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NEW QUESTION: 1

Which configuration tells a switch port to send Cisco Discovery protocol packets that configure an attached Cisco IP phone to trust tagged traffic that is received from a device that is connected to the access port on the Cisco IP phone?

A)

```
Router# configure terminal
Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos trust extend cos 5
```

B)

```
Router# configure terminal
Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos trust extend
```

C)

```
Router# configure terminal
Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos trust extend cos 3
```

D)

```
Router# configure terminal
Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos extend trust
```

A. Option C

B. Option B

C. Option A

D. Option D

Answer: B ([LEAVE A REPLY](#))

NEW QUESTION: 2

Refer to the exhibit.

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

v=0
o=Cisco-SIPUA 26529 0 IN IP4 10.10.10.84
s=SIP Call
b=AS:4064
t=0 0
m=audio 32136 RTP/AVP 114 9 124 113 115 0 8 116 18
c=IN IP4 10.10.10.84
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114
maxplaybackrate=16000;sprop-maxcapture=16000;maxaveragebitrate=64000;stereo=0;sprop-
stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0;mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1;mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=sendrecv

Call-ID: 381c1aba-7a78000d-4ca6894a-41dd3e0f@10.10.10.84
Max-Forwards: 70
CSeq: 101 INVITE
Contact: <sip:1234@10.10.10.84:50083;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Allow-Events: kpml,dialog
Content-Type: application/sdp
Content-Length: 658

v=0
o=Cisco-SIPUA 26529 0 IN IP4 10.10.10.84
s=SIP Call
b=AS:4064
t=0 0
m=audio 32136 RTP/AVP 114 9 124 113 115 0 8 116 18
c=IN IP4 10.10.10.84
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114
maxplaybackrate=16000;sprop-maxcapture=16000;maxaveragebitrate=64000;stereo=0;sprop-
stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0;mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1;mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=sendrecv
```

When a UC Administrator is troubleshooting DTMF negotiated by this SIP INVITE, which two messages should be examined next to further troubleshoot the issue? (Choose two.)

- A. NOTIFY
- B. UPDATE
- C. SUBSCRIBE
- D. REGISTER
- E. PRACK

Answer: A,C ([LEAVE A REPLY](#))

NEW QUESTION: 3

How can an administrator stop Cisco Unified Communications Manager from advertising the OPUS codec for recording enabled devices?

- A. In CUCM Service Parameters set "Opus codec Enables" to "Enabled for all devices Except Recording-Enabled devices."
- B. Integrate the Cisco Unified CM with a recording solution that does not support OPUS.
- C. Go to the phone's configuration page and set "Advertise OPUS Codec" to be "false"
- D. Router recorded calls through Cisco Unified Border Element because it does not support OPUS.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 4

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

Answer: C ([LEAVE A REPLY](#))

NEW QUESTION: 5

When a phone is registered over Mobile and Remote Access, where does it register?

- A. Cisco Unified Presence Server
- B. Expressway-E
- C. Cisco Unified Communications Manager
- D. Expressway-C

Answer: ([SHOW ANSWER](#))

Explanation/Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_b_system-configuration-guide-1201/cucm_b_system-configuration-guide-1201_chapter_01011010.html

NEW QUESTION: 6

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 7

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 161
- B. TCP 161
- C. TCP 80
- D. UDP 162

Answer: D ([LEAVE A REPLY](#))

[https://www.cisco.com/c/en/us/td/docs/net_mgmt/prime/collaboration/12-](https://www.cisco.com/c/en/us/td/docs/net_mgmt/prime/collaboration/12-1/assurance/advanced/guide/cpco_b_cisco-prime-collaboration-assurance-guide-advanced-12-1/cpco_b_cisco-prime-collaboration-assurance-guide-advanced-12-1_chapter_01111.html)

[1/assurance/advanced/guide/cpco_b_cisco-prime-collaboration-assurance-guide-advanced-12-1/cpco_b_cisco-prime-collaboration-assurance-guide-advanced-12-1_chapter_01111.html](https://www.cisco.com/c/en/us/td/docs/net_mgmt/prime/collaboration/12-1/assurance/advanced/guide/cpco_b_cisco-prime-collaboration-assurance-guide-advanced-12-1/cpco_b_cisco-prime-collaboration-assurance-guide-advanced-12-1_chapter_01111.html)

NEW QUESTION: 8

Given this H.323 gateway configuration and using cisco best practices, how must the called party transformation pattern 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
```

A)

Pattern Definition

Pattern*

Partition

Description

Numbering Plan

Route Filter

☒ Urgent Priority

☐ MLPP Preemption Disabled

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type*

Called Party Numbering Plan*

B)

Pattern Definition

Pattern*

Partition

Description

Numbering Plan

Route Filter

☒ Urgent Priority

☐ MLPP Preemption Disabled

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type*

Called Party Numbering Plan*

C)

Pattern Definition

Pattern*

Partition

Description

Numbering Plan

Route Filter

☒ Urgent Priority

☐ MLPP Preemption Disabled

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type*

Called Party Numbering Plan*

D)

Pattern Definition

Pattern*

Partition

Description

Numbering Plan

Route Filter

☒ Urgent Priority

☐ MLPP Preemption Disabled

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type

Called Party Numbering Plan

A. Option B

B. Option D

C. Option A

D. Option C

Answer: B ([LEAVE A REPLY](#))

NEW QUESTION: 9

Refer to the exhibit.

```
voice class dpg 2000
dial-peer 2001 preference 1
dial-peer 2002 preference 2
dial-peer 2003 preference 3

dial-peer voice 1001 voip
description INBOUND
session protocol sipv2
session target ipv4:10.0.0.1
destination dpg 2000
incoming called-number 57

dial-peer voice 2001 voip
destination-pattern 5506
session protocol sipv2
session target ipv4:10.0.0.2

dial-peer voice 2002 voip
destination-pattern 5507
session protocol sipv2
session target ipv4:10.0.0.3

dial-peer voice 2003 voip
destination-pattern 5507
session protocol sipv2
session target ipv4:10.0.0.4
```

A Cisco UCM user with directory number 4401 dials 5507. and the call is routed to a Cisco Unified Border Element. Which IP address will the call be sent to?

A. 10.0.0.3

B. 10.0.0.4

C. 10.0.0.1

D. 10.0.0.2

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 10

Refer to the exhibit.

```
sent:
INVITE sip:2004@192.168.100.100:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.100.200:5060;branch=z9hG4bKFI1ED
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
Seq: 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

m=0
c=CiscoSystemsSIP-GW-UserAgent 5871 9974 IN IP4 192.168.100.200
s=SIP Call
o=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 Unified Border Element is sending the incorrect media IP address. The IP address of the loopback interface must be advertised in the SDP
- B.** Cisco Unity Connection is configured on G.729 only. Cisco Unity Connection must be reconfigured to support iLBC.
- C.** 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
- D.** Cisco Unified Border Element is not sending any support for DTMF. OTMF configuration must be added to the appropriate dial peer.

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 11

How does an administrator make Cisco IP phone display the last 10 digits of the calling number when the call is in the connected state, and also display the calling number in the E.164 format within call history on the phone?

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 12

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 it.
- B.** When using media flow-around, both call signaling and media do not go through the Cisco Unified Border Element
- C.** When using media flow-through, call signaling goes through the Cisco Unified Border Element but media does not
- D.** When using media flow-around. the call signaling goes through the Cisco Unified Border Element, but media is not passed through it
- E.** When using media flow-through, the call signaling and media are passed through the Cisco Unified Border Element

Answer: D,E ([LEAVE A REPLY](#))

NEW QUESTION: 13

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

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

Answer: B ([LEAVE A REPLY](#))

NEW QUESTION: 14

Which protocol is used between Cisco Jabber clients for instant messaging and presence?

- A.** XMPP
- B.** P2P
- C.** SIP/SIMPLE
- D.** Jabber

Answer: ([SHOW ANSWER](#))

Section: Collaboration Applications

Explanation:

An XMPP connection handles the presence information exchange and instant messaging operations for XMPP-based clients.

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/10_5_1/CUP0_BK_CE43108E_00_config-admin-guide-imp-105/CUP0_BK_CE43108E_00_config-admin-guide-imp-

NEW QUESTION: 15

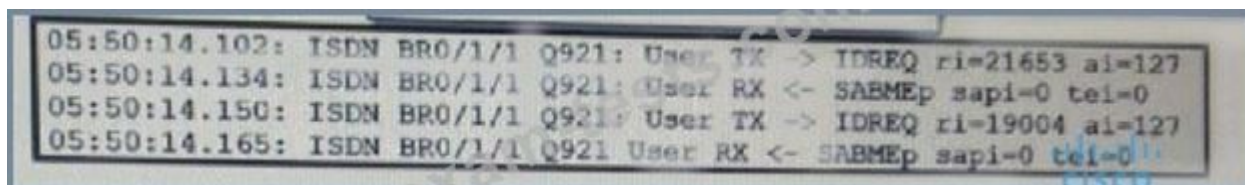
An engineer is configuring Cisco Jabber for Windows and must implement desk phone control mode for some of the users. Which access control group must be configured for those users to enable this functionality?

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

Answer: A ([LEAVE A REPLY](#))

NEW QUESTION: 16

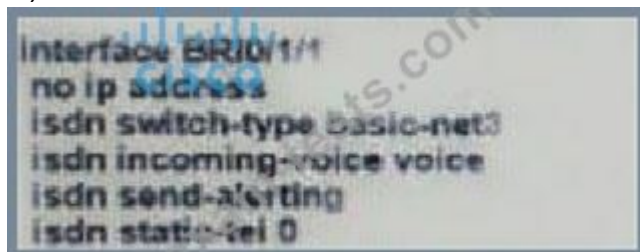
Refer to the exhibit.



```
05:50:14.102: ISDN BR0/1/1 Q921: User TX -> IDREQ ri=21653 ai=127
05:50:14.134: ISDN BR0/1/1 Q921: User RX <- SABMEp sapi=0 tei=0
05:50:14.150: ISDN BR0/1/1 Q921: User TX -> IDREQ ri=19004 ai=127
05:50:14.165: ISDN BR0/1/1 Q921: User RX <- SABMEp sapi=0 tei=0
```

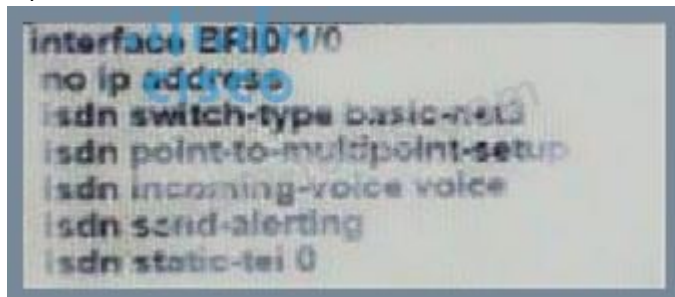
A customer submits this output, captured on a Cisco IOS router. Assuming that an MGCP gateway is configured with a ISDN BRI interface, which BRI changes resolve the issue?

A)



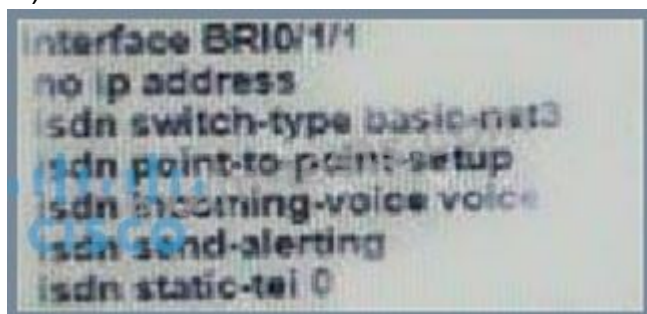
```
interface BRI0/1/1
no ip address
isdn switch-type basic-net3
isdn incoming-voice voice
isdn send-alerting
isdn static-tei 0
```

B)



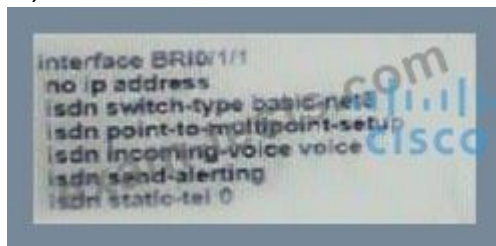
```
interface BRI0/1/0
no ip address
isdn switch-type basic-net3
isdn point-to-multipoint-setup
isdn incoming-voice voice
isdn send-alerting
isdn static-tei 0
```

C)



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

D)



A. Option C

B. Option A

C. Option B

D. Option D

Answer: A ([LEAVE A REPLY](#))

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<https://www.braindumpsPASS.com/Cisco/350-801-practice-exam-dumps.html> (408 Q&As Dumps, **40%OFF** Special Discount: **Exam-Tests**)

NEW QUESTION: 17

Which action is 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 (tunnelled overSSH 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: 18

An administrator must configure the local route group feature on cisco UCM. Which step will enable this feature?

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

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 19

Which action is required if an engineer wants to have Cisco Unified Communications Manager control the configuration for an MGCP gateway?

- A. Configure the Cisco Unified CM's IP in voice service VoIP.
- B. From Cisco Unified CM > Device > Gateway > Add gateway, check the auto-configuration check box.
- C. Apply the ccm-manager configuration commands to the gateway.
- D. Upload the custom configuration in the TFTP server in Cisco Unified CM.

Answer: (SHOW ANSWER)

NEW QUESTION: 20

Refer to the exhibit.



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

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

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 21

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

- A. Mobility Cross Cluster

- B. Intercluster Lookup Service
- C. Location Conveyance on intercluster SIP trunks
- D. intra-Cluster Communication Signaling

Answer: B ([LEAVE A REPLY](#))

NEW QUESTION: 22

Refer to the exhibit.



An engineer is troubleshooting a codec negotiation issue where both endpoints that are involved in the call support the codecs listed in the exhibit. Which audio codec is selected if a call between two endpoints in Region 1 is placed?

- A. G.729
- B. G.711u
- C. G.722
- D. G.711a

Answer: C ([LEAVE A REPLY](#))

NEW QUESTION: 23

A customer wants to deploy Cisco Jabber Phone Mode with contacts. Which two configurations are needed to accomplish this goal? (Choose two.)

- A. Remove the IM and Presence servers from the Cisco UCM service profile
- B. Enable Cisco UCM end-users for IM and Presence
- C. Disable Instant Messaging on IM and Presence
- D. Add the Application Client Users role to the Cisco UCM end users
- E. Import Contacts.xml into Jabber

Answer: B,C ([LEAVE A REPLY](#))

Section: Collaboration Applications

NEW QUESTION: 24

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 IM and Presence Server, and Cisco Video Communication Server
- B. Cisco Unified Border Element, Cisco UCM, and Cisco Video Communication Server
- C. Cisco Expressway-E, Cisco Expressway-C, and Cisco UCM
- D. Cisco Expressway-E, Cisco IM and Presence Server, and Cisco Video Communication Server

Answer: C ([LEAVE A REPLY](#))

<https://www.cisco.com/c/en/us/td/docs/solutions/CVD/Collaboration/enterprise/12x/120/collbcvd/dge.html>

NEW QUESTION: 25

A Cisco UCM administrator wants to enable the Self-Provisioning feature for end users. Which two prerequisites must be met first? (Choose two.)

- A. End users must be associated to a user profile or feature group template that includes a universal line template and universal device template
- B. Cisco Extended Functions service must be running
- C. End users must have a primary extension
- D. End users must belong to Standard CCM Admin Users group, the Standard CCM End Users group, and the Standard CCM Self-Provisioning group
- E. End users must have a secondary extension

Answer: A,C ([LEAVE A REPLY](#))

NEW QUESTION: 26

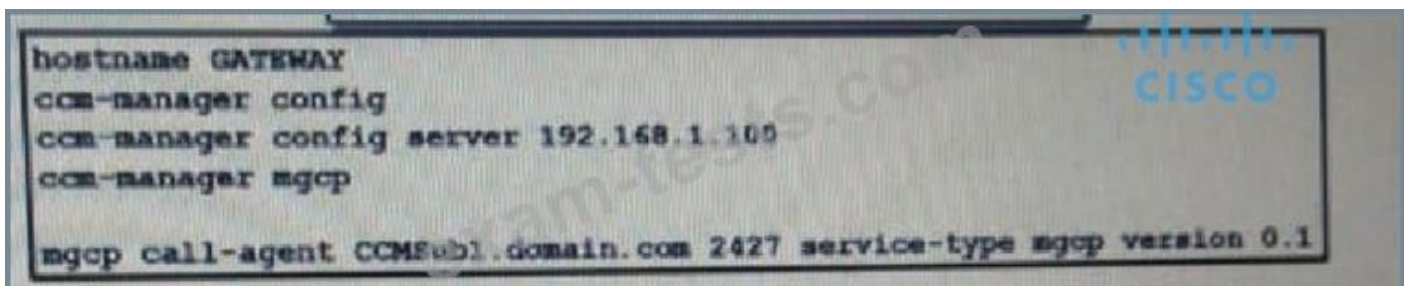
Refer to the exhibit. Which codec should an engineer select for a call made between "Dallas-REG" & "Austin-REG"?

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

Answer: C ([LEAVE A REPLY](#))

NEW QUESTION: 27

Refer to the exhibit.



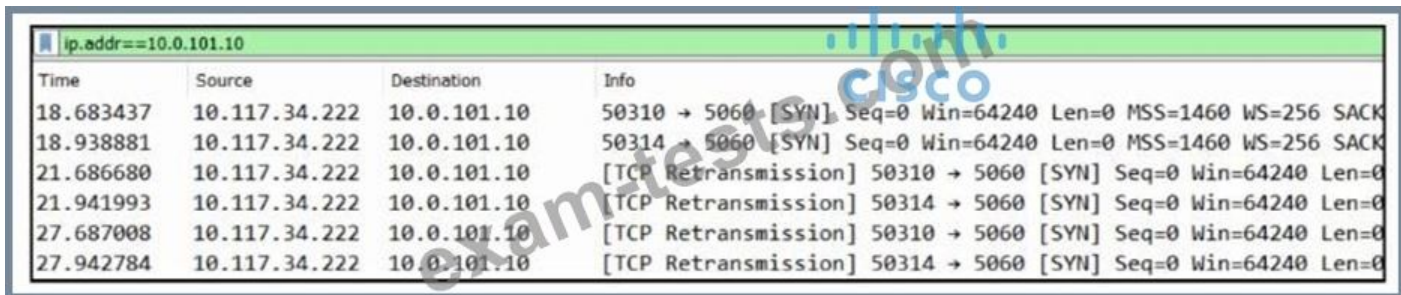
An engineer verifies the configuration of an MGCP gateway. The commands are already configured. Which command is necessary to enable MGCP?

- A. Device(config)# ccm-manager enable
- B. Device (config)# mgcp
- C. Device (config) # com-manager active
- D. Device(config)# mgcp enable

Answer: B ([LEAVE A REPLY](#))

NEW QUESTION: 28

Refer to the exhibit.



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

An administrator is attempting to register a SIP phone to a Cisco UCM but the registration is failing. The IP address of the SIP Phone is 10.117.34.222 and the IP 1 address of the Cisco UCM is 10.0.101.10. Pings from the SIP phone to the Cisco UCM are successful. What is the cause of this issue and how should it be resolved?

- A. An NTP mismatch is preventing connection of the TCP session between the SIP phone and the Cisco UCM. The SIP phone and Cisco UCM must be set with identical NTP sources.
- B. The certificates on the SIP phone are not trusted by the Cisco UCM. The SIP phone must generate new certificates.
- C. DNS lookup for the Cisco UCM FQDN is failing. The SIP phone must be reconfigured with the proper DNS server.
- D. An network device is blocking TCP port 5060 from the SIP phone to the Cisco UCM. This device must be reconfigured to allow traffic from the IP phone.

Answer: ([SHOW ANSWER](#)**)**

NEW QUESTION: 29

A network administrator with ID392116981 has determined that a WAN link between two Cisco UCM clusters supports only 1 Mbps of bandwidth for voice traffic.

How many calls does this link support if G.711 as the audio codec is used?

- A. 13
- B. 12
- C. 16
- D. 15

Answer: ([SHOW ANSWER](#)**)**

NEW QUESTION: 30

How many DNS SRV entries can be defined in the SIP trunk destination address field in Cisco Unified Communications Manager?

- A. 16
- B. 4
- C. 1
- D. 8

Answer: C ([LEAVE A REPLY](#))

NEW QUESTION: 31

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

- A. Use multiple egress queues to provide priority queuing of RTP voice packet streams and the ability to classify or reclassify traffic and establish a network trust boundary.
- B. Use 808.IQ trunking and 802.Ip for proper treatment of Layer 2 CoS packet marking on ports with phones connected.
- C. 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. Map audio and video streams of video calls (AF41 and AF42) to a class-based queue with weighted random early detection.
- E. Implement IP RTP header compression on the serial interface to reduce the bandwidth required per voice call on point-to-point links.

Answer: A,B ([LEAVE A REPLY](#))

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40%OFF Special Discount: Exam-Tests)

NEW QUESTION: 32

Where is the default for maximum session Bit Rate for a region configured?

- A. Enterprise Phone configuration
- B. Enterprise parameters configuration
- C. Service parameter configuration
- D. Region configuration

Answer: C ([LEAVE A REPLY](#))

NEW QUESTION: 33

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 destination phone signals back to the Cisco UCM that it is ringing, and the Cisco UCM signals back to the source phone.
- B.** The source phone must signal to the destination phone that it is ringing, and the destination phone signals back with a 180 Ringing message. The same process applies to 11 BYE messages.
- C.** The calls have an MTP in the call path due to different codec support. The calls are subsequently split into two call legs.
- D.** The 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.

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 34

What is a valid class included in the 8-class QoS strategy in a VoIP network?

- A.** Broadcast video
- B.** Multimedia conferencing
- C.** Real-Time Interactive
- D.** Assured Forwarding

Answer: ([SHOW ANSWER](#)**)**

NEW QUESTION: 35

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

- A.** Alternate Names
- B.** Alternate Extensions
- C.** Attempt Forward routing rule
- D.** Mobile User

Answer: B ([LEAVE A REPLY](#))

NEW QUESTION: 36

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

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

Answer: A,C ([LEAVE A REPLY](#))

NEW QUESTION: 37

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

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

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 38

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

- A. SRST
- B. Route Group
- C. Device Pool
- D. Cisco UCM Group

Answer: A ([LEAVE A REPLY](#))

Section: Infrastructure and Design

Explanation:

Cisco Unified SRST provides Cisco Unified CM with fallback support for Cisco Unified IP phones that are attached to a Cisco router on your local network. Cisco Unified SRST enables routers to provide call-handling support for Cisco Unified IP phones when they lose connection to remote primary, secondary, or tertiary Cisco Unified CM installations or when the WAN connection is down.

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusrst/admin/sccp_sip_srst/configuration/guide/SCCP_and_SIP_SRST_Admin_Guide/srst_overview.html

NEW QUESTION: 39

Which type of greeting in the Call Handler configuration in Cisco Unity Connection overtimes all other greetings?

- A. priority
- B. supervisory
- C. holidays
- D. alternate

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 40

Refer to the exhibit.



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

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

Answer: C ([LEAVE A REPLY](#))

NEW QUESTION: 41

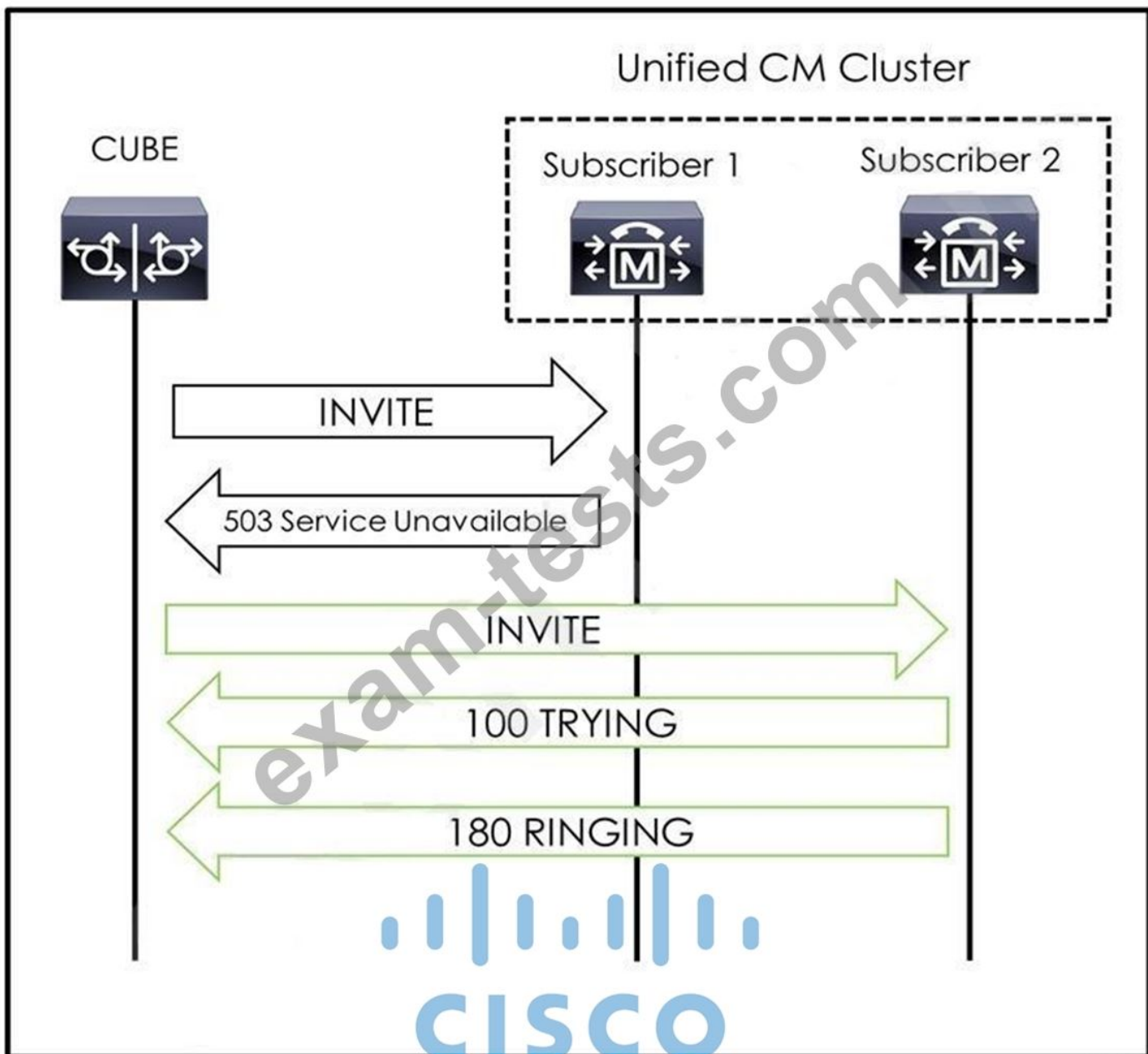
Which Cisco Unified Communications Manager feature is to determine the maximum bit rate for a call between two video-endpoints?

- A. Partitions
- B. Regions
- C. transformations
- D. Locations

Answer: B ([LEAVE A REPLY](#))

NEW QUESTION: 42

Refer to the exhibit. Cisco Unified Border Element is attempting to establish a call with Subscriber 1, but the call fails. Cisco Unified Border Element then retries the same call with Subscriber 2, and the call proceeds normally. Which action resolves the issue?



- A. Verify that the Run On All Active Unified CM Nodes checkbox is enabled.
- B. Verify that the PSTN Access checkbox is enabled.
- C. Verify that the correct calling search space is selected for the Inbound Calls section.
- D. Verify that the Significant Digits field for Inbound Calls is set to All.

Answer: C ([LEAVE A REPLY](#))

NEW QUESTION: 43

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-kpmi
codec aacld
```

B)

```
dial-peer voice 1 voip
translation-profile outgoing Strip9
max-conn 5
destination-pattern 91[2-9].[2-9]...$
session protocol sipv2
session target ipv4:142.45.10.1
dtmf-relay rtp-nte sip-notify sip-kpmi
codec ilbc mode 20
```

C)

```
dial-peer voice 1 voip
translation-profile outgoing Strip9
max-conn 5
destination-pattern 91[2-9].[2-9]...$
session protocol sipv2
session target ipv4:142.45.10.1
dtmf-relay rtp-nte sip-notify sip-kpmi
codec mp4a-latm
```

D)

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

A. Option D

B. Option A

C. Option B

D. Option C

Answer: C ([LEAVE A REPLY](#))

NEW QUESTION: 44

Multiple router patterns match a number. How does Cisco Unified Communications Manager determine which pattern to use?

A. the one that comes first in numerical order

B. the one with the closed match

C. the one with the longest match

D. the one that discards everything PreDot

Answer: ([SHOW ANSWER](#)**)**

NEW QUESTION: 45

What is a QoS requirement for VoIP calls?

A. $\leq 15\%$ packet loss

B. ≤ 150 bps per phone of guaranteed bandwidth

C. ≤ 150 ms of jitter

D. ≤ 150 ms of one-way latency from mouth to ear

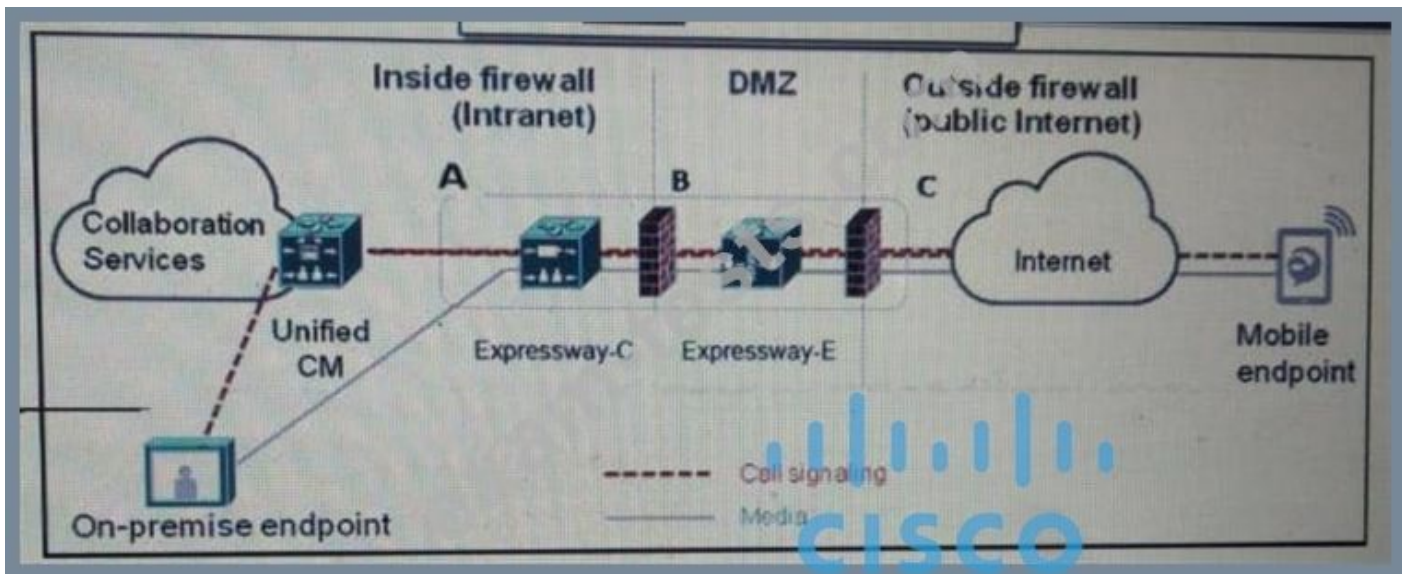
Answer: D ([LEAVE A REPLY](#))

Section: QoS

Explanation/Reference: [https://www.ciscopress.com/articles/article.asp?p=357102#:~:text=One%2Dway%20latency%20\(mouth%20to,and%20Layer%202%20media%20overhead\).](https://www.ciscopress.com/articles/article.asp?p=357102#:~:text=One%2Dway%20latency%20(mouth%20to,and%20Layer%202%20media%20overhead).)

NEW QUESTION: 46

Refer to the exhibit.



When making a call to a MRA client, what are the combinations of protocol on each of the different sections A-B-C?

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

Answer: C ([LEAVE A REPLY](#))

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NEW QUESTION: 47

Refer to the exhibit.

```

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

```

```

v=0
o=Cisco-SIPUA 26529 0 IN IP4 10.10.10.84
s=SIP Call
b=AS:4064
t=0 0
m=audio 32136 RTP/AVP 114 9 124 113 115 0 8 116 18
c=IN IP4 10.10.10.84
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114 maxplaybackrate=16000;spop-maxcapture=16000;maxaveragebitrate=64000;stereo=0;spop-
stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0;mode-change-capability=2
a=fmtp:113 octet-align=1;mode-change-capability=2

```

```

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;spop-maxcapture=16000;maxaveragebitrate=64000;stereo=0;spop-
stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0;mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1;mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000

```

```

c=IN IP4 10.10.10.84
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114 maxplaybackrate=16000;spop-maxcapture=16000;maxaveragebitrate=64000;stereo=0;spop-
stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0;mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1;mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=sendrecv

```

When a UC Administrator is troubleshooting DTMF negotiated by this SIP INVITE, which two messages should be examined next to further troubleshoot the issue? (Choose two.)

- A. SUBSCRIBE
- B. NOTIFY
- C. PRACK
- D. REGISTER
- E. UPDATE

Answer: A,B ([LEAVE A REPLY](#))

NEW QUESTION: 48

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\(.*\)/ /&/
```

A.

```
voice translation-rule 1
 rule 1 /9?911/ /911/
 rule 2 /^9\(.*\)/ /\0/
```

B.

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

C.

D.

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 49

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

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

Answer: B ([LEAVE A REPLY](#))

NEW QUESTION: 50

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

- A. alternate extensions
- B. message settings

- C. transfer rules
- D. greetings

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 51

What dialed number match this cisco UCM route pattern?

1[23]XX

- A. 1200 through 1300 only
- B. 12300 through 12399 only
- C. 1200 through 1399 only
- D. 1230 through 1239 only

Answer: C ([LEAVE A REPLY](#))

NEW QUESTION: 52

Refer to the exhibit.

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

Refer to the exhibit. An administrator configured a codec preference list with G.722 and ILBC codecs. Which change must the administrator make in the dial-peer section of the configuration to use this list?

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 53

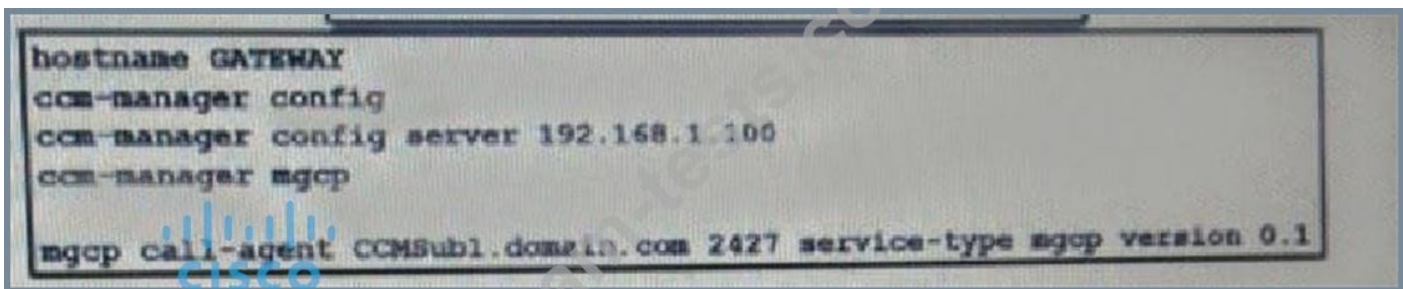
Which datastore and protocol is used for saving back-up files within the Disaster Recovery System of Cisco UCM?

- A. remote disk on the SFTP server
- B. remote disk on a CIFS share
- C. remote disk on an NFS share
- D. local disk on the Cisco UCM server

Answer: A ([LEAVE A REPLY](#))

NEW QUESTION: 54

Refer to the exhibit.



An engineer verifies the configured of an MGCP gateway. The commands are already configured. Which command is necessary to enable MGCP?

- A. Device(config)# ccm-manager enable
- B. Device (config)# mgcp
- C. Device (config) # com-manager active
- D. Device(config)# mgcp enable

Answer: ([SHOW ANSWER](#)**)**

NEW QUESTION: 55

Refer to the exhibit. Endpoint A calls endpoint B.

What is the only audio codec that can be used for the call?

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
```

- A. G722/8000
- B. PCMA/8000
- C. Telephone-event/8000
- D. G7221/16000

Answer: ([SHOW ANSWER](#))

Endpoint B is not supported G722/8000, both endpoints only supported G721/16000.

NEW QUESTION: 56

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

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

Answer: C ([LEAVE A REPLY](#))

Section: Infrastructure and Design

Explanation/Reference:

https://www.cisco.com/c/en/us/td/docs/switches/lan/catalyst2960/software/release/12-2_40_se/configuration/guide/scg/swvoip.pdf

NEW QUESTION: 57

A DTMS 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. Trusted relay point

B. Annunciator

C. Conference Bridge

D. Media Termination point

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 58

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

A. Reboot the VCS server and attempt reregistration.

B. Correct the restriction policy settings.

C. Upload a new policy In VCS.

D. Change the SIP trunk configuration.

Answer: B ([LEAVE A REPLY](#))

NEW QUESTION: 59

How many DNS SRV entries can be defined in the SIP trunk destination address field in Cisco Unified Communications Manager?

A. 1

B. 8

C. 16

D. 4

Answer: ([SHOW ANSWER](#)**)**

Explanation/Reference:

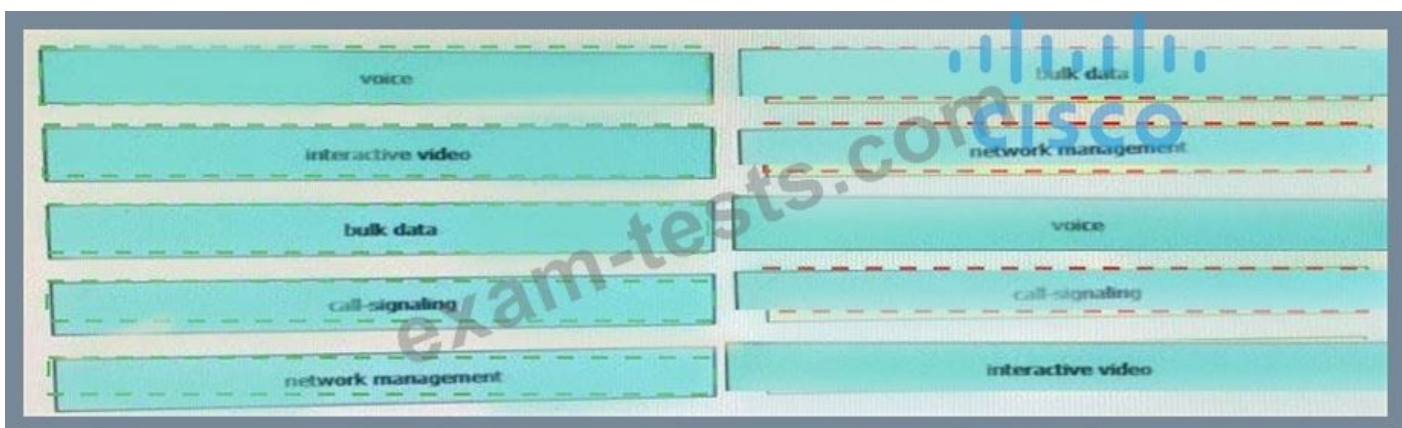
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/CUCM_BK_SE5DAF88_00_cucm-system-configuration-guide-1151/CUCM_BK_SE5DAF88_00_cucm-system-configuration-guide-1151_chapter_01110.html

NEW QUESTION: 60

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



Answer:



Explanation

Voice = EF

Interactive Video = AF41

Bulk Data = AF11

Call-Signaling = AF31/CS3

Network Management = CS2

https://www.cisco.com/c/en/us/td/docs/solutions/Enterprise/WAN_and_MAN/QoS_SRND/QoS-SRND-Book/Qo

NEW QUESTION: 61

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

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

Answer: ([SHOW ANSWER](#))

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<https://www.braindumpspass.com/Cisco/350-801-practice-exam-dumps.html> (408 Q&As Dumps, **40%OFF Special Discount: Exam-Tests**)

NEW QUESTION: 62

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

- A. Switch# copy running-config startup-config
- B. Switch(config)# interface GigabitEthernet1/0/1
- C. Switch(config)# no lldp run
- D. Switch# configure terminal
- E. Switch(config)# no lldp transmit

Answer: C,D ([LEAVE A REPLY](#))

NEW QUESTION: 63

An engineer must configure a cisco ISR 4000 as an MGCP gateway to download its MGCP- specific configuration from cisco UCM. Which Cisco IOS configuration snippet accomplishes this task?

1 ISR4K(config)# ip tftp source-interface GigabitEthernet 0/0/1
ISR4K(config)# mgcp call-agent 10.1.2.3 2427 service-type mgcp version 0.1
ISR4K(config)# ccm-manager config server 10.1.2.3
ISR4K(config)# ccm-manager config
ISR4K(config)# ccm-manager mgcp
ISR4K(config)# mgcp

2 ISR4K(config)# ip tftp source-interface GigabitEthernet 0/0/1
ISR4K(config)# mgcp call-agent 10.1.2.3 2427 service-type mgcp version 11.5.1
ISR4K(config)# ccm-manager config server 10.1.2.3
ISR4K(config)# ccm-manager config
ISR4K(config)# ccm-manager mgcp
ISR4K(config)# mgcp

3 ISR4K(config)# ip tftp source-interface GigabitEthernet 0/0/1
ISR4K(config)# mgcp call-agent 10.1.2.3 2427 service-type mgcp version 0.1
ISR4K(config)# ccm-manager config server 10.1.2.3
ISR4K(config)# ccm-manager config download
ISR4K(config)# ccm-manager mgcp
ISR4K(config)# mgcp

4 ISR4K(config)# ip tftp source-interface GigabitEthernet 0/0/1
ISR4K(config)# mgcp call-agent 10.1.2.3 2428 service-type mgcp version 0.1
ISR4K(config)# ccm-manager config server 10.1.2.3
ISR4K(config)# ccm-manager config
ISR4K(config)# ccm-manager mgcp
ISR4K(config)# mgcp

A. Option B

B. Option C

C. Option A

D. Option D

Answer: C ([LEAVE A REPLY](#))

NEW QUESTION: 64

Refer to the exhibit.



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

- A. G7221/16000
- B. PCMA/8000
- C. Telephone-event/8000
- D. G277/8000

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 65

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

- A. Set the Call Classification to OnNet for the international route pattern.
- B. Block international dial patterns in the SIP trunk CSS.
- C. Configure a Forced Authorization Code on the international route pattern.
- D. Set Call Forward All CSS to restrict international dial patterns.
- E. Check Route Next Hop By Calling Party Number on the international route pattern.

Answer: ([SHOW ANSWER](#)**)**

Explanation/Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmetrans.html

NEW QUESTION: 66

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

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

Answer: ([SHOW ANSWER](#)**)**

NEW QUESTION: 67

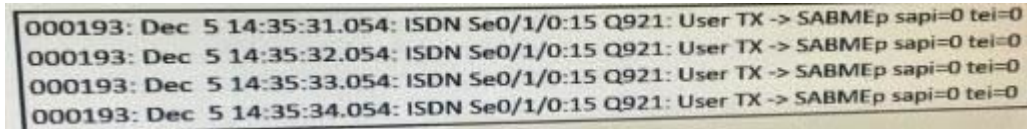
Which endpoint feature is supported using Mobile and Remote Access through Cisco Expressway?

- A. SSO
- B. SRST
- C. MGCP gateway registration
- D. H.323 registration proxy to Cisco Unified Communications Manager

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 68

Refer the exhibit.



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

Given this "debug isdn q921" output, what is the problem with the PRI?

- A. Layer 1 is down on the controller.
- B. PRI does not have an IP address configured on the interface.
- C. Noting, the PRI is sending keepalives.
- D. Layer 2 is down on the controller.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 69

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

Region Configuration

Save Delete Reset Apply Config Add New

Region Information

Name *

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls
SanJose-REG	Use System Default (Factory Default low loss)	24 kbps (AMR-WB)	Use System Default (384 kbps)
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default

Modify Relationship to other Regions

Regions	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls
Austin-REG Dallas-REG Default SanJose-REG	Keep Current Setting	Keep Current Setting	Keep Current Setting

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

Answer: ([SHOW ANSWER](#))

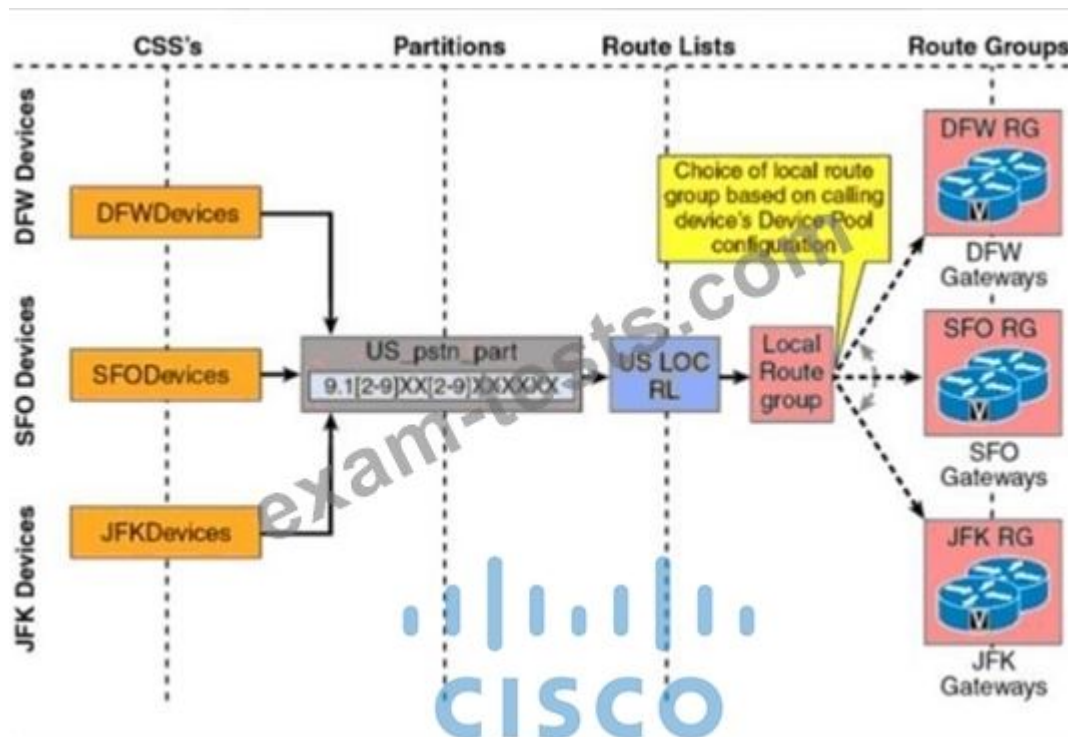
NEW QUESTION: 70

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

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

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 71



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 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
- B. The call egresses in New York because the device automatically is assigned a New York device pool and uses the local gateway
- C. 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
- D. The call fails because the Standard Local Route Group is being used only if no configuration is set for the device pools

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 72

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

- A. Balanced mode provides user load failover in the event of an outage. Active/standby mode provides an always on standby node in the event of an outage, and is also provides load balancing.
- B. Balanced mode provides user load balancing and user failover only for manually generated failovers. Active/standby mode provides an unconfigured standby node in the event of an outage, but it does not provide load balancing.

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

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

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

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

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 73

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. 38.4 kbps

B. 31.2 kbps

C. 87.2 kbps

D. 55.2 kbps

Answer: C ([LEAVE A REPLY](#))

NEW QUESTION: 74

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. UDP 161

D. TCP 161

Answer: A ([LEAVE A REPLY](#))

Section: Protocols, Codecs, and Endpoints

Explanation/Reference:

https://www.cisco.com/c/en/us/td/docs/net_mgmt/prime/collaboration/12-1/assurance/advanced/guide/cpco_b_cisco-prime-collaboration-assurance-guide-advanced-12-1/cpco_b_cisco-prime-collaboration-assurance-guide-advanced-12-1_chapter_01111.html

NEW QUESTION: 75

Which two types of device are supported by the Bulk Administration Tool? (Choose two.)

A. H.322

- B. Cisco Unified IP phones (all models)
- C. SIP trunks
- D. H.225 trunks
- E. music on hold servers

Answer: (SHOW ANSWER)

Section: Protocols, Codecs, and Endpoints

NEW QUESTION: 76

```
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=381claba7a78002c558eda31-12b8af63
To: <sip:1@10.10.10.219>
Call-ID: 381claba-7a78000d-4ca6894a-41dd3e0f@10.10.10.84
Max-Forwards: 70
CSeq: 101 INVITE
Contact: <sip:1234@10.10.10.84:50083;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Allow-Events: kpml,dialog
Content-Type: application/sdp
Content-Length: 658

v=0
o=Cisco-SIPUA 26529 0 IN IP4 10.10.10.84
s=SIP Call
b=AS:4064
t=0 0
m=audio 32136 RTP/AVP 114 9 124 113 115 0 8 116 18
c=IN IP4 10.10.10.84
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114
maxplaybackrate=16000;sprop-maxcapture=16000;maxaveragebitrate=64000;stereo=0;sprop-
stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0,mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1,mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=sendrecv
```

Refer to the exhibit. When a UC Administrator is troubleshooting DTMF negotiated by this SIP INVITE, which two messages should be examined next to further troubleshoot the issue? (Choose two.)

- A. REGISTER
- B. UPDATE
- C. PRACK
- D. NOTIFY
- E. SUBSCRIBE

Answer: D,E (LEAVE A REPLY)

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<https://www.braindumpsPass.com/Cisco/350-801-practice-exam-dumps.html> (408 Q&As Dumps, **40%OFF Special Discount: Exam-Tests**)

NEW QUESTION: 77

How does an administrator make a Cisco IP phone display the last 10 digits of the calling number when the call is in the connected state, and also display the calling number in the E.164 format within call history on the phone?

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 78

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 79

Which configuration tells a switch port to send Cisco Discovery protocol packets that configure an attached Cisco IP phone to trust tagged traffic that is received from a device that is connected to the access port on the Cisco IP phone?

A)

```
Router# configure terminal
Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos trust extend cos 5
```

B)

```
Router# configure terminal
Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos trust extend
```

C)

```
Router# configure terminal
Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos trust extend cos 3
```

D)

```
Router# configure terminal
Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos extend trust
```

A. Option D

B. Option C

C. Option A

D. Option B

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 80

Refer to the exhibit.

```
dspfarm profile 1 transcode universal
  codec g729r8
  codec g729ar8
  codec g711ulaw
  codec g711alaw
  maximum sessions 8
  associate application SCCP
```

Which two codec permutations should be transcoded by this dspfarm? (Choose two)

A. iLBC to G.711ulaw

B. G.729ar8 to G.711alaw

C. G.729r8 to G711ulaw

D. G.729br8toG711alaw

E. G.722 to G 729r8

Answer: ([SHOW ANSWER](#)**)**

NEW QUESTION: 81

Refer to the exhibit.

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

What is the result of applying these two rules to a voice translation-profile for use with an ISDT1 PRI on a Cisco Voice gateway?

A. The leading Plus is stripped from the numeric phone number.

B. The ISDN Plan is modified to the administrators defined value.

C. The ISDN Type is modified to the administrators defined value.

D. Any Zero is stripped from the numeric phone number.

Answer: C ([LEAVE A REPLY](#))

NEW QUESTION: 82

You are adding regions in Cisco Unified Communications Manager. Which codec(s) are selected when a call is placed if you set up the max audio bit rate to use 8 kbps?

A. G.722

B. G.729 and G.711ulaw

C. G.729

D. G.711ulaw and G.711alaw

Answer: C ([LEAVE A REPLY](#))

NEW QUESTION: 83

Refer to the exhibit.

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

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

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. UPDATE

B. NOTIFY

C. PRACK

D. REGISTER

E. SUBSCRIBE

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 84

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

A)

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

B)

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

C)

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

D)

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

A. Option B

B. Option C

C. Option A

D. Option D

Answer: C ([LEAVE A REPLY](#))

NEW QUESTION: 85

Refer to the exhibit.

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

Which Codec is negotiated?

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 86

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 = 8005551212.1122113
- B. Pattern = 8005xxxxxx
- C. Pattern = 3.800xxxxxx
- D. Pattern = 1122113.8005551212

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 87

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

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

Answer: C ([LEAVE A REPLY](#))

NEW QUESTION: 88

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

- A. Cisco XCP XMPP Federation Manager
- B. Cisco XCP Text Conference Manager
- C. Cisco XCP Router
- D. Cisco SIP Proxy

Answer: B ([LEAVE A REPLY](#))

NEW QUESTION: 89

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

- A. Maximum Number of Phones service parameter on the Publisher
- B. Maximum Number of Registered and Unregistered Devices service parameter on each node
- C. Maximum Number of Registered Devices service parameter on each node
- D. Minimum Number of Phones service parameter on each node

Answer: C ([LEAVE A REPLY](#))

NEW QUESTION: 90

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

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

Answer: C ([LEAVE A REPLY](#))

Explanation/Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0100111.pdf

NEW QUESTION: 91

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



Answer:



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NEW QUESTION: 92

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 93

Calls are being delivered to the end user in the globalized format. Where does an engineer configure the calling number into a localized format?

- A. gateway
- B. service parameters
- C. route pattern
- D. IP phone

Answer: [\(SHOW ANSWER\)](#)

NEW QUESTION: 94

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

Answer: B,C [\(LEAVE A REPLY\)](#)

NEW QUESTION: 95

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

- A. Do not wait for the tones immediately dial the FAC and CMC
- B. Change the code if the problem persists i/i-Advise the user to hang up and try again
- C. Adjust the T302 timer from the default of 15 seconds to 5 seconds to shorten the interdigit timer
- D. Advise the user to press the "#" button after dialling the FAC and CMC codes

Answer: C,D [\(LEAVE A REPLY\)](#)

NEW QUESTION: 96

Where is urgent priority enabled to bypass the T302 timer?

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

Answer: A [\(LEAVE A REPLY\)](#)

NEW QUESTION: 97

Which Cisco Unified Communications Manager feature is to determine the maximum bit rate for a call between two video-endpoints?

- A. Partitions
- B. Locations

D. transformations

Regions provide capacity controls for Unified Communications Manager multi-site deployments where you may need to limit the bandwidth for certain calls. For example, you can use regions to limit the bandwidth for calls that are sent across a WAN link, while maintaining a higher bandwidth for internal calls. You can use regions to limit the bandwidth for audio and video calls by setting the maximum bitrate for intraregional or interregional calls to whatever the region(s) can provide.

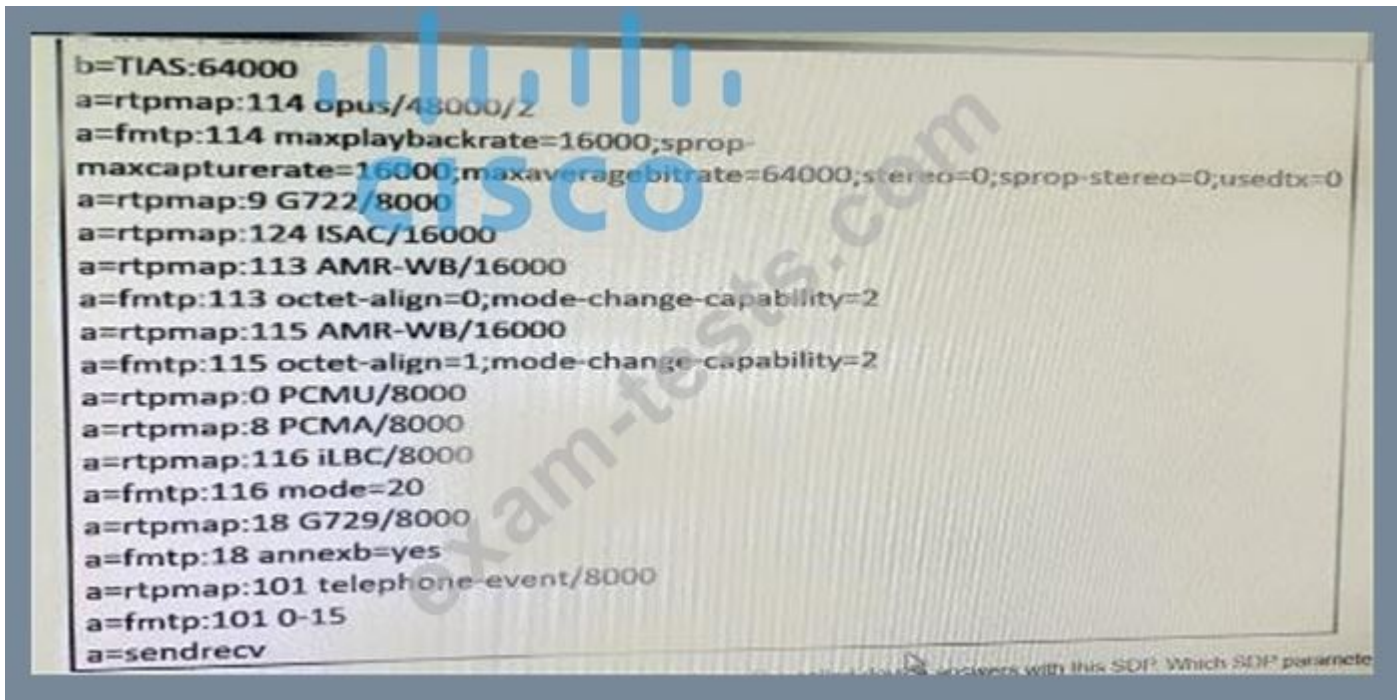
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/11_5_1_SU1/cucm_b_system-configuration-guide-1151su1/cucm_b_system-configuration-guide-1151su1_chapter_0111.html

Which packet delay is the maximum supported between Cisco Unified Communications Manager nodes for clustering over WAN deployments?

- Answer: D (LEAVE A REPLY)**

Refer to the exhibit.





A call is failing to establish between two SIP Devices. The Called device answer with this SDP. Which SDP parameter causes this issue?

- A. The media stream is set to sendonly.
- B. The payload for G.711ulaw must be 18.
- C. The RTP port is set to 0.
- D. The calling device did not offer aptime value.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 100

Which SNMP service must be activated manually on the Cisco Unified Communications Manager after installation?

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

Answer: A ([LEAVE A REPLY](#))

SNMP Master Agent serves as the primary service for the MIB interface. You must manually activate Cisco CallManager SNMP service; all other SNMP services should be running after installation.

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/service/9_0/admin/CUCM_BK_C136FE37_00_cisco-unified-serviceability-administration-90/CUCM_BK_C136FE37_00_cisco-unified-serviceability-administration-guide_chapter_0101.html

NEW QUESTION: 101

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

B. dial-peer voice 1 voip
translation-profile outgoing Strip9
max-conn 5
destination-pattern 91[2-9]..[2-9].....\$
session protocol sipv2
session target ipv4:142.45.10.1
dtmf-relay rtp-nte sip-notify sip-kpml
codec ilbc mode 20

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

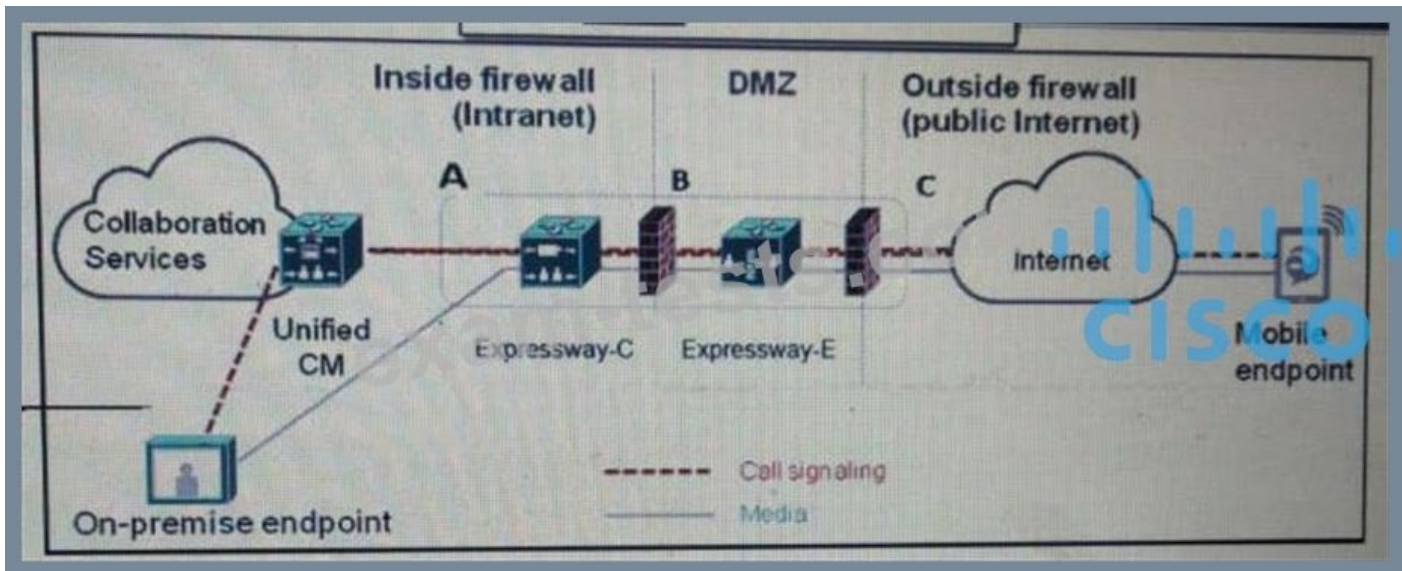
D. dial-peer voice 1 voip
translation-profile outgoing Strip9
max-conn 5
destination-pattern 91[2-9]..[2-9].....\$
session protocol sipv2
session target ipv4:142.45.10.1
dtmf-relay rtp-nte sip-notify sip-kpml
codec mp4a-latm

Answer: B ([LEAVE A REPLY](#))

Explanation/Reference:

NEW QUESTION: 102

Refer to the exhibit.



When making a call to a MRA client, what are the combinations of protocol on each of the different sections A-B-C?

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 103

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

- A. Configure all forced code on all router.
- B. Set call forward All CSS to restrict international dial patterns.
- C. Block international dial patterns in the SIP trunk CSS.
- D. Configure a Forced Authorization Code on the international route pattern.
- E. Set the call Classification to OnNet for the international route pattern.

Answer: B,D ([LEAVE A REPLY](#))

NEW QUESTION: 104

An end user at a remote site is trying to initiate an ad hoc conference call to an end user at the main site. The conference bridge is configured to support G.711 remote user phone only. Support G.729. The remote end user receives an error message on the phone "cannot complete conference call." What is the cause of the issue?

- A. A software conference bridge is not assigned
- B. The transcoder resource is missing.
- C. A Media Termination Point is missing
- D. The remote phone does not have the conference feature assigned.

Answer: B ([LEAVE A REPLY](#))

NEW QUESTION: 105

Refer to the exhibit.

```

INVITE sip:40008172.16.2.143 SIP/2.0
Via: SIP/2.0/TLS 172.16.2.143:5061;branch=c9b9483f3c15k7
Remote-Party-ID: <sip:140083350008172.16.2.143>;party=calling;screen=no;privacy=off
From: <sip:140863350008172.27.2.143>;tag=784255f6-980
To: <sip:40008172.16.2.143>
Date: Tue, 06 Aug 2019 15:03:05 GMT
Call-ID: 4KA4343-87711189-8A4AFFCF-10B6D71B8172.16.2.143
Supported: 100rel,timer,resource-priority,replaces,rdp-annot
Min-SE: 1000
Cisco-Quid: 0002391505-3077640661-2319777743-0380410075
User-Agent: Cisco-SIPGateway/IOS-15.8.3-Sub
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1545089565
Contact: <sip:140863350008172.16.2.143:5061>;transport=tls
Expires: 100
Allow-Events: telephone-event
Max-Forwards: 40
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 416
v=0
o=CiscoSystemsSIP-GW-UserAgent 8406 0200 IN IP4 172.16.2.143
s=SIP Call
c=IN IP4 172.16.2.143
t=0 0
m=audio 44612 RTP/SAVP 0 101
a=rtpmap:101 telephone-event/8000
m=telephone-event/8000
a=rtpmap:101 telephone-event/8000
a=extmap:101 urn:ietf:params:cioc What must be done for this call to succeed?
a=extmap:20 urn:ietf:params:cioc

```

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

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

Answer: B (LEAVE A REPLY)

NEW QUESTION: 106

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

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

Answer: C (LEAVE A REPLY)

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NEW QUESTION: 107

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.711alaw
- B. G.711ulaw
- C. G.722
- D. iLBC

Answer: B ([LEAVE A REPLY](#))

NEW QUESTION: 108

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



Answer:



NEW QUESTION: 109

Which two DNS records must be created to configure Service Discovery for on-premises Jabber? (Choose two.)

- A. _cuplogin._tcp.cisco.com pointing to a record of IM&P

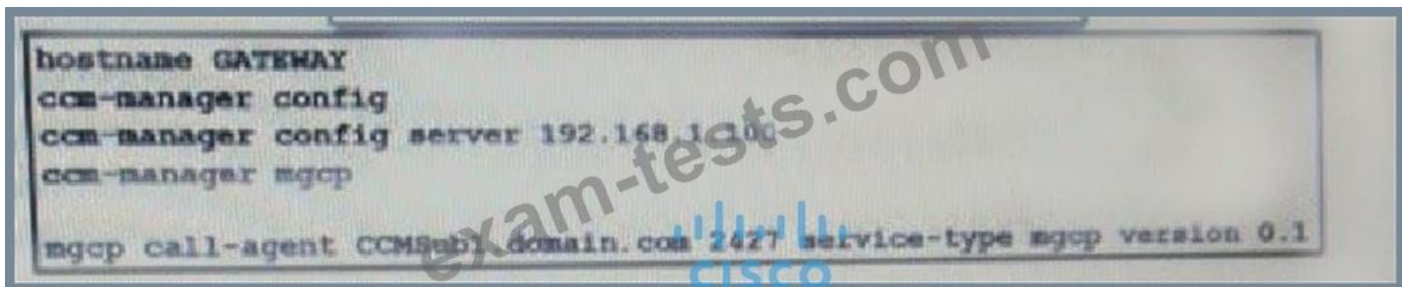
- B. _cisco-uds._tcp.cisco.com pointing to a record of Cisco Unified CM
- C. _cuplogin._tls.cisco.com pointing to the IP address of IM&P
- D. _cisco-uds._tls.cisco.com pointing to the IP address of Cisco Unified Communications manager
- E. _xmpp._tls.cisco.com pointing to a record of IM&P

Answer: A,B (LEAVE A REPLY)

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/jabber/Windows/9_7/CJAB_BK_C606D8A9_00_cisco-jabber-dns-configuration-guide/CJAB_BK_C606D8A9_00_cisco-jabber-dns-configuration-guide_chapter_010.html

NEW QUESTION: 110

Refer to the exhibit.



An engineer verifies the configured of an MGCP gateway. The commands are already configured. Which command is necessary to enable MGCP?

- A. Device (config) # com-manager active
- B. Device(config)# ccm-manager enable
- C. Device(config)# mgcp enable
- D. Device (config)# mgcp

Answer: D (LEAVE A REPLY)

NEW QUESTION: 111

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

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

Answer: (SHOW ANSWER)

NEW QUESTION: 112

How is Video Call Traffic Class defined on Cisco Unified CM for Enhanced Location CAC bandwidth deductions?

- A. On Device Profile page
- B. On each device or Device Pool's configuration page
- C. On SIP Profile configuration page
- D. On Location configuration page

E. On each device's configuration page

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 113

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

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

Answer: A ([LEAVE A REPLY](#))

NEW QUESTION: 114

A SIP phone has been configured in the system with MAC address 0030 96D2 D5CB The phone retrieves the configuration file from the Cisco UCM. Which naming format is the file that is downloaded?

- A. SIP0030406706705.cnf
- B. SIPOO3O96D2D5CB.cnf.xml
- C. SEP0030406706705.cnf
- D. SEP003096D2D5CB.cnf.xml

Answer: ([SHOW ANSWER](#)**)**

NEW QUESTION: 115

An engineer implements QoS in the enterprise network. Which command can to verify the correct classification and marking on a cisco IOS switch?

- A. show policy-map
- B. show class-map interface GigabitEthernet 1/0/1
- C. show access-lists
- D. show policy-map interface GigabitEthernet 1/0/1

Answer: D ([LEAVE A REPLY](#))

https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/qos_classn/configuration/xr-16/qos-classn-xr-16-book/qos-classn-mrkg-ntwk-trfc-xr.html

NEW QUESTION: 116

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. Create a centralized dial plan with a Cisco UCM Session Management Edition cluster or a Cisco gatekeeper.
- B. Implement ILS with GDPR so the dial plan can dynamically replicate across clusters.

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

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

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 117

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



Answer:



NEW QUESTION: 118

Which open-standard protocol is the basis for the Extensible Messaging and Presence Protocol?

A. XML

B. HTTP

C. Json

D. SIP

Answer: A ([LEAVE A REPLY](#))

NEW QUESTION: 119

Which two DNS records must be created to configure Service Discovery for on-premises Jabber? (Choose two.)

A. _xmpp.tls.cisco.com pointing to a record of IM and Presence

- B. _cisco-uds.tcp.cisco.com pointing to a record of Cisco UCM
- C. _cisco-uds._tls.cisco.com pointing to the IP address of Cisco UCM
- D. _cuplogin._tls.cisco.com pointing to the IP address of IM and Presence
- E. _cuplogin_tcp.cisco.com pointing to a record of IM and Presence

Answer: B,E ([LEAVE A REPLY](#))

NEW QUESTION: 120

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

```
voice translation-rule 40
rule 1 /3...$/ /4085558/
!
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)

The screenshot shows the Cisco Configuration Assistant (CCA) interface for configuring a dial-peer. The 'Pattern Definition' section is expanded, showing the following fields:

- Pattern*: \+1
- Partition: PT_US_VG_CD_Out_xForm
- Description: US International calling
- Numbering Plan: E.164
- Route Filter: < None >
- ☒ Urgent Priority
- ☐ MLPP Preemption Disabled

The 'Called Party Transformations' section is also expanded, showing the following fields:

- Discard Digits: PreDot
- Called Party Transformation Mask:
- Prefix Digits: 9011
- Called Party Number Type*: Unknown
- Called Party Numbering Plan*: Unknown

B)

Pattern Definition

Pattern*

Partition

Description

Numbering Plan

Route Filter

☒ Urgent Priority

☐ MLPP Preemption Disabled

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type*

Called Party Numbering Plan*

C)

Pattern Definition

Pattern*

Partition

Description

Numbering Plan

Route Filter

☒ Urgent Priority

☐ MLPP Preemption Disabled

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type*

Called Party Numbering Plan*

D)

Pattern Definition

Pattern*

Partition

Description

Numbering Plan

Route Filter

☒ Urgent Priority

☐ MLPP Preemption Disabled

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type*

Called Party Numbering Plan*

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

D. Option D

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 121

An engineer is configuring a BOT device for a Jabber user in Cisco Unified Communication Manager. Which phone type must be selected?

- A. Cisco Unified Client Services Framework
- B. Cisco Dual Mode for Android
- C. third-party SIP device
- D. Cisco Dual Mode for iPhone

Answer: B ([LEAVE A REPLY](#))

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NEW QUESTION: 122

Which command in the MGCP gateway configuration defines the secondary Cisco Unified Communications Manager server?

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

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 123

Which recommendation is the best practice for marking video and voice media in a cisco Unified Communications network?

A)

Voice Cos 6 (IP Precedence 4, PHB AF41, or DSCP 24)
Video Cos 5 (IP Precedence 4, PHB EF, or DSCP 34)

B)

Voice Cos 5 (IP Precedence 5, PHB EF, or DSCP 46)
Video Cos 4 (IP Precedence 4, PHB AF41, or DSCP 34)

C)

Voice Cos 5 (IP Precedence 2, PHB EF, or DSCP 48)
Video Cos 4 (IP Precedence 4, PHB AF41, or DSCP 46)

D)

Voice Cos 5 (IP Precedence 6, PHB AF41, or DSCP 16)
Video Cos 4 (IP Precedence 5, PHB EF, or DSCP 32)

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

Answer: A ([LEAVE A REPLY](#))

NEW QUESTION: 124

Which configuration on Cisco Unified Communication Manager is required for SIP MWI to work?

- A. The line partition must be inside the inbound CSS assigned to the CUC SIP trunk.
- B. Assign a MWI extension on the mailbox.
- C. The line partition must be inside the rerouting CSS assigned to the Cisco assigned to the Cisco unity connection SIP trunk.
- D. Set the "Enable message waiting indicator" on the port group.

Answer: A ([LEAVE A REPLY](#))

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/10x/integration/guide/cucm_sip/cucintcucmsip/cucintcucmsip060.html

NEW QUESTION: 125

An engineer is designing a load balancing solution for two Cisco Unified Border routers. The first router (cube1.abc.com) takes 60% of the calls and the second router (cube2.abc.com) takes 40% of the calls, Assume all DNS A records have been created.

Which two SRV records are needed for a load balanced solution? (Choose two.)

- A. sip_udp.abc.com 60 IN SRV 2 60 5060 cube1.abc.com
- B. sip_udp.abc.com 60 IN SRV 60 1 5060 cube1.abc.com
- C. sip_udp.abc.com 60 IN SRV 1 60 5060 cube1.abc.com
- D. sip_udp.abc.com 60 IN SRV 1 40 5060 cube2.abc.com
- E. sip_udp.abc.com 60 IN SRV 3 60 5060 cube2.abc.com

Answer: C,D ([LEAVE A REPLY](#))

NEW QUESTION: 126

An administrator recently upgraded a Cisco Webex DX80 through its web interface but discovered the next morning that the unit has received to its previous version. What must the administrator do to prevent this from happening again?

- A. Set the prepare cluster for rollback to pre-8-0 enterprise parameter to true.
- B. Assign a phone security profile with secure SIP.

- C. Confirm the phone load name in the phone configuration.
- D. Assign a universal device template to the phone.

Answer: C ([LEAVE A REPLY](#))

NEW QUESTION: 127

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

- A. Not more than 8%
- B. Not more than 3%
- C. Not more than 5%
- D. Not more than 1%

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 128

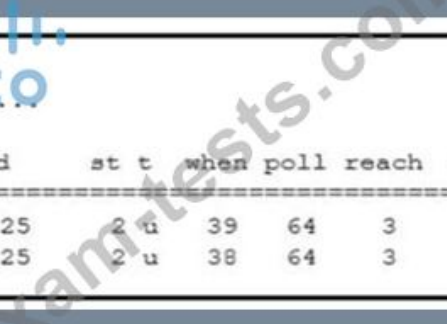
Which task is required when configuring self-provisioning for an end user in Cisco UCM?

- A. Associate the end user to the Standard CCM Super Users group.
- B. Enable Auto-Registration.
- C. Associate the end user to a SIP Profile.
- D. Disable Auto-Registration.

Answer: B ([LEAVE A REPLY](#))

NEW QUESTION: 129

Refer to the exhibit.



admin:utils ntp status									
ntpd (pid 17428) is running...									
remote	refid	st	t	when	poll	reach	delay	offset	jitter
*192.168.1.1	17.253.14.125	2	u	39	64	3	0.456	-0.236	0.116
*192.168.1.2	17.253.14.125	2	u	38	64	3	0.817	-0.695	0.395

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

- A. Stop the NTP service on Cisco UCM.
- B. Enable NTP authentication for the new NTP server on Cisco UCM.
- C. Delete the original NTP server from Cisco UCM.
- D. Restart the NTP service on Cisco UCM.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 130

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](#))

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