

Cisco.350-801.v2023-06-21.q172

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https://www.exam-tests.com/350-801-exam/Cisco.350-801.v2023-06-21.q172.html	

NEW QUESTION: 1

Refer to the exhibit. An administrator configures fax dial-peers on a Cisco IOS gateway and finds that faxes are not working correctly. Which change should be made to resolve this issue?

- A. codec g723ar63
- B. codec g711ulaw
- C. codec g729br81
- D. codec g726r32

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

NEW QUESTION: 2

Which transport protocol does the application layer protocol SNMP use?

- A. HTTP
- B. XML
- C. UDP
- D. SIP

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

<https://www.geeksforgeeks.org/simple-network-management-protocol-snmp/>

NEW QUESTION: 3

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

NEW QUESTION: 4

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

Section: Protocols, Codecs, and Endpoints

NEW QUESTION: 5

Which two steps should be taken to provision a phone after the Self-Provisioning feature was configured for end users? (Choose two.)

- A. Enter settings menu on the phone and press *,*,# (star, star, pound).
- B. Dial the hunt pilot extension and associate the phone to an end user
- C. Plug the phone into the network.
- D. Dial the self-provisioning IVR extension and associate the phone to an end user.
- E. Ask the Cisco UCM administrator to associate the phone to an end user.

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

NEW QUESTION: 6

As a voice engineer, which two recommendations do you make to your company to optimize Cisco Unified Communications Manager configuration to reduce the number of toll fraud incidents? (Choose two.)

- A. Inbounds CSS on any gateway typically should have access to internal destinations only and not PSTN destinations.
- B. Classify all route patterns as on-net and prohibit on-net to on-net call transfers in Cisco Unified CM service parameters.
- C. Classify all route pattern as off-net and prohibit off-net to off-net call transfers in Cisco Unified CM service parameters.
- D. Classify all route patterns as on-net or off-net and prohibit off-net call transfers in Cisco Unified CM Service parameters.
- E. Inbound CSS on any gateway typically should have access to internal destinations and PSTN destinations.

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

NEW QUESTION: 7

AN administrator needs to help 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 USM?

- A. Set service parameter "Drop Ad Hoc Conference" to "When Conference Controller leaves"
- B. Set service parameter "Drop Ad Hoc Conference" to "Do not allow outside parties "
- C. Set service parameter "Advanced Ad Hoc Conference" to 2.
- D. Set service parameter "Advanced Ad Hoc Conference" to FALSE

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

NEW QUESTION: 8

Which two features of Cisco Prime Collaboration Assurance require advanced licensing?

(Choose two.)

- A. customizable events
- B. email notifications
- C. real time alarm browse
- D. multicluster support
- E. call quality monitoring

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

NEW QUESTION: 9

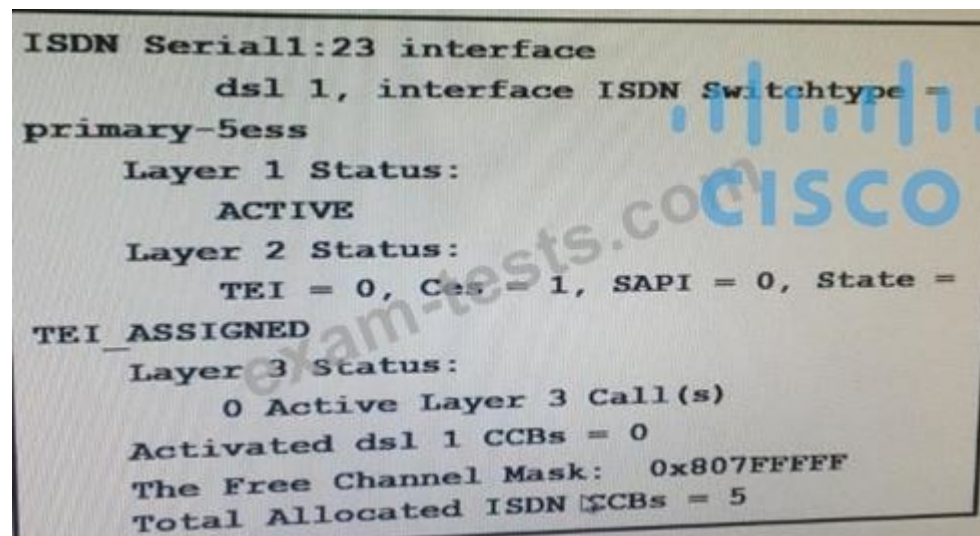
A customer has Cisco Unity Connections that is integrated with LDAP. As a Unity Connection administrator, you have received a request to change the first name for VM user. Where must the change be performed?

- A. Cisco Unity Connection
- B. Active Directory
- C. Cisco IM and Presence
- D. Cisco Unified Communications Manager end user

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 10

Refer to the exhibit.



What is a possible cause of the PRI issue?

- A. The framing is configured incorrectly.
- B. The clock source is incorrect.
- C. The cable is unplugged.
- D. The controller shut down.

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

NEW QUESTION: 11

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

- A. It provides additional granularity by allowing a user to create classes
- B. It minimizes jitter by implementing a priority queue for voice traffic

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

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

Answer: A (LEAVE A REPLY)

https://www.cisco.com/en/US/docs/ios/12_0t/12_0t5/feature/guide/cbwfq.html

NEW QUESTION: 12

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 A

B. Option C

C. Option B

D. Option D

Answer: A (LEAVE A REPLY)

NEW QUESTION: 13

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.722

B. G.711a

C. G.711u

D. G.729

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

NEW QUESTION: 14

Where in Cisco UCM is restrictions on audio bandwidth configured?

- A. region
- B. location
- C. partition
- D. serviceability

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 15

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
```

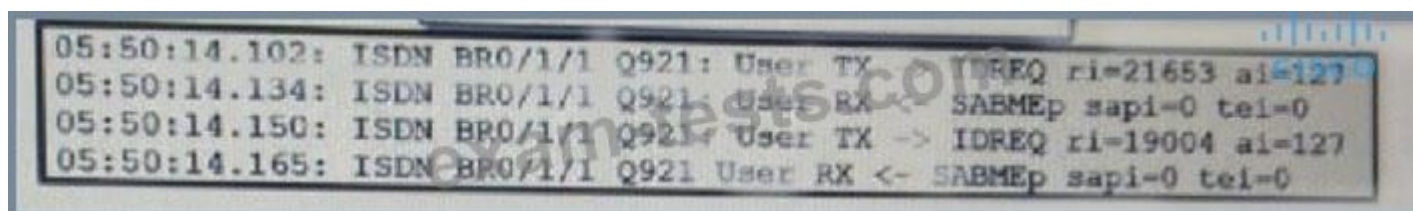
You deploy Mobile and Remote Access for Jabber and discover that Jabber for Windows does not register to cisco Unified Communications Manager while outside of the office. What is a cause of this issue?

- A. The DNS record type should be changed from SRV to A.
- B. The DNS record should be changed from _collab-edge_tcp 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: 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)

```

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

```

A. Option A

B. Option B

C. Option D

D. Option C

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

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<https://www.braindumpsPass.com/Cisco/350-801-practice-exam-dumps.html> (630 Q&As Dumps, **40%OFF** Special Discount: **Exam-Tests**)

NEW QUESTION: 17

Refer to the exhibit. The SDP offer/answer has been completed successfully but there is no DTMF when users press keys.

What is the cause of the issue?

```
INVITE sip:2002@10.10.10.10:5060 SIP/2.0
[..truncated..]
v=0
o=UAC 6107 7816 IN IP4 10.10.10.11
s=SIP Call
c=IN IP4 10.10.10.11
t=0 0
m=audio 8190 RTP/AVP 18 110
c=IN IP4 10.10.10.11
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:110 telephone-event/8000
a=fmtp:110 0-16
a=ptime:20

SIP/2.0 200 OK
[..truncated..]
v=0
o=UAS 4692 9609 IN IP4 10.10.10.10
s=SIP Call
c=IN IP4 10.10.10.10
t=0 0
m=audio 8056 RTP/AVP 18
c=IN IP4 10.10.10.10
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
```

- A. Payload type 110 was negotiated rather than type 101.
- B. DTMF was negotiated properly in these messages.
- C. G.729 rather than G.711ulaw was negotiated.
- D. DTMF was not negotiated on the call.

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

It is offered 18 (g729) and 110 (type-event), however the answer only contains 18. Therefore, DTMF was not negotiated.

NEW QUESTION: 18

What are two QoS requirements for VoIP traffic? (Choose two.)

- A. Voice traffic must be marked to DSCP EF.
- B. Voice traffic must be marked to DSCP AF41.
- C. Average one-way jitter is greater than 50 ms.
- D. One-way latency must be no more than 200 ms.
- E. Loss must be no more than 1 percent.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 19

An employee of company ABC just quit. The IT administrator deleted the employee's userid 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 userid 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 10 pm on March 4th when the user is automatically removed from Cisco UCM.
- C. Wait until 3:15 a.m. on March 6th for garbage collection to remove the user from Cisco UCM.
- D. Wait until 3:15 a.m. on March 5th for garbage collection to remove the user from Cisco UCM.

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

NEW QUESTION: 20

Which protocol does Cisco Prime Collaboration Assurance use to poll the health status of different systems in the Collaboration environment?

- A. SIP
- B. SNMP
- C. SCCP
- D. SMTP

Answer: (SHOW ANSWER)

Explanation/Reference: https://www.cisco.com/c/en/us/products/collateral/cloud-systems-management/prime-collaboration/guide-c07-736946.html#_Toc446633083

NEW QUESTION: 21

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

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

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

There is no @ for a wildcard in SIP route patterns. Because no default SIP route patterns exist in Cisco Unified Communications Manager, you must set them up. Domain name examples are cisco.com, my-pc.cisco.com, *.com, rtp-ccm[1-5].cisco.com. Valid characters for domain names are [, -, ., 0-9, A-Z, a-z, *, and]. IPv4 address examples 172.18.201.119 or 172.18.201.119/32 (explicit IP host address); 172.18.0.0/16 (IP subnet); 172.18.201.18/21 (IP subnet). Valid characters for IP addresses: 0-9, ., and /

NEW QUESTION: 22

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_C606D8A

9_00_cisco-jabber-dns-configuration-guide/CJAB_BK_C606D8A9_00_cisco-jabber-dns-configuration-guide_chapter_010.html

NEW QUESTION: 23

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. Add the FQDN or hostname of the device

C. Select the DTMF relay type for the gateway

D. Select a device pool for the new gateway

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

NEW QUESTION: 24

Which issue can occur if QoS is not Deployed on a Cisco Collaboration architecture across the WAN?

A. Excessive jitter

B. Unexpected shut-down on Cisco Unified Communications Manager

C. 403 Forbidden errors on SIP calls

D. Packet fragmentation

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

NEW QUESTION: 25

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?

```

INVITE sip:4000@172.16.1.1:5061 SIP/2.0
Via: SIP/2.0/TLS 172.16.2.143:5061;branch=z9hG4bK8FD315E7
Remote-Party-ID: <sip:+14088335000@172.16.2.143>;party=calling;screen=no; privacy=off
From: <sip:+14088335000@172.27.2.143>;tag=7B42E5F6-9B8
To: <sip:4000@172.16.1.1>
Date: Tue, 06 Aug 2019 15:03:05 GMT
Call-ID: 4EA4363-B77111E9-8A4AFFCF-10B6D71B@172.16.2.143
Supported: 100rel,timer,resource-priority, replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 0082391505-3077640681-2319777743-0280418075
User-Agent: Cisco-SIPGateway/IOS-15.5.3.S4b
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1565089565
Contact: <sip:+ 14088335000@172.16.2.143:5061;transport=tls>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 68
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 416
v=0
o=CiscoSystemsSIP-GW-UserAgent 8486 8298 IN IP4 172.16.2.143
s=SIP Call
c=IN IP4 172.16.2.143
t=0 0
m=audio 44612 RTP/SAVP 0 101
c=IN IP4 172.16.2.143
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20

```

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 26

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

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

Answer:



NEW QUESTION: 27

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 A

C. Option D

D. Option B

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

NEW QUESTION: 28

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


```
dial-peer voice 4100 voip
destination-pattern 4100
session protocol sipv2
session target ipv4:10.1.10.1
dtmf-relay sip-notify
no vad
```

How is this issue resolved?

- A. The dial-peer needs to be reconfigured to support vad.
- B. The destination pattern needs to be changed to match the dialed number of 4100.
- C. The dial-peer needs to be reconfigured to support H 323 instead of SIP.
- D. The dial-peer needs to be reconfigured to support the G.711 voice codec.

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

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NEW QUESTION: 32

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

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

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

NEW QUESTION: 33

```
rule 1 /^([025]..)\-([...])\-([...$])/ /\1\2\3/
```

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

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

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

Section: Call Control

NEW QUESTION: 34

Which external DNS SRV record must be present for Mobile and Remote Access?

- A.** _collab-edge._tcp.example.com
- B.** _cisco-uds._tls.example.com
- C.** _cisco-uds.Jcp.example.com
- D.** _collab-edge._tls.example.com

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 35

Refer to the exhibit.



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

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

Answer: C (LEAVE A REPLY)

NEW QUESTION: 36

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.** service parameters
B. gateway
C. route pattern
D. IP phone

Answer: (SHOW ANSWER)

NEW QUESTION: 37

What makes Cisco Unified Border Element a better choice than a conventional Session border controller?

- A. DTMF interworking
- B. Address hiding
- C. Voice policy
- D. SIP security

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 38

Which issue cause slips on a PRI?

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

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

NEW QUESTION: 39

What dialed numbers 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: ([SHOW ANSWER](#))

Section: Call Control

NEW QUESTION: 40

An administrator is integrating a Cisco Unity Express module to a Cisco UCME system A test call is placed to the Cisco Unity Express pilot number, but the administrator receives a busy signal.

The dial- peer is configured as follows:

```
dial-peer voice 4100 voip
destination-pattern 410
session protocol sipv2
session target ipv4:10.1.10.1
dtmf-relay sip-notify
no vad
```

How is this issue resolved?

- A. The dial-peer needs to be reconfigured to support vad.
- B. The destination pattern needs to be changed to match the dialed number of 4100.
- C. The dial-peer needs to be reconfigured to support the G 711 voice codec.
- D. The dial-peer needs to be reconfigured to support H 323 instead of SIP.

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

NEW QUESTION: 41

Refer to the exhibit. An engineer configures a VoIP dial peer on a Cisco gateway.

Which codec is used?

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

- A. G.711ulaw
- B. No codec is used (missing codec command).
- C. G.711alaw
- D. G.729r8

Answer: ([SHOW ANSWER](#))

<https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/dialpeer/configuration/15-mt/vd-15-mt-book/vd-dp-cfg-examp.pdf>

NEW QUESTION: 42

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

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

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-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: A (LEAVE A REPLY)

<https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/cube- codec-basic.html>

Table 2. Codec and Bandwidth Information									
Codec & Bit Rate (kbps)	Codec Sample Size (Bytes)	Codec Sample Interval (ms)	Mean Opinion Score (MOS)	Voice Payload Size (Bytes)	Voice Payload Size (ms)	Payload Size (ms) Packets Per Second (PPS)	Bandwidth MP or FRF.12 (kbps)	Bandwidth w/cRTP MP or FRF.12 (kbps)	Bandwidth Ethernet (kbps)
G.711 (64 kbps)	80	10	4.1	160	20	50	82.8	67.6	87.2
G.729 (8 kbps)	10	10	3.92	20	20	50	26.8	11.6	31.2
G.723.1 (6.3 kbps)	24	30	3.9	24	30	33.3	18.9	8.8	21.9
G.723.1 (5.3 kbps)	20	30	3.8	20	30	33.3	17.9	7.7	20.8
G.726 (32 kbps)	20	5	3.85	80	20	50	50.8	35.6	55.2
G.726 (24 kbps)	15	5		60	20	50	42.8	27.6	47.2
G.728 (16 kbps)	10	5	3.61	60	30	33.3	28.5	18.4	31.5
G722_64k(64 kbps)	80	10	4.13	160	20	50	82.8	67.6	87.2
ilbc_mode_20(15.2 kbps)	38	20	NA	38	20	50	34.0	18.8	38.4
ilbc_mode_30(13.33 kbps)	50	30	NA	50	30	33.3	25.867	15.73	28.8

NEW QUESTION: 44

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

A. G.711alaw

- B. G.729A
- C. G.722.1
- D. iLBC

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

NEW QUESTION: 45

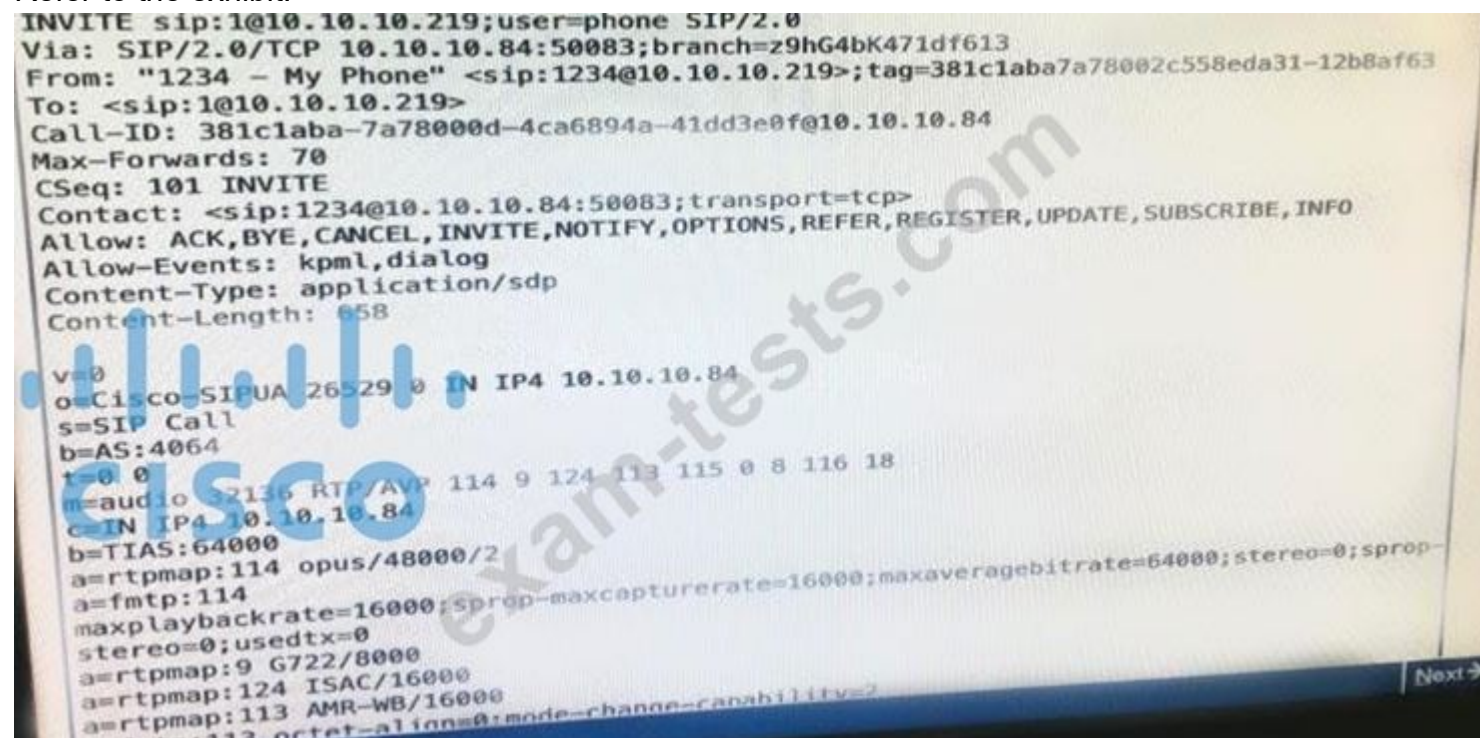
When configuring Cisco Unified Communications Manager, which configuration enables phones to automatically register to a Cisco Unified Communications publisher when the connection to the subscriber is lost?

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

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

NEW QUESTION: 46

Refer to the exhibit.



The image shows a network packet capture of a SIP INVITE message. The packet is from 10.10.10.84 to 10.10.10.219. The SDP content is as follows:

```
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=rtpmap:113 octet-align=0;mode=change-capability=2
```

```

Call-ID: 381c1a0a-7a700000-4c00034d-41005e01@10.10.10.04
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

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

Answer: ([SHOW ANSWER](#))

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<https://www.braindumpsPass.com/Cisco/350-801-practice-exam-dumps.html> (630 Q&As Dumps, **40%OFF** Special Discount: **Exam-Tests**)

NEW QUESTION: 47

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

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

NEW QUESTION: 48

A network administrator deleted a user from the LDAP directory of a company. The end user shows as Inactive LDAP Synchronize User in Cisco Unified Communications Manager. Which step is next to remove this user from Cisco Unified Communications Manager?

- A. Delete the user directly from Cisco Unified Communications Manager.
- B. Wait 24 hours for the garbage collector to remove the user.
- C. Restart the Dirsync service after the user is deleted from LDAP directory.
- D. Execute a manual sync to refresh the local databased and delete the end user.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 49

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-group timeslots 1-7
```

B.

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

C.

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

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

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

NEW QUESTION: 50

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: ([SHOW ANSWER](#))

Section: Call Control

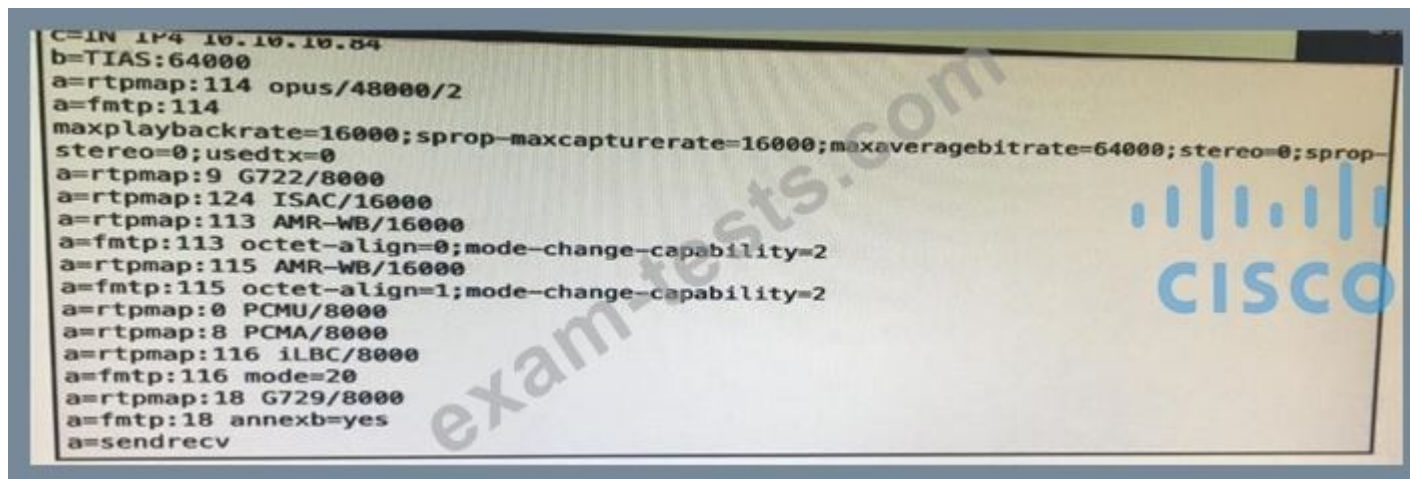
NEW QUESTION: 51

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=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=fmtp:113 octet-align=0;mode-change-capability=2
a=rtpmap:113 octet-align=0;mode-change-capability=2
a=fmtp:113 octet-align=0;mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
```

Next →



```
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-
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. PRACK
- C. REGISTER
- D. NOTIFY
- E. UPDATE

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

NEW QUESTION: 52

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. TCP port 5060
- B. Unsolicited NOTIFY
- C. 200 OK
- D. SIP Register

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

NEW QUESTION: 53

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: 0x807FFFFFFF
  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: 54

Refer to the exhibit.

```
INVITE sip:2002@10.10.10.10:5060 SIP/2.0
[..truncated..]
v=0
o=UAC 6107 7816 IN IP4 10.10.10.11
s=SIP Call
c=IN IP4 10.10.10.11
t=0 0
m=audio 8190 RTP/AVP 18 110
c=IN IP4 10.10.10.11
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:110 telephone-event/8000
a=fmtp:110 0-16
a=ptime:20

SIP/2.0 200 OK
[..truncated..]
v=0
o=UAS 4692 9609 IN IP4 10.10.10.10
s=SIP Call
c=IN IP4 10.10.10.10
t=0 0
m=audio 8056 RTP/AVP 18
c=IN IP4 10.10.10.10
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
```

The SOP offer/answer has been completed successfully but there is no DTMF when users press keys. What is the cause of the issue?

- A. Payload type 110 was negotiated rather than type 101.
- B. DTMF was negotiated properly in these messages.
- C. DTMF was not negotiated on the call.
- D. G.729 rather than G.711ulaw was negotiated.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 55

Refer to the exhibit.

SIP Trunk Security Profile Information

Name* CUP Non Secure SIP Profile

Description

Device Security Mode Non Secure

Incoming Transport Type* TCP+UDP

Outgoing Transport Type TCP

☐ Enable Digest Authentication

Nonce Validity Time (mins)* 600

Secure Certificate Subject or Subject Alternate Name

Incoming Port* 5060

☐ Enable Application level authorization

☒ Accept presence subscription

☒ Accept out-of-dialog refer**

☐ Accept unsolicited notification

☐ Accept replaces header

☐ Transmit security status

☐ Allow charging header

SIP V.150 Outbound SDP Offer Filtering* Use Default Filter

Refer to the exhibit. A collaboration engineer is configuring the Cisco UCM IM and Presence Service. Which two steps complete the configuration of the SIP trunk security profile? (Choose two.)

- A. Check the box to transmit security status.
- B. Check the box to allow charging header.
- C. Check the box to accept unsolicited notification.
- D. Check the box to enable application-level authorization.
- E. Check the box to accept replaces header.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 56

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 and media are passed through the Cisco Unified Border Element

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 57

A user reports transfer failures from an IP phone for calls received from a PSTN to another PSTN number.

What is a reason for these failures?

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

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

NEW QUESTION: 58

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

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

Answer: ([SHOW ANSWER](#))

Section: Cisco IOS XE Gateway and Media Resources

NEW QUESTION: 59

```
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
```

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

Answer: ([SHOW ANSWER](#))

Section: Cisco IOS XE Gateway and Media Resources

Explanation/Reference:

<https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-callmanager/42105-vg200-cfg.html>

NEW QUESTION: 60

Which configuration step is necessary for a Cisco SIP phone to synchronize its time with a specific source?

- A. Change the Time Zone from "America/Los_Angeles" to "Etc/GMT+8".
- B. Add a phone NTP Reference to the date/time Group.
- C. Assign the device to the correct region.
- D. Change the time format from 24-hour to 12-hour.

Answer: B (LEAVE A REPLY)

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_0110.html

NEW QUESTION: 61

A Cisco Unity Connection Administrator must set a voice mailbox so that it can be accessed from a secondary device.

Which configuration on the voice mailbox makes this change?

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

Answer: D (LEAVE A REPLY)

If you elect to make changes to the Cisco Unity system, there are two options available:

- Call Routing Rules (available in all Cisco Unity versions)
- Using the Alternate Extension feature (available in Cisco Unity 3.0 and later).

<https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unity/41600-2-ip-phones-1-mbox.html#ts1>

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NEW QUESTION: 62

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)

```

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

```

A. Option A

B. Option B

C. Option D

D. Option C

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

NEW QUESTION: 63

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&P

B. `_cisco-uds._tcp.cisco.com` pointing to a record of Cisco Unified CM

C. `_cuplogin._tcp.cisco.com` pointing to a record 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: ([SHOW ANSWER](#))

NEW QUESTION: 64

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?

SIP Trunk Security Profile Information

Name*	Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null String
Device Security Mode	Encrypted ▼
Incoming Transport Type*	TLS ▼
Outgoing Transport Type	TLS ▼
☐ Enable Digest Authentication	
Nonce Validity Time (mins)*	600
Secure Certificate Subject or Subject Alternate Name	
Incoming Port*	5061

- A. the common name of the remote device certificates
- B. the fully 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 fully qualified domain name of the remote device that is configured on the SIP trunk

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

NEW QUESTION: 65

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

- A. G.711ulaw
- B. iLBC
- C. G.722.1
- D. G.729A

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

Section: Protocols, Codecs, and Endpoints

Explanation/Reference: <https://community.cisco.com/t5/collaboration-voice-and-video/summary-of-cucm-supported-codecs/ ta-p/3162905>

NEW QUESTION: 66

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. TCP port 5060
- B. SIP Register
- C. Unsolicited NOTIFY
- D. 200 OK

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

<https://community.cisco.com/t5/collaboration-voice-and-video/understanding-troubleshooting-mwi-on-unity-connection/ta-p/3162948>

NEW QUESTION: 67

Which Cisco Unified CM service is installed by default and authenticates certificates on behalf of IP phones and other endpoints?

- A. Cisco CallManager
- B. Cisco CTL Provider
- C. Cisco Trust Verification
- D. Cisco TFTP
- E. Cisco Certificate Authority Proxy Function

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

NEW QUESTION: 68

A presence redundancy group is deployed, and an engineer with ID012345678 initiates a Manager fallback. Which statement about Cisco server Recovery Manager is true?

- A. restarts critical services on the secondary node
- B. restart the Cisco Presence Engine
- C. disconnects all users that had been failed over
- D. disconnects all users that had been failed over, and the users must log in again

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

NEW QUESTION: 69

Refer to the exhibit. Which option describes the security encryption status of this active call on a Cisco IP phone?

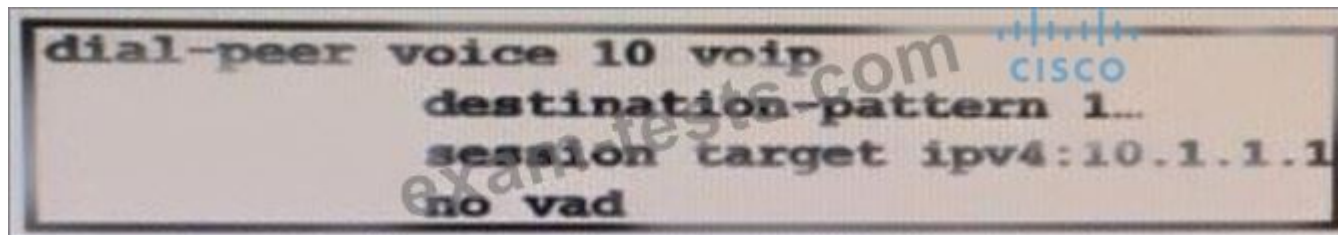


- A. encrypted call signaling and media
- B. encrypted call signaling but unencrypted call media
- C. unencrypted call signaling and media
- D. encrypted call media but unencrypted call signaling
- E. Not enough information provided to answer this question.

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

NEW QUESTION: 70

Refer to the exhibit.



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

The image shows a Cisco IOS configuration snippet for a VoIP dial-peer. The commands are: `dial-peer voice 10 voip`, `destination-pattern 1...`, `session target ipv4:10.1.1.1`, and `no vad`. A Cisco logo is visible in the top right corner of the exhibit.

AN engineer configures a VoIP dial peer on a Cisco gateway. which codec is used?

- A. No codec is used (missing codec command).
- B. G.729r8
- C. G.711ulaw
- D. G.711alaw

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

NEW QUESTION: 71

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/edgedge.html>

NEW QUESTION: 72

When a 8800 series phone is registered over MRA, where does it register?

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

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

NEW QUESTION: 73

Refer to the exhibit.

```
-----
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-
2dbce74ae7w2-
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. ILBC
- B. 729
- C. 711ulaw
- D. 728

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

NEW QUESTION: 74

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=fmtp:113 octet-align=0;mode-change-capability=2
```

Next →

```
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
```

Next

```
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
```

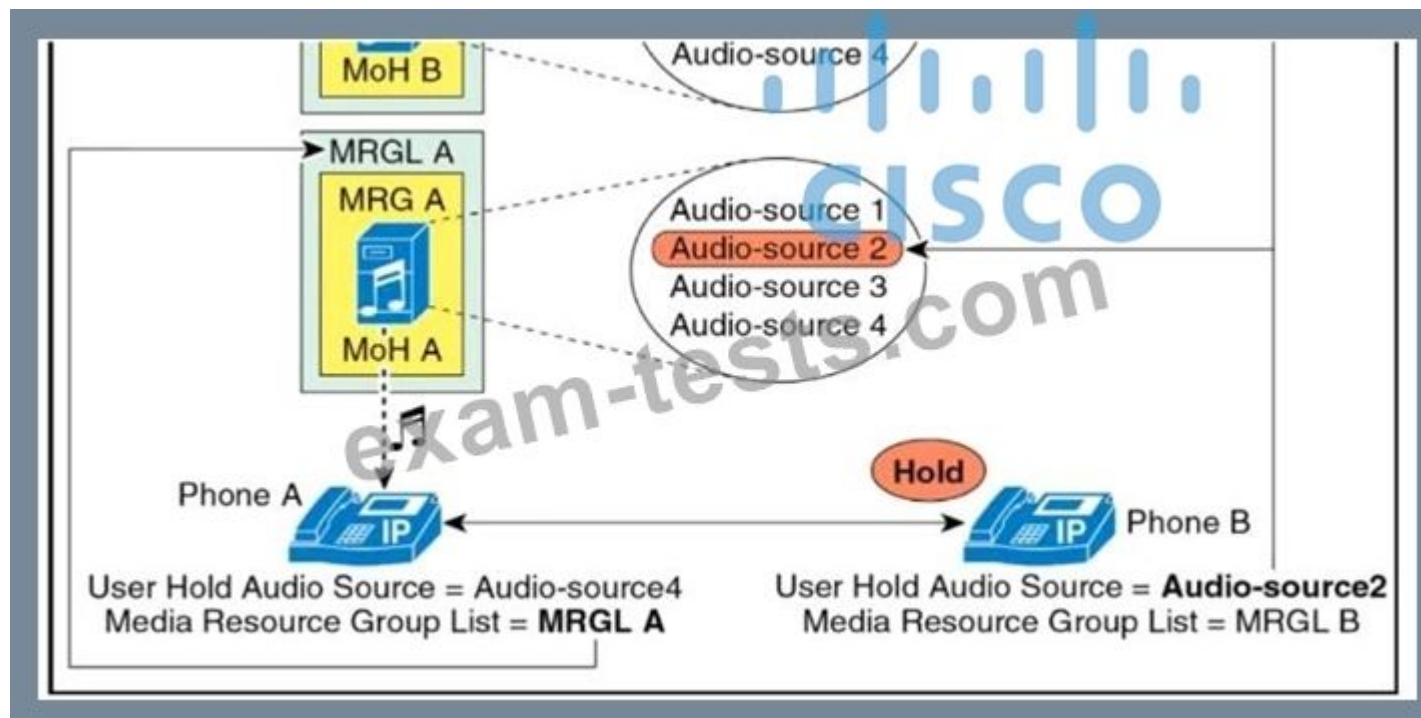
When a UC Administrator is troubleshooting DTMF negotiated by this SIP INVITE, which two messages should be examined next to further troubleshoot the issue? (Choose two.)

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

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

NEW QUESTION: 75

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 2
- B. MRGL A and Audio Source 4
- C. MRGL B and Audio Source 4
- D. MRGL B and Audio Source 2

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

NEW QUESTION: 76

Refer to the exhibit.

```
hqcucmpub.pkinane.com - PuTTY
login as: admin
admin@hqcucmpub.pkinane.com's password:
Command Line Interface is starting up, please wait ...

Welcome to the Platform Command Line Interface

VMware Installation:
  2 vCPU: Intel(R) Xeon(R) CPU E5-2699 v3 @ 2.30GHz
  Disk 1: 110GB, Partitions aligned
  8192 Mbytes RAM
  WARNING: DNS unreachable

admin:
```

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

- A. utils diagnose module validate Network
- B. show cdp neighbor
- C. utils ntp status
- D. utils diagnose test
- E. show version active

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

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NEW QUESTION: 77

Which command is used in Cisco IOS XE TDM gateway to configure the voice T1/E1 controller to provide clocking?

- A. clock source line
- B. Cisco IOS XE TDM gateway T1/E1 controller cannot provide clocking.
- C. clocking source internal
- D. clocking source network

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

Explanation/Reference: <https://www.cisco.com/c/en/us/td/docs/routers/access/interfaces/NIM/software/configuration/guide/4gen-t1-e1-nim-guide.html>

NEW QUESTION: 78

A user reports transfer failures from an IP phone for calls received from a PSTN to another PSTN number. What is a reason for these failures?

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 79

A user reports transfer failures from an IP phone for calls received from a PSTN to another PSTN number. What is a reason for these failures?

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

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

https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_1_3/ccmfeat/fsxfer.html#wp1048448

NEW QUESTION: 80

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 rtp-nte sip-notify
- B. sip-notify dtmf-relay rtp-nte
- C. dtmf-relay sip-kpml cisco-rtp
- D. dtmf-relay cisco-rtp

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 81

What is a capability of Cisco Expressway?

- A. It functions as an analog telephony adapter
- B. It provides access to on-premises Cisco Unified Communications infrastructure for remote endpoints
- C. It has remote endpoint enrollment with Certificate Authority Proxy Function
- D. It gives directory access for remote users via Cisco Directory Integration

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 82

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

- in the Calling Party Transformation Patterns section,
configure the Pattern as 9.011841234567
- A.
 - configure the Discard Digits as Predot 10-10-Dialing
- in the Calling Party Transformation Patterns section,
configure the Pattern as a 011841234557
- B.
 - configure the Discard Digits as Predot
- in the Called Party Transformation Pattern Configuration section,
configure the Pattern as 9.011841234567
- C.
 - configure the Discard Digits as Predot
- in the Called Party Transformation Pattern Configuration section,
configure the Pattern as 9.011841234587
- D.
 - configure the Discard Digits as Predot 10-10-Dialing

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 83

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

- A. option 15
- B. option 6
- C. option 66
- D. option 120

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

Explanation/Reference: <https://blog.router-switch.com/2013/03/dhcp-option-150-dhcp-option-66/>

NEW QUESTION: 84

Which program is required to deploy the Cisco Jabbar client on an on-premises Cisco collaboration solution?

- A. Cisco UCM
- B. Cisco Unity Connection
- C. Cisco UCM IM and Presence
- D. Cisco Expressway-C

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

NEW QUESTION: 85

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 86

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. Enable application level authorization.
- C. Allow charging header
- D. Accept replaces header
- E. Accept unsolicited notification

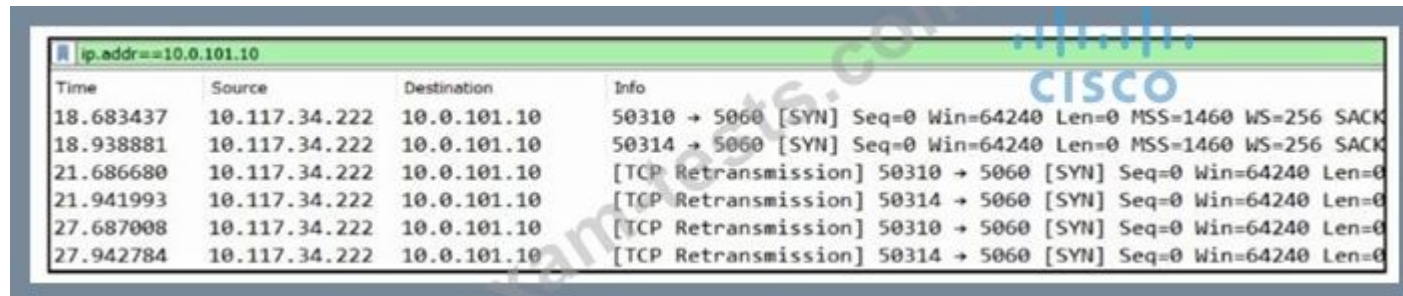
Answer: D,E (LEAVE A REPLY)

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/11x/integration/guide/cucm_sip/b_cucintcucmsip/b_cucintcucmsip_chapter_011.html

NEW QUESTION: 87

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 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?



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

- 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. 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.
- 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: 88

Which description of the Mobile and Remote Access feature is true?

- A. Collaboration Edge feature that enables remote individuals to perform international calls from Jabber with a VPN connection
- B. Collaboration Edge feature that enables remote individual to access enterprise collaboration services via Jabber with the use of a VPN connection
- C. Collaboration Edge feature that enables remote individuals to access all enterprise collaboration services using a PC within the corporate environment
- D. Collaboration Edge feature that enables remote individuals to access enterprise collaboration services via Jabber without the use of a VPN connection

Answer: D (LEAVE A REPLY)

NEW QUESTION: 89

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. Deploy RSVP to improve VoIP QoS only where it can have a positive impact on quality and functionality where there is limited bandwidth.
- C. Map audio and video streams of video calls (AF41 and AF42) to a class-based queue with weighted random early detection.
- D. Use 802.1Q trunking and 802.1p 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 required per voice call on point-to-point links.

Answer: A,D (LEAVE A REPLY)

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/srnd/design/guide/cmestnd/nstrct.html

NEW QUESTION: 90

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

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

Answer: (SHOW ANSWER)

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/jabber/11_5/CJAB_BK_D00D8CBD_00_deployment-installation-guide-cisco-jabber115/CJAB_BK_D00D8CBD_00_deployment-installation-guide-cisco-jabber115_chapter_01000.html

NEW QUESTION: 91

Which Cisco Unified Communications Manager configuration is required for SIP MWI integrations?

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

Answer: B (LEAVE A REPLY)

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NEW QUESTION: 92

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

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

NEW QUESTION: 93

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

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

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

NEW QUESTION: 94

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

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

NEW QUESTION: 95

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. SCCP
- C. SIP
- D. HTTPS

E. H.323

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 96

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

A. G.711alaw

B. G.722.1

C. G.729A

D. iLBC

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 97

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

A. third-party SIP device

B. Cisco Dual Mode for iPhone

C. Cisco Dual Mode for Android

D. Cisco Unified Client Services Framework

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

Section: Collaboration Applications

Explanation/Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/jabber/11_5/CJAB_BK_D00D8CBD_00_deployment-installation-guide-cisco-jabber115/CJAB_BK_D00D8CBD_00_deployment-installation-guide-cisco-jabber115_chapter_01000.html

NEW QUESTION: 98

What is a characteristic of video traffic that governs QoS requirements for video?

A. Voice and video traffic are different, but they have the same QoS requirements.

B. Voice and video are the same, so they have the same QoS requirements.

C. Video is typically variable bit rate.

D. Video is typically constant bit rate.

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

NEW QUESTION: 99

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

A. SNMP

B. LLDP

C. TFTP

D. LACP

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

NEW QUESTION: 100

Which two descriptions of TFTP option 66 and 150 for Cisco IP phones are true? (Choose two.)

- A. Option 150 supports multiple entries of TFTP server IP addresses, but option 66 supports only one of TFTP server DNS names.
- B. Option 150 supports only TFTP server IP addresses, but option 66 supports only TFTP server DNS names.
- C. Option 150 supports multiple entries of TFTP server IP addresses, but option 66 supports only one entry of either TFTP server IP address or DNS name.
- D. Option 66 is Cisco proprietary, but option 150 is standard-based.
- E. Option 150 is Cisco proprietary, but option 66 is standard-based.
- F. Option 150 and 66 are standard-based.

Answer: C,E (LEAVE A REPLY)

Used mostly with phone DHCP configurations, Option 66 provides a single TFTP server address. Option 150 provides a list of TFTP server addresses. Option 150 is Cisco proprietary, option 66 is IEEE.

NEW QUESTION: 101

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

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

Answer:

TRAP	SET
SET	TRAP
GET	INFORM
INFORM	GET

NEW QUESTION: 102

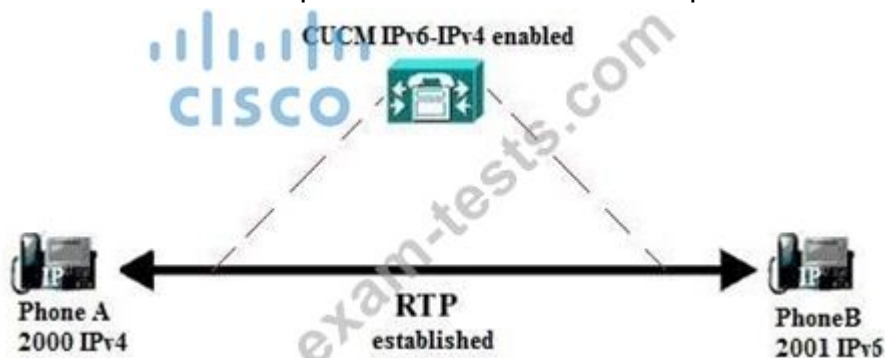
A collaboration engineer troubleshoots issues with a Cisco IP Phone 7800 Series. The IPv4 address of the phone is reachable via ICMP and HTTP and the phone is registered to Cisco UCM. However, the engineer cannot reach the CLI of the phone. Which two actions in Cisco UCM resolve the issue? (Choose two)

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 103

Refer to the exhibit. IP phone A makes a call to IP phone B and RTP voice stream is successfully established between the endpoints.



What option describes how to accomplish this call on phones with incompatible IP addressing versions?

- A. Set common device configuration to IPv4 and IPv6
- B. Assign a media resource group with available MTPs
- C. Enable alternative network address types
- D. Set IP addressing mode preference for signalling to IPv4

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 104

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

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

Answer: ([SHOW ANSWER](#))

Section: Call Control

NEW QUESTION: 105

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

- A. music on hold servers

- B. H.225 trunks
- C. Cisco Unified IP phones (all models)
- D. SIP trunks
- E. H.322 clients

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 106

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. Advise the user to press the "#" button after dialling the FAC and CMC codes
- 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. Do not wait for the tones immediately dial the FAC and CMC

Answer: ([SHOW ANSWER](#))

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NEW QUESTION: 107

Refer to the exhibit.



```

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=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv

```

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

NEW QUESTION: 108

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

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

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

NEW QUESTION: 109

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

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

Answer: (SHOW ANSWER)

Cisco recommends using G.729a as the low-bandwidth codec because it is supported on all Cisco Unified IP Phone models as well as most other Cisco Unified Communications devices, therefore it can eliminate the need for transcoding.

NEW QUESTION: 110

Which DTMF relay method configured on a SIP dial-peer will ensure that a media resource is not invoked by Unified CM for calls to UCCX IVRs?

- A. dtmf-relay slp-kpml
- B. dtmf-relay sip-notify
- C. sdtmf-relay h245-signal
- D. dtmf-relay rtp-nte

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 111

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. Conference Bridge
- B. Media Termination point
- C. Annunciator
- D. Trusted relay point

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 112

An engineer is designing a high availability and failover solution for two Cisco Unified Border Element 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 1 40 cube2.abc.com
- B. _sip_udp.abc.com 60 IN SRV 60 1 cube1.abc.com
- C. _sip_udp.abc.com 60 IN SRV 3 60 cube2.abc.com
- D. _sip_udp.abc.com 60 IN SRV 2 60 cube1.abc.com
- E. _sip_udp.abc.com 60 IN SRV 1 60 cube1.abc.com

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

NEW QUESTION: 113

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 114

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

- A. Enterprise parameters configuration
- B. Service parameter configuration

- C. Region configuration
- D. Enterprise Phone configuration

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

NEW QUESTION: 115

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: 116

A collaboration engineer troubleshoots issues with a Cisco IP Phone 7800 Series. The IPv4 address of the phone is reachable via ICMP and HTTP and the phone is registered to Cisco UCM. However, the engineer cannot reach the CLI of the phone.

Which two actions in Cisco UCM resolve the issue? (Choose two)

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

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

<https://www.cisco.com/c/en/us/support/docs/collaboration-endpoints/ip-phone-7800-series/200850-Troubleshoot-Cisco-Phone-7800-8800-Serie.pdf>

NEW QUESTION: 117

Refer to the exhibit.

SIP Trunk Security Profile Information

Name*	Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null String
Device Security Mode	Encrypted ▼
Incoming Transport Type*	TLS ▼
Outgoing Transport Type	TLS ▼
☐ Enable Digest Authentication	
Nonce Validity Time (mins)*	600
Secure Certificate Subject or Subject Alternate Name	
Incoming Port*	5061

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 common name of the remote device certificates
- B. the fully qualified domain name of the remote device that is configured on the SIP trunk
- C. the common name of the Cisco UCM CallManager certificate
- D. the fully qualified domain name of all Cisco UCM nodes that run the CallManager service

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

NEW QUESTION: 118

Refer to the exhibit. An engineer is confining class of control for a user in Cisco UCM.

Which change will ensure that the user is unable to call 2143?

User's phone has:

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

Configured Route Patterns:

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

- A. Set the user's line CSS to <None>
- B. Change line CSS to only contain Partition_B

- C. Set the users device CSS to <None>
- D. Change line partition to Partition_A

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 119

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

- A. Balanced mode provides user load balancing and user failover in the event of an outage. Active/standby mode provides an always on standby node in the event of an outage, and it also provides load balancing.
- B. Balanced mode provides user load balancing and user failover only for manually generated failovers. Active/standby mode provides an unconfigured standby node in the event of an outage, but it does not provide load balancing.
- C. Balanced mode provides user load balancing and user failover in the event of an outage. Active/standby mode provides an always on standby node in the event of an outage, but it does not provide load balancing.
- D. Balanced mode does not provide user load balancing, but it provides 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: C ([LEAVE A REPLY](#))

Section: Collaboration Applications

NEW QUESTION: 120

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

- A. If a call uses G.711, Cisco UCM subtracts 64k
- B. If a call uses G.729. Cisco UCM subtracts 40k
- C. If a call uses G.729. Cisco UCM subtracts 16k.
- D. If a call uses G.729. Cisco UCM subtracts 24k.
- E. If a call uses G.711, Cisco UCM subtracts 80k

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 121

A user reports transfer failure from an IP phone for calls received from a PSTN to another PSTN number.

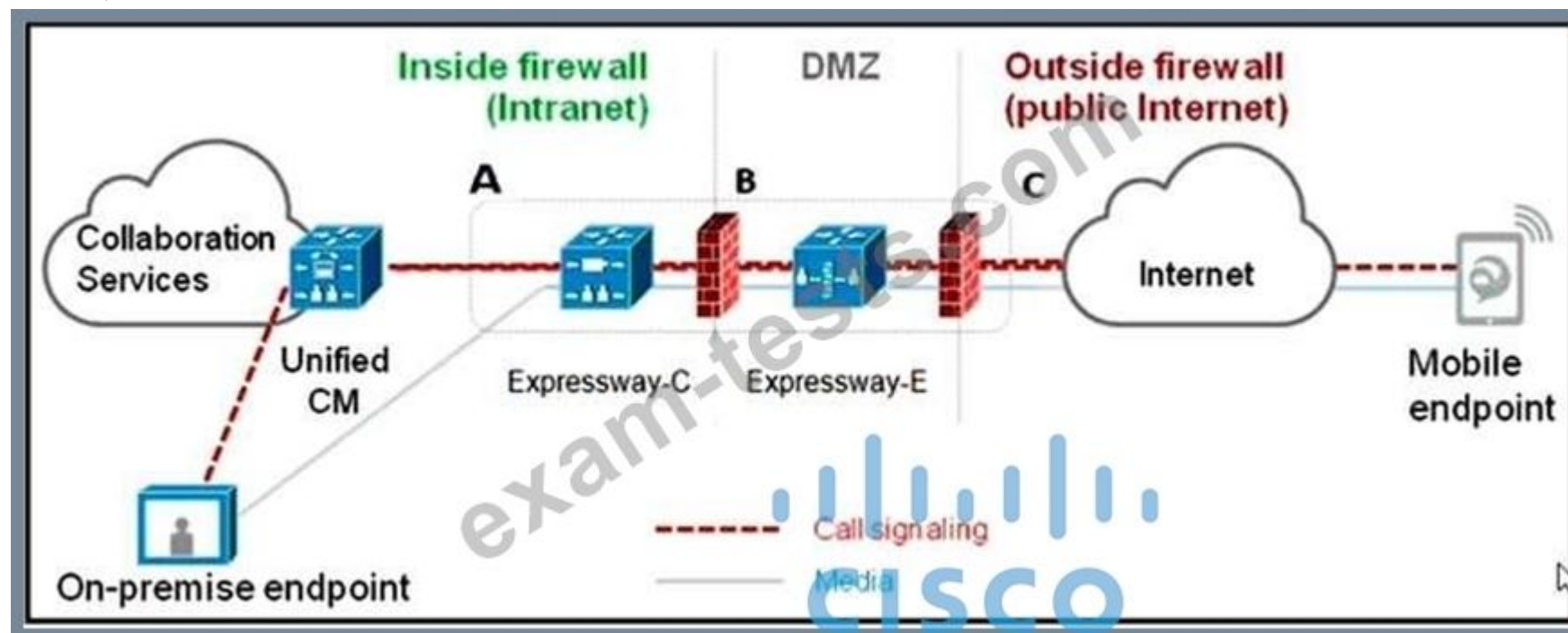
What is a reason for these failures?

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

Answer: ([SHOW ANSWER](#))

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NEW QUESTION: 122



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

Answer: A (LEAVE A REPLY)

Section: Protocols, Codecs, and Endpoints

Explanation/Reference:

https://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/expressway/config_guide/X8-9/Mobile-Remote-Access-via-Expressway-Deployment-Guide-X8-9-1.pdf

NEW QUESTION: 123

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

- A. linecode esf
- B. linecode ami
- C. linecode hdb3
- D. linecode b8zs

Answer: D (LEAVE A REPLY)

Explanation/Reference: [https://www.cisco.com/en/US/docs/ios/dial/configuration/guide/](https://www.cisco.com/en/US/docs/ios/dial/configuration/guide/dia_cfg_isdn_pri_external_docbase_0900e4b1806c752c_4container_external_docbase_0900e4b18216dd1b.html)

[dia_cfg_isdn_pri_external_docbase_0900e4b1806c752c_4container_external_docbase_0900e4b18216dd1b.html](https://www.cisco.com/en/US/docs/ios/dial/configuration/guide/dia_cfg_isdn_pri_external_docbase_0900e4b1806c752c_4container_external_docbase_0900e4b18216dd1b.html)

NEW QUESTION: 124

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=z9hG4bKFIKED
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-ana
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
```

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 not sending any support for DTMF. DTMF configuration must be added to the appropriate dial peer.
- 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 Unity Connection is configured on G.729 only. Cisco Unity Connection must be reconfigured to support iLBC.

Answer: (SHOW ANSWER)

NEW QUESTION: 125

Refer to the exhibit.

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

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

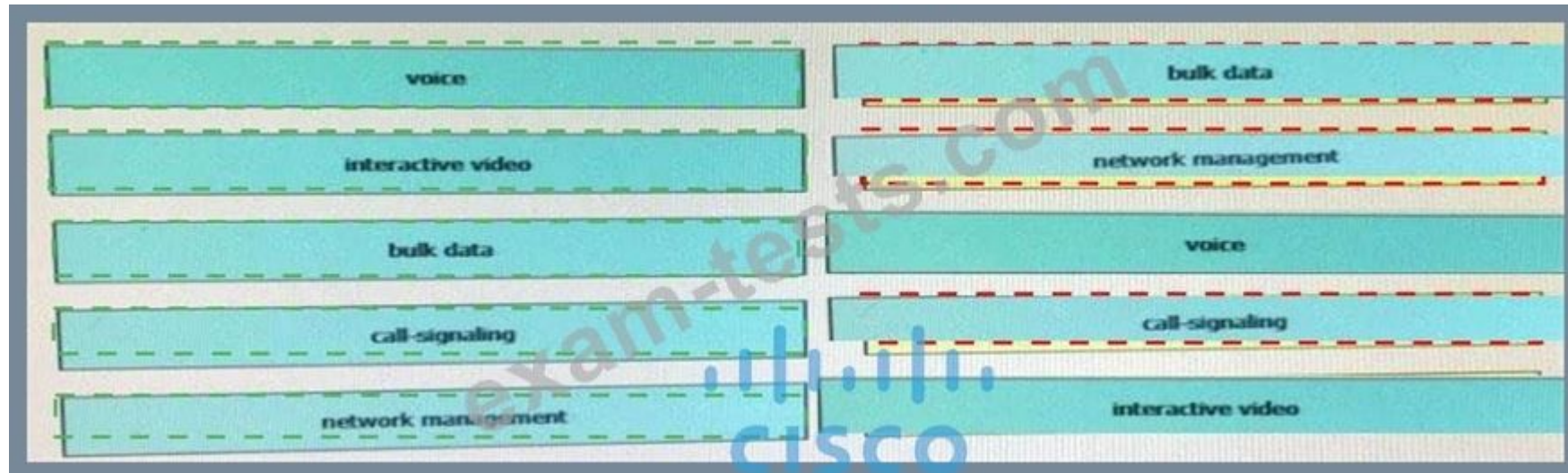
Answer: C (LEAVE A REPLY)

NEW QUESTION: 126

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/QoS

NEW QUESTION: 127

An engineer with ID378163512 is designing a new dial plan for a customer that has offices in several countries on four continents around the world. This client also want to integrate with a Microsoft Lync backend, Which dial plan type does the engineer recommend?

- A. TEHO
- B. H.323
- C. E.164
- D. SIP URI

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 128

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

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

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

NEW QUESTION: 129

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

NEW QUESTION: 130

What is the maximum DNS SRV entries that should be defined in the SIP Trunk destination address field in Cisco UCM?

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 131

When configuring Cisco Unified Communications Manager, which configuration enables phones to automatically register to a Cisco Unified Communications publisher when the connection to the subscriber is lost?

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

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

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: 132

Given the 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
```

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*

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*

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.

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

NEW QUESTION: 133

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

- A. Expressway-C
- B. Cisco Unified Communications Manager
- C. Cisco Unified presence server
- D. Expressway-E

Answer: ([SHOW ANSWER](#))

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: 134

Endpoint A is attempting to call endpoint B. Endpoint A only supports G.711 ulaw with a packetization rate of

20 ms, and endpoint B supports packetization rate of 30 ms for G.711ulaw. Which two media resources are allocated to normalize packetization rates through transrating?

- A. Software transcoder on Cisco unified Communications manager
- B. Hardware MTP on Cisco IOS Software
- C. Software MTP on Cisco IOS Software
- D. Hardware transcoder on Cisco IOS Software
- E. Software MTP on Cisco Unified Communication Manager

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 135

An administrator troubleshoots call flows and suspects that there are issues with the dial plan. Which tool enables a quick analysis of the dial plan and provides call flows of dialled digits?

- A. Cisco Dial Plan Analyzer
- B. Dialed Number Analyzer
- C. Digit Analysis Analyzer
- D. Dial Plan Analyzer

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 136

A customer wants to conduct B2B video calls with a partner using an on-premises conferencing solution. Which two devices are needed to facilitate this request? (Choose two.)

- A. Expressway-E
- B. Cisco TelePresence Management Suite
- C. Cisco Unified Border Element
- D. Expressway-C
- E. MGCP gateway

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

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NEW QUESTION: 137

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. iLBC

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

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

NEW QUESTION: 138

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. Any Zero is stripped from the numeric phone number.
- B. The ISDN Plan is modified to the administrators defined value.
- C. The leading Plus is stripped from the numeric phone number.
- D. The ISDN Type is modified to the administrators defined value.

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

NEW QUESTION: 139

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

- A. Cisco Bulk provisioning Service
- B. Cisco DirSync
- C. Cisco CallManager SNMP Service
- D. Cisco AXL Web Service

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 140

Refer to the exhibit.

```
INVITE sip:2002@10.10.10.10:5060 SIP/2.0
[..truncated..]
v=0
o=UAC 6107 7816 IN IP4 10.10.10.11
s=SIP Call
c=IN IP4 10.10.10.11
t=0 0
m=audio 8190 RTP/AVP 18 110
c=IN IP4 10.10.10.11
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:110 telephone-event/8000
a=fmtp:110 0-16
a=ptime:20

SIP/2.0 200 OK
[..truncated..]
v=0
o=UAS 4692 9609 IN IP4 10.10.10.10
s=SIP Call
c=IN IP4 10.10.10.10
t=0 0
m=audio 8056 RTP/AVP 18
c=IN IP4 10.10.10.10
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
```

Refer to the exhibit. The SDP offer/answer has been completed successfully but there is no DTMF when users press keys. What is the cause of the issue?

- A. Payload type 110 was negotiated rather than type 101.
- B. DTMF was negotiated properly in these messages.
- C. DTMF was not negotiated on the call.
- D. G.729 rather than G.711ulaw was negotiated.

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

NEW QUESTION: 141

Refer to the exhibit.

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

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

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

NEW QUESTION: 144

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 is not required.
- B. Set up for each group to use IPv4 and IPv6.
- C. Set up media ports for each port group to use IPv4.
- D. Set up IPv4 and IPv6 in Cisco Unified CM.

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

Section: Collaboration Applications

NEW QUESTION: 145

Refer to the Exhibit.

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

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

- A. video codec h264
- B. associate application Cisco Unified Border Element
- C. maximum sessions hardware 50
- D. codec pass-through

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

NEW QUESTION: 146

Refer to the exhibit.

The exhibit displays three screenshots of the Cisco Configuration Assistant (CCA) interface, showing the configuration of Calling Search Spaces (CSS) and their associated route partitions.

Global-CSS Configuration:

- Calling Search Space Information:** Name: Global-CSS, Description: Line Level CSS for calls including International.
- Route Partitions for this Calling Search Space:**
 - Available Partitions: 8851, BlockFraud-PT, BlockGlobal-PT, BlockGlobal-PT, BlockLD-PT.
 - Selected Partitions: BlockFraud-PT, BlockSpecial-PT, Test1-Svc-PT, Test2-Svc-PT.

Intl_CSS Configuration:

- Calling Search Space Information:** Name: Intl_CSS, Description: Calls including INTL.
- Route Partitions for this Calling Search Space:**
 - Available Partitions: 8851, BlockFraud-PT, BlockFraud-PT, BlockGlobal-PT, BlockGlobal-PT.
 - Selected Partitions: LOCAL_CALLS, International_PT.

Unrestricted-CSS Configuration:

- Calling Search Space Information:** Name: Unrestricted-CSS, Description: Line Level CSS for calls including Unrestricted.
- Route Partitions for this Calling Search Space:**
 - Available Partitions: 8851, BlockFraud-PT, BlockGlobal-PT, BlockGlobal-PT, BlockLD-PT.
 - Selected Partitions: BlockFraud-PT.

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

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

NEW QUESTION: 147

A customer has Cisco Unity Connections that is integrated with LDAP. As a Unity Connection administrator, you have received a request to change the first name for VM user. Where must the change be performed?

- A. Cisco Unity Connection
- B. Cisco Unified Communications Manager end user
- C. Active Directory
- D. Cisco IM and Presence

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

Section: Infrastructure and Design

NEW QUESTION: 148

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?



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

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

NEW QUESTION: 149

Which feature is enabled by Cisco Mobile and Remote Access?

- A. Cisco UCC Express clients can obtain VPN connectivity to Cisco UCC Enterprise.
- B. VPN connectivity can be established to Cisco UCM.
- C. Clients such as Cisco Jabber can use call control on Cisco UCM.
- D. Internal SIP clients registered to Cisco UCM can call external companies.

Answer: (SHOW ANSWER)

NEW QUESTION: 150

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

- A. IPP bits

- B. DCSP bits
- C. IP ECN bits
- D. 802.1p User Priority bits

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

Section: QoS

NEW QUESTION: 151

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

- A. On Cisco UCM Administrator page system > Device Pool > Cisco UCM group
- B. On Cisco UCM Administrator page system > Enterprise Parameters
- C. On Cisco UCM Administration Page Device > Phone > Phone Configuration page
- D. On Cisco UCM Administrator Page server > Cisco UCM

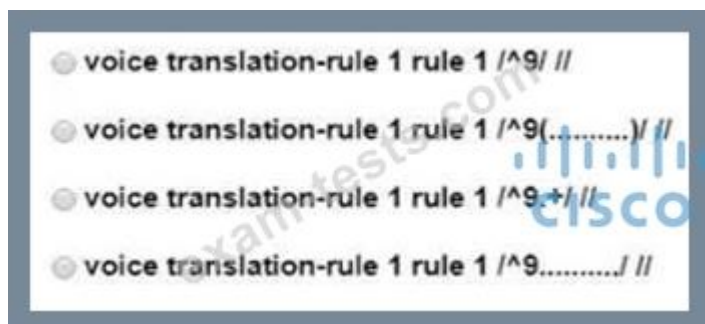
Answer: ([SHOW ANSWER](#)**)**

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NEW QUESTION: 152

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

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

NEW QUESTION: 153

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 is a possible cause of the PRI issue?

- A. The controller shut down.
- B. The cable is unplugged.
- C. The framing is configured incorrectly.
- D. The clock source is incorrect.

Answer: D (LEAVE A REPLY)

NEW QUESTION: 154

An engineer must configure a route pattern that can route all +E.164 globalized international numbers for the plans of all countries. Which Cisco UCM configuration accomplish this task?

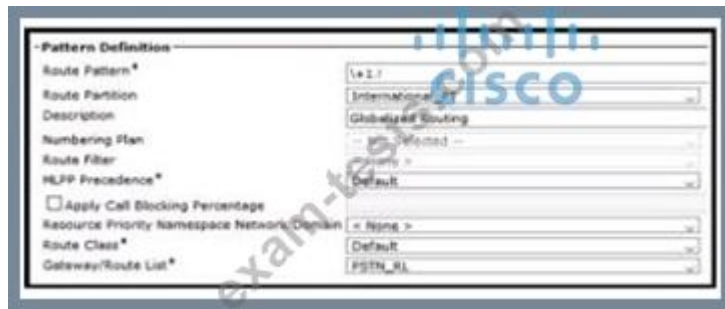
A)

Pattern Definition	
Route Pattern*	+0111
Route Partition	International_PT
Description	Globalized Routing
Numbering Plan	-- Not Selected --
Route Filter	None
HLPP Precedence*	Default
☐ Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	PSTN_RL

B)

Pattern Definition	
Route Pattern*	0111
Route Partition	International_PT
Description	Globalized Routing
Numbering Plan	-- Not Selected --
Route Filter	None
HLPP Precedence*	Default
☐ Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	PSTN_RL

C)



D)



A. Option C

B. Option D

C. Option A

D. Option B

Answer: B (LEAVE A REPLY)

NEW QUESTION: 155

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 extend trust

C. platform qos trust extend cos 3

D. platform qos trust extend

Answer: D (LEAVE A REPLY)

NEW QUESTION: 156

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

A. third-party SIP device

B. Cisco Dual Mode for iPhone

C. Cisco Dual Mode for Android

D. Cisco Unified Client Services Framework

Answer: (SHOW ANSWER)

Explanation/Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/jabber/11_5/CJAB_BK_D00D8CBD_00_deployment-installation-guide-cisco-jabber115/CJAB_BK_D00D8CBD_00_deployment-installation-guide-cisco-jabber115_chapter_01000.html

NEW QUESTION: 157

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. Upload the custom configuration in the TFTP server in Cisco Unified CM.
- D. Apply the ccm-manager configuration commands to the gateway.

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

NEW QUESTION: 158

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

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

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

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.

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/11_5_1_S_U1/cucm_b_system-configuration-guide-1151su1/cucm_b_system-configuration-guide-1151su1_chapter_0111.html

NEW QUESTION: 159

An administrator troubleshoots call flows and suspects that there are issues with the dial plan.

Which tool enables a quick analysis of the dial plan and provides call flows of dialed digits?

- A. Dialed Number Analyzer
- B. Digit Analysis Analyzer
- C. Cisco Dial Plan Analyzer
- D. Dial Plan Analyzer

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

NEW QUESTION: 160

Which configuration step is necessary for a Cisco SIP phone to synchronize its time with a specific source?

- A. Add a Phone NTP Reference to the Date/Time Group.
- B. Assign the device to the correct region.
- C. Change the Time Format from 24-hour to 12-hour.
- D. Change the Time Zone from "America/Los_Angeles" to "Etc/GMT+8".

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

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_0110.html

NEW QUESTION: 161

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

- A. linecode ami
- B. linecode hdb3
- C. linecode b8zs
- D. linecode esf

Answer: C (LEAVE A REPLY)

AMI is the default so to change the default you would need to set to something else and the most common linecode is b8zs.

Step 7	<code>linecode {ami b8zs hdb3}</code> Example: <code>Router(config-controller)# linecode ami</code>	Sets the line-encoding method to match that of your telephone-company service provider. Keywords are as follows: • ami --Alternate mark inversion (AMI), valid for T1 or E1 controllers. Default for T1 lines. • b8zs --B8ZS, valid for T1 controllers only. • hdb3 --High-density bipolar 3 (hdb3), valid for E1 controllers only. Default for E1 lines.
Step 8		

<https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/isdn/configuration/15-mt/vi-15-mt-book/vi-isdn-vi-cfg.html>

NEW QUESTION: 162

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)

NEW QUESTION: 163

An incoming off-net call to a user fails. An engineer notices that the off-net call is G.711, but the phone accepts only G.729. Which media resource on a Cisco Unified Border Element and Cisco Unified Communications Manager must the engineer configure to Manage the engineer configure to manage the codec negotiation?

- A. CFB
- B. MOH
- C. transcoder
- D. MTP

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 164

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

- A. The DNS server was not specified, which is needed to resolve the DHCP server IP address.
- B. The Cisco IP Phone does not know the IP address of any of the Cisco UCM Subscriber nodes.
- 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: 165

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

- A. IM and Presence subclusters (configured on Cisco UCM)
- B. Presence Redundancy Groups (configured on Cisco UCM)
- C. IM and Presence subclusters (configured on Cisco Unified IM and Presence)
- D. Presence Redundancy Groups (configured on Cisco Unified IM and Presence)

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

NEW QUESTION: 166

A collaboration engineer troubleshoots issues with a Cisco IP Phone 7800 Series. The IPv4 address of the phone is reachable via ICMP and HTTP, and the phone is registered to Cisco UCM. However the engineer cannot reach the CU of the phone.

Which two actions in Cisco UCM resolve the issue? (Choose two)

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

Answer: ([SHOW ANSWER](#))

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NEW QUESTION: 167

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 when calling the pilot number. Where should an engineer configure this feature?

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

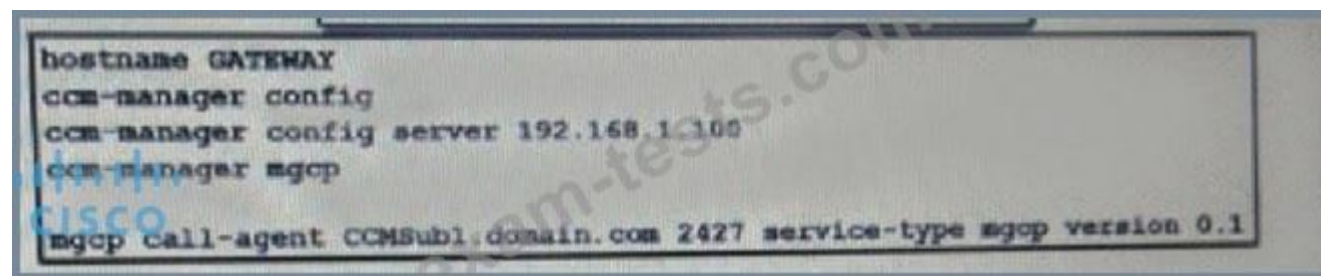
Answer: D (LEAVE A REPLY)

Section: Collaboration Applications

Explanation/Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/10x/administration/guide/10xcucsagx/10xcucsag170.html

NEW QUESTION: 168

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: B (LEAVE A REPLY)

NEW QUESTION: 169

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?

- ☐ voice translation-rule 1 rule 1 /^9/ //
- ☐ voice translation-rule 1 rule 1 /^9(.....)/ //
- ☐ voice translation-rule 1 rule 1 /^9.+/ //
- ☐ voice translation-rule 1 rule 1 /^9...../ //

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

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

NEW QUESTION: 170

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

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

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

NEW QUESTION: 171

Refer to the Exhibit.

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

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

- A. associate application Cisco unified border element
- B. codec pass-through
- C. maximum sessions hardware 50
- D. video codec h264

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 172

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

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

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

Answer: ([SHOW ANSWER](#))

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