# Cisco.350-801.v2022-03-26.q108

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|---|---|
| **Exam Code:** | 350-801 |
| **Exam Name:** | Implementing and Operating Cisco Collaboration Core Technologies |
| **Certification Provider:** | Cisco |
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| https://www.freepdfdumps.com/Cisco.350-801.v2022-03-26.q108.html | |

**NEW QUESTION: 1**
When a remote office location is set up with limited bandwidth resources, which codec carries the most voice calls?
**A.** G.711
**B.** G.723
**C.** G.729
**D.** G.722
**Answer: B (LEAVE A REPLY)**

**NEW QUESTION: 2**
What is a valid class included in the 8-Class QoS Strategy in a VoIP network?
**A.** Assured Forwarding
**B.** Broadcast Video
**C.** Multimedia Conferencing
**D.** Real-Time Interactive
**Answer: (SHOW ANSWER)**
Explanation/Reference: https://www.ciscopress.com/articles/article.asp?p=2756478&seqNum=8

**NEW QUESTION: 3**
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)**
Section: Infrastructure and Design
Explanation/Reference: https://blog.router-switch.com/2013/03/dhcp-option-150-dhcp-option-66/

**NEW QUESTION: 4**

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: 5**

Which two functionalities does Cisco Expressway provide in the Cisco Collaboration architecture? (Choose two.)

**A.** Survivable Remote Site Telephony functionality

**B.** customer interaction management services

**C.** secure firewall and NAT traversal for mobile or remote Cisco Jabber and TelePresence Video endpoints

**D.** MGCP gateway registration

**E.** Secure business-to-business communications

**Answer: C,E (LEAVE A REPLY)**

Section: Infrastructure and Design

**NEW QUESTION: 6**

User A Calls user. The call gets connected, but the quality is bed. What are two reasons for this issue? (Choose two)

**A.** No region relationship

**B.** Incompatible codec

**C.** Network congestion

**D.** Incorrect QoS

**E.** Incorrect partition

**Answer: C,D (LEAVE A REPLY)**

**NEW QUESTION: 7**

Refer to the exhibit.

```
admin:utils ntp status
ntpd (pid 17428) is running...

     remote           refid      st t when poll reach   delay   offset  jitter
==============================================================================
*192.168.1.1     17.253.14.125    2 u   36   64   377    0.435   0.039   0.047
 192.168.1.2     .INIT.          16 u    -   64     0    0.000   0.000   0.000
```

A collaboration engineer adds a redundant NTP server to an existing Cisco Collaboration solution On the Cisco UCM OS Administration page the new NTP server shows as "Not Accessible" Which action resolves this issue?

**A.** Delete and re-add the new NTP server via the Cisco UCM command-line interface
**B.** Configure the "reach" value as "377" for the new NTP server
**C.** Start the NTP service on the new NTP server
**D.** Restart NTPD on the Cisco UCM server

**Answer: A (LEAVE A REPLY)**

**NEW QUESTION: 8**

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 g711ulaw
**B.** codec g726r32
**C.** codec g723ar63
**D.** codec g729br81

**Answer: A (LEAVE A REPLY)**

**NEW QUESTION: 9**

```
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: D (LEAVE A REPLY)**

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: 10**

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

**A.** show policy-map

**B.** show class-map interface GigabitEthernet 1/0/1

**C.** show policy-map interface GigabitEthernet 1/0/1

**D.** show access-lists

**Answer: C (LEAVE A REPLY)**

**NEW QUESTION: 11**

Which two steps should be taken to provision after the Self-Provisioning feature was configured for end users?

**A.** Plug the phone into the network.

**B.** Dial the hunt pilot extension and associate the phone to an end user.

**C.** Dial the self-provisioning IVR extension and associate the phone to an end user.

**D.** Ask the Cisco UCM administrator to associate the phone to an end user.

**E.** Enter settings menu on the phone and press *,*,# (star, star, pound).

**Answer: (SHOW ANSWER)**

**NEW QUESTION: 12**

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

**A.** .

**B.** !

**C.** @

**D.** *

**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_0100111.html

**NEW QUESTION: 13**

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

**A.** RTP

**B.** SCCP

**C.** MGCP

**D.** H.323

**E.** SIP

**Answer: (SHOW ANSWER)**

Section: Protocols, Codecs, and Endpoints

Explanation/Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/11x/design/guide/b_11xcucdg/b_11xcucdg_chapter_00.html#ID-2342-00000133

**NEW QUESTION: 14**

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

**A.** *

**B.** @

**C.** .

**D.** !

**Answer: (SHOW ANSWER)**

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 (IPsubnet); 172.18.201.18/21 (IP subnet). Valid characters for IP addresses: 0-9, ., and /

**NEW QUESTION: 15**

Which issue cause slips on a PRI?

**A.** incorrect clock source

**B.** incorrectly encapsulation

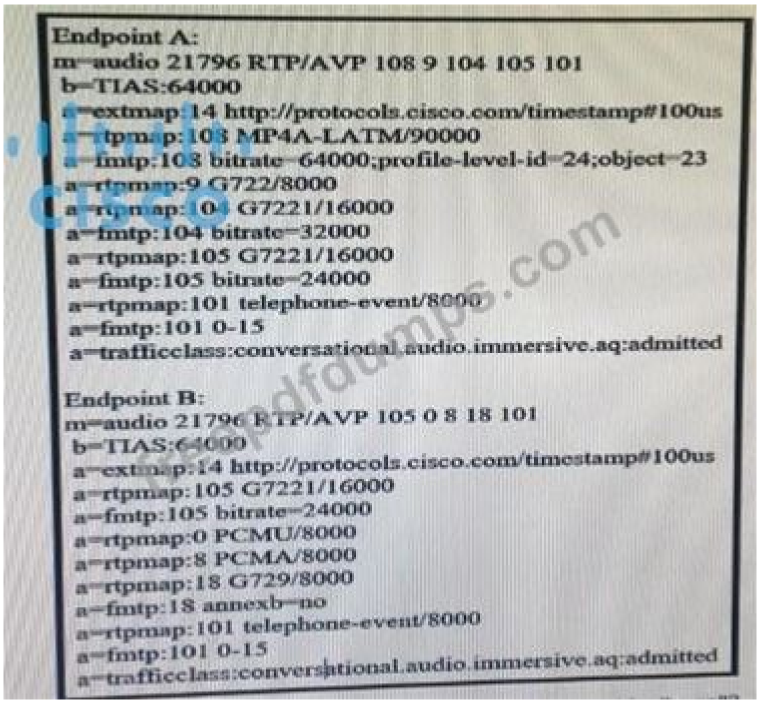
**C.** change in the line code

**D.** incorrectly configured time zone.

**Answer: (SHOW ANSWER)**

**NEW QUESTION: 16**

Refer to the exhibit.



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

Endpoint B:
m=audio 21796 RTP/AVP 105 0 8 18 101
b=TIAS:64000
a=extmap:14 http://protocols.cisco.com/timestamp#100us
a=rtpmap:105 G7221/16000
a=fmtp:105 bitrate=24000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=trafficclass:conversational.audio.immersive.aq:admitted

Endpoint A calls endpoint B.

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

**A.** Telephone-event/8000

**B.** PCMA/8000

**C.** G277/8000

**D.** G7221/16000

**Answer: (SHOW ANSWER)**



**NEW QUESTION: 17**

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

**A.** 1

**B.** 8

**C.** 16

**D.** 4

**Answer: C (LEAVE A REPLY)**

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

**NEW QUESTION: 18**
Which method is used to avoid toll fraud with Cisco Unified Communications Manager calls?
**A.** Call policy service
**B.** class of service
**C.** TOLLFRAUD_APP
**D.** Default zone access rules
**Answer: B (LEAVE A REPLY)**

**NEW QUESTION: 19**
Which recommendation is the best practice for marking video and voice media in a cisco Unified Communications network?
A)
Voice Cos 6 (IP Precedence 4, PHB AF41, or DSCP 24)
Video Cos 5 (IP Precedence 4, PHB EF, or DSCP 34)
B)
Voice Cos 5 (IP Precedence 5, PHB EF, or DSCP 46)
Video Cos 4 (IP Precedence 4, PHB AF41, or DSCP 34)
C)
Voice Cos 5 (IP Precedence 2, PHB EF, or DSCP 48)
Video Cos 4 (IP Precedence 4, PHB AF41, or DSCP 46)
D)
Voice Cos 5 (IP Precedence 6, PHB AF41, or DSCP 16)
Video Cos 4 (IP Precedence 5, PHB EF, or DSCP 32)
**A.** Option D
**B.** Option A
**C.** Option B
**D.** Option C
**Answer: C (LEAVE A REPLY)**

**NEW QUESTION: 20**

```
NVITE sip:1@10.10.10.219;user=phone SIP/2.0
ia: SIP/2.0/TCP 10.10.10.84:50083;branch=z9hG4bK471df613
rom: "1234 - My Phone" <sip:1234@10.10.10.219>;tag=381c1aba7a78002c558eda31-12b8af63
o: <sip:1@10.10.10.219>
all-ID: 381c1aba-7a78000d-4ca6894a-41dd3e0f@10.10.10.84
ax-Forwards: 70
Seq: 101 INVITE
ontact: <sip:1234@10.10.10.84:50083;transport=tcp>
llow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
llow-Events: kpml,dialog
ontent-Type: application/sdp
ontent-Length: 658

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



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

**A.** REGISTER

**B.** UPDATE

**C.** PRACK

**D.** NOTIFY

**E.** SUBSCRIBE

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

Section: Protocols, Codecs, and Endpoints

**NEW QUESTION: 21**

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

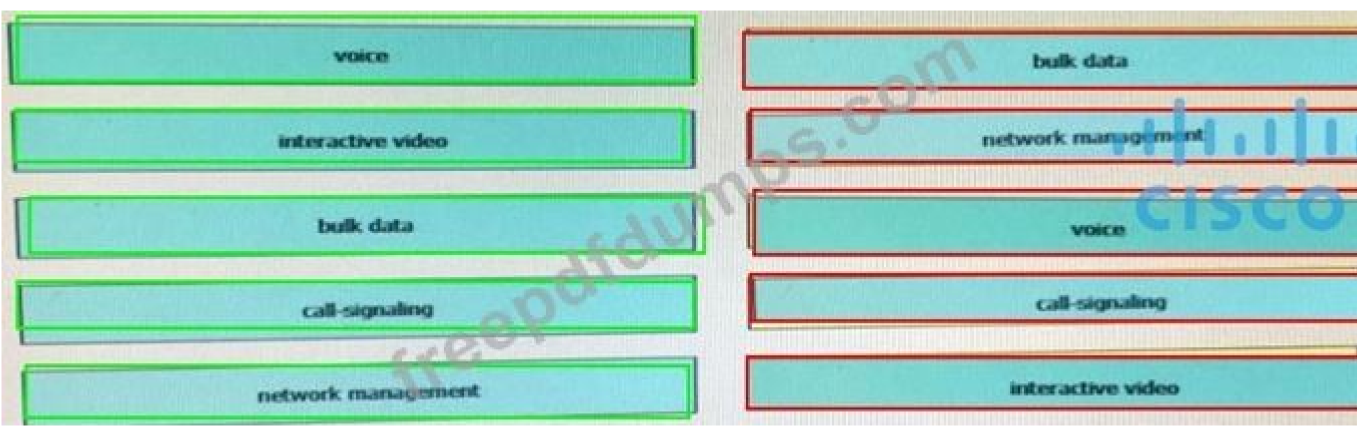
**Answer: C ([LEAVE A REPLY])**

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: 22**

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


**Answer:**




**NEW QUESTION: 23**

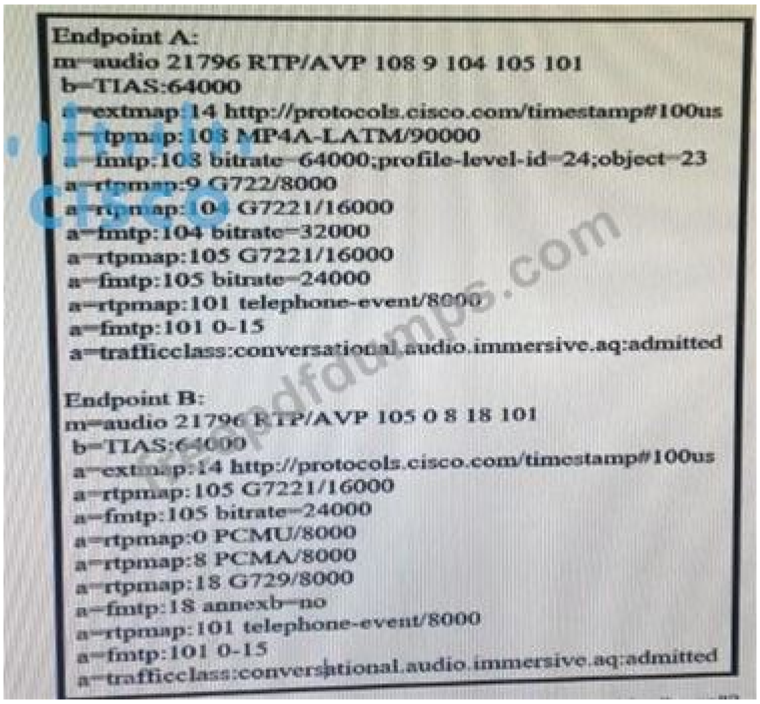
Which DTMF relay method configured on a SIP dial-peer will ensure that a media resource is not invoked by Unified CM for calls to UCCX IVRs?

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

**Answer: C (LEAVE A REPLY)**

**NEW QUESTION: 24**

Refer to the exhibit.



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

Endpoint B:
m=audio 21796 RTP/AVP 105 0 8 18 101
b=TIAS:64000
a=extmap:14 http://protocols.cisco.com/timestamp#100us
a=rtpmap:105 G7221/16000
a=fmtp:105 bitrate=24000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=trafficclass:conversational.audio.immersive.aq:admitted
```

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

**A.** Telephone-event/8000

**B.** G7221/16000

**C.** PCMA/8000

**D.** G277/8000

**Answer: D (LEAVE A REPLY)**

**NEW QUESTION: 25**

Refer the exhibit.

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

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

**A.** PRI does not have an IP address configured on the interface.

**B.** Layer 1 is down on the controller.

**C.** Noting, the PRI is sending keepalives.

**D.** Layer 2 is down on the controller.

**Answer: D (LEAVE A REPLY)**

**NEW QUESTION: 26**

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.729A

**B.** G.722.1

**C.** iLBC

**D.** G.711alaw

**Answer: A (LEAVE A REPLY)**

**NEW QUESTION: 27**

Refer to the exhibit.

```
hostname GATEWAY
ccm-manager config
ccm-manager config server 192.168.1.100
ccm-manager mgcp

mgcp call-agent CCMSub1.domain.com 2427 service-type mgcp version 0.1
```

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

**A.** Device(config)# mgcp enable

**B.** Device(config)# ccm-manager enable

**C.** Device (config)# mgcp

**D.** Device (config) # com-manager active

**Answer: (SHOW ANSWER)**

**NEW QUESTION: 28**

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

**A.** XMPP

**B.** SIP/SIMPLE

**C.** Jabber

**D.** P2P

**Answer: A (LEAVE A REPLY)**

**NEW QUESTION: 29**

Refer to the exhibit.

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

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

**A.** G.711alaw

**B.** G.729r8

**C.** No codec is used (missing codec command).

**D.** G.711ulaw

**Answer: (SHOW ANSWER)**

**NEW QUESTION: 30**

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

**A.** Change the time format from 24-hour to 12-hour.

**B.** Add a phone NTP Reference to the date/time Group.

**C.** Assign the device to the correct region.

**D.** Change the Time Zone from "America/Los_Angeles" to "Etc/GMT+8".

**Answer: B (LEAVE A REPLY)**

**NEW QUESTION: 31**

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

Refer to the exhibit. A call is failing to establish between two SIP Devices The called device answers with this SOP. Which SDP parameter causes this issue?

**A.** The calling device did not offer a ptime value.

**B.** The RTP port is set to 0.

**C.** The media stream is set to sendonly.

**D.** The payload for G.711ulaw must be 18.

**Answer: (SHOW ANSWER)**

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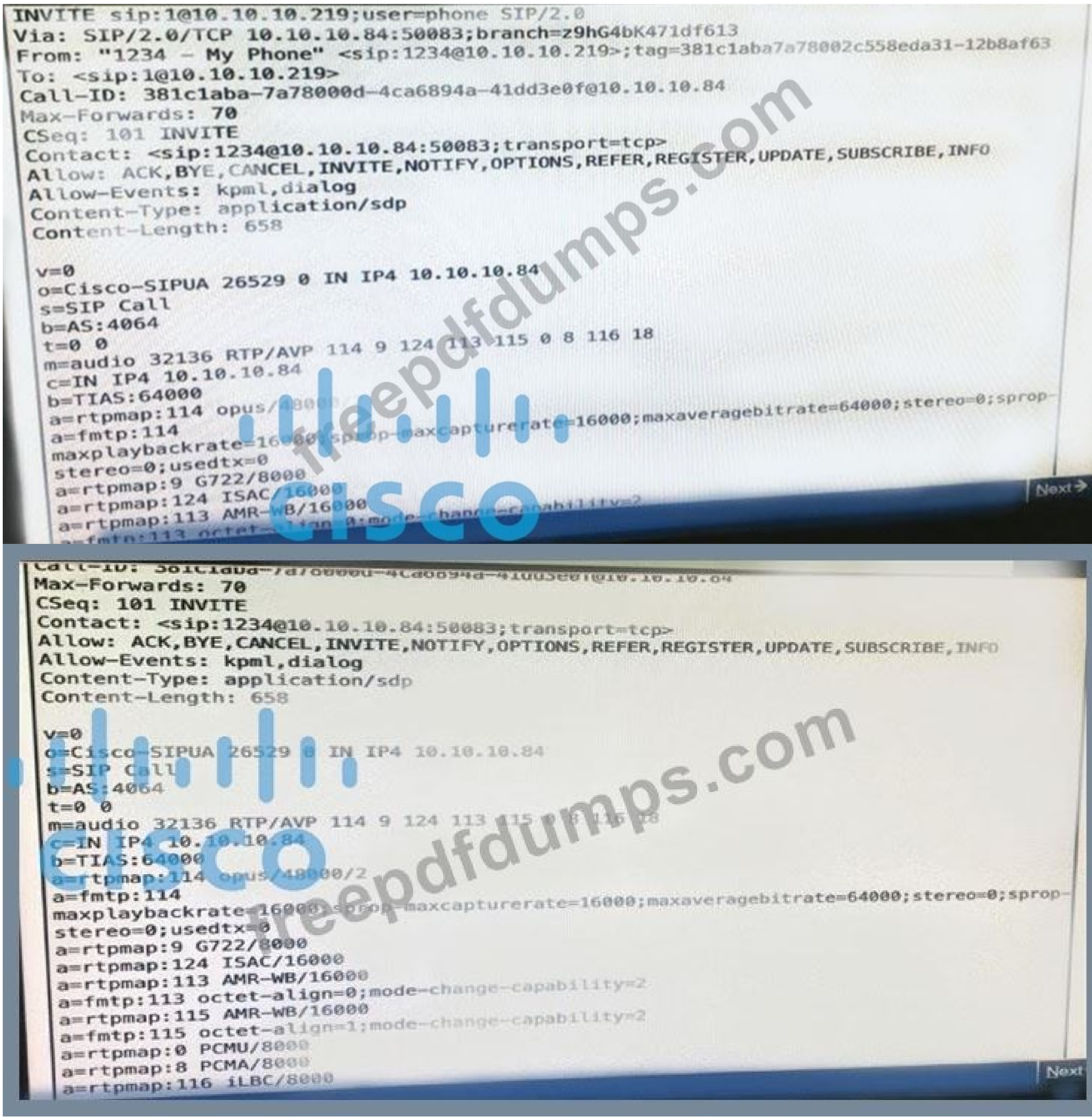
**Special Discount: Freepdfdumps**)

**NEW QUESTION: 32**

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-maxcapturerate=16000;maxaveragebitrate=64000;stereo=0;sprop-
stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0;mode-change-capability=2
```

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

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

```
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114
maxplaybackrate=16000;sprop-maxcapturerate=16000;maxaveragebitrate=64000;stereo=0;sprop-
stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0;mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=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.** PRACK
**C.** SUBSCRIBE
**D.** NOTIFY
**E.** UPDATE

**Answer: C,D (LEAVE A REPLY)**

**NEW QUESTION: 33**

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

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

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.** Cisco Unity Connection is configured on G.729 only. Cisco Unity Connection must be reconfigured to support iLBC.
**C.** The Cisco Unity Connection Call Handler is configured for a -Release to Switch" transfer type. Transfers must be disabled for the Cisco Unity Connection Call Handler.
**D.** Cisco Unified Border Element is sending the incorrect media IP address. The IP address of the loopback interface must be advertised in the SDP.
**Answer: A (LEAVE A REPLY)**

**NEW QUESTION: 34**
Which settings are needed to configure the SIP route pattern in Cisco Unified Communications Manager?
**A.** pattern usage, IPv6 pattern, and SIP trunk/Route list
**B.** pattern usage, IPv4 pattern, IPv6 pattern, and description
**C.** pattern usage, IPv4 pattern, and SIP trunk/Route list
**D.** SIP trunk/Route list, description, and IPv4 pattern
**Answer: C (LEAVE A REPLY)**
Explanation/Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0100111.pdf

**NEW QUESTION: 35**
What is a software-based media resource that is provided by the Cisco IP Voice Media Streaming Application?
**A.** video conference bridge
**B.** auto-attendant
**C.** transcoder
**D.** annunciator
**Answer: D (LEAVE A REPLY)**
Explanation/Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab09/clb09/media.html

**NEW QUESTION: 36**
Which DSCP class selector is necessary to mark scavenger traffic?
**A.** CS1
**B.** AF11
**C.** CS2
**D.** AF21
**Answer: (SHOW ANSWER)**

**NEW QUESTION: 37**

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

**A.** Ensure that applications will perform QoS.

**B.** Deploy the VMs to a server running Cisco UCM.

**C.** Provide a guaranteed bandwidth of 10 Mbps.

**D.** Avoid hardware oversubscription.

**Answer: (SHOW ANSWER)**

**NEW QUESTION: 38**

Refer to the exhibit.

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

**A.** Add a transcoder that supports G.711ualw and G.729.

**B.** ADD a media termination point that supports G.711ulaw and G.729.

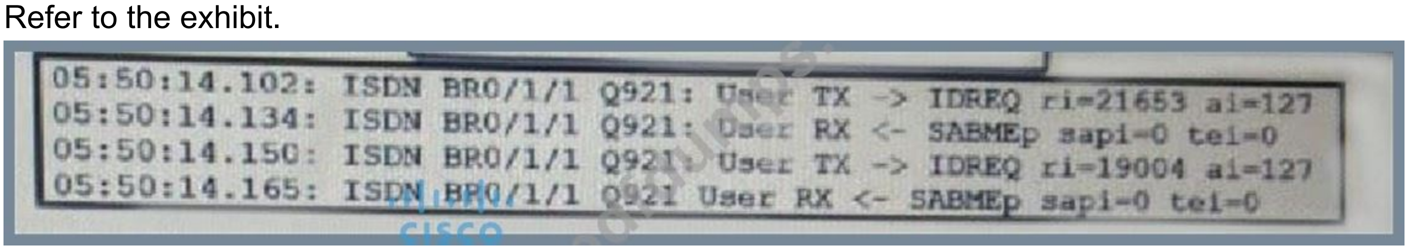
**C.** Nothing; both sides support payload type.101.

**D.** Noting; both sides support G.729.

**Answer: A (LEAVE A REPLY)**

**NEW QUESTION: 39**

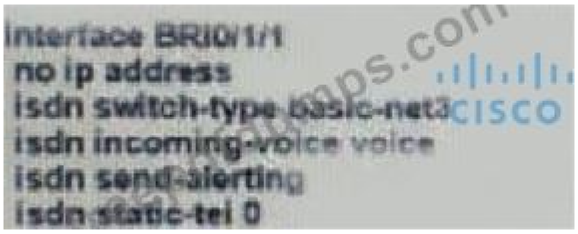
Refer to the exhibit.



```
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
```

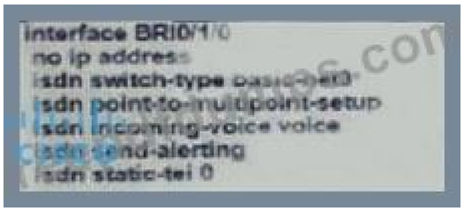
A customer submits this output, captured on a Cisco IOS router. Assuming that an MGCP gateway is configured with a ISDN BRI interface, which BRI changes resolve the issue?

A)



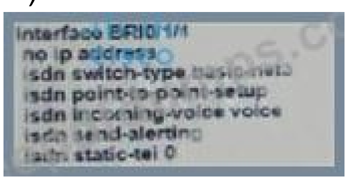
```
interface BRI0/1/1
 no ip address
 isdn switch-type basic-net3
 isdn incoming-voice voice
 isdn send-alerting
 isdn static-tei 0
```

B)



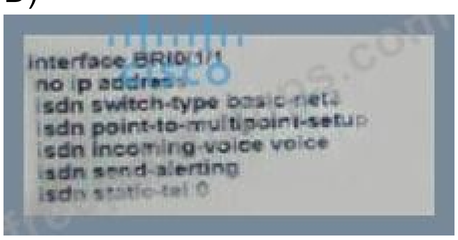
```
interface BRI0/1/0
 no ip address
 isdn switch-type basic-net3
 isdn point-to-multipoint-setup
 isdn incoming-voice voice
 isdn send-alerting
 isdn static-tei 0
```

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 C

**C.** Option D

**D.** Option B

**Answer: (SHOW ANSWER)**

**NEW QUESTION: 40**

Which attribute contains an XMPP stanza?

**A.** iq

**B.** message

**C.** presence

**D.** type

**Answer: D (LEAVE A REPLY)**

**NEW QUESTION: 41**

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

**A.** mgcapp

**B.** ccm-manager fallback-mgcp

**C.** mgcp call-agent

**D.** ccm-manager redundant-host

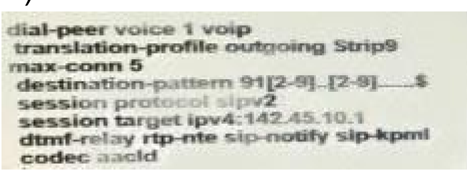
**Answer: B (LEAVE A REPLY)**

**NEW QUESTION: 42**

A collaboration engineer must configure Cisco Unified Border Element to support up to five concurrent outbound calls across an Ethernet Link with a bandwidth of 160 kb to the Internet Telephony service provider.

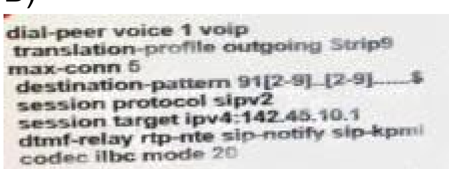
Which set of commands allows the engineer to complete the task without compromising voice quality?

A)



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

B)

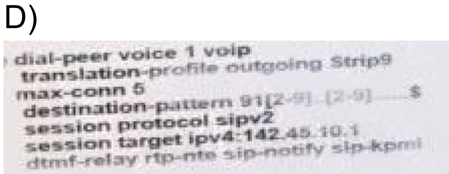


```
dial-peer voice 1 voip
 translation-profile outgoing Strip9
 max-conn 5
 destination-pattern 91[2-9]..[2-9]......$
 session protocol sipv2
 session target ipv4:142.45.10.1
 dtmf-relay rtp-nte sip-notify sip-kpml
 codec ilbc mode 20
```

C)

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

D)



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

**A.** Option D

**B.** Option C

**C.** Option B

**D.** Option A

**Answer: C (LEAVE A REPLY)**

**NEW QUESTION: 43**

Configuration of DNS is required to achieve a fully functional Cisco UCM system Cisco UCM uses DNS to resolve fully qualified domain names to IP addresses for which destinations?

**A.** Application server name

**B.** Cisco Unified Communications Manager Name

**C.** Sip trunk

**D.** Primary TFTP Server for option 150

**Answer: C (LEAVE A REPLY)**

**NEW QUESTION: 44**

Which configuration on Cisco Unified Communication Manager is required for SIP MWI to work?
**A.** The line partition must be inside the rerouting CSS assigned to the Cisco assigned to the Cisco unity connection SIP trunk.
**B.** Assign a MWI extension on the mailbox.
**C.** The line partition must be inside the inbound CSS assigned to the CUC SIP trunk.
**D.** Set the "Enable message waiting indicator" on the port group.
**Answer: (SHOW ANSWER)**

**NEW QUESTION: 45**
Refer the exhibit.

```
000193: Dec 5 14:35:31.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:32.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:33.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:34.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
```
Given this "debug isdn q921" output, what is the problem with the PRI?
**A.** Noting, the PRI is sending keepalives.
**B.** Layer 2 is down on the controller.
**C.** Layer 1 is down on the controller.
**D.** PRI does not have an IP address configured on the interface.
**Answer: (SHOW ANSWER)**

**NEW QUESTION: 46**
When a new SIP phone Is registered to Cisco Unified Communications Manager, it keeps failing and showing an "unprovisioned" error message in the phone display. Which problem is a possible cause of this issue?
**A.** Auto-registration is disabled on the Cisco Unified Communications Manager nodes and the phone device does not have a DN configured.
**B.** The DN configuration for this phone is shared with an SCCP phone. which is not supported.
**C.** The DHCP settings are set incorrectly and the phone does not have an alternate TFTP defined.
**D.** The phone cannot download and install the latest firmware.
**E.** The DN assigned to the phone is already in use by another SIP phone.
**Answer: D (LEAVE A REPLY)**

**NEW QUESTION: 47**
Refer to the exhibit.

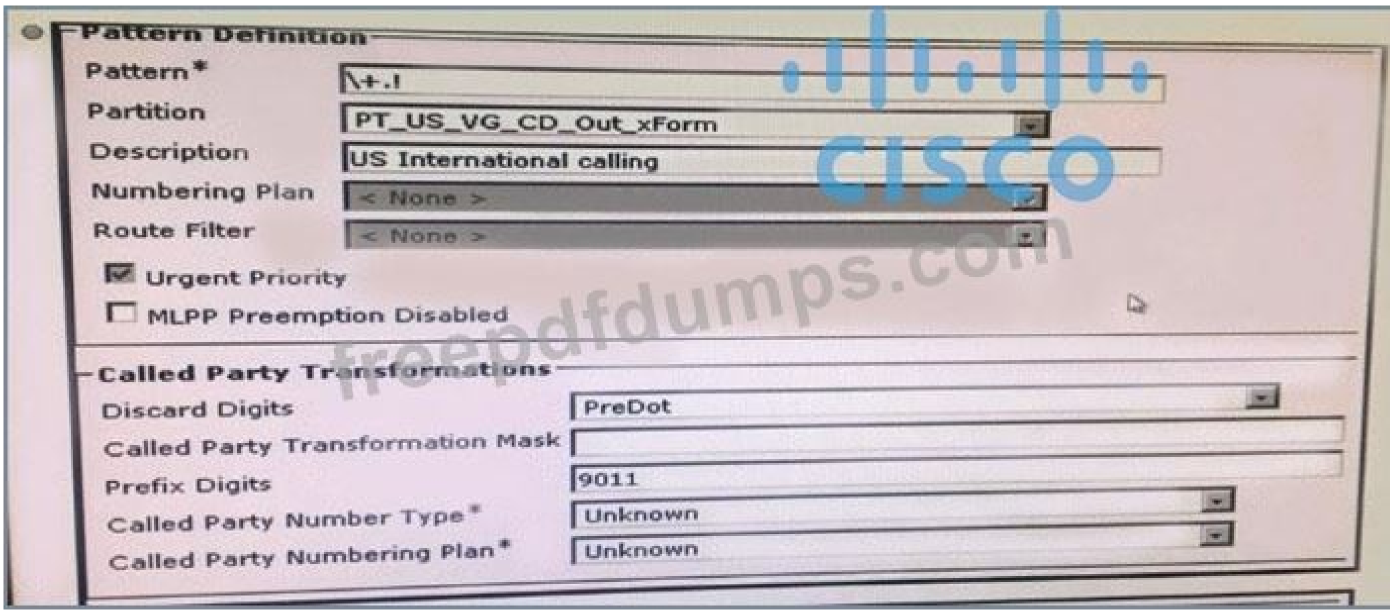
```
INVITE sip:2002@10.10.10.10:5060 SIP/2.0
[..truncated..]
v=0
o=UAC 6107 7816 IN IP4 10.10.10.11
s=SIP Call
c=IN IP4 10.10.10.11
t=0 0
m=audio 8190 RTP/AVP 18 110
c=IN IP4 10.10.10.11
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:110 telephone-event/8000
a=fmtp:110 0-16
a=ptime:20

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.** DTMF was not negotiated on the call.
**B.** DTMF was negotiated properly in these messages.
**C.** Payload type 110 was negotiated rather than type 101.
**D.** G.729 rather than G.711ulaw was negotiated.
**Answer: A (LEAVE A REPLY)**

**NEW QUESTION: 48**
Given this H.323 gateway configuration and using cisco best practices, how must the called party transformation pattern to ensure that a proper ISDN type of number is set?

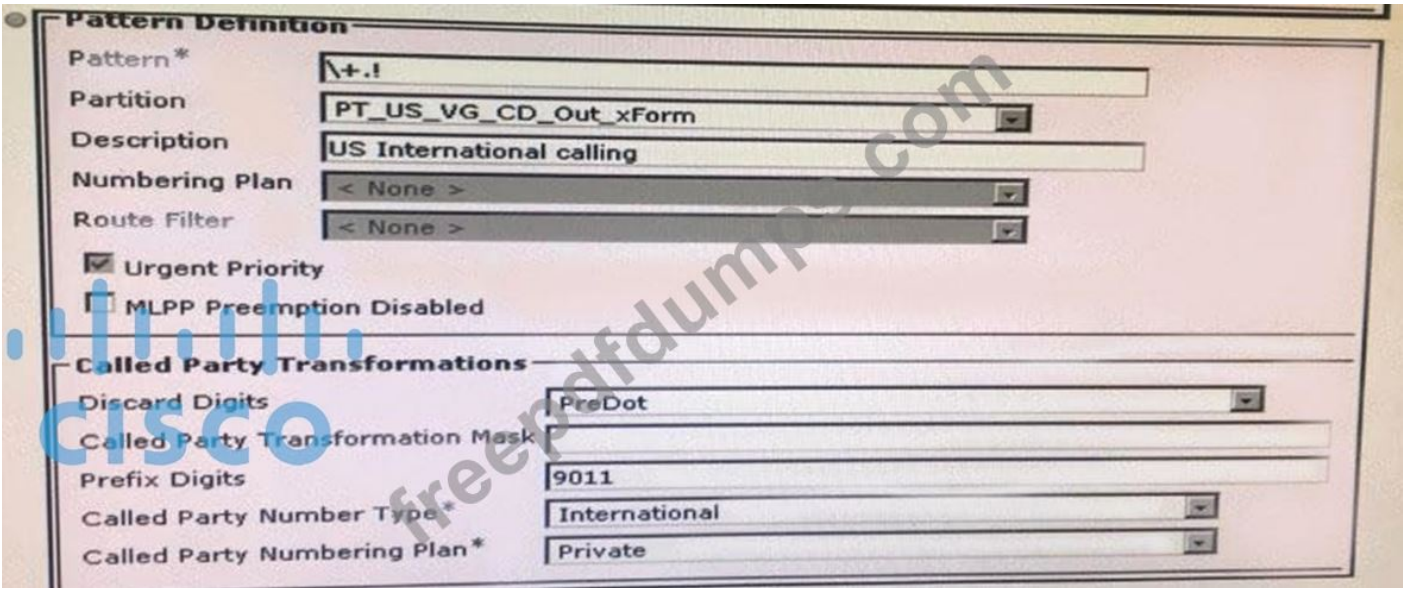
```
voice translation-rule 40
 rule 1 /3...$/ /4085558/
!
voice translation-profile INT
 translate calling 40
!
dial-peer voice 9011 pots
 translation-profile outgoing INT
 destination-pattern 9011T
 port 0/1/0:23
```

A)

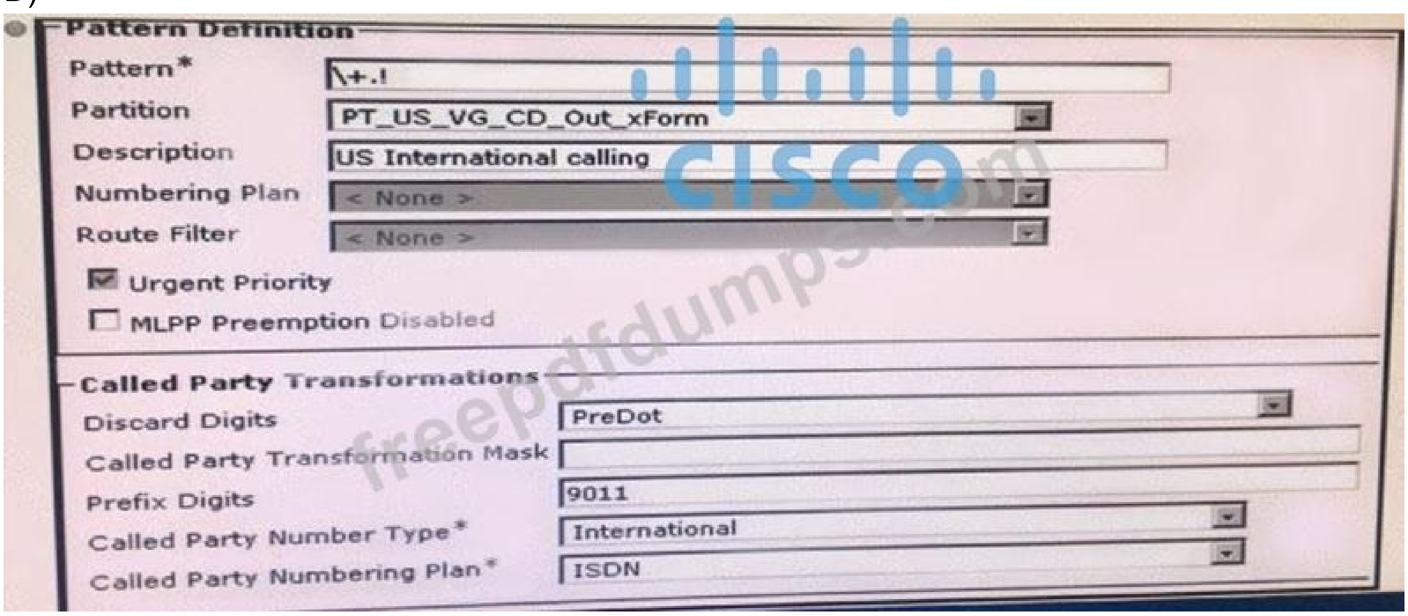



B)


C)




D)




**A.** Option B
**B.** Option D
**C.** Option A
**D.** Option C
**Answer: B (LEAVE A REPLY)**

**NEW QUESTION: 49**
What is the function of the Cisco Unity Connection Call Handler?
**A.** searches a list of extensions until the call is answered
**B.** queues calls
**C.** allows customized scripts for IVR capabilities
**D.** routes calls to a user based on caller input
**Answer: D (LEAVE A REPLY)**

**NEW QUESTION: 50**

What are two functions of Cisco Expressway in the Collaboration Edge? (Choose two)
**A.** Expressway-C provides encryption for Mobile and Remote Access but not for business-to-business communications.
**B.** Expressway-E provides a perimeter network that separates the enterprise network from the Internet.
**C.** The Expressway-C and Expressway-E pair can enable connectivity from the corporate network to the PSTN via a T1/E1 trunk.
**D.** Expressway-E provides a VPN entry point for Cisco IP phones with a Cisco AnyConnect client using authentication based on certificates.
**E.** The Expressway-C and Expressway-E pair can interconnect H 323-to-SIP calls for voice.
**Answer: (SHOW ANSWER)**

**NEW QUESTION: 51**
Refer the exhibit.

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

Given this "debug isdn q921" output, what is the problem with the PRI?
**A.** Noting, the PRI is sending keepalives.
**B.** Layer 1 is down on the controller.
**C.** PRI does not have an IP address configured on the interface.
**D.** Layer 2 is down on the controller.
**Answer: D (LEAVE A REPLY)**

**NEW QUESTION: 52**
Which two conditions must a user meet to provision a new device using the Self-Provisioning feature? (Choose two.)
**A.** The user must have a primary extension.
**B.** At least two DNs must be assigned to the user device.
**C.** The user must be part of "Standard CCM Super User".
**D.** The user must have the appropriate universal device template linked to the user profile.
**E.** The user must have at least one user device profile assigned.
**Answer: A,D (LEAVE A REPLY)**
Section: Infrastructure and Design

**NEW QUESTION: 53**
Which IP precedence value maps to DSCP EF?
**A.** 0
**B.** 1
**C.** 3
**D.** 5

**Answer: D (LEAVE A REPLY)**

**NEW QUESTION: 54**

A secondary device. Which configuration on the voice mailbox makes this change?

**A.** Mobile User

**B.** Alternate Extensions

**C.** Attempt Forward routing rule

**D.** Alternate Names

**Answer: B (LEAVE A REPLY)**

**NEW QUESTION: 55**

Refer to the exhibit.


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

**A.** Strip the leading 011 from the called party number

**B.** Add the isdn switch-type primart-dms100 command to the serial interface.

**C.** Add the bearer-cap speech command to the voice port.

**D.** Assign a transcoder to the MRGL of the gateway.

**Answer: A (LEAVE A REPLY)**

**NEW QUESTION: 56**

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

A)

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

B)

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

C)

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

D)

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

**A.** Option B

**B.** Option A

**C.** Option D

**D.** Option C

**Answer: B** (**LEAVE A REPLY**)

**NEW QUESTION: 57**

Refer to the exhibit.

```
INVITE sip:4000@172.16.1.1:5061 SIP/2.0
Via: SIP/2.0/TLS 172.16.2.143:5061;branch=z9hG4bK0FD315E7
Remote-Party-ID: <sip:+14088335000@172.16.2.143>;party=calling;screen=no;privacy=off
From: <sip:+14088335000@172.27.2.143>;tag=7B42E5F6-9B0
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:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
```

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

**A.** Noting; both sides support G.729.

**B.** Add a transcoder that supports G.711ualw and G.729.

**C.** ADD a media termination point that supports G.711ulaw and G.729.

**D.** Nothing; both sides support payload type.101.

**Answer: B** (**LEAVE A REPLY**)

**NEW QUESTION: 58**

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: 59**
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 should be created for _ cisco-uds _tcp example.com.
**B.** The DNS record should be changed from _collab-edge_tcp example.com.
**C.** The DNS record type should be changed from SRV to A.
**D.** Server 4.2.2.2 is not a valid DNS server.
**Answer: B (LEAVE A REPLY)**

**NEW QUESTION: 60**
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 B
**B.** Option C
**C.** Option A
**D.** Option D
**Answer: A (LEAVE A REPLY)**

**NEW QUESTION: 61**
An administrator installed a Cisco Unified IP 8831 Conference Phone that is failing to register Which two actions are taken to troubleshoot the problem? (Choose two)
**A.** Verify that the phone's network can access the option 150 server.
**B.** Verify that the switch port of the phone is enabled
**C.** Check the RJ-65 cable.
**D.** Disable HSRP on lhe access layer switch.
**E.** Verify that the RJ-11 cable is plugged into the PC port.
**Answer: A,B (LEAVE A REPLY)**



**NEW QUESTION: 62**
Given this H.323 gateway configuration and using cisco best practices, how must the called party transformation pattern to ensure that a proper ISDN type of number is set?

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

A)


Pattern Definition

Pattern* \+.!

Partition PT_US_VG_CD_Out_xForm

Description US International calling

Numbering Plan < None >

Route Filter < None >

Urgent Priority

MLPP Preemption Disabled

Called Party Transformations

Discard Digits PreDot

Called Party Transformation Mask

Prefix Digits 9011

Called Party Number Type* Unknown

Called Party Numbering Plan* Unknown

B)


Pattern Definition

Pattern* \+.!

Partition PT_US_VG_CD_Out_xForm

Description US International calling

Numbering Plan < None >

Route Filter < None >

Urgent Priority

MLPP Preemption Disabled

Called Party Transformations

Discard Digits PreDot

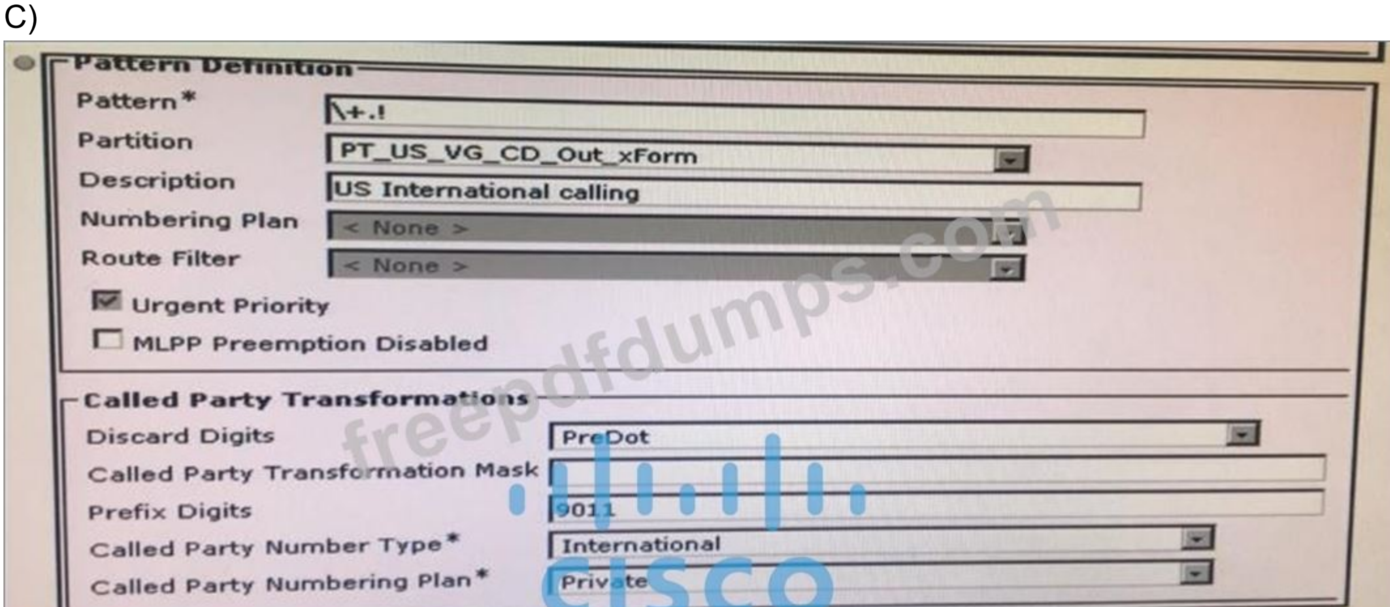
Called Party Transformation Mask

Prefix Digits 9011

Called Party Number Type* Cisco CallManager

Called Party Numbering Plan* Cisco CallManager

C)


Pattern Definition

Pattern* \+.!

Partition PT_US_VG_CD_Out_xForm

Description US International calling

Numbering Plan < None >

Route Filter < None >

Urgent Priority

MLPP Preemption Disabled

Called Party Transformations

Discard Digits PreDot

Called Party Transformation Mask

Prefix Digits 9011

Called Party Number Type* International

Called Party Numbering Plan* Private

D)


**A.** Option A

**B.** Option C

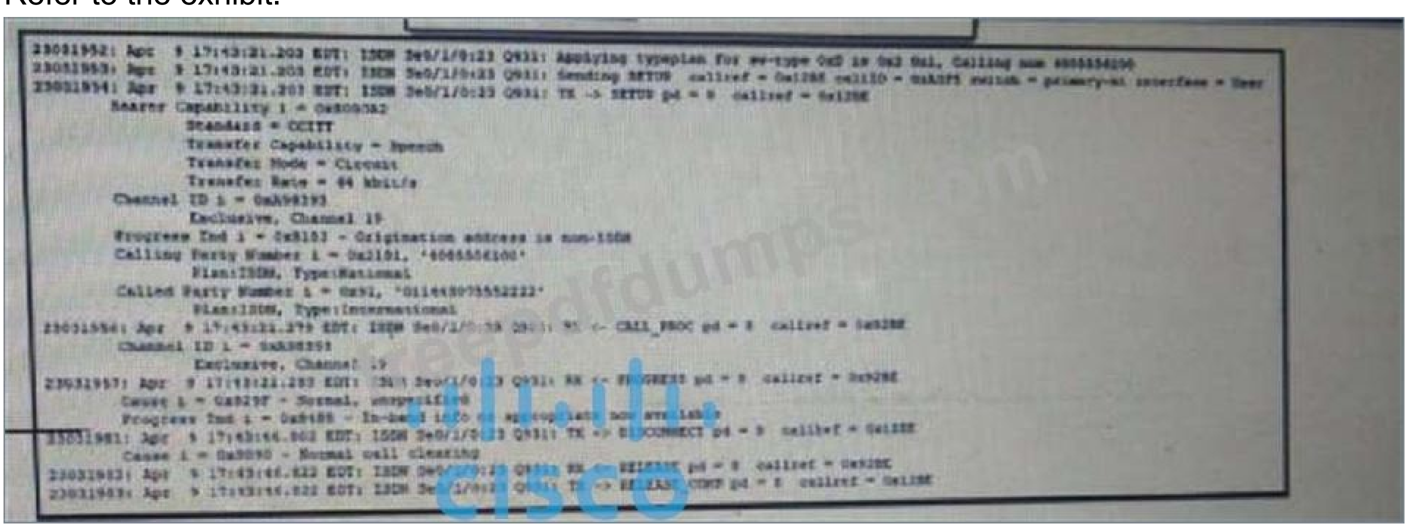
**C.** Option D

**D.** Option B

**Answer: C (LEAVE A REPLY)**

**NEW QUESTION: 63**

Refer to the exhibit.



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

**A.** Add the bearer-cap speech command to the voice port.

**B.** Assign a transcoder to the MRGL of the gateway.

**C.** Add the isdn switch-type primart-dms100 command to the serial interface.

**D.** Strip the leading 011 from the called party number

**Answer: D (LEAVE A REPLY)**

**NEW QUESTION: 64**

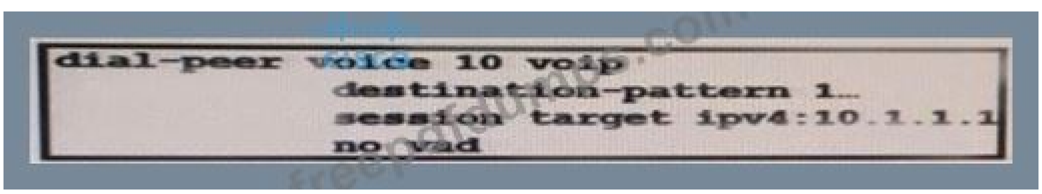
Refer to the exhibit. Which codec should an engineer select for a call mode between "Dallas-REG" & "Austin-REG"?

**A.** G.729

**B.** OPUS

**C.** G.711
**D.** MP4A-LATM
**Answer: A (LEAVE A REPLY)**

**NEW QUESTION: 65**
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.** Enable SSH Access under Product Specific Configuration Layout in Cisco UCM
**C.** Enable Settings Access under Product Specific Configuration Layout in Cisco UCM.
**D.** Enable FIPS Mode under Product Specific Configuration Layout in Cisco UCM
**E.** Set a username and password under Secure Shell Information in Cisco UCM
**Answer: (SHOW ANSWER)**

**NEW QUESTION: 66**
Refer to the exhibit.

```
dial-peer voice 10 voip
 destination-pattern 1..
 session target ipv4:10.1.1.1
 no vad
```
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.711ulaw
**C.** G.729r8
**D.** G.711alaw
**Answer: C (LEAVE A REPLY)**

**NEW QUESTION: 67**
Which description of the function of call handlers in Cisco Unity Connection is true?
**A.** They provide access to a corporate directory by playing an audio list that users and outside callers users and leave messages.
**B.** They collect information from callers by playing a series of questions and recording the answers.
**C.** They answer calls, take message, and provide menus of options.
**D.** They control outgoing calls by allowing you to specify the numbers that Cisco Unity Connection can dial to transfer calls, notify users of massages, and delivery faxes.
**Answer: C (LEAVE A REPLY)**

**NEW QUESTION: 68**

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

**A.** Assign a phone security profile with secure SIP.

**B.** Set the prepare cluster for rollback to pre-8.0 enterprise parameter to true.

**C.** Confirm the phone load name in the phone configuration.

**D.** Assign a universal device template to the phone.

**Answer: (SHOW ANSWER)**

Section: Infrastructure and Design

**NEW QUESTION: 69**

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

**A.** SRST

**B.** SSO

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

**D.** MGCP gateway registration

**Answer: C (LEAVE A REPLY)**

Section: Call Control

**NEW QUESTION: 70**

Which two protocols should be configured for the cisco unity connection and cisco UCM integration?

**A.** MGCP

**B.** H.323

**C.** SIP

**D.** RTP

**E.** SCCP

**Answer: (SHOW ANSWER)**

**NEW QUESTION: 71**

Refer to the exhibit.

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



Which Codec is negotiated?

**A.** 711ulaw

**B.** ILBC

**C.** 728

**D.** 729

**Answer: (SHOW ANSWER)**

**NEW QUESTION: 72**

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

**A.** Wait 24 hours for the garbage collector to remove the user.

**B.** Restart the Dirsync service after the user is deleted from LDAP directory.

**C.** Delete the user directly from Cisco Unified Communications Manager

**D.** Execute a manual sync to refresh the local database and delete the end user.

**Answer: (SHOW ANSWER)**

**NEW QUESTION: 73**

Which action prevents toll fraud in Cisco Unified Communications Manager?

**A.** Implement toll fraud restriction in the Cisco IOS router.

**B.** Implement route patterns in Cisco Unified CM.

**C.** Allow off-net to off-net transfers.

**D.** Configure ad hoc conference restriction.
**Answer: (SHOW ANSWER)**
Explanation/Reference: https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified- communications-manager-express/107626-cme-toll-fraud.html

**NEW QUESTION: 74**
An engineer must manually provision a Cisco IP Phone 8845 using SIP. Which two fields must be configured for a successful provision? (Choose two.)
**A.** CSS
**B.** Device security profile
**C.** SIP profile
**D.** Location
**E.** Media resources group list
**Answer: B,C (LEAVE A REPLY)**

**NEW QUESTION: 75**
An engineer configures Cisco Unified Communications Manager to prevent toll fraud. At which two points does the engineer block the pattern in Cisco Unified CM to complete this task? (Choose two.)
**A.** route pattern
**B.** route group
**C.** translation pattern
**D.** partition
**E.** CSS
**Answer: C,E (LEAVE A REPLY)**
Section: Call Control

**NEW QUESTION: 76**
Which application traffic does the DiffServ AF41 class according to the Cisco Collaboration System Solution Reference Network Design?
**A.** signalling
**B.** video call
**C.** messaging
**D.** audio call
**Answer: (SHOW ANSWER)**

**NEW QUESTION: 77**
Which two conditions must a user meet to provision a new device using the self-provisioning feature? (Choose two.)
**A.** The user must have a primary extension.
**B.** The user must be part of "standard CCM Super User".
**C.** At least DNs must be assigned to the user device.
**D.** The user must have at least one user device profile assigned.
**E.** The user must have the appropriate universal device template linked to the linked to the user profile.
**Answer: (SHOW ANSWER)**

**NEW QUESTION: 78**
A company's employees have been complaining that they have been unable to select options on the internal IVR of the help desk IT support has been given Cisco UCM traces and below is the snippet of the SDP of the INVITE packet.

```
m=audio 25268 RTP/AVP 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=ptime:20
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

How is this issue resolved?
**A.** Configure CODEC for G.729
**B.** Configure CODEC for G 722
**C.** Configure DTMF for RFC 2833.
**D.** Configure DTMF for KPML
**Answer: (SHOW ANSWER)**

**NEW QUESTION: 79**
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: 80**

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

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



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: A ([LEAVE A REPLY]())**

**NEW QUESTION: 81**

How can an engineer determine location-based CAC bandwidth requirements for Cisco Unified Communications Manager?

**A.** Set the requirements in the service parameters.

**B.** Add the requirements for each audio and video codec and multiply how many calls must be supported.

**C.** Execute the Resource Reservation Protocol to return location-based requirements.
**D.** Calculate the number of calls against the license for Cisco Unified Border Element to determine calls per location.
**Answer: A (LEAVE A REPLY)**
Section: QoS
Explanation

**NEW QUESTION: 82**
Which datastore and protocol is used for saving back-up files within the Disaster Recovery System of Cisco UCM?
**A.** remote disk on an NFS share
**B.** local disk on the Cisco UCM server
**C.** remote disk on a CIFS share
**D.** remote disk on the SFTP server
**Answer: D (LEAVE A REPLY)**

**NEW QUESTION: 83**
A company has an excessive number of call transfers to local and long-distance PSTN from Cisco Unity Connection voicemail Which action in the Cisco Unity Connection restriction table resolves this issue?
**A.** Create a custom restriction table ??????????? and block it
**B.** Implement password complexity on voicemail boxes to prevent accounts from being compromised
**C.** Create a custom restriction table *********** and Wock it
**D.** Block PSTN patterns on Default Transfer. Default Outdial. and Default System Transfer
**Answer: D (LEAVE A REPLY)**

**NEW QUESTION: 84**
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._tls.cisco.com pointing to the IP address of Cisco Unified Communications manager
**C.** _xmpp._tls.cisco.com pointing to a record of IM&P
**D.** _cuplogin._tls.cisco.com pointing to the IP address of IM&P
**E.** _cisco-uds._tcp.cisco.com pointing to a record of Cisco Unified CM
**Answer: (SHOW ANSWER)**

**NEW QUESTION: 85**
An administrator with ID392116981 is receiving complaints of pixilation smearing, and pulsing of video calls between two offices that are connected by a WAN. Assuming that QoS is implemented

on the WAN connection, which classification is used to mark the video traffic, according to the Cisco QoS baseline?

**A.** CS3

**B.** AF31

**C.** AF41

**D.** EF

**Answer: C (LEAVE A REPLY)**

**NEW QUESTION: 86**

An engineer is configuring a Cisco Unified Border Element to allow the video endpoints to negotiate without the Cisco Unified Border Element interfering in the process. What should the engineer configure on the Cisco Unified Border Element to support this process?

**A.** Configure a hardcoded codec on the dial peers.

**B.** Configure codec transparent on the dial peers.

**C.** Configure path-thru content sdp on the voice service.

**D.** Configure a transcoder for video protocols.

**Answer: B (LEAVE A REPLY)**

**NEW QUESTION: 87**

Which issue causes slips on a PRI?

**A.** incorrect encapsulation

**B.** incorrectly configured time zone

**C.** change in the line code

**D.** incorrect clock source

**Answer: (SHOW ANSWER)**

**NEW QUESTION: 88**

Refer to the exhibit.

```
INVITE sip:4000@172.16.1.1:5061 SIP/2.0
Via: SIP/2.0/TLS 172.16.2.143:5061;branch=z9hG4bK0FD315E7
Remote-Party-ID: <sip:+14088335000@172.16.2.143>;party=calling;screen=no;privacy=off
From: <sip:+14088335000@172.27.2.143>;tag=7842E5F6-9B8
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: 41[illegible]
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:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
```

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

**A.** ADD a media termination point that supports G.711ulaw and G.729.

**B.** Add a transcoder that supports G.711ualw and G.729.

**C.** Nothing; both sides support payload type.101.

**D.** Noting; both sides support G.729.

**Answer: B (LEAVE A REPLY)**

**NEW QUESTION: 89**

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

**A.** The call egresses in San Francisco because the user uses device mobility and is allowed to roam while still keeping the number and resources assigned in San Francisco

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

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

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

**Answer: (SHOW ANSWER)**

**NEW QUESTION: 90**

A network administrator with ID392116981 has determined that a WAN link between two Cisco UCM clusters supports only 1 Mbps of bandwidth for voice traffic How many calls does this link support if G.711 as the audio codec is used?

**A.** 13

**B.** 15

**C.** 16

**D.** 12

**Answer: (SHOW ANSWER)**

**NEW QUESTION: 91**

Refer to the exhibit.

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

**A.** Add the bearer-cap speech command to the voice port.

**B.** Add the isdn switch-type primart-dms100 command to the serial interface.

**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: 92**

Which action prevent toll fraud in Cisco Unified Communications Manager?

**A.** Implement toll fraud restriction in the Cisco IOS router.

**B.** Allow off-net to off-net transfers.
**C.** Configure ad hoc conference restriction.
**D.** Implement route patterns in Cisco Unified CM.
**Answer: C (LEAVE A REPLY)**

**NEW QUESTION: 93**
An administrator must configure the local route group feature on cisco UCM. Which step will enable this feature?
**A.** For each route list configure a route group to use as a Local Route Group.
**B.** For each device pool. configure a route group to use as a Local Route Group for that device pool
**C.** For each route group, check the box for the Local Route Group feature.
**D.** For each route pattern, select the Local Route Group as the destination.
**Answer: B (LEAVE A REPLY)**

**NEW QUESTION: 94**
An engineer is designing a high availability and failover solution for two Cisco Unified Border Element routers. The first router (cube 1.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 cube1abc.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: 95**
How does traffic policing respond to violations?
**A.** Excess traffic is queued.
**B.** Excess traffic is retransmitted.
**C.** AH traffic is treated equally.
**D.** Excess traffic is dropped.
**Answer: D (LEAVE A REPLY)**

**NEW QUESTION: 96**
An engineer is notified that the Cisco TelePresence MX800 that is registered in Cisco Unified communications Manager shows an empty panel, and the Touch 10 shows a corresponding icon with no action when pressed.
Where does the engineer go to remove the inactive custom panel?
**A.** The software Upgrades page in CUCM OS Administration

**B.** The phone configuration page in CUCM Administration
**C.** The In-Room control Editor on the webpage of the MX800
**D.** The SIP Trunk security profile page in CUCM Administration
**Answer: C (LEAVE A REPLY)**

**NEW QUESTION: 97**
Refer to the exhibit.

```
05:50:14.102: ISDN BR0/1/1 Q921: User TX -> IDREQ ri=216[illegible]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
```

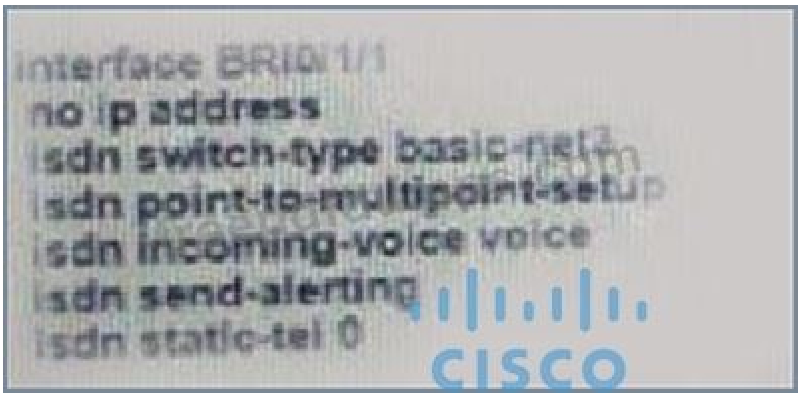
B)

```
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 D
**C.** Option B
**D.** Option C
**Answer: (SHOW ANSWER)**

**NEW QUESTION: 98**
A customer has Cisco Unity Connection 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: (SHOW ANSWER)**

**NEW QUESTION: 99**
What dialed number match this cisco UCM route pattern?
1[23]XX
**A.** 1200 through 1399 only
**B.** 12300 through 12399 only
**C.** 1200 through 1300 only
**D.** 1230 through 1239 only
**Answer: A (LEAVE A REPLY)**

**NEW QUESTION: 100**
A collaboration engineer must configure Cisco Unified Border Element to support up to five concurrent outbound calls across an Ethernet Link with a bandwidth of 160 kb to the Internet Telephony service provider.
Which set of commands allows the engineer to complete the task without compromising voice quality?
A)

```
dial-peer voice 1 voip
 translation-profile outgoing Strip9
 max-conn 5
 destination-pattern 91[2-9]..[2-9]......$
 session protocol sipv2
 session target ipv4:142.45.10.1
 dtmf-relay rtp-nte sip-notify sip-kpml
 codec aacld
```
B)
```
dial-peer voice 1 voip
 translation-profile outgoing Strip9
 max-conn 5
 destination-pattern 91[2-9]..[2-9]......$
 session protocol sipv2
 session target ipv4:142.45.10.1
 dtmf-relay rtp-nte sip-notify sip-kpml
 codec ilbc mode 20
```
C)
```
dial-peer voice 1 voip
 translation-profile outgoing Strip9
 max-conn 5
 destination-pattern 91[2-9]..[2-9]......$
 session protocol sipv2
 session target ipv4:142.45.10.1
 dtmf-relay rtp-nte sip-notify sip-kpml
 codec mp4a-latm
```
D)
```
dial-peer voice 1 voip
 translation-profile outgoing Strip9
 max-conn 5
 destination-pattern 91[2-9]..[2-9]......$
 session protocol sipv2
 session target ipv4:142.45.10.1
 dtmf-relay rtp-nte sip-notify sip-kpml
```
**A.** Option A
**B.** Option D
**C.** Option C
**D.** Option B
**Answer: D (LEAVE A REPLY)**

**NEW QUESTION: 101**
How many DNS SRV entries can be defined in the SIP trunk destination address field in Cisco Unified Communications Manager?
**A.** 1
**B.** 8
**C.** 4
**D.** 16
**Answer: (SHOW ANSWER)**

**NEW QUESTION: 102**
Which statement describes the outcome when the trust boundary is defined at the Cisco IP phone?
**A.** Packets of Ethernet frames are not remarked at the access layer switch.
**B.** Packets or Ethernet frames are remarked at the distribution layer switch.
**C.** Packets or Ethernet frames are not remarked by the IP phone.
**D.** Packets of Ethernet frames are remarked at the access layer switch.
**Answer: A (LEAVE A REPLY)**

**NEW QUESTION: 103**

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

**A.** Configure the Cisco Unified CM's IP in voice service VoIP.

**B.** Upload the custom configuration in the TFTP server in cisco Unified CM.

**C.** Apply the ccm-manager configuration commands to the gateway.

**D.** From Cisco Unified CM >Device > Gateway > Add gateway, check the auto-configuration check box.

**Answer: C (LEAVE A REPLY)**

**NEW QUESTION: 104**

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

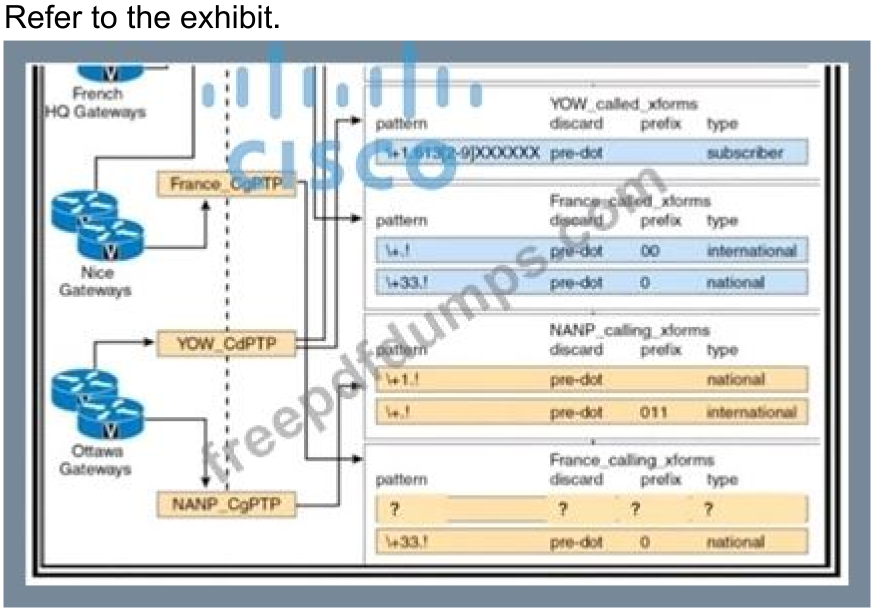
**Answer: (SHOW ANSWER)**

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: 105**

Refer to the exhibit.




A call from +1 613 555 1234 that is sent out through the Nice Gateways should result in a calling party of 001 613 555 1234 with the numbering type "international" Which configuration accomplishes this goal?

**A.** \+.l pre-dot 00 international

**B.** \+ 0011 pre-dot 1 international

**C.** 613XXXXXXX none +011 international
**D.** \+1 ! none pre-dot 001 international
**Answer: D (LEAVE A REPLY)**

**NEW QUESTION: 106**
An engineer configures Cisco Unified Communications Manager to prevent toll fraud. At which two point does the engineer block the pattern in Cisco Unified CM to complete this task? (Choose two.)
**A.** partition
**B.** route group
**C.** translation pattern
**D.** CSS
**E.** route pattern
**Answer: A,D (LEAVE A REPLY)**



**NEW QUESTION: 107**
Due to service provider restriction, Cisco Unified Communications Manager cannot send video in the SDP.
Which two options on Cisco Unified CM are configured to suppress video in the SDP is outgoing invites?
(Choose two.)
**A.** Set Video bandwidth in the Region settings to 0.
**B.** Change the Video Capabilities dropdown on the endpoint to Disabled.
**C.** Add the audio forced command to voice service voip on the Unified Border Elements.
**D.** Check the Retry Call as Audio on the SIP trunk
**E.** Check the Send send-receive SDP in mid-call INVITE check box on the SIP trunk SIP profile.
**Answer: (SHOW ANSWER)**

**NEW QUESTION: 108**
Refer to the exhibit.

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

Endpoint B:
m=audio 21796 RTP/AVP 105 0 8 18 101
 b=TIAS:64000
 a=extmap:14 http://protocols.cisco.com/timestamp#100us
 a=rtpmap:105 G7221/16000
 a=fmtp:105 bitrate=24000
 a=rtpmap:0 PCMU/8000
 a=rtpmap:8 PCMA/8000
 a=rtpmap:18 G729/8000
 a=fmtp:18 annexb=no
 a=rtpmap:101 telephone-event/8000
 a=fmtp:101 0-15
 a=trafficclass:conversational.audio.immersive.aq:admitted
```

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

**A.** PCMA/8000

**B.** G7221/16000

**C.** G277/8000

**D.** Telephone-event/8000

**Answer: C (LEAVE A REPLY)**