a Cisco Unified Communications Manager with default configuration receives a call setup message. Which codec is negotiated when the SDP offer includes this line of text?

M=audio 498181 RTP/AVP 0 8 97

A. G.711ulaw

B. iLBC

C. G.711alaw

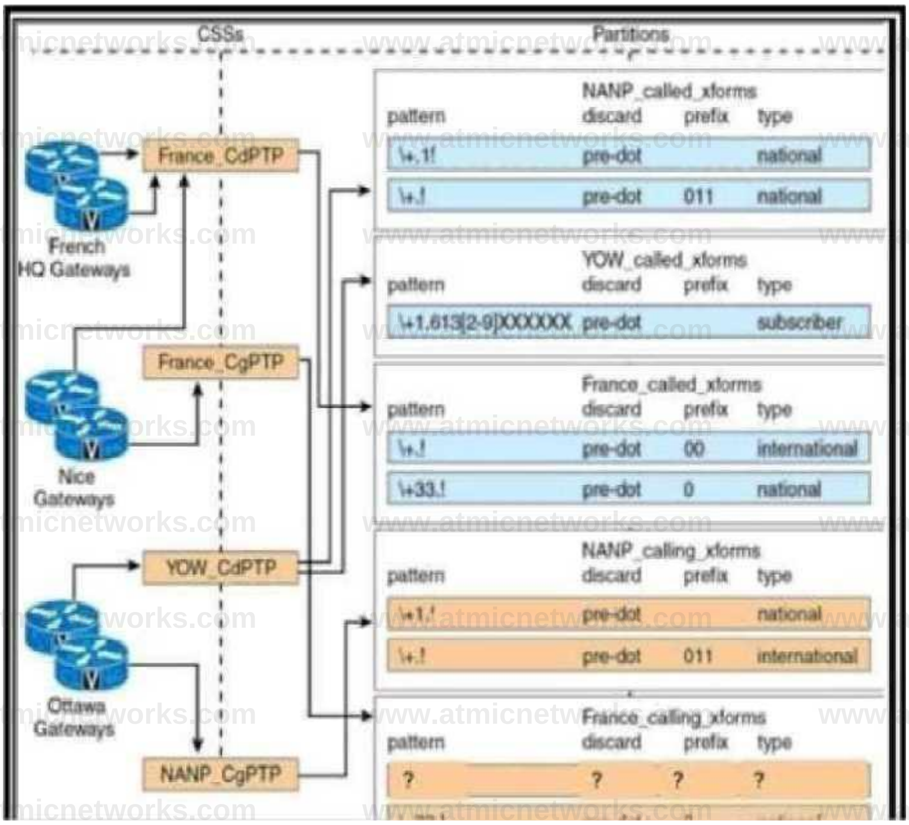
D. G.722

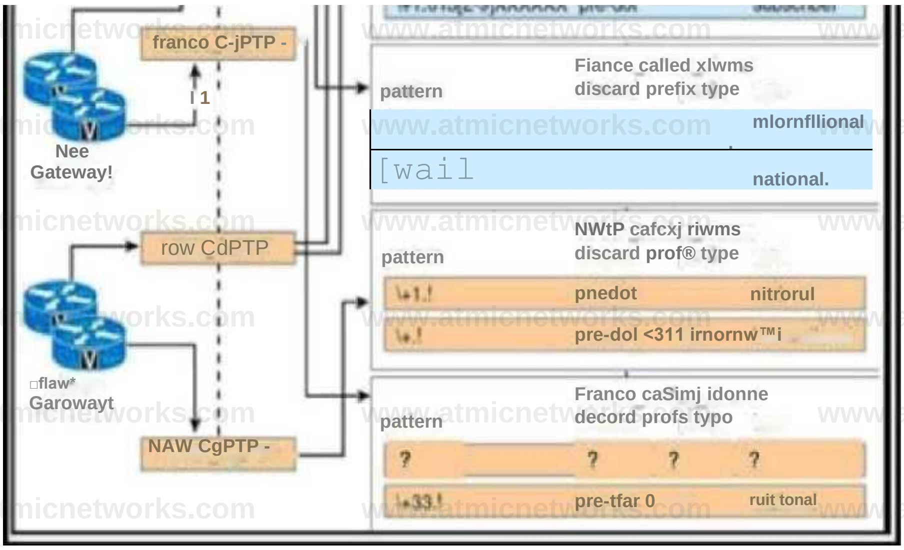
Answer: A

Explanation:

Question: 187

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

Explanation:

Question: 188

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

Explanation:

Question: 189

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

A. SNMPv1

B. SNMPv3

C. SNMPv2

D. SNMPv2c

Answer: B

Explanation:

Question: 190

Refer to exhibit.

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

Pattern® VU96929810. CSS=Unreslricted-CSS. PreDot. Prefix=777011

Pattern® W.777011496929810. CSS=Inll_CSS

Paltern= W.011496929810. CSS=Global-CSS. PreDot. Prefix=777

PaHem= \+.496929810, CSS=Intl CSS, PreDot. Preflx=777011

A. Option A

B. Option B

C. Option C

D. Option D

Answer: C

Explanation:

Question: 191

Refer to exhibit.

□erSrrmSuj^aTrUaTeSaTTTvTTSTTTjTs*"
CSep: 181 OPTIONS
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, MUCA, UPDATE, REFER. SUBSCRIBE, NOTIFY, INFO, AltI STEP
Allow-Events: telephone-event
Acc«i>t: oppl>cetlon/*dp
Supportedi 188:el,finer,resource-prior ity,rec laces,spp-anat

Content-Type: 8ppllcatioh/sdp
Content-Length: 368

r-t
o-ClscoSysteesSIP-CW-UserAqent MM 4117 IN IP4 1».8.140.23 s*SIP Call
c=]N IM IP.8.140.23 t«8 »
n^auOLo • RTP/AVP 18 8 8 4 15
C-IN IPS 18,8.148.23
n^loage 0 udptl t3B C=1N IP4 18.8.148.23
a»T38fa<Versl0n:8
a=T3«N»«0|tJlate:96M
«*T3BP««R»teMpnpgeeentttr8MferredTCF
a-T3«PxNM8ufter:28«
a=T38Fi«NMOatagran:32e
4=T38Fn»Utfp£C:138U0PRedwn8«ncy

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

"0 729 G.711U G.Hla G723 G728

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

b^uM 0IWWW o 10 fl 4 ft

ifi^ORTPMlbBJSie

mwrtlc 0 fi TP^-f a & LMU

m-BUd&O kTPAL^ 1&$ &4 Efl

A. Option A

B. Option B

C. Option C

D. Option D

Answer: D

Explanation:

Question: 192

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: C, D

Explanation:

Question: 193

An administrator is designing a new Cisco UCM for a company with many departments and firm structure on their communications policies. The administrator must make sure that these communication policies are reflected in the phone system setup. Certain departments cannot be accessed directly, even if they have dedicated DID numbers. Some phones, especially public phones, must not be able to dial international numbers Which type of function is configured to control which device is allowed to call another device in Cisco UCM?

A. partitions and calling search spaces

B. calling patterns and route patterns

C. regions and device pools

D. links and pipes

Answer: A

Explanation:

Question: 194

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

Explanation:

Question: 195

Where in Cisco UCM is restrictions on audio bandwidth configured?

A. location

B. partition

C. region

D. serviceability

Answer: C

Explanation:

Question: 196

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

A. 5 min

B. 10 min

C. 30 min

D. 60 min

Answer: C

Explanation:

Question: 197

Why does Cisco UCM use DNS?

A. It provides certificate-based security for media

B. It resolves FQDN to IP address resolution for trunks

C. it connects endpoints to single sign-on services.

D. It provides SRV resolution to the endpoints registered

Answer: D

Explanation:

Question: 198

An administrator is configuring a new Cisco UCM with PSTN capabilities. Due to bandwidth constraints, audio compression is used on the codec. DTMF must work as expected because the customer is calling many call centers where the users must select options in the call. Where is DTMF out-of-band in a CCM 12.5 with SIP-based gateway configured?

A. in the DTMF setting under SIP profile on the Cisco Unified Border Element

B. in the dial peer on the Cisco IOS router

C. in regions on the Cisco UCM where the appropriate codec to use is set

D. in DTMF settings in the audio codec preference list under regions in the Cisco UCM

Answer: B

Explanation:

Question: 199

A high-speed network is often configured with a five-class QoS model. Which classes are used in the model?

A. real-time, call-signaling, critical data, best-effort, and scavenger

B. real-time, signaling, critical data, best-effort and drop-class

C. call-signaling, real-time, critical data, best-effort, and drop-class

D. voice, video, signaling, critical data, and best-effort

Answer: A

Explanation:

Question: 200

Which characteristic of distributed class- based weighted fair queueing addresses jitter prevention?

A. It provides additional granularity by allowing a user to create classes

B. It minimizes jitter by implementing a priority queue for voice traffic

C. It uses a priority queue for voice traffic to avoid jitter.

D. It provides additional granularity by allowing a user to define custom class

Answer: B

Explanation:

Question: 201

What are two characteristics of jitter in voice and video over IP communications? (Choose two.)

A. The packets arrive with frame errors.

B. The packets arrive at varying time intervals.

C. The packets arrive out of sequence.

D. The packets never arrive due to tail drop.

E. The packets arrive at uniform time intervals.

Answer: B, C

Explanation:

Question: 202

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

A. highly secure freal-traversal technology to extend organizational reach.

B. additional visibility of the edge traffic in an organization.

C. complete endpoint registration and monitoring capabilities for devices that are local and remote.

D. session-based access to comprehensive collaboration for remote workers, without the need for a separate VPN client.

E. utilization and adoption metrics of all remotely connected devices.

Answer: A, D

Explanation:

Question: 203

Exhibit.

admin:utils ntp status ntpd (pid 14550) is running...

remote	refid	st	t	when	poli	reach	delay	offset	jitter
*192.168.1.1	17.253.14.125	2	u	39	64	3	0.456	-0.236	0.116
*192.168.1.2	17.253.14.125	2	u	38	64	3	0.817	-0.695	0.395

Refer the exhibit. A collaboration engineer needs to replace the original, single NTP server that was configured during the inital install of a Cisco UCM server. What is the first step to accomplish this task?

A. Restart the NTP service on Cisco UCM

B. Delete the original NTP server from Cisco UCM

C. Stop the NTP service on Cisco UCM

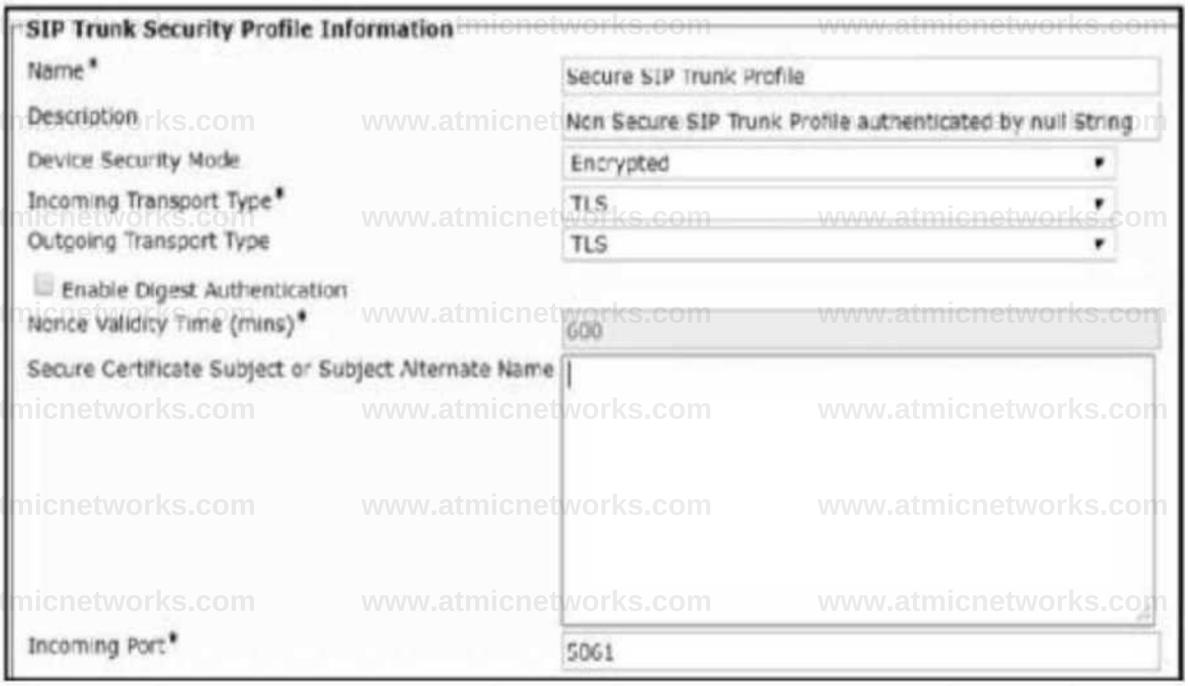
D. Enable NTP authentication for the new NTP server on Cisco UCM

Answer: B

Explanation:

Question: 204

Refer to the exhibit.



An administrator configures a secure SIP trunk on Cisco UCM.

Which value is needed in the secure certificate subject or subject alternate name field to accomplish this task?

A. The fully qualified domain name of the remote device that is configured on the SIP trunk.

B. The common name of the Cisco UCM CallManager certificate.

C. The full qualified domain name of all Cisco UCM nodes that run the CallManager service.

D. The common name of the remote device certificates.

Answer: B

Explanation:

Question: 205

A SIP phone has been configured in the system with MAC address 0030.96D2.D5CB. The phone retrieves the configuration file from the Cisco UCM. Which naming format is the file that is downloaded?

A. SIP003096D2D5CB.cnf.xml

B. SEP003096D2D5CB.cnf.xml

C. SEP003096D2D5CB.cnf

D. SIP003096D2D5CB.cnf

Answer: B

Explanation:

Question: 206

Which actions required for a firewall configuration on a Mobile and Remote Access through Cisco Expressway deployment?

A. The traversal zone on Expressway-c points to Expressway-e through the peer address field on the traversal zone, which specifies the Expressway-e server address. For dual NIC deployments, set the Expressway-e address using an FQDN that resolves the IP address of the internal interface

B. The external firewall must allow these inbound connections to Expressway: SIP: TCP 5061; HTTPS: TCP 8443; XMPP TCP 5222; media: UDP 36002 to 59999

C. Do not use a shared address for Expressway-e and Expressway-c, as the firewall cannot distinguish between them. If static NAT for IP addressing on Expressway-e is used, ensure that any NAT operation on expressway-c does not resolve the same traffic IP address. Shared NAT IS not supported

D. The internal firewall must allow these inbound and outbound connections between

expressway - c and Expressway-e :sip;HTTPS(tunneled over SSH between C and E.

TCP 2222: TCP 7001: Traversal Media: UDP 2776 to 2777(or 36000 to 36011 for large VM/appliance); XMPP:TCP 7400

Answer: B

Explanation:

Question: 207

An administrator configures Cisco UCM to use UDP for SIP signaling and finds that an endpoint cannot make calls. Which action resolves this issue?

A. Change the common phone profile.

B. Change the SIP dial rules.

C. Change the SIP profile.

D. Change the phone security profile.

Answer: D

Explanation:

Question: 208

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

A. iLBC to G.711ulaw

B. G.728br8 to G.711alaw

C. G.729r8 to G.711ulaw

D. G.722 to G.729r8

E. G.729ar8 to G.711alaw

Answer: C, E

Explanation:

Question: 209

Refer to the exhibit.

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

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

A. bandwidth 170 to reserve 170

B. bandwidth 170 to LL1 170

C. bandwidth 170 to priority 170

D. bandwidth 170 to percent 170

Answer: C

Explanation:

Question: 210

Which type of input is required when configuring a third-party SIP phone?

A. digest user

B. manufacturer

C. serial number350-801 2023-4

D. authorization code

Answer: A

Explanation:

Question: 211

Callers from a branch report getting busy tones intermittetly when trying to reach colleagues in other office branches during peak hours. An engineer collects Cisco

CallManager service traes to examine the situation. The traces show:

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

What should be fixed to resolve the issue?

A. class of service configuration

B. region configuration

C. geolocation configuration

D. codec configuration

Answer: B

Explanation:

Question: 212

After an engineer implements the FAC and CMC features together, users report that calls take almost one minute to complete and that they occasionally hear the reorder tone. Which two actions address this issue? (Choose two)

A. Adjust the T302 timer from the default of 15 seconds to 5 seconds to shorten the interdigit timer

B. Change the code if the problem persists

C. Advise the user to hang up and try again

D. Do not wait for the tones immediately dial the FAC and CMC

E. Advise the user to press the"#"button after dialing the FAC and CMC codes

Answer: A, B

Explanation:

Question: 213

Which Cisco unity Connection handler plays a greeting at announces the option to dial a user extension by default?

A. the operator call handler

B. the Interview handler

C. the Goodbye call handler

D. the Directory handler

Answer: A

Explanation:

Question: 214

Which type of message must an administrator configure in the SIP Trunk Security Profile for a Message Waiting Indicator light to work with a SIP integration between Cisco UCM and Cisco Unity Connection?

A. Unsolicited NOTIFY

B. 200 ok

C. SIP Register

D. TCP port 5060

Answer: A

Explanation:

Question: 215

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

A. Ensure that ad hoc conference calls are dropped if an external user is add.

B. Call forward settings (ALL/Busy/No Answer) are restricted to internal extensions in the network

C. Add an additional firewall between the Cisco UCM server and the Expressway Core server.

D. For extension mobility, logged-out CSS is restricted to internal extensions and emergencies.

E. Forced authorization code is used to recognize a dialing extension and authorize an international call.

Answer: B, E

Explanation:

Question: 216

Which Cisco IM and Presence service handles failover and state changes in the cluster?

A. XCP Sync Agent

B. Cisco Server Recovery Manager

C. Cisco XCP Connection Manager

D. XCP router

Answer: B

Explanation:

Question: 217

SIP proxies have operations defined in RFC 3261 and supporting extensions. Though no IETF RFC completely defines how SBCs must function. SBCs evolved over the years.

Which two operations demonstrate the high-level differences between SBCs and SIP proxies?

(Choose two.)

A. Stateful proxies are context-aware and can terminate communication sessions by themselves

B. SIP proxies add a Via header and optionally a Record-Route header, and the rest of the headers are left untouched

C. SBCs can modify headers such as To, From, Contact, and Call-D.It can introduce new headers into the SIP message

D. SBCs are capable of interworking completely different protocols to set up, modify, and tear down communication sessions. It includes SIP, H.323,and MGCP protocols

E. SIP proxies are SDP-aware and can change the SDP bodies

Answer: B, D

Explanation:

Question: 218

Refer to the exhibit.

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

When a call is received on Cisco Unified Border Element. from which

IP does it allow a connection?

A. 192.168.100.103

B. 192.168.102.102

C. 192.168.100.102

D. 192.168.101.201

Answer: B

Explanation:

Question: 219

What are two common attributes of XMPP XML stanzas? (Choose two.)

A. from

B. to

C. destination

D. version

E. Source

Answer: A, B

Explanation:

Question: 220

Which two devices are supported by the Flexible DSCP Marking and Video Promotion feature? (Choose two.)

A. MGCP devices

B. SCCP devices

C. pass-through MTPs

D. H.323 trunks

E. DX80

Answer: B, C

Explanation:

Question: 221

A Cisco voice gateway is configured to use a sip-kpml DTMF relay in global settings. A new SIP dial peer is configured for a third-party application that only supports an in-band DTMF relay. Which commands must an engineer run on the dial peer?

A. dtmf-relay sip-info

B. dtmf-relay sip-notify

C. dtmf-relay rtp-net

D. no dtmf-relay sip-kpml

Answer: C

Explanation:

Question: 222

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

A. Video traffic cannot tolerate any packet loss and has a latency of 150 miliseconds

B. Video traffic can tolerate up to 10% packet loss and latency of 10 seconds

C. Video trafic can tolerate up to 5% packet loss and latency of 5 seconds

D. Video traffic can tolerate a packet loss of up to 1% and latency of 150 milliseconds

Answer: D

Explanation:

Question: 223

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

A. QoS trust boundaries include all the devices directly attached to the access switch ports

B. QoS trust boundaries can be extended to Jabber running on a PC

C. QoS trust boundaries exclude Jabber softphone running on a PC

D. QoS trust boundaries can be extended to voice and video devices if the connected PCs are included

E. QoS trust boundaries can be extended to voice and video devices exclusively

Answer: C, D

Explanation:

Question: 224

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

A. Block

B. Transfer to Number

C. Reject

D. Restrict

Answer: D

Explanation:

The Restrict setting in the Webex Calling dial plan prevents users from placing certain types of outbound calls. For example, you can use the Restrict setting to prevent users from making international calls or calls to premium-rate numbers.

The Block setting in the Webex Calling dial plan prevents users from placing any outbound calls. The Transfer to

Number setting in the Webex Calling dial plan transfers all outbound calls to a specified number. The Reject setting in the Webex Calling dial plan rejects all outbound calls.

Here is a table summarizing the different dial plan settings and their effects:

Dial Plan Setting	Effect
Block	Prevents users from placing any outbound calls.
Transfer to Number	Transfers all outbound calls to a specified number.
Reject	Rejects all outbound calls.
Restrict	Prevents users from placing certain types of outbound calls.

Question: 225

When configure Cisco UCM,which configuration enables phones to automatically reregister to a Cisco UCM publisher 'when the connection to the subscriber is lost?

A. SRST

B. Route Group

C. Cisco UCM

D. Device Pool

Answer: A

Explanation:

SRST, or Survivable Remote Site Telephony, is a feature that allows Cisco IP phones to continue to function even when the connection to the Cisco UCM publisher is lost. When SRST is configured, the phones will automatically reregister to the publisher when the connection is restored.

Route groups are used to route calls to different destinations based on the caller's phone number or other criteria. Cisco UCM is the call management system that controls the IP phones. Device pools are used to group phones together for administrative purposes.

Question: 226

Which option must be used when configuring the Local Gateway for a Cisco Webex Calling trunk?

A. local authentication

B. certificate-based

C. mutual TLS

D. Auth-based

Answer: B

Explanation:

[A certificate-based trunk is a type of trunk that uses certificates to authenticate the connection between Webex Calling and the Local Gateway1]. [A Local Gateway is a supported session border controller that terminates the trunk on the premises2]. [A certificate-based trunk requires a certificate authority (CA) to issue and manage the certificates for both Webex Calling and the Local Gateway1].

Question: 227

What is set when using COS to mark an Ethernet frame?

A. Ipp bits

B. IP ECN bits

C. DCSP bits

D. 802.1 p User Priority bits

Answer: D

Explanation:

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

Question: 228

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

A. Set the DELETEUSERDATA=r installation argument

B. Set the "HKEY_LOCAL_MACHINE,.Software'«WOW6432Node .CiscoCollabHost -Eula_disable

C. Set the "HKEY_LCX^M._MACHINE .SoftwareCiscoCollabHo$t€ula Setting registry Eula_disable

D. Set the DEFAULT^'THEMEs-Dark"" installation argument

E. Set the "/quiet installation argument

Answer: BC

Explanation:

The correct answers are B and C.

To ensure that end users are not prompted to accept the end-user license agreement (EULA) when they start the Webex app, the engineer must set the following two registry keys:

HKEY_LOCAL_MACHINE\Software\WOW6432Node\CiscoCollabHost\Eula_disable

HKEY_LOCAL_MACHINE\Software\CiscoCollabHost\Eula Setting\Eula_disable

Setting these registry keys will disable the EULA prompt for all users who start the Webex app.

The other options are not valid actions to ensure that end users are not prompted to accept the EULA.

Question: 229

Which two protocols can be configured for the Cisco Unity Connection and Cisco UCM integration? (Choose two.)

A. H. 323

B. SIP

C. SCCP

D. MGCP

E. RTP

Answer: BC

Explanation:

The two protocols that can be configured for the Cisco Unity Connection and Cisco UCM integration are SIP and SCCP.

SIP, or Session Initiation Protocol, is a signaling protocol used for initiating, maintaining, and terminating real-time sessions, including voice, video, and messaging applications.

SCCP, or Skinny Client Control Protocol, is a Cisco proprietary signaling protocol used for controlling Cisco IP phones.

H.323 is an older signaling protocol that is no longer widely used. MGCP, or Media Gateway Control Protocol, is a protocol used for controlling media gateways. RTP, or Real-time Transport Protocol, is a protocol used for transporting real-time data, such as voice and video

Question: 230

What are two access management mechanisms in Cisco Webex Control Hub? (Choose two.)

A. multifactor authentication

B. Active Directory synchronization

C. attribute-based access control

D. single sign-on with Google

E. Client ID/Client Secret

Answer: AB

Explanation:

The correct answers are A and B.

The two access management mechanisms in Cisco Webex Control Hub are multifactor authentication and Active Directory synchronization.

Multifactor authentication is a security measure that requires users to provide two or more pieces of evidence to verify their identity. This can include something they know, such as a password, and something they have, such as a security token.

Active Directory synchronization is a process that allows Cisco Webex Control Hub to automatically synchronize user accounts from an Active Directory domain. This can simplify user management and provide users with single sign-on access to Cisco Webex Control Hub and other applications.

Question: 231

What must be configured on a Cisco Unity Connection voice mailbox to access the mailbox from a secondary device?

A. mobile user

B. alternate names

C. alternate extensions

D. Attempt Forward routing rule

Answer: C

Explanation:

To access a Cisco Unity Connection voice mailbox from a secondary device, you must configure an alternate extension for the mailbox. This is a phone number that is different from the mailbox's primary extension. When you call the alternate extension, you will be prompted to enter the mailbox's PIN. Once you have entered the PIN, you will be able to access the mailbox just as you would if you were calling from the primary device.

Question: 232

What is an advantage of using Cisco Webex Control HuB?

A. enables the provisioning, administration, and management of Webex services and Webex Hybrid Services

B. brings Video, audio, and web communication together to meet the collaboration needs of the modern workplace

C. provides streamlined communication and collaboration for a hybrid workforce

D. offers easy contact management, centralized administration, and centralized configuration management

Answer: A

Explanation:

Cisco Webex Control Hub is a cloud-based management platform that enables you to provision, administer, and manage Webex services and Webex Hybrid Services. It provides a single pane of glass for managing all of your Webex services, including Webex Meetings, Webex Teams, and Webex Calling.

Webex Control Hub offers a number of features and benefits, including:

A single pane of glass for managing all of your Webex services

Centralized user management

Simplified provisioning and administration

Real-time analytics and reporting

Enhanced security and compliance

Webex Control Hub is a powerful tool that can help you manage your Webex services more effectively. It is easy to use and provides a number of features and benefits that can help you improve your productivity and efficiency.

Question: 233

Which Webex Calling construct is used to organize calling features within a physical site?

A. client settings

B. locations

C. service settings

D. call routing

Answer: B

Explanation:

A location is a physical site that contains users, devices, and resources. Locations are used to organize calling features within a physical site. For example, you can create a location for each of your offices and then assign users, devices, and resources to that location. This will allow you to manage calling features for each office separately.

Question: 234

Refer to the exhibit.

```
ISDN Serial1:23 interface
        dsl 1, interface ISDN Switchtype =
primary-5ess
    Layer 1 Status:
        ACTIVE
    Layer 2 Status:
        TEI = 0, Ces = 1, SAPI = 0, State =
TEI_ASSIGNED
    Layer 3 Status:
        0 Active Layer 3 Call(s)
    Activated dsl 1 CCBs = 0
    The Free Channel Mask:  0x807FFFFF
    Total Allocated ISDN CCBs = 5
```

What causes the PRI issue?

A. The controller shut down

B. The cable is unplugged

C. The framing is configured incorrectly

D. The clock source is incorrect.

Answer: **B**

Explanation:

is that the cable is unplugged.

Question: 235

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

A. voice codec

B. session capacity of the PVDM module

C. number of protocol data units

D. software version Of the router

E. license types

Answer: **AB**

Explanation:

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

Question: 236

An engineer roust configure DTMF relay on a Cisco Unified Border Element by using RFC2833 as the preferred relay mechanism and KPML as the next preferred relay mechanism. The engineer logs in to the CUBE and enters the dial-peer configuration level. Which command should be run at dial-peer configuration level?

A. dtmf-relay sip-kvmi rtp-nte

B. dtmf- relay rtp-nte sip-kpml

C. dtmf-relay sip-kgml rtp-inband

D. dtmf-relay rtp-inband sip-kvmi

Answer: B

Explanation:

The dtmf-relay command is used to configure DTMF relay on a Cisco Unified Border Element. The rtp- nte option specifies that RFC2833 is the preferred relay mechanism, and the sip-kpml option specifies that KPML is the next preferred relay mechanism.

Question: 237

Where is Directory Connector hosted in a Cisco Webex Hybrid Services deployment?

A. on a server in the Webex Data Center

B. on a dedicated on-premises server

C. on a Cisco Expressway-C connector host server

D. on an on-premises Microsoft Active Directory server

Answer: B

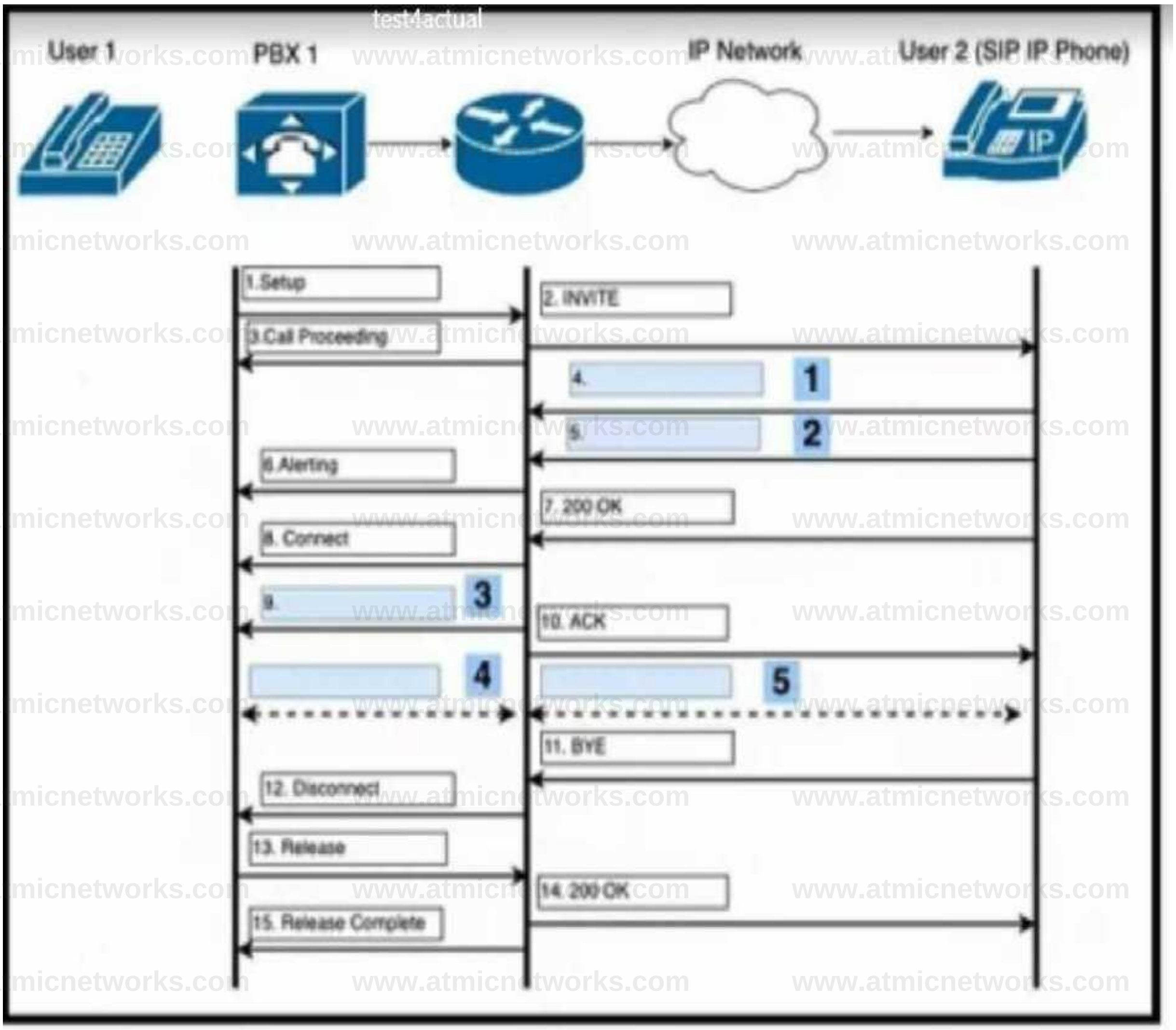
Explanation:

The Cisco Directory Connector is a software application that is installed on a dedicated on-premises server. It synchronizes user identities between the on-premises directory and the Cisco Webex cloud.

Question: 238

DRAG DROP

Refer to the exhibit.



https://i.postimg.cc/wMYy0Fhm/image.pn

Drag and drop the flow step labels from the left into the correct order on the right to establish this call flow:

- User 1 calls user 2.
- User 2 answers the call
- user 2 disconnects the call

two-way voice path

two-way RTP channel

100 Trying

Connect ACK

180 Ringing

Answer:

Explanation:

1. 100 Trying
2. 180 Ringing
3. two-way voice path
4. Connect ACK
5. two-way RTP channel

Question: 239

What should be used to detect common issues on a Cisco IOS XE-based Local Gateway and generate an email?

A. Real-Time Monitoring Tool

B. diagnostic signatures

C. syslog

D. SNMP

Answer: B

Explanation:

Diagnostic signatures are a feature that proactively detects commonly observed issues in the IOS XE- based Local Gateway and generates email, syslog, or terminal message notification of the event12. You can also install the diagnostic signatures to automate diagnostics data collection and transfer collected data to the

Cisco TAC case to accelerate resolution time12.

Question: 240

What is a capability of the call forwarding feature in a Cisco Webex dial plan?

A. device pool selection

B. Call Admission Control

C. business continuity

D. ringtone selection

Answer: C

Explanation:

Call forwarding is a feature that allows users to forward incoming calls to another number. This can be useful in a number of situations, such as when a user is not available to take a call, or when a user wants to forward calls to a different number during certain times of the day.

Call forwarding can be used to improve business continuity by ensuring that calls are always answered, even if the user is not available. For example, if a user is out of the office, they can forward their calls to their voicemail or to another employee. This ensures that customers and clients can always reach someone, even if the user is not available.

Question: 241

An engineer is deploying the Webex app on a Microsoft Windows computer that has multiple user accounts. Which CLI argument allows the engineer to Install the application as a "per machine" Installation?

A. ACCEPT_EULA

B. INSTALL ROOT

C. ALLUSERS

D. FORCELOCKDOVN

Answer: C

Explanation:

The ALLUSERS property is a command-line argument that can be used to install the Webex app as a "per machine" installation. This means that the app will be installed for all users on the computer, regardless of which user account is currently logged in.

The ACCEPT_EULA property is a command-line argument that can be used to accept the end-user license agreement (EULA) for the Webex app. The INSTALL_ROOT property is a command-line argument that can be used to specify the installation directory for the Webex app. The FORCELOCKDOWN property is a command-line argument that can be used to prevent users from uninstalling the Webex app.

Question: 242

The chief officer at a company must reduce collaboration infrastructure costs by on bording all on - permises equipment to the cloud by using CISCO Webex Control Hub. Administrators need the ability to manage upgrades and set up hot desking for on-premises devices.

Which action must be taken before on boardng devices by using the Control Hub?

A. Configure tie Control Hub organization ID on the devices

B. Acquire a license for each device.

C. Allow HTTP traffic from each device to Control Hub.

D. Upgrade all the devices to software version CE9.15 or later

Answer: D

Explanation:

This is a prerequisite for using the Device Connector tool, which allows you to onboard and register several devices simultaneously to the Webex Control Hub1. The Device Connector tool creates a workspace, an activation code, and activates all of your devices in one go1. This way you don't need to be physically present in the same room to activate the devices.

The other options are not required before onboarding devices by using the Control Hub:

Configuring the Control Hub organization ID on the devices is not necessary, as the Device Connector tool will send the device information to your Webex organization and generate activation codes for them1.

Acquiring a license for each device is not necessary, as you can assign licenses to users and devices after they are registered to the Webex Control Hub2.

Allowing HTTP traffic from each device to Control Hub is not necessary, as HTTPS connectivity is required for the Device Connector tool to communicate with the devices1.

Question: 243

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

A. any two nodes

B. any node

C. only nodes that have Cisco UCM service enabled

D. any subscriber nodes

Answer: C

Explanation:

TFTP is a network protocol that is used to transfer files between devices. It is often used to transfer firmware and configuration files to network devices. In order to use TFTP, the device must have a TFTP server configured.

In Cisco UCM, the TFTP server is configured on the serviceability page. The TFTP server can be configured on any node that has Cisco UCM service enabled. The TFTP server cannot be configured on nodes that do not have Cisco UCM service enabled.

Question: 244

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

A. ALLUSERS=0

B. ENABLEVDI=1

C. ALLUSERS=1

D. ENABLEVDI=2

E. ROAMINGENABLED=1

Answer: B, E

Explanation:

The Cisco Webex app can be installed on a Windows Virtual Desktop Infrastructure (VDI) environment by using the following command-line arguments:

ENABLEVDI=1 - This argument enables VDI mode for the Webex app.

ROAMINGENABLED=1 - This argument enables roaming for the Webex app.

The ALLUSERS argument is not required when installing the Webex app on a VDI environment. The ENABLEVDI argument must be set to 1, and the ROAMINGENABLED argument must be set to 1.

The following is an example of the command that can be used to install the Webex app on a VDI environment:

Code snippet

msiexec /i WebexApp.msi ENABLEVDI=1 ROAMINGENABLED=1

Question: 245

An engineer must configure codec on a Cisco Unified Border Element to prefer the G.711 ulaw and use G.711 codec as the next The engineer logs In to the CUBE, enters the dial.peer configuration level, and runs the voice- class codec 100 command. Which set of commands completes the configuration?

A. voice class codec 100 codec g7llalaw preference 1 codec a7Hulaw preference 2

B. voice class codec 11j codec <?7iiulaw preferred codec g7iialaw

C. vice class codec 100

codec preference 1 g7llulaw

codec preference 2 o711alaw

D. voice class codec ::: codec g711ulaw g711alaw

Answer: C

Explanation:

The following commands are used to configure the codec on a Cisco Unified Border Element to prefer the G.711 ulaw and use G.711 alaw as the next codec:

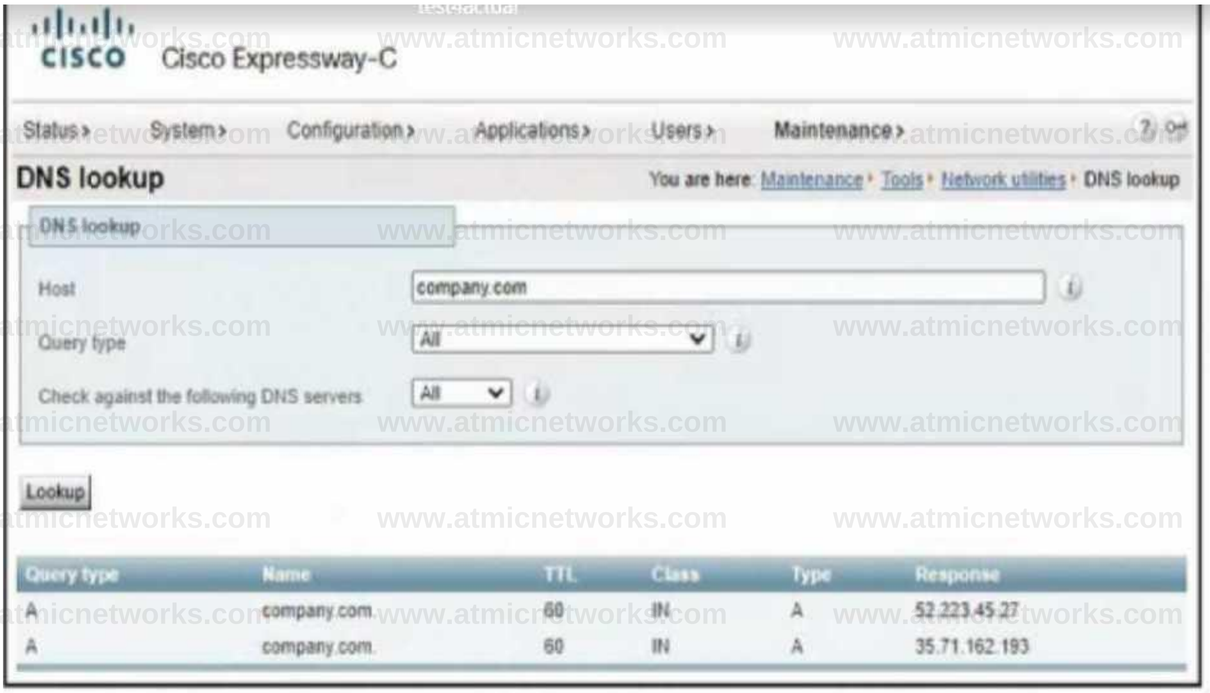
Code snippet

```
voice class codec 100
codec preference 1 g711ulaw
codec preference 2 g711alaw
```

The voice class codec 100 command creates a new voice class with the ID of 100. The codec preference 1 g711ulaw command sets the preference for the G.711 ulaw codec to 1. The codec preference 2 g711alaw command sets the preference for the G.711 alaw codec to 2.

Question: 246

Refer to exhibit.



A company recently deployed CISCO Jabber Users log in to Jabber by using their email address in a domain named company com. The users report that they cannot register their telephony services when working from unless they use a VPN. An engineer runs DNS lookup tool in Cisco Expressway-C to troubleshoot tie What IS me cause of the issue?

A. The company com domain must be resolved only in Expressway-E

B. There is a missing SRV record for the company.com domain.

C. The TTL value for the company.com is too short.

D. There must be only response for the company.com domain

Answer: B

Explanation:

Question: 247

What is the purpose of a hybrid Local Gateway?

A. to handle calls between Webex Calling and Cisco Calling Plans

B. to handle calls between Webex Calling and Cloud Connected PSTN

C. to handle calls between Cisco IJCM and Webex Calling

D. to handle calls between the Public Switched Telephone Network and Webex Calling

Answer: D

Explanation:

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

Question: 248

An engineer must enable onboarding of on-premises devices by using activation to a Cisco UCM server The engineer activated the CISCO Device Activation Service and set the default registration method to use the codes. Which action completes the configuration?

A. Set Enable Activation Code enterprise parameter to True-

B. Manually provision new phones that have an activation code requirement

C. Create a Bulk Administration Tool provisioning template.

D. Generate 16-digit codes by using the Bilk Administration Tool

Answer: A

Explanation:

The engineer must set the Enable Activation Code enterprise parameter to True. This will enable the use of activation codes for onboarding on-premises devices to a Cisco UCM server. The other options are not necessary to complete the configuration.

Here are the steps to complete the configuration:

Log in to the Cisco Unified Communications Manager (CUCM) Administration interface.

Go to System > Enterprise Parameters.

Set the Enable Activation Code enterprise parameter to True.

Click Save.

The activation code onboarding feature is now enabled. You can use it to onboard new phones to the CUCM server.

Question: 249

Which two types of trunks can be used when configuring a hybrid Local Gateway for Cisco Webex Calling? (Choose Two.)

A. TLS-based

B. certificate-based

C. registration-based

D. authentication-based

E. OAuth-based

Answer: A, C

Explanation:

These are the two types of trunks that can be used when configuring a hybrid local gateway for Cisco Webex Calling1. A TLS-based trunk uses Transport Layer Security (TLS) to secure the SIP signaling between the hybrid local gateway and Webex Calling1. A registration-based trunk uses SIP registration to authenticate the hybrid local gateway with Webex Calling and receive calls from the cloud1.

Question: 250

An engineer configures a new phone in Cisco UCM The phone boots and gets IP when it connects to the network, however the phone fails to register With CISCO UCM, The engineer observes that the phone has a status Rejected In CISCO UCM

What must be verified first 'Mien troubleshooting the issue?

A. whether auto-registration is enabled in Cisco UCM

B. whether the Initial Trust List and Certificate Trust List files on the phone are correct

C. whether the phone ts in the correct VLAN

D. whether the phone's MAC address is correct In Cisco UCM

Answer: A

Explanation:

This is the first thing that must be verified when troubleshooting the issue of phone status showing rejected in Cisco UCM1. Auto-registration allows new phones to register with Cisco UCM without manual configuration1. If auto-registration is disabled, the phone will not be able to register and will show a rejected status1.

The other options are not the first things that must be verified when troubleshooting the issue:

B. whether the Initial Trust List and Certificate Trust List files on the phone are correct is not the first thing to verify, but it may be a possible cause of the issue if the phone has an ITL file from another cluster that prevents it from registering with Cisco UCM2. To resolve this issue, the ITL file needs to be deleted from the phone or exchanged between the clusters2.

C. whether the phone is in the correct VLAN is not the first thing to verify, but it may be a possible cause of the issue if the phone is not in the same VLAN as the Cisco UCM server or cannot reach it due to network issues3. To resolve this issue, the network connectivity and VLAN configuration need to be checked and fixed3.

D. whether the phone's MAC address is correct in Cisco UCM is not the first thing to verify, but it may be a possible cause of the issue if the phone's MAC address does not match the one configured in Cisco UCM. To resolve this issue, the MAC address needs to be corrected and updated in Cisco UCM.

Question: 251

What is a capability of a Cisco IOS XE media resource?

A. It provides a hardware conferencing solution.

B. It provides call forwarding capabilities.

C. It provides redundancy for voice calls.

D. It provides a voice packet optimization solution.

Answer: A

Explanation:

A Cisco IOS XE media resource provides a hardware conferencing solution. It can be used to mix multiple media streams, such as audio and video, into a single stream that can be sent to all participants in a conference call. This is done using a digital signal processor (DSP), which is a specialized processor that is designed to handle the processing of digital signals, such as audio and video.

Question: 252

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

A. debug isdn q921

term mon

B. debug voip ecapi inout

show controller ti

C. debug isdn q931

show isdn status

D. debug isdn q921

debug voip ecapi inout

Answer: C

Explanation:

The engineer should run the following debugs or commands in the voice gateway to troubleshoot the issue:

debug isdn q931 - This debug will show the ISDN Q.931 messages that are being exchanged between the voice gateway and the ISDN switch. This can be used to identify the cause of the "Incomplete Destination" error.

show isdn status - This command will show the status of the ISDN PRI circuit. This can be used to verify that the circuit is up and running.

The other options are not correct. The debug isdn q921 command will show the ISDN Q.921 messages that are being exchanged between the voice gateway and the ISDN switch. This is not necessary for troubleshooting the issue. The term mon command will show the terminal monitor output. This is not necessary for troubleshooting the issue. The debug voip ecapi inout command will show the VoIP ECAP messages that are being exchanged between the voice gateway and the VoIP server. This is not necessary for troubleshooting the issue. The show controller ti command will show the status of the T1 controller. This is not necessary for troubleshooting the issue.

Question: 253

What is the default TCP port for SIP OAuth mode in Cisco UCM?

A. 5011
B. 3174
C. 8443
D. 5090

Answer: D

Explanation:

The Cisco Unified Communications Manager (CUCM) uses SIP Phone OAuth Port (5090) to listen for SIP line registration from Jabber OnPremise devices over TLS. However, CUCM uses SIP Mobile Remote Access Port (default 5091) to listen for SIP line registrations from Jabber over Expressway through mTLS. Both of these ports are configurable.

Question: 254

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

A. Cisco Unified Border Element

B. Cisco Unity Connection

C. Cisco Meeting Server

D. Edge

Answer: D

Explanation:

The Cisco Webex DX80 Video endpoint can be provisioned in two ways:

Automatically, using the Cisco Unified Communications Manager (CUCM) or Cisco Video Communication Server (VCS)

Manually, using the Edge provisioning mode

The Edge provisioning mode is used when the endpoint is not connected to the CUCM or VCS. In this mode, the endpoint is configured with the necessary settings, such as the IP address, SIP/H.323 parameters, and time and date.

The Cisco Unified Border Element (Cisco UBE) is a network element that provides security and call control for IP telephony networks. The Cisco Unity Connection is a unified messaging system that provides voicemail, email, and fax services. The Cisco Meeting Server is a video conferencing system that provides high-quality video and audio conferencing.

Question: 255

How is bandwidth allocated to traffic flows in a flow-based WFQ solution?

A. All the bandwidth is divided based on the QoS marking of the packets.
B. Each type of traffic flow has equal bandwidth.
C. Bandwidth is divided among traffic flows. Voice has priority.
D. Voice has priority and the other types of traffic share the remaining bandwidth.

Answer: D

Explanation:

In a flow-based WFQ solution, bandwidth is allocated to traffic flows based on the following criteria: The priority of the traffic flow

The amount of bandwidth that is available

The number of traffic flows that are competing for bandwidth

Voice traffic is typically given a higher priority than other types of traffic, such as data traffic. This is because voice traffic is more sensitive to latency and jitter than data traffic.

When there is not enough bandwidth to accommodate all of the traffic flows, the WFQ algorithm will prioritize the traffic flows based on their priority. The traffic flows with the highest priority will be given the most bandwidth, and the traffic flows with the lowest priority will be given the least bandwidth.

If there is still not enough bandwidth to accommodate all of the traffic flows, the WFQ algorithm will start to drop packets. The packets that are dropped will be the packets from the traffic flows with the lowest priority.

Question: 256

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

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

D. For each route list, configure a route group to use as a Local Route Group.

Answer: C

Explanation:

The Local Route Group feature allows you to use a route group as the destination for calls that are placed from a device pool. The route group that you use as the destination for calls from a device pool is called the Local Route Group for that device pool.

To configure the Local Route Group feature, you must first create a route group. You can then configure the Local Route Group feature for a device pool by selecting the route group that you want to use as the Local Route Group for that device pool.

Question: 257

An engineer encounters third-party devices that do not support Cisco Discovery Protocol. What must be configured on the network to allow device discovery?

A. LLDP
B. TFTP
C. LACP
D. SNMP

Answer: A

Explanation:

LLDP (Link Layer Discovery Protocol) is a vendor-neutral network discovery protocol that is used to discover the topology of a network. LLDP is similar to CDP (Cisco Discovery Protocol), but it is not proprietary to Cisco. LLDP is supported by a wide range of network devices, including switches, routers, and firewalls.

To configure LLDP on a network, you must enable LLDP on the devices that you want to discover. You can then use a network management tool, such as Cisco Network Assistant, to view the topology of the network.

The other options are incorrect. TFTP (Trivial File Transfer Protocol) is a network protocol that is used to transfer files between devices. LACP (Link Aggregation Control Protocol) is a network protocol that is used to aggregate multiple network links into a single logical link. SNMP (Simple Network Management Protocol) is a network protocol that is used to manage network devices.

Question: 258

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

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

Answer: A

Explanation:

The call flow for a Mobile and Remote Access registered endpoint to central call control is as follows: The endpoint registers with the Expressway-C.

The Expressway-C forwards the registration request to the Expressway-E.

The Expressway-E forwards the registration request to the Cisco UCM.

The Cisco UCM registers the endpoint.

When the endpoint places a call, the call flow is as follows:

The endpoint sends the call request to the Expressway-C.

The Expressway-C forwards the call request to the Expressway-E.

The Expressway-E forwards the call request to the Cisco UCM.

The Cisco UCM places the call.

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

Question: 259

An administrator installs a new Cisco TelePresence video endpoint and receives this error:"AOR is not permitted by Allow/Deny list. Which action should be taken to resolve this problem?

A. Reboot the VCS server and attempt reregistration.
B. Change the SIP trunk configuration.
C. Correct the restriction policy settings.
D. Upload a new policy in VCS.

Answer: C

Explanation:

The error message "AOR is not permitted by Allow/Deny list" indicates that the endpoint is not allowed to register with the VCS server because it is not on the Allow List or it is on the Deny List. To resolve this problem, you must correct the restriction policy settings.

Question: 260

Which two configuration elements are part of the Cisco UCM toll-fraud prevention?(Choose two.)

A. feature control policy
B. partition
C. SIP trunk security profile
D. SUBSCRIBE Calling Search Space
E. Calling Search Space

Answer: AE

Explanation:

The following are the configuration elements that are part of the Cisco UCM toll-fraud prevention: Feature control policy - This policy controls the features that are available to users. For example, you can use this policy to prevent users from making international calls.

Calling Search Space - This space defines the numbers that users can call. For example, you can use this space to prevent users from calling premium-rate numbers.

Question: 261

Refer to the exhibit.

https://i.postimg.cc/C57TkczG/image.png

```
v=0
o=Cisco-SIPUA 13439 0 IN IP4 10.10.10.10
s=SIP Call
b=AS:4064
t=0 0
m=audio 0 RTP/AVP 114 9 124 113 115 0 8 116 18 101
c=IN IP4 10.10.10.10
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114 maxplaybackrate=16000;sprop-
maxcapturerate=16000;maxaveragebitrate=64000;stereo=0;sprop-stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0;mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=l;mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
```

A call is falling to establish between two SIP Devices The called device answers with these SOP Which SOP parameter causes issue?

A. The calling device did not offer a ptime value

B. The media stream is set to send only

C. The payload for G.711ulaw must be 18.

D. The RTP port is set to 0.

Answer: D

Explanation:

The RTP port is used to send and receive media packets during a call. If the RTP port is set to 0, the called device will not be able to send or receive media packets, and the call will fail.

The other options are not correct because:

A) The calling device did not offer a ptime value: The ptime value is used to specify the amount of time between each media packet. If the calling device does not offer a ptime value, the called device will use the default value of 20 milliseconds.

B) The media stream is set to sendonly: The media stream is set to sendonly when the called device is only able to send media packets, and not receive them. This is not a problem, and the call will still succeed.

C) The payload for G.711ulaw must be 18: The payload for G.711ulaw is the type of media packet that is used. The payload must be set to 18 for G.711ulaw, but this is not a problem, and the call will still succeed.

Question: 262

Which wildcard must an engineer configure to match a whole domain in SIP route patterns?

A. *
B. @
C. !
D. .

Answer: A

Explanation:

The asterisk (*) wildcard is used to match any sequence of characters, including an empty sequence. Therefore, it can be used to match any domain name in a SIP Route Pattern.
The other options are not correct because:
B) @: The @ symbol is used to separate the user name from the domain name in an email address.
C) !: The ! symbol is used to negate a character class.
D) .: The . symbol is used to match any single character.

Question: 263

An administrator is asked to implement toll fraud prevention in Cisco UCM, specifically to restrict off- net to off-net call transfers. How is this implemented?

A. Enforce ad-hoc conference restrictions.
B. Set the appropriate service parameter.
C. Implement time-of-day routing.
D. Use the correct route filters.

Answer: B

Explanation:

To restrict off-net to off-net call transfers, an administrator can set the "Block Offnet to Offnet Transfer" service parameter to "On". This will prevent users from transferring calls from one external number to another external number.
The other options are not correct because:
A) Enforce ad-hoc conference restrictions: This will prevent users from creating ad-hoc conferences, but it will not prevent them from transferring calls.
C) Implement time-of-day routing: This will allow calls to be routed to different destinations based on the time of day, but it will not prevent users from transferring calls.
D) Use the correct route filters: This will allow calls to be filtered based on the destination, but it will not prevent users from transferring calls.

Question: 264

An administrator must implement toll fraud prevention on Cisco UCM using these parameters:

- Enable Forced Authorization Code 112211.

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

How must the route pattern be implemented?

A. Pattern = 1122113.8005551212

B. Pattern = 8005551212.1122113

C. Pattern = 8005xxxxxx

D. Pattern = 3.800xxxxxxx

Answer: A

Explanation:

To implement toll fraud prevention on Cisco UCM, an administrator can use the following parameters:
Enable Forced Authorization Code 112211.
Set an authorization level of 3 for the route pattern 8005551212.
Require no access code to dial 10-digit numbers.
The route pattern must be implemented as follows:
Pattern = 1122113.8005551212
This will require users to enter the authorization code 112211 followed by the number 8005551212 to dial this number.
The authorization level of 3 will prevent users from transferring calls to this number.

Question: 265

What is required when deploying co-resident VMs by using Cisco UCM?

A. Provide a guaranteed bandwidth of 10 Mbps.
B. Deploy the VMs to a server running Cisco UCM.

C. Avoid hardware oversubscription.
D. Ensure that applications will perform QoS.

Answer: C

Explanation:

When deploying co-resident VMs by using Cisco UCM, it is important to avoid hardware oversubscription. This means that you should not assign more resources to the VMs than the physical hardware can provide. For example, if you have a server with 16 CPU cores, you should not assign more than 16 CPU cores to the VMs.

If you oversubscribe the hardware, the VMs will not be able to get the resources they need to run properly. This can lead to performance problems and even outages.

To avoid hardware oversubscription, you should carefully plan your VM deployments. You should also monitor the performance of the VMs to make sure that they are not overusing the resources.

Here are some additional tips for deploying co-resident VMs by using Cisco UCM:

Use a virtualization platform that supports Cisco UCM.

Make sure that the VMs have the correct operating system and software installed.

Configure the VMs to use the correct network settings.

Monitor the performance of the VMs to make sure that they are running properly.

Question: 266

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

A. configure single route pattern for international calls

B. set up all international route patterns to 0.!

C. reduce the T302 timer to less than 4 seconds

D. create a second route pattern followed by the # wildcard

Answer: D

Explanation:

When building a variable-length route pattern, you need to create a second route pattern followed by the # wildcard. This will allow the user to indicate the end of the number by dialing #. For example, if you want to create a route pattern for international calls, you would create a route pattern like this:

9.011!#

This route pattern will match any number that starts with 9.011, followed by any number of digits, and then ends with #.

The other options are incorrect because:

Configuring a single route pattern for international calls will not allow the user to indicate the end of the number.

Setting up all international route patterns to 0.! will not allow the user to indicate the end of the number.

Reducing the T302 timer to less than 4 seconds will not allow the user to indicate the end of the number.

Question: 267

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

A) Balanced mode provides user load balancing and user failover in the event of an outage. Active/standby mode provides an always on standbynode in the event of an outage, and it also provides load balancing.

B) Balanced mode provides user load balancing and user failover only for manually generated failovers. Active/standby mode provides anunconfigured standby node in the event of an outage, but it does not provide load balancing.

C) Balanced mode provides user load balancing and user failover in the event of an outage.

Active/standby mode provides an always on standbynode in the event of an outage, but it does not provide load balancing.

D) Balanced mode does not provide user load balancing, but it provides user failover in the event of an outage.

Active/standby mode provides analways on standby node in the event of an outage, but it does not provide load balancing.

Answer: C

Explanation:

Balanced mode provides user load balancing and user failover in the event of an outage.

Active/standby mode provides an always on standby node in the event of an outage, but it does not provide load balancing.

Here is a more detailed explanation of the two modes:

Balanced mode: In balanced mode, the IM and Presence Service nodes are configured to work together to provide high availability. The nodes are configured in a redundancy group, and the system automatically balances the load of users across the nodes in the group. If one of the nodes fails, the system automatically fails over the users to the other nodes in the group.

Active/standby mode: In active/standby mode, one of the IM and Presence Service nodes is designated as the active node, and the other nodes are designated as standby nodes. The active node handles all of the user traffic, and the standby nodes are only used if the active node fails. If the active node fails, the system automatically fails over to one of the standby nodes.

Question: 268

What is the function of the Cisco Unity Connection Call Handler?

A. routes calls to a user based on caller input
B. queues calls
C. allows customized scripts for IVR capabilities
D. searches a list of extensions until the call is answered

Answer: A

Explanation:

A Cisco Unity Connection Call Handler is a software application that answers calls, plays greetings, and routes calls to users based on caller input. Call handlers can be used to create automated attendants, voice menus, and other interactive voice response (IVR) applications.

Call handlers are created and managed using the Cisco Unity Connection Administration interface.

When creating a call handler, you can specify a variety of settings, including the greeting that is played, the caller input options that are available, and the destination that calls are routed to. Call handlers are a powerful tool that can be used to create a variety of IVR applications. By using call handlers, you can improve the efficiency of your organization's communications and provide a better experience for your callers.

Here are some additional tips for using call handlers:

Use call handlers to create automated attendants that can answer calls and route them to the appropriate person or department.

Use call handlers to create voice menus that can provide callers with information or options.

Use call handlers to create interactive voice response (IVR) applications that can collect information from callers and process their requests.

Question: 269

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

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

Answer: D

Explanation:

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

Question: 270

Where is urgent priority enabled to bypass the T302 timer?

A. route partition
B. transformation pattern
C. directory number
D. CTI port

Answer: C

Explanation:

Urgent priority is enabled on the directory number configuration page. This allows the call to be

routed at once to the fully qualified DN without any necessity to wait for inter-digit-timeout. If the Urgent Priority checkbox is disabled and you have overlap patterns configured, then CUCM waits for the user to dial further digits.

The other options are incorrect because:

Route partitions are used to group route patterns and route lists.

Transformation patterns are used to convert dialed digits into a different format.

CTI ports are used to connect Cisco Unified Communications Manager to third-party applications.

https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications- manager-callmanager/200477-Urgent-Priority-Configuration-on-Directo.html

Question: 271

Which configuration on Cisco UCM is required for SIP MWl to work?

A. Assign an MWl extension on the mailbox.
B. The line partition must be inside the inbound CSS assigned to the CUC SIP trunk.
C. The line partition must be inside the rerouting CSS assigned to the Cisco Unity Connection SIP trunk.
D. Set the"Enable message waiting indicator" on the port group.

Answer: B

Explanation:

The line partition must be inside the inbound CSS assigned to the CUC SIP trunk. This ensures that the SIP MWI messages are sent to the correct destination.

The other options are incorrect because:

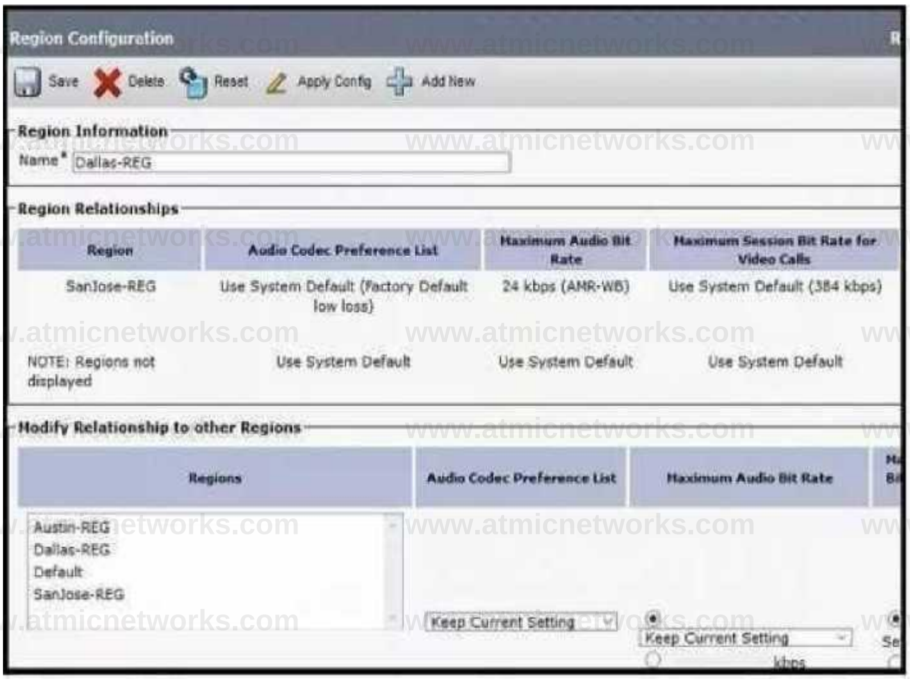
Assigning an MWI extension on the mailbox is not required for SIP MWI to work.

The line partition does not need to be inside the rerouting CSS assigned to the Cisco Unity Connection SIP trunk.

Setting the "Enable message waiting indicator" on the port group is not required for SIP MWI to work.

Question: 272

Refer to the exhibit.



Which codec should an engineer select for a call made between "Dallas-REG' & "Austin-REG'

A. MP4A-LATM

B. G.711

C. OPUS

D. G.729

Answer: D

Explanation:

The codec preference list for the "Dallas-REG" region is "Factory Default low loss". This list includes the following codecs in order of preference:

G.729

G.711

OPUS

MP4A-LATM

The codec preference list for the "Austin-REG" region is "Factory Default low loss". This list includes the following codecs in order of preference:

G.729

G.711

OPUS
MP4A-LATM

Since both regions have the same codec preference list, the codec that will be used for a call made between "Dallas-REG" and "Austin-REG" is G.729.

G.729 is a narrowband speech codec that was developed by the ITU-T in 1988. It is a low-bitrate codec that provides good quality speech at a bitrate of 8 kbps. G.729 is widely used in VoIP applications and is the default codec for many VoIP systems.

G.711 is a wideband speech codec that was developed by the ITU-T in 1972. It is a high-bitrate codec that provides excellent quality speech at a bitrate of 64 kbps. G.711 is not as widely used as G.729 due to its high bitrate requirements.

OPUS is a lossy audio codec that was developed by the IETF in 2012. It is a low-bitrate codec that provides good quality speech at a bitrate of 6 kbps. OPUS is widely used in VoIP applications and is the default codec for many VoIP systems.

MP4A-LATM is a lossy audio codec that was developed by the IETF in 1999. It is a high-bitrate codec that provides excellent quality speech at a bitrate of 24 kbps. MP4A-LATM is not as widely used as G.729 or OPUS due to its high bitrate requirements.

Question: 273

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

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

Answer: AC

Explanation:

The Cisco Expressway Series provides the following functions:

Voice and video transcoding
Interworking of SIP and H.323
Firewall traversal
Session border controller (SBC) functionality
Endpoint registration
Call admission control (CAC)
Quality of service (QoS)
Security

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

Question: 274

Cisco Prime Collaboration Manager is using Cisco Discovery Protocol to discover devices in an automated setup. What must be used to get alerts and faults via traps from devices at regular polling intervals?

A. HTTPS
B. CDP
C. SNMP
D. XML

Answer: C

Explanation:

Question: 275

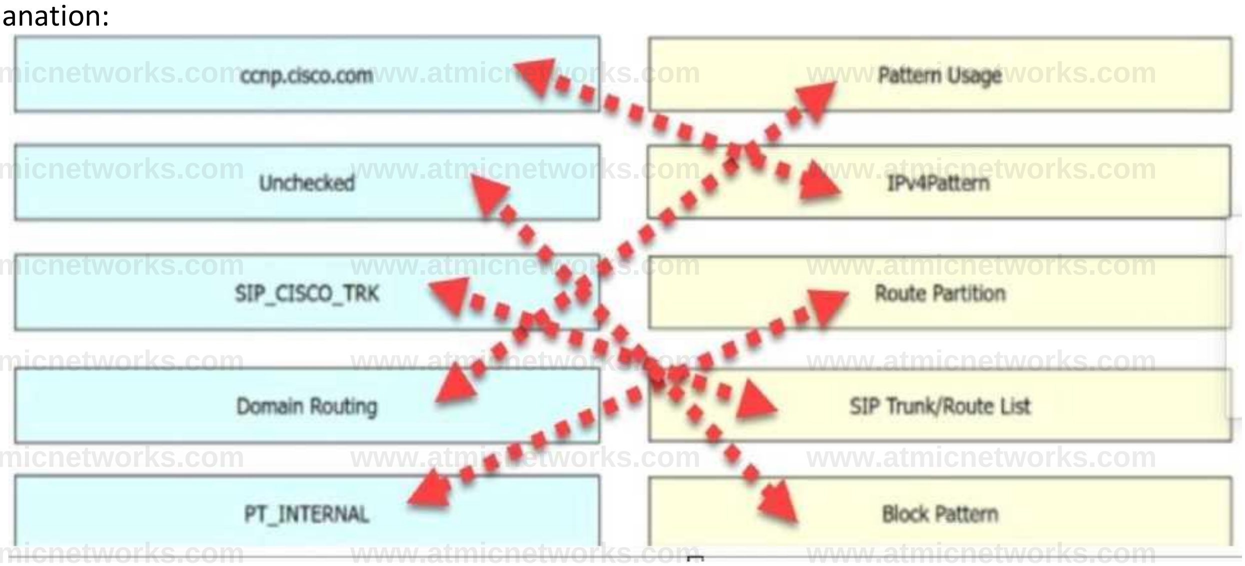
DRAG DROP

In Cisco UCM, an engineer must create a new SIP route pattern to allow ccnp.cisco com for all internal callers The internal callers have a calling search space named CSSJNTERNAL that includes the partition named PTJNTERNAL If allowed, the calls must be routed to a SIP trunk named SIP_CISCO_TRK. Drag and drop the values from the left onto the corresponding fields on the right.

ccnp.cisco.com	Pattern Usage
Unchecked	IPv4Pattern
SIP_CISCO_TRK	Route Partition
Domain Routing	SIP Trunk/Route List
PT_INTERNAL	Block Pattern

Answer:

Explanation:



Question: 276

How does Cisco define jitter?

A. voice packet loss that is greater than 1 %
B. excessive network latency
C. varying delay of received voice packets
D. varying delay of sent voice packets

Answer: C

Question: 277

What is the maximum one-way packet loss and jitter for voice before quality issues arise?

A. < 10 percent packet loss and < 10 ms jitter
B. < 1 percent packet loss and < 30 ms jitter
C. < 9 percent packet loss and < 30 ms jitter
D. < 3 percent packet loss and < 60 ms jitter

Answer: B

Question: 278

What will Differserv provide?

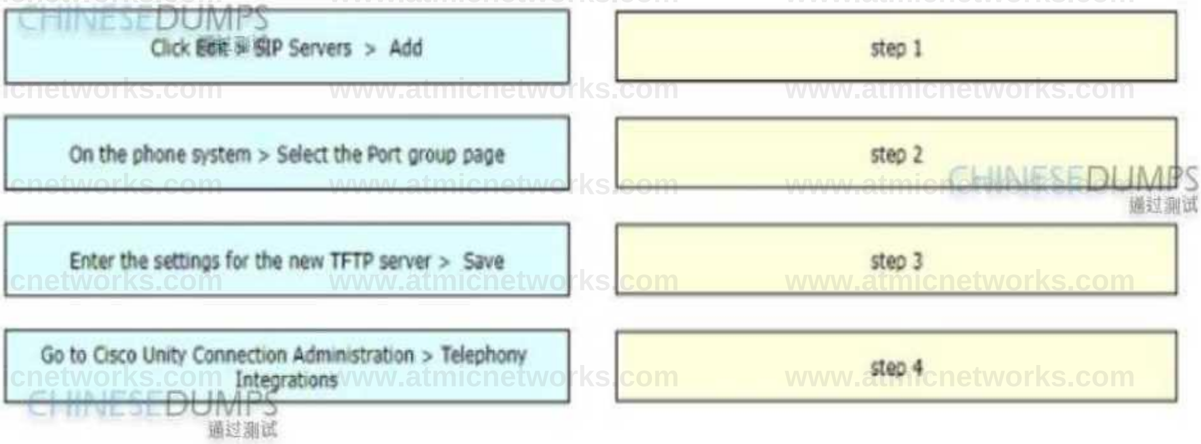
A. traffic classification
B. scheduling technique
C. service differentiation
D. queuing mechanism

Answer: A

Question: 279

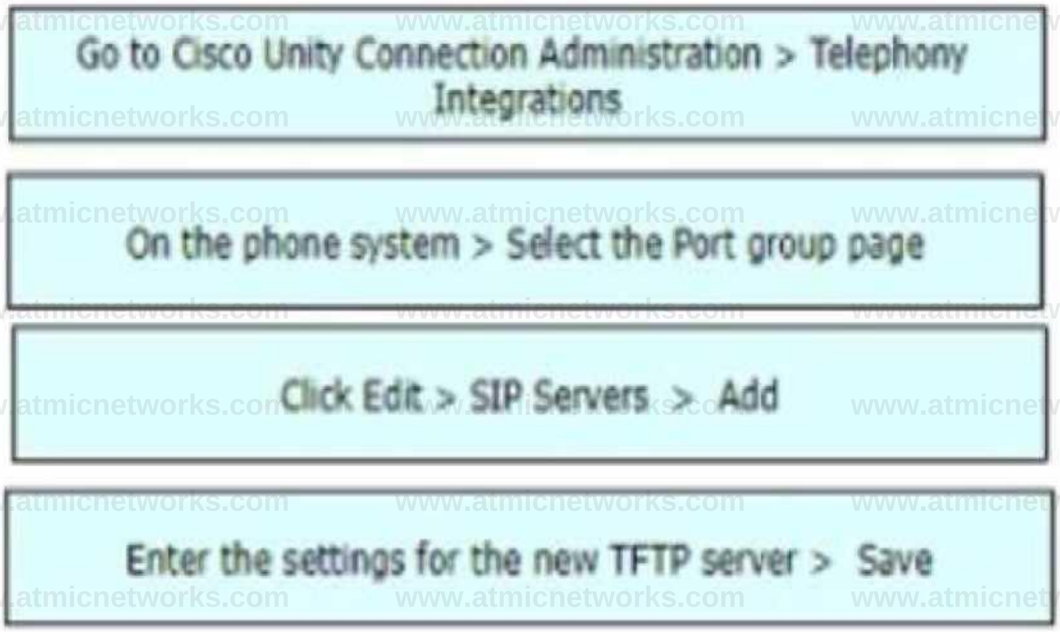
DRAG DROP

An engineer is working on a SIP integration between Cisco UCM and Cisco Unity Connection and must add an additional TFTP server Drag and drop the configuration steps from the left into the correct order on the right.



Answer:

Explanation:



Question: 280

Location-based call admission control is used in Cisco UCM to control the amount of bandwidth allowed between predefined locations. What must be considered when calculating the number of calls that can be made?

A. A limit of 900 kb is the total amount of bandwidth used for the sum of calls. The use of compression does not impact the amount of calls because only the codec definition is considered when calculating the CAC in Cisco UCM

B. A limit of 900 kb is the total amount of actual used RTP traffic so the oversubscription can be done for multiple calls adding its maximum RTP stream requirements.

C. A limit of 900kb is the total amount of bandwidth used for the sum of the media and signal in the calls. The system allows for an actual of 20% overhead of IP packet header. This is to compare the use of ISDN lines with no IP overhead.

D. A limit of 900 kb is the total amount of bandwidth used for the sum of calls of the defined codec including headers and media traffic

Answer: C

Question: 281

Where does an administrator establish the trust boundary on a LAN?

A. core router
B. voice VLAN
C. access switch
D. distribution switch

Answer: C

Question: 282

An engineer must deploy the Cisco Webex app to a Virtual Desktop Infrastructure (VDI) environment and support full-featured meetings. Which action must the engineer take to meet these requirements?

A. Configure the VDI thin client and install the Webex app
B. From Control Hub. configure VDI optimization for the Webex app
C. Install three separate VDI plugins on the VDI thin client
D. Install two separate VDI plugins on the VDI thin client

Answer: D

Question: 283

A collaboration engineer must optimize the dial plan within Cisco UCM. There are multiple remote sites. And each site has its own route patterns and local gateways. What should the engineer do on the cisco UCM to optimize the dial plan?

A. Configure a Standard Local Route Group to use a single route pattern for all calls within the cluster.
B. Create a centralized dial plan with a Cisco UCM Session Management Edition cluster or a Cisco gatekeeper.
C. Implement ILS with GDPR so the dial plan can dynamically replicate across clusters.
D. Leverage Cisco UCM Express as SRST so the phones can have more features at each remote site

Answer: C

Question: 284

What is the cisco UCM service parameter default value for DSCP for Telepresence calls

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

Answer: A

Question: 285

Which data security feature is included with the Pro Pack for Cisco Webex Control Hub?

A. encryption in transit
B. end-to-end encryption for meetings
C. Hybrid Data Security end-to-end
D. encryption of content

Answer: C

Question: 286

Which issue cause slips on a PRI?

A. incorrect dock source
B. incorrectly configured time zone
C. change in the line code
D. incorrect encapsulation

Answer: A

Question: 287

Which Cisco UCM service pararmeter is enabled to disconnect a multiparty call when the call initiator hangs up?

A. H.225 Block Setup Destination
B. Drop Ad Hoc Conference
C. Enterprise Feature Access Code for Conference
D. Block OffNet To OffNet Transfer

Answer: B

Question: 288

Which SNMP service must be active manually after Cisco UCM is installed?

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

Answer: A

Question: 289

Refer to the exhibit.

```
!
voice service voip
 ip address trusted list
 ipv4 172.19.245.1
 supplementary-service media-renegotiate
 sip
 registrar server expires max 120 min 120
!
!
dial-peer voice 1 voip
 destination-pattern 4422...
 session protocol sipv2
 session target ipv4:192.168.10.10
 incoming called-number 4433...
 direct-inward-dial
 dtmf-relay sip-notify
 codec g711ulaw
!
dial-peer voice 100 pots
 destination-pattern 91...
 incoming called-number 2...
 forward-digits 4
!
```

Refer to the exhibit. An engineer must implement toll fraud prevention on a Cisco UCM cluster. Only the SIP protocol must be allowed for connections passing through Cisco Unified Border Element. What must be configured?

```
voice service voip
 session protocol sipv2
```

```
voice service voip
 allow-connections sip to sip
```

```
dial-peer voice 100 pots
 session protocol sipv2
```

```
dial-peer voice 100 pots
 allow-connections sip to sip
```

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

Answer: B

Question: 290

Which Cisco Unity Connection call management element can be used to answer calls, provide callers with a list of menu options, and transfer calls?

A. directory handler B. call routing table C. interview handler D. call handler

Answer: D

Question: 291

During a conference call, lhe CTO of a company adds an external user to the call. The CTO leaves at the end of the call, but the external user stays on and dials out to a premium rate number. Several weeks later, the CTO receives an unexpectedly high phone bill. The CTO decides to Implement an Ad Hoc conference restriction. Which parameters must be adjusted?

A. Drop Ad Hoc Conference
B. Limit Ad Hoc Conference
C. Restrict Ad Hoc Conference
D. Monitor Ad Hoc Conference

Answer: A

Question: 292

What is a benefit of Easy/QoS?

A. It configures QoS on the network.
B. It provisons QoS for Layer 3 devices automatically.
C. It expresses business relevance for applications.

D. It allows single command QoS deployment.

Answer: A

Question: 293

What is the most important consideration when designing a QoS voice framework?

A. band width calculations

B. codec selection

C. speed propagation

D. Layer 2 overhead consumption

Answer: B

Question: 294

Which set of four functions are performed by Cisco Unified Border Element?

A. session control, security, interworking, and demarcation
B. PRI connections, security, faxing, and border enforcement
C. session control, meet-me conferencing, interworking. and demarcation
D. session control, security. SIP. and H.323

Answer: A

Question: 295

Which TCP port is used by default for SIP OAuth on a Cisco UCM server when using Mobile and Remote Access?

A. 5060
B. 5061
C. 5090
D. 5091

Answer: D

Question: 296

DRAG DROP

An engineer must add a new user account and mailbox in Cisco Unity Connection. Drag and drop the configuration steps from the left into the correct order on the right.

Answer:

Question: 297

```
admin:utils diagnose test

Log file: platform/log/diagl.log

Starting diagnostic test(s)

test -    disk-space              : Passed (available; 6463 MB, used:        12681 MB)
skip -    disk_files              : This module must be run directly        and off    hours
test -    service-manager         : Passed
test -    toecat                  : Passed
test -    t oecat deadlocks       : Passed
test -    tomcat_keystore         : Passed
test -    toecateonnec t ors      ; Passed
test -    toncatthreads           : Passed
test -    tomcat-memory           : Passed
test -    tomcat_sessions         : Passed
skip -    tdmcat_heapdump         ; This module must be run directly        and off    hours
test -    validate_network        : Passed
test -    raid                    : Passed
test -    systeminfo              : Passed (Collected system informat      ion in     diagnostic log)
test •    ntp_reachability        : Passed
test *    ntp_clock_drjft         : Passed
test -    ntpstratum              : Failed
The reference NTP server          is a stratum 5 clock.
```

An engineer is trying to add a new subscriber to the cluster of Cisco UCM. and the synchronization is failing. The engineer obtained the output from the publisher and noticed the error Based on the output, what is the issue?

A. The NTP server is running out of space in the hard drive, which prevents the cluster service from starting.
B. The NTP service is pointing to the wrong publisher.
C. The NTP server for Cisco UCM must be stratum 6 or higher.
D. The NTP server must be replaced by another server with NTP stratum 1, 2, or 3 running version NTPv4.

Answer: D

Question: 298

What requires a Cisco IOS XE transcoder?

A. DTMF conversion
B. SIP early offer
C. transrating
D. call queuing

Answer: A

Question: 299

```
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=fmtp:101 0-15
 a=trafficclass:conversational.audio.immersive.aq:admitted
```

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

A. G722/8000
B. Telephone-event/8000
C. G7221/46000
D. PCMA/8000

Answer: C

Question: 300

Which DSCP PHB classification must be used to configure QoS for voice on a high-speed network?

A. EF
B. CS4
C. AF43
D. 43

Answer: A

Question: 301

What is a function of a call handler in Cisco Unity Connection?

A. It collects information from callers by playing a series of questions and recording the answers.

B. It controls outgoing calls by allowing an administrator to specify the numbers that Cisco Unity Connection dials to transfer calls, notify users of messages, and deliver faxes.

C. It provides access to a corporate directory by playing an audio list that users and outside callers use to reach users and leave messages.

D. It answers cans, take messages, and provide menus of options.

Answer: D

Question: 302

A collaboration engineer inherits the responsibilities of another engineer. The previous engineer implemented a SIP integration between Cisco Unity Connection and Cisco UCM. Users can leave messages, but the Message Waiting Indicator is unconfigured. Which two actions must the engineer perform to enable MWI? (Choose two.)

A. From Cisco UCM, select Advanced Features, select Voice Mail, select Message Waiting, and configure the message waiting numbers.

B. From Cisco UCM. open the SIP Trunk Security Profile Information menu and select Accept Unsolicited Notification.

C. From Message Waiting Indicator Settings in Unity Connection, enable MWI for each user.

D. From Port Group in Unity Connection, select Enable Message Waiting Indicators.

E. From Message Waiting Indicators in Unity Connection, select Send Message Counts

Answer: BD

Question: 303

An administrator is troubleshooting an issue where an 8845 is not joining the correct voice VLAN configuration from a switch and they believe that LLDP is the issue Which command allows the administrator to see the packets sent from the phone to the switch?

A. show lldp status

B. show lldp traffic

C. show lldp

D. show lldp run

Answer: B

Question: 304

Which mark is used when the Attendant Console Client application and the Cisco TSP send their traffic?

A. Best Effort (DSCP=0)
B. AF23(DSCP=22)

C. Expedited ForwafOng (DSCP=U)
D. AF11 (DSCP=10)

Answer: B

Question: 305

Refer to the exhibit.

```
s=SIP Call
c=IN IP4 1.2.3.4
t=0 0
m=audio 16444 RTP/AVP 0 8 101
```

Refer to tho exhibit. A customer reports that SIP voice calls are using G.711 codec instead of G.729. The administrator checks the SIP signaling in the logs and sees the SDP shown. What will the administrator see in the logs if the right configuration change was made to resolve the issue?

A. m=audio 16444 RTP/AVP 4 101
B. m=audio 16444 RTP/AVP 15 101
C. m=audio 16444 RTP/AVP 9 101
D. m=audio 16444 RTP/AVP 18 101

Answer: D

Question: 306

The chief technology officer at a company must onboard devices to the cloud A workspace is created in Cisco Webex Control Hub for every device. What must be generated to complete the on boarding process?

A. 16-digit activation code to share with all the devices
B. 16-digit activation code for each devices
C. 15-digit activation code for each devices
D. 15-digit activation code to share with at all devices

Answer: B

Question: 307

What are two characteristics of QoS traffic policing? (Choose two.)

A. Traffic policing can be applied to outbound traffic on an interface.
B. Traffic policing can be applied to inbound traffic on an interface.

C. Traffic policing uses scheduling functions to transmit delayed packets.
D. When traffic rate equals the configured maximum rate, excess traffic is remarked
E. Traffic policing queues excess packets.

Answer: AB

Question: 308

What are two characteristics of numbers in Webex Calling? (Choose two)

A. A number marked as Main is not eligible to be moved of deleted
B. A number can only belong to one location
C. A number must be marked Active to be eligible for deletion.
D. A number can belong to multiple locations.
E. A number cannot be moved while marked Active.

Answer: AB

Question: 309

Refer to the exhibit.

```
9:41:32a TFTP Timeout : SEP0CDEA7824C58.cnf.xml
9:45:15a No Trust List Installed
9:46:51a TFTP Timeout : SEP0CDEA7824C58.cnf.xml
9:50:34a No Trust List Installed
9:52:10a TFTP Timeout : SEP0CDEA7824C58.cnf.xml
9:52:30a File Not Found : SEP0CDEA7824C58.cnf.xml
9:52:31a XMLDefault.cnf.xml
```

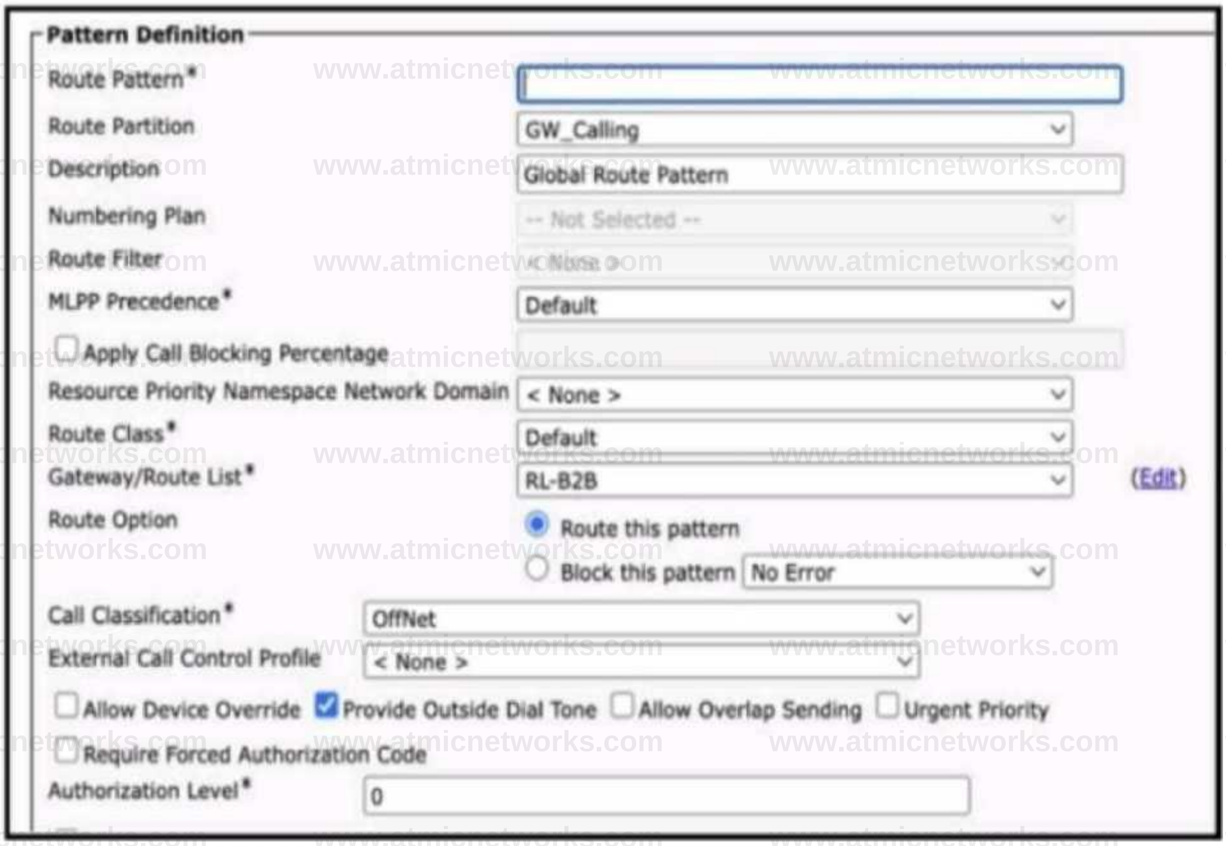
An administrator is configuring a new phone on a Cisco UCM cluster and the phone failed to register. The Cisco UCM cluster is in non-secure mode. Which action resolves the issue?

A. Update phone configuration in Cisco UCM with the correct MAC address.
B. Ensure that DHCP scope has option 150 configured.
C. Delete phone ITL file.
D. Manually configure voice VLAN on the phone.

Answer: C

Question: 310

Refer to the exhibit.



Refer to the exhibit. Which route pattern can the Cisco UCM administrator apply to the configuration to allow calls to "1306.1316.1326,13*6.13#6" only?

A. 13XX
B. 13[25-8]6
C. 13!#
D. 13[A3-9)6

Answer: D

Question: 311

An engineer must configure a new Cisco UCM translation pattern to convert 915555555555 into 1665 Which translation pattern must be used?

Translation pattern 915555555 555
Called-parly transformation discard digits None
Calling-party transformation prefix digts 5

Translation pattern" 9155555555 55
Callee-party transformation discard digits PreOot
Calling-party transformation mask 1665

Translation pattern 91555555 5555

Calted-party transformation discard digits PreAf
Calling-party transformation prefix digits 1665

Translation pattern 91555555555 5
Called-party transformation discard digits ReDot
Called-party transformation prefix digits 166

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

Answer: D

Question: 312

For Webex Calling, how are outbound transfers and forwards restricted for a particular call type?

A. together on a organization-by-organization basis

B. together on a site-by-site basis

C. independently on an organization-by-organization basis

D. independently on a site-by-site basis

Answer: C

Question: 313

Which two conditions must a user meet to provision a new device using the self-provisioning feature? (Choose two.)

A. The user must have the appropriate universal device template linked to the linked to the user profile.
B. The user must have at least one user device profile assigned.
C. The user must have a primary extension.
D. At least DNs must be assigned to the user device.
E. The user must be part of “standard CCM Super User”.

Answer: A, C

Question: 314

Refer to the exhibit.

```
Router#show isdn status

Global ISDN Switchtype = primary-ni
ISDN Serial0:23 interface
      dsl 3, interface ISDN Switchtype = primary-ni
   Layer 1 Status:
      SHUTDOWN
   Layer 2 Status:
      TEI = 0, Ces = 1, SAPI = 0, State = TEI_ASSIGNED
   Layer 3 Status:
      0 Active Layer 3 Call(s)
```

Refer to the exhibit. Users at a company located In New York cannot place calls. The New York gateway is configured with a T1 IQSDN PRI card with 24 channels. The engineer runs the show isdn status command and receives output. Which action must the engineer take to resolve the issue?

A. Change the switch type on the ISDN card to primary-5ess.
B. Increase the number of timeslots in the controller t1 interface.
C. Change the card type from T1 to E1.
D. Issue the no shut command Into the controller t1 interface.

Answer: D

Question: 315

A user reports that when receiving an inbound call on their IP Phone from the PSTN they are unable to transfer this call to another PSTN number What is the reason for this failure?

A. The incoming calling search space of the SIP trunk does not include the partition of the line in the IP phone
B. The gateway is configured with the wrong device pool
C. The service parameter related to Öffnet to Offnet Call Transfer is set to TRUE
D. The IP phone is configured with the wrong region

Answer: C

Question: 316

An administrator needs to stop the leading 9 from outbound calls on an IOS Voice Gateway and ensure that the system handles 911 emergency calls Which configuration is needed to accomplish this task?

voice translation-rule 1 rule 1 79’911/ /911/ rule 2 /*9\(.*\)/ /\0/

voice translation-rule 1 rule 1 /9?911/ /911/ rule 2 /^A9\(J\)/ /\1/

voice translation-rule 1 rule 1 /^A9\(.‘\)/ /\1/ rule 2 /9?911/ /911/

voice translation-rule 1 rule 1 /9?911/ /911/ rule 2 /^A9\(.*\)/ *III*

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

Answer: C

Question: 317

Refer to the exhibit.

Content-Type: application/sdp	Content-Type: application/sdp
Content-Length: 254	Content-Length: 254
v=0	v=0
o=151 9655 9655 IN IP4 172.16.1.1	o=151 9655 9655 IN IP4 10.10.10.1
s=-	s=-
c=IN IP4 172.16.1.1	c=IN 10.10.10.1
t=0 0	t=0 0
m=audio 50024 RTP/AVP 8 0 2 18 9	m=audio 50024 RTP/AVP 18 8 0 2
a=rtpmap:8 PCMA/8000	a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000	a=rtpmap:8 PCMA/8000
a=rtpmap:2 G726-32/8000/1	a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000	a=rtpmap:2 G726-32/8000/1
a=rtpmap:9 G722/8000	a=ptime:20
a=ptime:20	a=maxptime:80
a=maxptime:80	a=sendrecv
a=sendrecv	a=rtcp:48523
a=rtcp:50025	

Refer to the exhibit An engineer is troubleshooting codec negotiation between two Cisco phones in different branch offices that connect over a WAN Which codec is used for the call?

- G711alaw
- G726-32
- G711ulaw
- G729

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

Answer: C

Question: 318

What happens to voice packets from an 8845 Cisco IP Phone at a properly configured QoS trust boundary?

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

Answer: C

Question: 319

A company has a Cisco collaboration infrastructure that includes Cisco UCM and a Cisco Unified Border Element. The company CTO asks an engineer to improve security and implement toll fraud prevention. The engineer configures partitions and a calling search space to provide segmentation and access control Which two additional configurations are needed to block all OffNet to OffNet transfers? (Choose two.)

A. Set the Allow offNet to OffNet Transfer Cisco CallManager service parameter to false.
B. Classify an SIP devices as OffNet
C. Set the Block OffNet to OffNet Transfer CallManager service parameter to true.
D. Set the Call Classification Cisco CallManager service parameter to OffNet
E. Classify the SIP trunk to Cisco Unified Border Element as OffNet

Answer: CE

Question: 320

Which method is used to avoid toll fraud with Cisco Unified Communications Manager calls?

A. Call policy service
B. Default zone access rules
C. TOLLFRAUD_APP
D. Class of service

Answer: D

Question: 321

What is a benefit of using DNS in a Cisco collaboration infrastructure?

A. provides high availability for the collaboration infrastructure
B. provides certificate-based security
C. reduces latency for audio and video communications
D. enables the directory address book in Cisco UCM

Answer: B

Question: 322

Why is an end user's PC device not included 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 may Incorrectly tag their traffic k> be prioritized over other network traffic.
C. The end user could incorrectly tag their traffic to bypass firewalls.
D. There is no reason not to include an end user's PC device in a QoS trust boundary.

Answer: B

Question: 323

Which settings are needed to configure the SIP route pattern in Cisco Unified Communications Manager?

A. Pattern usage, IPv4 pattern, IPv6 pattern, and description
B. Pattern usage, IPv4 pattern, and SIP trunk/Route list
C. Pattern usage, IPv6 pattern, and SIP trunk/Route list
D. SIP trunk/Route list, description, and IPv4 pattern

Answer: B

Question: 324

An engineer is trying to implement a new IP phone system in a company The phones connected to the internal network are reporting a status of "unknown " There is no firewall between the Cisco UCM and the phones What is a possible reason for the unknown status?

A. Cisco Discovery Protocol is not enabled, and ping sweep is enabled
B. DNS is not enabled for the phones
C. DHCP option 150 is not set up correctly
D. The IP phone management service is not enabled under Services

Answer: C

Question: 325

Which values must be enabled to set up QoS for video conferencing on a VoIP network?

A. CS4 and DSCP 32
B. AP41 and DSCP 34
C. AF31 and DSCP 34
D. CS3 and DSCP 24

Answer: B

Question: 326

What is the purpose of bandwidth management when designing a Cisco collaboration solution?

A. It improves monitoring and troubleshooting.
B. It determines the least cost path between sites.
C. It optimizes the minimum required bandwidth.
D. It ensures the best possible user experience

Answer: D

Question: 327

Refer to the exhibit. Which codec should an engineer select for a call mode between “Dallas- REG” & “Austin-REG”?

A. MP4A-LATM
B. G.729
C. OPUS
D. G.711

Answer: B

Explanation:

Question: 328

Which two features will access layer switches provide support providing high-quality voice and taking advantage of the full voice feature set? (Choose two.)

A. Deploy RSVP to improve VoIP QoS only where it will have a positive impact on quality and functionality where there is limited bandwidth and frequent network congestion.

B. 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.

C. Map audio and video streams of video calls (AF41 andAF42) to a class-based queue with weighted random early detection.

D. Implement IP RTP header compression on the serial interface to reduce the bandwidth required per voice call on point-to-point links.

E. Use 802 1Q trucking and 802.1p for proper treatment of Layer 2 CoS packet marking on ports with phones connected.

Answer: CE

Explanation:

Question: 329

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: C, E

Explanation:

Question: 330

An engineer is configuring a Cisco IOS XE gateway to join three callers together in the same conversation. Which IOS XE media resource must the engineer configure to accomplish this goal?

A. IOS XE Hardware MTP
B. IOS XE Software MTP
C. IOS XE Hardware Transcoder
D. IOS XE Hardware Conference Bridge

Answer: D

Explanation:

Question: 331

How does an administrator change the DNS settings on Cisco UCM?

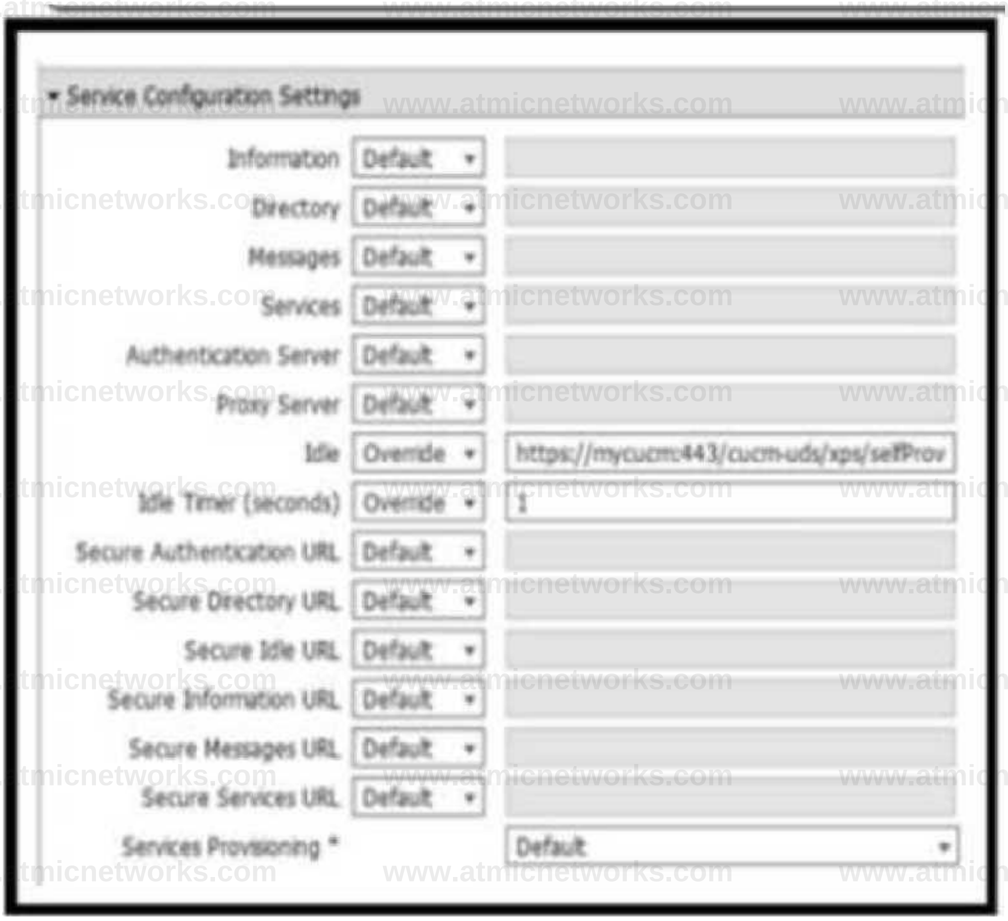
A. Go to Network > IP > Cisco Serviceability and change the DNS settings.
B. Find the DNS setting in the system parameter settings in the Network section.
C. Go to System > Network > DNS settings.
D. Open an SSH session to each node in the Cisco UCM cluster and change the DNS setting using CLI.

Answer: D

Explanation:

Question: 332

Refer to the exhibit.



Refer to the exhibit. An administrator is asked to allow new phones to use self-provisioning Interactive Voice Response. Which change to the Universal Device Template implements this feature?

A. Configure Idle URL with default.
B. Configure Authentication Server URL to the authentication server.
C. Configure the Directory URL to a valid directory.
D. Configure Secure Information URL to info URL.

Answer: A

Explanation:

Question: 333

Which command enables SIP OAuth mode on a Cisco UCM publisher node?

- `utils sip-OAuth mode activate`
- `utils sipOAuth-mode activate`
- `utils sip-OAuth-mode enable`
- `utils sipOAuth-mode enable`

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

Answer: D

Explanation:

Question: 334

An engineer must deploy the Cisco Webex app to a shared Windows Virtual Desktop Infrastructure environment in the C:\Program Files folder. Which command line argument must be run when running the installer?

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

Answer: C

Explanation:

Question: 335

An engineer must create a route pattern for a SIP trunk with the pattern named

Unity_Con_SIP_Trunk and a voicemail pilot number 8000/< None > in Cisco Unity Connection. Which value must the engineer enter for the Gateway/Route List field?

○ SIP_Trunk_Route_List

○ 8000/< None >

○ SIP_Trunk_Route_Group

○ Unity_Con_SIP_Trunk

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

Answer: D

Explanation:

Question: 336

What is the function of DiffServ?

A. It automatically identifies traffic
B. It provides reserve bandwidth by using RSVP
C. It categorizes traffic.
D. It guarantees end-to-end bandwidth for traffic.

Answer: C

Explanation:

Question: 337

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

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

Answer: B

Explanation:

Question: 338

Which DiffServ value must be used to identify and classify audio and video from Cisco Telepresence endpoints according to the Cisco dual video queue approach?

A. AF42
B. CS3
C. CS4
D. AF41

Answer: C

Explanation:

Question: 339

An engineer configures Class-Based Weighted Fair Queuing, but must still configure Low Latency Queuing to prioritize voice audio traffic and guarantee 5 Mbps of bandwidth. Which Cisco IOS command must be run in the voice class map?

```
priority 5000
```

```
priority 5
```

```
priority level 5
```

```
priority
```

A. Option A

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

Answer: A

Explanation:

Question: 340

Where would an administrator assign the system shadow location for implementing intercluster enhanced locations CAC in Cisco UCM?

A. nongatekeeper-controlled intercluster trunk
B. SIP intercluster trunk
C. H.225 trunk
D. gatekeeper-controlled inter cluster trunk

Answer: C

Explanation:

Question: 341

A customer reports that the load balancing solution is not working (or two Cisco Unified Border Element routers. The (cube1.abc.com) router should take 60% of the calls, and the (cube2 abc.com) router should take 40% of the calls. Assuming all DNS A records are present, which two SRV configurations would provide the desired outcome? (Choose two.)

o sip ixt^d^ $)INSRV2 60 5060cube1.abc.com

Q sip udp abc com 60 IN SRV 1 40 5060 cube2.abc.com

□ sip udp abc com 60 IN SRV 1 60 5060 cubel abc com

C sip udp abc com 60 IN SRV 60 1 5060 cubel abc.com

□ sip udp abc com 60 IN SRV 3 60 5060 cube2.abc.com

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

Answer: B, C

Explanation:

Question: 342

Which transport protocol does the application layer protocol SNMP use?

A. HTTP
B. XML

C. UDP
D. SIP

Answer: C

Explanation:

Question: 343

A collaboration engineer is troubleshooting issues with a T1 PRi The engineer wants to confirm if the router received an ISON Set Asynchronous Balanced Mode Extended message and responded with an Unnumbered Acknowledge frame Which Cisco OS command is used to debug ISDN Laver 2 issues?

A. debug isdn q931
B. debug isdn q921
C. debug voice q921
D. debug voip ccapi mout

Answer: B

Explanation:

Question: 344

What is a capability of the call forwarding feature in a Cisco Webex dial plan?

A. It forwards calls when a line is busy by enabling the call waiting feature.
B. It forwards calls to the voicemail of a mobile phone when there is no answer.
C. It provides the ability to forward a Webex meeting to another user.
D. It chooses a tone to be played when forwarding all calls.

Answer: D

Explanation:

Question: 345

By default. IP phones upgrade their images using the HTTP Service from the TFTP Server integrated into one or more of the call processing platforms. Which Port / Transport Protocol does this service use?

A. 8080 / TCP
B. 6970 / UDP
C. 443 /TCP
D. 6970 / TCP

Answer: D

Explanation:

Question: 346

An administrator troubleshoots a DNS issue with Cisco UCM and observes that the hostname resolution returns the wrong IP address The DNS team Informs the administrator that the A'AAA record was recently updated on the DNS server Which action must be taken on Cisco UCM to reflect the correct entries?

A. Flush the current cache using this CLl command admin utils network name-service hosts cache Invalidate
B. Delete me arp entry of the host utils network arp delete
C. Flush the current cache vising this CLl command admin:utils network name-service hosts cache validate
D. Restart the DNS server to Hush any stale entries

Answer: A

Explanation:

Question: 347

After an administrator installs the Cisco jabber Client, it fails to register with the cisco UCM server.
Which two actions should be taken to troubleshoot this problem? (Choose two.)

A. Verify the installation of the LMHOST file on the PC.
B. Verify the configuration of the DNS SRV record.
C. Verify corporate firewall settings for client connections.
D. Verify Layer 3 connectivity on the gateway.
E. Reboot the Cisco Unified Border Element Gateway.

Answer: B, C

Explanation:

Question: 348

What is a characteristic of Cisco UCM translation patterns?

A. Configure routing based on the calling party number for the next hop.
B. When there is a match, the call is routed to the specified destination in the translation pattern.
C. When a dialed number is manipulated in a translation pattern, the dialed number is written to the call detail record.
D. Never configure a partition for a translation pattern.

Answer: B

Explanation:

Question: 349

What is the purpose of Mobile and Remote Access with Cisco Jabber in the cisco UCM?

A. to allow Cisco Jabber to register with Cisco UCM without trie use of a VPN from outside the corporate network
B. to allow Cisco Jabber to register with Cisco UCM via a remote access VPN from a branch office
C. to allow Cisco Jabber to register with Cisco UCM via a remote access VPN from outside the corporate network
D. to allow Cisco Jabber to register with Cisco UCM via a site-to-srte VPN from a branch office

ANSWER: A

Question: 350

A network engineer must convert an inbound number 14085551111 when it reaches Cisco UCM and then converts it to an internal extension 3022, so it can ring to a specific desk. Which action meets this requirement?

A. Create a route group and route list and assign it to the translation pattern.
B. Create translation pattern 14085551111, called party transform mask 3022.
C. Change the partition and calling search space on the DID for the specific desk.
D. Create a route pattern 9.1XXXXXXXXXX and assign it to the trunk.

Answer: B

Explanation:

Question: 351

What an advantage of using of the SIP OAuth mode in Cisco Jabber?

A. secure signaling and media
B. authentication of SIP messages
C. reduced authentication times
D. usage of self-signed certificates

Answer: A

Explanation:

Question: 352

An engineer with troubleshoots poor voice quality on multiple calls. After looking at packet captures, the engineer notices high levels of jitter. Which two areas does the engineer check to prevent jitter? (Choose two.)

A. Voice packets are classified and marked
B. All devices use wired connections instead of wireless connections
C. The network meets bandwidth requirements

D. Cisco UBE manages voice traffic, not data traffic

E. MTP is enable on the sip trunk to cisco Unified Border Elemnet

Answer: A, C

Explanation:

Question: 353

An engineer must configure a new route pattern. When a user dials 9-555-555-5555, only the last 10 digits must be sent to the gateway. All transformations must be done at the route pattern level.

Which route pattern must the engineer use?

A.
Route pattern: 9.@
Called party transformation discard digits: PreDot

B.
Route pattern: 9.(2-9].. [2-9] ....
Called party transformation discard digits: Nine

C.
Route pattern: 9.(5-5] [5-5]...
Called party transformation discard digits: PreDot

D.
Route pattern: 9T
Called party transformation discard digits: Nine

Answer: A

Explanation:

Question: 354

How many dial patterns can a Webex Calling dial plan support?

A. 1,000
B. 2,000
C. 10,000
D. 12,000

Answer: C

Explanation:

Question: 355

What are two differences between media flow-around and media flow-through on cisco unified Border element? (Choose two.)

A. When using media flow-around, the can signaling goes through the Cisco Unified Border Element, but media is not passed through It.

B. When using media flow-through, call signaling goes through the Cisco Unified Border Element, but media does not

C. When using media flow-through, the call signaling and media are passed through the Cisco Unified Border Element

D. When using media flow-through, the call signaling goes through the Cisco Unified Border Element, but media is not passed through When using media flow-around, both call signaling and media do not go through the Cisco Unified Border Element

Answer: AC

Explanation:

Question: 356

Refer to the exhibit.

```
92195463.002 |12:41:19.104 |AppInfo  |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.1.1.1 on
port 32814 index 198 with 1554 bytes:
[887563054,NET]
INVITE sip:+12222222222@20.2.2.2:5095 SIP/2.0
Via: SIP/2.0/TCP 10.1.1.1:5095;branch=z9hG4bK3dce30168ab3a48
From: <sip:+13333333333@abc.com>;tag=119028044-fe171de1-7ea6-4a70-adc1-5c722bb7010f-266705388
To: <sip:+12222222222@20.2.2.2>
Date: Tue, 22 Dec 2020 18:41:19 GMT
Call-ID: 46877700-1ec16288-36f2729-f0607a2@10.1.1.1
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM12.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Supported: X-cisco-srtp-fallback,X-cisco-original-called
Call-Info:
<urn:x-cisco-remotecc:callinfo>;x-cisco-loc-id=e2cc7a6e-99f7-6f4b-a868-c639c0a4998f;x-cisco-loc-name=OffNet_L
oc;x-cisco-fateshare-id=cucmsme:266705387;x-cisco-video-traffic-class=MIXED
Session-ID: e6cac8f0d533d35811ee2ab119828043;remote=00000000000000000000000000000000
Cisco-Guid: 1183282944-0000065536-0009379795-0252053410
Session-Expires:  1800
P-Asserted-Identity: <sip:+13333333333@abc.com>
Remote-Party-ID: <sip:+13333333333@abc.com>;party=calling;screen=yes;privacy=off
Contact: <sip:+13333333333@10.1.1.1:5095;transport=tcp>
Max-Forwards: 67
Content-Type: application/sdp
Content-Length: 275

v=0
o=CiscoSystemsCCM-SIP 119828044 1 IN IP4 10.1.1.1
s=SIP Call
c=IN IP4 10.1.1.1
b=TIAS:256000
b=AS:272
t=0 0
m=audio 54094 RTP/AVP 18 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

]
92195856.001 |12:41:19.127 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.1.1.1 on
port 32814 index 198
[887563059,NET]
SIP/2.0 488 Not Acceptable Media
Via: SIP/2.0/TCP 10.1.1.1:5095;branch=z9hG4bK3dce30168ab3a48
From: <sip:+13333333333@abc.com>;tag=119028044-fe171de1-7ea6-4a70-adc1-5c722bb7010f-266705388
To: <sip:+12222222222@20.2.2.2>;tag=300491835-3e79dabf-55a9-4df7-b733-dd0511327a60-98163145
Date: Tue, 22 Dec 2020 18:41:19 GMT
Call-ID: 46877700-1ec16288-36f2729-f0607a2@10.1.1.1
CSeq: 101 INVITE
Allow-Events: presence
Server: Cisco-CUCM12.5
Session-ID: 00000000000000000000000000000000;remote=e6cac8f0d533d35811ee2ab119828043
Warning: 370 20.2.2.2 "Insufficient Bandwidth"
Reason: Q.850; cause=125
Remote-Party-ID: "Branch Line 1 - 20000"
<sip:+12222222222@20.2.2.2;user=phone>;party=x-cisco-original-called;privacy=off
Content-Length: 0
```

Refer to the exhibit. What is the reason for this call failure?

A. The device pool contains more call processing agents in the CMG group than tie endpoint can support
B. MRGL contains more media groups than the endpoint can support.
C. The hunt group contains more devices than the endpoint can support
D. The available location bandwidth between sites is exceeded

Answer: D

Explanation:

Question: 357

A DTMF mismatch is occurring between an MGCP gateway registered FXS port and a Cisco Unified communications Manager SIP trunk. Which media resource can be leveraged to interwork this mismatch?

A. Media Termination point
B. Conference Bridge
C. Annunciator
D. Trust relay point

Answer: A

Explanation:

Question: 358

An engineer must configure a DHCP server to save automatic bindings on a remote host that has an IP address of 172.16.1.1. The IP address range of 172.16.1.100 to 172.16.1.108 must be blocked from the DHCP server. Which set of commands must the engineer run?

```
enable
configure terminal
ip dhcp database ftp://user:password@172.16.1.1/
router-dhcp timeout 80
ip dhcp excluded-address 172.16.1.100 172.16.1.108
end
```

```
enable
configure terminal
ip dhcp database ftp://user:password@172.16.1.1/
router-dhcp timeout 80
no ip dhcp conflict logging
ip dhcp excluded 172.16.1.108 172.16.1.100
end
```

```
enable
configure terminal
no ip dhcp conflict logging ftp://
user:password@172.16.1.1/router-dhcp timeout 80
ip dhcp excluded-address 172.16.1.108 172.16.1.100
end
```

```
enable
configure terminal
ip dhcp database timeout 80 ftp://
user:password@172.16.1.1/router-dhcp
no ip dhcp conflict logging
ip dhcp excluded-address 172.16.1.108 172.16.1.100
end
```

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

Answer: A

Explanation:

Question: 359

An engineer is troubleshooting a Primary Rate Interface that fails to come up on a H.323 production voice gateway. Layer 1 and Layer 2 configurations were validated successfully Which Cisco IOS command must be run to debug the issue with the least impact on network performance?

A. debug vtsp session
B. debug ccsip messages
C. debug all

D. debug isdn q931

Answer: D

Explanation:

Question: 360

DRAG DROP

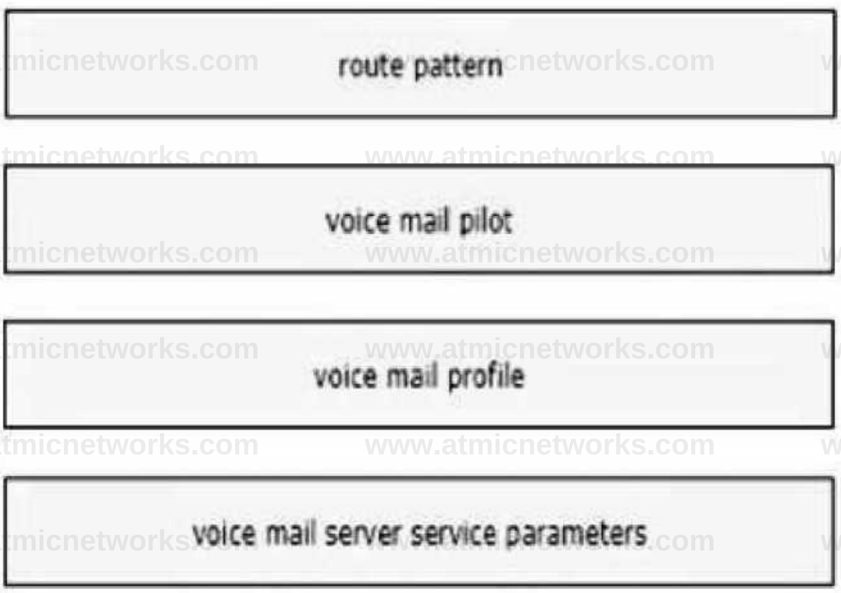
An engineer must configure a SIP integration between cisco UCM and cisco Unity Connection. These components are configured already:

- SIP 1/unk security profile
- SIP profile
- SIP trunk

Drag and drop the components from the left into the order they must be configured on the right.

Answer:

Explanation:



Summary of Steps:
Create a SIP Trunk Security Profile.
Create a SIP Profile.
Create the SIP Trunk in Cisco UCM.
Configure Route Patterns for routing calls to Unity Connection.
Configure SIP ports in Cisco Unity Connection.
Set the Voicemail Pilot Number in Cisco UCM.
Configure the Voicemail Ports in Cisco UCM.
Assign Voicemail Profile to the appropriate users.

Question: 361

What is a capability provided by the Cisco Expressway Series?

A. voice and video transcoding
B. endpoint registration
C. voice and video conferencing
D. Intercluster extension mobility

Answer: A