642-427 Q&As

Troubleshooting Cisco Unified Communications v8.0 (TVOICE v8.0)

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QUESTION 1

You have just obtained a list of the following options:

All Patterns Unassigned DN Call Park Conference Directory Number Translation Pattern Call Pickup Group Route Pattern Message Waiting Voice Mail Attendant Console

What have you selected in order to produce this list?

A. Control Center > Feature Services

B. Dialed Number Analyzer

C. Route Plan > Route Plan Report

D. Route Plan > External Route Plan Wizard

Correct Answer: C

QUESTION 2

You have received a trouble ticket stating that the MWI light is not coming on for a group of users. Further investigation reveals that the affected users are connected to the same subscriber in the cluster. Users that are connected to other subscribers in the same cluster are not experiencing this issue.

What is causing this problem?

A. The MWI numbers for this subscriber have been changed.

B. The Cisco Unity voice-mail services need to be restarted for the subscriber so all the users will receive proper MWI message indication.

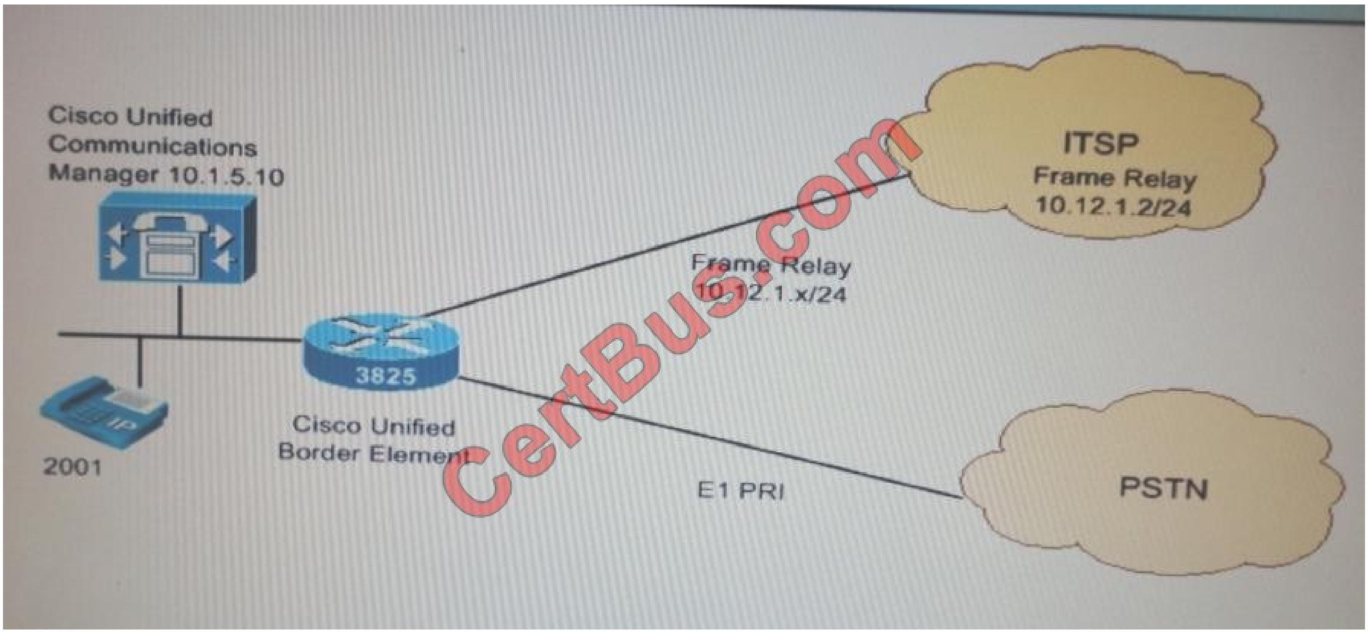
C. Database replication has an error or has failed.

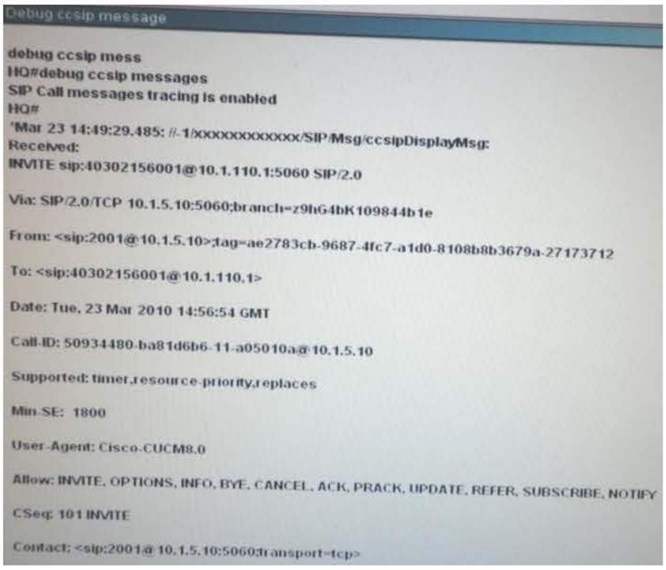
D. The voice-mail ports assigned to the subscriber are down.

Correct Answer: C

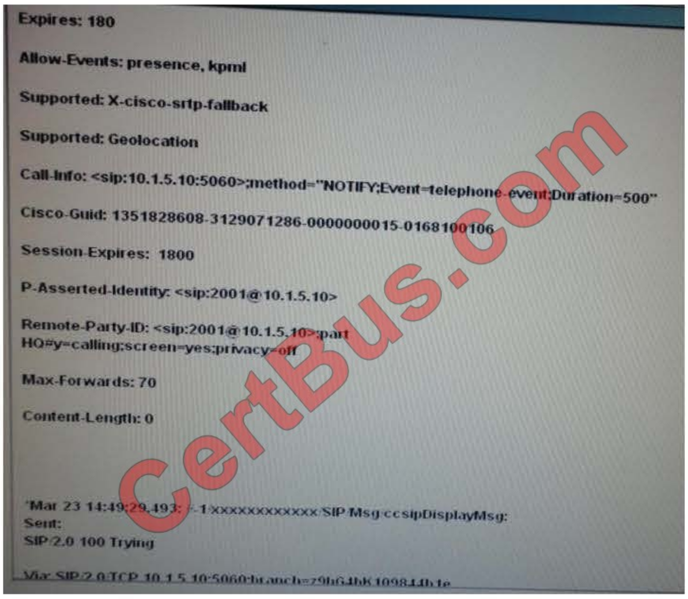
QUESTION 3

Refer to the exhibits.

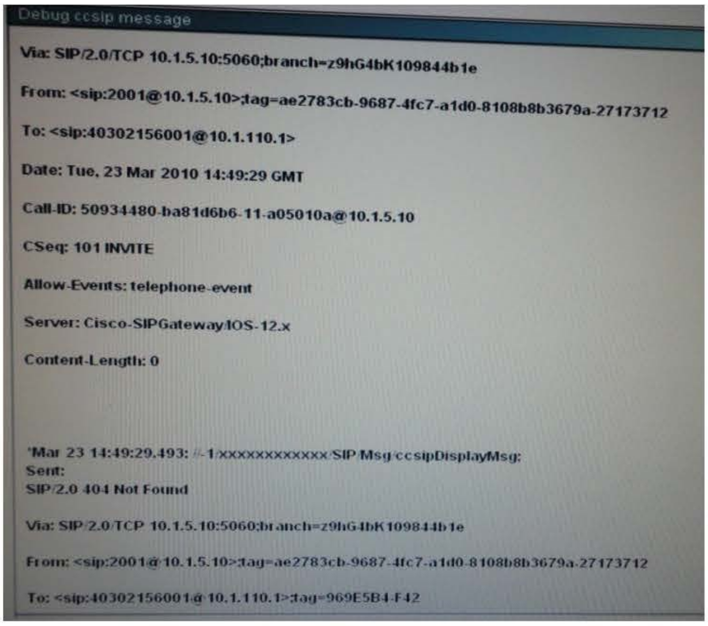

Cisco Unified
Communications
Manager 10.1.5.10
2001
3825
Cisco Unified
Border Element
Frame Relay
10.12.1.x/24
ITSP
Frame Relay
10.12.1.2/24
E1 PRI
PSTN



```
Debug ccsip message
debug ccsip mess
HQ#debug ccsip messages
SIP Call messages tracing is enabled
HQ#
*Mar 23 14:49:29.485: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
INVITE sip:40302156001@10.1.110.1:5060 SIP/2.0

Via: SIP/2.0/TCP 10.1.5.10:5060;branch=z9hG4bK109844b1e

From: <sip:2001@10.1.5.10>;tag=ae2783cb-9687-4fc7-a1d0-8108b8b3679a-27173712

To: <sip:40302156001@10.1.110.1>

Date: Tue, 23 Mar 2010 14:56:54 GMT

Call-ID: 50934480-ba81d6b6-11-a05010a@10.1.5.10

Supported: timer,resource-priority,replaces

Min-SE: 1800

User-Agent: Cisco-CUCM8.0

Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY

CSeq: 101 INVITE

Contact: <sip:2001@10.1.5.10:5060;transport=tcp>
```



```
Expires: 180

Allow-Events: presence, kpml

Supported: X-cisco-srtp-fallback

Supported: Geolocation

Call-Info: <sip:10.1.5.10:5060>;method="NOTIFY;Event=telephone-event;Duration=500"

Cisco-Guid: 1351828608-3129071286-0000000015-0168100106

Session-Expires: 1800

P-Asserted-Identity: <sip:2001@10.1.5.10>

Remote-Party-ID: <sip:2001@10.1.5.10>;part
HQ#y=calling;screen=yes;privacy=off

Max-Forwards: 70

Content-Length: 0


*Mar 23 14:49:29.493: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 100 Trying

Via: SIP/2.0/TCP 10.1.5.10:5060;branch=z9hG4bK109844b1e
```

```
Debug ccsip message
Via: SIP/2.0/TCP 10.1.5.10:5060;branch=z9hG4bK109844b1e

From: <sip:2001@10.1.5.10>;tag=ae2783cb-9687-4fc7-a1d0-8108b8b3679a-27173712

To: <sip:40302156001@10.1.110.1>

Date: Tue, 23 Mar 2010 14:49:29 GMT

Call-ID: 50934480-ba81d6b6-11-a05010a@10.1.5.10

CSeq: 101 INVITE

Allow-Events: telephone-event

Server: Cisco-SIPGateway/IOS-12.x

Content-Length: 0


*Mar 23 14:49:29.493: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 404 Not Found

Via: SIP/2.0/TCP 10.1.5.10:5060;branch=z9hG4bK109844b1e

From: <sip:2001@10.1.5.10>;tag=ae2783cb-9687-4fc7-a1d0-8108b8b3679a-27173712

To: <sip:40302156001@10.1.110.1>;tag=969E5B4-F42
```

```
Call-ID: 50934480-ba81d6b6-11-a05010a@10.1.5.10

CSeq: 101 INVITE

Allow-Events: telephone-event

Server: Cisco-SIPGateway/IOS-12.x

Reason: Q.850;cause=3

Content-Length: 0


*Mar 23 14:49:29.493: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
ACK sip:40302156001@10.1.110.1:5060 SIP/2.0

Via: SIP/2.0/TCP 10.1.5.10:5060;branch=z9hG4bK109844b1e

From: <sip:2001@10.1.5.10>;tag=ae2783cb-9687-4fc7-a1d0-8108b8b3679a-27173712

To: <sip:40302156001@10.1.110.1>;tag=969E5B4-F42

Date: Tue, 23 Mar 2010 14:56:54 GMT

Call-ID: 50934480-ba81d6b6-11-a05010a@10.1.5.10

Max-Forwards: 70
```

```
CUBE Config
!
interface Loopback0
 ip address 10.1.111.1 255.255.255.0
!
interface GigabitEthernet0/0
 no ip address
 ip pim sparse-dense-mode
 duplex auto
 speed auto
 media-type rj45
!
interface GigabitEthernet0/0.5
 encapsulation dot1Q 5
 ip address 10.1.5.1 255.255.255.0
 ip pim sparse-dense-mode
!
interface GigabitEthernet0/0.10
 encapsulation dot1Q 10
 ip address 10.1.10.1 255.255.255.0
 ip pim sparse-dense-mode
!
interface GigabitEthernet0/0.110
 encapsulation dot1Q 110
 ip address 10.1.110.1 255.255.255.0
 ip helper-address 10.1.5.10
 ip pim sparse-dense-mode
!
interface GigabitEthernet0/1
 ip address 10.140.1.2 255.255.255.0
 duplex auto
 speed auto
```



```
CUBE Config
!
interface Serial0/0/0:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-net5
 isdn incoming-voice voice
 isdn bind-l3 ccm-manager
 no cdp enable
!
interface Serial0/1/0
 no ip address
 ip pim sparse-dense-mode
 encapsulation frame-relay IETF
!
interface Serial0/1/0.101 point-to-point
 ip address 10.12.1.1 255.255.255.0
 ip pim sparse-dense-mode
 snmp trap link-status
 frame-relay interface-dlci 101
!
interface Serial0/1/0.102 point-to-point
 ip address 10.13.1.1 255.255.255.0
 snmp trap link-status
 frame-relay interface-dlci 102
!
interface Serial0/1/1
 no ip address
 shutdown
 clock rate 2000000
!
!
```

CUBE Config

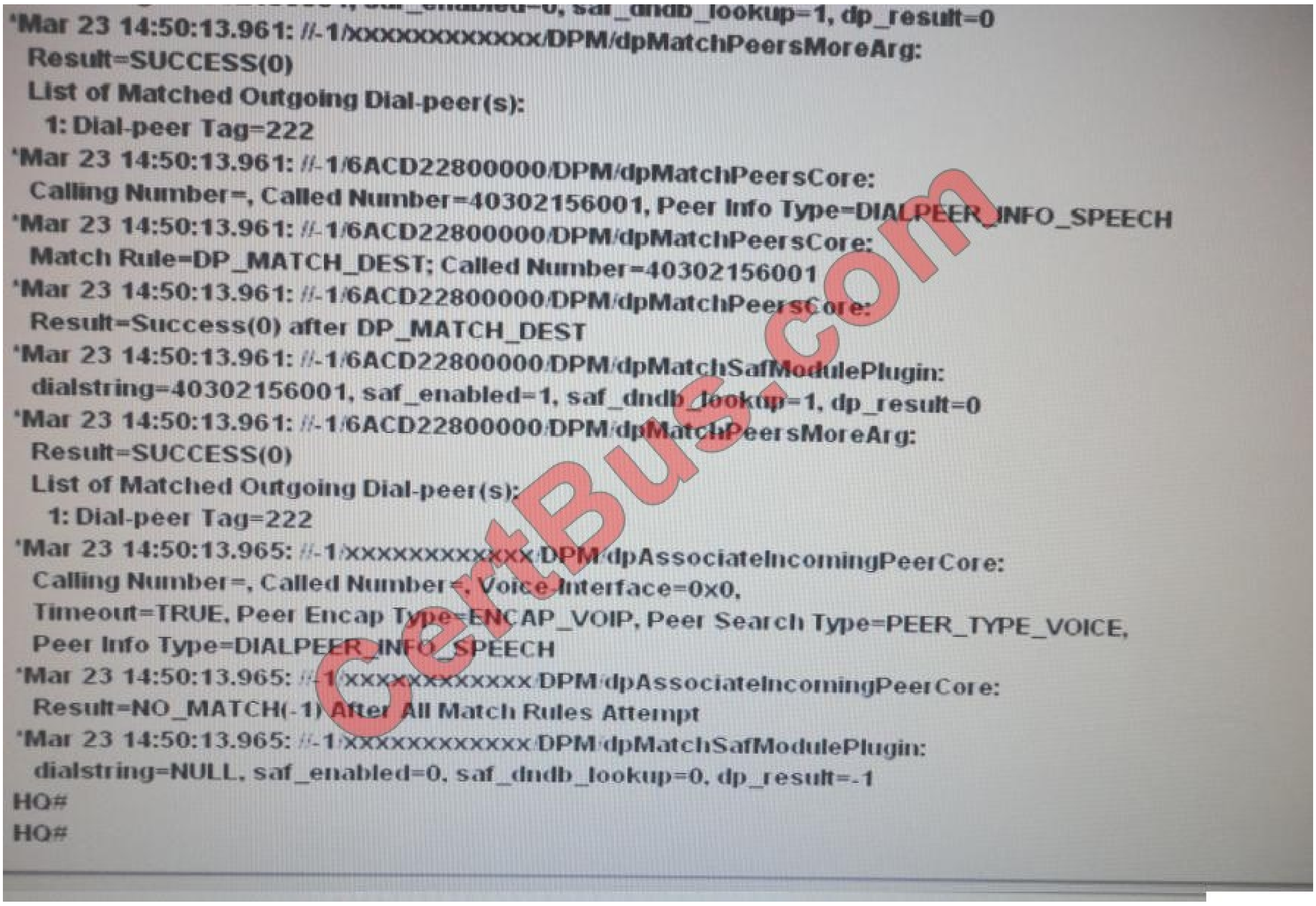
```
router eigrp 10
network 10.0.0.0
!
!
!
!
control-plane
!
!
voice-port 0/0/0:15
!
ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
ccm-manager config server 10.1.5.10
!
mgcp
mgcp call-agent 10.1.5.10 service-type mgcp version 0.1
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp package-capability pre-package
no mgcp package-capability res-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 ecm
mgcp rtp payload-type g726r16 static
mgcp behavior g729-variants static-pt
!
mgcp profile default
```

CUBE Config

```
!
dial-peer voice 1111 voip
 session protocol sipv2
 incoming called-number .
!
dial-peer voice 222 voip
 destination-pattern 40.........
 session target ipv4:10.12.1.2
!
!
gateway
 timer receive-rtp 1200
!
!
gatekeeper
 shutdown
!
!
line con 0
 exec-timeout 0 0
 logging synchronous
line aux 0
line vty 0 4
 exec-timeout 0 0
 password cisco123
 login
!
scheduler allocate 20000 1000
end

HQ#
```


Debug voice dial

```
 Result=SUCCESS(0)
 List of Matched Outgoing Dial-peer(s):
  1: Dial-peer Tag=222
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpMatchPeersCore:
 Calling Number=40302156001, Called Number=40302156001, Peer Info Type=DIALPEER_INFO_SPEECH
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpMatchPeersCore:
 Match Rule=DP_MATCH_DEST; Called Number=40302156001
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpMatchPeersCore:
 Result=Success(0) after DP_MATCH_DEST
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpMatchSafModulePlugin:
 dialstring=40302156001, saf_enabled=0, saf_dndb_lookup=1, dp_result=0
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpMatchPeersMoreArg:
 Result=SUCCESS(0)
 List of Matched Outgoing Dial-peer(s):
  1: Dial-peer Tag=222
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpAssociateIncomingPeerCore:
 Calling Number=40302156001, Called Number=, Voice-Interface=0x0,
 Timeout=TRUE, Peer Encap Type=ENCAP_VOIP, Peer Search Type=PEER_TYPE_VOICE,
 Peer Info Type=DIALPEER_INFO_SPEECH
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpAssociateIncomingPeerCore:
 Result=Success(0) after DP_MATCH_ORIGINATE; Incoming Dial-peer=222
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpMatchSafModulePlugin:
 dialstring=NULL, saf_enabled=0, saf_dndb_lookup=0, dp_result=0
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpAssociateIncomingPeerCore:
 Calling Number=40302156001, Called Number=, Voice-Interface=0x0,
 Timeout=TRUE, Peer Encap Type=ENCAP_VOIP, Peer Search Type=PEER_TYPE_VOICE,
 Peer Info Type=DIALPEER_INFO_SPEECH
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpAssociateIncomingPeerCore:
 Result=Success(0) after DP_MATCH_ORIGINATE; Incoming Dial-peer=222
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpMatchSafModulePlugin:
 dialstring=NULL, saf_enabled=0, saf_dndb_lookup=0, dp_result=0
```



```
saf_dndb_lookup=1, dp_result=0
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpMatchPeersMoreArg:
  Result=SUCCESS(0)
  List of Matched Outgoing Dial-peer(s):
   1: Dial-peer Tag=222
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchPeersCore:
  Calling Number=, Called Number=40302156001, Peer Info Type=DIALPEER_INFO_SPEECH
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchPeersCore:
  Match Rule=DP_MATCH_DEST; Called Number=40302156001
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchPeersCore:
  Result=Success(0) after DP_MATCH_DEST
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchSafModulePlugin:
  dialstring=40302156001, saf_enabled=1, saf_dndb_lookup=1, dp_result=0
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchPeersMoreArg:
  Result=SUCCESS(0)
  List of Matched Outgoing Dial-peer(s):
   1: Dial-peer Tag=222
*Mar 23 14:50:13.965: //-1/xxxxxxxxxxxx/DPM/dpAssociateIncomingPeerCore:
  Calling Number=, Called Number=, Voice-Interface=0x0,
  Timeout=TRUE, Peer Encap Type=ENCAP_VOIP, Peer Search Type=PEER_TYPE_VOICE,
  Peer Info Type=DIALPEER_INFO_SPEECH
*Mar 23 14:50:13.965: //-1/xxxxxxxxxxxx/DPM/dpAssociateIncomingPeerCore:
  Result=NO_MATCH(-1) After All Match Rules Attempt
*Mar 23 14:50:13.965: //-1/xxxxxxxxxxxx/DPM/dpMatchSafModulePlugin:
  dialstring=NULL, saf_enabled=0, saf_dndb_lookup=0, dp_result=-1
HQ#
HQ#
```

When the IP phone 2001 places a call to 9011 49403021 56001, the call is sent to the Cisco Unified Border Element as 40302156001 which is what the ITSP expects to receive. The ITSP support personnel claim that they never saw the call. Issuing the debug CCSIP message command on the Cisco Unified Border Element results in the message "SIP/2 0 404 Not Found".

Refer to the Cisco Unified Border Element configuration, debug voice dial and ccsip messages exhibits. Which situation can cause this issued?

A. The Cisco Unified Bolder Element is configured as an MGCP gateway also so that the call is attempted via the PSTN

B. The command allow-connections sip to h323 is missing

C. SIP error 404 means that a codec mismatch occurred Cisco Unified Communications Manager is sending the call as an early offer with G.711 codec.

D. The Cisco Unified Communications Manager is rnisconfigured. The SIP invite should be sent to the ITSP at 10.1.2.1.2. The debug ccsip message shows the SIP invite being sent to 10.12.1.2.

Correct Answer: B

QUESTION 4

You have received a trouble ticket stating that an IP phone is not working. When asked, the user informs you the phone is displaying the message "Registration rejected." Which two issues are possible causes of this problem? (Choose two.)

A. The IP phone is not getting an IP address.

B. The IP phone\\'s primary Cisco Unified CallManager has a database replication issue.

C. The primary Cisco Unified CallManager is unavailable and the CallManager group assigned to the IP phone does not include a secondary CallManager.

D. The IP phone has not been defined in Cisco Unified CallManager.

E. The IP phone is not associated with a valid user profile.

Correct Answer: BD

QUESTION 5

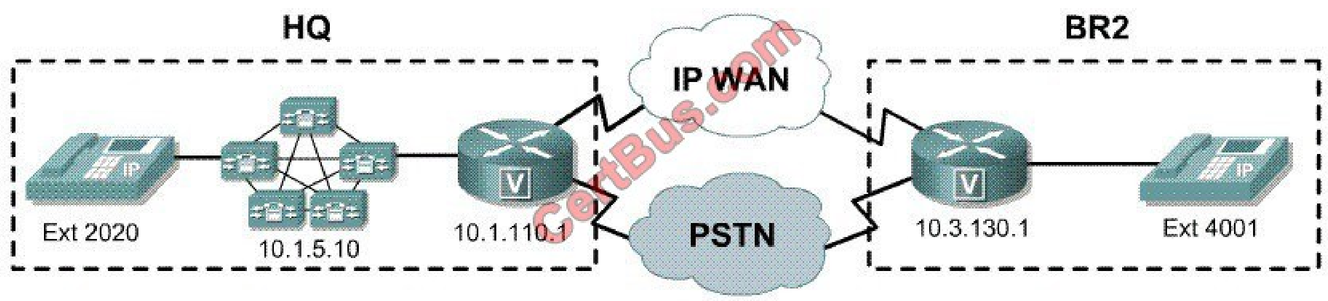
Look at the following exhibit carefully.

You can click the Voice Gateway for the BR2 location to see the output from the debug voice ccapi inout command and click on 10.1.5.10 to view and search the trace file output. You can also enter a string in the Search box and click the Find

button to search the output. X can be clicked to back to the item.

As a network technician, you have recently configured a trunk between the Cisco Unified CallManager cluster at 10.1.5.10 and a CME at a 10.3.130.1. However, in the testing of this configuration, you discover that you cannot complete any

call when dialing from ext. 2020 to ext.4001 or from ext.4001 to ext.2020. Please choose the most possible reason from the following statements.



A. A transaction rule has been applied that is keeping the call from being completed.

B. The CSS has been omitted from the trunk configured to BR2.

C. An incorrect CSS has been applied to the gateway at HQ.

D. The trunk IP address in the Cisco Unified CallManager information field is incorrect.

Correct Answer: D

QUESTION 6

Users report poor audio quality when they call across a WAN connection despite QoS being configured on the WAN links. Which command verifies if RTP packets are being dropped?

A. show policy-map

B. show class-map

C. show policy-map interface serial 0/0/0

D. show interface serial 0/0/0

Correct Answer: C

QUESTION 7

Which two troubleshooting tools would initially be the best to use when troubleshooting the PSTN gateway side of a call routing issue while using Cisco Unified Communications Manager? (Choose two)

A. RTMT trace output

B. Cisco IOS debug commands

C. Dialed Number Analyzer output

D. Cisco Unified Communications Manager alerts

E. Cisco IOS show commands

Correct Answer: BE

QUESTION 8

Refer to the exhibit

```
router eigrp SAF
 !
 service-family ipv4 autonomous-system 1
  !
  topology base
  exit-sf-topology
 exit-service-family
!
voice service saf
 !
 channel 1 vrouter SAF asystem 1
!
```


Which statement is true about the Cisco SAF Forwarder configuration on a Cisco Unified Communications Manager Express?

A. The Cisco Unified Communications Manager Express will not be able to register with the Cisco SAF Forwarder because the service-family external-client configuration is missing.

B. The Cisco Unified Communications Manager Express will be able to register with the Cisco SAF Forwarder

C. The QSCC Unified Communications Manager Express will not be able to register with the Cisco SAF Forwarder because the trunk configuration under voice service saf is missing

D. The Cisco Unified Communications Manager Express will not be able to register with the Cisco SAF Forwarder because call control services not been configured

E. The Cisco Communications Manager Express will be able to register with the Cisco SAF Forwarder, but an error message will be displayed regarding the missing trunk configuration

Correct Answer: B

QUESTION 9

Which default switchover method is used by the SCCP client to connect to another Cisco Unified Communications Manager after losing connectivity with the first Cisco Unified Communications Manager?

A. immediate

B. urgent

C. graceful

D. panic

E. recovery

F. static

Correct Answer: C

QUESTION 10

Acme Corporation has Cisco IP Phones configured in a single-site Cisco Unified Communications Manager cluster. The 10 phones in the accounting department can call each other. Another 10 phones in the finance department can call each other. However, no phones in accounting can call the phones in finance.

Which change is needed to allow all accounting phones to call all finance phones?

A. Ensure that all phones use SIP signaling.

B. Ensure that all phones use SCCP signaling.

C. Ensure that all phones are the same Cisco IP Phone model.

D. Ensure that all accounting phones are in a CSS that contains the partition that includes the finance phones.

E. Ensure that all finance phones are in a CSS that is in a partition that includes the accounting phones.

Correct Answer: D

QUESTION 11

Please choose the function of an H.323 gateway from the following items.

A. Converts an alias address to an IP address

B. Responds to bandwidth requests and modifications

C. Transmits and receives G.711 PCM-encoded voice

D. Performs translation between audio, video, and data formats

Correct Answer: D

QUESTION 12

You have placed all DNs in the Phones partition. During testing you discover that you cannot place calls between IP phones, but you can place calls to the PSTN and voice mail. What is one possible cause of this issue?

A. A database replication issue is preventing calls between phones.

B. An access list is blocking RTP streams in your voice VLAN.

C. An access list is blocking SCCP packets in your voice VLAN.

D. The IP phones have not been assigned a CSS.

Correct Answer: D