

300-080 Dumps

Troubleshooting Cisco IP Telephony and Video

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NEW QUESTION 1

- (Exam Topic 1)

Company XYZ reports that their SAF calls are being routed through the PSTN. Which cause of the issue is true?

- A. TCP Connection Failure has occurred between the SAF Forwarder and Cisco Unified CommunicationsManager.
- B. Cisco Unified Communications Managed did not increment the service number correctly.
- C. The maximum number of learned patterns have being reached.
- D. Cisco Unified Communications Manager did not construct the SAF message correctly.

Answer: A

NEW QUESTION 2

- (Exam Topic 1)

Users report intermittent call failures. All calls must invoke an IOS-based transcoder, registered to Cisco Unified Communications Manager, to be successful. Which two commands can be used to rule out issues with the transcoder? (Choose two.)

- A. debug voip ccapi inout
- B. debug ccsip message
- C. show dspfarm profile
- D. show version
- E. debug sccp messages

Answer: AB

NEW QUESTION 3

- (Exam Topic 1)

Replication is failing between the Cisco Unified Communications Manager Publisher and Subscriber servers. In which two ways can you verify the database replication status? (Choose two.)

- A. TRACERT
- B. CLI
- C. APIC-EM
- D. PING
- E. RTMT

Answer: BE

Explanation:

Reference:

<https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communicationsmanager-callmanager/200396-Steps-to-Troubleshoot-Database-Replicati.html#anc2>

NEW QUESTION 4

- (Exam Topic 1)

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. The Cisco VCS Control is down.
- B. The interworking setting is turned off.
- C. The audio feature in the Cisco TelePresence SX10 is turned off.
- D. The SIP trunk is not configured on the gateway.
- E. 01 is not a valid prefix.
- F. ISDN is not enabled on the Cisco TelePresence SX10.
- G. The Cisco TelePresence SX10 is not registered to the Cisco VCS Control.
- H. The Cisco TelePresence SX10 is not registered to the Cisco Express C.

Answer: ABG

NEW QUESTION 5

- (Exam Topic 1)

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: //92672/70DBC0B09D1A/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-alphanumeric

- B. rtp-nte
- C. sip-kpml
- D. sip-notify
- E. cisco-rtp
- F. h245-signal

Answer: B

NEW QUESTION 6

- (Exam Topic 1)

When parsing trace output after the call routing decision and path selection have been made, which two records can be found in the CCM|RouteList? (Choose two.)

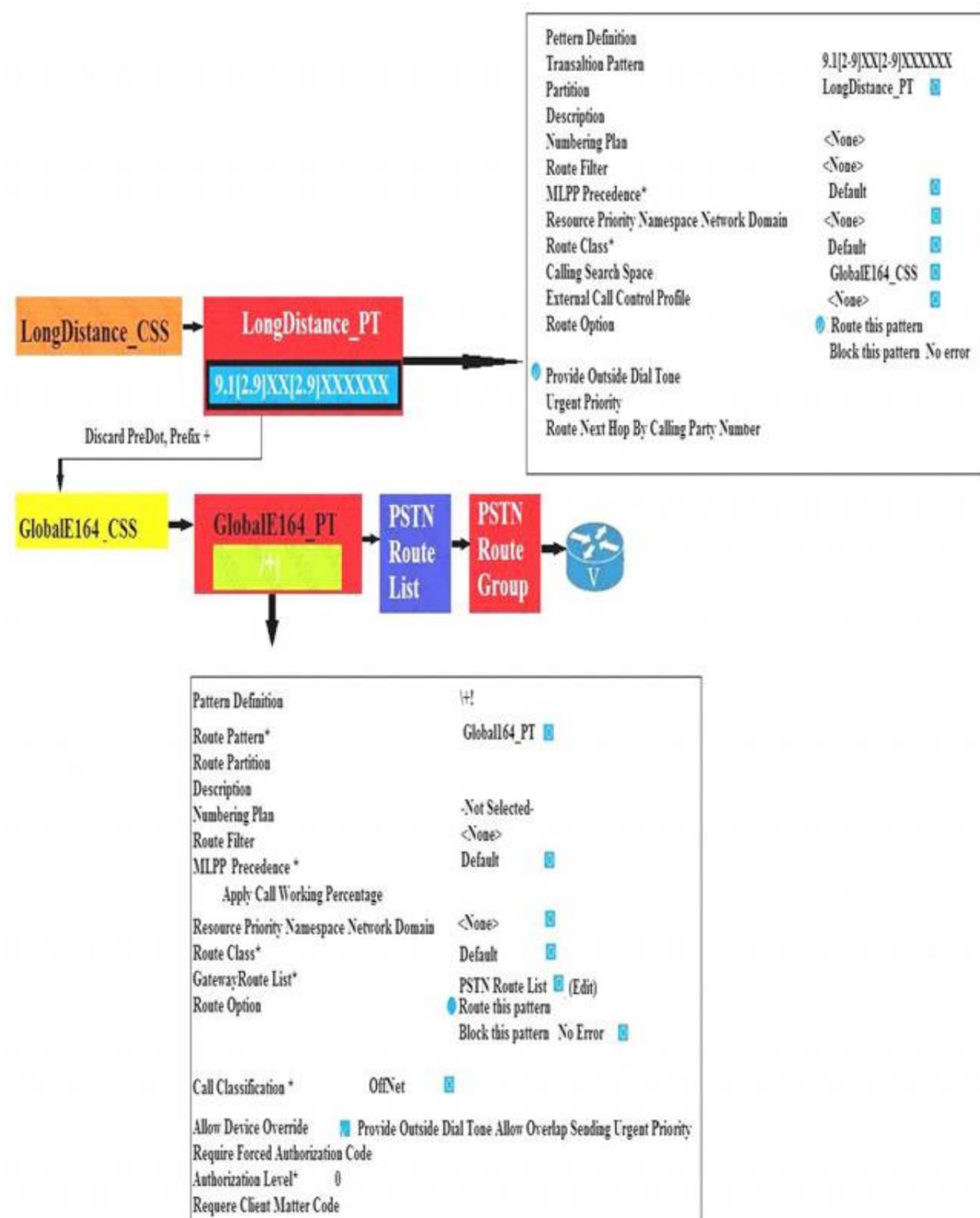
- A. PretransformDigitString
- B. CallingPartyNumber
- C. PretransformCallingPartyNumber
- D. RouteListName
- E. findLocalDevice
- F. RouteListCdr :

Answer: DF

NEW QUESTION 7

- (Exam Topic 1)

Refer to the exhibit.



In this CUCM dial-plan, a localized translation pattern 9.1[2-9]XX[2-9]XXXXXX is configured in the Calling Search Space named LongDistance_CSS. This Translation Pattern is used to globalize the User's dialed digits followed by matching a+! Route Pattern as seen in the Exhibit. Users with Devices/Lines assigned to the LongDistance_CSS are reporting their calls are not proceeding until after the inter-digit timeout of 15 seconds. For example, after dialing digits 91555678309 the user does not hear ringback until after 15 seconds. What two reasons could lead to this inter-digit timeout condition? (Choose two.)

- A. Route Pattern Call Classification set to OffNet.
- B. Route Pattern does not have "Allow Overlap Sending" checked
- C. Translation Pattern has "Provide Outside Dial Tone" checked
- D. Translation Pattern does not have "Urgent Priority" checked
- E. Route Pattern does not have "Urgent Priority" checked

Answer: DE

NEW QUESTION 8

- (Exam Topic 1)

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. 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 configureDFirst on the Cisco VCS Control.
- B. 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.
- C. 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."
- D. Both systems need to register to the Cisco Unified Communications Manager, as the Cisco VCS Control is used only for firewall traversal.
- E. You need Cisco TelePresence Management Suite to register Cisco TelePresence systems.
- F. You need Cisco TelePresence Server to register Cisco TelePresence systems.

Answer: B

NEW QUESTION 9

- (Exam Topic 1)

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 Serviceability
- B. Cisco Unified Communications Manager Dialed Number Analyzer
- C. Cisco Unified Communications Manager Real Time Monitoring Tool
- D. Cisco Unified Syslog Viewer
- E. Cisco IOS debugs

Answer: B

NEW QUESTION 10

- (Exam Topic 1)

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. Change the significant digits for inbound calls to 4 on the SIP trunk configuration in Cisco Unified Communications Manager.
- B. Configure the digit strip 4 on the SIP trunk under Incoming Called Party Settings in Cisco Unified Communications Manager.
- C. 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.
- D. 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.
- E. 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.
- F. Configure the Cisco IOS command num-exp 2288223001 3001 on the gateway ISDN interface.

Answer: ACE

NEW QUESTION 10

- (Exam Topic 1)

Refer to topology and Exhibits below:

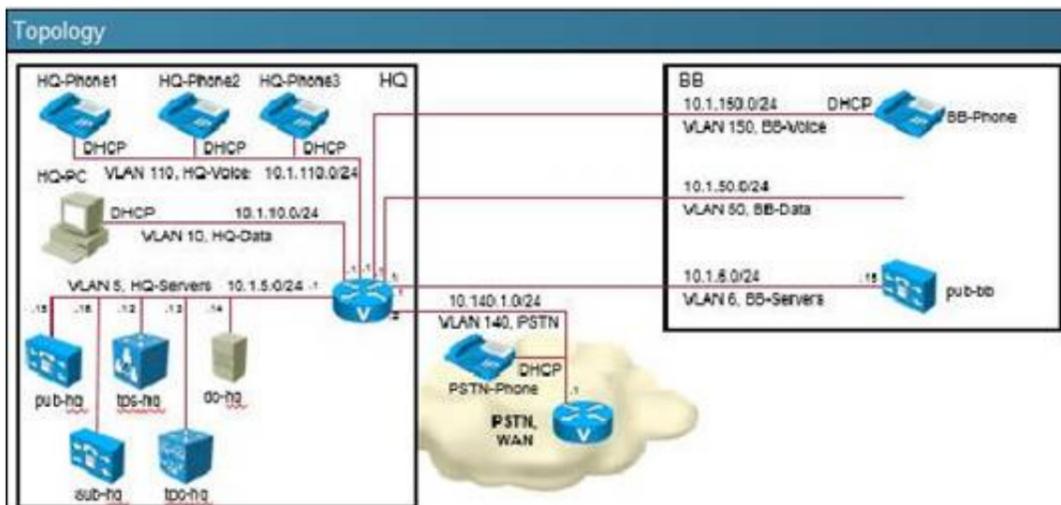


Exhibit2

Gateway Details

Product	Cisco 3925
Gateway	MGCP-GW
Protocol	MGCP
⚠ Device is not trusted	
Domain Name *	MGCP-GW
Description	MGCP-GW
Cisco Unified Communications Manager Group *	Default

Configured Slots, VICs and Endpoints

Module in Slot 0: NM-4VWIC-MBRD
Subunit 0: VWIC2-1MFT-T1E1-T1 0/0/0

Exhibit3

Device Information

Product	Cisco MGCP T1 Port
Gateway	MGCP-GW
Device Protocol	Digital Access PRI
Registration:	Unknown
IPv4 Address:	None
⚠ Device is not trusted	
End-Point Name *	S0/SU0/DS1-0@MGCP-GW
Description	S0/SU0/DS1-0@MGCP-GW

Exhibit4

```

MGCP Domain Name: HQ
Priority      Status      Host
-----
Primary      Backup Ready  10.1.5.25
First Backup Registering with CM 10.1.5.26
Second Backup None

Current active Call Manager: None
Backhaul/Redundant link port: 2428
Failover Interval: 30 seconds
keepalive Interval: 15 seconds
Last keepalive sent: 21:44:55 UTC Feb 10 2015 (elapsed time: 00:20:03)
Last MGCP traffic time: 22:04:42 UTC Feb 10 2015 (elapsed time: 00:00:15)
Last failover time: 22:04:42 UTC Feb 10 2015 from (10.1.5.25)
Last switchback time: 22:04:12 UTC Feb 10 2015 from (10.1.5.26)
Switchback mode: Graceful
MGCP fallback mode: Not Selected
Last MGCP Fallback start time: None
Last MGCP Fallback end time: None
MGCP Download Tones: Disabled
TFTP retry count to shut Ports: 2

FAX mode: disable
Configuration Error History:
    
```

After making AChange to a manually configured MGCP gateway, what step should you take to ensure that the gateway accepts the changes and be operational?

- A. Issue the commands no MGCP, then MGCP.
- B. Issue the commands no SCCP, then SCCP.
- C. Issue the commands Shut, then no shut.
- D. Issue the commands no CCM, then CCM.

Answer: A

NEW QUESTION 12

- (Exam Topic 1)

You are troubleshooting an ILS connectivity issue. All clusters are set to "Use TLS Certificates". Which certificates must be exchanged between Cisco Unified Communications Manager clusters?

- A. Tomcat certificates between all nodes in all clusters.
- B. TLS certificates between publisher nodes in all clusters.
- C. Call Manager certificates between publisher nodes in all clusters.
- D. Tomcat certificates between publisher nodes in all clusters.

Answer: D

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmfeat/CUCM_BK_F3AC1C0F_00_cucm-features

NEW QUESTION 15

- (Exam Topic 1)

URI dialing is enabled between two clusters with the default options. The engineer that set up the URI dialing verified that all was working properly. However, a user from one of the clusters cannot dial using URI to a user in the same cluster. What do you do to resolve this issue?

- A. Verify the password that is used by the authentication under Intercluster Lookup Service configuration.
- B. Find out if the URI address of the called user has ACapital letter in the URI string.
- C. Verify that Intercluster Lookup Service is set up correctly.
- D. Verify USN Data Synchronization Status.

Answer: B

NEW QUESTION 18

- (Exam Topic 1)

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. No incoming and outgoing calls are possible.
- C. Cisco Unified SRST is able to receive incoming calls.
- D. The current call is not disconnected.
- E. The phones display "CM Fallback Service Operating."

Answer: BD

NEW QUESTION 23

- (Exam Topic 1)

After you deploy a new Cisco Collaboration solution, users report echoes and choppy voice quality. Which two actions correct the problem? (Choose two.)

- A. Upgrade the Cisco IOS version and Flash memory on the Cisco IOS router.
- B. Deploy additional hardware resources
- C. Deploy an echo canceller.
- D. Upgrade Cisco Unified Communications Manager.
- E. Enable QoS on the network.

Answer: CE

NEW QUESTION 26

- (Exam Topic 1)

Refer to the exhibit.

Pattern	TimeStamp	Status	Protocol	AgentId	IP
+7XXX	2015/10/06 16:25:26	Reachable	SIP		172.17.0.

An engineer configured CCD between a Cisco Unified Communications Manager (CUCM) cluster and Cisco Unified Communications Manager Express (CME). When the CUCM agents try to dial ACME extension on the range 7XXX, they receive ABusy signal as soon as the 7 is dialed. What can be done to resolve this issue?

- A. Set the CUCM CCD Requesting Service PSTN prefix to 9 instead of 7
- B. Set the CME profile dn-block type to extension instead of global
- C. Set the CUCM Hosted DN Pattern PSTN Failover Strip Digits field to 1 instead of 0
- D. Set the CME subscribe callcontrol to instance instead of wildcarded

Answer: A

NEW QUESTION 30

- (Exam Topic 1)

If a user receives a reorder tone after dialing a number, the Cisco Unified Communications Manager bandwidth allocation for the location of one of the call end devices may have been exceeded. Which option is true?

- A. If the call is using G.711, Cisco Unified CM subtracts 64k. If the call is using G.723, Cisco Unified CM subtracts 16k.
- B. If the call is using G.729, Cisco Unified CM subtracts 8k.
- C. If the call is using G.711, Cisco Unified CM subtracts 80k. If the call is using G.723, Cisco Unified CM subtracts 24k.
- D. If the call is using G.729, Cisco Unified CM subtracts 24k.
- E. If the call is using G.711, Cisco Unified CM subtracts 80k. If the call is using G.723, Cisco Unified CM subtracts 24k.
- F. If the call is using G.729, Cisco Unified CM subtracts 16k.
- G. If the call is using G.711, Cisco Unified CM subtracts 64k. If the call is using G.723, Cisco Unified CM subtracts 24k.
- H. If the call is using G.729, Cisco Unified CM subtracts 8k.

Answer: B

NEW QUESTION 33

- (Exam Topic 1)

Which configuration can be dynamically set using the Cisco Unified Communications Manager Device Mobility feature?

- A. phone model and protocol
- B. SRST reference and directory number
- C. CSS and local gateway
- D. partition and CSS
- E. media resources and permanent bridges

Answer: C

NEW QUESTION 35

- (Exam Topic 1)

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. There is no search pattern to route the call to System B.
- B. There is no valid route pattern to route from System A to System B.
- C. System B is registered as H.323 and needs to use an E.164 alias number only.
- D. The Cisco VCS Control should be neighbor to the Cisco Unified Communications Manager.
- E. You need an MGCP gateway to route from the Cisco Unified Communications Manager to the Cisco VCS Control.
- F. The Cisco VCS Control is missing the Cisco Unified Communications Manager interop option.
- G. The Cisco VCS Control is missing the interworking option.

Answer: G

NEW QUESTION 36

- (Exam Topic 1)

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

- A. missing the configuration command isdn bind-l3 ccm-manager under the ISDN interface
- B. mismatched domain name on the MGCP gateway and Cisco Unified Communications Manager gatewayconfiguration
- C. misconfigured route group in Cisco Unified Communications Manager
- D. incorrect MGCP IP address specified in the gateway configuration in Cisco Unified Communications Manager

Answer: B

NEW QUESTION 38

- (Exam Topic 1)

Which Cisco Unified Communications Manager tool verifies configured route patterns, calling search spaces, and route groups?

- A. Cisco Unified Reporting
- B. Cisco Unified Communications Manager CDR Analysis and Reporting
- C. Cisco Unified Communications Manager Dialed Number Analyzer
- D. Cisco Unified Real-Time Monitoring Tool

Answer: C

NEW QUESTION 40

- (Exam Topic 1)

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 a software MTP.
- B. Implement a hardware MTP.
- C. Implement transcoding with the router DSP resources.
- D. Implement transcoding with the Cisco Unified Communications Manager resources.

Answer: AB

NEW QUESTION 45

- (Exam Topic 1)

You observe that EMCC restriction on the cluster fails when a user with a different profile logs in to another cluster. Which action can you take to correct the problem?

- A. Enable logical partitioning.
- B. Set SIP session timers to 0.
- C. Set SIP session timers to 3800.
- D. Configure the regions in Cisco Unified Communications Manager to allow EMCC.

Answer: D

NEW QUESTION 46

- (Exam Topic 1)

During ABusiness-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 Cisco Expressway does not have a Rich Media Session license.
- B. The firewall is blocking SIP signaling.
- C. The Cisco Unified Communications Manager is not configured for business-to-business calling.
- D. The firewall is blocking inbound RTP ports.
- E. The Advanced Networking option is not installed on the Expressway Edge.

Answer: D

NEW QUESTION 48

- (Exam Topic 1)

Which reason for calls being disconnected just after connection is true?

- A. An incompatible MTP type is allocated to the call.
- B. Codec are mismatched between phones.
- C. Firewall is blocking RTP packets.
- D. Phone A is using SCCP and phone B is using SIP.

Answer: B

NEW QUESTION 51

- (Exam Topic 1)

What is the default hold time for two peering SAF Forwarders that are connected on a LAN?

- A. 240 seconds
- B. 15 seconds
- C. 60 seconds
- D. 45 seconds
- E. 120 seconds

Answer: B

NEW QUESTION 55

- (Exam Topic 1)

Which two issues can prevent an IP Phone from receiving an IP address via DHCP? (Choose two.)

- A. The DHCP server is not on the same VLAN as the phone.
- B. The DHCP scope's leases are exhausted.
- C. DHCP Option 150 is incorrect.
- D. The TFTP server is not reachable.
- E. The DHCP scope is not defined for the subnet of the phone.

Answer: AB

NEW QUESTION 60

- (Exam Topic 1)

Which configuration is required on Cisco TelePresence Server, in order to support 1080p resolution?

- A. Screen licenses must be configured.
- B. Cisco TelePresence Server must be in remotely managed mode.
- C. Cisco TelePresence Server must be in HD mode.
- D. Cisco TelePresence Server must be configured with Cisco TelePresence Conductor.
- E. Cisco TelePresence Server must be in Full HD mode.

Answer: E

NEW QUESTION 65

- (Exam Topic 1)

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. H.323 ID
- B. Configuration Template
- C. SIP URI
- D. Active Cisco Unified Communications Manager Address
- E. System Name
- F. System Contact
- G. E.164 alias
- H. IEEE 802.1x Authentication Password

Answer: ABCEG

NEW QUESTION 70

- (Exam Topic 1)

Endpoints are configured for both H.323 and SIP using the same URI and Cisco VCS settings, but the endpoints register only as H.323 endpoints. What is causing this issue?

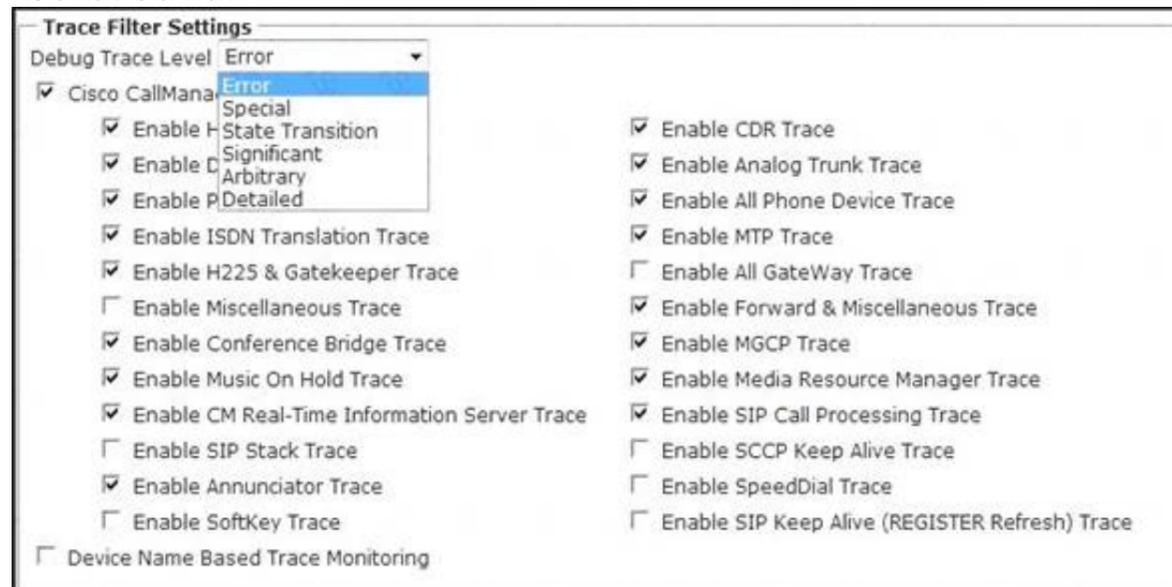
- A. A firewall is blocking all traffic from the endpoints to the Cisco VCS.
- B. The Cisco VCS has no SIP domains configured.
- C. The Cisco VCS is blocking the endpoints because of duplicate ID entries.
- D. The endpoints do not have the SIP option key installed.
- E. SIP does not work, because SIP is used for Cisco Unified Communications Manager registration only.

Answer: B

NEW QUESTION 74

- (Exam Topic 1)

Refer to the exhibit.



Which Cisco Unified Communications Manager trace file level should be selected when enabling traces to send to Cisco TAC for analysis?

- A. State Transition
- B. Arbitrary
- C. Significant
- D. Error
- E. Detailed
- F. Special

Answer: E

NEW QUESTION 77

- (Exam Topic 1)

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. CS6
- B. CS2
- C. AF41
- D. AF31
- E. EF
- F. CS3

Answer: C

NEW QUESTION 78

- (Exam Topic 1)

Refer to the exhibit.

<http://172.16.1.1:8080/emapp/EMAppServlet?device=#DEVICENAME#&EMCC=#EMCC#>

a Cisco Unified Communications Manager Extension Mobility enabled user attempts to log in to a new phone where Cisco Extension Mobility Cross Cluster Service is configured with the URL that is displayed in the exhibit. The service is configured on a Cisco Unified CM 9.X or later Unified CM cluster. After the user enters the user ID and PIN, the phone displays "Login is Unavailable (23)."

Which configuration requirement supports Cisco EMCC and avoids this error condition?

- A. The device must be subscribed to Cisco EMCC Services.
- B. The given user ID is not found in the remote cluster.
- C. The user must associate a device profile.
- D. The Cisco IP Phone Services configuration for Cisco EMCC must set the enterprise subscription.

Answer: A

NEW QUESTION 79

- (Exam Topic 1)

You are troubleshooting media issues during SIP calls on AC-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 AC-series endpoint? (Choose four.)

- A. Stop the capture by pressing the key combination Ctrl + C.
- B. Set up administrator access on the endpoint.
- C. Start the capture by entering the tcpdump command string and then start the call.
- D. Set up root access on the endpoint.
- E. Start the call, and then enter the tcpdump command string to begin the capture.
- F. On the endpoint, set the default transport to TLS.
- G. Set the default transport to TCP.

Answer: ACDG

NEW QUESTION 80

- (Exam Topic 1)

An endpoint cannot connect to a valid TFTP server during the registration process. Which two statements describe possible causes? (Choose two.)

- A. The DHCP configuration contains TFTP server 66, but no DNS server is available.
- B. The DHCP configuration contains TFTP server 150, but no DNS server is available.
- C. No separate voice VLAN configuration on the switch port connected to the endpoint.
- D. Cisco Unified Communications Manager CallManager services are not started.
- E. Cisco Unified Communications Manager TFTP services are not started.

Answer: DE

NEW QUESTION 83

- (Exam Topic 1)

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 ip interface details
- D. show cdp neighbor details
- E. show eigrp service-family ipv4 clients details
- F. show service-family asf-forwarder details

Answer: E

NEW QUESTION 87

- (Exam Topic 1)

An engineer is troubleshooting an issue where aliases that contain an identity "@abc.com" are unable to register with an endpoint due to an entry in the registration restriction configuration in Expressway-E. Where is the alias being blocked in Expressway-E?

- A. regex list
- B. hunt list
- C. black list
- D. allow list

Answer: C

NEW QUESTION 88

- (Exam Topic 1)

In a network with two Cisco Unified Communications Manager clusters, Phone 1 on Cluster A dials jsmith@cisco.com to reach Phone 2 on Cluster B, and the call fails. What are two possible causes for this problem? (Choose two.)

- A. Cluster A does not have a SIP trunk to reach Cluster B.
- B. Cisco UDS is not enabled on Cluster B.
- C. Cluster B does not have a SIP trunk to reach Cluster A.
- D. The calling space on Phone 1 contains the partition with a pattern that matches jsmith@cisco.com
- E. Cisco UDS is not enabled on Cluster A.

Answer: AC

NEW QUESTION 91

- (Exam Topic 1)

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 Cisco Extension Mobility service has not been configured on the phone.
- B. The user device profile is not subscribed to the Cisco Extension Mobility service.
- C. The CTI service is not running.
- D. The logout URL that is defined for the Cisco Extension Mobility service is incorrect or does not exist under the IP Phone Services configuration.

Answer: B

NEW QUESTION 92

- (Exam Topic 1)

Which command is used to check if an MGCP gateway is currently registered with Cisco CallManager?

- A. Router# show ccm-manager gateway

- B. Router# show mgcp ccm-manager
- C. Router# show ccm-manager
- D. Router# show ccm manager

Answer: C

NEW QUESTION 94

- (Exam Topic 1)

Refer to the exhibit.

Audio/Video Call: Video Stream Statistics					
Local	10.10.10.100:25654				
Remote	10.10.10.26:16654				
Average Latency (Call)	0				
Average Latency (Period)	0				
	Center	LEGACY	Presentation (1 or 5 FPS)	Presentation (30 FPS)	
Transmit					
Is Active	0	0	0	0	
Media Type	H.264	H.264	H.264	H.264	
Frames Per Second	30.00	30.00	5.00	30.00	
Total Bytes	256836441	5386	0	0	
Total Packets	279459	62	0	0	
Receive					
Is Active	0	0	0	0	
Media Type	H.264	N/A	N/A	H.264	
Frames Per Second	30.00	30.00	5.00	30.00	
Total Bytes	1710890943	0	0	350683593	
Total Packets	1749089	0	0	370595	
Lost Packets	2129	0	0	332	
Lost Packets % (Call)	0.1216	0.0000	0.0000	0.0895	
Lost Packets % (Period)	0.0000	0.0000	0.0000	0.2058	
Duplicate Packets	0	0	0	0	
Late Packets	0	0	0	0	
Failed SRTP Authentication Packets	0	0	0	0	
Average Jitter (Call)	9	0	0	9	
Average Jitter (Period)	7	0	0	9	

Which timeframe does the Lost Packets % (Period) value refer to?

- A. total packets lost during the active call
- B. total packets lost within the last 10 seconds
- C. total packets lost within the last 10 minutes
- D. total packets lost within the last second

Answer: B

NEW QUESTION 98

- (Exam Topic 1)

How does an IP Phone react upon initialization, if there is no CTL and ITL files present on the device?

- A. The phone downloads only the files that are signed to that device.
- B. The phone blindly trusts the next series of downloaded files.
- C. The file matches the signature against the files on TFTP.
- D. The phone displays the message, Unprovisioned.

Answer: D

NEW QUESTION 101

- (Exam Topic 1)

Which two statements about Cisco Unified CM location bandwidth deduction are true? (Choose two.)

- A. If a Call uses G.711, Cisco Unified Communications Manager subtracts 64k.
- B. If a Call uses G.711, Cisco Unified Communications Manager subtracts 80k.
- C. If a Call uses G.723, Cisco Unified Communications Manager subtracts 16k.
- D. If a Call uses G.729, Cisco Unified Communications Manager subtracts 16k.
- E. If a Call uses G.729, Cisco Unified Communications Manager subtracts 24k.

Answer: BE

NEW QUESTION 103

- (Exam Topic 1)

Which action can you take to prevent users from transferring external calls to external devices?

- A. Enable the Block OffNet to OffNet Transfer feature in Cisco Unified Communications Manager
- B. Block the National route pattern.
- C. Remove the route pattern to prevent inbound calls from matching the gateway.
- D. Create a new translation pattern to block external call transfer.

Answer: A

NEW QUESTION 104

- (Exam Topic 1)

An inbound call from the PSTN is not reaching the directory number that it is calling. When the PSTN phone calls the correct DID, only a dial tone is heard. Which

command resolves this issue?

- A. (config-dial-peer)#direct-inward-dial
- B. (config-controller)# no provide-outside-dialtone
- C. (config-if)#no dial-tone
- D. (config-dial-peer)# no dial-tone
- E. (config-if)#direct-inward-dial
- F. (config) allow inbound dial-peer 1

Answer: A

NEW QUESTION 107

- (Exam Topic 1)

Which two statements indicate something that can cause an IP phone to fail roaming when device mobility has been configured? (Choose two.)

- A. Device Mobility Mode is set to Off in the Cisco Unified Communications Manager service parameters while the device mobility configuration on the phone is set to default.
- B. No device mobility groups have been configured.
- C. No locations have been configured and assigned to the device pools.
- D. No physical locations have been configured and assigned to the device pools.
- E. No device mobility-related information settings were configured under the device pools.

Answer: AD

NEW QUESTION 109

- (Exam Topic 1)

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. View the `activelog` file.
- B. Run the same command on all nodes of the cluster.
- C. Restart the Cisco CallManager service.
- D. The command `utils dbreplication runtimestate` must be run on the publisher.
- E. The command `utils dbreplication runtimestate` must be run on the subscriber.

Answer: A

NEW QUESTION 113

- (Exam Topic 1)

Several users at your site have reported that they receive a fast busy when they call another site within the same cluster. Which action can you take to connect the problem?

- A. Rebuild the SIP trunk between Cisco Unified Communications Manager and the gateway.
- B. Run RTMT to check the status of the Cisco Unified Communications Manager services.
- C. Reset the intercluster trunk from your site to the remote site.
- D. Correct the grouping of the site partitions and calling search spaces.

Answer: D

NEW QUESTION 116

- (Exam Topic 1)

Refer to topology and Exhibits below:

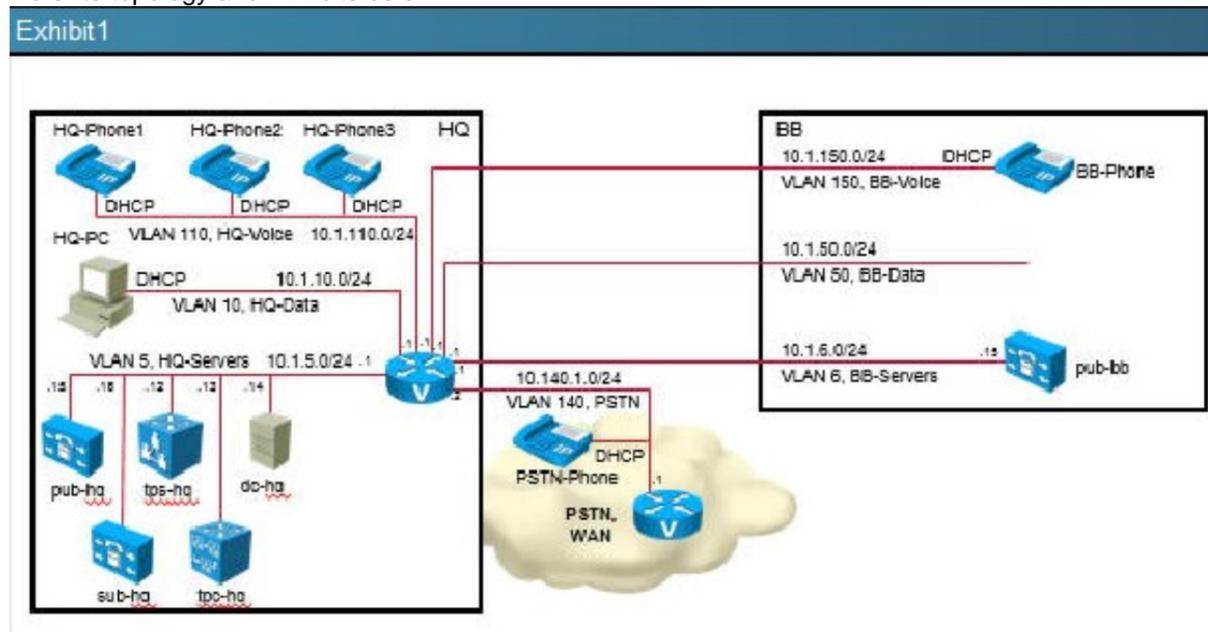


Exhibit2

Pattern Definition

Pattern Usage: Domain Routing

IPv4 Pattern*: v360.cisco.com

IPv6 Pattern: [Empty]

Description: [Empty]

Route Partition: < None >

SIP Trunk/Route List*: Trunk-VCS (Edit)

Block Pattern

Exhibit3

Trunk Configuration

Save | Delete | Reset | Add New

Destination

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	10.1.5.29		5060

MTP Preferred Originating Codec*: 711ulaw

BLF Presence Group*: Standard Presence group

SIP Trunk Security Profile*: VCS Non Secure SIP Trunk Profile

Rerouting Calling Search Space: < None >

Out-Of-Dialog Refer Calling Search Space: < None >

SUBSCRIBE Calling Search Space: < None >

SIP Profile*: Standard SIP Profile (View Details)

DTMF Signaling Method*: RFC 2833

Exhibit4

Name	Type	Calls	Bandwidth used	H323 status	SIP status	Search rule status
DefaultZone	Default zone	0	0 kbps	On	On	
<input type="checkbox"/> UCM	Neighbor	0	0 kbps	Off	Active	Enabled search rules: 2

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
<input type="checkbox"/> 50	LocalZoneMatch	Any	Any	No	Any alias				Continue	LocalZone	Disabled	View/Edit Clone
<input type="checkbox"/> 100	UCM	Any	Any	No	Any alias				Continue	UCM	Enabled	View/Edit Clone
<input type="checkbox"/> 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
<input type="checkbox"/> 50	LocalZoneMatch	Any	Any	No	Any alias				Continue	LocalZone	Enabled	View/Edit Clone
<input type="checkbox"/> 100	UCM	Any	Any	No	Any alias				Continue	UCM	Enabled	View/Edit Clone
<input type="checkbox"/> 100	UCM2	SIP	Any	No	Alias pattern match	Regex	2...	Leave	Stop	UCM	Disabled	View/Edit Clone

a Call from a SX20 in the BackBone (not shown) with a URI extension is dialing a HQ Ph 1 that is registered to the HQ CUCM. 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 SIP port is incorrect on the Cisco Unified Communications Manager CUCM SIP trunk.
- E. The Local Zone Match Rule state is disabled.
- F. Rule name UCM2 is set to stop on Match

Answer: AC

NEW QUESTION 120

- (Exam Topic 1)

Which Cisco Unified Communications Manager troubleshooting tool can be used to look at detailed specific Events, such as dial plan digit analysis, as they are happening?

- A. RTMT real-time trace
- B. Cisco Unified Dialed Number Analyzer
- C. syslog output
- D. RTMT performance log viewer

Answer: B

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/dna/8_6_1/dnaguide/dnai.html

NEW QUESTION 124

- (Exam Topic 1)

Cisco Unified Communications Manager failed to register with the Cisco SAF Forwarder. Assuming that the Cisco IOS SAF Forwarder is configured correctly, which minimum configuration would be needed on Cisco Unified Communications Manager to test registration?

- A. SAF trunk, SAF security profile, Cisco SAF Forwarder, and CCD advertising service
- B. SAF trunk, SAF security profile, Cisco SAF Forwarder, and CCD requesting service
- C. SAF trunk, SAF security profile, Cisco SAF Forwarder, CCD requesting service, and CCD advertising service
- D. SAF trunk, SAF security profile, and Cisco SAF Forwarder
- E. SAF trunk, CCD requesting service, and CCD advertising service

Answer: B

NEW QUESTION 127

- (Exam Topic 1)

An endpoint has a SIP trunk configured between Cisco Unified Communications Manager (CUCM) and a Cisco VCS cluster. When a Call is made from a Cisco TelePresence System EX60 that is registered to the Cisco VCS to a Cisco IP Phone 9971 that is registered to CUCM, it rings. But upon picking up the call, a Busy tone is heard. What should be checked to resolve this issue?

- A. CUCM zone on the Cisco VCS.
- B. SIP trunk and phone region settings.
- C. SIP trunk registration.
- D. authentication on the SIP trunk.

Answer: B

NEW QUESTION 129

- (Exam Topic 1)

Which four performance counters are available when monitoring a Cisco MTP device using the Cisco Unified Communications Manager RTMT? (Choose four.)

- A. Resource Total
- B. Resource Available
- C. Out of Resources
- D. Resource Idle
- E. Resource Active
- F. MTP Streams Active
- G. MTP Connection Lost
- H. MTP Instances Active

Answer: ABCE

NEW QUESTION 132

- (Exam Topic 1)

Refer to the exhibit.

```
voice-card 0
no local-bypass

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
codec g711ulaw
codec g711alaw
codec g729ar8
codec g729br8
codec g729r8
maximum session 5
associate application SCCP
no shutdown
```

You're tasked with staging configuration changes to add conference bridge functionality to an existing IOS voice gateway deployment. What command is missing for the configuration to be accepted by the IOS CLI?

- A. The command maximum conference-participants must be configured under the dspfarm profile.
- B. The Enhanced IOS Conference Bridge is not configured in Cisco Unified Communications Manager.
- C. The command dsp services dspfarm must be configured under the voice-card configuration.
- D. The dspfarm command under the voice card is missing.
- E. The dsp tdm pooling command under the voice-card is missing.

Answer: C

NEW QUESTION 133

- (Exam Topic 1)

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?

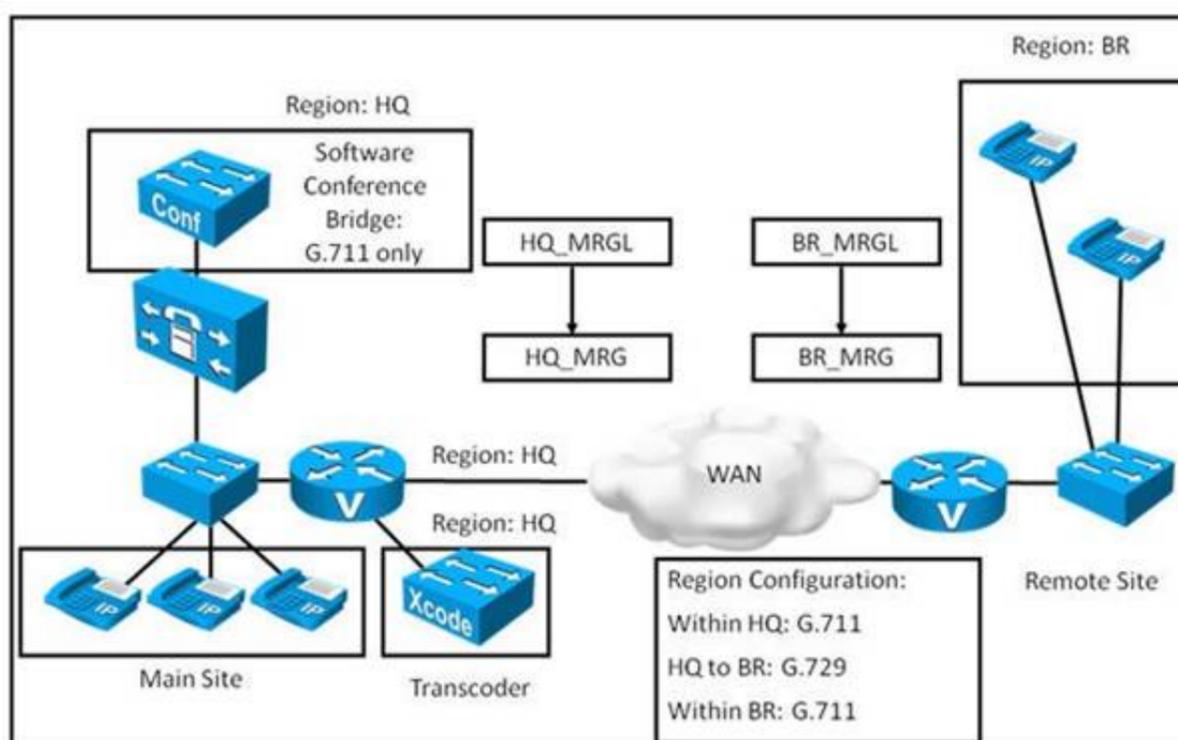
- A. Cisco Unified Communications Manager does not support SIP gateways.
- B. The gateway does not have the UC license installed.
- C. The gateway does not have Cisco Unified Border Element session licensing.
- D. Cisco Unified Communications Manager does not show a SIP gateway as registered if it is not properly configured.
- E. Cisco Unified Communications Manager never shows a SIP gateway as registered. Even when it is properly configured.
- F. The Cisco ISR G2 cannot be a SIP gateway.

Answer: E

NEW QUESTION 134

- (Exam Topic 1)

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. The BR_MRG must contain the transcoder device
- B. The BR_MRGL must be assigned to the BR phones.
- C. The HQ_MRG must contain the transcoder device
- D. The HQ_MRGL must be assigned to the HQ phones.

- E. A transcoder should be configured at the remote site and assigned to all remote phones through the BR_MRGL.
- F. The HQ_MRGL must contain the transcoder device
- G. The HQ_MRGL must be assigned to the software conference bridge.
- H. Enable the software conference bridge to support G.711 and G.729 codecs in Cisco Unified Communications Manager service parameters.

Answer: D

NEW QUESTION 136

- (Exam Topic 1)

Refer to the exhibit.

```
(output omitted)

controller T1 0/0/0
 framing esf
 linecode b8zs
 cablelength short 133
 pri-group timeslots 1-24
 description PRI to PSTN

interface Serial0/0/0:23
 description PSTN
 no ip address
 encapsulation hdlc
 isdn switch-type primary-ni
 isdn incoming-voice voice
 isdn negotiate-bchan
 no cdp enable

dial-peer voice 210 pots
 destination-pattern 91456.....
 port 0/0/1:23
 prefix 456

(output omitted)
```

Local 10-digit dialing in the North American Numbering Plan to area Code 456 is failing. Which two changes are needed to enable 10-digit dialing? (Choose two.)

- A. Change dial peer 210 to destination-pattern 456.....
- B. Change dial peer 210 to destination-pattern 9456.....
- C. Change dial peer 210 to port 0/0/0:23.
- D. Remove the command prefix 456 from dial peer 210.

Answer: BC

NEW QUESTION 141

- (Exam Topic 1)

In Cisco Unified Communications Manager, what is the default maximum number of learned patterns for the call control discovery feature parameter?

- A. 5000
- B. 10000
- C. 20000
- D. 500
- E. 50000

Answer: C

NEW QUESTION 143

- (Exam Topic 1)

In an MCU call with three Cisco TelePresence MX800 systems and a mobile phone calling in, the three TelePresence MX800 systems suddenly experience low audio levels, but the mobile phone audio levels are correct. What can you do to correct this issue?

- A. Turn off the audio processors on the TelePresence MX800.
- B. Use the mobile phone audio option on the TelePresence MX800 to adjust the mobile phone levels.
- C. Mobile phone audio levels can vary, so you cannot correct the issue.
- D. Turn on AGC on the MCU to adjust the audio levels.
- E. Turn on ALG on the MCU to adjust the audio levels.
- F. Turn on the Auto Adjust levels under "Settings > Audio" on the MCU.

Answer: D

NEW QUESTION 145

- (Exam Topic 1)

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 progress_ind progress disable command in the gateway.
- B. Issue the voice call disc-pi-off command in the gateway.
- C. Enable mobile connect on the user profile.
- D. Assign Resume softkey on the desk phone.

Answer: B

NEW QUESTION 149

- (Exam Topic 1)

Refer to the exhibit. An engineer is troubleshooting a SAF client that is not registering with the SAF forwarder.

Sep 30 19:40:32.130: SAF-EC: Client Label attribute found

Sep 30 19:40:32.130: SAF-EC: Client found CUCMI

Sep 30 19:40:32.130:

Class: Request Method: 0x0001 (Register)

Sep 30 19:40:32.134: Packet Length: 124 bytes Not including 20 byte Saf Header

Sep 30 19:40:32.134: Magic Cookie: 0x7F5A9BC7 Transaction ID: 4B43474D58584743574D5A4C

Sep 30 19:40:32.134: Username: 006: Length: 5: "SAF"

Sep 30 19:40:32.134: Client Name: 1001: Length: 41:

UCM/172.16.100.50/NodeId=1/10.5.1.10000-7

Sep 30 19:40:32.134: Protocol Version: 1003: Length: 4: 65536

Sep 30 19:40:32.134: Page Size: 1004: Length: 4: 7

Sep 30 19:40:32.134: Client Label: 1005: Length: 5: CUCMI

Sep 30 19:40:32.134: Message Integrity: 008: Length: 20:

90BD7AE9E02193BB717E24015EB3C95B4862EEFF

Sep 30 19:40:32.134: SAF-EC: Message integrity check failed

Sep 30 19:40:32.134: Class: Error Response Method: 0x0001 (Register)

Sep 30 19:40:32.134: Packet Length: 68 bytes Not including 20 byte Saf Header

Sep 30 19:40:32.138: Magic Cookie: 0x7F5A9BC7 Transaction ID: 4B43474D58584743574D5A4C

Sep 30 19:40:32.138: Realm: 014: Length: 5: "SAF"

Sep 30 19:40:32.138: Error Code: 009: Length: 27: Error Class: 4 Error Code: 31

Sep 30 19:40:32.138: Error Reason: Integrity Check Failure

Sep 30 19:40:32.138: Message Integrity: 008: Length: 20:

52C03E5DE27EA25A5E7684EA5C615247E5F62D89

What is the root cause of the issue?

- A. The packets are being dropped on the network.
- B. The SAF password is incorrectly configured.
- C. The SAF port is incorrectly configured.
- D. The client SAF label is incorrectly configured.

Answer: A

NEW QUESTION 153

- (Exam Topic 2)

An Extension Mobility user successfully logs in to an Extension Mobility (Hoteling) phone, but now the user cannot log in to an additional Extension Mobility (Hoteling) phone. Why is the user unable to log in to the next phone?

- A. The user is set for autologout for Extension Mobility.
- B. The user is set to a maximum of one login.
- C. The user does not have login privileges for the new phone
- D. The user does not have permission to use Extension Mobility
- E. The first phone does not have a logout button, due to an incorrect phone button template.
- F. The user is not set to allow multiple logins.

Answer: B

NEW QUESTION 157

- (Exam Topic 2)

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 IP phone that the user attempted to call.
- B. Verify that the partition and calling search space are correct.
- C. Reboot the user's IP phone.
- D. Reboot the Cisco Unified communications Manager Cluster.
- E. Ping the remote gateway to verify connectivity.
- F. Use RTMT to trace the call from DN to DN.

Answer: BE

NEW QUESTION 161

- (Exam Topic 2)

You need to add transcoding support for g711alaw between two different sites. The current configuration is as follows:

dspfarm profile 1 transcode codec g711 ulaw

codec g729ar8 codec g729abr8 codec g729br8 codec g729r8 maximum sessions 4

associate application SCCP

Which two steps do you need to take in order to add support for this codec? (Choose two)

- A. Issue the command no associate application SCCP.
- B. Issue the command shutdown under the dspfarm profile.
- C. Add the command codec g711alaw and issue the no shutdown command.
- D. Disassociate the profile under the CCM group.
- E. Add the command codec g711alaw and issue the command no sccp followed by the command sccp.

Answer: CE

NEW QUESTION 163

- (Exam Topic 2)

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 > Domains > SIP
- B. VCS configuration > Protocols > SCCP > Domains
- C. VCS Domains > Protocols > SIP
- D. VCS configuration > Protocols > SIP > Domains

Answer: D

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/telepresence/infrastructure/articles/vcs_endpoint_registration_problems_kb_460.html

NEW QUESTION 165

- (Exam Topic 2)

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 Ad Hoc Conference, Call Manager Service Parameter
- B. Maximum Ad Hoc Conference, Enterprise Parameters Configuration
- C. Maximum Meet-Me Conference, Call Manager Service Parameter
- D. Maximum Meet-Me Conference, Enterprise Parameters Configuration

Answer: C

NEW QUESTION 168

- (Exam Topic 2)

If you are required to configure a router to use MGCP on a digital port, which measure will you take?

- A. Add the application mgcpapp subcommand to the dial peer
- B. Add the service mgcp subcommand to the dial peer
- C. Add the parameter application mgcpapp to the dsO-group controller subcommand.
- D. Add the service mgcp parameter to the dsO-group controller subcommand

Answer: D

NEW QUESTION 170

- (Exam Topic 2)

Cisco TelePresence Systems that are calling from a Cisco Unified Communications Manager cluster to an

- A. H.323 registered device on a Cisco Expressway Control do not work
- B. However, calls from the H.323 Expressway registered devices to the same Cisco Unified Communications Manager registered systems do work. What are two possible reasons? (Choose two.)
- C. The SIP trunk has not enabled bidirectional mode.
- D. The call from Cisco Unified Communications manager to the Cisco Expressway Control does not match any search rule.
- E. Calls from Cisco Unified Communications Manager add ".5060" or ".5061" after the SIP address, unlike the Cisco Expressway Control.
- F. The Cisco Expressway Control is registered in the wrong partition.
- G. Both systems do not support TLS encryption

Answer: AC

NEW QUESTION 173

- (Exam Topic 2)

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

- A. username (alias)
- B. extension
- C. first name and last name
- D. employee ID
- E. title

Answer: AB

NEW QUESTION 178

- (Exam Topic 2)

While troubleshooting a transcoding issue, where you need 32 G.711 to G.729a sessions, you realize the DSP capacity may be undersized. Which two can support your requirement? (Choose two.)

- A. PVDM4-128
- B. PVDM4-32
- C. PVDM4-48
- D. PVDM4-16
- E. PVDM4-64

Answer: BE

NEW QUESTION 179

- (Exam Topic 2)

An engineer is troubleshooting an image quality issue for a Call on an SX20 and must verify the video protocol. Which menu navigation sequence does the engineer use to find the video protocol on the web interface?

- A. Call Control > Call Control > Participants and click the triangle to get more call detail
- B. Configuration > Video > Input > Participants and click the triangle to get more call detail
- C. Configuration > Conference > Rate > Participants and click the triangle to get more call detail
- D. Configuration > System Configuration > Conference > Participants and click the triangle to get more call detail

Answer: A

NEW QUESTION 182

- (Exam Topic 2)

You install a second Cisco TelePresence PrecisionHD 1080p camera on your C-Series Codec, but you are unable to control the camera. What should you do to fix this issue?

- A. Ensure that the distance between the end user and the camera is between 4 feet and 6 feet.
- B. Purchase and install a Camera with full pan, tilt, and zoom capabilities.
- C. Order a VISCACascading cable from Cisco, which connects the first camera to the second camera.
- D. Ensure that the RJ45 end of the camera Cable goes to the connector that is marked "HD Video Out Codec."
- E. Ensure that the HD-SDI cable between the codec and the camera is not longer than 100 meters (330 feet).

Answer: C

NEW QUESTION 186

- (Exam Topic 2)

You are configuring the Call Control Discovery feature. You need to configure Hosted DN Group Configuration settings. What is the range of values that you can define in the PSTN Failover Strip Digits field?

- A. 0-10
- B. 0-12
- C. 0-14
- D. 0-16
- E. 0-18
- F. 0-20

Answer: D

NEW QUESTION 189

- (Exam Topic 2)

What does the debug output show? Refer to the exhibit.

```

047155: *Sep 25 08:57:32: sccp_connect_to_ccm_on_priority_basis: Trying connecting to CCM
on priority basis prof id 1, appl type 8
047156: *Sep 25 08:57:32: sccp_connect_to_ccm_on_priority_basis: Trying CCM with ipaddr
10.0.2.3, priority 1, port 2000
047157: *Sep 25 08:57:32: sccp_tcp_socket_connect: Trying tcp soc connect for appl_type 8,
prof id 1, to ipaddr 10.0.2.3
047158: *Sep 25 08:57:32: sccp_get_ccm_intf_vrf_id: ccm sccp local interface vrfid=0appl
type is 8, appl profile 1
047159: *Sep 25 08:57:32: sccp_get_local_address_by_idb: get_physical_ip:1, use IP addr
10.0.144.15
047160: *Sep 25 08:57:32: sccp_tcp_open_and_set_option: Socket 3 opened and binded to addr
10.0.2.3 - for appl type 8, prof id 1 w/ local ddr 10.0.144.15
047161: *Sep 25 08:57:32: sccp_socket_connect_to_ccm:: connecting to port 2000
sccp_socket_connect_to_ccm: soc conn to 10.0.2.3 port 2000 in progress for appl_type 8,
state 1, soc fd 3 appl 431E4B5C
047162: *Sep 25 08:57:32: sccpapp_process_socket_events: appl_type 8, soc fd 3, soc 3,
smb soc -1
047163: *Sep 25 08:57:32: sccpapp_process_socket_events: TCP_SOCKET_READ: appl_type 8, eve
4, state 1
047166: *Sep 25 08:57:32: sccp_setup_appl_to_tcp_conn_state: soc connected to 10.0.2.3,
for prof id 1, appl type 8
047167: *Sep 25 08:57:32: sccp_register_with_connected_ccm:
047168: *Sep 25 08:57:32: sccp_register_with_connected_ccm: Sending regis msg for
appl type 8, state 5
047169: *Sep 25 08:57:32: sccp_get_appl_serv_by_prof_id: prof_id 1 found in the CCM group
1
047187: *Sep 25 08:57:32: sccpapp_process_socket_events: TCP_SOCKET_READ: appl_type 8, eve
4, state 5
047188: *Sep 25 08:57:32: sccp_get_control_message: bytes_to_read 8, bytes_read 8,
appl_total_bytes 8, appl_bytes_read 8, soc_read_mode 0, soc 3
047189: *Sep 25 08:57:32: sccp_get_control_message: bytes_to_read 36, bytes_read 36,
appl_total_bytes 36, appl_bytes_read 36, soc_read_mode 1, soc 3
047190: *Sep 25 08:57:32: sccp_get_control_message: sccp ctrl msg rcvd, prof_id 1,
appl_type 8, msg len 36
047191: *Sep 25 08:57:32: sccp_spi_ha_is_not_ready_to_handle_new_msg:
sccp_spi_ha_is_not_ready_to_handle_new_msg
047192: *Sep 25 08:57:32: sccp_parse_control_msg: msg_ptr 437C1BC4, msg_len 36, msg_id 157
047193: *Sep 25 08:57:32: sccp_parse_control_msg: glob_ccm->version 9
047194: *Sep 25 08:57:32: SCCP:rcvd RegisterRejectMessage
047195: *Sep 25 08:57:32: sccp_appl_service_stop_timer: Stop 431E4BB4 timer
047196: *Sep 25 08:57:32: sccp_parse_control_msg_v1: rcvd register rej, reason: Error: DB
Config
047197: *Sep 25 08:57:32: sccp_initiate_ccm_connect_interval_timer: Start CCM connect
interval timer for prof_id 1, appl_type 8, interval 41 by random number 377
    
```

- A. a DSP farm profile shutdown
- B. a DSP farm profile registration failure due to a TCP connection error
- C. a DSP farm profile registration failure Message due to a mismatched name
- D. that SCCP is not activated

Answer: C

NEW QUESTION 190

- (Exam Topic 2)

An end user at a remote site is trying to initiate an Ad Hoc conference call to an end user at the main site. The conference bridge is configured to support G.711. Remote user's phone only supports G.729. The remote end user receives an error message on the phone: "Cannot Complete Conference Call." What is one cause of the issue?

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

Answer: D

NEW QUESTION 195

- (Exam Topic 2)

Refer to the exhibit.

	Dallas	New York	San Jose	San Francisco
DID Range	(972)555-1XXX	(212)555-1XXX	(408)555-1XXX	(415)555-1XXX
Site Code	1	2	3	4

Intersite calls that are placed across the IP WAN are dialed with the on-net access code 8, followed by the one-digit site code and the four-digit extension of the called party. In order to preserve these dialing habits when the IP WAN is down and Cisco SRST is active, the internal numbers must be converted into E.164 numbers before sending them to the PSTN.

Which configuration allows PSTN calls from the New York site to the San Jose site over the PSTN, while preserving the end user's six-digit dialing habit?

A)

```
!  
voice translation-rule 1  
  rule 1 /^91/ /81972555/  
  rule 2 /^93/ /81408555/  
  rule 3 /^94/ /81415555/  
voice translation-profile on-net-xlate  
  translate called 1  
call-manager-fallback  
  translation-profile incoming on-net-xlate  
dial-peer voice 2 pots  
  destination-pattern 91[2-9]..[2-9].....  
port 1/0:0  
  direct-inward-dial  
  forward-digits 11  
!
```

B)

```
!  
voice translation-rule 1  
  rule 1 /^81/ /91972555/  
  rule 2 /^83/ /91408555/  
  rule 3 /^84/ /91415555/  
voice translation-profile on-net-xlate  
  translate called 1  
call-manager-fallback  
  translation-profile outgoing on-net-xlate  
dial-peer voice 2 pots  
  destination-pattern 91[2-9]..[2-9].....  
port 1/0:0  
  direct-inward-dial  
  forward-digits 11  
!
```

C)

```
!  
voice translation-rule 1  
  rule 1 /^81/ /91972555/  
  rule 2 /^83/ /91408555/  
  rule 3 /^84/ /91415555/  
voice translation-profile on-net-xlate  
  translate called 1  
call-manager-fallback  
  translation-profile incoming on-net-xlate  
dial-peer voice 2 pots  
  destination-pattern 91[2-9]..[2-9].....  
port 1/0:0  
  direct-inward-dial  
  forward-digits 11  
!
```

D)

```

!
voice translation-rule 1
  rule 1 /^81/ /91972555/
  rule 2 /^83/ /91212555/
  rule 3 /^84/ /91415555/
voice translation-profile on-net-xlate
  translate called 1
call-manager-fallback
  translation-profile incoming on-net-xlate
dial-peer voice 2 pots
  destination-pattern 91[2-9]..[2-9].....
port 1/0:0
  direct-inward-dial
  forward-digits 11
!

```

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

Answer: B

NEW QUESTION 199

- (Exam Topic 2)

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. User provided the wrong UserID or PIN
- B. User has no Extension Mobility profiles assigned.
- C. The given user ID is not found in any of the remote clusters.
- D. Phone is not subscribed to Extension Mobility phone service.

Answer: C

NEW QUESTION 203

- (Exam Topic 2)

Refer to the exhibit.

```

dijohn-RTRA#debugisdn q931
*Jun 18 15:57:36.825: ISDN Se0/0/0:15 Q931:RX<SETUP pd = 8 callref=0x0068
  Bearer Capabilityi = 0x8090A3
  Standard = CCITT
  Transfer Capability = Speech
  Transfer Mode = Circuit
  Transfer Rate = 64 kbit/s
  Channel ID I = 0xA98381
  Exclusive, Channel 1
  Calling PartyNumber I = 0x0183, '0981080968'
  Plan: ISDN, Type: National
  Sending Complete
*Jun 18 15:57:36.825: ISDN Se0/0/0:15 Q931:Received SETUP callref = 0x8068 cal
  IID = 0x0088 switch = primary-net5 interface = User

The Voice Gateway has the following dial-peer configuration:
dial-peer voice 1 pots
  incoming called number 0981080968
  port 0/0/0:15

dial-peer voice 2 pots
  incoming called-number 3334445010
  port 0/0/1:15

dial-peer voice 3 pots
  incoming called-number
  port 0/0/0:15

dial-peer voice 4 pots
  destination-pattern 0981080968

```

Which dial peer is matched on the incoming leg (POTS/ISDN)?

- A. dial-peer voice 4 pots
- B. dial-peer voice 3 pots
- C. dial-peer voice 2 pots
- D. dial-peer voice 1 pots

Answer: D

NEW QUESTION 204

- (Exam Topic 2)

Refer to the exhibit.

```
00018161.010 |20:49:58.585 |AppInfo |Digit analysis: patternUsage=5
00018161.011 |20:49:58.585 |AppInfo |Digit analysis: match(pi="2", fqcn="1001",
cn="1001", plv="5", pss="HQ-TRANSLATIONS:HQ-911:HQ-INTERNAL", TodFilteredPss="HQ-
TRANSLATIONS:HQ-911:HQ-INTERNAL", dd="1010", dac="0")
00018161.012 |20:49:58.585 |AppInfo |Digit analysis: analysis results
00018161.013 |20:49:58.585 |AppInfo ||PretransformCallingPartyNumber=1001
|CallingPartyNumber=1001
|DialingPartition=HQ-INTERNAL
|DialingPattern=1010
|FullyQualifiedCalledPartyNumber=101!
|DialingPatternRegularExpression=(1010)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=ForegoPotentialMatches
|DialingsSdlProcessId=(0,0,0)

|PretransformDigitString=1010
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=1010
|CollectedDigits=12310101
```

An engineer recently configured a Cisco Unified Communications Manager cluster. The users are reporting that extensions starting with 10 are routing to a different office. Based on the output, what is the root cause of the issue?

- A. The destination partition is missing from the assigned calling search space
- B. Urgent priority is chosen on a translation or route pattern
- C. The incorrect calling search space was assigned to the phones
- D. The extension is a shared line, and one of the phones is unregistered

Answer: A

NEW QUESTION 208

- (Exam Topic 2)

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. The inter site location is a member of the same cisco unity connection site
- B. The FQDN does not match the remote site gateway configuration file.
- C. The FQDN does not match the server name on the remote site gateway web SSL certificate.
- D. A partition is missing from search spaces on the Cisco Unity Connection location.
- E. DNS is not configured on the Cisco Unity Connection site gateway.

Answer: C

NEW QUESTION 212

- (Exam Topic 2)

Refer to the exhibit.

```
!
sccp local Vlan5
sccp ccm 10.10.5.1 identifier 5 version 8.6.2
sccp
!
sccp ccm group 5
bind interface Vlan1
associate ccm 5 priority 1
associate profile 5 register MTPHW1
!
dspfarm profile 5 mtp
codec g711ulaw
codec pass-through
maximum sessions software 500
!
```

You are trying to configure a new Cisco IOS MTP to register with Cisco Unified Communications Manager, but it fails to register. Which two changes are required so that the MTP registers successfully? (Choose two.)

- A. Restart the router.
- B. Bind the interface to Vlan5.
- C. The sccp ccm group should be 1.
- D. Associate the application with SCCP.
- E. Change the maximum sessions to 1.

Answer: BD

NEW QUESTION 217

- (Exam Topic 2)

A system administrator observes that settings on an endpoint revert to unknown values every day, without the administrator's knowledge. What are two likely causes of this issue? (Choose two.)

- A. Persistent settings on the endpoint are causing the issue.
- B. A persistent template is applied for the endpoint on the TMS.
- C. The external server periodically restores the configuration on the endpoint.
- D. The TMS reverts the configuration backup of the endpoint.
- E. The endpoint can be set periodically revert to a specific configuration.

Answer: AD

NEW QUESTION 219

- (Exam Topic 2)

Refer to the exhibit.

```

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

NAME: "VVIC3-IMFT-T1/E1 - 1-Port RJ-48 Multiflex Trunk - T1/E1 on slot 0 subslot 0", DESCR: "VVIC3-IMFT-T1/E1"
PID: VVIC3-IMFT-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 36, 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. 21
- B. 14
- C. 60
- D. 15

Answer: B

NEW QUESTION 221

- (Exam Topic 2)

Calls drop right after the called party tries to answer the call. Which of the below reasons is true?

- A. Phone A is using SCCP and phone B is using SIP
- B. Firewall is blocking RTP packets
- C. An incompatible MTP type is allocated to the call
- D. Codec are mismatched between phones

Answer: D

NEW QUESTION 224

- (Exam Topic 2)

You have sites across a WAN and because of a recent change on the access control lists, your SIP phones are having registration issues. Which cause is likely?

- A. TCP/UDP 0506 is blocked by the ACL.
- B. TCP/UDP 5006 is blocked by the ACL.
- C. TCP/UDP 5060 is blocked by the ACL.
- D. TCP/UDP 6050 is blocked by the ACL.

Answer: C

NEW QUESTION 227

- (Exam Topic 2)

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 DMI is configured with the correct IP subnets.
- B. Verify that the Cisco IOS devices on the VPN support audio connections.
- C. Verify that the VLANs at remote locations are configured correctly.
- D. Verify that users at the remote locations are connecting to the closest enterprise VPN concentrator.
- E. Verify that the firewall is allowing RTP traffic flow.

Answer: BE

NEW QUESTION 229

- (Exam Topic 2)

Which debug command analyzes messages that are produced by SIP during the call setup process in IOS?

- A. Show isdn status
- B. debug voip ccapi inout
- C. show sip-ua register status
- D. debug isdn q931
- E. debug ccsip messages
- F. debug voice dialpeer

Answer: E

NEW QUESTION 233

- (Exam Topic 2)

What does this trace indicate? Refer to the exhibit.

```

17:12:38.525 (StationID: (0003212) OpenReceiveChannelAck Status=0, ipAddr=ipAddr type0
ipAddr:0xac545077000000000000000000000000(172.100.80.119), Port=29458, PartyID=33712903(2,100,50,1,41500994
17:12:38.529 (StationID: (0003212) CallStateCallState=3 lineInstance=1 callReference=35181381 privacy=0 scp_precedence=4
precedence=0(2,100,50,1,41500715
17:12:38.529 (StationID: (0003212) SelectSoftKeys instance=1 reference=35181381 softKeySetIndex=8 validKeyMask=ffffff(
2,100,50,1,41500715
17:12:38.529 (StationID: (0003212) DisplayPromptStatus timeout=0 Status='C' content='Ring Out' line=1 C=35181381 ver=85720014(
2,100,50,1,41500715
17:12:38.529 (StationID: (0003212) DEBUG_star_DSetCallState(7) State of cdpc(156589) is 6 (2,100,50,1,41500715
17:12:43.029 (StationID: (0003212) restart0_CcSetupRes: updateACall=35181381, new cm_prec=5(2,100,50,1,41500715
17:12:43.029 (StationID: (0003212) ReceivedCSetupRes - callSecurityStatus=1 (2,100,50,1,41500715
17:12:43.029 (StationID: (0003212) star_CcNotifyReq - CallSecurityStatus=1 (2,100,50,1,41500715
17:12:43.029 (StationID: (0003212) CallStateCallState=5 lineInstance=1 callReference=35181381 privacy=0 scp_precedence=4
precedence=0(2,100,50,1,41500715
17:12:43.029 (StationID: (0003212) SelectSoftKeys instance=1 reference=35181381 softKeySetIndex=1 validKeyMask=ffffff(
2,100,50,1,41500715
17:12:43.029 (StationID: (0003212) DisplayPromptStatus timeout=0 Status='C' content='Connected' line=1 C=35181381 ver=85720014(
2,100,50,1,41500715
17:12:43.029 (StationID: (0003212) StopTone(2,100,50,1,41500715
17:12:43.029 (StationID: (0003212) DEBUG_star_DSetCallPhase updateACall=35181381 from Phase=0 to callPhase=1 (2,100,50,1,41500715
17:12:43.029 (StationID: (0003212) DEBUG_star_DSetCallState(10) State of cdpc(156589) is 7 (2,100,50,1,41500715
17:12:54.031 (StationID: (0003212) StationMediaPathEvt: Handset (2) = Off (2)(2,100,50,1,41501251
17:12:54.039 (StationID: (0003212) OnHook(2,100,50,1,41501252
17:12:54.039 (StationID: (0003212) StopTone(2,100,50,1,41501252
17:12:54.039 (StationID: (0003212) CloseReceiveChannel conferenceID=35181381 passThruPartyID=33712903 myIP: ipAddr type0
ipAddr:0xac545077(172.100.80.119) (2,100,50,1,41501252
17:12:54.039 (StationID: (0003212) StopMediaTransmission conferenceID=35181381 passThruPartyID=33712903 myIP: ipAddr type0
ipAddr:0xac545077(172.100.80.119) (2,100,50,1,41501252

```

- A. a disconnected call
- B. a Completed call setup
- C. a failed call setup
- D. an interrupted call setup

Answer: B

NEW QUESTION 234

- (Exam Topic 2)

Refer to the exhibit.

The image shows two configuration pages from Cisco Unified Communication Manager. The top page is 'SAF Security Profile Configuration' with fields for Name (SAF Security Profile 1), Description, User Name (safuser), and User Password (secretsauce). The bottom page is 'SAF Forwarder Configuration' with fields for Name (SAF-Forwarder), Description, Client Label (cucm), SAF Security Profile (SAF Security Profile 1), SAF Forwarder Address (192.168.91.220), and SAF Forwarder Port (5050). To the right is a snippet of router configuration for R1:

```
#R1
!
interface Loopback0
 ip address 192.168.91.220 255.255.255.255
!
router eigrp saf
!
 service-family ipv4 autonomous-system 1
!
 topology base
 external-client cucm
 exit-sf-topology
 exit-service-family
!
 service-family external-client listen ipv4 5050
 external-client cucm
 username safuser
!
#R1
```

Cisco Unified Communication Manager has been configured with a SAF Security Profile and a SAF Forwarder. In which section of the R1 router configuration should the SAF Forwarder password be entered?

- A. interface loopback 0
- B. router eigrp saf
- C. service-family external client
- D. service-family ipv4

Answer: C

NEW QUESTION 235

- (Exam Topic 2)

Which command do you use to confirm that a router interlace is enableDFor SAP?

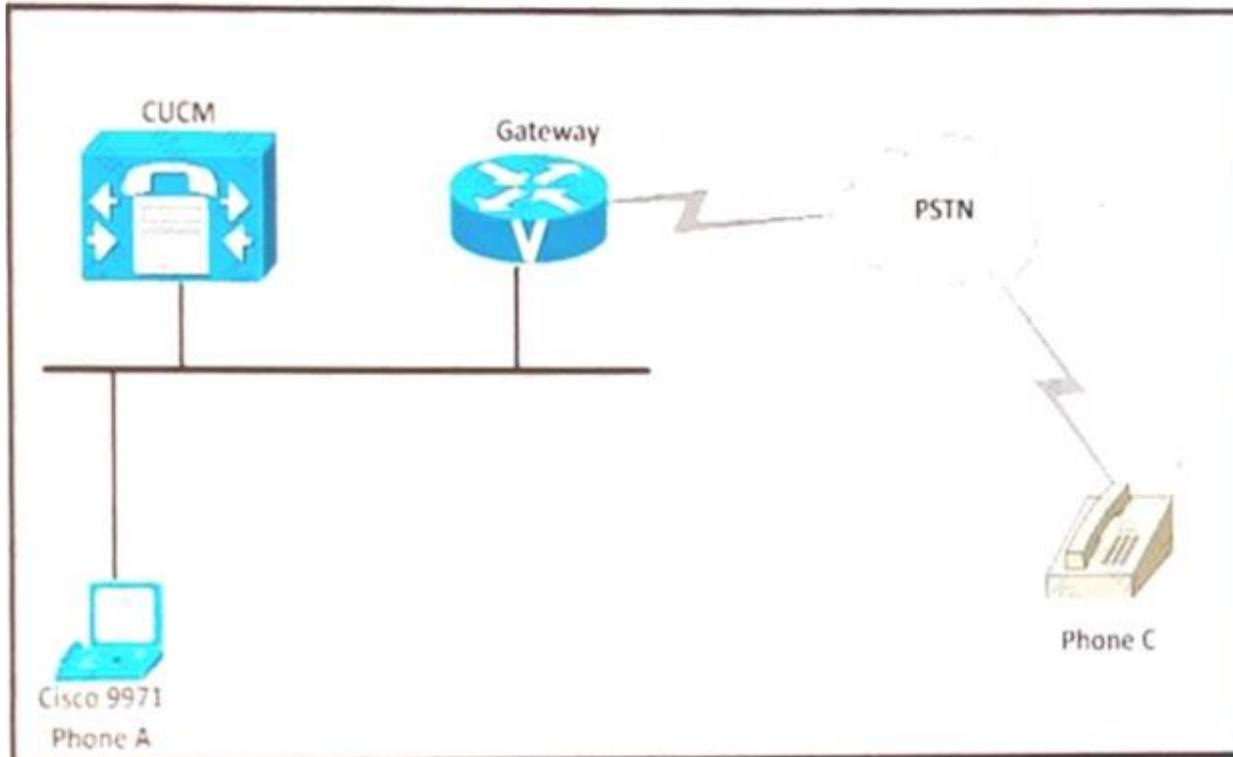
- A. show eigrp service-family ipv4 client details.
- B. show ip interface details
- C. show ip saf-service-family interface
- D. show run
- E. show eigrp service -family interface
- F. show eigrp service-family ipv4 <AS number interfaces

Answer: E

NEW QUESTION 238

- (Exam Topic 2)

Refer to the exhibit.



When Phone A makes a Call to Phone C, this message is observed Cause i = 0x80C1 – Bearer capability not implemented. How do you resolve this error?

- A. Reset Phone A
- B. Migrate Phone A from SCCP to SIP

- C. Set the bearer capability on the gateway to match the provider capability
- D. Open a Cisco TAC case

Answer: C

NEW QUESTION 240

- (Exam Topic 2)

Refer to the exhibit.

```

May 17 22:02:45.043: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 203.84.23.194:5060;branch=z9hG4bK1sdq7c20dgrhm3l0u2n1cdhpsvsj2.1
From: "John Ford" <sip:734561290@10.88.124.108;user=phone>;tag=433298465-1431900233544-
To: "882931316 882931316" <sip:882931316@cisco.com>;tag=15428C-15CF
Date: Sun, 17 May 2015 22:02:45 GMT
Call-ID: BW080353544180515-1909915730@10.83.154.138
Server: Cisco-SIPGateway/IOS-15.4.3.M
CSeq: 838061991 BYE
Reason: Q.850;cause=16
Content-Length: 0

May 17 22:02:45.043: //19/3B918B06800E/SIP/Msg/ccsipDisplayMsg:
Sent:
BYE sip:882931316@10.0.10.2:5060 SIP/2.0
Via: SIP/2.0/UDP 10.0.10.8:5060;branch=z9hG4bKA22B1
From: "John Ford" <sip:00734561290@cisco.com>;tag=15424C-1FB
To: <sip:882931316@10.0.10.2>;tag=1130116-90b05bfa-5f5d-4e05-8607-205de3dbe498-20784137
Date: Sun, 17 May 2015 22:02:26 GMT
Call-ID: 3B92C356-FC1711E4-8014AC6D-9A5A5580@10.0.10.8
User-Agent: Cisco-SIPGateway/IOS-15.4.3.M
Max-Forwards: 70
Timestamp: 1431900165
CSeq: 103 BYE
Reason: Q.850;cause=16
Content-Length: 0
    
```

According to the log diagram, what is the reason that the call ended?

- A. The call was put on hold.
- B. The call was experiencing one-way audio.
- C. The call was completed successfully.
- D. The call was transferred.
- E. The call dropped out.

Answer: C

NEW QUESTION 243

- (Exam Topic 2)

Which two tasks are performed by the RAS signaling function of H.225.0? (Select two.)

- A. Performs bandwidth changes.
- B. Transports audio messages between endpoints.
- C. Performs disengage procedures between endpoints and a gatekeeper.
- D. Allows endpoints to create connections between call agents.

Answer: AC

NEW QUESTION 248

- (Exam Topic 2)

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 desensitive.
- B. Adjust the URI lookup policy to case sensitive.
- C. Adjust the URI lookup policy to case insensitive.
- D. Adjust the URI lookup policy to case nonsensitive.

Answer: C

NEW QUESTION 250

- (Exam Topic 2)

You try to register a new H.323 endpoint to a Cisco VCS server, but the endpoint does not register correctly. You successfully check network connectivity and VCS registration with other devices. Which issue and solution will resolve this registration problem?

- A. Cisco VCS Server H.323 service must be restarted to activate the new endpoint.
- B. The Registration Policy is set to "Deny List" so all new endpoints are rejecte
- C. Set the Policy to "Allow List" so that the endpoint can register themselves and then reassign the "Deny List" Setting for security reasons.
- D. The new endpoint has not been assigned with the correct registration domain, so configure the correct domain on the endpoint.
- E. The Registration Policy is set to "Allow List" and the newly addeDEndpoint has to be added to the list, because only listed devices are allowed to register.

Answer: D

NEW QUESTION 254

- (Exam Topic 2)

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. Verify that the RJ-11 cable is plugged into the PC port.
- B. Verify that the phone's network can access the option 150 server.
- C. Disable HSRP on the access layer switch.
- D. Verify that the correct drivers are installed on the switch port of the phone.
- E. Verify that the switch port of the phone is enabled.
- F. Check the RJ-65 cable.

Answer: BE

NEW QUESTION 259

- (Exam Topic 2)

Refer to the exhibit.

```
dspfarm profile 1 mtp
  codec g711ulaw
  maximum sessions software 100
  associate application SCCP
```

Media Termination Point is required when the H.264 codec-capable video endpoints route calls via a specific Cisco Unified Communications Manager SIP trunk, in order to provide DTMF interworking capabilities. In the exhibit, a designated IOS Software MTP is configured. However, calls establish as audio only when using this particular MTP. Which codec is missing from this IOS MTP configuration?

- A. codec pass-through
- B. codec h264 cif
- C. codec h263 cif
- D. codec h264 720p

Answer: A

NEW QUESTION 264

- (Exam Topic 2)

An engineer is troubleshooting an existing dial plan and encounters the dial pattern: 12[0-5]X Which directory numbers are matched by this pattern?

- A. 120 through 125
- B. 1200 through 1250
- C. 1200 through 1259
- D. all numbers starting with 12

Answer: A

NEW QUESTION 267

- (Exam Topic 2)

Which three statements about SAF forwarding are true? (Choose three.)

- A. It supports DNS and IP addresses.
- B. The SAF Forwarder configuration can be updated without connecting to Cisco Unified Communications Manager.
- C. It requires the SAF Forwarders to use the same configuration as the IOS switch.
- D. It is supported on IPv4 only.
- E. It is supported on IPv6 only.
- F. It requires a unique IP address for each SAF Forwarder.

Answer: ABF

NEW QUESTION 272

- (Exam Topic 2)

What is the meaning of this alarm snippet

```
CCM_CALLMANAGER-CALLMANAGER-3-EndPointTransientConnection?
```

- A. The device was successfully unregistered.
- B. The device was reset via the reset button in Cisco Unified Communications Manager.
- C. The device was misconfigured.
- D. The device was successfully registered.

Answer: C

NEW QUESTION 277

- (Exam Topic 2)

Refer to the exhibit.

```
2015-02-02T19:46:31+01:00 collabedge tvcs: UTCTime="2015-02-02 18:46:31,144"  
Module="network.tcp" Level="DEBUG": Src-ip="JabberPubIP" Src-port="4211"  
Dst-ip="VCS-E_IP" Dst-port="5061" Detail="TCP Connecting"  
2015-02-02T19:46:31+01:00 collabedge tvcs: UTCTime="2015-02-02 18:46:31,144"  
Module="network.tcp" Level="DEBUG": Src-ip="JabberPubIP" Src-port="4211" Dst-ip=  
"VCS-E_IP" Dst-port="5061" Detail="TCP Connection Established"2015-02-  
02T19:46:49+01:00  
collabedge tvcs: UTCTime="2015-02-02 18:46:49,606"  
Module="network.tcp" Level="DEBUG": Src-ip="92.90.21.82" Src-port="4211" Dst-ip=  
"VCS-E_IP" Dst-port="5061" Detail="TCP Connection Closed" Reason="Idle countdown  
expired"
```

While reviewing Expressway-E logs, an engineer encounters an idle countdown expireDError that occurred when a Jabber client tried to register. There is a firewall in front of the Jabber client. Which cause of this error is true?

- A. SIP inspection is enabled on the firewall
- B. SIP inspection is disabled on the firewall
- C. UDP port 5061 is not allowed through the firewall
- D. TCP port 5061 is not allowed through the firewall

Answer: A

NEW QUESTION 281

- (Exam Topic 2)

A network administrator is troubleshooting a support ticket with ID33118456 regarding video bandwidth issues. On a Cisco TelePresence VCS, the administrator can configure bandwidth control for which two VCS configuration options? (Choose two)

- A. links and pipes
- B. subzones
- C. CPL
- D. Bandwidth restrictions can be configured only on endpoints.
- E. zones

Answer: AB

NEW QUESTION 286

- (Exam Topic 2)

In a Cisco Unified Communications Manager call trace, which string indicates an intrasite call?

- A. OutsideDialtone = [1]
- B. cmDeviceType=[AccessDevice]
- C. OutsideDialtone= [0]
- D. OutsideDialtone-[2]

Answer: A

NEW QUESTION 291

- (Exam Topic 2)

On a VCS, which CLI command resolves the issue represented by this error message?

"Cluster replication error: this peer's configuration conflicts with the master's configuration, manual synchronization of configuration is required"

- A. xCommandFeedbackDeregister ID: 1
- B. xcommandForceConfigUpdate
- C. xCommandEdgessopurgetokens
- D. xCommand PolicyServiceDelete PolicyServiceid: 1

Answer: B

NEW QUESTION 295

- (Exam Topic 2)

Partitions can be assigned to which two items? (Choose two)

- A. directory numbers
- B. trunks
- C. devices
- D. gateways
- E. IP phones

Answer: AD

NEW QUESTION 296

- (Exam Topic 2)

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. Enable the endpoint voice interface for backhauling in interface configuration mode with the isdn bind-l3 ccm-manager command
- D. Ensure that you have configured the ccm-manager call-agent command with the correct Cisco Unified Communications Manager server address or hostname.
- E. Activate the MGCP Feature service on all call-processing servers in the cluster.

Answer: BD

NEW QUESTION 301

- (Exam Topic 2)

Users on your network report fast busy signals when they attempt to place calls. While troubleshooting, you verify that the codecs are configured correctly and determine that the network carries up to 200 concurrent calls at different times of the day. Which two actions correct the problem? (Choose two)

- A. Add more Cisco Unified Communications Managers to the cluster.
- B. Add more DSPs to the NM-HD
- C. Increase the number of sessions to the maximum allowed by system resources.
- D. Implement an SRST-CMF gateway to improve resource allocation.
- E. Add a Cisco Catalyst switch to the environment to increase port density
- F. Bind the network PRIs to add additional bandwidth

Answer: CF

NEW QUESTION 303

- (Exam Topic 2)

A user takes an IP phone from one office to another and just called you to inform that the phone does not work in the new location. Which three things should you check to resolve the issue? (Choose three)

- A. Make sure that the device mobility information contains the correct IP subnet information.
- B. Make sure that physical locations are properly assigned under device pools.
- C. Recreate the phone under a new cluster.
- D. Make sure that the device pool is assigned to a device mobility group.
- E. Reconfigure the remote destination profile.
- F. Check whether the phone is registered under a different MAC address.
- G. Check the local DHCP information for possible clues.

Answer: ABG

NEW QUESTION 308

- (Exam Topic 2)

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 ras
- C. debug voip ccapi inout
- D. debug mgcp errors
- E. debug ccsip error

Answer: C

NEW QUESTION 313

- (Exam Topic 2)

After a user successfully logs in to a phone by using Cisco Extension Mobility, the user does not have the option to log out. What can cause this issue?

- A. The login/logout button must be added to the user device profile.
- B. The Cisco Extension Mobility service is malfunctioning.
- C. The user device profile is not subscribed to the Cisco Extension Mobility service.
- D. The phone must be reset instead of restarted.

Answer: C

NEW QUESTION 314

- (Exam Topic 2)

Refer to the exhibit.

```
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 28052 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
```

Based on the ACK SDP output, what can be concluded?

- A. There is no DTMF support for the call.
- B. The total amount of bandwidth for the video channels is 192 kbps.
- C. High-definition video is configured for the endpoints
- D. This event is a multipoint video call with three participants.

Answer: D

NEW QUESTION 318

- (Exam Topic 2)

When troubleshooting a "disconnect code 65" on Cisco Unified Border Element, which option is the likely cause in the configuration?

- A. missing dial-peer
- B. codec mismatch
- C. dtmf-relay misconfiguration
- D. no IP address trust list

Answer: B

NEW QUESTION 321

- (Exam Topic 2)

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 the Cisco Unified Border Element is running.
- B. Verify the search pattern.
- C. Verify the route pattern, route list, and route group.
- D. Check that Cisco VCS Control and Cisco VCS Express are getting through the firewall.
- E. Verify that the PSTN line is connected to the Cisco Unified Communications Manager.
- F. Verify that the connection to and from the Cisco Unified Border Element is good.

Answer: ACE

NEW QUESTION 326

- (Exam Topic 2)

After you deploy SAF in a megACLuster and you verify that TCP port 5050 is open, you notice that SAF connections are failing. Which two functions must you check to troubleshoot the problem? (Choose two.)

- A. Cisco AXL Web service
- B. Cisco UP XCP Router service
- C. Connection conversation Messenger
- D. SMTP server
- E. CCD advertising service
- F. CCD requesting service

Answer: AB

NEW QUESTION 328

- (Exam Topic 2)

Which two Cisco Unified Communications Manager operations will test reverse DNS lookup? (Choose two.)

- A. Installation
- B. Reboot
- C. Upgrade
- D. Switch version
- E. Changing IP address or hostname

Answer: AE

NEW QUESTION 330

- (Exam Topic 2)

Refer to the exhibit.

```
SIP URL, Alphanumeric User, uri=JOE @cisco.com
28022445.003 |16:03:40.531 |AppInfo |Digit Analysis: star_DaReq: Matched
URI in remoteroutingpatternable.uri = JOE @cisco.com, routeString=Americas
28022445.004 |16:03:40.531 |AppInfo |star_DaReq: Attempt Match on Route
String Host: Americas
28022445.005 |16:03:40.531 |AppInfo |star_DaReq: new ils callableEndpoint
28022445.006 |16:03:40.531 |AppInfo |star_DaReq: sendDMPidReq
28022446.000 |16:03:40.531 |SdISig |DmPidReq                |initialized
|DeviceManager(2,100,205,1)    |Da(2,100,211,1)
|2,100,14,1042.580^172.24.18.24^SEP1CDF0F772424 |[T:N-
H:0,N:0,L:0,V:0,Z:0,D:0] Cepn=Id=3832024920
28022447.000 |16:03:40.531 |SdISig |DmPidErr                |wait
|Da(2,100,211,1)              |DeviceManager(2,100,205,1)
|2,100,14,1042.580^172.24.18.24^SEP1CDF0F772424 |[R:N-
H:0,N:0,L:0,V:0,Z:0,D:0] Cepn=Id=3832024920 Pid=0,0,0,0
28022448.000 |16:03:40.531 |SdISig |DaRes                    |setup_da
|Cdcc(2,100,219,5315)         |Da(2,100,211,1)
|2,100,14,1042.580^172.24.18.24^SEP1CDF0F772424 |[R:N-
H:0,N:0,L:0,V:0,Z:0,D:0] CI=37720843 Block NoPotentialMatchesExist
OnNetpatternUsage=28requestID=0
28022449.000 |16:03:40.531 |SdISig |CcDisconnReq
```

A network administrator in Cisco recently configured ILS in the network. The administrator attempted a Call to Joe's SIP URI (JOE@cisco.com), but the call failed. The administrator collected Cisco CallManager traces using Cisco Unified Communications Manager RTMT. What should the administrator do to solve this issue?

- A. Create a route pattern for "americas" to route out the SIP trunk that is configured for the remote ILS cluster.
- B. Joe's phone is currently using the SIP protocol
- C. Change the protocol to SCCP.
- D. Use the utils dbreplication reset all command to synchronize databases between ILS clusters.
- E. Check the "MTP Required" check box on the ILS trunk.

Answer: A

NEW QUESTION 333

- (Exam Topic 2)

An engineer implemented a new voice gateway using H.323 to integrate with Cisco Unified Communications Manager (CUCM) and is experiencing issues routing calls via the gateway, even though the network connectivity is working. The engineer wants to verify that the gateway is properly configured. What is an indication of correct configuration?

- A. Successful ping from CUCM to the gateway
- B. Gateway IP address in CUCM
- C. Trace route from the gateway to CUCM
- D. Gateway registration status in CUCM

Answer: B

NEW QUESTION 336

- (Exam Topic 2)

Which two statements about software MTP devices are true? (Choose two.)

- A. The RTMT can monitor the number of registered and in-use MTPs.

- B. A single MTP can register with multiple Cisco Unified Communications Managers.
- C. Each server can support up to four instances of the Cisco IP Voice Media Streaming Application.
- D. The Cisco IP Voice Media Streaming Application can reduce the performance of the Cisco Unified Communications Manager if it is installed on the same server.
- E. When you configure and restart the MTP, the changes are applied immediately to active calls.
- F. When you configure and reset the MTP, the changes are applied immediately to active calls.

Answer: AE

NEW QUESTION 341

- (Exam Topic 2)

In a Cisco UCM multisite WAN with centralized call-processing deployment model, what redundancy feature should be configured on remote site routers to provide basic IP telephony services in the event of a WAN outage?

- A. AAR
- B. SRST
- C. CAC
- D. V3PN

Answer: B

NEW QUESTION 343

- (Exam Topic 2)

An administrator decides to factory reset a Cisco TelePresence System EX90 after the log analysis failed to provide the reason for an intermittent one-way audio issue.

What can the Administrator expect after the reset is performed?

- A. The release keys and option keys are deleted.
- B. The release keys and option keys are preserved.
- C. The system needs to be manually powered up after the factory reset.
- D. The call logs are preserved.

Answer: C

NEW QUESTION 344

- (Exam Topic 2)

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. undebg all
- C. no logging console
- D. logging buffered
- E. show debug
- F. show logging info

Answer: CD

NEW QUESTION 349

- (Exam Topic 2)

A Call transfer from a video phone on Cisco Unified CM to VCS fails. Which is true?

- A. The DTMF signaling method was set to RFC 2833
- B. The UCM security mode was misconfigured
- C. The DTMF signaling method was set to no preference
- D. The SIP trunk between UCM and VCS was reset

Answer: A

NEW QUESTION 352

- (Exam Topic 2)

What is the default maximum login time for a user in Cisco Extension Mobility?

- A. 1 hour
- B. 2 hours
- C. 4 hours
- D. 8 hours
- E. 10 hours
- F. 12 hours
- G. 4 hours
- H. 8 hours
- I. 10 hours
- J. 12 hours
- K. 8 hours
- L. 10 hours
- M. 12 hours
- N. 10 hours

- O. 12 hours
- P. 12 hours

Answer: D

NEW QUESTION 353

- (Exam Topic 2)

During a Conference that is hosted on a Cisco TelePresence Server, which three circumstances determine that some participants do not have video, but they do have audio? (Choose three.)

- A. The participants have no video component.
- B. No free screen licensing ports are available.
- C. No video ports are available in a slave Cisco TelePresence Server configuration.
- D. The maximum number of participants is exceeded.
- E. In all Cisco TelePresence deployments, some participants are connected only via audio.
- F. No video ports are available in a single Cisco TelePresence Server configuration.

Answer: BCF

NEW QUESTION 354

- (Exam Topic 2)

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 card type t1 command
- B. Use the show call active voice brief command
- C. Use the show inventory command
- D. Use the feature activate isdn command
- E. Use the clock source line command

Answer: BE

NEW QUESTION 356

- (Exam Topic 2)

ACustomer reports trial calls that are made through the PSTN gateway drop after few seconds of being placed on mute. Which MGCP configuration command can you issue in the gateway to resolve this problem?

Refer to the exhibit.

```

mgcp
mgcp call-agent 172.16.240.124 2427 service-type mgcp version 0.1
cm-manager mgcp
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode cisco
mgcp sdp simple
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp timer receive-rtcp 10
no mgcp explicit hookstate
isdn switch-type primary-ni
call rsvp-sync
  
```

- A. mgcp explicit hookstate
- B. mgcp max-walling-delay
- C. no mgcp rtp unreachable timeout
- D. no mgcp timer receive-rtcp

Answer: C

NEW QUESTION 360

- (Exam Topic 2)

A user reports that video quality is good when video is available, but there are constant interruptions to video. Which possible cause of this issue is true?

- A. The bandwidth is too high
- B. A satellite connection joins the meeting
- C. The codeCExperiences packet loss
- D. The multiplexor negotiation timeout occurs

Answer: D

NEW QUESTION 363

- (Exam Topic 2)

When you we dialing from an internally registered device on Cisco VCS, m which order are the options processed lo determine the call routing?

- A. CPL, Transforms, FindMe, Search Rules
- B. CPL, FindMe, Transforms, Search Rules
- C. Transforms, CPL, FindMe, Search Rules

- D. Search Rules, Transforms, CPL, FindMe
- E. FindMe, CPL, Transforms, Search Rules

Answer: C

NEW QUESTION 366

- (Exam Topic 2)

Which Cisco Unified Communications Manager tool can you use to troubleshoot issues with international calling?

- A. Cisco Prime
- B. Cisco Deployment Tool
- C. RTMT
- D. Dialed Number Analyzer

Answer: C

NEW QUESTION 367

- (Exam Topic 2)

In Cisco Unified Communications Manager (CUCM), you have three Service Advertisement Framework (SAF) forwarders configured, what happens when the primary and the backup SAF forwarders fail?

- A. You will need to designate another primary SAF forwarder.
- B. CUCM continues to work without connecting to the third SAF forwarder.
- C. The third SAF forwarder automatically becomes the primary SAF forwarder.
- D. CUCM tries to reconfigure the existing primary and backup SAF forwarders.
- E. TCP timer continues to initiate connection with the primary and backup SAF forwarders.
- F. Primary and backup SAF forwarders are re-initiated automatically.

Answer: B

NEW QUESTION 372

- (Exam Topic 2)

Which two protocols are required for a Cisco 8800 series IP phone to successfully register with CUCM? (Choose two)

- A. CDP
- B. HTTP
- C. TFTP
- D. LLDP
- E. SIP
- F. DHCP
- G. SCCP

Answer: CE

NEW QUESTION 374

- (Exam Topic 2)

The link from your local Cisco Unity Connection site to another site has gone down. While troubleshooting, you discover that the local gateway is unable to reach a DNS server. Which action can you take to reestablish the link?

- A. Restart the Connection Manager series on the local gateway
- B. Create an intersite link manually using the IP address of the remote gateway.
- C. Synchronize the two Cisco Unity Connection sites.
- D. Create an intersite link manually using the FQDN of the remote gateway.
- E. Configure the directory synchronization task schedule on the remote gateway.

Answer: A

NEW QUESTION 377

- (Exam Topic 2)

Refer to the exhibit.

19:14:54 Configuring IP

19:15:00 Updating Trust List

19:15:01 Trust List update failed

After migrating a Cisco IP phone to a new cluster, the phone continues to register with its old Cisco Unified Communications Manager cluster. The Cisco IP phone status message indicates the error that is shown. What is a possible cause for this error condition?

- A. The phone cannot reach its new TFTP server.
- B. The phone is not provisioned correctly on the destination cluster.
- C. The new TFTP server is not in the ITL file.
- D. Cisco CallManager services are not running on the destination cluster.

Answer: C

NEW QUESTION 382

- (Exam Topic 2)

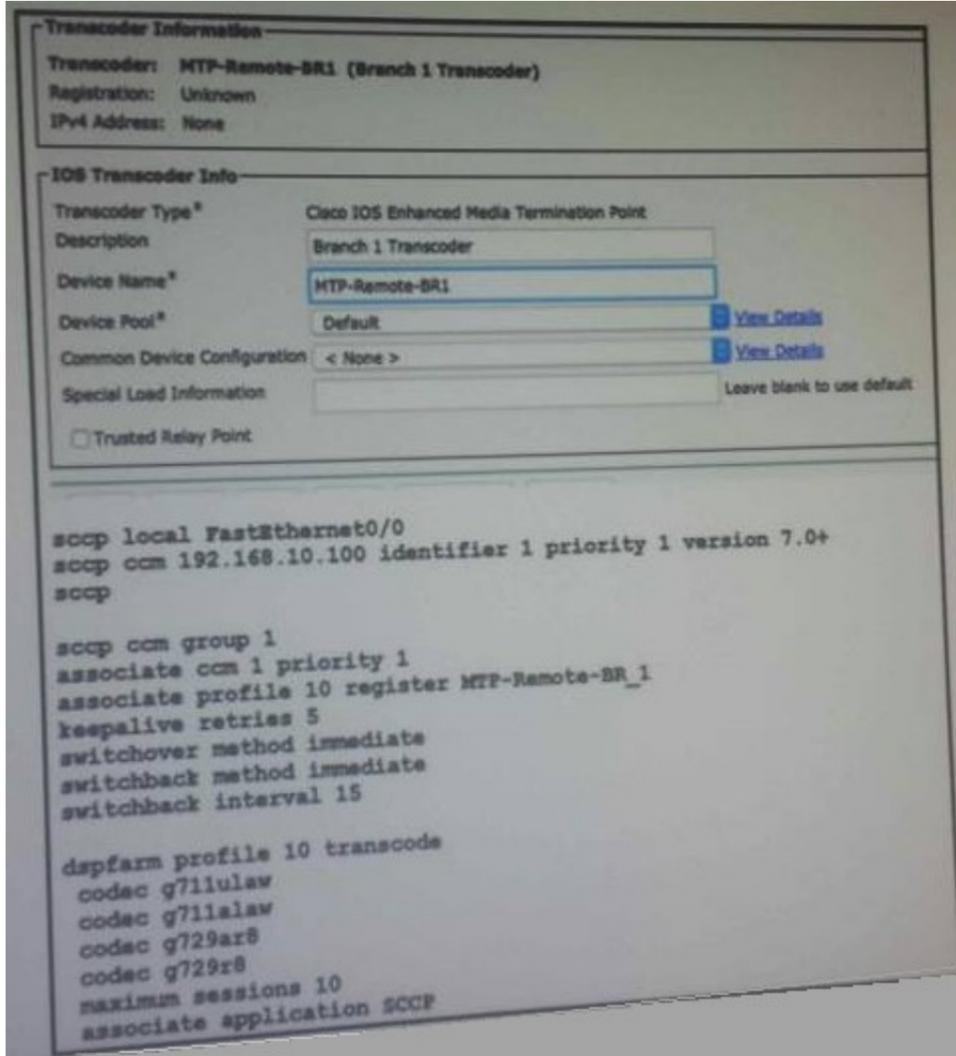
You discover that ACCD client is failing to learn patterns with RTMT. Which two actions can you take to troubleshoot the problem? (Choose two.)

- A. Verify that the SIP trunk between Cisco Unified Communications Manager and the Cisco Unified Presence Server is configured correctly.
- B. Verify that the trunk is selected for the CCD advertising service.
- C. Verify that the trunk is selected for the CCD requesting service
- D. Verify that EIGRP topology information is available to the SAF Forwarders.
- E. Verify the SAF configuration.

Answer: CE

NEW QUESTION 383

- (Exam Topic 2)



Refer to the exhibit. An engineer is troubleshooting an issue with a newly configured transcoder. After the engineer configures the IOS device and Cisco Unified Communications Manager (CUCM), the device will not register. What must the engineer reconfigure to fix this issue?

- A. the device type on CUCM
- B. the device pool on the transcoder
- C. the router to match the profile name
- D. the router to remove any unnecessary codecs

Answer: A

NEW QUESTION 388

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