Cisco.350-801.v2026-04-24.q174

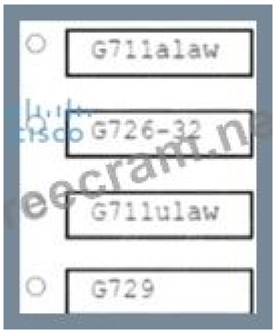
Exam Code:	350-801
Exam Name:	Implementing and Operating Cisco Collaboration Core Technologies
Certification Provider:	Cisco
Free Question Number:	174
Version:	v2026-04-24
# of views:	103
# of Questions views:	1801
https://www.freecram.net/torrent/Cisco.350-801.v2026-04-24.q174.html	

NEW QUESTION: 1

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?

```
Content-Type: application/sdp
Content-Length: 254

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

```
Content-Type: application/sdp
Content-Length: 254

v=0
o=151 9655 9655 IN IP4 10.10.10.1
s=-
c=IN 10.10.10.1
t=0 0
m=audio 50024 RTP/AVP 18 8 0 2
a=rtpmap:18 G729/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000/1
a=ptime:20
a=maxptime:80
a=sendrecv
a=rtcp:48523
```

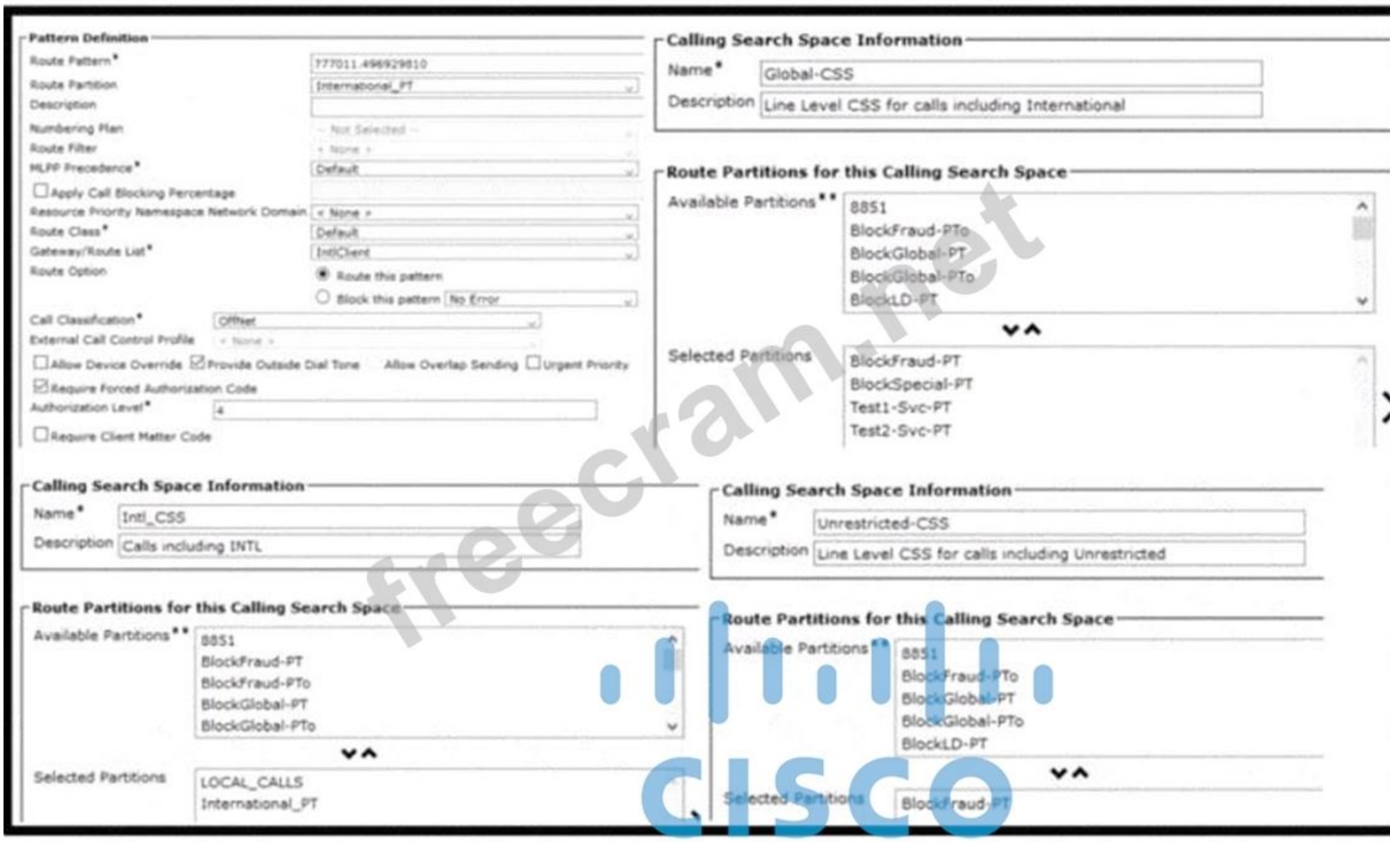


A. Option C
B. Option D
C. Option B
D. Option A
Answer: (SHOW ANSWER)

NEW QUESTION: 2
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. Wait until 10 pm on March 5th when the user is automatically removed from Cisco UCM.
B. Wait until 3 15 a.m. on March 6th for garbage collection to remove the user from Cisco UCM.
C. Wart until 10 pm on March 4th when the user is automatically removed from Cisco UCM.
D. Wait until 315am on March 5th for garbage collection to remove the user from Cisco UCM.
Answer: (SHOW ANSWER)

NEW QUESTION: 3
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-sip
B. ip dhcp pool
C. transport preferred none
D. ip domain lookup
E. ip dhcp excluded-address
Answer: (SHOW ANSWER)

NEW QUESTION: 4



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

- Pattern= \+.496929810, CSS=Unrestricted-CSS, PreDot, Prefix=777011
- Pattern= \+.777011496929810, CSS=Intl_CSS
- Pattern= \+.011496929810, CSS=Global-CSS, PreDot, Prefix=777
- Pattern= \+.496929810, CSS=Intl_CSS, PreDot, Prefix=777011

A. Option A
B. Option B
C. Option D
D. Option C
Answer: (SHOW ANSWER)

NEW QUESTION: 5
Which command in the MGCP gateway configuration defines the secondary Cisco UCM server?
A. mgcpapp
B. ccm-manager fallback-mgcp
C. mgcp call-agent
D. ccm-manager redundant-host
Answer: (SHOW ANSWER)

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

A. Voice is classified as CoS 5, and video conferencing is CoS 4.
B. Voice is classified as CoS 4, and video conferencing is CoS 5.
C. Voice and video conferencing are both classified is CoS 3.
D. Video conferencing is classified as CoS 1, and voice is CoS 2.
Answer: (SHOW ANSWER)

NEW QUESTION: 7
What are two QoS requirements for VoIP traffic?
A. Voice traffic must be marked to DSCP AF41.
B. Average one-way jitter is greater than 50 ms.
C. Voice traffic must be marked "to DSCP EF.
D. One-way latency must be no more than 200 ms.
E. Loss must be no more man 1 percent.
Answer: (SHOW ANSWER)

NEW QUESTION: 8
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: (SHOW ANSWER)

NEW QUESTION: 9

```
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: (SHOW ANSWER)

NEW QUESTION: 10

ip.addr==10.0.101.10

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

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 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.

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

C. 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.

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

Answer: (SHOW ANSWER)

NEW QUESTION: 11
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(config)# no lldp run
B. Switch(config># no lldp transmit
C. Switch# configure terminal
D. Switch# copy running-config startup-config
E. Switch(conlicj)# interface GigabitEthernet1/0/1
Answer: (SHOW ANSWER)

NEW QUESTION: 12
According to QoS guidelines, what is the packet loss for streaming video?
A. Not more than 3%
B. Not more than 1%
C. Not more than 8%
D. Not more than 5%
Answer: (SHOW ANSWER)

NEW QUESTION: 13
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: (SHOW ANSWER)
The Cisco Expressway Series provides the following functions:
Voice and video transcoding
Interworking of SIP and H.323
Firewall traversal
Session border controller (SBC) functionality
Endpoint registration
Call admission control (CAC)
Quality of service (QoS)
Security
The Cisco Expressway Series does not provide voice and video conferencing or intercluster extension mobility.

NEW QUESTION: 14

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 mp4a-latm
```

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
```

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 ilbc mode 20
```

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
 codec aacld
```

Answer: (SHOW ANSWER)

NEW QUESTION: 15

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

A. 12,000

B. 1,000

C. 10,000

D. 2,000

Answer: (SHOW ANSWER)

NEW QUESTION: 16

An engineer troubleshoots outbound can failure on an ISDN-PRI circuit. The engineer ts 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: (SHOW ANSWER)
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.

NEW QUESTION: 17
After an engineer runs the utils ntp status command on the Cisco Unified Communications Manager publisher, the stratum value is 16. Which issue can the Cisco Unified CM cluster experience?
A. The date/time group on all phones defaults to the time zone of the engineer.
B. The cluster loses access to port 124 at the firewall.
C. Database replication is not synchronized on the Unified CM nodes.
D. Unified CM sends an NTPv4 packet.
Answer: (SHOW ANSWER)

NEW QUESTION: 18

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$tEula Setting registry Eula_disable

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

E. Set the "/quiet installation argument

Answer: (SHOW ANSWER)

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.

NEW QUESTION: 19

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

A. XCP router

B. Cisco Server Recovery Manager

C. Cisco XCP Connection Manager

D. XCP Sync Agent

Answer: (SHOW ANSWER)

NEW QUESTION: 20

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: (SHOW ANSWER)

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

NEW QUESTION: 21

An administrator set up Cisco UCM LDAP Directory and already synced the users to Cisco UCM from the Active Directory, but they forgot to set up feature group templates when. doing the import. Which action resolves the issue?
A. Configure feature group templates in the Cisco UCM LDAP Directory and do a manual sync.
B. Add another LDAP Directory in Cisco UCM, configure feature group templates, and do a manual sync for that new LDAP Directory.
C. Use the Bulk Administration Tool to update those users with the feature group template.
D. Configure feature group templates in the Cisco UCM LDAP Directory and configure the resync interval.
Answer: (SHOW ANSWER)

NEW QUESTION: 22
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 (LEAVE A REPLY)
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.

NEW QUESTION: 23
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. Wait 24 hours for the garbage collector to remove the user.
B. Restart the Dirsync service after the user is deleted from LDAP directory.
C. Delete the user directly from Cisco UCM.
D. Execute a manual sync to refresh the local database and delete the end user.

Answer: (SHOW ANSWER)

NEW QUESTION: 24

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 SIP dial rules.

B. Change the common phone profile.

C. Change the phone security profile.

D. Change the SIP profile

Answer: (SHOW ANSWER)

NEW QUESTION: 25

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: (SHOW ANSWER)

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

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

NEW QUESTION: 26

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. Ensure that DHCP scope has option 150 configured.

B. Delete phone ITL file.

C. Update phone configuration in Cisco UCM with the correct MAC address.

D. Manually configure voice VLAN on the phone.
Answer: (SHOW ANSWER)

NEW QUESTION: 27
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 destination phone signals back to the Cisco UCM that it is ringing, and the Cisco UCM signals back to the source phone.
C. 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.
D. The calls have an MTP in the call path due to different codec support. The calls are subsequently split into two call legs.
Answer: (SHOW ANSWER)

NEW QUESTION: 28
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. SIP
B. H.323
C. HTTPS
D. SCCP
E. SRTP
Answer: (SHOW ANSWER)

NEW QUESTION: 29
Which two DNS records must be created to configure Service Discovery for on-premises Jabber? (Choose two.)
A. _cuplogin._tls.cisco.com pointing to the IP address of IM and Presence
B. _cisco-uds._tls.cisco.com pointing to the IP address of Cisco UCM
C. _cisco-uds.tcp.cisco.com pointing to a record of Cisco UCM
D. _xmpp.tls.cisco.com pointing to a record of IM and Presence
E. _cuplogin_tcp.cisco.com pointing to a record of IM and Presence
Answer: (SHOW ANSWER)

NEW QUESTION: 30
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 Layer 3 connectivity on the gateway.
B. Verify the configuration of the DNS SRV record.
C. Verify the installation of the LMHOST file on the PC
D. Reboot the Cisco Unified Border Element Gateway.
E. Verify corporate firewall settings for client connections.
Answer: (SHOW ANSWER)

NEW QUESTION: 31
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 can be extended to voice and video devices if the connected PCs are included
B. QoS trust boundaries can be extended to voice and video devices exclusively
C. QoS trust boundaries exclude Jabber softphone running on a PC
D. QoS trust boundaries include all the devices directly attached to the access switch ports
E. QoS trust boundaries can be extended to Jabber running on a PC
Answer: (SHOW ANSWER)

NEW QUESTION: 32
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: (SHOW ANSWER)

NEW QUESTION: 33
Why does Cisco UCM use DNS?
A. It provides certificate-based security for media
B. It provides SRV resolution to the endpoints registered
C. It resolves FQDN to IP address resolution for trunks

D. it connects endpoints to single sign-on services.
Answer: (SHOW ANSWER)

NEW QUESTION: 34
Which action must an administrator take to provide a method that plays prerecorded messages when setting up a voicemail server in a Cisco collaboration on-premises system?
A. Configure a telephony service in Cisco Unity Connection to play prerecorded messages applying this service to a number.
B. Configure a directory handler to respond to the calls in Cisco UCM.
C. Create a recording that is applied to a service voicemailbox in Cisco UCM.
D. Create a call handler template to apply to a call handler in Cisco Unity Connection.
Answer: (SHOW ANSWER)

NEW QUESTION: 35
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. Create a custom restriction table ??????????? and block it.
B. Create a custom restriction table *********** and block it.
C. Block PSTN patterns on Default Transfer, Default Outdial, and Default System Transfer.
D. Implement password complexity on voicemail boxes to prevent accounts from being compromised.
Answer: (SHOW ANSWER)

NEW QUESTION: 36
What happens to voice packets from an 8845 Cisco IP Phone at a properly configured QoS trust boundary?
A. The voice packets are not trusted, and the access layer switch reclassifies the packets.
B. Cisco UCM determines how the voice packets are classified.
C. The phone and access layer switch negotiate the classification of packets.
D. The voice packets are classified by the phone and the classification is accepted.
Answer: (SHOW ANSWER)

NEW QUESTION: 37
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: (SHOW ANSWER)
The SDP offer includes the following line of text:
M=audio 498181 RTP/AVP 0 8 97
This line of text indicates that the following codecs are available:
0: G.711ulaw
8: G.711alaw
97: iLBC
The Cisco IP Phone 7841 is registered to a Cisco Unified Communications Manager with default configuration. This means that the phone will negotiate the G.711ulaw codec.
The G.711ulaw codec is a standard codec that is used for voice communication. It is a low-bandwidth codec that provides good quality.
The iLBC codec is a newer codec that is designed for use in low-bandwidth environments. It provides good quality, but it is not as widely supported as the G.711ulaw codec.
The G.722 codec is a high-quality codec that is used for voice communication. It provides excellent quality, but it requires more bandwidth than the G.711ulaw codec.

NEW QUESTION: 38

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 = 5, Weight = 70
B. Priority = 10, Weight = 10
C. Priority = 100, Weight = 90
D. Priority = 10, Weight = 5
Answer: A (LEAVE A REPLY)

NEW QUESTION: 39
An administrator is trying to change the default LINECODE for a voice ISDN T1 PRI. Which command makes this change?
A. linecode hdb3
B. linecode b8zs
C. linecode ami
D. linecode esf
Answer: (SHOW ANSWER)

NEW QUESTION: 40

Which DSCP PHB classification must be used to configure QoS for voice on a high-speed network?
A. AF43
B. CS4
C. 43
D. EF
Answer: (SHOW ANSWER)

NEW QUESTION: 41
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 (LEAVE A REPLY)
In environments with poor WAN conditions-including packet loss, delay, and jitter-Cisco documentation consistently recommends codecs that are loss-tolerant, resilient, and designed for low- bandwidth or impaired networks. Among the available options, iLBC (internet Low Bitrate Codec) is specifically engineered to handle significant packet loss without producing severe audio distortion.
Cisco Collaboration design guides describe iLBC as a codec that uses independent encoding of each speech frame, allowing audio to remain understandable even when multiple packets are lost in transit. This is different from predictive waveform codecs like G.711 or G.729A, which depend on sequential packet flow.
When packets are lost in those codecs, audio becomes choppy or unintelligible.

NEW QUESTION: 42
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. dtmf-relay rtp-nte sip-notify
D. sip-notify dtmf-relay rtp-nte
Answer: (SHOW ANSWER)

NEW QUESTION: 43
Where does an administrator establish the trust boundary on a LAN?
A. access switch
B. core router
C. voice VLAN

D. distribution switch
Answer: (SHOW ANSWER)

NEW QUESTION: 44
What happens to voice packets from a Cisco 8845 IP phone in the QoS trust boundary?
A. The voice and access layer switch negotiate the classification of packets
B. The voice packets are not trusted, and the access layer switch reclassifies the packets.
C. The voice packets are classified by the phone, and the classification is accepted
D. Cisco UCM determines how the voice packers are classified.
Answer: (SHOW ANSWER)

NEW QUESTION: 45
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: (SHOW ANSWER)
A hybrid local gateway handles calls between the Public Switched Telephone Network (PSTN) and Webex Calling. It is commonly deployed on the customer's premises but can also be hosted by a partner. The local gateway registers with Webex Calling and handles all calls between the PSTN and Webex Calling. It gives customers the flexibility to bring their own service provider or continue using their existing provider for a smooth and effective transition to the cloud.

NEW QUESTION: 46
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 access-lists
B. show policy-map
C. show class-map interface GigabitEthernet 1/0/1
D. show policy-map interface GigabitEthernet 1/0/1
Answer: (SHOW ANSWER)

NEW QUESTION: 47
Which issue can occur if QoS is not Deployed on a Cisco Collaboration architecture across the WAN?
A. Unexpected shut-down on Cisco Unified Communications Manager
B. Packet fragmentation
C. 403 Forbidden errors on SIP calls
D. Excessive jitter
Answer: (SHOW ANSWER)

NEW QUESTION: 48

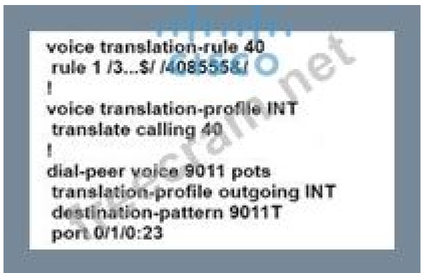
```
admin:utils diagnose test

Log file: platform/log/diag1.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 - tomcat              : Passed
test - tomcat_deadlocks    : Passed
test - tomcat_keystore     : Passed
test - tomcat_connectors   : Passed
test - tomcat_threads      : Passed
test - tomcat_memory       : Passed
test - tomcat_sessions     : Passed
skip - tomcat_heapdump     : This module must be run directly and off hours
test - validate_network    : Passed
test - raid                : Passed
test - system_info         : Passed (Collected system information in diagnostic log)
test - ntp_reachability    : Passed
test - ntp_clock_drift     : Passed
test - ntp_stratum         : 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 server for Cisco UCM must be stratum 6 or higher.
C. The NTP service is pointing to the wrong publisher.
D. The NTP server must be replaced by another server with NTP stratum 1, 2, or 3 running version NTPv4.

Answer: (SHOW ANSWER)

NEW QUESTION: 49

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

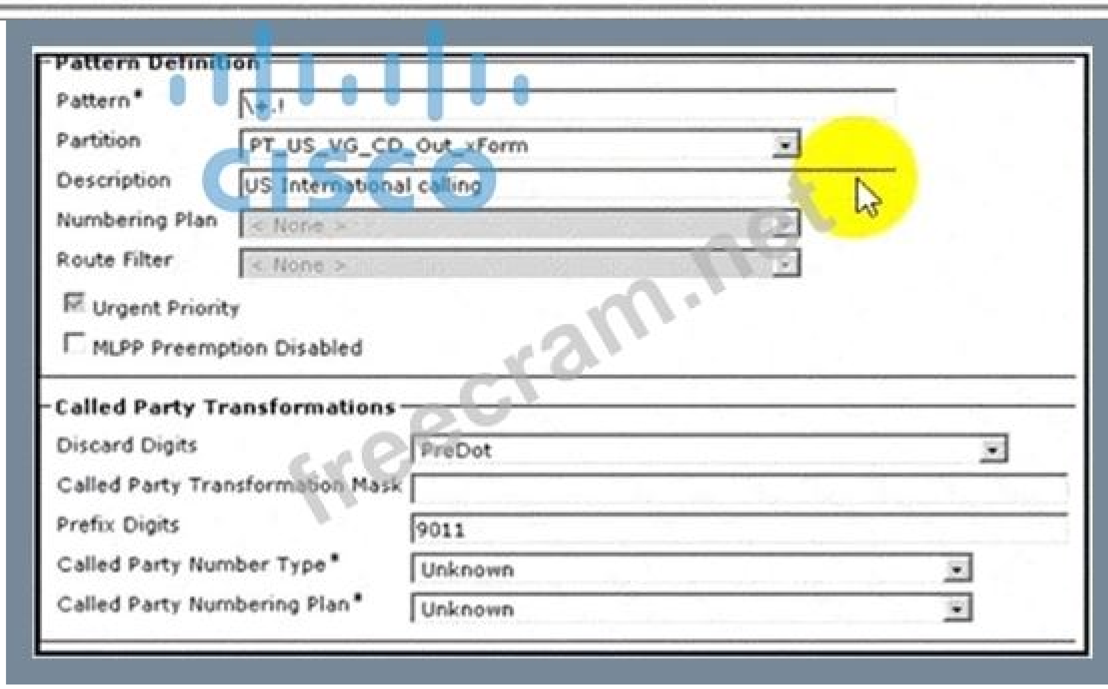


```
voice translation-rule 40
 rule 1 /3...$/ /408555&/
!
voice translation-profile INT
 translate calling 40
!
dial-peer voice 9011 pots
 translation-profile outgoing INT
 destination-pattern 9011T
 port 0/1/0:23
```

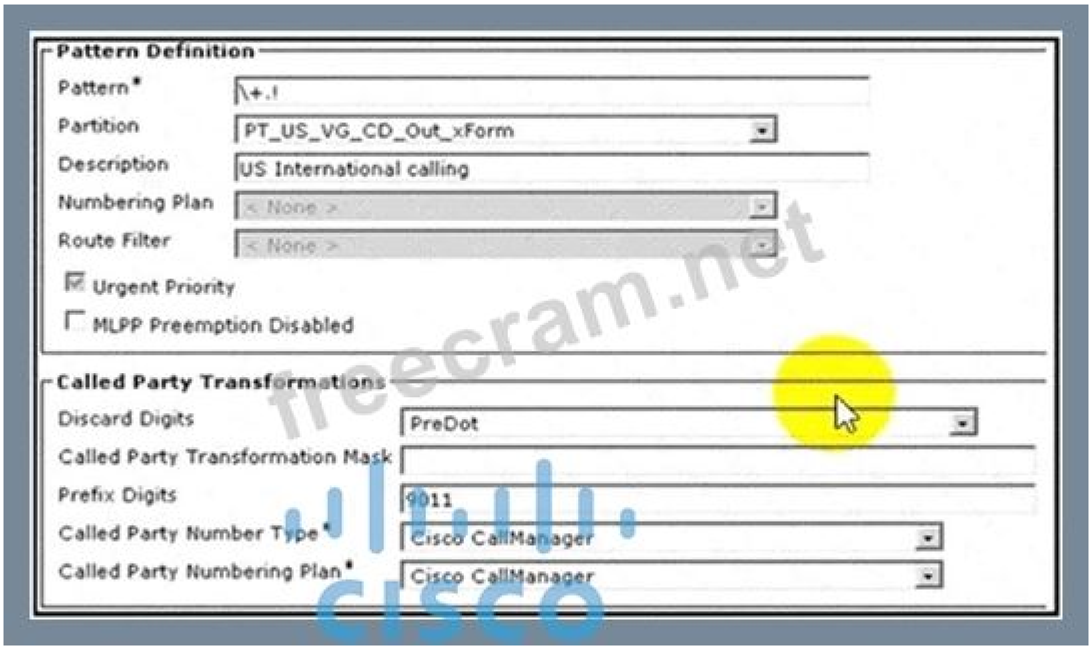
A.

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*
International
Called Party Numbering Plan*
ISDN


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*
Unknown
Called Party Numbering Plan*
Unknown

C.

D. 

Answer: (SHOW ANSWER)

NEW QUESTION: 50

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 software transcoder on Cisco UCM to support voice calls to and from both remote sites

B. configure the hardware transcoder on the site B router

C. configure the hardware transcoder on the hub location router

D. configure the hardware transcoder on the site A router

Answer: (SHOW ANSWER)

NEW QUESTION: 51

```
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 ISDN 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: (SHOW ANSWER)

NEW QUESTION: 52
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.)

- ☐ _sip._udp.abc.com 60 IN SRV 2 60 5060 cube1.abc.com
- ☐ _sip._udp.abc.com 60 IN SRV 1 40 5060 cube2.abc.com
- ☐ _sip._udp.abc.com 60 IN SRV 1 80 5060 cube1.abc.com
- ☐ _sip._udp.abc.com 60 IN SRV 60 1 5060 cube1.abc.com
- ☐ _sip._udp.abc.com 60 IN SRV 3 60 5060 cube2.abc.com

A. Option A
B. Option E
C. Option B
D. Option C
E. Option D
Answer: (SHOW ANSWER)

NEW QUESTION: 53
Which actions required for a firewall configuration on a Mobile and Remote Access through Cisco Expressway deployment?
A. 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
B. 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
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 external firewall must allow these inbound connections to Expressway: SIP: TCP 5061; HTTPS:
TCP 8443; XMPP TCP 5222; media: UDP 36002 to 59999
Answer: (SHOW ANSWER)

NEW QUESTION: 54
Which attribute contains an XMPP stanza?
A. type
B. presence
C. iq
D. message

Answer: (SHOW ANSWER)

NEW QUESTION: 55

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

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

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

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

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

Answer: (SHOW ANSWER)

NEW QUESTION: 56

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 translation pattern 14085551111, called party transform mask 3022.

B. Create a route group and route list and assign it to the translation pattern.

C. Create a route pattern 9.1XXXXXXXXXX and assign it to the trunk.

D. Change the partition and calling search space on the DID for the specific desk.

Answer: (SHOW ANSWER)

NEW QUESTION: 57

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. CS3

B. AF42

C. AF41

D. CS4

Answer: (SHOW ANSWER)

NEW QUESTION: 58

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. Implement ILS with GDPR so the dial plan can dynamically replicate across clusters.

B. Configure a Standard Local Route Group to use a single route pattern for all calls within the cluster.

C. Create a centralized dial plan with a Cisco UCM Session Management Edition cluster or a Cisco gatekeeper.

D. Leverage Cisco UCM Express as SRST so the phones can have more features at each remote site

Answer: (SHOW ANSWER)

NEW QUESTION: 59

```
!
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 B

B. Option A
C. Option D
D. Option C
Answer: (SHOW ANSWER)

NEW QUESTION: 60
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 Router
C. Cisco XCP XMPP Federation Manager
D. Cisco XCP Text Conference Manager
Answer: (SHOW ANSWER)

NEW QUESTION: 61
Refer to the exhibit. Which two codec permutations should be transcoded by this dspfarm? (Choose two.)
A. G.729r8 to G.711ulaw
B. G.728br8 to G.711alaw
C. G.729ar8 to G.711alaw
D. iLBC to G.711ulaw
E. G.722 to G.729r8
Answer: (SHOW ANSWER)

NEW QUESTION: 62
What is the purpose of the Disaster Recovery System in Cisco UCM?
A. to transparently present failover process events to users
B. to create and manage Cisco UCM backup clone servers
C. to provide data backup and restore by using an SFTP server
D. to speed up the rebuilding of a crashed Cisco UCM server
Answer: (SHOW ANSWER)

NEW QUESTION: 63

```
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 15 101
B. m=audio 16444 RTP/AVP 9 101
C. m=audio 16444 RTP/AVP 4 101
D. m=audio 16444 RTP/AVP 18 101
Answer: (SHOW ANSWER)

NEW QUESTION: 64

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. call signaling
D. voice
Answer: (SHOW ANSWER)

NEW QUESTION: 65

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 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: (SHOW ANSWER)

NEW QUESTION: 66

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

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

NEW QUESTION: 67
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 dialed string.
B. The pattern does not match the dialed string.
C. The pattern represents the smallest number of endpoints.
D. The pattern represents the largest number of endpoints.
E. The pattern matches the shortest explicit prefix.
Answer: (SHOW ANSWER)

NEW QUESTION: 68
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)#controller e1 0/0/0
2900(config-controller)#pri-timeslots 1-7
```

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

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

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

Answer: (SHOW ANSWER)

NEW QUESTION: 69

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

```
Router# configure terminal
Router(config)# interface gigabitethernet 5/1
Router config-if)# description Cube E41.228-0097
```

A. platform qos trust extend cos 5

B. platform qos trust extend

C. platform qos extend trust

D. platform qos trust extend cos 3

Answer: (SHOW ANSWER)

NEW QUESTION: 70

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

A. speed propagation

B. Layer 2 overhead consumption

C. bandwidth calculations

D. codec selection

Answer: (SHOW ANSWER)

NEW QUESTION: 71

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

A. Ipp bits

B. IP ECN bits

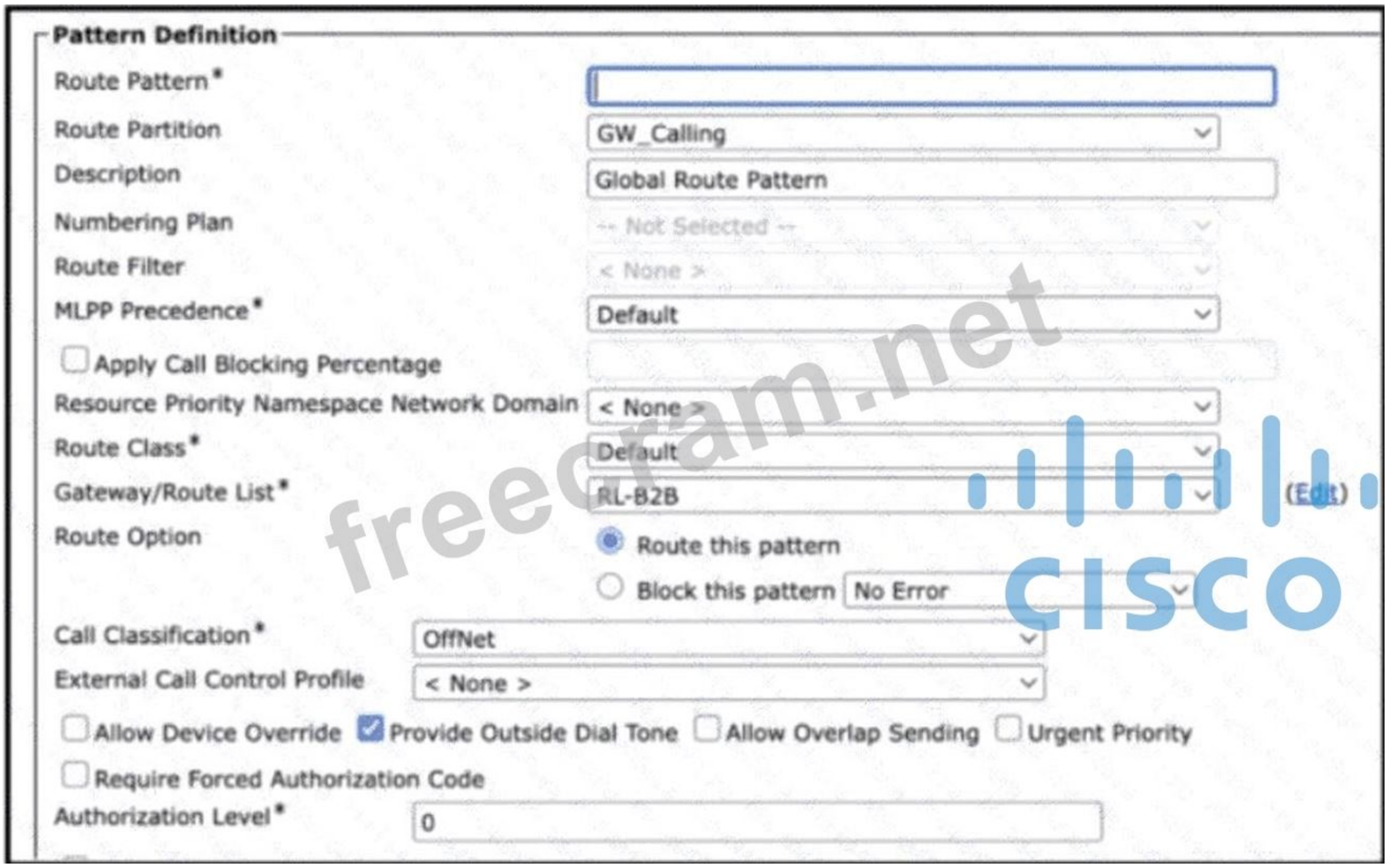
C. DCSP bits

D. 802.1 p User Priority bits

Answer: (SHOW ANSWER)

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

NEW QUESTION: 72



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. 13[25-8]6

B. 13XX

C. 13!#

D. 13[A3-9)6

Answer: (SHOW ANSWER)

NEW QUESTION: 73

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

(Choose two.)

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

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

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

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

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

Answer: (SHOW ANSWER)

NEW QUESTION: 74

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 clock source is incorrect.

D. The framing is configured incorrectly

Answer: (SHOW ANSWER)

NEW QUESTION: 75

Refer to the exhibit.

```
C:\Users\CISCO>nslookup
Default Server:  dns.example.com
Address:  192.168.100.1

> set type=SRV
> _collab-edge._tcp.example.com
Server:  dns.example.com
Address:  192.168.100.1

Non authoritative answer:
_collab-edge._tcp.example.com       SRV service location:
          priority         = 10
          weight           = 10
          port             = 8443
          svr hostname     = expe.example.com
```

You deploy Mobile and Remote Access for Jabber and discover that Jabber for Windows does not register to cisco Unified Communications Manager while outside of the office. What is a cause of this issue?

A. The DNS record type should be changed from SRV to A.

B. The DNS record should be changed from _collab-edge._tls example.com.

C. The DNS record should be created for _ cisco-uds._tcp example.com.

D. Server 4.2.2.2 is not a valid DNS server.
Answer: (SHOW ANSWER)

NEW QUESTION: 76
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. 87.2 kbps
C. 38.4 kbps
D. 55.2 kbps
Answer: (SHOW ANSWER)

NEW QUESTION: 77

```
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 ISDN Type is modified to the administrator's defined value.
B. The leading Plus is stripped from the numeric phone number.
C. The ISDN Plan is modified to the administrator's defined value.
D. Any zero is stripped from the numeric phone number.
Answer: (SHOW ANSWER)

NEW QUESTION: 78

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. The IP phone management service is not enabled under Services

B. DHCP option 150 is not set up correctly

C. DNS is not enabled for the phones

D. Cisco Discovery Protocol is not enabled, and ping sweep is enabled

Answer: (SHOW ANSWER)

NEW QUESTION: 79

Which IP precedence value maps to DSCP EF?

A. 3

B. 5

C. 0

D. 1

Answer: (SHOW ANSWER)

NEW QUESTION: 80

Refer to the Exhibit.

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

```
dspfarm profile 1 mtp
 codec g711ulaw
 maximum sessions software 50
 associate application SCCP
```

A. codec pass-through

B. associate application Cisco unified border element

C. video codec h264

D. maximum sessions hardware 50

Answer: (SHOW ANSWER)

NEW QUESTION: 81

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

Which command is required to complete the configuration?

A. Codec g729cr81

B. Codec g711ulaw

C. Codec g726r32

D. Codec g723ar63

Answer: (SHOW ANSWER)

NEW QUESTION: 82

How does an administrator change the DNS settings on Cisco UCM?
A. Find the DNS setting in the system parameter settings in the Network section.
B. Open an SSH session to each node in the Cisco UCM cluster and change the DNS setting using CLI.
C. Go to Network > IP > Cisco Serviceability and change the DNS settings.
D. Go to System > Network > DNS settings.
Answer: (SHOW ANSWER)

NEW QUESTION: 83
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 "Drop Ad Hoc Conference" to "Do not allow outside parties."
B. Set service parameter "Drop Ad Hoc Conference" to "When Conference Controller leaves."
C. Set service parameter 'Advanced Ad Hoc Conference" to FALSE.
D. Set service parameter "Advanced Ad Hoc Conference" to 2.
Answer: (SHOW ANSWER)

NEW QUESTION: 84
A high-speed network is often configured with a five-class QoS model. Which classes are used in the model?
A. call-signaling, real-time, critical data, best-effort, and drop-class
B. real-time, signaling, critical data, best-effort and drop-class
C. real-time, call-signaling, critical data, best-effort, and scavenger
D. voice, video, signaling, critical data, and best-effort
Answer: (SHOW ANSWER)

NEW QUESTION: 85
An engineer must deploy the Webex app to a Windows Virtual Desktop Infrastructure environment by using these settings:
* Thin client plugin: Installed
* Call service-enabled user: Media is optimized
* Calls on Webex app: Media is optimized
Which two command-line arguments are valid when running the installer? (Choose two.)
A. ENABLEVDI=1
B. ENABLEVDI=0
C. ALLUSERS=0
D. ENABLEVDI=2
E. ALLUSERS=1
Answer: (SHOW ANSWER)

NEW QUESTION: 86

```
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 session codec 20

B. add voice-class codec 20

C. add voice-codecs 20

D. add codec preference 20

Answer: (SHOW ANSWER)

NEW QUESTION: 87

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 g729r8
 codec preference 2 g711ulaw
 video codec h264

dial-peer voice 101 voip
 session protocol sipv2
 destination e164-pattern-map 1
 voice-class codec 10
```

B.

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

dial-peer voice 101 voip
 session protocol sipv2
 destination e164-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 e164-pattern-map 1
 voice-class codec 100
```

D.

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

dial-peer voice 101 voip
 session protocol sipv2
 destination e164-pattern-map 1
 voice-class codec 10
```

Answer: ([SHOW ANSWER])

NEW QUESTION: 88

```
controller t1 0/0/1
 pri-group timeslots 1-24
 clock source line
 linecode b8zs
 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 line
 linecode crc4
 framing hd3
```

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

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

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

Answer: B (LEAVE A REPLY)

NEW QUESTION: 89

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

A. Variable bit rate of the video stream

B. Provisioned bandwidth of the link

C. Bursly behavior of video traffic

D. Low response time between endpoints
Answer: (SHOW ANSWER)

NEW QUESTION: 90
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: (SHOW ANSWER)
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.

NEW QUESTION: 91
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 (LEAVE A REPLY)
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.

NEW QUESTION: 92
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. Disable HSRP on the access layer switch.
C. Check the RJ-65 cable.
D. Verify that the RJ-11 cable is plugged into the PC port.
E. Verify that the phone's network can access the option 150 server.
Answer: A,E (LEAVE A REPLY)

NEW QUESTION: 93
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 a packet loss of up to 1% and latency of 150 milliseconds
C. Video traffic can tolerate up to 10% packet loss and latency of 10 seconds
D. Video trafic can tolerate up to 5% packet loss and latency of 5 seconds
Answer: B (LEAVE A REPLY)

NEW QUESTION: 94
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. UDP 162
B. TCP 80
C. UDP161
D. TCP 161
Answer: (SHOW ANSWER)

NEW QUESTION: 95

```
INVITE sip:1@10.10.10.219;user=phone SIP/2.0
Via: SIP/2.0/TCP 10.10.10.84:50083;branch=z9hG4bK471df613
From: "1234 - My Phone" <sip:1234@10.10.10.219>;tag=381c1aba7a78002c558eda31-12b8af63
To: <sip:1@10.10.10.219>
Call-ID: 381c1aba-7a78000d-4ca6894a-41dd3e0f@10.10.10.84
Max-Forwards: 70
CSeq: 101 INVITE
Contact: <sip:1234@10.10.10.84:50083;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Allow-Events: kpml,dialog
Content-Type: application/sdp
Content-Length: 658

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

B. SUBSCRIBE

C. UPDATE

D. PRACK

E. REGISTER

Answer: (SHOW ANSWER)

NEW QUESTION: 96

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. voice

B. signaling

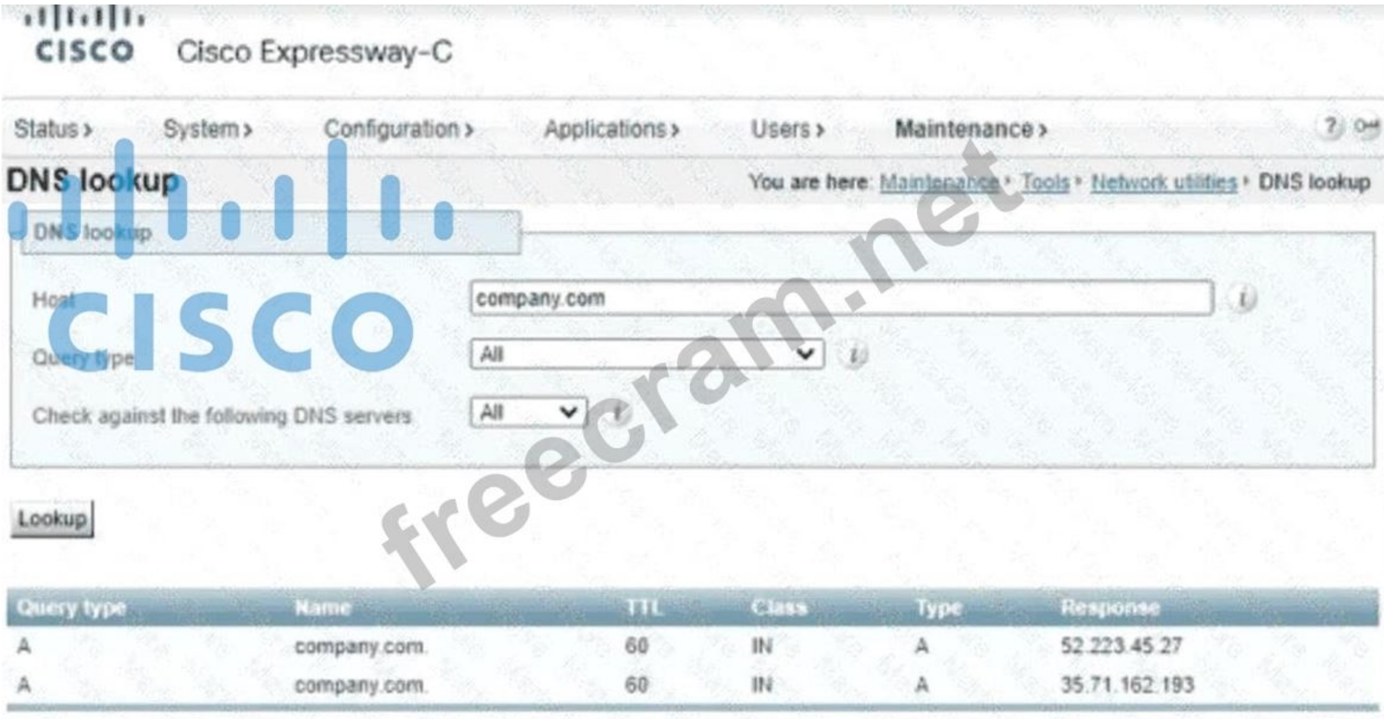
C. video conference

D. streaming video

Answer: (SHOW ANSWER)

NEW QUESTION: 97

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

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

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

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

Answer: (SHOW ANSWER)

NEW QUESTION: 98

Which layer is the optimal trust boundary for an untrusted Cisco endpoint?

A. access

B. distribution

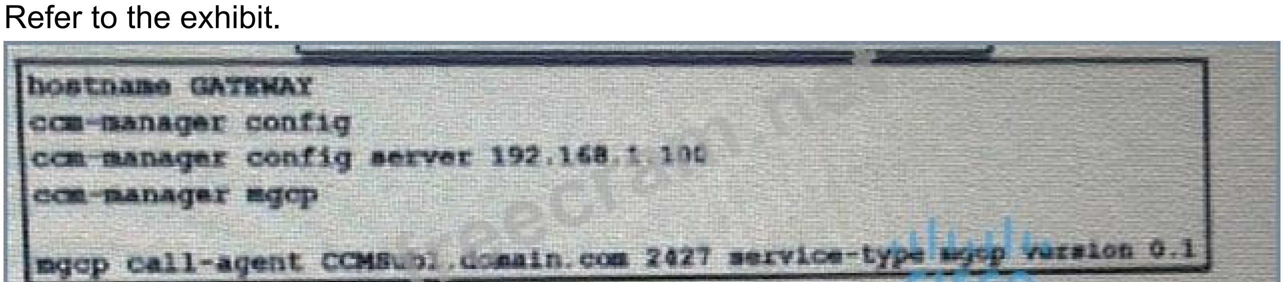
C. WAN

D. core

Answer: (SHOW ANSWER)

NEW QUESTION: 99

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) # com-manager active

C. Device(config)# ccm-manager enable

D. Device (config)# mgcp

Answer: (SHOW ANSWER)

NEW QUESTION: 100

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. 200 OK

B. SIP Register

C. TCP port 5060

D. Unsolicited NOTIFY

Answer: (SHOW ANSWER)

NEW QUESTION: 101

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. Classify the SIP trunk to Cisco Unified Border Element as OffNet

B. Set the Call Classification Cisco CallManager service parameter to OffNet

C. Classify an SIP devices as OffNet

D. Set the Allow offNet to OffNet Transfer Cisco CallManager service parameter to false.

E. Set the Block OffNet to OffNet Transfer CallManager service parameter to true.

Answer: (SHOW ANSWER)

NEW QUESTION: 102

Which DHCP option must be set up tor new phones to obtain the TFTP server IP address?

A. option 120

B. option 66

C. option 15

D. option 6

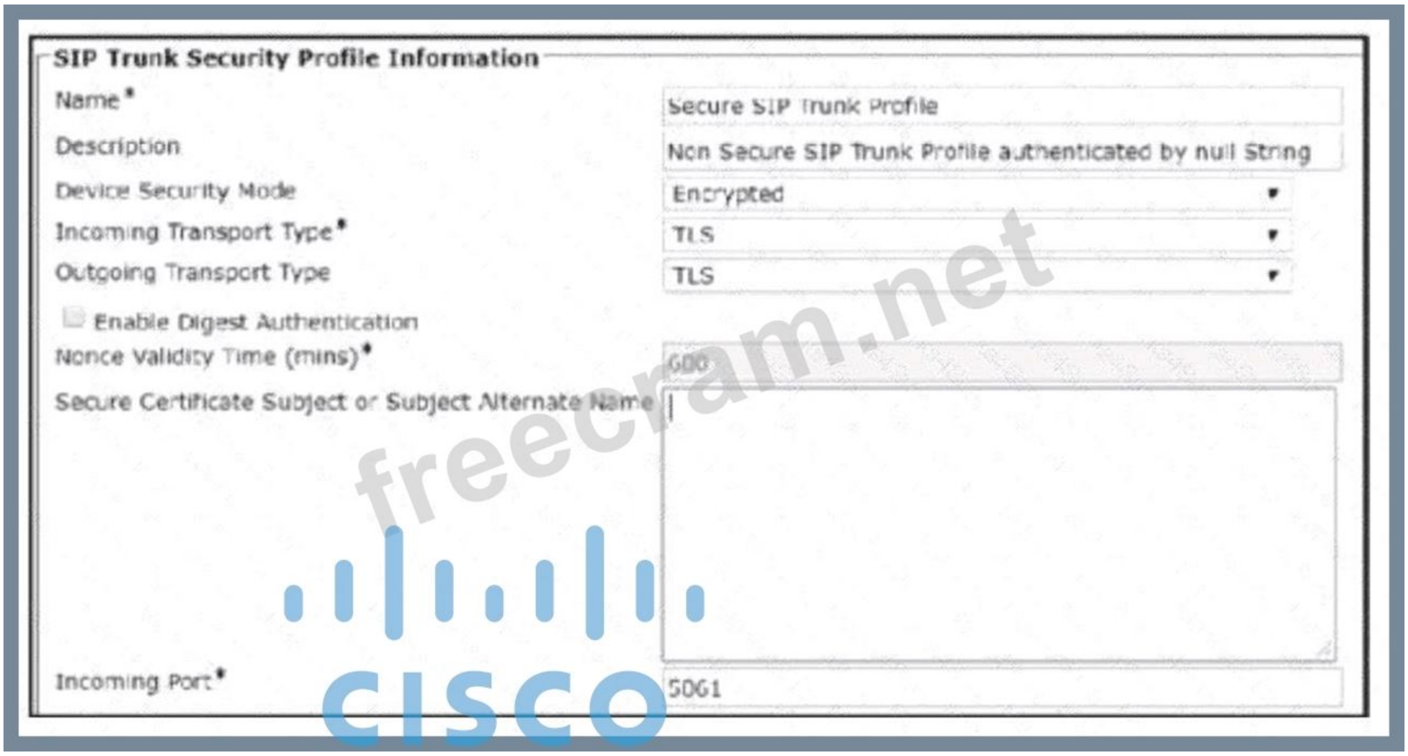
Answer: B (LEAVE A REPLY)

NEW QUESTION: 103

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: (SHOW ANSWER)

NEW QUESTION: 104
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 off-net and prohibit off-net to off-net call transfers in Cisco UCM service parameters.
B. Classify all route patterns as on-net and prohibit on-net to on-net call transfers in Cisco UCM service parameters.
C. Inbound CSS on any gateway typically should have access to internal destinations only and not PSTN destinations.
D. Inbound CSS on any gateway typically should have access to internal destinations and PSTN destinations.
E. Classify all route patterns as on-net or off-net and prohibit off-net to off-net call transfers in Cisco UCM service parameters.
Answer: (SHOW ANSWER)

NEW QUESTION: 105
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 full qualified domain name of all Cisco UCM nodes that run the CallManager service.
C. The common name of the Cisco UCM CallManager certificate.
D. The common name of the remote device certificates.
Answer: (SHOW ANSWER)

NEW QUESTION: 106
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 UCM, 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 IM and Presence Server, and Cisco Video Communication Server
D. Cisco Expressway-E Cisco Expressway-C and Cisco UCM
Answer: (SHOW ANSWER)

NEW QUESTION: 107
Which call routing pattern is used for phone numbers that are in the E.164 format?
A. \+.! Route pattern
B. \+.! Translation pattern
C. /+.! Route Pattern
D. \+1. [2-9]XX[2-9]XXXXXXX called Party Transformation pattern
Answer: (SHOW ANSWER)

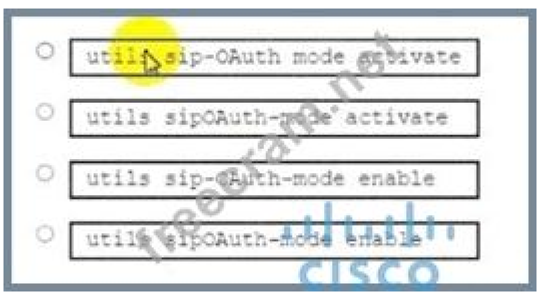
NEW QUESTION: 108
Which DSCP value and PHB equivalent are the default for audio calls?
A. 32 and AF41
B. 48 and EF
C. 34 and AF41
D. 32 and CS4
Answer: (SHOW ANSWER)

NEW QUESTION: 109
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. Map audio and video streams of video calls (AF41 and AF42) to a class-based queue with weighted random early detection.
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. 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.
D. Use 808.IQ trunking and 802.lp for proper treatment of Layer 2 CoS packet marking on ports with phones connected.
E. Implement IP RTP header compression on the serial interface to reduce the bandwidth requlreq per voice call on point-to-point links.
Answer: (SHOW ANSWER)

NEW QUESTION: 110
For Webex Calling, how are outbound transfers and forwards restricted for a particular call type?
A. independently on a site-by-site basis
B. independently on an organization-by-organization basis
C. together on an organization-by-organization basis
D. together on a site-by-site basis

Answer: (SHOW ANSWER)

NEW QUESTION: 111
Which behavior occurs when Cisco UCM has a CallManager group that consists of two subscribers?
A. Endpoints attempt to register with the top subscriber in the list.
B. Endpoints attempt to register with the bottom subscriber in the list.
C. If a subscriber is rebooted, endpoints deregister until the rebooted system is back in service.
D. Endpoints attempt to register with both subscribers in a load-balanced method.
Answer: (SHOW ANSWER)

NEW QUESTION: 112
Which command enables SIP OAuth mode on a Cisco UCM publisher node?


A. Option A
B. Option D
C. Option B
D. Option C
Answer: (SHOW ANSWER)

NEW QUESTION: 113
Which DiffServe PHB preserves backward compatibility with any IP precedence scheme?
A. expedited forwarding
B. class selector
C. default
D. assured forwarding
Answer: (SHOW ANSWER)

NEW QUESTION: 114
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. CSS
B. device pool
C. route list
D. route pattern
Answer: (SHOW ANSWER)

NEW QUESTION: 115

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. Annunciator

B. Conference Bridge

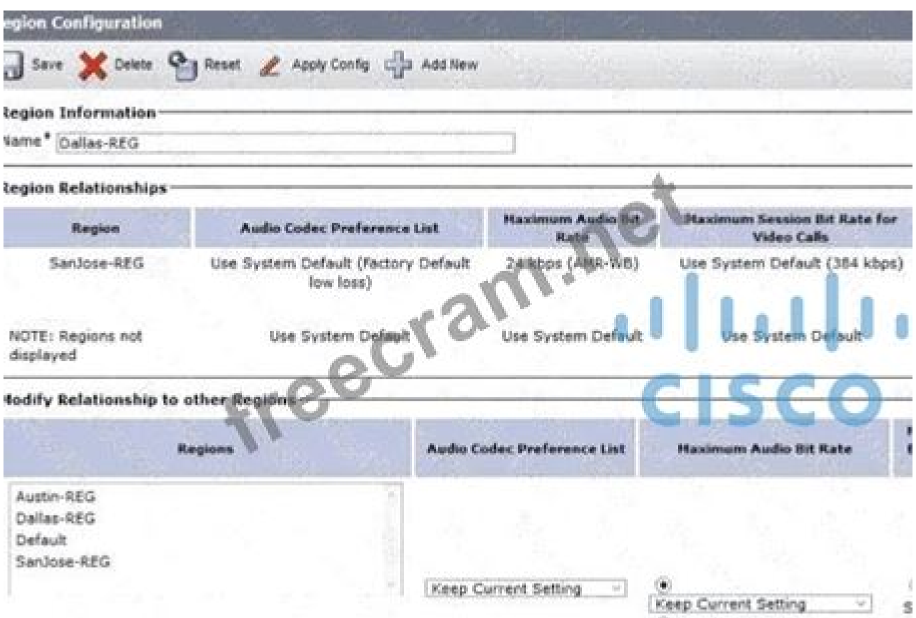
C. Media Termination point

D. Trust relay point

Answer: (SHOW ANSWER)

NEW QUESTION: 116

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: (SHOW ANSWER)

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

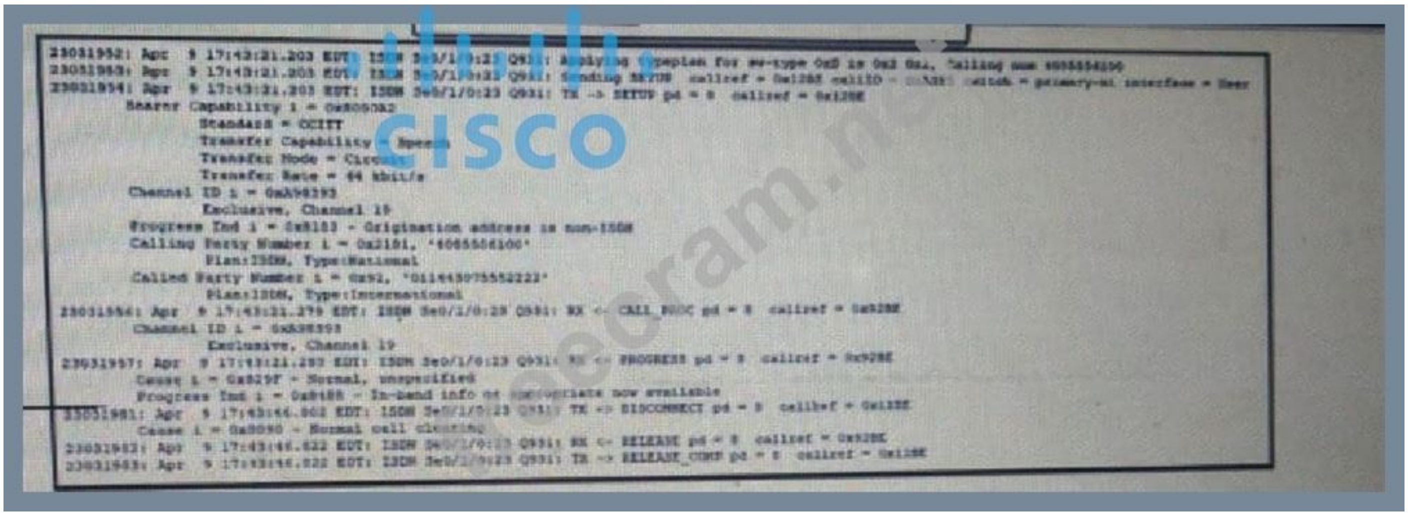
* G.711
* OPUS
* MP4A-LATM
Since both regions have the same codec preference list, the codec that will be used for a call made between
"Dallas-REG" and "Austin-REG" is G.729.
* G.729 is a narrowband speech codec that was developed by the ITU-T in 1988. It is a low-bitrate codec that provides good quality 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.

NEW QUESTION: 117
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: (SHOW ANSWER)
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.

NEW QUESTION: 118
How are E.164 called-party numbers normalized on a globalized call-routing environment in Cisco UCM?
A. Normalization is achieved by setting up calling search spaces and partitions at the SIP trunks for PSTN connection.
B. Call ingress must be normalized before the call being routed.

C. Normalization is not required.
D. Normalization is achieved by stripping or translating the called numbers to internally used directory numbers.
Answer: (SHOW ANSWER)

NEW QUESTION: 119
Refer to the exhibit.



```
Bearer Capability i = 0x8090A2
        Standard = CCITT
        Transfer Capability = Speech
        Transfer Mode = Circuit
        Transfer Rate = 64 kbit/s
Channel ID i = 0xA98393
        Exclusive, Channel 19
Progress Ind i = 0x8183 - Origination address is non-ISDN
Calling Party Number i = 0x2181, '4085556100'
        Plan:ISDN, Type:National
Called Party Number i = 0x91, '01144307555222'
        Plan:ISDN, Type:International
Channel ID i = 0xA98393
        Exclusive, Channel 19
Cause i = 0x829F - Normal, unspecified
Cause i = 0x8090 - Normal call clearing
```

A call to an international number has failed. Which action corrects this problem?
A. Strip the leading 011 from the called party number
B. Add the isdn switch-type primart-dms100 command to the serial interface.
C. Add the bearer-cap speech command to the voice port.
D. Assign a transcoder to the MRGL of the gateway.
Answer: (SHOW ANSWER)

NEW QUESTION: 120
Which certificate does the Disaster Recovery System in Cisco UCM use to encrypt its communications?
A. Cisco CallManager
B. Cisco Tomcat
C. IPsec
D. CAPF
Answer: (SHOW ANSWER)

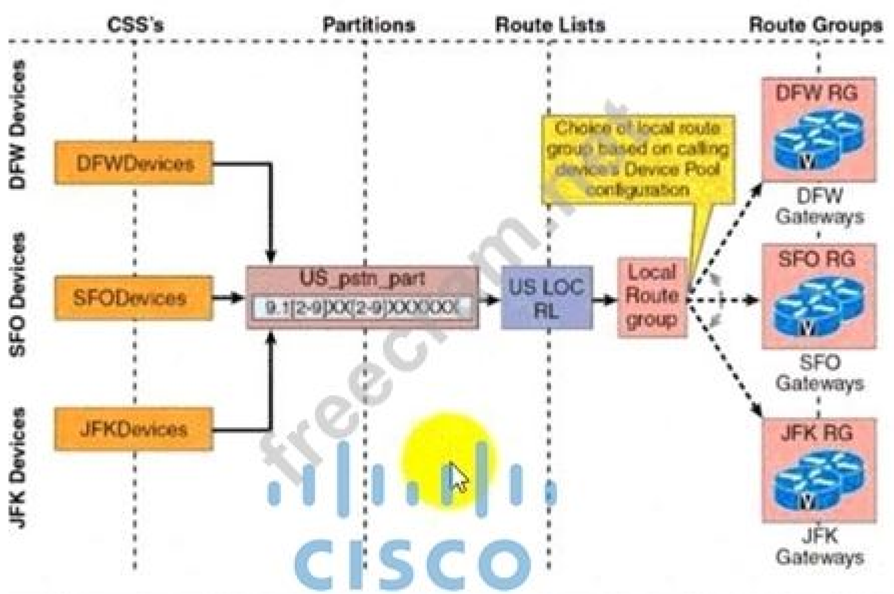
NEW QUESTION: 121
What should be used to detect common issues on a Cisco IOS XE-based Local Gateway and generate an email?
A. Real-Time Monitoring Tool
B. diagnostic signatures
C. syslog
D. SNMP
Answer: (SHOW ANSWER)

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.

NEW QUESTION: 122
Where in Cisco UCM is restrictions on audio bandwidth configured?
A. location
B. partition
C. region
D. serviceability
Answer: (SHOW ANSWER)

NEW QUESTION: 123
What is the function of DiffServ?
A. It provides reserve bandwidth by using RSVP
B. It categorizes traffic.
C. It guarantees end-to-end bandwidth for traffic.
D. It automatically identifies traffic
Answer: (SHOW ANSWER)

NEW QUESTION: 124



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 the Standard Local Route Group is being used only it no configuration is set for the device pools.
B. 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.
C. The call egresses in New York because the device automatically is assigned a New York device pool and uses the local gateway.
D. 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.
Answer: (SHOW ANSWER)

NEW QUESTION: 125
Regarding SIP integrations with Cisco Unified Communications Manager, if the Cisco Unity Connection is configured to listen for incoming IPv4 and IPv6 traffic, how should the addressing mode be set up in the Cisco Unity Connection?
A. Set up IPv4 and IPv6 in cisco Unified CM.
B. Set up for each group to use IPv4 and IPv6.
C. Set up is not required.
D. Set up media ports for each port group to use IPv4.
Answer: (SHOW ANSWER)

NEW QUESTION: 126

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 /9?911/ /911/
 rule 2 /^9\(.*\)/ /\0/
```

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

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

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

A. Option D
B. Option A
C. Option B
D. Option C
Answer: (SHOW ANSWER)

NEW QUESTION: 127
Which values must be enabled to set up QoS for video conferencing on a VoIP network?
A. AP41 and DSCP 34
B. CS3 and DSCP 24
C. AF31 and DSCP 34
D. CS4 and DSCP 32
Answer: (SHOW ANSWER)

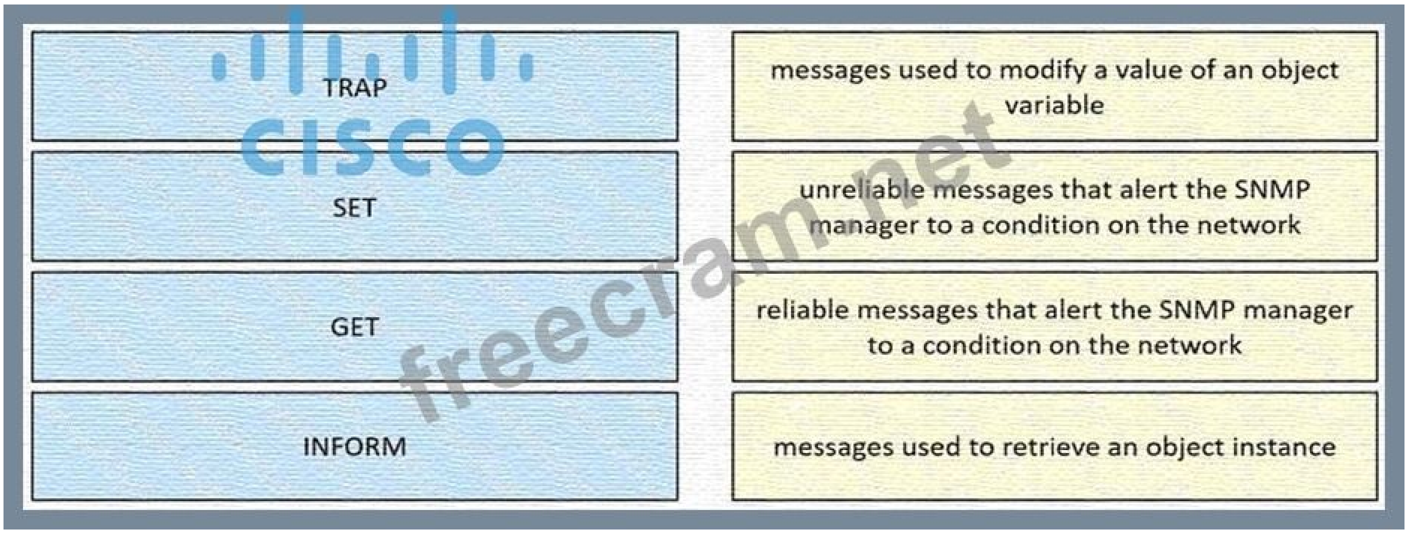
NEW QUESTION: 128
Which action prevent toll fraud in Cisco Unified Communication Manager?
A. Configure ad hoc conference restriction
B. Implement route patterns in Cisco Unified CM
C. Allow off-net to off-net transfer

D. Implement toll fraud restriction in the Cisco IOS router

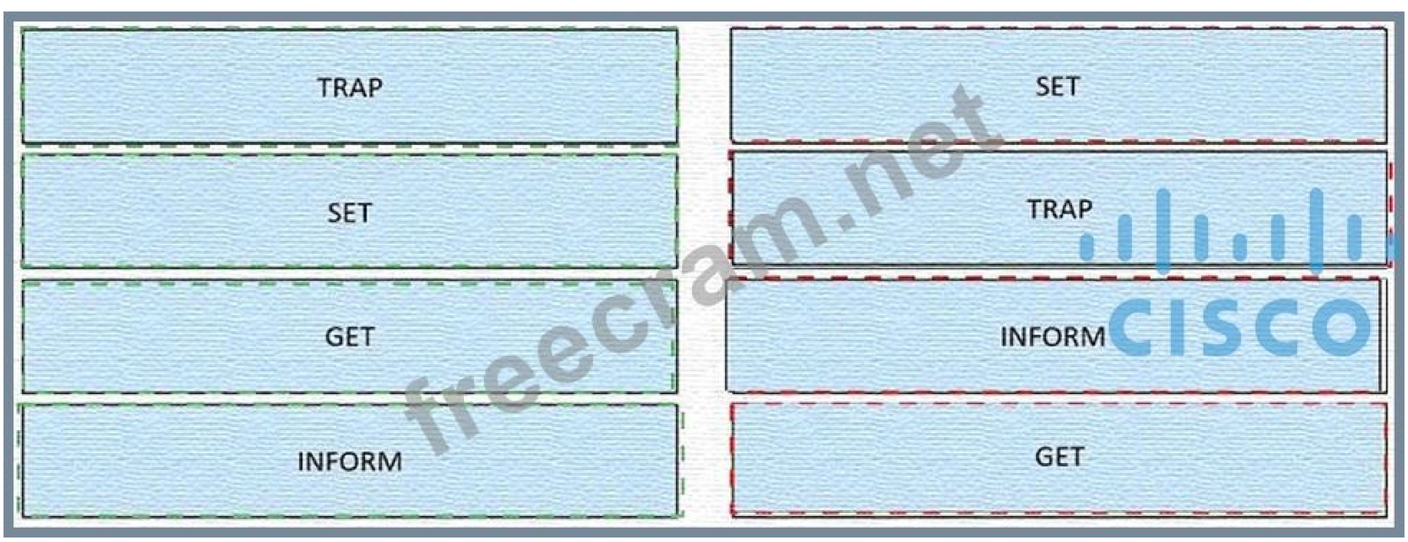
Answer: (SHOW ANSWER)

NEW QUESTION: 129

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



Answer:



NEW QUESTION: 130

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

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

C. 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.

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

Answer: (SHOW ANSWER)

NEW QUESTION: 131

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: (SHOW ANSWER)

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

NEW QUESTION: 132
An administrator must implement a new Cisco Unity Connection cluster integrated with LDAP for users and SIP with Cisco UCM. Assuming a phone system and LDAP directory has been configured, how must the administrator add a new user mailbox with message waiting indication functionality to this setup?
A. Manually configure the new user with required parameters, and set up MWI on and off DNs in Cisco UCM.
B. Import the new user from LDAP, configure the mailbox, and set up MWI on and off DNs in Cisco UCM.
C. Sync the new user from Cisco UCM using AXL setup, and set up MWI on and off DNs in Cisco UCM.
D. Import the new user from LDAP, configure the mailbox, and configure a SIP trunk for MWI functionality.
Answer: (SHOW ANSWER)

NEW QUESTION: 133

```
Endpoint A:
m=audio 21796 RTP/AVP 108 9 104 105 101
 b=TIAS:64000
 a=extmap:14 http://protocols.cisco.com/timestamp#100us
 a=rtpmap:108 MP4A-LATM/90000
 a=fmtp:108 bitrate=64000;profile-level-id=24;object=23
 a=rtpmap:9 G722/8000
 a=rtpmap:104 G7221/16000
 a=fmtp:104 bitrate=32000
 a=rtpmap:105 G7221/16000
 a=fmtp:105 bitrate=24000
 a=rtpmap:101 telephone-event/8000
 a=fmtp:101 0-15
 a=trafficclass:conversational.audio.immersive.aq:admitted

Endpoint B:
m=audio 21796 RTP/AVP 105 0 8 18 101
 b=TIAS:64000
 a=extmap:14 http://protocols.cisco.com/timestamp#100us
 a=rtpmap:105 G7221/16000
 a=fmtp:105 bitrate=24000
 a=rtpmap:0 PCMU/8000
 a=rtpmap:8 PCMA/8000
 a=rtpmap:18 G729/8000
 a=fmtp:18 annexb=no
 a=rtpmap:101 telephone-event/8000
```

```
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=trafficclass:conversational.audio.immersive.aq:admitted
```


Refer to the exhibit. Endpoint A calls endpoint B. What is the only audio codec that is used for the call?

A. G7221/16000

B. Telephone-event/8000

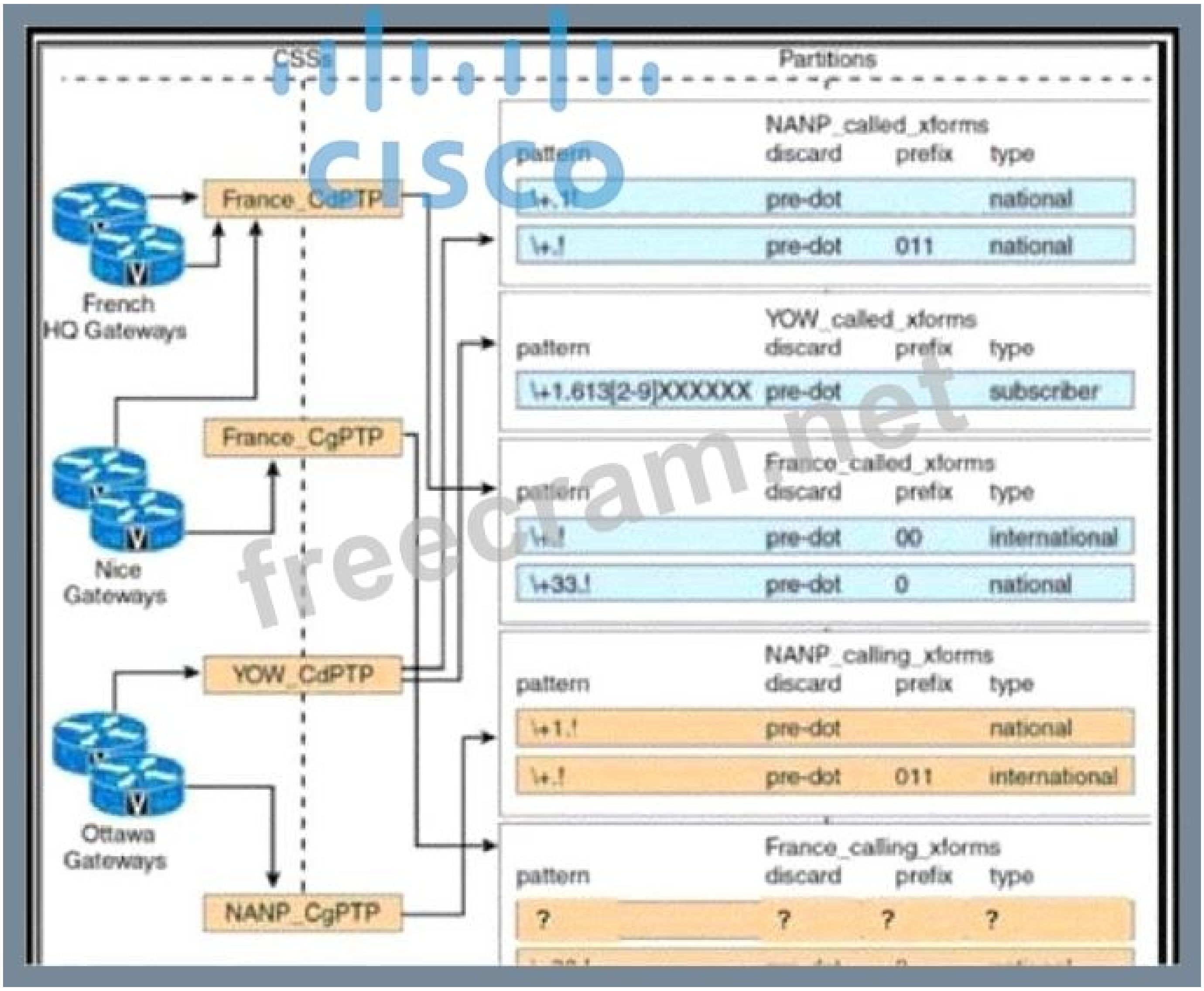
C. PCMA/8000

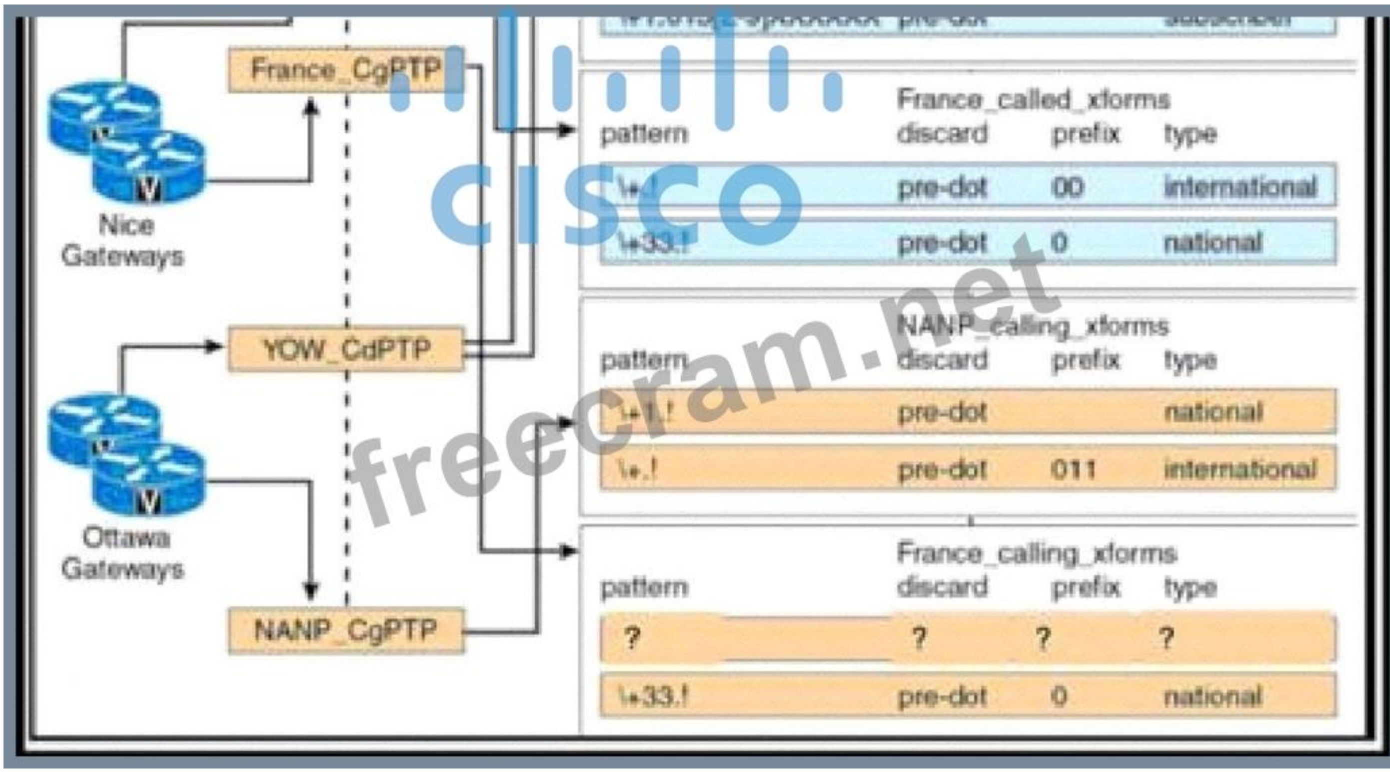
D. G722/8000

Answer: (SHOW ANSWER)

NEW QUESTION: 134

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. \+1.1 none pre-dot 001 international

B. \+.001! pre-dot 1 international

C. \+.! pre-dot 00 international

D. 613XXXXXXX none +011 internationa

Answer: (SHOW ANSWER)

NEW QUESTION: 135

```
Gateway1#show sccp
SCCP Admin State: UP
Gateway Local Interface: Loopback0
        IPv4 Address: 192.168.12.1
        Port Number: 2000


Gateway1#
Gateway1#show ccm-manager
% Call Manager Application is not enabled
Gateway1#


Gateway1#show mgcp
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 "mgcp" on the gateway in configuration mode.
C. Delete and re-add the gateway configuration in Cisco UCM.
D. Enter "ccm-manager config" on the gateway in configuration mode.
E. Enter "ccm-manager mgcp" on the gateway in configuration mode.
Answer: (SHOW ANSWER)

NEW QUESTION: 136
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 PVDM4-128
B. Cisco Webex Meetings Server
C. software conference bridge on Cisco UCM
D. Cisco Meeting Server
Answer: (SHOW ANSWER)

NEW QUESTION: 137

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

Refer to the exhibit. Given this "debug isdn q921" output, what is the problem with the PRI?
A. PRI does not have an IP address configured on the interface.
B. Layer 1 is down on the controller.
C. Layer 2 is down on the controller.
D. Nothing, the PRI is sending keepalives.
Answer: (SHOW ANSWER)

NEW QUESTION: 138

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: ([SHOW ANSWER]())

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.

NEW QUESTION: 139

```
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
```

Refer to the exhibit.

Endpoint A calls endpoint B. What is the only audio codec that can be used for the call?
A. Telephone-event/8000
B. G7221/46000
C. G722/8000
D. PCMA/8000
Answer: (SHOW ANSWER)

NEW QUESTION: 140
Callers from a branch report getting busy tones intermittetly when trying to reach colleagues in other office branches during peak hours. An engineer collects Cisco CallManager service traes to examine the situation. The traces show:

```
50805567.000 |07:35:39.676 |Sdl Sig |StationOoutputDisplayNotify |restart0
|StaatinD(1,100,63,6382) |StionCdpc(1,100,64,4725) |1,100,40,6.709919^*^*
|[R:N-H:0,L:0,V:0,Z:0,D:0] TimeOutValue=10 Status=x807 Unicode Status=Locale=1
50805567.001 |07:35:39.676 |AppInfo |StationD: (0006382) DisplayNotify
timeOutValue=10 notify='x807' content='Not Enough Bandwidth' ver=85720014.
```

What should be fixed to resolve the issue?
A. region configuration
B. geolocation configuration
C. codec configuration
D. class of service configuration
Answer: (SHOW ANSWER)

NEW QUESTION: 141
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. Ensure that you have set up SIP Digest Authentication on the SIP trunk security profile.
B. Increase the number of voice ports.
C. Add dedicated dial-out ports with the allow trap connections setting selected.
D. The phone system to which the phones are registered in Unity Connection has the Default Trap Switch check box enabled.
E. If it's a Unity Connection Cluster, ensure that replication is fine and not in split-brain mode.
Answer: (SHOW ANSWER)

NEW QUESTION: 142
Which action prevents toll fraud in Cisco UCM?
A. Implement route patterns in Cisco UCM.
B. Configure ad hoc conference restriction.

C. Allow off-net to off-net transfers.
D. Implement toll fraud restriction in the Cisco IOS router.
Answer: (SHOW ANSWER)

NEW QUESTION: 143
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. 30 min
C. 10 min
D. 60 min
Answer: (SHOW ANSWER)

NEW QUESTION: 144
In the cisco expressway solution, which two features does mobile and Remote access provide? (Choose two)
A. secure reverse proxy firewall traversal connectivity
B. VPN-based enterprise access for a subset of Cisco Unified IP Phone models
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: (SHOW ANSWER)

NEW QUESTION: 145
Which issue cause slips on a PRI?
A. incorrectly configured time zone
B. incorrect encapsulation
C. incorrect dock source
D. change in the line code
Answer: C (LEAVE A REPLY)

NEW QUESTION: 146

```
    Bearer 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 = 0xA1, '2145551212'
          Plan:ISDN, Type:National
Mar  1 02:35:37: ISDN Se0/1/1:23 Q931: RX <- CALL_PROC pd = 8  callref = 0x809A
     Channel ID i = 0xA98388
          Exclusive, Channel 8

interface Serial0/1/1:23
 description PRI Circuit to R1
 no ip address
 encapsulation hdlc
 isdn switch-type primary-ni
 isdn protocol-emulate network
 isdn incoming-voice voice
 no cdp 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 send display le

B. isdn supp-service name calling

C. isdn outgoing display-ie

D. isdn enable did

Answer: (SHOW ANSWER)

NEW QUESTION: 147

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

A. \+13135551212@192.168.1.3

B. 192.168.1.3

C. ciscocm1.jupiter.com

D. +13135551212@192.168.1.3

Answer: (SHOW ANSWER)

NEW QUESTION: 148

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

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


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. 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

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

Answer: (SHOW ANSWER)

NEW QUESTION: 149

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

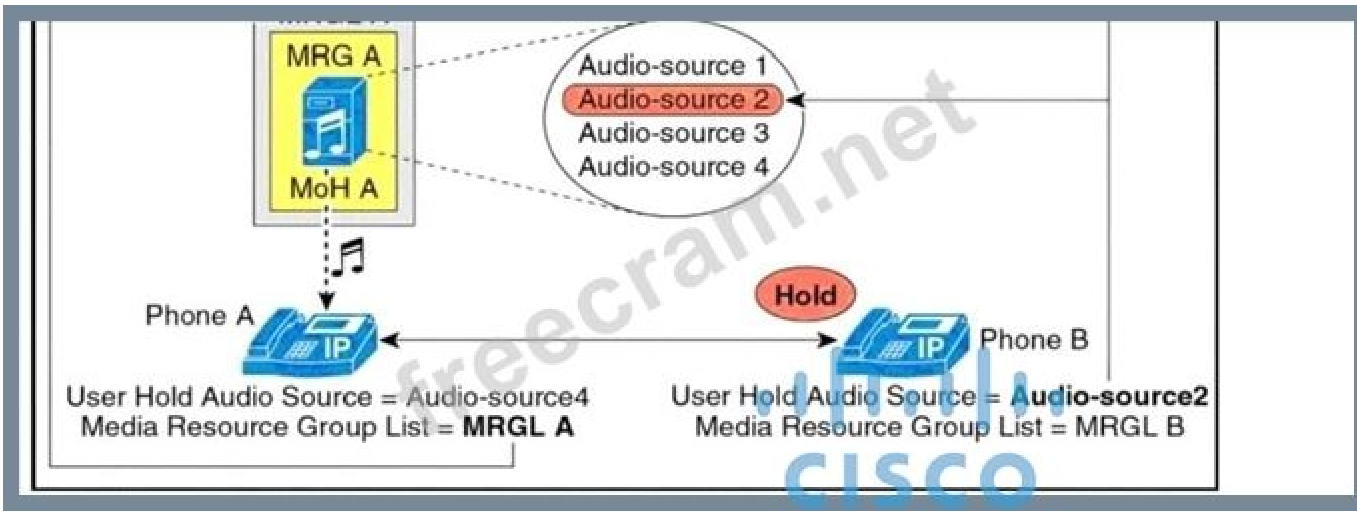
B. Device(config~if)# switchport trunk allowed vlan 221

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

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

Answer: D (LEAVE A REPLY)

NEW QUESTION: 150



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: (SHOW ANSWER)

NEW QUESTION: 151

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 Unified Border Element, Cisco UCM, 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 IM and Presence Server and Cisco Video Communication Server

Answer: (SHOW ANSWER)

NEW QUESTION: 152

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 switch-type primary-qsig
B. isdn switch-type basic-qsig
C. isdn incoming-voice voice
D. isdn switch-type basic-ni
Answer: (SHOW ANSWER)

NEW QUESTION: 153
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. 192.168.102.102
B. 192.168.101.201
C. 192.168.100.103
D. 192.168.100.102
Answer: (SHOW ANSWER)

NEW QUESTION: 154

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. Set the service parameter Use Video Bandwidth Pool for Immersive Video Calls to "false".

B. Check for duplex/speed mismatches between the network port settings of the system and network switch.

C. Use direct IP address calls between two endpoints to troubleshoot call quality issues.

D. Restart the Cisco Location Bandwidth Manager service on the Cisco UCM publisher.

Answer: (SHOW ANSWER)

NEW QUESTION: 155

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

A. SSO

B. MGCP gateway registration

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

D. SRST

Answer: (SHOW ANSWER)

NEW QUESTION: 156

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. open the SIP Trunk Security Profile Information menu and select Accept Unsolicited Notification.

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

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

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

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

Answer: (SHOW ANSWER)

NEW QUESTION: 157

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

A. 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
B. When using media flow-through, call signaling goes through the Cisco Unified Border Element, but media does not
C. When using media flow-around, the can signaling goes through the Cisco Unified Border Element, but media is not passed through It.
D. When using media flow-through, the call signaling and media are passed through the Cisco Unified Border Element
Answer: (SHOW ANSWER)

NEW QUESTION: 158
Exhibit.

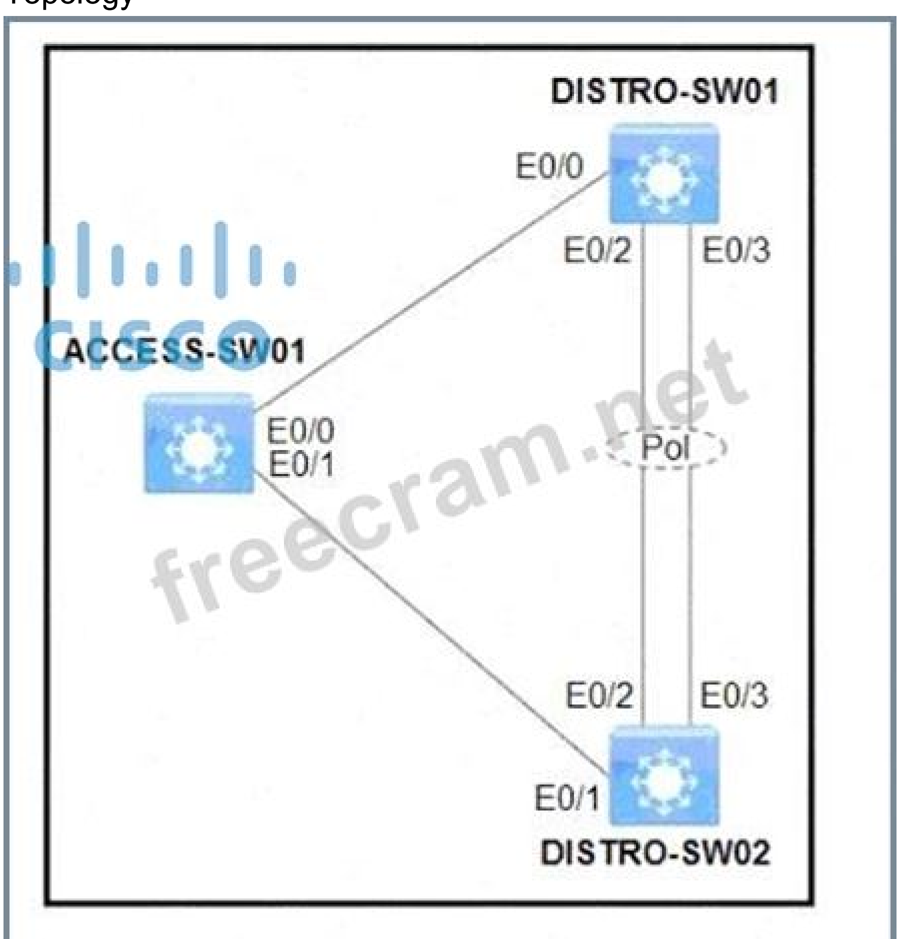
```
!
voice class codec 10
 codec preference 1 g711ulaw
 codec preference 2 g729r8
 codec preference 3 ilbc
 video codec h264
!
```

Configuration of DNS is required to achieve a fully functional Cisco UCM system. Cisco UCM uses DNS to resolve fully qualified domain names to IP addresses for which destinations?
A. MRA
B. AAR
C. trunk
D. H.323
Answer: (SHOW ANSWER)

NEW QUESTION: 159
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 regular route pattern with URI feature enable in the configuration page. The pattern must look like this:(*@company.com)

B. A regular route pattern with URI feature enable in the configuration page. The pattern must look like this:MATCH(*@company.com)
C. A SIP route pattern that looks like the *@company.com
D. A SIP route pattern that looks like this company.com
Answer: A (LEAVE A REPLY)

NEW QUESTION: 160
Guidelines
This is a lab item in which tasks will be performed on virtual devices.
* Refer to the Tasks tab to view the tasks for this lab item.
* Refer to the Topology tab to access the device console(s) and perform the tasks.
* Console access is available for all required devices by clicking the device icon or using the tab(s) above the console window.
* All necessary preconfigurations have been applied.
* Do not change the enable password or hostname for any device.
* Save your configurations to NVRAM before moving to the next item.
* Click Next at the bottom of the screen to submit this lab and move to the next question.
* When Next is clicked, the lab closes and cannot be reopened.
Topology



Tasks
The operations team started configuring network devices for a new site.

Complete the configurations to achieve these goals:

1. Ensure that port channel Po1 between DISTRO-SW01 and DISTRO-SW02 is operational using the LACP protocol. Configuration changes for this task must be made on DISTRO-SW01.
2. Ensure that traffic on VLAN 10 is carried as untagged traffic between DISTRO-SW01 and DISTRO-SW02.
3. Complete the Rapid-PVST+ configuration on DISTRO-SW2 by ensuring it it the secondary root switch for all VLANs in the range of 1 to 1005.

DISTRO-SW01

```
DISTRO-SW01>
DISTRO-SW01>
DISTRO-SW01>en
DISTRO-SW01#config t
Enter configuration commands, one per line. End with CNTL/Z.
DISTRO-SW01(config)# in
DISTRO-SW01(config)# interface e0/0
DISTRO-SW01(config-if)#no cha
DISTRO-SW01(config-if)#no channel-g
DISTRO-SW01(config-if)#no channel-group 1
DISTRO-SW01(config-if)#no channel-group 1 mo
DISTRO-SW01(config-if)#no channel-group 1 mode pas
DISTRO-SW01(config-if)#no channel-group 1 mode passive
DISTRO-SW01(config-if)#
* Jul 10 20:07:53.469: %LINEPROTO-5-UPDOWN: Line protocol on Interface Ethernet0/0,
changed state to up
DISTRO-SW01(config-if)#
DISTRO-SW01(config-if)#exit
DISTRO-SW01(config)#
DISTRO-SW01(config)# in
DISTRO-SW01(config)# interface ran
DISTRO-SW01(config) # interface range e
DISTRO-SW01(config)# interface range ethernet 0/2-3
DISTRO-SW01(config-if-range)#ch
DISTRO-SW01(config-if-range)#channel-gr
DISTRO-SW01(config-if-range)#channel-group 1
DISTRO-SW01(config-if-range)#channel-group 1 mo
DISTRO-SW01(config-if-range)#channel-group 1 mode ac
DISTRO-SW01(config-if-range)#channel-group 1 mode active
DISTRO-SW01(config-if-range)#
* Jul 10 20:08:25.239: %LINEPROTO-5-UPDOWN: Line protocol on Interface Ethernet0/2,
changed state to up
* Jul 10 20:08:25.239: %LINEPROTO-5-UPDOWN: Line protocol on Interface Ethernet0/3,
changed state to up
DISTRO-SW01(config-if-range)#exit
DISTRO-SW01(config)#
* Jul 10 20:08:31.447: %LINK-3-UPDOWN: Interface Port-channel, changed state to up
* Jul 10 20:08:32.451: %LINEPROTO-5-UPDOWN: Line protocol on Interface Ethernet0/3,
changed state to up
DISTRO-SW01(config)#
DISTRO-SW01(config)#exit
DISTRO-SW01#
* Jul 10 20:08:39.212: %SYS-5-CONFIG_I: Configured from console by console
DISTRO-SW01#sh eth
DISTRO-SW01#sh etherc
DISTRO-SW01#sh etherchannel s
DISTRO-SW01#sh etherchannel summary
```

```
DISTRO-SW01#sh etherchannel s
DISTRO-SW01#sh etherchannel summary
Flags:  D - down        P - bundled in port-channel
        I - stand-alone s - suspended
        H - Hot-standby (LACP only)
        R - Layer3      S - Layer2
        U - in use      N - not in use, no aggregation
        f - failed to allocate aggregate

        M - not in use, minimum links not met
        m - not in use, port not aggregated due to minimum links not met
        u - unsuitable for bundling
        w - waiting to be aggregated
        d - default port

      A - formed by Auto LAG

Number of channel-groups in use: 1
Number of aggregators:           1

Group  Port-channel  Protocol    Ports
------+-------------+-----------+-----------------------------------------------
1      Po1(SU)         LACP      Et0/2(P)    Et0/3(P)
```

```
DISTRO-SW01#
DISTRO-SW01#config t
Enter configuration commands, one per line. End with CNTL/Z.
DISTRO-SW01(config)#inte
DISTRO-SW01(config)#interface po
DISTRO-SW01(config)#interface port-
DISTRO-SW01(config)#interface port-channel 1
DISTRO-SW01(config-if)#sw
DISTRO-SW01(config-if)#switchport tr
DISTRO-SW01(config-if)#switchport trunk na
DISTRO-SW01(config-if)#switchport trunk native v1
DISTRO-SW01(config-if)#switchport trunk native vlan 10
DISTRO-SW01(config-if)#
* Jul 10 20:09:27.352: %CDP-4-NATIVE_VLAN_MISMATCH: Native VLAN mismatch
discovered on Ethernet0/3 (10), with DISTRO-SW02 Ethernet0/3 (1).
DISTRO-SW01(config-if)#
* Jul 10 20:09:27.857: %SPANTREE-2-RECV_PVID_ERR: Received BPDU with inconsistent
peer vlan id 1 on Port-channel1 VLAN10.
* Jul 10 20:09:27.857: %SPANTREE-2-BLOCK_PVID_PEER: Blocking Port-channel1 on
VLAN0001. Inconsistent peer vlan.
* Jul 10 20:09:27.857: %SPANTREE-2-BLOCK_PVID_LOCAL: Blocking Port-channel1 on
VLAN0010. Inconsistent local vlan.
DISTRO-SW01(config-if)#
```

```
* Jul 10 20:09:27.857: %SPANTREE-2-RECV_PVID_ERR: Received BPDU with inconsistent peer vlan
id 1 on Port-channel1 VLAN10.
* Jul 10 20:09:27.857: %SPANTREE-2-BLOCK_PVID_PEER: Blocking Port-channel1 on VLAN0001.
Inconsistent peer vlan.
* Jul 10 20:09:27.857: %SPANTREE-2-BLOCK_PVID_LOCAL: Blocking Port-channel1 on VLAN0010.
Inconsistent local vlan.
DISTRO-SW01(config-if)#
* Jul 10 20:09:30.710: %CDP-4-NATIVE_VLAN_MISMATCH: Native VLAN mismatch
discovered on Ethernet0/2 (10), with DISTRO-SW02 Ethernet0/2 (1).
DISTRO-SW01(config-if)#
* Jul 10 20:10:03.864: %SPANTREE-2-UNBLOCK_CONSIST_PORT: Unblocking Port-channel on
VLAN0001. Port consistently restored
* Jul 10 20:10:03.864: %SPANTREE-2-UNBLOCK_CONSIST_PORT: Unblocking Port-channel on
VLAN0010. Port consistently restored
DISTRO-SW01(config-if)#
DISTRO-SW01(config-if)#exit
DISTRO-SW01(config)#
DISTRO-SW01(config)#exit
DISTRO-SW01#co
DISTRO-SW01#co
* Jul 10 20:10:35.644: %SYS-5-CONFIG_I: Configured from console by console
DISTRO-SW01#copy ru
DISTRO-SW01#copy running-config st
DISTRO-SW01#copy running-config startup-config
Destination filename [startup-config]?
Building configuration…
Compressed configuration from 1447 bytes to 886 bytes[OK]
DISTRO-SW01#
DISTRO-SW01#
```

DISTRO-SW02

```
DISTRO-SW02>
* Jul 10 20:08:25.239: %LINEPROTO-5-UPDOWN: Line protocol on Interface Ethernet0/2,
changed state to up
* Jul 10 20:08:25.239: %LINEPROTO-5-UPDOWN: Line protocol on Interface Ethernet0/3,
changed state to up
DISTRO-SW02>
* Jul 10 20:08:31.447: %LINK-3-UPDOWN: Interface Port-channel, changed state to up
* Jul 10 20:08:32.448: %LINEPROTO-5-UPDOWN: Line protocol on Interface Port-channel,
changed state to up
DISTRO-SW02>
* Jul 10 20:09:27.848: %SPANTREE-2-RECV_PVID_ERR: Received BPDU with inconsistent peer
vlan id 10 on Port-channel1 VLAN1.
* Jul 10 20:09:27.848: %SPANTREE-2-BLOCK_PVID_PEER: Blocking Port-channel1 on VLAN0010.
Inconsistent peer vlan.
* Jul 10 20:09:27.857: %SPANTREE-2-BLOCK_PVID_LOCAL: Blocking Port-channel1 on VLAN0001.
Inconsistent local vlan.
DISTRO-SW02>
* Jul 10 20:09:31.330: %CDP-4-NATIVE_VLAN_MISMATCH: Native VLAN mismatch
discovered on Ethernet0/3 (1), with DISTRO-SW01 Ethernet 0/3 (10).
```

```
DISTRO-SW02>en
DISTRO-SW02#config t
Enter configuration commands, one per line. End with CNTL/Z.
DISTRO-SW02(config)#
DISTRO-SW02(config)#inter
DISTRO-SW02(config)#interface por
DISTRO-SW02(config)#interface port-ch
DISTRO-SW02(config)#interface port-channel 1
DISTRO-SW02(config)#interface port-channel 1
DISTRO-SW02(config-if)#sw
DISTRO-SW02(config-if)#switchport tr
DISTRO-SW02(config-if)#switchport trunk na
DISTRO-SW02(config-if)#switchport trunk native vl
DISTRO-SW02(config-if)#switchport trunk native vlan 10
DISTRO-SW02(config-if)#exit
DISTRO-SW02(config)#
DISTRO-SW02(config)#sp
DISTRO-SW02(config)#spanning-tree vl
DISTRO-SW02(config)#spanning-tree vlan 1-
* Jul 10 20:10:04.934: %SPANTREE-2-UNBLOCK_CONSIST_PORT: Unblocking
Port-channel1 on VLAN0010. Port consistently restored.
* Jul 10 20:10:04.934: %SPANTREE-2-UNBLOCK_CONSIST_PORT: Unblocking
Port-channel1 on VLAN0001. Port consistently restored.
DISTRO-SW02(config)#spanning-tree vlan 1-1005 ro
DISTRO-SW02(config)#spanning-tree vlan 1-1005 root se
DISTRO-SW02(config)#spanning-tree vlan 1-1005 root secondary
DISTRO-SW02(config)#exit
```

```
DISTRO-SW02 con0 is now available

Press RETURN to get started
DISTRO-SW02>
DISTRO-SW02>
DISTRO-SW02>en
DISTRO-SW02#copy ru
DISTRO-SW02#copy running config st
DISTRO-SW02#copy running config startup-config
Destination filename [startup-config]?
Building configuration…
Compressed configuration from 1449 bytes to 889 bytes[OK]
DISTRO-SW02#
DISTRO-SW02#
```

Answer:

see the answer in explanation below.

Suggested Answer:

Task 1. Ensure that port channel Po1 between DISTRO-SW01 and DISTRO-SW02 is operational using the LACP protocol.
Configuration changes
for this task must be made on DISTRO-SW01.
The port towards to ACCESS-SW1 was part of port channel 1 which was used towards to DISTRO-SW02, so remove the port channel config pointing to the ACCESS-SW1 first.

```
DISTRO-SW01(config)#int e0/0
DISTRO-SW01(config-if)#no channel-group 1
```

Set the LACP mode active on this switch:

```
DISTRO-SW01(config-if)#int range e0/2 - 3
DISTRO-SW01(config-if)#channel-group 1 mode active
```

Verification:
We can verify if the Port-channel was formed with the "show etherchannel summary" command

```
DISTRO-SW01# show etherchannel summary
-- output omitted --
 Number of channel-groups in use: 1
 Number of aggregators:       1
 Group  Port-channel  Protocol    Ports
------+-------------+-----------+-----------------------------------------------
 1      Po1(SU)        LACP      Et0/2(P)   Et0/3(P)
```

If we see the "Po1(SU)" means our configuration worked correctly.

Task 2. Ensure that traffic on VLAN 10 is carried as untagged traffic between DISTRO-SW01 and DISTRO-SW02.

```
DISTRO-SW01,DISTRO-SW02(config)#interface port-channel 1
DISTRO-SW01,DISTRO-SW02(config-if)#switchport trunk native vlan 10
```

Task 3. Complete the Rapid-PVST+ configuration on DISTRO-SW2 by ensuring it is the secondary root switch for all VLANs in the range of 1 to 1005.

```
DISTRO-SW02(config)#spanning-tree vlan 1-1005 root secondary
```

Save the configuration

```
DSW1,DSW2(config)#copy running-config startup-config
```


NEW QUESTION: 161

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

A. SIP intercluster trunk

B. H.225 trunk

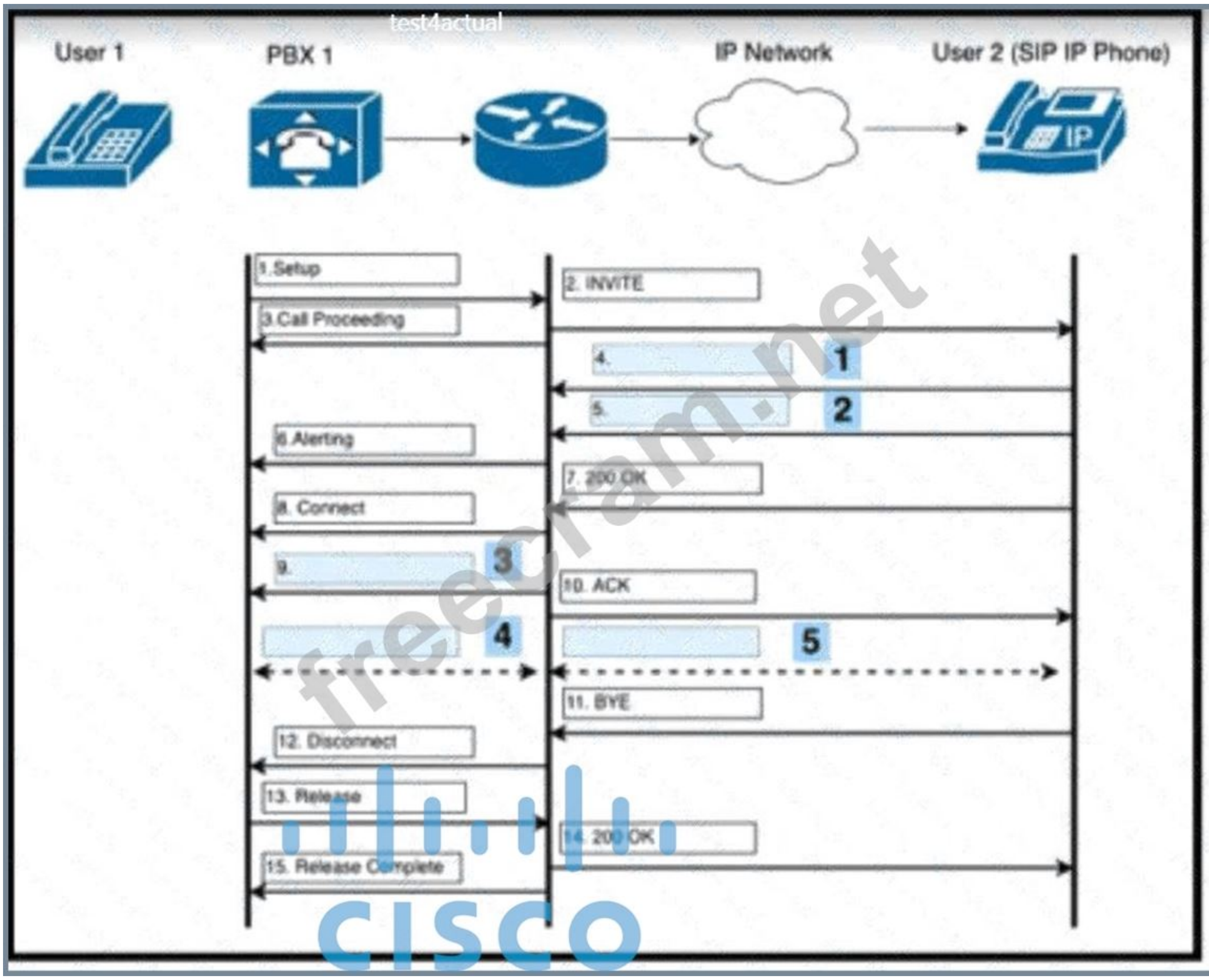
C. gatekeeper-controlled inter cluster trunk

D. nongatekeeper-controlled intercluster trunk

Answer: (SHOW ANSWER)

NEW QUESTION: 162

Refer to the exhibit.



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

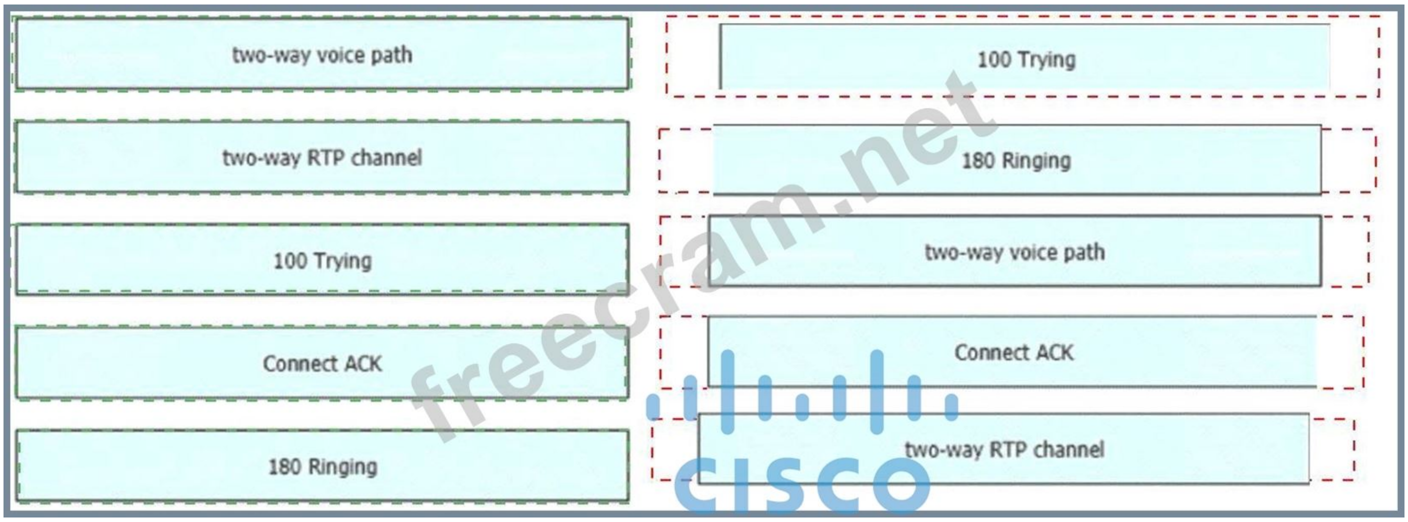
Drag and drop the flow step labels from the left into the correct order on the right to establish this call flow:

* User 1 calls user 2.

* User 2 answers the call

* user 2 disconnects the call

Answer:



Explanation:
1.100 Trying
2.180 Ringing
3.two-way voice path
4.Connect ACK
5.two-way RTP channel

NEW QUESTION: 163

Which Cisco UCM configuration is required for SIP MWI integrations?

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

Answer: (SHOW ANSWER)

NEW QUESTION: 164

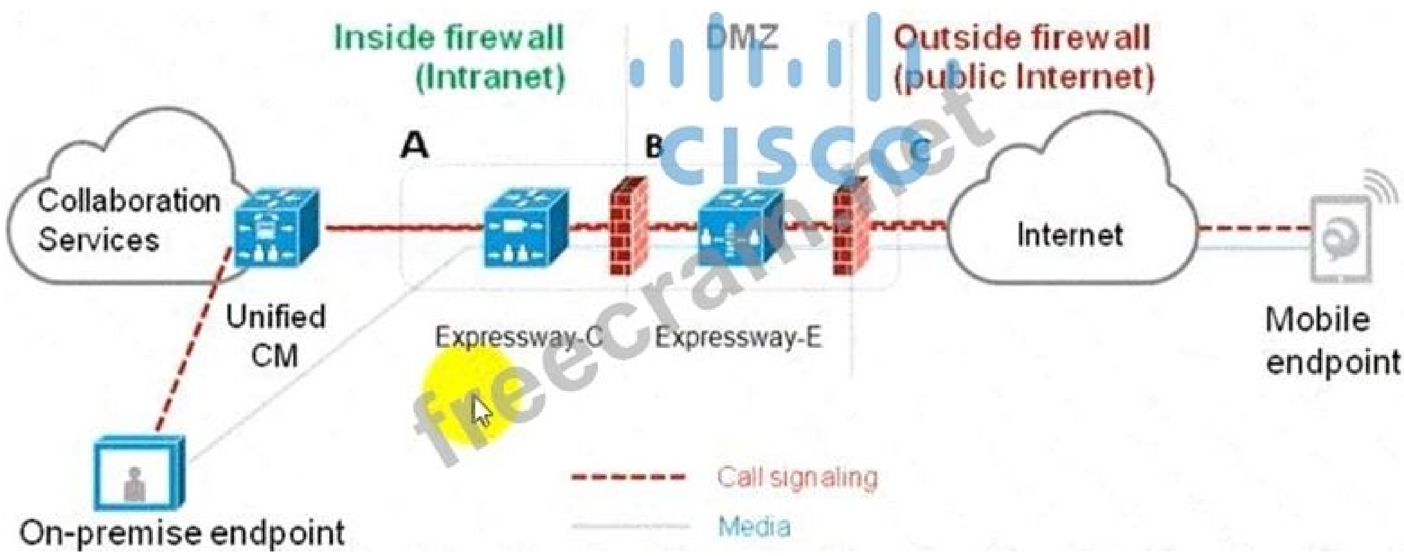
What is a capability of a Cisco IOS XE media resource?

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 (LEAVE A REPLY)

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.

NEW QUESTION: 165



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. SIP TLS (A) + SIP TLS (B) + SIP TLS (C)

B. SIP TCP/TLS (A) + SIP TCP/TLS (B) + SIP TCP/TLS (C)

C. SIP TCP/TLS (A) + SIP TLS (B) + SIP TLS (C)

D. IP TCP/TLS (A) + SIP TCP/TLS (B) + SIP TLS (C)

Answer: (SHOW ANSWER)

NEW QUESTION: 166

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 Conference Bridge

B. IOS XE Hardware Transcoder

C. IOS XE Software MTP

D. IOS XE Hardware MTP

Answer: (SHOW ANSWER)

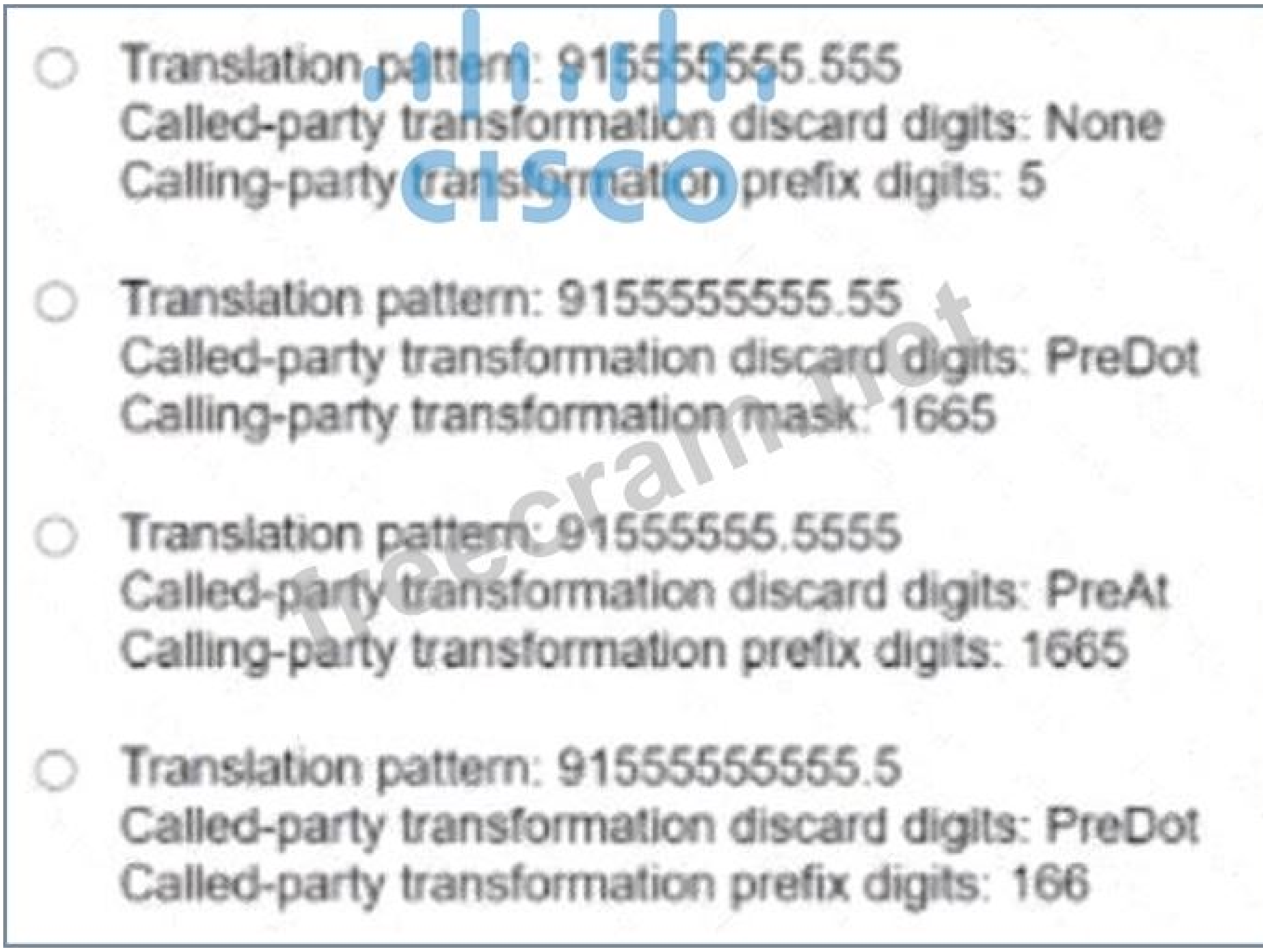
NEW QUESTION: 167

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

A. AF41

B. CS4
C. CS3
D. EF
Answer: (SHOW ANSWER)

NEW QUESTION: 168
An engineer must configure a new Cisco UCM translation pattern to convert 915555555555 into 1665 Which translation pattern must be used?



Translation pattern: 91555555.555
Called-party transformation discard digits: None
Calling-party transformation prefix digits: 5

Translation pattern: 9155555555.55
Called-party transformation discard digits: PreDot
Calling-party transformation mask: 1665

Translation pattern: 91555555.5555
Called-party transformation discard digits: PreAt
Calling-party transformation prefix digits: 1665

Translation pattern: 91555555555.5
Called-party transformation discard digits: PreDot
Called-party transformation prefix digits: 166

A. Option D
B. Option A
C. Option C
D. Option B
Answer: (SHOW ANSWER)

NEW QUESTION: 169
Which Cisco unity Connection handler plays a greeting at announces the option to dial a user extension by default?
A. the Interview handler
B. the Goodbye call handler
C. the Directory handler

D. the operator call handler
Answer: (SHOW ANSWER)

NEW QUESTION: 170
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. accept replaces header
C. accept unsolicited notification
D. enable application-level authorization
E. allow changing header
Answer: (SHOW ANSWER)

NEW QUESTION: 171
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. There is no reason not to include an end user's PC device in a QoS trust boundary.
C. The end user may Incorrectly tag their traffic k> be prioritized over other network traffic.
D. The end user could incorrectly tag their traffic to bypass firewalls.
Answer: (SHOW ANSWER)

NEW QUESTION: 172
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: (SHOW ANSWER)

NEW QUESTION: 173
The chief officer at a company must reduce collaboration infrastructure costs by onbording 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: (SHOW ANSWER)

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 them
1.
* 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.

NEW QUESTION: 174
Which transport protocol does the application layer protocol SNMP use?
A. HTTP
B. UDP
C. SIP
D. XML
Answer: (SHOW ANSWER)