Cisco.300-080.v2018-03-16.q100

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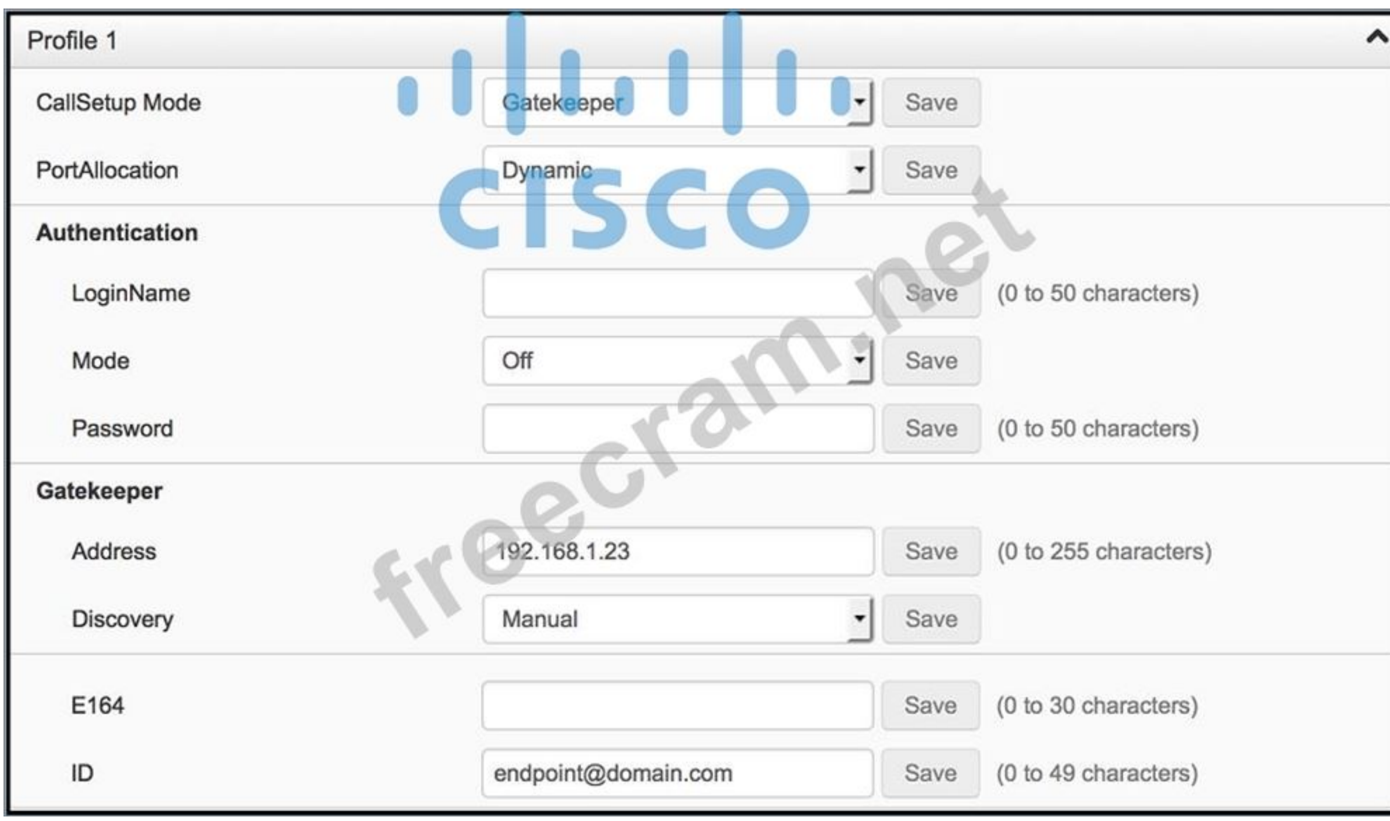
NEW QUESTION: 1
Refer to the exhibit.



These settings are configured on a Cisco TelePresence System EX90. What is the result?

A. The endpoint successfully registers to Cisco VCS as an H.323 endpoint.

B. The endpoint does not register to Cisco Unified Communications Manager as a SIP endpoint, because the domain information is missing.

C. 323 endpoint.

D. The endpoint successfully registers to Cisco Unified Communications Manager as an

E. The endpoint successfully registers to Cisco Unified Communications Manager as a SIP endpoint.

F. The endpoint does not register to Cisco VCS as a SIP endpoint, because the domain information is missing.
G. The endpoint successfully registers to Cisco VCS as a SIP endpoint.
Answer: (SHOW ANSWER)

NEW QUESTION: 2
An engineer configured a Cisco TelePresence server with two Cisco acquired Codian devices. Users are reporting that the image looks frozen and the audio contains static and is garbled on one of the devices. What is the root cause of the issue?
Refer to the exhibit.

```
SWG-POE#sh int gi 0/12
GigaEthernet0/12 is up, line protocol is up (connected)
  Hardware is Gigabit Ethernet, address is 000c.854d.a78c (bia 000c.854d.a78c)
  MTU 1500 bytes, BW 1000000 Kbit, DLY 1000 usec,
     reliability 255/255, txload 34/255, rxload 23/255
  Encapsulation ARPA, loopback not set
  Keepalive set (10 sec)
  Half-duplex, 1000Mb/s, media type is 10/100/1000BaseTX
  input flow-control is off, output flow-control is unsupported
  ARP type: ARPA, ARP Timeout 04:00:00
  Last input never, output never, output hang never
  Last clearing of "show interface" counters never
  Input queue: 0/75/19872/0 (size/max/drops/flushes); Total output drops: 17526
  Queueing strategy: fifo
  Output queue: 0/40 (size/max)
  5 minute input rate 121221168 bits/sec, 1201221 packets/sec
  5 minute output rate 9879879 bits/sec, 76876 packets/sec
```
A. There is a mismatch in the port configuration.
B. There are packet drops in the ingress queues
C. The input flow control is off.
D. The QoS that is configured on the port is set to FIFO
Answer: (SHOW ANSWER)

NEW QUESTION: 3
What is a common reason that an IP Phone cannot get its configuration from Cisco Unified Communications Manager after it obtains the correct IP address information?
A. The DHCP scope is exhausted.
B. The DHCP scope has the incorrect Option 150 or 66 defined.
C. The DHCP scope is on the wrong subnet.
D. The DHCP server is not reachable.
Answer: (SHOW ANSWER)

NEW QUESTION: 4
You need to collect debug Information in a production environment on urgent basis to troubleshoot a problem. Which two commands are recommended by Cisco that you should run? (Choose two.)
A. no debug all

B. no logging console

C. undebug all

D. show debug

E. logging buffered

F. show logging info

Answer: (SHOW ANSWER)

NEW QUESTION: 5

Refer to the exhibit.

```
R1#show inventory
NAME: "CISCO2901/K9 chassis", DESCR: "CISCO2901/K9 chassis"
PID: CISCO2901/K9      , VID: V04 , SN: FGL160521F8

NAME: "VWIC3-1MFT-T1/E1 - 1-Port RJ-48 Multiflex Trunk - T1/E1 on Slot 0 SubSlot 0", DESCR: "VWIC3-1MFT-T1/E1
PID: VWIC3-1MFT-T1/E1  , VID: V01 , SN: FOC16020NRF

NAME: "PVDM3 DSP DIMM with 32 Channels on Slot 0 SubSlot 4", DESCR: "PVDM3 DSP DIMM with 32 Channels"
PID: PVDM3-32          , VID: V01 , SN: FOC16021305

NAME: "C1941/C2901 AC Power Supply", DESCR: "C1941/C2901 AC Power Supply"
PID: PWR-1941-2901-AC  , VID:     , SN:

R1#
R1#
R1#show voice dsp capabilities slot 0 dsp 1
DSP Type: SP2600 -32
Card 0 DSP id 1 Capabilities:
    Credits   480 ,  G711Credits 15,  HC Credits  34,  MC Credits  22,
    FC Channel  32,  HC Channel  14,  MC Channel  21,
    Conference 8-party credits:
     G711  36 ,  G729  96 ,  G722  96 ,  ILBC  120
Secure Credits:
   Sec LC Xcode  20,     Sec HC Xcode  34,
   Sec MC Xcode  26,     Sec LC UNIV Xcode  20,
   Sec HC UNIV Xcode  68,Sec MC UNIV Xcode  40,
   Sec G729 conf 120,    Sec G722 conf 120,     Sec ILBC conf 160,
   Sec G711 conf 68 ,
Max Conference Parties per DSP:
    G711 104,  G729 40,  G722 40,  ILBC 32,
    Sec G711 56,  Sec G729 32,
    Sec G722 32  Sec ILBC 24,
Voice Channels:
    g711perdsp = 32,  g726perdsp = 21, g729perdsp = 14, g729aperdsp = 21,
    g723perdsp = 0 ,  g728perdsp = 14, g711_5msperdsp = 22, gsmamrnbperdsp = 14,
    ilbcperdsp = 14,  isacperdsp = 7 , modemrelayperdsp = 14,
    g72264Perdsp = 21, h324perdsp = 14,
    m_f_thruperdsp = 32, faxrelayperdsp = 21,
    maxchperdsp = 32, minchperdsp = 14,
    srtp_maxchperdsp = 18, srtp_minchperdsp = 9 , faxrelay_srtp_perdsp = 9 ,
    g711_srtp_perdsp = 18, g729_srtp_perdsp = 9 , g729a_srtp_perdsp = 16
    gnx64_srtp_perdsp = 18

R1#
```

How many high-complexity transcoding sessions can this Cisco ISR G2 support?

A. 60

B. 14

C. 21

D. 15

Answer: (SHOW ANSWER)

NEW QUESTION: 6

An engineer integrated a voice gateway with Cisco Unified Communications Manager using H.323, but the calls through it are failing. Which debug command helps isolate this issue?

A. debug call-mgmt

B. debug mgcp errors

C. debug voip ccapi inout

D. debug ras

E. debug ccsip error
Answer: (SHOW ANSWER)

NEW QUESTION: 7
To achieve 720p (HD) quality at 30 frames per second on an endpoint that is running TC software, what is the minimum configured call rate?
A. 1152 kbps
B. 768 kbps
C. 512 kbps
D. 2560 kbps
Answer: (SHOW ANSWER)

NEW QUESTION: 8
Which configuration is required on Cisco TelePresence Server, in order to support 1080p resolution?
A. Cisco TelePresence Server must be in HD mode.
B. Cisco TelePresence Server must be configured with Cisco TelePresence Conductor.
C. Cisco TelePresence Server must be in remotely managed mode.
D. Screen licenses must be configured.
E. Cisco TelePresence Server must be in Full HD mode.
Answer: E (LEAVE A REPLY)

NEW QUESTION: 9
When user attempted to call a colleague at the same site, the caller received a recording that the call could not be completed as dialed. Which two actions can you take to troubleshoot the problem? (Choose two.)
A. Reboot the Cisco Unified communications Manager Cluster.
B. Ping the remote gateway to verify connectivity.
C. Reboot the user's IP phone.
D. Reboot the IP phone that the user attempted to call.
E. Use RTMT to trace the call from DN to DN.
F. Verity that the partition and calling search space are correct.
Answer: B,F (LEAVE A REPLY)

NEW QUESTION: 10
For which two reasons does music on hold fail to play? (Choose two.)
A. Analog lines are terminated into the voice gateway.
B. The DRAM is corrupted on the Cisco IOS router.
C. iPv4 addressing is in use.
D. A SIP trunk is configured between the phone and the music on hold server.
E. Unicast music on hold is configured and the IP addressing mode is set to IPv4 Only.

F. Multicast music on hold is configured and the IP addressing mode is set to IPv6 Only.
Answer: (SHOW ANSWER)

NEW QUESTION: 11
You are trying to register an H.323-based Cisco TelePresence system to Cisco Unified Communications Manager and a Cisco DX70 system to the Cisco VCS Control. Why do neither of the units want to register?
A. Both systems need to register to the Cisco VCS Control, but the H.323-based system needs to have the gatekeeper setting set to "Direct."
B. You need Cisco TelePresence Management Suite to register Cisco TelePresence systems.
C. You need Cisco TelePresence Server to register Cisco TelePresence systems.
D. Both systems need to register to the Cisco Unified Communications Manager, as the Cisco VCS Control is used only for firewall traversal.
E. The H.323-based system needs to register to the Cisco VCS Control with an E.164 number, and the Cisco DX70 needs the TFTP address to register on the Cisco Unified Communications Manager.
F. The H.323-based system needs an E164 number to register to Cisco Unified Communications Manager, and the Cisco DX70 needs to have the MAC address configured first on the Cisco VCS Control.
Answer: (SHOW ANSWER)

NEW QUESTION: 12
Which CLI command monitors ILS replication progress?
A. utils ils showpeerinfo
B. utils ils show peer info
C. utils ils findxnode
D. utils ils lookup
Answer: A (LEAVE A REPLY)

NEW QUESTION: 13
You have just configured Cisco CallManager.
A few users in your organization complain that when they log out, the phones are reset.
You need to provide a solution to this problem.
What should you do?
A. Restart the TFTP service.
B. Restart the Internet Information Service (IIS).
C. Change the DNS address to IP address.
D. Reboot the phones.
E. Edit the form.jsp file and make appropriate changes.
F. Restart the Computer Telephony Integration (CTI).
Answer: (SHOW ANSWER)

NEW QUESTION: 14

```
Content-Type: application/sdp
Content-Length: 879

v=0
o=CiscoSystemsCCM-SIP 277 1 IN IP4 172.16.100.11
s=SIP Call
c=IN IP4 172.16.100.101
b=AS:128
t=0 0
m=audio 24068 RTP/AVP 9 101
a=rtpmap:9 G722/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 27882 RTP/AVP 126
b=TIAS:64000
a=rtpmap:126 H264/90000
a=fmtp:126 profile-level-id=42E01F;packetization-mode=1;max-
fs=3601;max-rcmd-nalu-size=32000;level-asymmetry-allowed=1
a=imageattr:126 recv
[x=[32:1:1280],y=[18:1:720],par=1.7778,q=1.00]
a=content:main
m=video 28032 RTP/AVP 126
b=TIAS:64000
a=rtpmap:126 H264/90000
a=fmtp:126 profile-level-id=42E01F;packetization-mode=1;max-
fs=8101;max-rcmd-nalu-size=32000;level-asymmetry-allowed=1
a=content:slides
m=application 5748 UDP/BFCP *
m=application 33516 RTP/AVP 125
a=rtpmap:125 H224/4800
a=rtcp:33517
```

Refer to the exhibit. Based on the ACK SDP output, what can be concluded?

A. There is no DTMF support for the call.

B. High-definition video is configured for the endpoints

C. The total amount of bandwidth for the video channels is 192 kbps.

D. This event is a multipoint video call with three participants.

Answer: (SHOW ANSWER)

NEW QUESTION: 15

An engineer is configuring Cisco Unified Communications Manager (CUCM) Version 10.0. The office manager of a remote office sends a request to set up a new phone that contains the MAC address of the phone (0AFC00021AB2) and the extension (2324). The engineer manually creates a new Cisco Unified IP Phone 7942 in CUCM, but the phone displays "Registration Rejected" when the user plugs the phone into the LAN. What should be done to fix this issue if autoregistration is already enabled in CUCM?

A. Check the MAC address to make sure that it matches and that there are no extra characters in the MAC address field.

B. Ensure that the phone extension is not already used on another set.

C. Verify that the phone model being connected is the same type that is configured in CUCM.

D. Check the Enterprise License Manager to ensure that the system is not out of licenses.

Answer: (SHOW ANSWER)

NEW QUESTION: 16

Refer to the exhibit.

```
*Mar 24 16:17:54.190: ISDN Se0/0/0:15 Q931: RX <- SETUP pd = 8  callref = 0x00AA
        Bearer Capability i = 0x8090A3
                Standard = CCITT
                Transfer Capability = Speech
                Transfer Mode = Circuit
                Transfer Rate = 64 kbit/s
        Channel ID i = 0xA98381
                Exclusive, Channel 1
        Progress Ind i = 0x8183 - Origination address is non-ISDN
        Calling Party Number i = 0x1180, '4940302156001'
                Plan:ISDN, Type:International
        Called Party Number i = 0x81, '2288223001'
                Plan:ISDN, Type:Unknown
*Mar 24 16:17:54.210: ISDN Se0/0/0:15 Q931: TX -> RELEASE_COMP pd = 8  callref =
                            0x80AA
        Cause i = 0x8081 - Unallocated/unassigned number
```

The exhibit shows the output of debug isdn q931. An inbound PSTN call was received by a SIP gateway that is reachable via a SIP trunk that is configured in Cisco Unified Communications Manager. The call failed to ring extension 3001. If the phone at extension 3001 is registered and reachable through the gateway inbound CSS, which three actions can resolve this issue? (Choose three.)

A. Configure a voice translation profile in the SIP Cisco IOS gateway with a voice translation rule that truncates the number from ten digits to four digits.

B. Change the significant digits for inbound calls to 4 on the SIP trunk configuration in Cisco Unified Communications Manager.

C. Configure a called-party transformation CSS on the gateway in Cisco Unified Communications Manager that includes a pattern that transforms the number from ten digits to four digits.

D. Configure a translation pattern in Cisco Unified Communications Manager that can be accessed by the trunk CSS to truncate the called number to four digits.

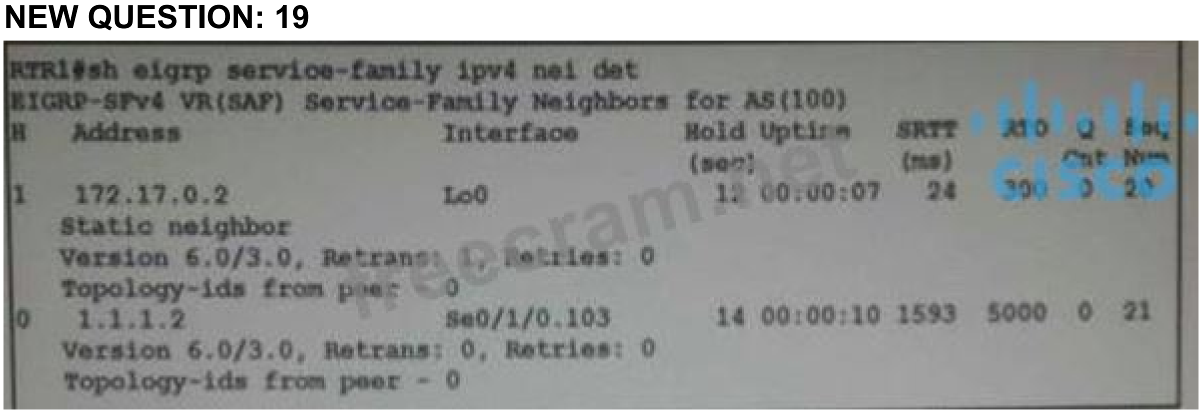
E. Configure the digit strip 4 on the SIP trunk under Incoming Called Party Settings in Cisco Unified Communications Manager.

F. Configure the Cisco IOS command num-exp 2288223001 3001 on the gateway ISDN interface.

Answer: (SHOW ANSWER)

NEW QUESTION: 17
You are configuring Troubleshooting Perfmon Data-Logging parameters.
You want to specify the Cisco recommended file size that can be stored in a perfmon log file.
What is the maximum file size value that you must specify?
A. 5 MB
B. 600 MB
C. 150 MB
D. 100 MB
E. 500 MB
F. 50 MB
Answer: (SHOW ANSWER)

NEW QUESTION: 18
Which Cisco Unified Communications Manager tool can you use to troubleshoot issues with international calling?
A. Cisco Deployment Tool
B. Cisco Prime
C. Dialed Number Analyzer
D. RTMT
Answer: (SHOW ANSWER)

NEW QUESTION: 19


Refer to the exhibit. An engineer is troubleshooting a SAF forwarder adjacency in which two interfaces from the same router are being advertised to all other neighbors. Which set of IOS commands successfully stops the router from advertising the Loopback0 interface?
A. router eigrp SAF
service-family ipv4 autonomous-system 100
sf-interface Loopback0
exit-sf-interface
B. router eigrp SAF

service-family ipv4 autonomous-system 100
sf-interface Loopback0
shutdown
C. router eigrp SAF
service-family ipv4 autonomous-system 100
sf-interface Loopback0
no split-horizon
D. route eigrp SAF
service-family ipv4 autonomous-system 100
sf-interface Loopback0
no dampening-change
Answer: (SHOW ANSWER)

NEW QUESTION: 20
After a Cisco Unified Communications Manager system is installed, users report problems when more than four users attempt to join a Meet-Me conference. Which parameter should you increase?

A. Maximum Meet-Me Conference, Enterprise Parameters Configuration
B. Maximum Meet-Me Conference, Call Manager Service Parameter
C. Maximum Ad Hoc Conference, Enterprise Parameters Configuration
D. Maximum Ad Hoc Conference, Call Manager Service Parameter
Answer: (SHOW ANSWER)

NEW QUESTION: 21
You are experiencing bad video quality in calls, and you suspect packet loss in the network. Both the network switch and the collaboration endpoint network interfaces are set to autonegotiation, but the reported port speed and duplex settings do not match. Which action should you take?
A. Configure the switch port to use a manual speed and duplex setting of 100 Mbps and half duplex, and keep the collaboration endpoint setting as "auto."
B. Set both the collaboration endpoint and the switch port to use manual speed and duplex settings of 100 Mbps and full duplex.
C. Keep both the switch port and collaboration endpoint settings as "auto," because this configuration is not responsible for the mismatched settings that were reported.
D. Set the collaboration endpoint to use a manual speed and duplex setting of 100 Mbps and full duplex, and keep the switch port setting as "auto."
Answer: (SHOW ANSWER)

NEW QUESTION: 22
Users of your local Cisco Unified Communications Manager cluster report that they receive error "Login is unavailable (23)" when they try to log in to Extension Mobility. Which reason for this error is true?
A. The given user ID is not found n any of the remote clusters.

B. User has no Extension Mobility profiles assigned.
C. Phone is not subscribed to Extension Mobility phone service.
D. User provided the wrong UserID or PIN
Answer: (SHOW ANSWER)

NEW QUESTION: 23
As a Voice Administrator, you have configured the Service Advertisement Call Control Discovery (CCD) feature. When you encounter CCD issues, which command would you use on the IOS interface to check all the learned routes on the IOS device?
A. show voice saf client
B. show eigrp service-family ipv4 interfaces
C. show voice saf dial-peer outbound
D. show voice safdndb all
E. show ccm-manager
F. show eigrp service-family ipv4 topology
Answer: (SHOW ANSWER)

NEW QUESTION: 24
An engineer created an inter site link in cisco unity connection by using the Cisco Unity Connection Site by manually exchanging configuration files option.
The error appears "Hostname entered does not match that on the remote site certificate".
What is the one cause of the issue?
A. DNS is not configured on the Cisco Unity Connection site gateway.
B. The FQDN does not match the remote site gateway configuration file.
C. A partition is missing from search spaces on the Cisco Unity Connection location.
D. The FQDN does not match the server name on the remote site gateway web SSL certificate.
E. The inter site location is a member of the same cisco unity connection site
Answer: (SHOW ANSWER)

NEW QUESTION: 25
You are 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 should you use to mark the video traffic, according to the Cisco QoS baseline?
A. EF
B. AF31
C. CS3
D. CS2
E. CS6
F. AF41
Answer: (SHOW ANSWER)

NEW QUESTION: 26

You need to increase the maximum number of Meet-Me conference participants on Cisco Unified Communications Manager. Where do you configure this increase?

A. Media Resources > Conference Bridge

B. Call Routing > Meet-Me Number/Pattern

C. System > Service Parameters > [Publisher Server] > Cisco CallManager (Active) > Clusterwide Parameters (Feature - Conference)

D. Device > Conference Bridge

E. Media Resources > Media Resource Group

Answer: (SHOW ANSWER)

NEW QUESTION: 27

Refer to Exhibit:

```
Exhibit1
2015/02/09 11:13:27.672|SIPL|0|TCP|IN|10.1.5.25|5060|SEP8478ACE71D1F|10.1.111.3|50740|1,100,14,146.36^10.1.111.3^*|173|8478ace7-1d1f0006-43f1b7ba-45b75221b@10.1.111.3|INVITE
2015/02/09 11:13:27.672|SIPL|0|TCP|OUT|10.1.5.25|5060|SEP8478ACE71D1F|10.1.111.3|50740|1,100,14,146.36^10.1.111.3^*|174|8478ace7-1d1f0006-43f1b7ba-45b77521b@10.1.111.3|100 Trying
2015/02/09 11:13:29.371|CC|SETUP|23329196|23329197|2001|2003|2003
2015/02/09 11:13:29.438|SIPL|23329197|TCP|OUT|10.1.5.25|5060|SEP64AE0CF7BC79|10.1.111.2|49499|1,100,14,1.12^*^*|186|b9a7d800-4d9106d9-9-1905010a@10.1.5.25|INVITE
2015/02/09 11:13:29.439|CC|OFFERED|23329196|23329197|2001|2003|2003|SEP8478ACE71D1F|SEP64AE0CF7BC79
2015/02/09 11:13:29.530|SIPL|23329197|TCP|IN|10.1.5.25|5060|SEP64AE0CF7BC79|10.1.111.2|49499|1,100,14,119.25^10.1.111.2^*|188|b9a7d800-4d9106d9-9-19056010a@10.1.5.25|100 Trying
2015/02/09 11:13:29.667|SIPL|23329197|TCP|IN|10.1.5.25|5060|SEP64AE0CF7BC79|10.1.111.2|49499|1,100,14,119.26^10.1.111.2^*|191|b9a7d800-4d9106d9-9-19056010a@10.1.5.25|180 Ringing
2015/02/09 11:13:29.670|SIPL|23329196|TCP|OUT|10.1.5.25|5060|SEP8478ACE71D1F|10.1.111.3|50740|1,100,14,119.26^10.1.111.2^*|192|8478ace7-1d1f0006-43f1b7ba-45b7521b@10.1.111.3|180 Ringing
2015/02/09 11:13:38.651|SIPL|23329196|TCP|IN|10.1.5.25|5060|SEP8478ACE71D1F|10.1.111.3|50740|1,100,14,146.45^10.1.111.3^*|195|8478ace7-1d1f0006-43f1b7ba-45b7521b@10.1.111.3|CANCEL
2015/02/09 11:13:38.651|SIPL|23329196|TCP|OUT|10.1.5.25|5060|SEP8478ACE71D1F|10.1.111.3|50740|1,100,14,146.45^10.1.111.3^*|196|8478ace7-1d1f0006-43f1b7ba-45b7521b@10.1.111.3|200 OK
2015/02/09 11:13:38.670|SIPL|23329197|TCP|OUT|10.1.5.25|5060|SEP64AE0CF7BC79|10.1.111.2|49499|1,100,14,146.45^10.1.111.3^*|197|b9a7d800-4d9106d9-9-19005010a@10.1.5.25|CANCEL
2015/02/09 11:13:38.671|SIPL|23329196|TCP|OUT|10.1.5.25|5060|SEP8478ACE71D1F|10.1.111.3|50740|1,100,14,146.45^10.1.111.3^*|198|8478ace7-1d1f0006-43f1b7ba-45b7521b@10.1.111.3|487 Request Cancelled
2015/02/09 11:13:38.772|SIPL|23329196|TCP|IN|10.1.5.25|5060|SEP8478ACE71D1F|10.1.111.3|50740|1,100,14,146.46^10.1.111.3^*|199|8478ace7-1d1f0006-43f1b7ba-45b7521b@10.1.111.3|ACK
2015/02/09 11:13:38.774|SIPL|23329197|TCP|IN|10.1.5.25|5060|SEP64AE0CF7BC79|10.1.111.2|49499|1,100,14,119.27^10.1.111.2^*|200|b9a7d800-4d9106d9-9-19056010a@10.1.5.25|200 OK
2015/02/09 11:13:38.775|CC|RELEASE|23329196|23329197|16
2015/02/09 11:13:38.851|SIPL|23329197|TCP|IN|10.1.5.25|5060|SEP64AE0CF7BC79|10.1.111.2|49499|1,100,14,119.28^10.1.111.2^*|201|b9a7d800-4d9106d9-9-19056010a@10.1.5.25|487 Request Cancelled
2015/02/09 11:13:38.851|SIPL|23329197|TCP|OUT|10.1.5.25|5060|SEP64AE0CF7BC79|10.1.111.2|49499|1,100,14,119.28^10.1.111.2^*|202|b9a7d800-4d9106d9-9-19005010a@10.1.5.25|ACK
```

What type of trace file is this considered to be?

A. sdi

B. dbnotify

C. calllogs

D. dbl

E. sdl

Answer: (SHOW ANSWER)

NEW QUESTION: 28

Users are complaining of problems when they make SIP calls by dialing URIs. To help users complete calls, what must you do?

A. Adjust the URI lookup policy to case nonsensitive.

B. Adjust the URI lookup policy to case sensitive.

C. Adjust the URI lookup policy to case desensitive.

D. Adjust the URI lookup policy to case insensitive.

Answer: (SHOW ANSWER)

NEW QUESTION: 29

A user is trying to call a mobile phone using the number 547895341, where 5 is the prefix to can off-net numbers. Calls to mobile phones have worked in past, but now the call does not work.
Which three areas should you check, to resolve the issue? (Choose three)
A. Verify that me PSTN line is connected to the cisco Unified Communications Manager.
B. Verify that the connection to and from the Cisco Unified Border Element is good.
C. Check that Cisco VCS Control and Cisco VCS Express are getting through the firewall.
D. Verify the search pattern.
E. Verify that the Cisco Unified Border Element is running.
F. Verity the route pattern, route list, and route group.
Answer: (SHOW ANSWER)

NEW QUESTION: 30
Partitions can be assigned to which two items? (Choose two)
A. directory numbers
B. IP phones
C. trunks
D. devices
E. gateways
Answer: (SHOW ANSWER)

NEW QUESTION: 31
System A at Company 1 is calling System B at Company 2. The call completes, but only audio and video are present on System A from System B.
What are three possible causes? (Choose three.)
A. An access list is blocking video and audio somewhere in the video and audio path between System A and System B.
B. System A has turned off the camera and the microphone.
C. The firewall at Company 1 is blocking outgoing traffic.
D. System A cannot call System B because it is at a different company.
E. There is a firewall in the path that is blocking audio and video traffic from Company 1 to Company 2.
Answer: (SHOW ANSWER)

NEW QUESTION: 32
Which two tasks are performed by the RAS signaling function of H.225.0? (Select two.)
A. Allows endpoints to create connections between call agents.
B. Performs disengage procedures between endpoints and a gatekeeper.
C. Transports audio messages between endpoints.
D. Performs bandwidth changes.
Answer: (SHOW ANSWER)

NEW QUESTION: 33
You have installed a Cisco Unified IP 8831 Conference Phone that is failing to register.
Which two actions must you take to troubleshoot the problem? (Choose two)
A. Verity that the correct drivers are installed on the switch port of the phone.
B. Verify that the RJ-11 cable is plugged into the PC port.
C. Check the RJ-65 cable.
D. Verify that the switch port of the phone is enabled.
E. Verity that the phone's network can access the option 150 server.
F. Disable HSRP on the access layer switch.
Answer: (SHOW ANSWER)

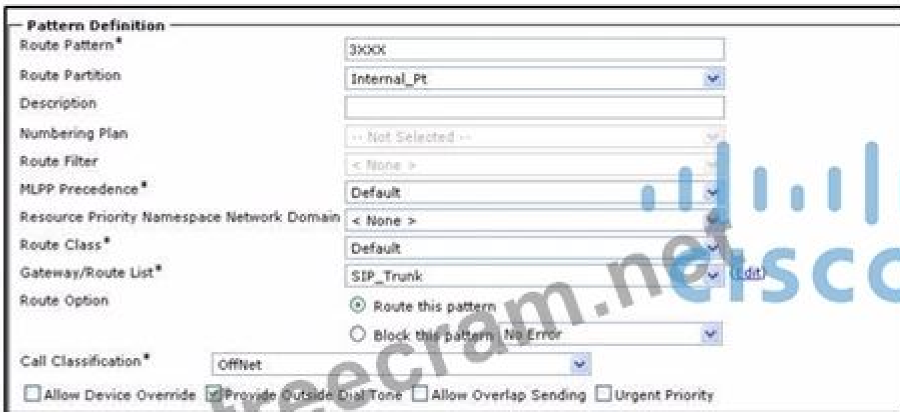
NEW QUESTION: 34
An engineer configured Cisco Extension Mobility for a user. The user can log in to the profile but notices that the extension is not correct. Which component should the engineer check first to verify the user extension?
A. Phone Service
B. SIP Profile
C. User Profile
D. Device Profile
E. UC Service
Answer: (SHOW ANSWER)

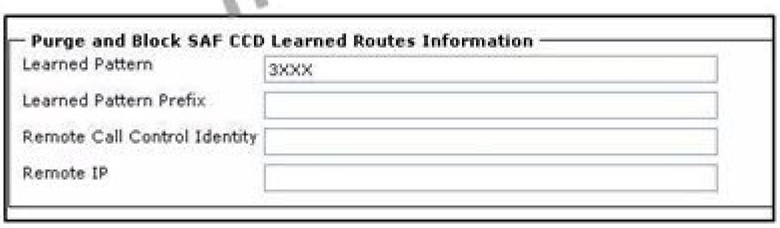
NEW QUESTION: 35
Which database replication value indicates that the server no longer has an active logical connection to receive database tables?
A. 4
B. 3
C. 2
D. 1
E. 0

Answer: (SHOW ANSWER)

NEW QUESTION: 36

Refer to the exhibits.





Assume that all learned SAF routes are placed in the SAF_Pt partition. The 3XXX directory number pattern is being advertised by a remote cluster and is also being blocked by the local cluster that is shown in the exhibit. An IP phone is attached to the local cluster and is configured with a CSS that contains the following partitions: SAF_Pt and Internal_Pt in this order. When the IP phone places a call to 3001, what will occur?

A. The call will fail because it will be blocked by the CCD Blocked Learned Route configuration.

B. The call will be placed in a round-robin fashion between the SAF network and SIP_Trunk.

C. The call will placed in a round-robin fashion between the SAF network and SIP_Trunk.
Every other call will fail.

D. The call will succeed and will be placed via the SIP_Trunk.

Answer: (SHOW ANSWER)

NEW QUESTION: 37

Endpoint A is registered to Cisco Unified Communications Manager as S1@company.com.
It is trying to call Endpoint B, which is registered to the same company's Cisco VCS Control with an H.323 ID of S2.internal@company.com. The route pattern is set to "*.*" and is pointed to a SIP trunk to the Cisco VCS Control. The search rule for (.*).internal@company.com is set to search the local zone. The call does not work. What is a possible reason?

A. The Cisco VCS Control is missing the Cisco Unified Communications Manager interop option.

B. The Cisco VCS Control should be neighbor to the Cisco Unified Communications Manager.

C. The Cisco VCS Control is missing the interworking option.

D. There is no search pattern to route the call to System B.

E. There is no valid route pattern to route from System A to System B.

F. You need an MGCP gateway to route from the Cisco Unified Communications Manager to the Cisco VCS Control.

G. System B is registered as H.323 and needs to use an E.164 alias number only.

Answer: (SHOW ANSWER)

NEW QUESTION: 38

Two phones in the same cluster and at the same site have a call currently connected. The site local H.323 PSTN gateway loses connection with Cisco Unified Communications Manager. Which two results do you expect? (Choose two.)

A. SRST is active, and all the phones enter SRST mode.

B. Cisco Unified SRST is able to receive incoming calls.

C. The current call is not disconnected.

D. The phones display "CM Fallback Service Operating."

E. No incoming and outgoing calls are possible.

Answer: (SHOW ANSWER)

NEW QUESTION: 39

A company has a headquarters and a remote site. Cisco Unified Communications Manager (CUCM) acts as a DHCP server. Both sites use their local voice gateways tor PSTN calls.

At the headquarters site, the PSTN prefix is 9, and the emergency number is 911. At the remote site, the PSTN prefix is 0, and the emergency number is 112. Here are the deployment policies for roaming devices:

* Softphones can roam between two sites.

* A roaming softphone uses the local gateway for all PSTN calls.

* The user keeps home dial habits on a roaming softphone, with the exception of the emergency number.

When a headquarters user uses a softphone at the remote site, prefix 9 can be used to make PSTN calls via voce gateway in the remote site, but emergency number 112 does not work.

Which setting on CUCM should be checked?

A. Physical Location

B. Device Mobility Info

C. Device Mobility Group

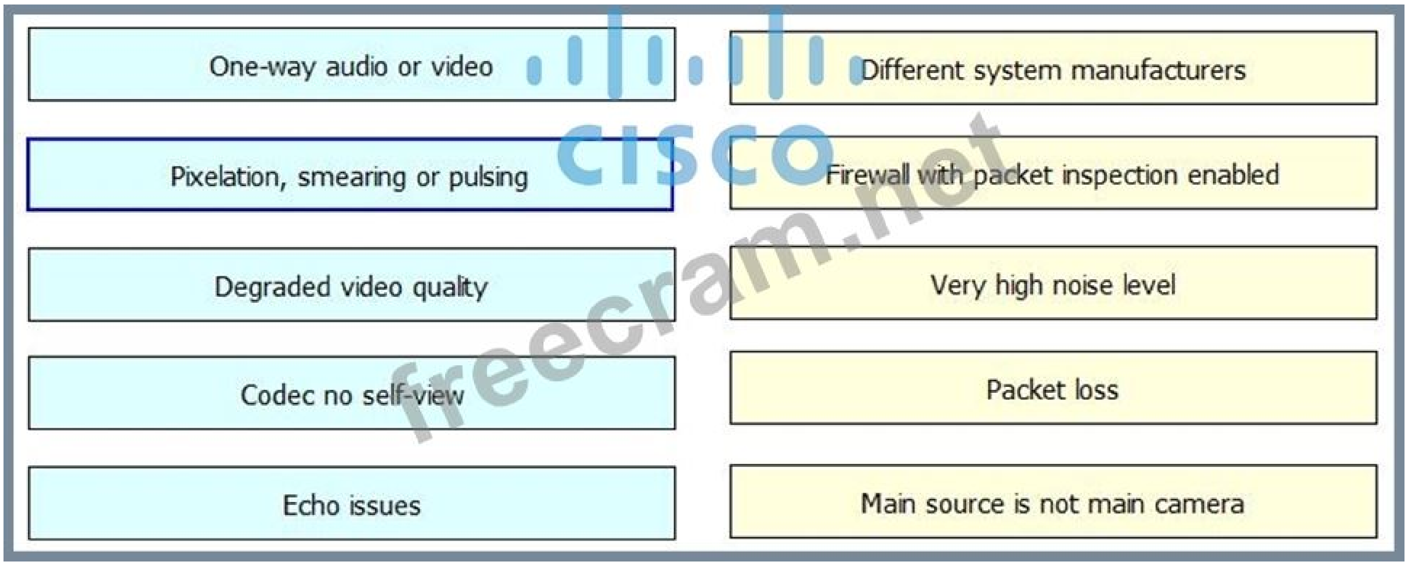
D. DHCP subnet

Answer: (SHOW ANSWER)

NEW QUESTION: 40

DRAG DROP

Drag the network-related video issue on the left to its root cause on the right.



Answer:

NEW QUESTION: 41

Refer to Exhibit:

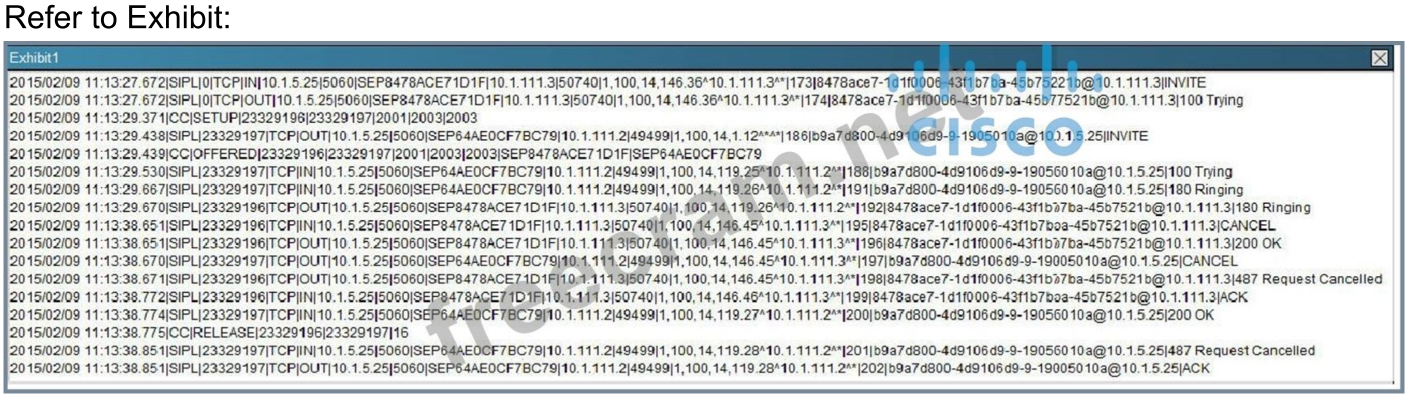


Exhibit1

```
2015/02/09 11:13:27.672|SIPL|0|TCP|IN|10.1.5.25|5060|SEP8478ACE71D1F|10.1.111.3|50740|1,100,14,146.36^10.1.111.3^*|173|8478ace7-1d1f0006-43f1b7ba-45b75221b@10.1.111.3|INVITE
2015/02/09 11:13:27.672|SIPL|0|TCP|OUT|10.1.5.25|5060|SEP8478ACE71D1F|10.1.111.3|50740|1,100,14,146.36^10.1.111.3^*|174|8478ace7-1d1f0006-43f1b7ba-45b77521b@10.1.111.3|100 Trying
2015/02/09 11:13:29.371|CC|SETUP|23329196|23329197|2001|2003|2003
2015/02/09 11:13:29.438|SIPL|23329197|TCP|OUT|10.1.5.25|5060|SEP64AE0CF7BC79|10.1.111.2|49499|1,100,14,1.12^*^*|186|b9a7d800-4d9106d9-9-1905010a@10.1.5.25|INVITE
2015/02/09 11:13:29.439|CC|OFFERED|23329196|23329197|2001|2003|2003|SEP8478ACE71D1F|SEP64AE0CF7BC79
2015/02/09 11:13:29.530|SIPL|23329197|TCP|IN|10.1.5.25|5060|SEP64AE0CF7BC79|10.1.111.2|49499|1,100,14,119.25^10.1.111.2^*|188|b9a7d800-4d9106d9-9-19056010a@10.1.5.25|100 Trying
2015/02/09 11:13:29.667|SIPL|23329197|TCP|IN|10.1.5.25|5060|SEP64AE0CF7BC79|10.1.111.2|49499|1,100,14,119.26^10.1.111.2^*|191|b9a7d800-4d9106d9-9-19056010a@10.1.5.25|180 Ringing
2015/02/09 11:13:29.670|SIPL|23329196|TCP|OUT|10.1.5.25|5060|SEP8478ACE71D1F|10.1.111.3|50740|1,100,14,119.26^10.1.111.2^*|192|8478ace7-1d1f0006-43f1b7ba-45b7521b@10.1.111.3|180 Ringing
2015/02/09 11:13:38.651|SIPL|23329196|TCP|IN|10.1.5.25|5060|SEP8478ACE71D1F|10.1.111.3|50740|1,100,14,146.45^10.1.111.3^*|195|8478ace7-1d1f0006-43f1b7ba-45b7521b@10.1.111.3|CANCEL
2015/02/09 11:13:38.651|SIPL|23329196|TCP|OUT|10.1.5.25|5060|SEP8478ACE71D1F|10.1.111.3|50740|1,100,14,146.45^10.1.111.3^*|196|8478ace7-1d1f0006-43f1b7ba-45b7521b@10.1.111.3|200 OK
2015/02/09 11:13:38.670|SIPL|23329197|TCP|OUT|10.1.5.25|5060|SEP64AE0CF7BC79|10.1.111.2|49499|1,100,14,146.45^10.1.111.3^*|197|b9a7d800-4d9106d9-9-19005010a@10.1.5.25|CANCEL
2015/02/09 11:13:38.671|SIPL|23329196|TCP|OUT|10.1.5.25|5060|SEP8478ACE71D1F|10.1.111.3|50740|1,100,14,146.45^10.1.111.3^*|198|8478ace7-1d1f0006-43f1b7ba-45b7521b@10.1.111.3|487 Request Cancelled
2015/02/09 11:13:38.772|SIPL|23329196|TCP|IN|10.1.5.25|5060|SEP8478ACE71D1F|10.1.111.3|50740|1,100,14,146.46^10.1.111.3^*|199|8478ace7-1d1f0006-43f1b7ba-45b7521b@10.1.111.3|ACK
2015/02/09 11:13:38.774|SIPL|23329197|TCP|IN|10.1.5.25|5060|SEP64AE0CF7BC79|10.1.111.2|49499|1,100,14,119.27^10.1.111.2^*|200|b9a7d800-4d9106d9-9-19056010a@10.1.5.25|200 OK
2015/02/09 11:13:38.775|CC|RELEASE|23329196|23329197|16
2015/02/09 11:13:38.851|SIPL|23329197|TCP|IN|10.1.5.25|5060|SEP64AE0CF7BC79|10.1.111.2|49499|1,100,14,119.28^10.1.111.2^*|201|b9a7d800-4d9106d9-9-19056010a@10.1.5.25|487 Request Cancelled
2015/02/09 11:13:38.851|SIPL|23329197|TCP|OUT|10.1.5.25|5060|SEP64AE0CF7BC79|10.1.111.2|49499|1,100,14,119.28^10.1.111.2^*|202|b9a7d800-4d9106d9-9-19005010a@10.1.5.25|ACK
```

What protocol was used in this call?

A. MGCP

B. SCCP

C. H.323

D. SIP

Answer: (SHOW ANSWER)

NEW QUESTION: 42

Refer to the exhibit.

A CUCM user has been configured to use the Mobility feature and is expecting their home phone 408-555-6016 to ring simultaneously when their office phone is called. What configuration change needs to be made to allow this happen?

A. Select the Line Association check box.

B. Select the Enable Move to Mobile check box.

C. Deselect the Enable Single Number Reach check box.

D. Select the Enable Extend and Connect check box.

Answer: A (LEAVE A REPLY)

NEW QUESTION: 43

Which two components can be software upgraded via Cisco TelePresence Management Suite? (Choose two.)

A. endpoints that are registered by Cisco Unified Communications Manager

B. endpoints that are registered by a partner company

C. endpoints that are provisioned by Cisco TelePresence Management Provisioning Extension

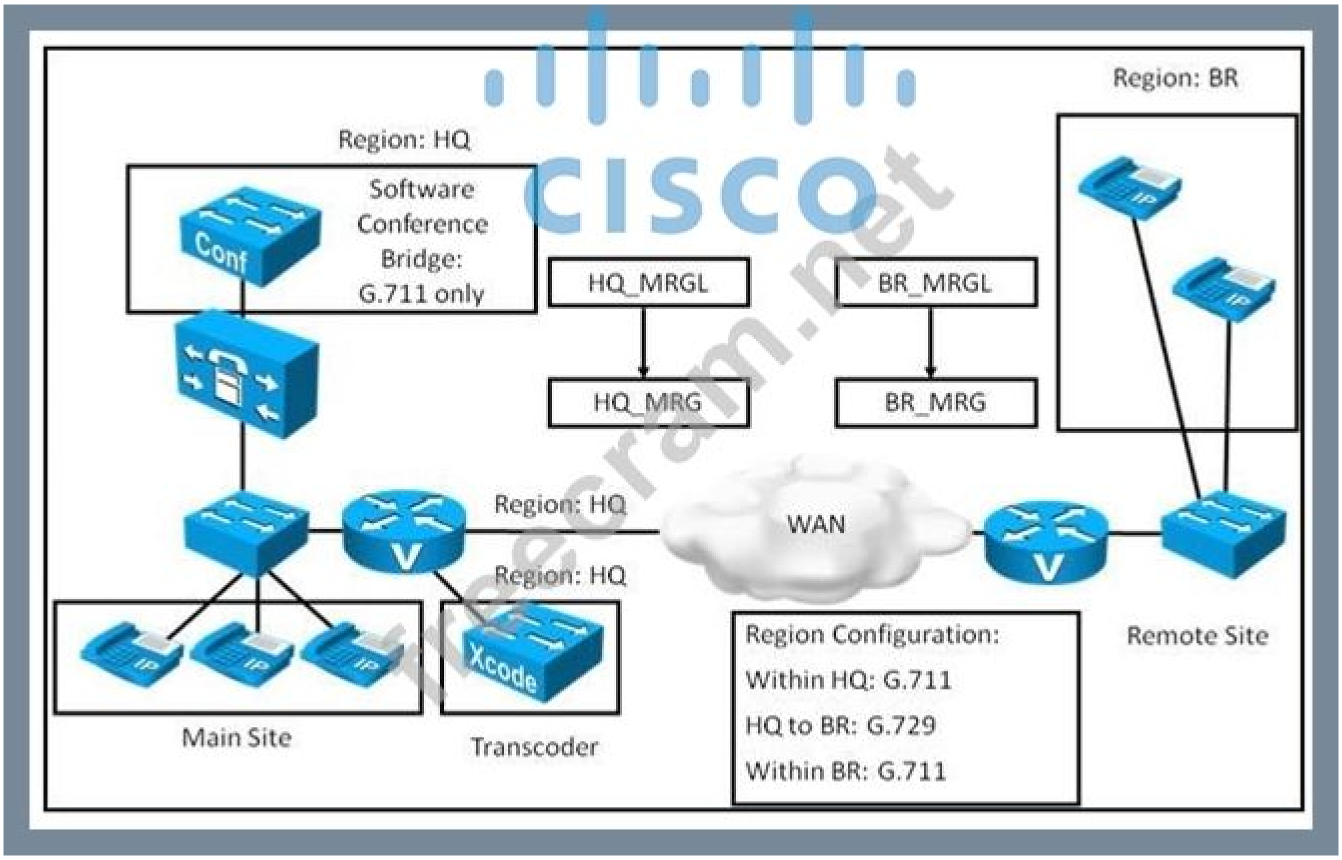
D. endpoints that are managed by Cisco TelePresence Management Suite

E. unmanaged endpoints

Answer: (SHOW ANSWER)

NEW QUESTION: 44

Refer to the exhibit.



When a call between two HQ users was being conferenced with a remote user at the BR site, the conference failed. Which configuration would be needed to solve the problem?

A. Enable the software conference bridge to support G.711 and G.729 codecs in Cisco Unified Communications Manager service parameters.

B. A transcoder should be configured at the remote site and assigned to all remote phones through the BR_MRGL.

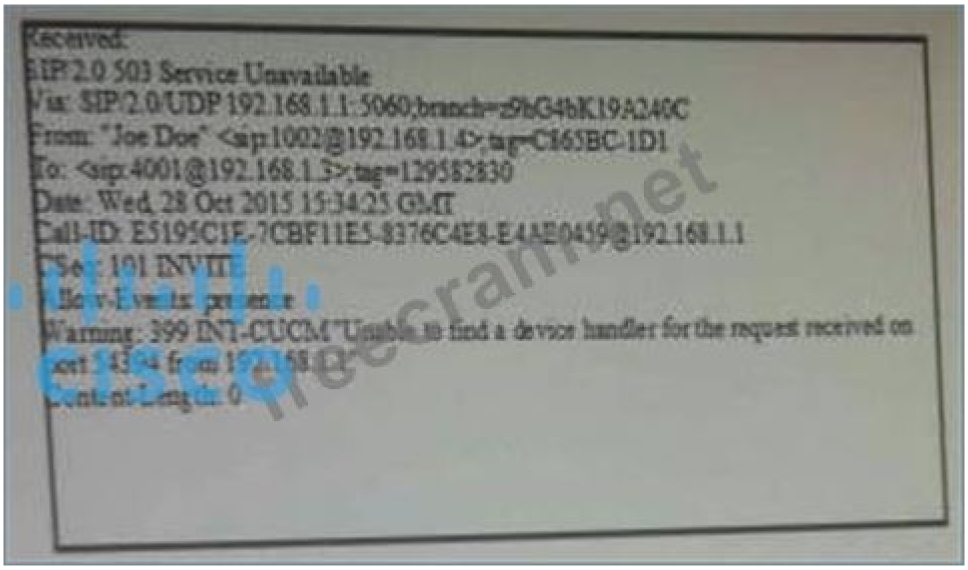
C. The HQ_MRG must contain the transcoder device. The HQ_MRGL must be assigned to the HQ phones.

D. The HQ_MRG must contain the transcoder device. The HQ_MRGL must be assigned to the software conference bridge.

E. The BR_MRG must contain the transcoder device. The BR_MRGL must be assigned to the BR phones.

Answer: (SHOW ANSWER)

NEW QUESTION: 45



Received:
SIP/2.0 503 Service Unavailable
Via: SIP/2.0/UDP 192.168.1.1:5060;branch=z9hG4bK19A240C
From: "Joe Doe" <sip:1002@192.168.1.4>;tag=C165BC-1D1
To: <sip:4001@192.168.1.3>;tag=129582830
Date: Wed, 28 Oct 2015 15:34:25 GMT
Call-ID: E5195C1E-7CBF11E5-8376C4E8-E4AE0459@192.168.1.1
CSeq: 101 INVITE
Allow-Events: presence
Warning: 399 INT-CUCM "Unable to find a device handler for the request received on port 54354 from 192.168.1.1"
Content-Length: 0

Refer to the exhibit. A customer has two Cisco Unified Communications Manager clusters (NY and CAL). The customer recently tried to enable intercluster communication between these clusters. When NY users call CAL, they get a fast-busy tone. A network administrator collects Cisco CallManager traces and sees the displayed SIP message coming from the remote Cisco Unified Communication Manager (CAL cluster). What is a likely reason for this?

A. The remote Cisco IP phone is not registered.

B. Cisco CallManager service is shut down on the CAL cluster.

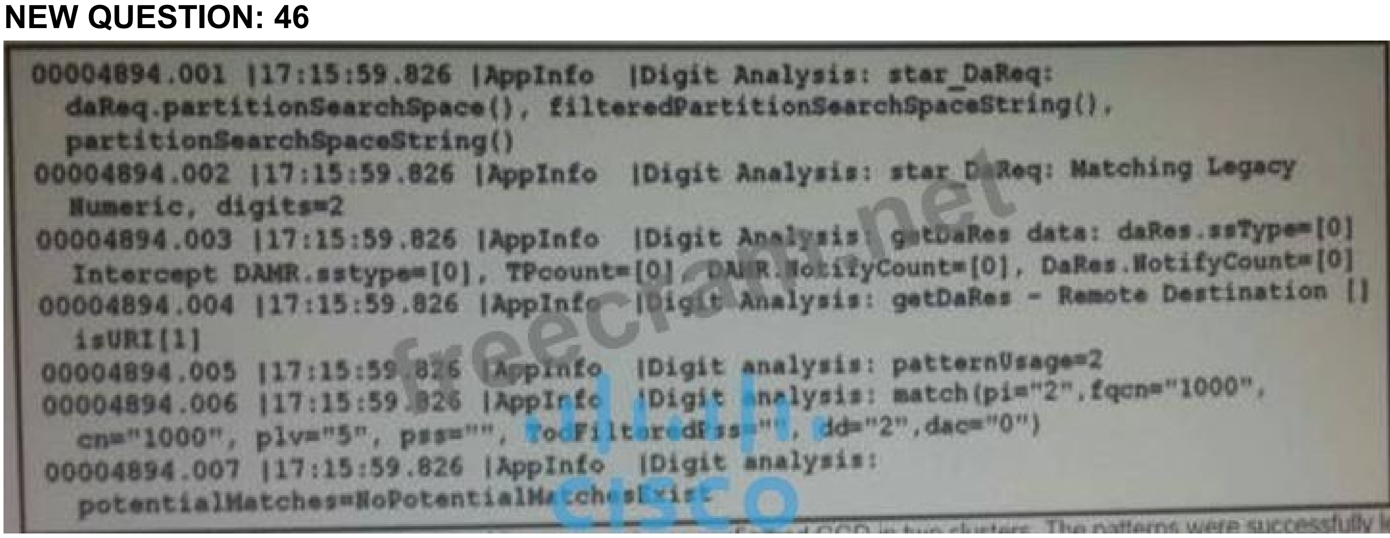
C. The SIP trunk on the CAL cluster is configured incorrectly.

D. The SIP service is shut down on the CAL cluster.

E. Cisco CTI Manager service is shut down on the CAL cluster.

Answer: (SHOW ANSWER)

NEW QUESTION: 46



```
00004894.001 |17:15:59.826 |AppInfo  |Digit Analysis: star_DaReq:
  daReq.partitionSearchSpace(), filteredPartitionSearchSpaceString(),
  partitionSearchSpaceString()
00004894.002 |17:15:59.826 |AppInfo  |Digit Analysis: star_DaReq: Matching Legacy
  Numeric, digits=2
00004894.003 |17:15:59.826 |AppInfo  |Digit Analysis: getDaRes data: daRes.ssType=[0]
  Intercept DAMR.sstype=[0], TPcount=[0], DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]
00004894.004 |17:15:59.826 |AppInfo  |Digit Analysis: getDaRes - Remote Destination []
  isURI[1]
00004894.005 |17:15:59.826 |AppInfo  |Digit analysis: patternUsage=2
00004894.006 |17:15:59.826 |AppInfo  |Digit analysis: match(pi="2",fqcn="1000",
  cn="1000", plv="5", pss="", TodFilteredPss="", dd="2",dac="0")
00004894.007 |17:15:59.826 |AppInfo  |Digit analysis:
  potentialMatches=NoPotentialMatchesExist
```

Refer to the exhibit. A Cisco Unified Communications Manager engineer configured CCD in two clusters. The patterns were successfully learned, however, the call attempt from the cluster 1 agent to the cluster 2 agent fails. Based on the output of this cluster 1 trace, what is the root cause of the issue?

A. The Activated Feature check in the CCD Requesting Service configuration is missing

B. The CSS is missing from the phones and line configuration

C. The pattern that was dialed has been added to the Blocked Learned Pattern configuration
D. The CCD partition is missing from the CCD Requesting Service configuration
Answer: (SHOW ANSWER)

NEW QUESTION: 47
When users in headquarters call branch office users over the WAN link, branch users report poor audio quality. Headquarters users consistently experience acceptable audio quality. Which troubleshooting approach most directly improves the audio quality of the branch users?
A. Make the headquarters router configuration for LLQ match the branch router.
B. Make the headquarters router configuration for CBWFQ match the branch router.
C. Make the branch router configuration for CBWFQ match the headquarters router.
D. Make the branch router configuration for LLQ match the headquarters router.
Answer: (SHOW ANSWER)

NEW QUESTION: 48

52661207.001 |15:14:31.997 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.86.224.118 on port 53151 index 600
[9789648,NET]
SIP/2.0 404 Not Found
Via: SIP/2.0/TLS 10.86.224.118:53151;branch=z9hG4bK78988629
From: <sip:85541234@10.86.224.118>;tag=54d814a485203e9272a2f1f9-27b3d79a
To: <sip:84431234@172.18.106...>;tag=1559694386
Date: Thu, 18 Jul 2015 19:14:31 GMT
Call-ID: 64d814a4-8520...-209b8352-3b583a7c@10.86.224.118
CSeq: 1000 INVITE
Server: Cisco-CUCM10.5
Content-Length: 0

Refer to the exhibit. Calls from Phone 1 (85541234), which is registered to Cluster A, are failing to Phone 2 (84431234), which is registered to Cluster B.
The exhibit was taken from Cluster B. What are two possible root causes for this problem? (Choose two.)
A. Incoming calling search space on the Cluster B SIP trunk is misconfigured.

B. The calling search space on the Cluster A SIP trunk is misconfigured
C. A translation pattern is misconfigured on Cluster B.
D. Cluster A is missing a route pattern to reach Cluster B.
E. Cluster B is missing a route pattern to reach Cluster A
Answer: B,D (LEAVE A REPLY)

NEW QUESTION: 49

Refer to the exhibit.

```
hostname HQ
!
card type e1 0 0
!
network-clock-participate wic 0
!
isdn switch-type primary-net5
!
voice-card 0
!
sccp local GigabitEthernet0/0.110
sccp ccm 10.1.5.10 identifier 1 version 7.0
sccp
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 1 register HQ_Conf
!
dspfarm profile 1 conference
 maximum conference-participants 0
 shutdown
```

```
HQ(config-dspfarm-profile)# maximum sessions 5
                                  ^
% Invalid input detected at '^' marker.
```

```
HQ(config-dspfarm-profile)#maximum sessions ?
  <0-0>  Number of sessions assigned to this profile
```


When the user tried to configure the command maximum sessions 5 under the dspfarm profile 1, the error shown in the exhibit was reported. Which course of action will resolve this issue?

A. Ensure that the conference bridge is not registered in Cisco Unified Communications Manager.
B. Configure the dspfarm all command under the voice card.
C. The maximum conference-participants value must be configured >0.
D. The command dsp services dspfarm must be configured under the voice-card configuration.
Answer: (SHOW ANSWER)

NEW QUESTION: 50

In a single-site deployment model, the internal endpoints are unable to dial from one to the other. What are two possible causes? (Choose two.)

A. The called endpoint is not in the partition of the calling endpoint.

B. The PSTN gateway is not configured.

C. The calling endpoint is not in the CSS of the called endpoint.

D. The calling endpoint is not configured for the correct CoS.

E. The called endpoint does not have the SIP trunk enabled.

F. The called endpoint is not registered.

Answer: (SHOW ANSWER)

NEW QUESTION: 51

Two Cisco Unified Communications Manager clusters have recently been installed. One is in Tokyo and the other in Chicago connected by a WAN link. You have started the configuration of a SIP trunk and you are integrating the dial plans. All phones in Tokyo can call all phones in Chicago, but no phones in Chicago can call Tokyo phones. Which two statements are possible causes? (Choose two.)

A. The Tokyo SIP trunk inbound CSS does not contain the partition of the Tokyo phones.

B. The CSS of the Chicago phones does not include the partition of the route pattern to point to Tokyo.

C. The SIP trunk is configured from Tokyo to Chicago, but not from Chicago to Tokyo.

D. The SIP trunk in Chicago is in a CSS that is not in the partition of the Chicago phones.

E. Tokyo phones are in a CSS that is not in the partition of the route pattern to call from Tokyo to Chicago.

Answer: (SHOW ANSWER)

NEW QUESTION: 52

Which two fields are required parameters when manually creating users on Cisco Unity Connection with predefined templates? (Choose two.)

A. first name and last name

B. extension

C. username (alias)

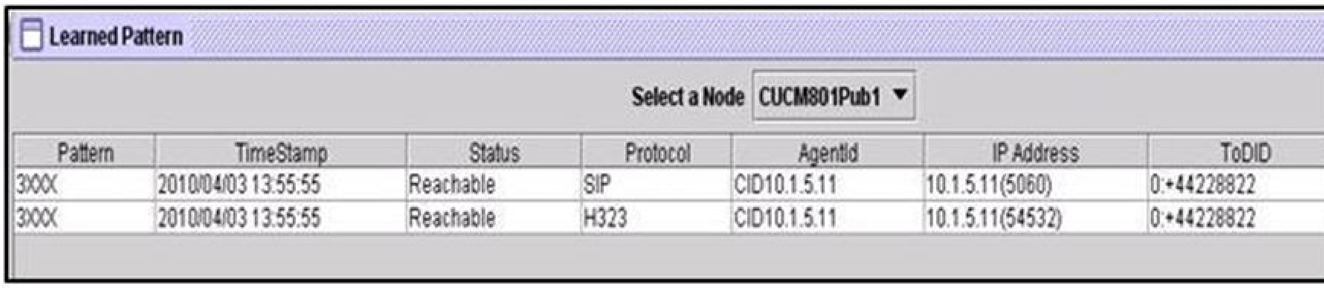
D. title

E. employee ID

Answer: (SHOW ANSWER)

NEW QUESTION: 53

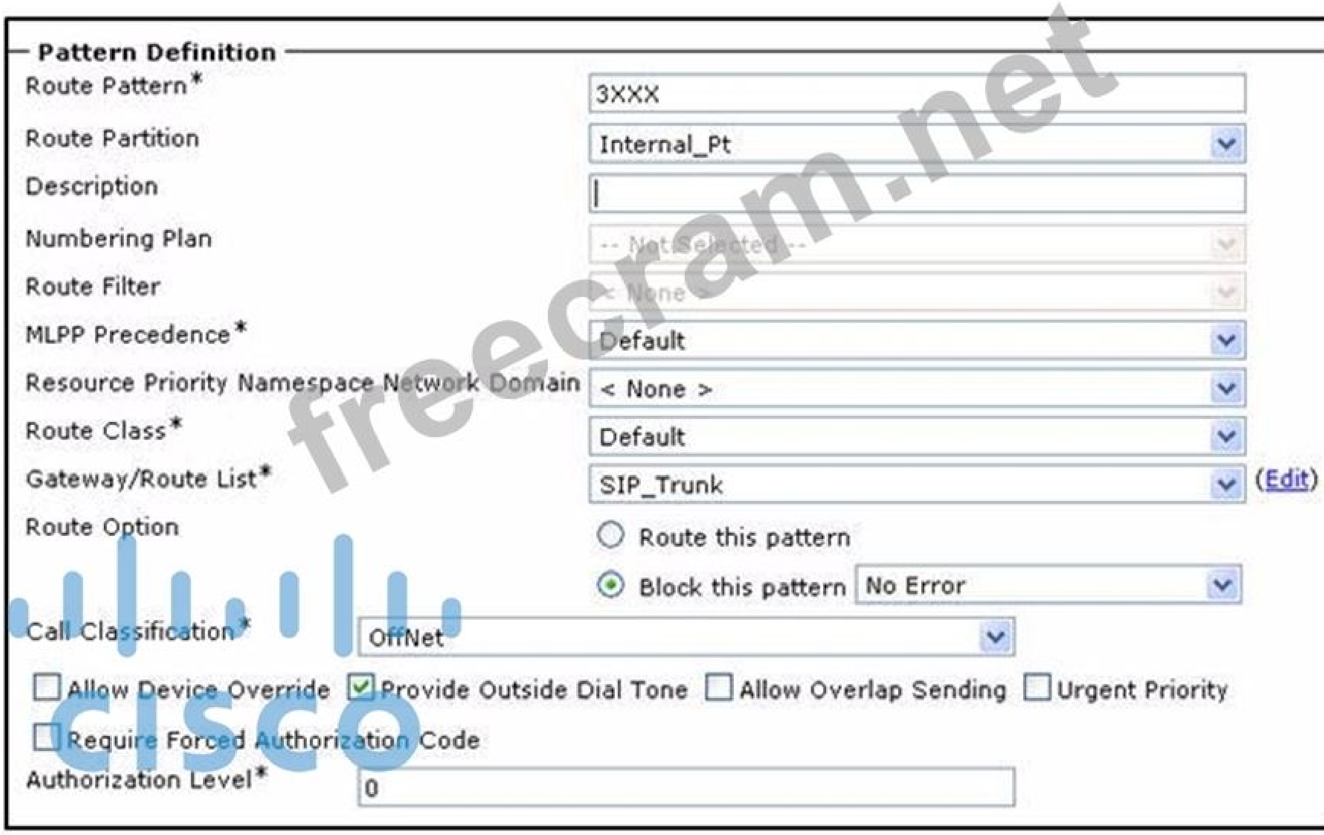
Refer to the exhibits.



Learned Pattern

Select a Node CUCM801Pub1

Pattern	TimeStamp	Status	Protocol	AgentId	IP Address	ToDID
3XXX	2010/04/03 13:55:55	Reachable	SIP	CID10.1.5.11	10.1.5.11(5060)	0:+44228822
3XXX	2010/04/03 13:55:55	Reachable	H323	CID10.1.5.11	10.1.5.11(54532)	0:+44228822



Assume that all learned SAF routes are placed in the SAF_Pt partition. An IP phone CSS contains the following partitions in this order: Internal_Pt, SAF_Pt. When the IP phone places a call to 3001, what will occur?

A. The call will fail because it will be blocked by the route pattern.

B. The call will be placed in a round-robin fashion between the SAF network and SIP_Trunk. Every other call will fail.

C. The call will be placed in a round-robin fashion between the SAF network and SIP_Trunk.

D. The call will succeed and will be placed via the SAF network. SAF-learned routes always take precedence.

Answer: (SHOW ANSWER)

NEW QUESTION: 54

ACME Corporation is deploying a new voice gateway. The network administrator is trying to configure the ISDN PRI T1 cards. However the router does not accept the controller t1

0 /0/0 command. The administrator sees this error message: "% Invalid input detected at '^' marker." To diagnose this problem, which two commands should the administrator use? (Choose two.)

A. Use the feature activate isdn command

B. Use the clock source line command

C. Use the show inventory command

D. Use the card type t1 command

E. Use the show call active voice brief command

Answer: (SHOW ANSWER)

NEW QUESTION: 55

In a Cisco Unified Communications Manager (CUCM) cluster, you receive a Database Communication Error when you attempt to login to the Publisher server. You need to resolve this issue.

Which two solutions should you try to resolve this issue? (Choose two.)

A. Restart the DNS service in the cluster.

B. Restart the DBL service on the publisher.

C. Modify the hostname for all Subscriber nodes in the cluster.

D. Revert the changes for hostname or IP address on the Publisher server.

E. Verify and ensure that the replication is working on all nodes in the cluster.

F. Reboot all nodes in the cluster.

Answer: (SHOW ANSWER)

NEW QUESTION: 56

You recently configured a system for B2B SIP URI calls, and users confirmed that they could make calls. You are receiving multiple reports that inbound calls are failing and that users are not receiving calls to their URI. You confirm that all zones between expressways are active, and the trunk between Cisco Unified Communications Manager and Cisco VCS Expressway is active. You also see that the inbound call is sent from Cisco VCS Expressway C to Cisco Unified Communications Manager. Why are the calls failing?

A. The certificate is not valid.

B. The cluster FQDN was not set in Enterprise Parameters.

C. The FQDN was not registered in DNS.

D. The Cisco Unified Communications Manager FQDN was not set.

E. The cluster FQDN was not set in Service Parameters.

Answer: (SHOW ANSWER)

NEW QUESTION: 57

You have been presented with a trouble ticket from an end user who works at a remote location that is served by a Cisco Unified Communications Manager Express. The user reports being unable to place calls to international numbers, but all other calls work properly and other users at this location can place international calls. Which two troubleshooting techniques would be helpful in resolving this issue? (Choose two.)

A. Cisco IOS debug tools

B. Class of Restriction baseline configuration for the user on Cisco Unified Communications Manager Express

C. show output of the voice translation rules in the voice gateway

D. show output for the T1 controller and voice port configuration in the voice gateway

E. show output of the ephone and ephone-dn configurations

Answer: (SHOW ANSWER)

NEW QUESTION: 58

Refer to the exhibit, which displays the output of the debug ccsip messages command on a Cisco router.

```
036294: Jun 26 13:46:07.936: //32672/70DBC0B09D1A/SIP/Call/sipSPIMediaCallInfo:
Number of Media Streams: 1
Media Stream              : 1
Negotiated Codec          : g711ulaw
Negotiated Codec Bytes    : 160
Nego. Codec payload       : 0 (tx), 0 (rx)
Negotiated Dtmf-relay     : 6
Dtmf-relay Payload        : 101 (tx), 101 (rx)
Source IP Address (Media): 192.168.84.5
Source IP Port    (Media): 16540
Destn  IP Address (Media): 192.168.84.12
Destn  IP Port    (Media): 31002
Orig Destn IP Address:Port (Media): [ - ]:0
```

What is the negotiated dual tone multifrequency on this call?

A. h245-signal

B. h245-alphanumeric

C. sip-notify

D. sip-kpml

E. rtp-nte

F. cisco-rtp

Answer: (SHOW ANSWER)

NEW QUESTION: 59

You must integrate a third-party H.323 system with your existing Cisco Unified Communications Manager cluster. When you create an H.323 trunk from the cluster, calls from the cluster to the third-party H.323 system are failing. The vendor of the third-party
H.323 device has confirmed that the H.323 call setup time must be reduced. Which two approaches reduce the call setup time from Cisco Unified Communications Manager to the third-party H.323 device? (Choose two.)

A. Implement transcoding with the Cisco Unified Communications Manager resources.

B. Implement transcoding with the router DSP resources.

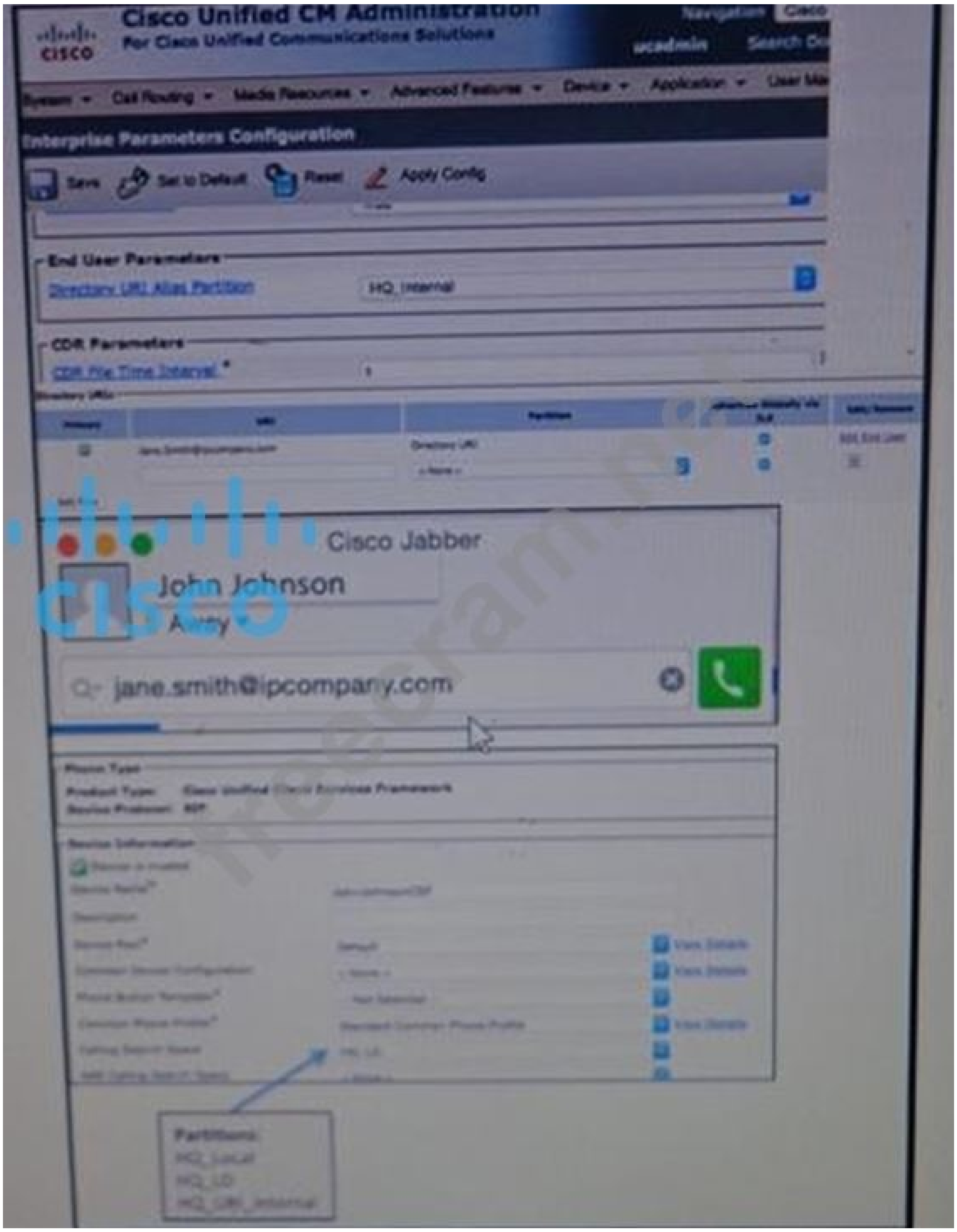
C. Implement a hardware MTP.

D. Implement a software MTP.

Answer: (SHOW ANSWER)

NEW QUESTION: 60

John Johnson is trying to leach Jane Smith at net Cisco Untied IP phone by dialing net directory URI, but the call fails. What are two possible reasons for the call failure? (Choose two) Refer to the exhibit.



A. Cisco Unified Communications Manager is blocking the call.

B. The CSS that is used by John's Cisco Jabber device does not contain the partition to teach Jane's endpoint.

C. Jane's endpoint does not support blended addressing.

D. The dialed directory URI and the provisioned directory URI do not match.

E. The directory URI should not be entered under the user configuration page Instead, the directory URI should be entered directly into Jane's line configuration page.
F. Cisco Unified Communions Manager matched the host portion of Jane's URI to a SIP trunk and fouled the call to another cluster.
Answer: (SHOW ANSWER)

NEW QUESTION: 61
In Cisco Unified Communications Manager, what is the default maximum number of learned patterns for the call control discovery feature parameter?
A. 500
B. 50000
C. 20000
D. 5000
E. 10000
Answer: (SHOW ANSWER)

NEW QUESTION: 62
You have an endpoint registration problem with VCS, and the event log reason of "unknown domain". The domain names that your endpoints are using to register with must be added to this list. Where do you check the list of defined domains?
A. VCS configuration > Protocols > SIP > Domains
B. VCS configuration > Domains > SIP
C. VCS configuration > Protocols > SCCP > Domains
D. VCS Domains > Protocols > SIP
Answer: (SHOW ANSWER)

NEW QUESTION: 63
Phone A is able to dial the directory number of Phone B and complete a call. However, when Phone B dials the directory number of Phone A, Phone B receives a fast-busy tone.
What is causing this issue?
A. Phone B is not registered.
B. Phone A and Phone B are in different partitions.

C. Phone B has the incorrect CSS to dial Phone A.
D. Phone B does not have the CSS of Phone A in its partition.
E. Phone B is not in the CSS of Phone A.
F. Phone B does not have Phone A in its partition.
Answer: (SHOW ANSWER)

NEW QUESTION: 64
After you deploy cisco Unified communications Manager Device Mobility across a VPN connection with Cisco Unified IP phones, users in remote locations report one-way audio issues.
Which two actions can you take to locate the problem? (Choose two.)
A. Verify that users at the remote locations are connecting to the closest enterprise VPN concentrator.
B. Verify that the firewall is allowing RTP traffic flow.
C. Verify that the VLANs at remote locations are configured correctly.
D. Verify that the Cisco IOS devices on the VPN support audio connections.
E. Verify that DMI is configured with the correct IP subnets.
Answer: (SHOW ANSWER)

NEW QUESTION: 65
You have configured a new MGCP gateway on your Cisco Unified Communications Manager servers but the endpoints are shown with status "None".
What are two possible solutions for this registration issue? (Choose two)
A. Configure the voice-VLAN interface of the MGCP gateway with mgcp-gateway voip interface command to set the source IP address of the MGCP packets correctly.
B. Ensure that you have configured the mgcp call-agent command with the Cisco Unified Communications Manager server address or hostname.
C. Ensure that you have configured the ccm-manager call-agent command with the correct Cisco Unified Communications Manager server address or hostname.
D. Enable the endpoint voice interface for backhauling in interface configuration mode with the isdn bind-l3 ccm-manager command
E. Activate the MGCP Feature service on all call-processing servers in the cluster.
Answer: (SHOW ANSWER)

NEW QUESTION: 66
Of the following persistent settings for Cisco TMS-controlled endpoints, TMS overwrites these settings if which five of them are altered on the endpoint? (Choose five.)
A. Configuration Template
B. IEEE 802.1x Authentication Password
C. System Name
D. System Contact
E. SIP URI

F. Active Cisco Unified Communications Manager Address
G. H.323 ID
H. E.164 alias
Answer: (SHOW ANSWER)

NEW QUESTION: 67
What is the maximum number of characters that you can use in Service Advertisement Framework (SAF) forwarder name?
A. 50
B. 20
C. 40
D. 30
E. 60
F. 10
Answer: (SHOW ANSWER)

NEW QUESTION: 68
Refer to the exhibit.


Which course of action will resolve the Mobile Connect issues that are shown in the exhibit?

A. Make the user an owner of the phone device in the phone device configuration page.
B. Enable the user for Cisco Mobile Connect.
C. Configure the Mobility softkey on the phone.
D. Enable the device mobility mode on the phone since it is disabled.
Answer: (SHOW ANSWER)

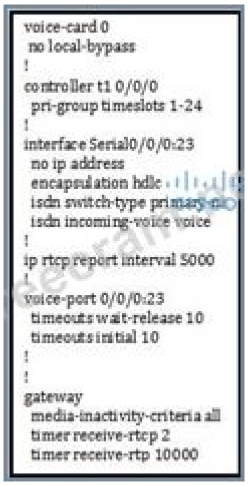
NEW QUESTION: 69
Which two protocols are required for a Cisco 8800 series IP phone to successfully register with CUCM? (Choose two)
A. SCCP
B. DHCP
C. SIP
D. LLDP
E. TFTP
F. CDP
G. HTTP
Answer: (SHOW ANSWER)

NEW QUESTION: 70
You have 50 hardware MTP resources and 200 software MTP resources. You want to use hardware resources first, but software is being used first. Where can you confirm the MTP selection order?

A. Cisco Unified Real-Time Monitoring Tool
B. MTP list
C. Media Resource Group List
D. MGCP gateway
E. calling search space
F. phone device pool
Answer: (SHOW ANSWER)

NEW QUESTION: 71
In a SIP direct call setup, the originating UAC sends what message to the UAS of the recipient?
A. OK
B. INVITE
C. RINGING
D. ACK
Answer: (SHOW ANSWER)

NEW QUESTION: 72
Refer to the exhibit.



```
voice-card 0
 no local-bypass
!
controller t1 0/0/0
 pri-group timeslots 1-24
!
interface Serial0/0/0:23
 no ip address
 encapsulation hdlc
 isdn switch-type primary-ni
 isdn incoming-voice voice
!
ip rtcp report interval 5000
!
voice-port 0/0/0:23
 timeouts wait-release 10
 timeouts initial 10
!
!
gateway
 media-inactivity-criteria all
 timer receive-rtcp 2
 timer receive-rtp 10000
```

Users are reporting that inbound calls from the PSTN are dropping when not answered within 10 seconds. Calls come in via ISDN T1 PRI. Which configuration change is needed to prevent the calls from dropping?

A. Modify the Call Forward No Answer setting in CUCM to redirect calls to Voicemail or another extension.

B. Remove the timeouts initial 10 command from under the voice-port.

C. Remove the timer receive-rtp 10000 command from under the gateway.

D. Remove the timeouts wait-release 10 command from under the voice-port.

E. Remove the timer receive-rtcp 2 command from under the gateway.

Answer: (SHOW ANSWER)

NEW QUESTION: 73

Upon completion of a failover task you notice that all the CTI devices are still pointing to the backup server rather than the primary server. How should you resolve this issue?

A. Stop the CTI Manager Service on the Secondary server so that all devices using CTI register with the primary.

B. Restart the subscriber with the CTI Manager Service on the secondary.

C. Do nothing and wait for the next failover assuming this will revert back to the primary CTI server.

D. Stop the CTI Manager Service on the Primary server so that all devices using CTI stay registered with the primary CTI server.

Answer: (SHOW ANSWER)

NEW QUESTION: 74

Which command is used on an IOS Router that is acting as a SAF Forwarder to confirm its registration status with a SAF Client?

A. show ip asf-forwarder status details

B. show ospf neighbor details

C. show service-family asf-forwarder details

D. show ip interface details

E. show eigrp service-family ipv4 clients details

F. show cdp neighbor details
Answer: (SHOW ANSWER)

NEW QUESTION: 75
When the command utils dbreplication status is executed on the Cisco Unified Communications Manager CLI, which step should be taken next to check the database replication status?
A. The command utils dbreplication runtimestate must be run on the subscriber.
B. The command utils dbreplication runtimestate must be run on the publisher.
C. Restart the Cisco CallManager service.
D. Run the same command on all nodes of the cluster.
E. View the activelog file.
Answer: (SHOW ANSWER)

NEW QUESTION: 76
Which Cisco Unified Communications Manager troubleshooting tool can be used to determine the digit manipulation path a call takes within the Cisco Unified Communications Manager system from the perspective of a specific directory number, without having the actual device at hand?
A. Cisco Unified Communications Manager Real Time Monitoring Tool
B. Cisco IOS debugs
C. Cisco Unified Syslog Viewer
D. Cisco Unified Communications Manager Dialed Number Analyzer
E. Cisco Unified Communications Manager Serviceability
Answer: (SHOW ANSWER)

NEW QUESTION: 77
When a remote endpoint dials in to join a conference that is configured on a Cisco TelePresence Server bridge, the endpoint receives only audio. Other users can successfully join the call with Voice and Video. What is causing this issue?
A. The bridge is out of all licenses.
B. The bridge is not able to host video calls.
C. The endpoint is assigned a region without enough configured bandwidth for video.
D. The endpoint does not have the partition of the bridge in its CSS.

E. The endpoint does not have the multisite option installed.
Answer: (SHOW ANSWER)

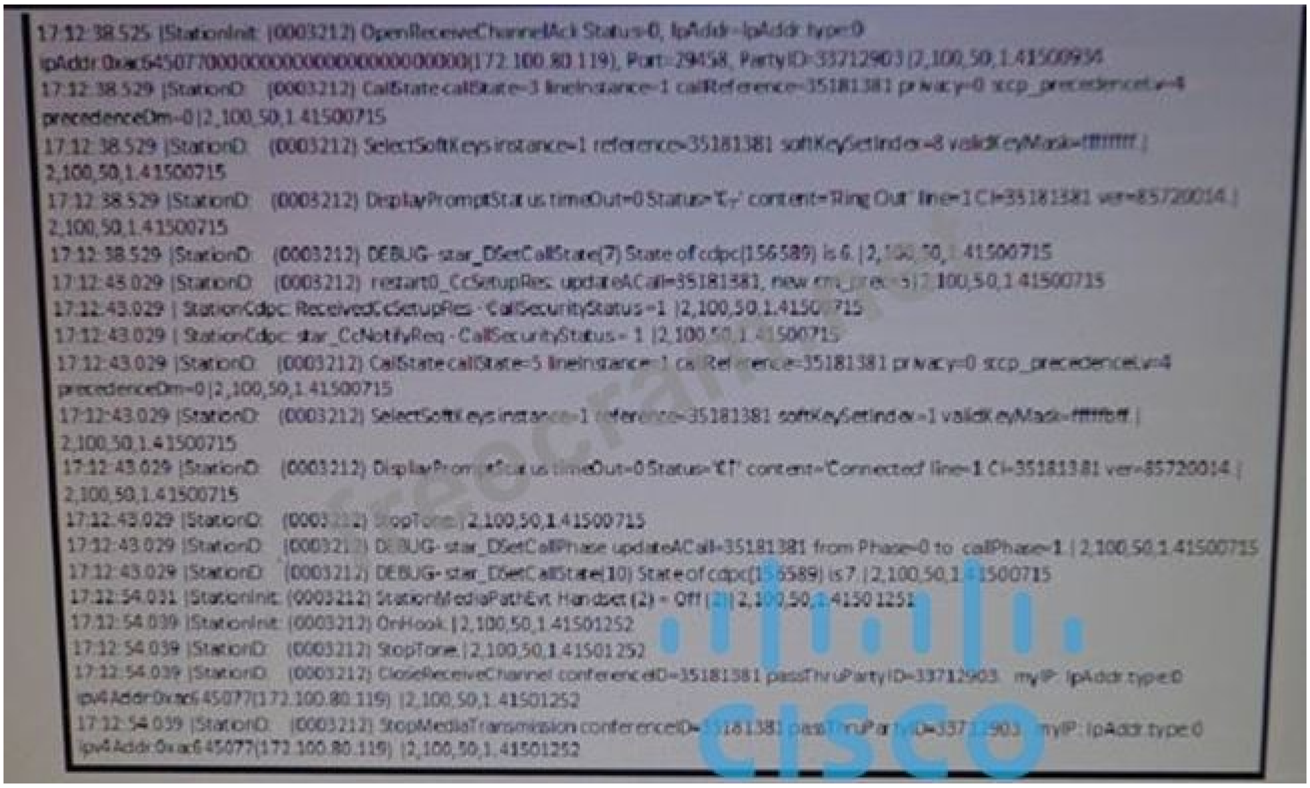
NEW QUESTION: 78
When a user attempts to log out from Cisco Extension Mobility service by pressing the services button and selecting the Cisco Extension Mobility service, the user is not able to log out. What is causing this issue?
A. The logout URL that is defined for the Cisco Extension Mobility service is incorrect or does not exist under the IP Phone Services configuration.
B. The Cisco Extension Mobility service has not been configured on the phone.
C. The CTI service is not running.
D. The user device profile is not subscribed to the Cisco Extension Mobility service.
Answer: (SHOW ANSWER)

NEW QUESTION: 79
You are troubleshooting video quality issues on a Cisco TelePresence TX9000 Series system. Which CLI command shows the total number of lost video packets and the received jitter during a call in progress?
A. show call statistics all
B. show call statistics video detail
C. show call statistics video
D. show call statistics detail
Answer: (SHOW ANSWER)

NEW QUESTION: 80
Which call control model does MGCP use?
A. Ad hoc
B. Distributed
C. Centralized
D. Hybrid
Answer: (SHOW ANSWER)

NEW QUESTION: 81
After you deploy cisco Unified Commutations Manager Device Mobility in your environment, you note that phones in a remote site are failing to register. Which two actions correct the problem? (Choose two.)
A. Verify that the phones are attempting to register with the correct mega cluster in the device pool of the central office.
B. Verify that the remote site is assigned to the device mobility group that matches its dialing pattern.
C. Verify that extension mobility is configured correctly for the remote site.

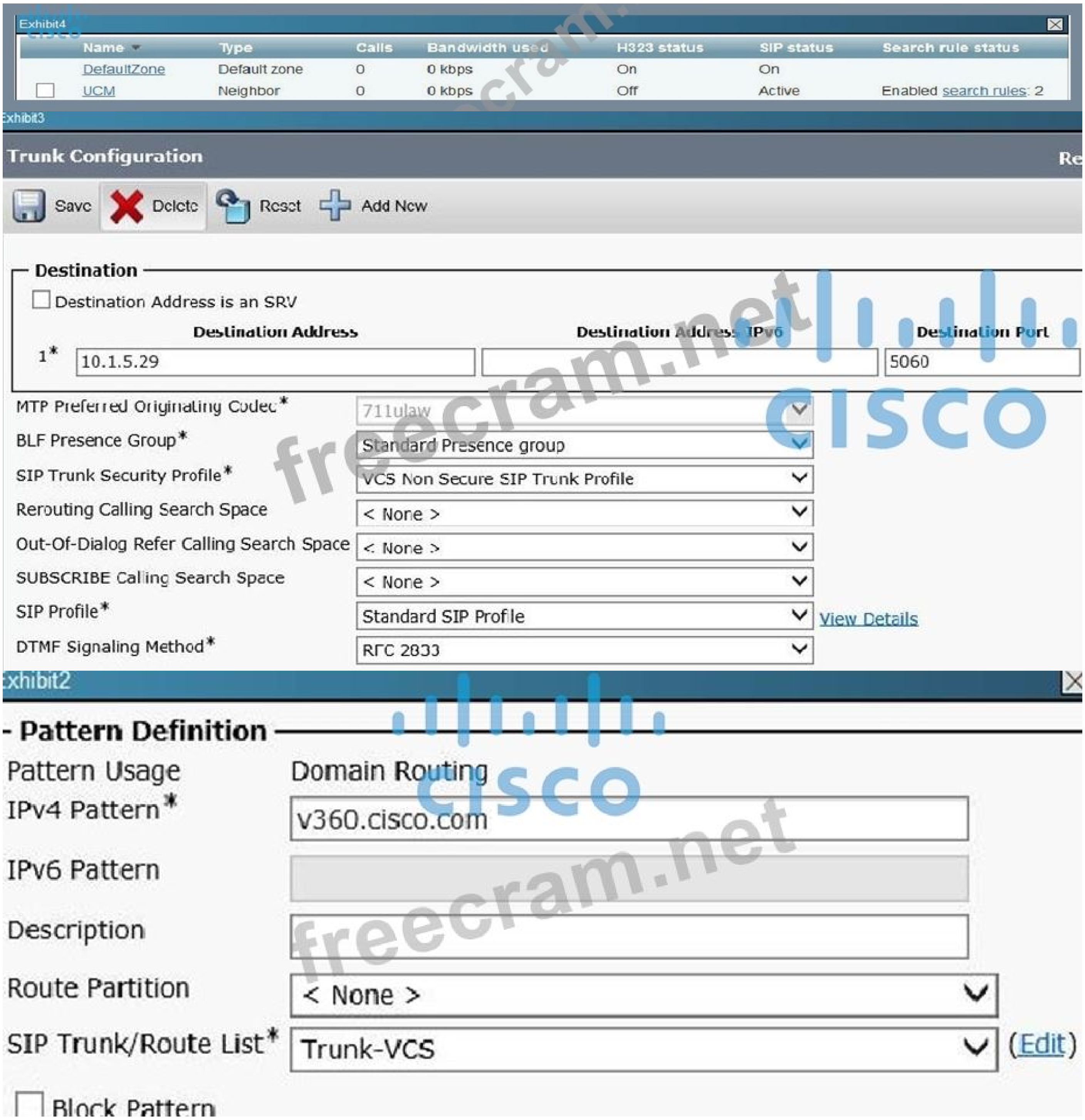
D. verify that the subnet of the remote site is configured in device mobility info.
E. verify that the LDAP server is configured to support device Mobility at the remote site.
Answer: (SHOW ANSWER)

NEW QUESTION: 82
What does this trace indicate?
Refer to the exhibit.

```
17:12:38.525 |StationInit: (0003212) OpenReceiveChannelAck Status=0, IpAddr=IpAddr type=0
ipAddr:0xac64507700000000000000000000000(172.100.80.119), Port=29458, PartyID=33712903 |2,100,50,1.41500934
17:12:38.529 |StationD:   (0003212) CallState callState=3 lineInstance=1 callReference=35181381 privacy=0 sccp_precedenceLv=4
precedenceDm=0 |2,100,50,1.41500715
17:12:38.529 |StationD:   (0003212) SelectSoftKeys instance=1 reference=35181381 softKeySetIndex=8 validKeyMask=ffffffff |
2,100,50,1.41500715
17:12:38.529 |StationD:   (0003212) DisplayPromptStatus timeOut=0 Status='Ring Out' content='Ring Out' line=1 CI=35181381 ver=85720014 |
2,100,50,1.41500715
17:12:38.529 |StationD:   (0003212) DEBUG- star_DSetCallState(7) State of cdpc(156589) is 6 |2,100,50,1.41500715
17:12:43.029 |StationD:   (0003212) restart0_CcSetupRes: updateACall=35181381, new cm_prec=5|2,100,50,1.41500715
17:12:43.029 | StationCdpc: ReceivedCcSetupRes - CallSecurityStatus=1 |2,100,50,1.41500715
17:12:43.029 | StationCdpc: star_CcNotifyReq - CallSecurityStatus= 1 |2,100,50,1.41500715
17:12:43.029 |StationD:   (0003212) CallState callState=5 lineInstance=1 callReference=35181381 privacy=0 sccp_precedenceLv=4
precedenceDm=0 |2,100,50,1.41500715
17:12:43.029 |StationD:   (0003212) SelectSoftKeys instance=1 reference=35181381 softKeySetIndex=1 validKeyMask=ffffbff |
2,100,50,1.41500715
17:12:43.029 |StationD:   (0003212) DisplayPromptStatus timeOut=0 Status='Connected' content='Connected' line=1 CI=35181381 ver=85720014 |
2,100,50,1.41500715
17:12:43.029 |StationD:   (0003212) StopTone |2,100,50,1.41500715
17:12:43.029 |StationD:   (0003212) DEBUG- star_DSetCallPhase updateACall=35181381 from Phase=0 to callPhase=1 |2,100,50,1.41500715
17:12:43.029 |StationD:   (0003212) DEBUG- star_DSetCallState(10) State of cdpc(156589) is 7 |2,100,50,1.41500715
17:12:54.031 |StationInit: (0003212) StationMediaPathEvt Handset (2) = Off (2)|2,100,50,1.41501251
17:12:54.039 |StationInit: (0003212) OnHook |2,100,50,1.41501252
17:12:54.039 |StationD:   (0003212) StopTone |2,100,50,1.41501252
17:12:54.039 |StationD:   (0003212) CloseReceiveChannel conferenceID=35181381 passThruPartyID=33712903 myIP: IpAddr type:0
ipv4Addr:0xac645077(172.100.80.119) |2,100,50,1.41501252
17:12:54.039 |StationD:   (0003212) StopMediaTransmission conferenceID=35181381 passThruPartyID=33712903 myIP: IpAddr type:0
ipv4Addr:0xac645077(172.100.80.119) |2,100,50,1.41501252
```
A. a disconnected call
B. an interrupted call setup
C. a failed call setup
D. a completed call setup
Answer: (SHOW ANSWER)

NEW QUESTION: 83
When viewing the configuration of Intercluster Lookup Service, this command is used to look up the URI of a remote cluster user:
Admin:utils ils lookup user1@acme.com
Routing string is cucm1.cisco.com
Which configuration is required on Cisco Unified Communications Manager, in order to allow routing calls to the URI of this remote cluster user?
A. SIP trunk with destination cucm1.cisco.com
B. SIP route pattern for domain*cisco.com
C. SIP route pattern for domain*.acme.com
D. end user with directory URI user1@acme.com
Answer: (SHOW ANSWER)

NEW QUESTION: 84

Refer to topology and Exhibits below:



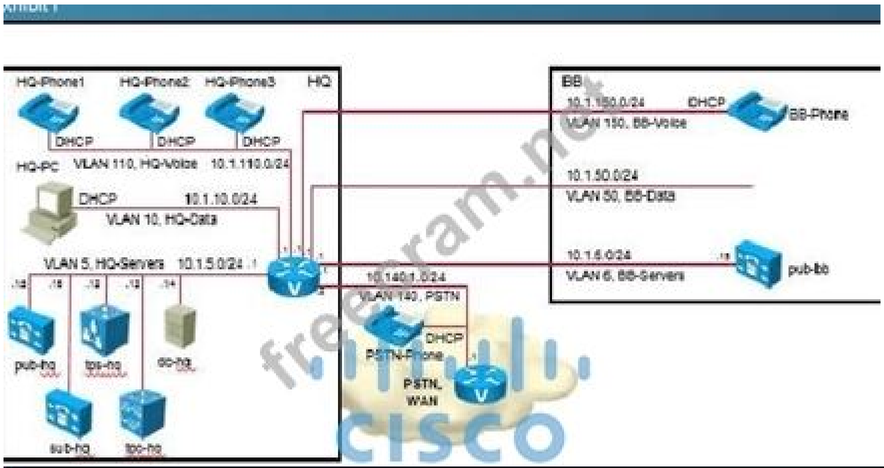


Exhibit5

tvcs: Event="Call Rejected" Service="SIP" Src-ip="10.1.5.25" Src-port="5060" Src-alias-type="SIP" Src-alias="sip:2001@10.1.5.25" Dst-alias-type="SIP" Dst-alias="sip:sx20-p16@v360.cisco.com" Call-serial-number="9183df89-ebd2-46c6-86ed-bbb6c2f68b43" Tag="3ff200fd-8a4e-4260-a7a1-65f6a5f86ca6" Detail="Not Found" Protocol="TCP" Response-code="404" Level="1" UTCTime="2015-02-12 21:22:47.131"

Exhibit6

	Priority	Rule name	Protocol	Source	Authentication required	Mode	Pattern type	Pattern string	Pattern behavior	On match	Target	State	Actions
☐	50	LocalZoneMatch	Any	Any	No	Any alias				Continue	LocalZone	Disabled	View/Edit | Clone
☐	100	UCM	Any	Any	No	Any alias				Continue	UCM	Enabled	View/Edit | Clone
☐	100	UCM2	SIP	Any	No	Alias pattern match	Regex	2...	Leave	Stop	UCM	Enabled	View/Edit | Clone

Exhibit7

tvcs: Event="Call Disconnected" Service="SIP" Src-ip="10.1.150.11" Src-port="5061" Src-alias-type="SIP" Src-alias="sip:sx20-p16@v360.cisco.com" Dst-alias-type="SIP" Dst-alias="sip:2001@v360.cisco.com" Call-serial-number="3045badb-7615-4a8a-b64a-98ad237497fc" Tag="2a8d5d57-8f8f-4e37-a817-7d4160affa3b" Protocol="TLS" Level="1" UTCTime="2015-02-12 21:37:29.082"

tvcs: Event="Call Disconnected" Service="SIP" Src-ip="10.1.5.29" Src-port="5073" Src-alias-type="SIP" Src-alias="sip:sx20-p16@v360.cisco.com" Dst-alias-type="SIP" Dst-alias="sip:2001@v360.cisco.com" Call-serial-number="280ddd00-3d67-4482-adb4-2c2587061def" Tag="2a8d5d57-8f8f-4e37-a817-7d4160affa3b" Protocol="TLS" Level="1" UTCTime="2015-02-12 21:37:28.614"

tvcs: Event="Search Completed" Service="SIP" Src-alias-type="SIP" Src-alias="sx20-p16@v360.cisco.com" Dst-alias-type="SIP" Dst-alias="sip:2001@v360.cisco.com" Call-serial-number="3045badb-7615-4a8a-b64a-98ad237497fc" Tag="2a8d5d57-8f8f-4e37-a817-7d4160affa3b" Detail="found:true, searchtype:INVITE" Call-routed="YES" Level="1" UTCTime="2015-02-12 21:37:18.592"

tvcs: Event="Call Connected" Service="SIP" Src-ip="10.1.150.11" Src-port="5061" Src-alias-type="SIP" Src-alias="sip:sx20-p16@v360.cisco.com" Dst-alias-type="SIP" Dst-alias="sip:2001@v360.cisco.com" Call-serial-number="3045badb-7615-4a8a-b64a-98ad237497fc" Tag="2a8d5d57-8f8f-4e37-a817-7d4160affa3b" Protocol="TLS" Call-routed="YES" Level="1" UTCTime="2015-02-12 21:37:18.592"

Exhibit8

	Priority	Rule name	Protocol	Source	Authentication required	Mode	Pattern type	Pattern string	Pattern behavior	On match	Target	State	Actions
☐	50	LocalZoneMatch	Any	Any	No	Any alias				Continue	LocalZone	Enabled	View/Edit | Clone
☐	100	UCM	Any	Any	No	Any alias				Continue	UCM	Enabled	View/Edit | Clone
☐	100	UCM2	SIP	Any	No	Alias pattern match	Regex	2...	Leave	Stop	UCM	Disabled	View/Edit | Clone

A call from HQ Phone 1 with the extension of 2001 is dialing a SX20 that is registered to the VCS in BackBone (not shown). Determine if the call fails and if so, what are the two causes? (Choose two.)

A. The call succeeds

B. The call fails

C. There are no issues, so the call succeeds.

D. The Local Zone Match Rule state is disabled.

E. The SIP port is incorrect on the Cisco Unified Communications Manager SIP trunk.

F. Rule name UCM2 is set to stop on Match.

Answer: (SHOW ANSWER)

NEW QUESTION: 85

Which issue would cause an MGCP gateway to fail to register with Cisco Unified Communications Manager?

A. misconfigured route group in Cisco Unified Communications Manager

B. mismatched domain name on the MGCP gateway and Cisco Unified Communications Manager gateway configuration

C. incorrect MGCP IP address specified in the gateway configuration in Cisco Unified Communications Manager

D. missing the configuration command isdn bind-l3 ccm-manager under the ISDN interface

Answer: (SHOW ANSWER)

NEW QUESTION: 86

In Cisco Unified Communications Manager (CUCM), the primary Service Advertisement Framework (SAF) forwarder is deleted from the database.

If there were total of three SAF forwarders configured, which two will happen next? (Choose two.)

A. Through an election, the backup and the third SAF forwarder will be designated as primary SAF forwarder.

B. The third SAF forwarder will become the primary SAF forwarder.

C. The third SAF forwarder will become the backup SAF forwarder.

D. You need to desinate the backup SAF forwarder as the primary SAF forwarder.

E. The backup SAF forwarder will automatically become the primary SAF forwarder.

F. CUCM will wait for the primary SAF forwarder to come alive.

Answer: (SHOW ANSWER)

NEW QUESTION: 87

A user is dialing an external PSTN number with a prefix of 01 from a Cisco TelePresence SX10 Quick Set in a Cisco VCS environment. In the past, the Cisco VCS and the ISDN gateway were correctly configured with a prefix of 01, but the calls are now failing. What are three possible causes? (Choose three.)

A. ISDN is not enabled on the Cisco TelePresence SX10.

B. The SIP trunk is not configured on the gateway.

C. The Cisco TelePresence SX10 is not registered to the Cisco VCS Control.

D. The audio feature in the Cisco TelePresence SX10 is turned off.

E. The Cisco TelePresence SX10 is not registered to the Cisco Express C.

F. 01 is not a valid prefix.

G. The Cisco VCS Control is down.

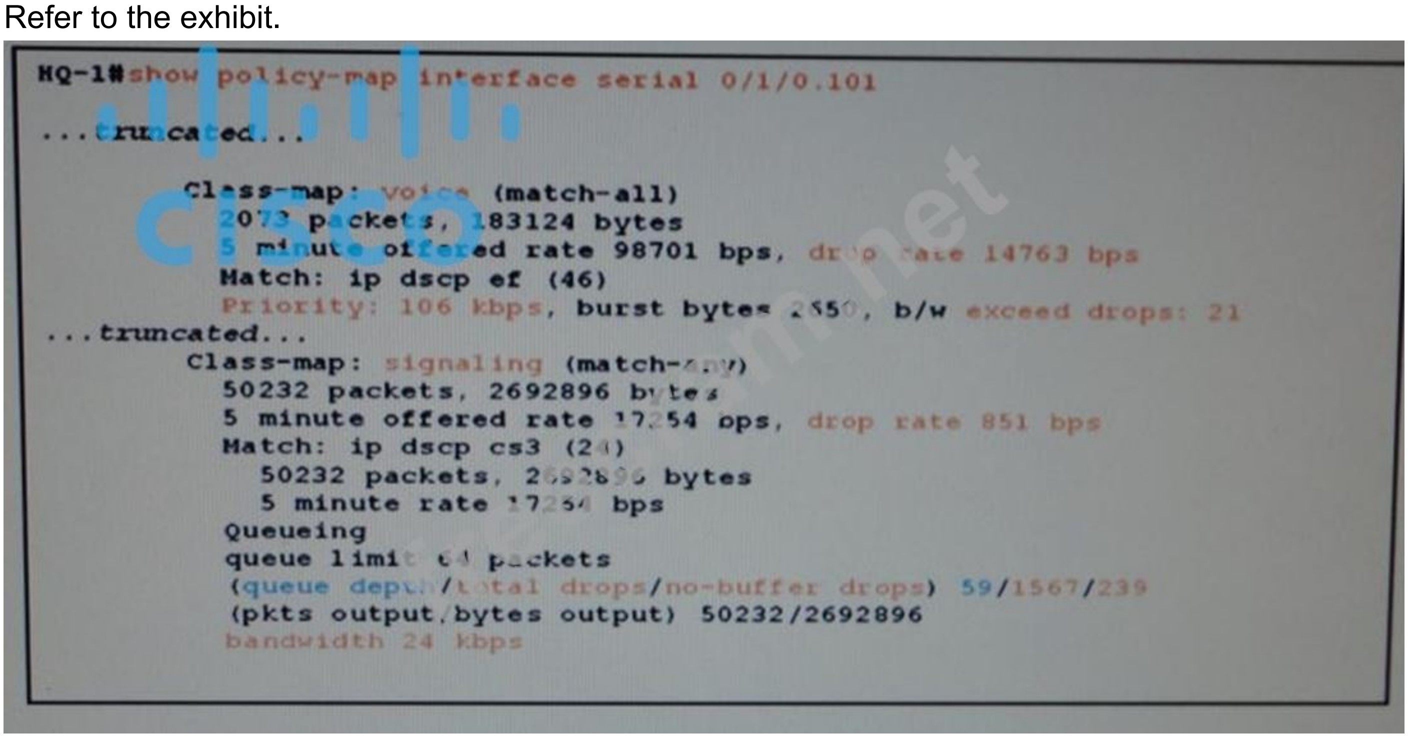
H. The interworking setting is turned off.

Answer: (SHOW ANSWER)

NEW QUESTION: 88

In a Cisco UCM multisite WAN with centralized call-processing deployment model, what redundancy feature should be configured on remote site routers to supply basic IP telephony services in the event of a WAN outage?
A. CAC
B. SRST
C. V3PN
D. AAR
Answer: (SHOW ANSWER)

NEW QUESTION: 89
You configured a Cisco ISR G2 as a SIP gateway, but the gateway does not show that it is registered with Cisco Unified Communications Manager. What is causing this issue?
A. Cisco Unified Communications Manager does not show a SIP gateway as registered if it is not properly configured.
B. Cisco Unified Communications Manager does not support SIP gateways.
C. Cisco Unified Communications Manager never shows a SIP gateway as registered even when it is properly configured.
D. The Cisco ISR G2 cannot be a SIP gateway.
E. The gateway does not have Cisco Unified Border Element session licensing.
F. The gateway does not have the UC license installed.
Answer: (SHOW ANSWER)

NEW QUESTION: 90
Refer to the exhibit.


Assume a centralized Cisco Unified Communications Manager topology. From the output of the show policy-map interface command, it can be seen that the voice traffic is experiencing drops.

The router is configured for low latency queuing and is located at the central site. Which course of action would rectify this issue?
A. Configure a higher rate bandwidth codec on all voice calls
B. Configure Call Admission Control to limit the maximum calls that are sent to the priority queue.
C. Place the signaling traffic in the priority queue.
D. split some of the excess voice traffic into other queues K avoid oversubscription of the priority queue.
Answer: (SHOW ANSWER)
Explanation: Link- http://www.iphelp.ru/faq/27/ch07lev1sec3.html

NEW QUESTION: 91
Which three statements about SAF forwarding are true? (Choose three.)
A. It requires the SAF Forwarders to use the same configuration as the IOS switch.
B. It is supported on IPv4 only.
C. It requires a unique IP address for each SAF Forwarder.
D. It is supported on IPv6 only.
E. It supports DNS and IP addresses.
F. The SAF Forwarder configuration can be updated without connecting to Cisco Unified Communications Manager.
Answer: (SHOW ANSWER)

NEW QUESTION: 92
During a business-to-business video call through the Cisco Expressway solution, the internal endpoint can call out to the remote endpoint on the Internet, but it does not receive audio or video. The remote endpoint receives both audio and video. What is causing the issue?
A. The firewall is blocking inbound RTP ports.
B. The Cisco Expressway does not have a Rich Media Session license.
C. The Cisco Unified Communications Manager is not configured for business-to-business calling.
D. The firewall is blocking SIP signaling.
E. The Advanced Networking option is not installed on the Expressway Edge.
Answer: (SHOW ANSWER)

NEW QUESTION: 93
Which Cisco IOS command is used to display information for active connections that are controlled by the Media Gateway Control Protocol?
A. show mgcp statistics
B. show mgcp endpoint
C. show mgcp nas
D. show mgcp connection
Answer: (SHOW ANSWER)

NEW QUESTION: 94
Which four performance counters are available when monitoring a Cisco MTP device using the Cisco Unified Communications Manager RTMT? (Choose four.)
A. Out of Resources
B. Resource Active
C. Resource Total
D. MTP Instances Active
E. MTP Streams Active
F. Resource Available
G. Resource Idle
H. MTP Connection Lost
Answer: A,B,C,F (LEAVE A REPLY)

NEW QUESTION: 95
Which command do you use to confirm that a router interlace is enabled for SAP?
A. show run
B. show eigrp service -family interface
C. show eigrp service-family ipv4 client details.
D. show ip saf-service-family interface
E. show eigrp service-family Ipv4 <AS number interfaces
F. show ip interface details
Answer: (SHOW ANSWER)

NEW QUESTION: 96
Cisco Unified Mobile Connect has been enabled, but users are not able to switch an in- progress call from their mobile phone to their desk phone. You find out that the Resume softkey option does not appear on the desk phone after users hang up the call on their mobile phone. What do you do to resolve this issue?
A. Issue the voice call disc-pi-off command in the gateway.
B. Enable mobile connect on the user profile.
C. Assign Resume softkey on the desk phone.
D. Issue the progress_ind progress disable command in the gateway.

Answer: (SHOW ANSWER)

NEW QUESTION: 97

You are troubleshooting media issues during SIP calls on a C-series collaboration endpoint that is running TC6.3 software. You are asked to recreate the problem and to provide a packet capture from the endpoint. Which four of the following are required to accurately perform a packet capture on a C-series endpoint? (Choose four.)

A. Stop the capture by pressing the key combination Ctrl + C.

B. Start the call, and then enter the tcpdump command string to begin the capture.

C. Set the default transport to TCP.

D. Set up administrator access on the endpoint.

E. On the endpoint, set the default transport to TLS.

F. Set up root access on the endpoint.

G. Start the capture by entering the tcpdump command string and then start the call.

Answer: (SHOW ANSWER)

NEW QUESTION: 98

In a SAF deployment, the registration status looks correct and the learned patterns appear reachable, but calls are not routed. What is causing this issue?

A. network connection failure between the primary and backup SAF Forwarders

B. network connection failure between the SAF Forwarder and Cisco Unified Communications Manager

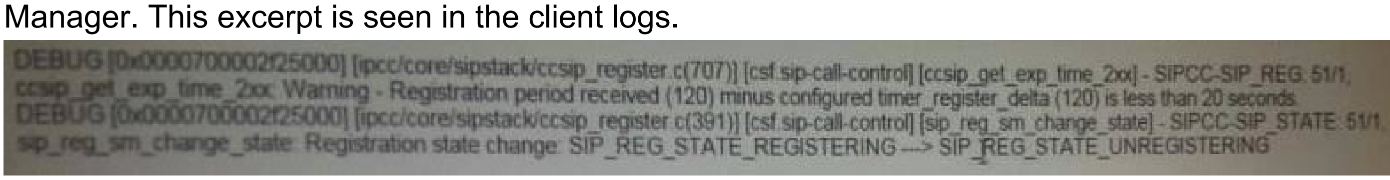
C. TCP connection failure with the backup SAF Forwarder

D. TCP connection failure with the primary SAF Forwarder

Answer: (SHOW ANSWER)

NEW QUESTION: 99

Jabber for Windows or MAC Soft Client fails to register with Cisco Unfed Communications Manager. This excerpt is seen in the client logs.


DEBUG [0x0000700002f25000] [ipcc/core/sipstack/ccsip_register.c(707)] [csf.sip-call-control] [ccsip_get_exp_time_2xx] - SIPCC-SIP_REG: 51/1, ccsip_get_exp_time_2xx: Warning - Registration period received (120) minus configured timer_register_delta (120) is less than 20 seconds
DEBUG [0x0000700002f25000] [ipcc/core/sipstack/ccsip_register.c(391)] [csf.sip-call-control] [sip_reg_sm_change_state] - SIPCC-SIP_STATE: 51/1, sip_reg_sm_change_state: Registration state change: SIP_REG_STATE_REGISTERING ---> SIP_REG_STATE_UNREGISTERING

Which configuration item needs to be adjusted to correct this condition?

A. "KeepAlive_Timer" parameter in jabber-config xml file

B. Unified CM Device Assigned SIP Profile "Timer Register Delta"

C. Unified CM Device Assigned SIP Profile "2XX Expiry Timer"

D. Unified CM Service Parameter "Station KeepAlive Interval"

Answer: (SHOW ANSWER)

NEW QUESTION: 100

When a caller dials 9 plus an external seven-digit number, the caller hears a fast-busy tone after a period of silence. What is causing the silence?

A. The T302 timer is waiting to expire.

B. To dial successfully, the caller must enter a Forced Authorization Code.

C. The gateway is not dropping the leading 9, and the PSTN fails.

D. The caller does not have the PSTN partition in the CSS.

E. There is no dial route for 9XXXXXXX on Cisco Unified Communications Manager.

F. The caller dialed the wrong number.

Answer: (SHOW ANSWER)