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**Exam** : **300-815**

**Title** : Implementing Cisco Advanced  
Call Control and Mobility  
Services

**Vendor** : Cisco

**Version** : DEMO

**NO.1** Which elements does Cisco cloud mobility for collaboration include?

- A. Cisco Webex Collaboration Cloud Services
- B. Cisco Collaboration Cloud
- C. Cisco Collaboration Cloud and Cisco Webex Collaboration Cloud Services
- D. Cisco Collaboration Cloud and Cisco Mobile and Remote Access Collaboration Cloud Services

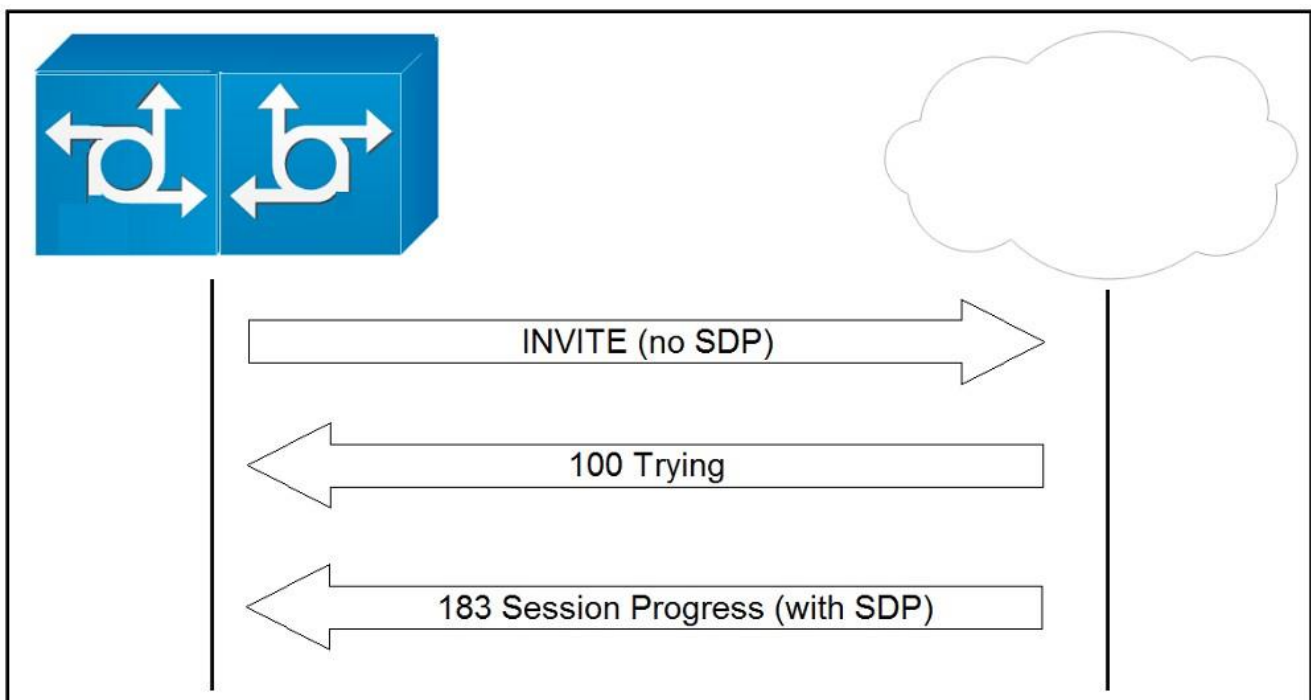
**Answer:** D

**NO.2** An engineer is troubleshooting an intersite call between two endpoints where the call fails and the message "Not Enough Bandwidth" is displayed. G.729 codec is in use on both sites. First, calls are being properly routed, and the issue happens after the third call is established and the bandwidth utilization between the two sites is under 50%. Which configuration in Cisco UCM must be adjusted to resolve the issue?

- A. transcoder
- B. location
- C. route pattern
- D. translation pattern

**Answer:** B

**NO.3** Refer to the exhibit.



An administrator is troubleshooting why users are not hearing audio when dialing long distance numbers across their Cisco Unified Border Element. The customer's carrier has a requirement that dialing long distance requires an access code to be entered. Looking at the exhibit, what two actions can be taken to correct signaling? (Choose two.)

- A. Enable PRACK.
- B. Enable Early Offer on the Cisco Unified Border Element.
- C. Enable the supplementary-service media-renegotiate command.
- D. Enable Media Flow Around

E. Enable Mid-Call Signaling Consumption.

**Answer:** A,B

**NO.4** An administrator is configuring a new deployment using Cisco Unified CME. The SCCP phones register without any issues, but SIP phones are not registering. Assume that all other configuration is valid. Which code allows SIP phones to register to Cisco UCME?

A. voice service voip

allow-connections sip to h323

B. voice service voip

sip

bind media source-interface Vlan100

C. voice service voip

sip

bind control source-interface Vlan100

D. voice service voip

sip

registrar server expires max 600 min 60

**Answer:** C

**NO.5** A company is experiencing an issue where calls from the Cisco UCM via Cisco Unified Border Element to the ITSP fail with a "500 server internal error" right after a 183 session in progress is sent back from the ITSP. What must be enabled to resolve the issue?

A. delayed offer on Cisco UCM

B. early offer at the CUBE

C. delayed offer on CUBE

D. early offer at the ITSP

**Answer:** B

**NO.6** Refer to the exhibit.

```

!
dial-peer voice 10 voip
description Inbound
session protocol sipv2
incoming called-number 2000
dtmf-relay rtp-nte
no vad
!
dial-peer voice 20 voip
description Outbound
destination-pattern 2.
session protocol sipv2
session target ipv4:192.168.100.101
voice-class sip options-keepalive
dtmf-relay rtp-nte
!

CUBE#show dial-peer voice summary
dial-peer hunt 0

```

TAG	TYPE	MIN	OPER	PREFIX	DEST-PATTERN	PRE FER	PASS THRU	SESS-THRU	SER-TARGET	GRP\ STAT	OUT PORT	KEEPALIVE	VRF
10	voip	up	up			0	syst					busyout	NA
20	voip	up	up		2.	0	syst	ipv4:192.168.100.101				busyout	NA

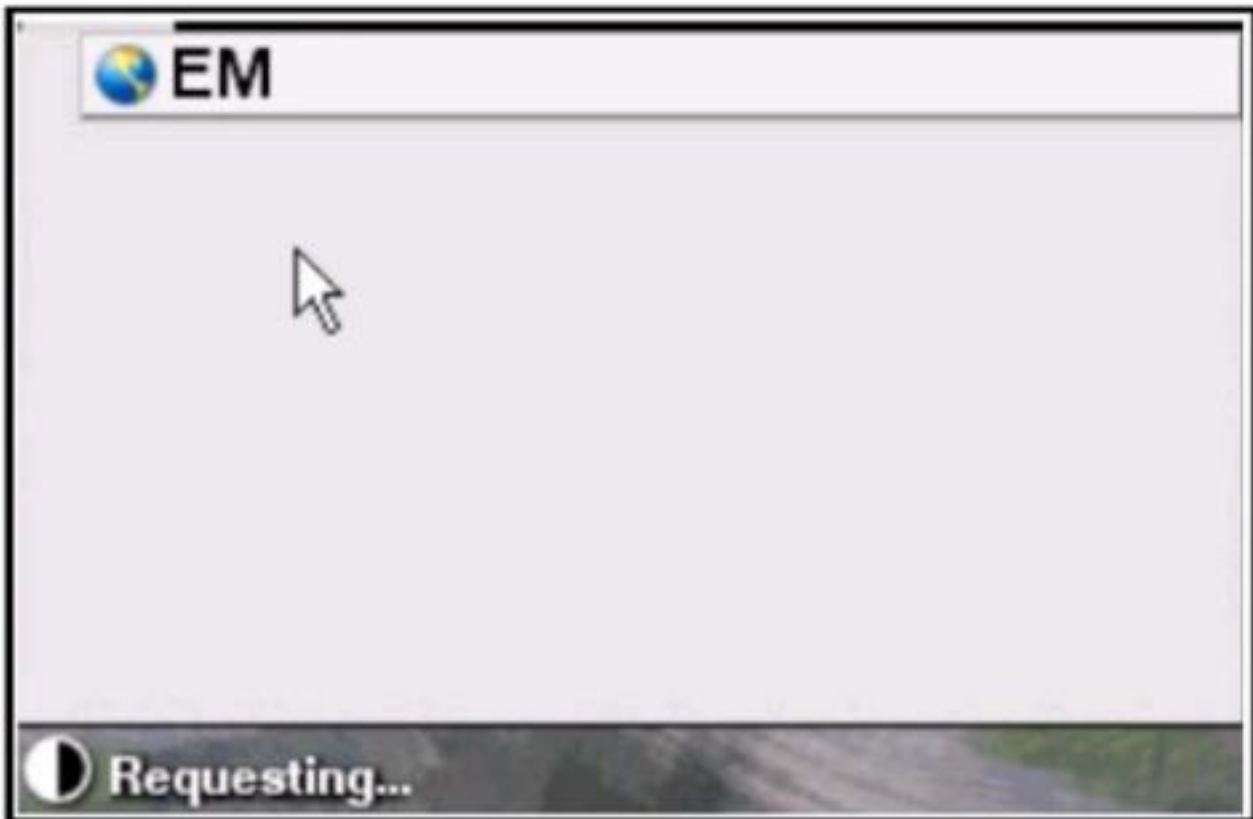
A call made through the Cisco Unified Border Element to pilot 2000 is failing. What is causing the call

to foil?

- A. No codecs are configured on the dial peers
- B. The Cisco Unified Border Element is not receiving a response to its OPTION keepalives.
- C. The destination pattern is incorrect for the dialed number.
- D. VAD was not disabled on the outgoing dial peer.

**Answer:** C

**NO.7** Refer to me exhibit.



A user cannot access Extension Mobility even after clicking the services button and selecting EM service. There is nowhere to place the username and password to sign in for this service. What is the cause of the issue?

- A. IP phone services Extension Mobility does not have the correct service URL configured.
- B. IP phone services Extension Mobility has XML Service Category configured.
- C. The Cisco IP phone has not checked the Extension Mobility checkbox.
- D. The Cisco IP phone is not subscribed to an Extension Mobility service.

**Answer:** A

**NO.8** A company has users that are logged in to hunt groups. However, there is a requirement for hunt group configurations to provide an option to turn on audible ringtones when calls to a line group arrive at a phone that is logged out and on a break. This ringtone alerts a logged-out user that there is an incoming call to a hunt list to which the line is a member, but the call does not ring at the phone of that line group member because of the logged-out status. Which action meets this requirement?

- A. Configure the HLog softkey on the phone so that while a user is logged off, it plays an audible tone

when a call is missed.

**B.** Set the service parameter Party Entrance Tone to True."

**C.** Configure the service parameter hunt group logoff notification and specify the name of the ringtone file.

**D.** Set the service parameter Enterprise Feature Access number for hunt group logout and set up an access number

**Answer:** C

**NO.9** An administrator configured Extension Mobility on Cisco UCM. Users report that logins are successful, but the phones do not have an Extension Mobility option after logging in. The administrator verified that Extension Mobility is enabled on the devices and that the log-out profile is valid. Which action must the administrator take to resolve the issue?

**A.** Subscribe all device profiles to the Extension Mobility phone service.

**B.** Delete the identity trust list file from the phone(s).

**C.** Change the Extension Mobility URL from HHTP to HTTPS.

**D.** Restart the Cisco Extension Mobility and Cisco Extension Mobility application services.

**Answer:** A

**NO.10** After configuring a Cisco CallManager Express with Cisco Unity Express, inbound calls from the PSTN SIP trunk receive a ring tone for 20 seconds and then a busy signal instead of voicemail. Which configuration fixes this problem?

**A.** Router(config)# voice service voip

Router(conf-voi-serv)#allow-connections h323 to h323

**B.** Router(config)#dial-peer voice 2 voip

Router(config-dial-peer)#no vad

**C.** Router(config)# voice service voip

Router(conf-voi-serv)#allow-connections voice-mail mod

**D.** Router(config)# voice service voip

Router(conf-voi-serv)#no supplementary-service sip moved-temporarily

**Answer:** D

**NO.11** An engineer configures several calling party transformation patterns in Cisco UCM so that calling numbers are shown in the +E.164 format. The route patterns match seven-digit internal numbers and prefix the numbers with 1408. The SIP trunk matches 1408XXXXXX and prefixes the number with +. When users make outbound calls, the calling number still shows as seven digits. What is the cause of this problem?

**A.** The calling transformation on the SIP trunk is applied first, and then the calling transformation on the route patterns is applied.

**B.** Calling transformations cannot be configured on both route patterns and a SIP trunk.

**C.** The calling transformation on the route patterns is ignored because there is already a calling transformation on the SIP trunk.

**D.** The engineer must also apply a calling transformation at the route group details level.

**Answer:** D

**NO.12** Which configuration element of a hunt group allows for changing Calling Party Transformations settings?

- A. line group
- B. hunt pilot
- C. route group
- D. hunt list

**Answer:** B

**NO.13** Calls to a particular extension are not routing to voicemail. The user reaches the voicemail system from the handset by pressing the Messages button Which configuration parameter causes this problem?

- A. The voicemail pilot number for call forwarding is missing from the ephone-dn
- B. The voicemail pilot number is missing from the call handling on Cisco Unity Express
- C. The voicemail pilot number is missing from the telephony service configuration on Cisco UCME
- D. The voicemail pilot number for call forwarding is missing from the ephone

**Answer:** A

**NO.14** Refer to the exhibit.

```
!  
  
dial-peer voice 1 voip  
description to ITSP  
destination-pattern 555.....  
session target ipv4:209.110.110.1  
incoming called-number .  
codec g711ulaw  
  
!
```

An engineer configures Cisco Unified Border Element to connect the enterprise VoIP network with a SIP telephony provider. Calls are not working in either direction. What must be configured in the dial peer 1 to fix the issue?

- A. answer-address 555 .....
- B. codec g729
- C. session-protocol sipv2
- D. incoming called number 555.....

**Answer:** C

**NO.15** What are two configuration features of the Client matter code setting in the route pattern configuration? (Choose two.)

- A. The client Matter Code feature supports overlap sending since the Cisco UCM can determine

when to prompt the user for the code.

**B.** Selecting the Allow Overlap Sending setting disables the Require Client Matter Code setting.

**C.** The Client Matter Code feature provides the option to configure Authorization Level such as in the Forced Authorization Code.

**D.** Selecting the Allow Overlap Sending setting allows a user to select the Require Client Matter Code setting.

**Answer:** B,C

**NO.16** Which two configuration parameters are prerequisites to set Native Call Queuing on Cisco Unified Communications Manager? (Choose two.)

**A.** Cisco IP Voice Media Streaming Service must be activated on at least one node in the cluster.

**B.** A unicast music on hold audio source must be configured.

**C.** Cisco RIS data collector service must be running on the same server as the Cisco CallManager service.

**D.** The maximum number of callers allowed in queue must be 10.

**E.** The phone button template must have the Queue Status Softkey configured.

**Answer:** A,C

**NO.17** Refer to the exhibit.

```
voice translation-profile incoming
  translate called 999
!
voice translation-rule 999
  rule 1/\ (^[1-2] [1-2] [1-2]\ ) 333\ ([4-5] [4-5] .\ ) $ / / \2333\1/
!
dial-peer voice 999 voip
  translation-profile outgoing incoming
  session protocol sipv2
  incoming called-number
  dtmf-relay rtp-nte
  codec transparent
  destination dpg 888
  no vad
!
voice class dpg 888
  dial-peer 888
!
dial-peer voice 888 voip
  destination-pattern 888
  session protocol sipv2
  session target ipv4:192.168.0.1
  codec transparent
  dtmf-relay rtp-nte
  no vad
```

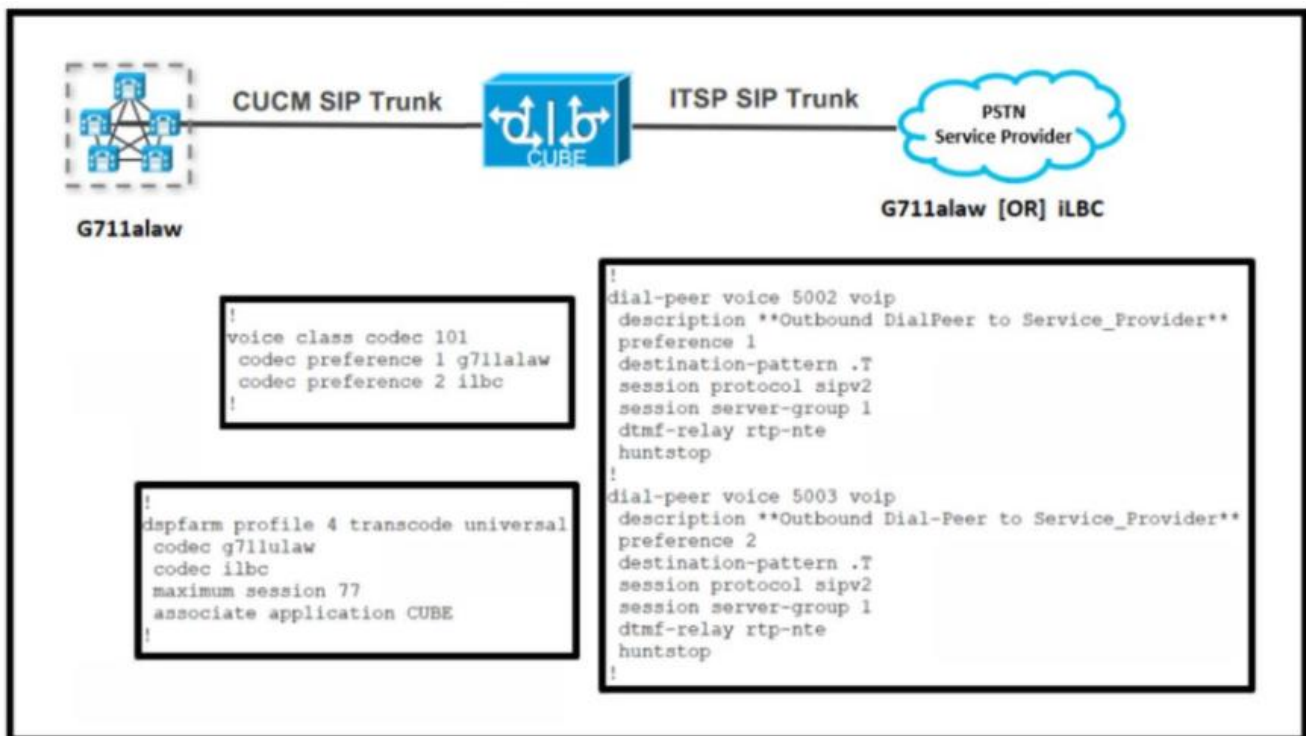
Calls incoming from the provider are not working through newly set up Cisco Unified Border Element. Provider engineers get the 404 Not Found SIP message. Incoming calls are coming from the provider with called number "222333444" and Cisco Unified Communications Manager is expecting the called number to be delivered as "444333222". The administrator already verified that the IP address of the

Cisco Unified CM is set up correctly and there are no dial peers configured other than those shown in the exhibit. Which action must the administrator take to fix the issue?

- A. Change the destination-pattern on the outgoing dial peer to match "444333222".
- B. Set up translation-profile on the incoming dial peer to match incoming traffic.
- C. Create specific matching for "222333444" on the incoming dial peer.
- D. Fix the voice translation-rule to match specifically number "222333444" and change it to "444333222".

**Answer:** B

**NO.18** Refer to the exhibit.



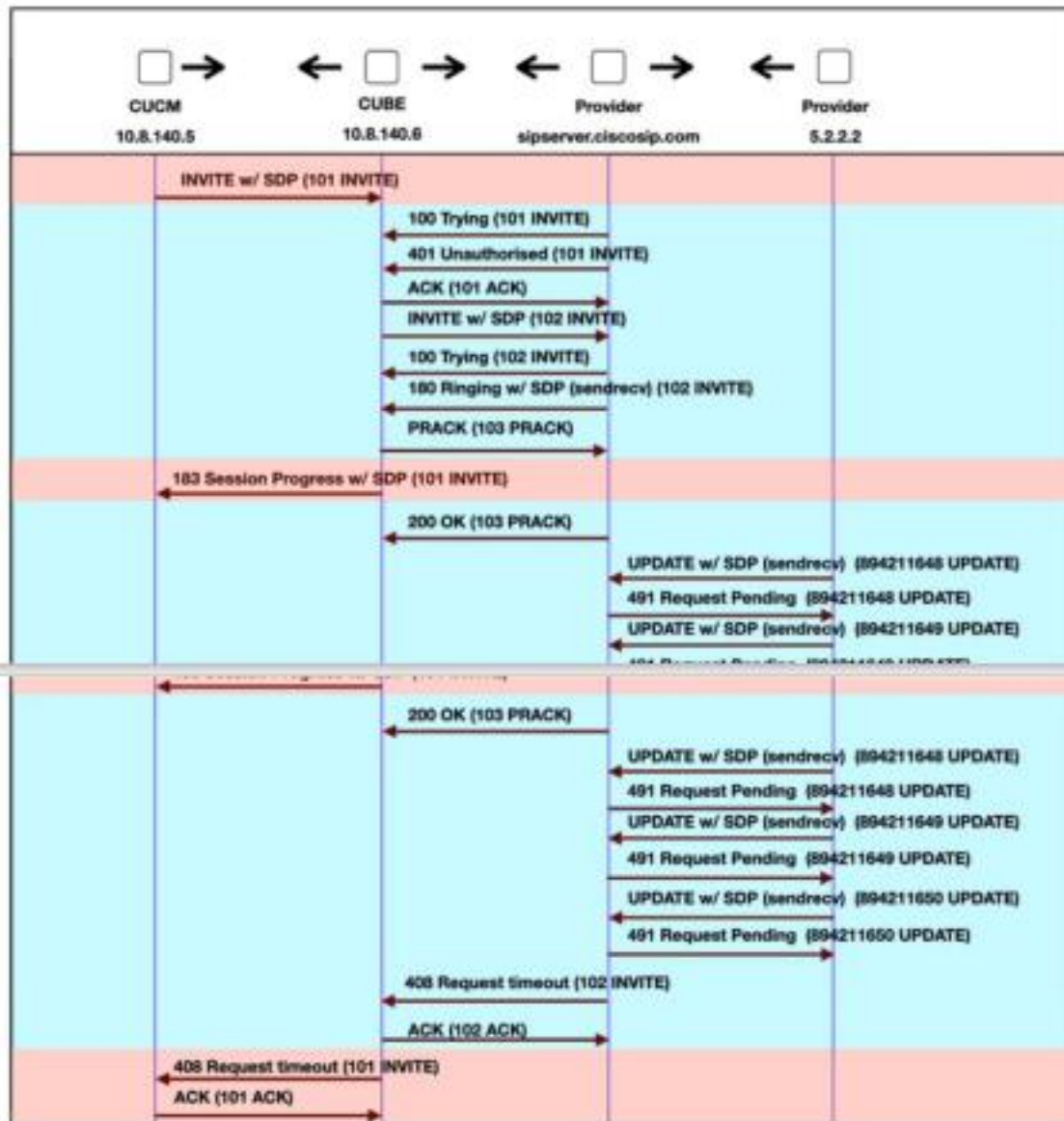
Outbound calls to the service provider cause intermittent errors due to a codec mismatch. The internal network sends early offer SDP that contains only G.711 A-law. The service provider reports that some destinations support only G.711 A-law while others support only iLBC. The service provider also allows only 20 active calls at a time. Which configuration allows successful media negotiation for all calls using outbound dial peers 5002 and 5003?

- dial-peer voice 5002 voip  
codec g711alaw ilbc  
!  
dial-peer voice 5003 voip  
codec g711alaw ilbc
  - dial-peer voice 5002 voip  
voice-class codec 101 offer-all  
!  
dial-peer voice 5003 voip  
voice-class codec 101 offer-all
  - dial-peer voice 5002 voip  
codec g711alaw  
!  
dial-peer voice 5003 voip  
codec ilbc
  - dial-peer voice 5002 voip  
voice-class codec 101  
!  
dial-peer voice 5003 voip  
voice-class codec 101
- 

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

**Answer:** D

**NO.19** Refer to the exhibit.



A Cisco Unified Border Element continues to send 180/183 with the required: 100rel header to Cisco UCM. and the call eventually disconnects How is the issue resolved?

- A. Enable "SIP Rel1XX Options\* and -Early Offer Support" on the SIP Profile Configuration Page in Cisco UCM.
- B. Enable \*Early Offer support for voice and video calls" on the SIP Profile Configuration Page in Cisco UCM.
- C. Disable "SIP Rel1XX Options\* and 'Early Offer Support\*" on the SIP Profile Configuration Page in Cisco UCM.
- D. Disable "Send send-receive SDP in mid-call INVITE\*" on the SIP Profile Configuration Page in Cisco UCM.

**Answer:** B

**NO.20** Due to a shortage of physical interfaces on a device the administrator requires that a loopback for RTP is used. Which command is required when using a loopback interface for RTP?

- A. voice-class sip resources priority mode passthrough
- B. voice-class sip bind control source-interface Loopback0

- C. voice-class sip early-offer forced.
- D. voice-class sip bind media source-interface Loopback0

**Answer:** D

**NO.21** Where is the dtmf-relay command configured on Cisco Unified Border Element?

- A. in the voice-class VoIP configuration
- B. in the VoIP dial peer
- C. in global SIP configuration
- D. in the VoIP or POTS dial peers

**Answer:** B

**NO.22** An engineer is troubleshooting local ringback on a Cisco SIP gateway. The gateway is not ignoring the SIP 180 response with SDP from the service provider, and the far end device is sending the 180 with SDP to play ringback from the IP address specified in the SDP. Which configuration change must be made on the gateway to resolve the issue?

- A. Router(conf-voi-serv)# disable-early-media 180
- B. Router(conf-tg-sip-ua)# disable-early-media 180
- C. Router(conf-voi-serv)# no disable-early-media 180
- D. Router(conf-tg-sip-ua)# no disable-early-media 180

**Answer:** B

**NO.23** Customers that call into a company's IVR report that when they try to select an option, none of the prompts work. The administrator determines that the calls are coming in across an H.323 gateway. While analyzing the dial peer that points toward Cisco UCM, the administrator notices that no DTMF method is configured. Which command resolves this issue?

- A. dial-peer voice 2 voip  
dtmf-relay sip-kpml
- B. dial-peer voice 2 voip  
dtmf-relay h245-alphanumeric
- C. dial-peer voice 2 voip  
dtmf-relay sip-notify
- D. dial-peer voice 2 pots  
dtmf-relay h245-alphanumeric

**Answer:** B

**NO.24** What is the relationship between partition, time schedule, and time period in Time-of-Day routing in Cisco Unified Communications Manager?

- A. A partition can have multiple time schedules assigned. A time schedule contains one or more time periods.
- B. A partition can have one time schedule assigned. A time schedule contains one or more time periods.
- C. A partition can have multiple time schedules assigned. A time schedule contains only one time period.
- D. A partition can have one time schedule assigned. A time schedule contains only one time period.

***Answer:*** A