

Exam Questions 300-080

Troubleshooting Cisco IP Telephony and Video

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NEW QUESTION 1

- (Exam Topic 1)

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

- A. RTMT trace output
- B. Cisco IOS debug commands
- C. Dialed Number Analyzer output
- D. Cisco Unified Communications Manager alerts
- E. Cisco IOS show commands

Answer: BE

NEW QUESTION 2

- (Exam Topic 1)

An IP phone that is connected through a Cisco Catalyst 3750 Series Switch is failing to register with the subscriber as ABackup server. When the user presses the settings button on the phone, only the Cisco Unified Communications Manager publisher shows as registered. What is the most likely cause for this issue?

- A. The phone does not have the correct Cisco Unified Communications Manager group in the device configuration page.
- B. The Cisco Unified Communications Manager group that is applied through the device pool is misconfigured.
- C. The ip-helper address command for the subscriber is not configured on the switch port.
- D. The subscriber does not have the correct device pool configured.
- E. The enterprise phone configuration does not have the call control redundancy enabled.

Answer: B

NEW QUESTION 3

- (Exam Topic 1)

When a remote endpoint dials in to join a Conference that is configured on a Cisco TelePresence Server bridge, the endpoint receives only audio. Other users can successfully join the call with Voice and Video. What is causing this issue?

- A. The endpoint does not have the multisite option installed.
- B. The endpoint does not have the partition of the bridge in its CSS.
- C. The bridge is out of all licenses.
- D. The endpoint is assigned a region without enough configured bandwidth for video.
- E. The bridge is not able to host video calls.

Answer: D

NEW QUESTION 4

- (Exam Topic 1)

Which meaning does the universal keyword have in this dspfarm profile configuration?

```
dspfarm profile 1 transcode universal
codec g711alaw
codec g711ulaw
codec g722-64
codec g729r8
codec g729ar8
maximum sessions 4
associate application SCCP
```

- A. The profile allocates DSP resources in flexible mode.
- B. The profile allows transcoding between any configured codecs.
- C. The profile can be used not only as transcoder but also as MTP.
- D. No special meaning, this is the default setting.

Answer: B

NEW QUESTION 5

- (Exam Topic 1)

Some users report that they cannot dial out from headquarters on their Cisco IP Phones to PSTN users, but others can. Which troubleshooting approach is the most direct to isolate the source of the failure of the users that cannot dial out to the PSTN?

- A. Use DNA to analyze the dialing permissions of the Cisco IP Phones.
- B. Use DNA to generate actual calls to the PSTN.
- C. Use RTMT to analyze the dialing permissions of the Cisco IP Phones.
- D. Use RTMT to generate actual calls to the PSTN.

Answer: A

NEW QUESTION 6

- (Exam Topic 1)

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

- A. There is no dial route for 9XXXXXXX on Cisco Unified Communications Manager.
- B. The gateway is not dropping the leading 9, and the PSTN fails.
- C. The T302 timer is waiting to expire.
- D. The caller does not have the PSTN partition in the CSS.
- E. The caller dialed the wrong number.
- F. To dial successfully, the caller must enter a Forced Authorization Code.

Answer: C

NEW QUESTION 7

- (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 8

- (Exam Topic 1)

To maintain proper database integrity, what is the recommended maximum round-trip delay between multiple Cisco VCS appliances in ACluster?

- A. 10 ms
- B. 15 ms
- C. 25 ms
- D. 30 ms
- E. 50 ms
- F. 80 ms

Answer: D

NEW QUESTION 9

- (Exam Topic 1)

After you install the Cisco Jabber client, it fails to register with the Cisco Unified Communications Manager server. Which two actions can you take to troubleshoot the problem? (Choose two.)

- A. Verify that the DNS SRV record is correctly configured.
- B. Verify that the LMHOST file is installed on the PC.
- C. Verify that the corporate firewall allows connections to andFrom the Jabber client.
- D. Reboot the Cisco Unified Border Element Gateway.
- E. Verify Layer 3 connectivity on the gateway.

Answer: AC

NEW QUESTION 10

- (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.)

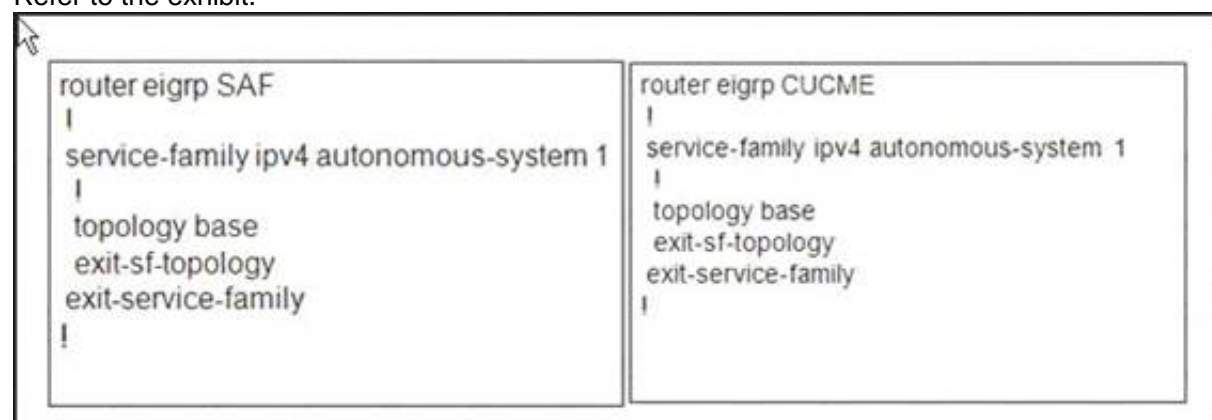
- A. debug voip ccapi inout
- B. debug ccsip message
- C. show dspfarm profile
- D. show version
- E. debug sccp messages

Answer: AB

NEW QUESTION 10

- (Exam Topic 1)

Refer to the exhibit.



Assuming that the two Cisco SAF Forwarders are adjacent to each other and that no SAF clients have been configured, which statement is true?

- A. The Cisco SAF Forwarders will not establish a neighbor relationship because the service-family external-client configuration is missing.
- B. The Cisco SAF Forwarders will not establish a neighbor relationship because the eigrp label CUCME should be replaced with SAF.
- C. The Cisco SAF Forwarders will not establish a neighbor relationship because the service-family external-client configuration is missing as well as the static neighbor configurations.
- D. The Cisco SAF Forwarders will establish a neighbor relationship.
- E. No further configuration is required.
- F. Cisco SAF Forwarders will not establish a neighbor relationship until the SAF clients are configured and registered to the Cisco SAF Forwarders.

Answer: D

NEW QUESTION 15

- (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 17

- (Exam Topic 1)

Refer to the exhibit.



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

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

Answer: C

NEW QUESTION 20

- (Exam Topic 1)

You enabled Cisco Unified Mobile Connect for a user, but the user is unable to send calls to a mobile phone from the desk phone. What do you do to resolve the issue?

- A. Restart the phone, and verify that the key is present.
- B. Under User Management > User, make sure that the Mobility option is selected.
- C. Make sure that the phone is subscribed to Extension Mobility.
- D. Add the mobility key to the softkey template that the phone is currently using.

Answer: D

NEW QUESTION 23

- (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 28

- (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 29

- (Exam Topic 1)

When dialing any external SIP URI for ABusiness-to-business call, an endpoint that is registered to the Cisco VCS Control fails to locate the remote endpoint. The same endpoint can successfully call another endpoint that is registered to the Cisco VCS Expressway. How do you resolve this issue?

- A. Add traversal call licensing on the Cisco VCS Expressway.
- B. Add traversal call licensing on the Cisco VCS Control.
- C. Add a multisite option to the endpoint.
- D. Configure a proper DNS zone on the Cisco VCS Expressway.
- E. Configure a traversal zone between the Cisco VCS Control and the Cisco VCS Expressway.
- F. Configure a SIP route pattern in Cisco Unified Communications Manager.

Answer: D

NEW QUESTION 31

- (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 36

- (Exam Topic 1)

An engineer is troubleshooting an intersite call between two endpoints where calls are intermediately failing with the error message: "488 Not Acceptable Media". Which option causes this error message to trigger?

- A. The device pool contains more call processing agents in the CMG group than the endpoint can support.
- B. MRGL contains more media groups than the endpoint can support.
- C. A lower bandwidth is set in the location than the endpoint can support.
- D. The hunt group contains more devices than the endpoint can support.

Answer: C

NEW QUESTION 40

- (Exam Topic 1)

Refer to the exhibit.

Learned Pattern					
			Select a Node cucmpub2.lab.vmware.home ▼		
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 +
- 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 44

- (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 46

- (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 50

- (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 54

- (Exam Topic 1)

To achieve 720p (HD) quality at 30 frames per second on an endpoint that is running TC software, what is the minimum configured call rate?

- A. 512 kbps
- B. 1152 kbps
- C. 768 kbps
- D. 2560 kbps

Answer: B

NEW QUESTION 56

- (Exam Topic 1)

When a user tries to initiate an Ad Hoc conference call from an IP phone, this message appears: No Conference Bridge Available. Which two actions resolve this issue? (Choose two.)

- A. Make sure that the Join softkey is assigned to the phone.
- B. Make sure that a Conference Bridge resource is registered to Cisco Unified Communications Manager.
- C. Reset the phone, to re-register resources.
- D. Make sure that a Conference Bridge Resource is assigned to the MRGL on the phone that initiates the conference call.

Answer: BD

NEW QUESTION 58

- (Exam Topic 1)

An engineer is investigating voice quality degradation on calls passing through a particular SIP gateway. To gather the necessary information, sample traffic captures are taken. Which information in the capture reveals the problem?

- A. destination port
- B. version
- C. ToS bits
- D. MTU

Answer: B

Explanation:

Reference:

<https://community.cisco.com/t5/collaboration-voice-and-video/how-to-troubleshoot-voice-qualityissues-in-a-ucm-environment/ta-p/3121613>

NEW QUESTION 63

- (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 gateway configuration
- 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 65

- (Exam Topic 1)

When identifying Cisco TelePresence Endpoint traffic characteristics, which three statements are true? (Choose three.)

- A. Latency, jitter, and loss are measured in a round-trip fashion.
- B. Latency, jitter, and loss are measured unidirectionally.
- C. Latency and loss are measured at a packet level, based on RTP header sequence numbers and time stamps.
- D. Latency and jitter are measured at a packet level, based on RTP header sequence numbers and time stamps.
- E. Jitter is measured at a video frame level, by measuring the arrival time of the video frame versus the expected arrival time.
- F. Jitter is measured at a packet level, by measuring the arrival time of the packet versus the expected arrival time.

Answer: BCE

NEW QUESTION 67

- (Exam Topic 1)

An engineer is analyzing an issue about system connection under Cisco TMS connection, where endpoints handled by TMS automatically change from reachable on LAN to behind the firewall status. Which protocol does the network engineer need to troubleshoot the network between managed device Cisco TMS?

- A. SDP
- B. XMPP
- C. FTP
- D. HTTP

Answer: D

NEW QUESTION 72

- (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 73

- (Exam Topic 1)

Refer to the exhibit.

System	
CCT Regression Test Only *	0
CDR Enabled Flag *	True
CDR Log Calls with Zero Duration Flag *	False
Digit Analysis Complexity *	StandardAnalysis
Database Debounce Timer *	0
Maximum Phone Fallback Queue Depth *	10
Maximum Number of Registered Devices *	5000
System Initialization Timer *	60

SDL Trace	
SDL Trace Data Flags *	0x00000111
SDL Trace Flush Immediately *	False
SDL Trace Data Size *	0
SDL Trace Flag *	True
SDL Trace Max File Size *	2
SDL Trace Total Number of Files *	375
SDL TraceType Flags *	0x8000EB15

An engineer is troubleshooting an outbound call and needs to see each step of the dial plan as it is being parsed in the system. The engineer is not able to see all of the steps in the trace output. How can this problem be resolved?

- A. Change the Digit Analysis Complexity to Translation And Alternate Pattern Analysis.
- B. Change the SDL Trace Max File Size to a higher number because it is not large enough to display dial plan steps.
- C. Change the SSL TraceType Flag to 0x9000EC44.
- D. Change the SSL Trace Flush Immediately to True.

Answer: A

NEW QUESTION 76

- (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 80

- (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 83

- (Exam Topic 1)

Refer to the exhibit.


```

CapiSetPayload Max=1000(rs), Payload Init(1000(rs), Payload Rm(1000(rs))
#PlayOut Max=1000(rs), Fax Num=200(rs))
23447: Jun 26 10:30:21.828: //92009/389C36000000/CCAPI/cc_api_caps_ack:
Destination Interface=0x2CFDE01C, Destination Call Id=92008, Source Call Id=92009,
Caps(Codec=0711u1a(0x1), Fax Rate=MAX_RATE_VOICE(0x2), Vad=OFF(0x1),
Modem=OFF(0x0), Codec Bytes=160, Signal Type=2, Seq Num Start=6427)
23448: Jun 26 10:30:21.828: //92009/389C36000000/CCAPI/cc_api_caps_ack:
Destination Interface=0x2CFDE01C, Destination Call Id=92008, Source Call Id=92009,
Caps(Codec=0pmefr(0x0), Fax Rate=Invalid(0x0), Vad=Invalid(0x0),
Modem=Invalid(0xFFFFFFF), Codec Bytes=0, Signal Type=0, Seq Num Start=-1)
23449: Jun 26 10:30:21.828: //92008/389C36000000/CCAPI/cc_api_event_indication:
Event=175, Call Id=92008
23450: Jun 26 10:30:21.828: //92008/389C36000000/CCAPI/cc_api_event_indication:
Event Is Sent To Conferenced SPI(s) Directly
23451: Jun 26 10:30:21.828: //92008/389C36000000/CCAPI/cc_api_event_indication:
Event=197, Call Id=92008
23452: Jun 26 10:30:21.828: //92008/389C36000000/CCAPI/cc_api_event_indication:
Event Is Sent To Conferenced SPI(s) Directly
23453: Jun 26 10:30:22.248: //92008/389C36000000/CCAPI/cc_generateToneInfo:
Stop Tone On Digit=FALSE, Tone=NULL,
Tone Direction=Sum Network, Params=0x0, Call Id=92008
23454: Jun 26 10:30:22.248: //92009/389C36000000/CCAPI/cc_api_event_indication:
Event=185, Call Id=92009
23455: Jun 26 10:30:22.248: //92009/389C36000000/CCAPI/cc_api_event_indication:
Event Is Sent To Conferenced SPI(s) Directly
23456: Jun 26 10:30:22.248: //92008/XXXXXXXXXXXX/CCAPI/cc_api_get_xcode_stream:
23457: Jun 26 10:30:22.248: cc_api_get_xcode_stream : 4803
23458: Jun 26 10:30:22.248: //92009/XXXXXXXXXXXX/CCAPI/cc_api_get_xcode_stream:
23459: Jun 26 10:30:22.248: cc_api_get_xcode_stream : 4803
23460: Jun 26 10:30:22.248: //92008/XXXXXXXXXXXX/CCAPI/cc_api_get_xcode_stream:
23461: Jun 26 10:30:22.248: cc_api_get_xcode_stream : 4803
23462: Jun 26 10:30:22.248: //92008/XXXXXXXXXXXX/CCAPI/cc_api_get_xcode_stream:
23463: Jun 26 10:30:22.248: cc_api_get_xcode_stream : 4803
23464: Jun 26 10:30:22.248: //92009/XXXXXXXXXXXX/CCAPI/cc_api_get_xcode_stream:
23465: Jun 26 10:30:22.252: cc_api_get_xcode_stream : 4803
23466: Jun 26 10:30:22.252: //92008/XXXXXXXXXXXX/CCAPI/cc_api_get_xcode_stream:
23467: Jun 26 10:30:22.252: cc_api_get_xcode_stream : 4803
23468: Jun 26 10:30:22.252: //92009/XXXXXXXXXXXX/CCAPI/cc_api_get_xcode_stream:
23469: Jun 26 10:30:22.252: cc_api_get_xcode_stream : 4803
23470: Jun 26 10:30:22.252: //92009/XXXXXXXXXXXX/CCAPI/cc_api_get_xcode_stream:
23471: Jun 26 10:30:22.252: cc_api_get_xcode_stream : 4803
23472: Jun 26 10:30:22.288: //92009/389C36000000/CCAPI/cc_api_call_facility:
Interface=0x2CFDE01C, Call Id=92009
23473: Jun 26 10:30:22.288: //92009/389C36000000/CCAPI/cc_api_event_indication:
Event=197, Call Id=92009
N-C2921-CME#
23474: Jun 26 10:30:22.288: //92009/389C36000000/CCAPI/cc_api_event_indication:
Event Is Sent To Conferenced SPI(s) Directly
N-C2921-CME#
23475: Jun 26 10:30:29.895: //92008/389C36000000/CCAPI/cc_generateToneInfo:
Stop Tone On Digit=FALSE, Tone=NULL,
Tone Direction=Sum Network, Params=0x0, Call Id=92008
23476: Jun 26 10:30:29.895: //92009/389C36000000/CCAPI/cc_api_call_disconnected:
Cause Value=16, Interface=0x2CFDE01C, Call Id=92009
23477: Jun 26 10:30:29.895: //92009/389C36000000/CCAPI/cc_api_call_disconnected:
Call Entry(Response=TRUE, Cause Value=16, Retry Count=0)
23478: Jun 26 10:30:29.895: //92008/389C36000000/CCAPI/cc_conferenceDestroy:
Conference Id=0x11A8, Tag=0x0
23479: Jun 26 10:30:29.895: //92008/389C36000000/CCAPI/cc_api_bridge_drop_done:
Conference Id=0x11A8, Source Interface=0x2CFDE01C, Source Call Id=92008,
Destination Call Id=92009, Disposition=0x0, Tag=0x0
23480: Jun 26 10:30:29.895: //92009/389C36000000/CCAPI/cc_api_bridge_drop_done:
Conference Id=0x11A8, Source Interface=0x2CFDE01C, Source Call Id=92009,
Destination Call Id=92008, Disposition=0x0, Tag=0x0
23481: Jun 26 10:30:29.895: //92008/389C36000000/CCAPI/cc_generic_bridge_done:
Conference Id=0x11A8, Source Interface=0x2CFDE01C, Source Call Id=92009,
Destination Call Id=92008, Disposition=0x0, Tag=0x0
23482: Jun 26 10:30:29.895: //92008/389C36000000/CCAPI/ccallDisconnect:
Cause Value=16, Tag=0x0, Call Entry(Previous Disconnect Cause=0, Disconnect Cause=0)
23483: Jun 26 10:30:29.895: //92008/389C36000000/CCAPI/ccallDisconnect:
Cause Value=16, Call Entry(Response=TRUE, Cause Value=16)
23484: Jun 26 10:30:29.899: //92009/389C36000000/CCAPI/ccallDisconnect:
Cause Value=16, Tag=0x0, Call Entry(Previous Disconnect Cause=0, Disconnect Cause=16)
23485: Jun 26 10:30:29.899: //92009/389C36000000/CCAPI/ccallDisconnect:
Cause Value=16, Call Entry(Response=TRUE, Cause Value=16)
23486: Jun 26 2015 10:30:29.899 CDT: SVOIPAAA-S-VOIP_CALL_HISTORY: CallLegType 2, ConnectionId 389C3600100003BAFC54A8C0, SetupTime
0:30:11.529 CDT Fri Jun 26 2015, PeerAddress 4445556666, PeerSubaddress , DisconnectCause 10 , DisconnectText normal call clearing
16), connecttime 10:30:21.799 CDT Fri Jun 26 2015, DisconnectTime 10:30:29.899 CDT Fri Jun 26 2015, CallOrigin 1, ChargedUnits 0,
InfoType 2, TransmitPackets 398, TransmitBytes 63840, ReceivePackets 384, ReceiveBytes 61440
23487: Jun 26 10:30:29.899: //-1/XXXXXXXXXXXX/CCAPI/cc_build_feature_vsa:
23488: Jun 26 10:30:29.899: :inside cc_build_feature_vsa
23489: Jun 26 10:30:29.899: //-1/XXXXXXXXXXXX/CCAPI/cc_build_feature_vsa:
23490: Jun 26 10:30:29.899: feature call basic
23491: Jun 26 10:30:29.899: //-1/XXXXXXXXXXXX/CCAPI/cc_build_feature_vsa:
23492: Jun 26 10:30:29.899: cc_build_feature_vsa attr is fn:TW,ft:06/26/2015
0:30:11.528,cgn:1112223333,cdn:4445556666,frs:0,fid:11393,fcid:389C3600100003BAFC54A8C0,legid:16769
23493: Jun 26 2015 10:30:29.899 CDT: SVOIPAAA-S-VOIP_FEAT_HISTORY: FEAT_VSA=fn:TW,ft:06/26/2015
0:30:11.528,cgn:1112223333,cdn:4445556666,frs:0,fid:11393,fcid:389C3600100003BAFC54A8C0,legid:16769,bguid:389C36000001000000003BAFC
54A8C0
23494: Jun 26 10:30:29.899: //92009/389C36000000/CCAPI/cc_api_call_disconnect_done:
Disposition=0, Interface=0x2CFDE01C, Tag=0x0, Call Id=92009,
Call Entry(Disconnect Cause=16, Voice Class Cause Code=0, Retry Count=0)
23495: Jun 26 10:30:29.899: //92009/389C36000000/CCAPI/cc_api_call_disconnect_done:
Call Disconnect Event Sent
23496: Jun 26 10:30:29.899: //-1/XXXXXXXXXXXX/CCAPI/cc_free_feature_vsa:
23497: Jun 26 10:30:29.899: :cc_free_feature_vsa freeing 280F1DC8
23498: Jun 26 10:30:29.899: //-1/XXXXXXXXXXXX/CCAPI/cc_free_feature_vsa:
23499: Jun 26 10:30:29.899: vsacount in free is 3
23500: Jun 26 10:30:29.899: //-1/XXXXXXXXXXXX/CCAPI/cvcmnPoolTDFreeHelper:
data = 34850B34
23501: Jun 26 10:30:29.899: cvcmnPoolTDFreeHelper:mem_ngr_neepool_free: mem_refcnt(28F04A0C)=0 - neepool cleanup
23502: Jun 26 2015 10:30:29.915 CDT: SVOIPAAA-S-VOIP_CALL_HISTORY: CallLegType 2, ConnectionId 389C3600100003BAFC54A8C0, SetupTime
0:30:11.525 CDT Fri Jun 26 2015, PeerAddress 1112223333, PeerSubaddress , DisconnectCause 10 , DisconnectText normal call clearing
16), connecttime 10:30:21.895 CDT Fri Jun 26 2015, DisconnectTime 10:30:29.915 CDT Fri Jun 26 2015, CallOrigin 2, ChargedUnits 0,
InfoType 2, TransmitPackets 384, TransmitBytes 61440, ReceivePackets 400, ReceiveBytes 64000
23503: Jun 26 10:30:29.915: //-1/XXXXXXXXXXXX/CCAPI/cc_build_feature_vsa:
23504: Jun 26 10:30:29.915: :inside cc_build_feature_vsa
23505: Jun 26 10:30:29.915: //-1/XXXXXXXXXXXX/CCAPI/cc_build_feature_vsa:
23506: Jun 26 10:30:29.915: feature call basic
23507: Jun 26 10:30:29.915: //-1/XXXXXXXXXXXX/CCAPI/cc_build_feature_vsa:
23508: Jun 26 10:30:29.915: cc_build_feature_vsa attr is fn:TW,ft:06/26/2015
0:30:11.524,cgn:1112223333,cdn:4445556666,frs:0,fid:11392,fcid:389C3600100003BAFC54A8C0,legid:16768
23509: Jun 26 2015 10:30:29.919 CDT: SVOIPAAA-S-VOIP_FEAT_HISTORY: FEAT_VSA=fn:TW,ft:06/26/2015
0:30:11.524,cgn:1112223333,cdn:4445556666,frs:0,fid:11392,fcid:389C3600100003BAFC54A8C0,legid:16768,bguid:389C36000001000000003BAFC
54A8C0

```

According to the output of the debug voip ccapi inout command, why was this call dropped?

- A. Normal call clearing
- B. User busy
- C. Call rejected
- D. Invalid number
- E. No circuit
- F. No resource

Answer: A

NEW QUESTION 87

- (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 90

- (Exam Topic 1)

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

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

Answer: A

NEW QUESTION 95

- (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 100

- (Exam Topic 1)

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

- A. Cisco IOS debug tools
- B. Class of Restriction baseline configuration for the user on Cisco Unified Communications Manager Express
- C. show output of the ephone anDEphone-dn configurations
- D. show output of the voice translation rules in the voice gateway
- E. show output for the T1 controller and voice port configuration in the voice gateway

Answer: AB

NEW QUESTION 105

- (Exam Topic 1)

Refer to the exhibit.

RTP Phone Device Configuration	Partitions	RTP Phone DN Configuration	Partitions
Device CSS	RTP_Emergency ALL_Phones	Line CSS	RTP_Local RTP_LongDistance RTP_International
AAR CSS	RTP_LongDistance	AAR Group	AAR

U.K. User Device Profile	Partitions	Partition	Route Pattern
Line CSS	U.K_Emergency ALL_Phones	RTP_Emergency	9.911
		RTP_Local	9.[2-9]XXXXXX
		RTP_LongDistance	9.1[2-9]XX[2-9]XXXXXX
		RTP_International	9.011#
		U.K_Emergency	0.000
AAR Group	AAR	U.K_PSTN	9.1

Assume a Centralized Cisco Unified Communications Manager topology with the headquarters at RTP and remote located at the U.K. All route patterns are assigned a route list that contains a route group pointing to the local gateway. RTP route patterns use the RTP gateway, and U.K. route patterns use the U.K. gateway.

When a U.K. user logs into an RTP phone using the Cisco Extension Mobility feature and places an emergency call to 0000, which statement about the emergency call is true?

- A. The call will match the U.K_Emergency route pattern partition and will egress at the RTP gateway.
- B. The call will match the U.K_Emergency route pattern partition and will egress at the U.K. gateway.
- C. The call will match the RTP_Emergency route pattern partition and will egress at the RTP gateway.
- D. The call will match the RTP_Emergency route pattern partition and will egress at the U.K. gateway.
- E. The call will match the RTP_Emergency route pattern partition and will egress at the U.K. gateway.

- F. gateway.
- G. The call will fail.

Answer: B

NEW QUESTION 108

- (Exam Topic 1)

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

- A. The PSTN gateway is not configured.
- B. The calleDEndpoint does not have the SIP trunk enabled.
- C. The calleDEndpoint is not registered.
- D. The calling endpoint is not in the CSS of the calleDEndpoint.
- E. The calleDEndpoint is not in the partition of the calling endpoint.
- F. The calling endpoint is not configuredFor the correct CoS.

Answer: CF

NEW QUESTION 113

- (Exam Topic 1)

Where in Cisco TMS would you see if a system is registered to a Cisco VCS or a Cisco Unified Communications Manager?

- A. Systems > Registration
- B. Navigation > Systems > Registrations
- C. under Registration on the System Administration tab
- D. System Overview
- E. Settings > Provisioning
- F. where you start the Cisco Unified Communications Manager RTMT under Systems and Reports

Answer: D

NEW QUESTION 115

- (Exam Topic 1)

Which three network conditions anDEquipment should you avoid to ensure a high-quality Cisco TelePresence experience? (Choose three.)

- A. network hubs
- B. Layer 3 switches
- C. duplex mismatch connections
- D. 10/100 access ports
- E. high utilization link with QoS
- F. network loops
- G. redundant network trunks

Answer: ACF

NEW QUESTION 117

- (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 118

- (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 120

- (Exam Topic 1)

Which two issues can cause a Cisco Unified Communications Manager to fail to register with its Cisco SAF Forwarder? (Choose two.)

- A. An H.323 SAF trunk was configured instead of a SIP SAF trunk.
- B. No directory number patterns were configured on the Cisco Unified Communications Manager.
- C. CCD advertising service was not activated.

- D. Incorrect user credentials were used on the SAF Forwarder.
- E. CCD requesting service was not activated.

Answer: DE

NEW QUESTION 125

- (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 129

- (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 132

- (Exam Topic 1)

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

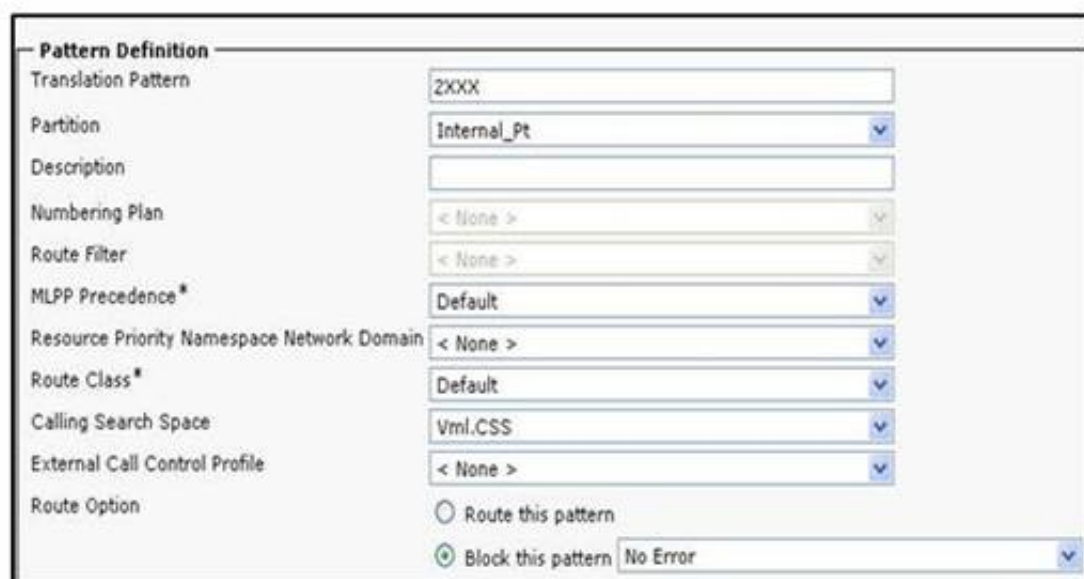
- A. Router# show ccm-manager gateway
- B. Router# show mgcp ccm-manager
- C. Router# show ccm-manager
- D. Router# show ccm manager

Answer: C

NEW QUESTION 137

- (Exam Topic 1)

Refer to the exhibit.



The image shows a 'Pattern Definition' configuration window. The fields are as follows:

Field	Value
Translation Pattern	2XXXX
Partition	Internal_Pt
Description	
Numbering Plan	< None >
Route Filter	< None >
MLPP Precedence *	Default
Resource Priority Namespace Network Domain	< None >
Route Class *	Default
Calling Search Space	Vml.CSS
External Call Control Profile	< None >
Route Option	<input type="radio"/> Route this pattern <input checked="" type="radio"/> Block this pattern
	No Error

All phones are placed in the Internal_Pt partition. The CSS for all phones contains the partition Internal_Pt, and Vml.CSS contains the voicemail hunt pilot. When a call is placed from extension 2001 to 2002, which statement is true?

- A. Extension 2002 will ring.
- B. The call will be blocked.
- C. The call will be answered by voicemail.
- D. Extension 2002 will ring, and if the call is not answered, the call will match the translation pattern and then be blocked.
- E. Extension 2002 will ring, and if the call is not answered, the call will match the translation pattern and then be forwarded to voicemail.

Answer: A

NEW QUESTION 140

- (Exam Topic 1)

Which CLI command is used to troubleshoot ILS network connection issues within the local Cisco Unified CM cluster to determine which server within the cluster is the xnode?

- A. utils ils lookup
- B. utils ils showpeerinfo
- C. utils ils display xnode
- D. utils ils find xnode

Answer: D

NEW QUESTION 142

- (Exam Topic 1)

Refer to the exhibit.



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

- A. The endpoint successfully registers to Cisco Unified Communications Manager as a SIP endpoint.
- B. The endpoint does not register to Cisco VCS as a SIP endpoint, because the domain information is missing.
- C. The endpoint does not register to Cisco Unified Communications Manager as a SIP endpoint, because the domain information is missing.
- D. The endpoint successfully registers to Cisco VCS as a SIP endpoint.
- E. The endpoint successfully registers to Cisco Unified Communications Manager as an H.323 endpoint.
- F. The endpoint successfully registers to Cisco VCS as an H.323 endpoint.

Answer: F

NEW QUESTION 147

- (Exam Topic 1)

You are troubleshooting video quality issues on a Cisco TelePresence TX9000 Series system. Which CLI command shows the total number of lost video packets and the received jitter during a Call in progress?

- A. show call statistics video
- B. show call statistics all
- C. show call statistics detail
- D. show call statistics video detail

Answer: D

NEW QUESTION 150

- (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 151

- (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 155

- (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 160

- (Exam Topic 1)

Refer to topology and Exhibits below:

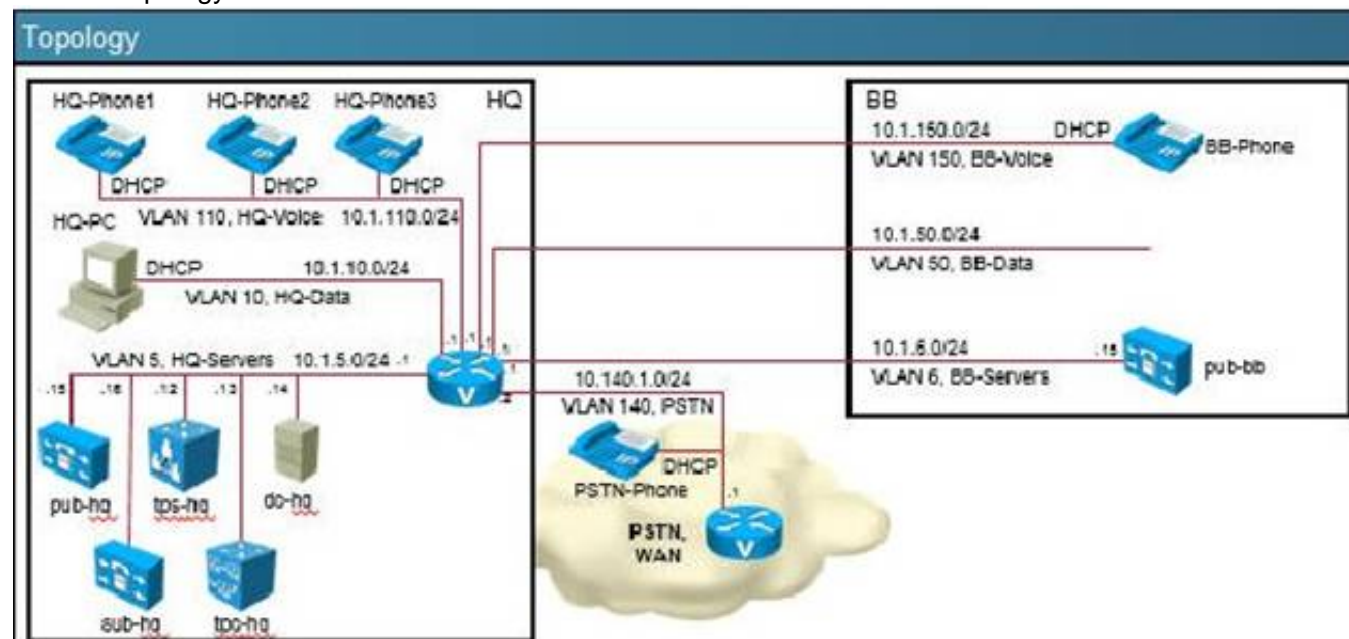


Exhibit2

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 T1PRI

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:
```

What is the reason that this MGCP gateway is not registered with Cisco Unified Communications Manager?

- A. The primary server address is incorrect.
- B. The MGCP domain name is incorrect on either the Cisco Unified Communications Manager or the router.
- C. Backhaul/Redundant link port is incorrect
- D. This MGCP gateway is not down; it is operational.

Answer: B

NEW QUESTION 165

- (Exam Topic 1)

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

- A. network connection failure between the SAF Forwarder and Cisco Unified Communications Manager
- B. network connection failure between the primary and backup SAF Forwarders
- C. TCP connection failure with the primary SAF Forwarder
- D. TCP connection failure with the backup SAF Forwarder

Answer: A

NEW QUESTION 169

- (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 172

- (Exam Topic 1)

System A at Company 1 is calling System B at Company 2. The call completes, but only audio and video are present on System A from System B. What are three possible causes? (Choose three.)

- A. System A cannot call System B because it is at a different company.
- B. There is a firewall in the path that is blocking audio and video traffic from Company 1 to Company 2.
- C. The firewall at Company 1 is blocking outgoing traffic.
- D. An access list is blocking video and audio somewhere in the video and audio path between System A and System B.
- E. System A has turned off the camera and the microphone.

Answer: BDE

NEW QUESTION 176

- (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

- (Exam Topic 1)

- A. `utils ils findxnode`
- B. `utils ils show peer info`
- C. `utils ils showpeerinfo`
- D. `utils ils lookup`

NEW QUESTION 185

- (Exam Topic 1)

Refer to Exhibit:

After reviewing the trace in the exhibit, what was the Directory number of the calling party?

A. 2001
B. 5010
C. 1905
D. 2003

NEW QUESTION 186

- (Exam Topic 1)

Refer to topology and Exhibits below:

Exhibit2

Pattern Definition

Pattern Usage

Domain Routing

IPv4 Pattern*

v360.cisco.com

IPv6 Pattern

Description

Route Partition

< None >

SIP Trunk/Route List*

Trunk-VCS

☐ 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

twcs: 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-bbb8c2f68b43" 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

twcs: 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"

twcs: 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="280dd300-3d67-4482-adb4-2c2587061de1" Tag="2a8d5d57-8f8f-4e37-a817-7d4160affa3b" Protocol="TLS" Level="1" UTCtime="2015-02-12 21:37:28.614"

twcs: 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"

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

- A. The call succeeds
- B. The call fails
- C. There are no issues, so the call succeeds.
- D. The SIP port is incorrect on the Cisco Unified Communications Manager SIP trunk.
- E. The Local Zone Match Rule state is disabled.
- F. Rule name UCM2 is set to stop on Match.

Answer: BF

NEW QUESTION 191

- (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 192

- (Exam Topic 1)

Which database replication value indicates that the server no longer has an active logical connection to receive database tables?

- A. 1
- B. 2
- C. 3
- D. 4

Answer: E

NEW QUESTION 197

- (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 199

- (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 201

- (Exam Topic 1)

Refer to the exhibits.

Learned Pattern						
				Select a Node	CUCM801Pub1 ▼	
Pattern	TimeStamp	Status	Protocol	AgentId	IP Address	To/DID
300X	2010/04/03 13:55:55	Reachable	SIP	CID10.1.5.11	10.1.5.11(5060)	0+44228822
300X	2010/04/03 13:55:55	Reachable	H323	CID10.1.5.11	10.1.5.11(54532)	0+44228822

Pattern Definition	
Route Pattern*	3XXX
Route Partition	Internal_Pt ▼
Description	
Numbering Plan	-- Not Selected -- ▼
Route Filter	< None > ▼
MLPP Precedence*	Default ▼
Resource Priority Namespace Network Domain	< None > ▼
Route Class*	Default ▼
Gateway/Route List*	SIP_Trunk ▼ (Edit)
Route Option	<input type="radio"/> Route this pattern <input checked="" type="radio"/> Block this pattern No Error ▼
Call Classification*	OffNet ▼
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority	
<input type="checkbox"/> Require Forced Authorization Code	
Authorization Level*	0

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

- A. The call will succeed and will be placed via the SAF network
- B. SAF-learned routes always take precedence.
- C. The call will fail because it will be blocked by the route pattern.
- D. The call will be placed in a round-robin fashion between the SAF network and SIP_Trunk.
- E. The call will be placed in a round-robin fashion between the SAF network and SIP_Trunk
- F. Every other call will fail.

Answer: B

NEW QUESTION 202

- (Exam Topic 1)

Refer to the exhibit.

```
interface GigabitEthernet1/0/4
description HQ Phone 1
switchport access vlan 10
switchport voice vlan 110
spanning-tree portfast
```

An IP phone that is connected through a Cisco Catalyst 3750 Series Switch is failing to register with Cisco Unified Communications Manager. When the user presses the settings button on the phone, the Operational VLAN ID shows ABlank entry. What is the most likely cause for this issue?

- A. The switch may not be supplying inline power.
- B. The spanning tree portfast command needs to be removed.
- C. The trunk encapsulation is missin
- D. The trunk must be configuredFor dot1.Q.
- E. Cisco Discovery Protocol is disabled on the switch.
- F. The Operational VLAN ID of the phone always shows as blan
- G. The Admi
- H. VLAN ID should be 110.

Answer: D

NEW QUESTION 205

- (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 209

- (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 213

- (Exam Topic 1)

After an IP Phone gets IP address information from DHCP, what is the next step in the initialization process?

- A. CTL and ITL files are downloaded.
- B. The phone requests its VLAN information.
- C. The DHCP offer is sent from the phone
- D. The TFTP server is contacted for configuration information.
- E. Nothing else is required, the phone is operational at this stage

Answer: D

NEW QUESTION 218

- (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 221

- (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 CUCM1
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: CUCM1
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 224

- (Exam Topic 1)

Cisco TelePresence System EX90-A and EX90-B are in a Call. EX90-A tries to call EX90-C. When the call is dialed, EX90-B is put on hold. EX90-A and EX90-C are connected, but there is no merge button on the touch panel. What is causing this issue?

- A. The multisite option key is missing.
- B. The multisite configuration is missing.
- C. The conference option key is missing.
- D. The conference configuration is missing.
- E. Cisco TelePresence systems cannot make multipoint calls without a Cisco TelePresence Server.
- F. The multipoint option key is missing.

Answer: A

NEW QUESTION 226

- (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 228

- (Exam Topic 2)

A remote office uses a VPN with the HQ office to register its phones for centralized management. The office has a local SIP gateway with PRI for calling and to support SRST. On the Cisco Unified Communications Manager, a device pool is created for this site with media resources that are local to Cisco Unified Communications Manager. A SIP trunk that requires an MTP connects to the gateway. Users report voice quality issues when making and receiving calls. How do you resolve this issue?

- A. Restart the phones at the remote office
- B. Restart the voice gateway and switch
- C. Enable QoS on the remote office
- D. Assign an MTP that is local to the remote office to the device pool that is used by the phones.

Answer: C

NEW QUESTION 232

- (Exam Topic 2)

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

- A. The CSS of the Chicago phones does not include the partition of the route pattern to point to Tokyo.
- B. The SIP trunk in Chicago is in ACSS that is not in the partition of the Chicago phones.
- C. The Tokyo SIP trunk inbound CSS does not contain the partition of the Tokyo phones.
- D. Tokyo phones are in ACSS that is not in the partition of the route pattern to call from Tokyo to Chicago.
- E. The SIP trunk is configured from Tokyo to Chicago, but not from Chicago to Tokyo.

Answer: AC

NEW QUESTION 236

- (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 238

- (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 239

- (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 242

- (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 247

- (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 251

- (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 255

- (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 258

- (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 259

- (Exam Topic 2)

After you deploy a hardware transcoder, users without a Common codec are unable to complete a call to one another. Which two actions must you take (Choose two.)

- A. Use RTMT to verify that transcoder resources are available
- B. verify that all SIP trunks are registered within Cisco Unified Communications Manager
- C. verify that media resources are assigned to the transcoder
- D. verify that the transcoder registered with Cisco Unified Communications Manager.
- E. verify that the transcoder is running the latest firmware and Flash NVRAM

Answer: AD

NEW QUESTION 261

- (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 263

- (Exam Topic 2)

Two Cisco DX80 collaboration endpoints that support high-definition video are registered to a Cisco Unified Communications Manager in a single-site deployment model. However, the video that is displayed is pixelated and appears as sub-HD. To enable the Cisco DX80s to make high-definition video calls, which configuration option do you check?

- A. Check the configuration in the Cisco DX80 settings menu, and enable high-definition video.
- B. Check the Maximum Session Bit Rate for Video Calls in the Cisco Unified Communications Manager Regions Configuration.
- C. Check that Video Capabilities are enabled under the Product Specific Configuration area for the device in Cisco Unified Communications Manager.
- D. Check the DSCP for Video Calls value in the Cisco Unified Communications Manager clusterwide parameters QoS configuration.

Answer: B

NEW QUESTION 266

- (Exam Topic 2)

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

- A. Stop the CTI Manager Service on the Secondary server so that all devices using CTI register with the primary.
- B. Stop the CTI Manager Service on the Primary server so that all devices using CTI stay registered with the primary CTI server.
- C. Restart the subscriber with the CTI Manager Service on the secondary.
- D. Do nothing and wait for the next failover assuming this will revert back to the primary CTI server.

Answer: A

NEW QUESTION 267

- (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 270

- (Exam Topic 2)

A Cisco TelePresence SX80 Dial is registered on Cisco Unified Communications Manager is calling a registered H.323 endpoint that is registered to a Cisco VCS Control within the same campus. The call is placed using 6 Mb, however only a 1 MB call is set up. Where can you find two possible causes for this issue? (Choose two)

- A. The Cisco TelePresence SX80, which needs an option key to call over 2 M
- B. which also enables multisite calls
- C. in the Region settings in Cisco Unified Communications Manager
- D. bandwidth settings in the SIP trunk profile
- E. legacy telepresence systems that are registered to the Cisco VCS Control and can only do 1 Mb towards a Cisco Unified Communications Manager system
- F. a link and pipe configuration on the VCS to Cisco Unified Communications Manager subzone
- G. in the subzone settings of the registered H.323 endpoint

Answer: C

NEW QUESTION 274

- (Exam Topic 2)

Which set ensures that all SAF clients receive all service advertisements?

A)

```
(config)#router eigrp saf
(config)#service-family ipv4 autonomous --system 208
(config-router-sf)#
(config-router-sf)#sf-interface gigabitEthernet0/0
(config-router-sf)#hello-interval 10
(config-router-sf)#split-horizon
```

B)

```
(config)#router eigrp saf
(config)#service-family ipv4 autonomous --system 208
(config-router-sf)#
(config-router-sf)#sf-interface gigabitEthernet0/0
(config-router-sf)#hello-interval 10
```

C)

```
(config)#router eigrp saf
(config)#service-family ipv4 autonomous --system 208
(config-router-sf)#
(config-router-sf)#sf-interface gigabitEthernet0/0
(config-router-sf)#hello-interval 10
(config-router-sf)#no split-horizon
```

D)

```
(config)#router eigrp saf
(config)#service-family ipv4 autonomous --system 208
(config-router-sf)#
(config-router-sf)#sf-interface gigabitEthernet0/0
(config-router-sf)#hello-interval 50
(config-router-sf)#split-horizon
```

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

Answer: C

NEW QUESTION 279

- (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", fqc="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=1011
|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 282

- (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 appliedFor 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 285

- (Exam Topic 2)

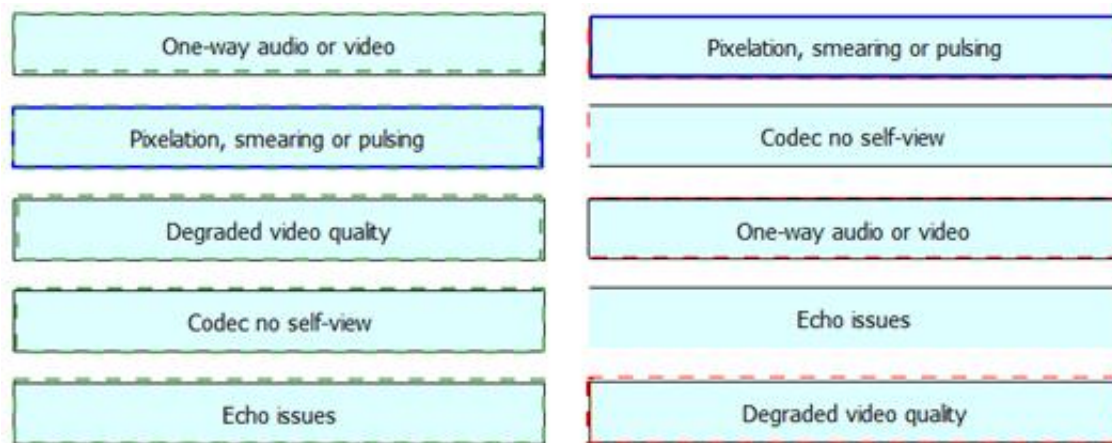
Drag the network-related video issue on the left to its root cause on the right.

One-way audio or video	Different system manufacturers
Pixelation, smearing or pulsing	Firewall with packet inspection enabled
Degraded video quality	Very high noise level
Codec no self-view	Packet loss
Echo issues	Main source is not main camera

- A. Mastered
- B. Not Mastered

Answer: A

Explanation:



NEW QUESTION 289

- (Exam Topic 2)

Users are reporting intermittent poor audio quality on VoIP calls. Which configuration area requires troubleshooting?

- A. network QoS
- B. media resource
- C. call routing
- D. phone setup

Answer: A

NEW QUESTION 293

- (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 295

- (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 300

- (Exam Topic 2)

Refer to the exhibit.

```
#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?

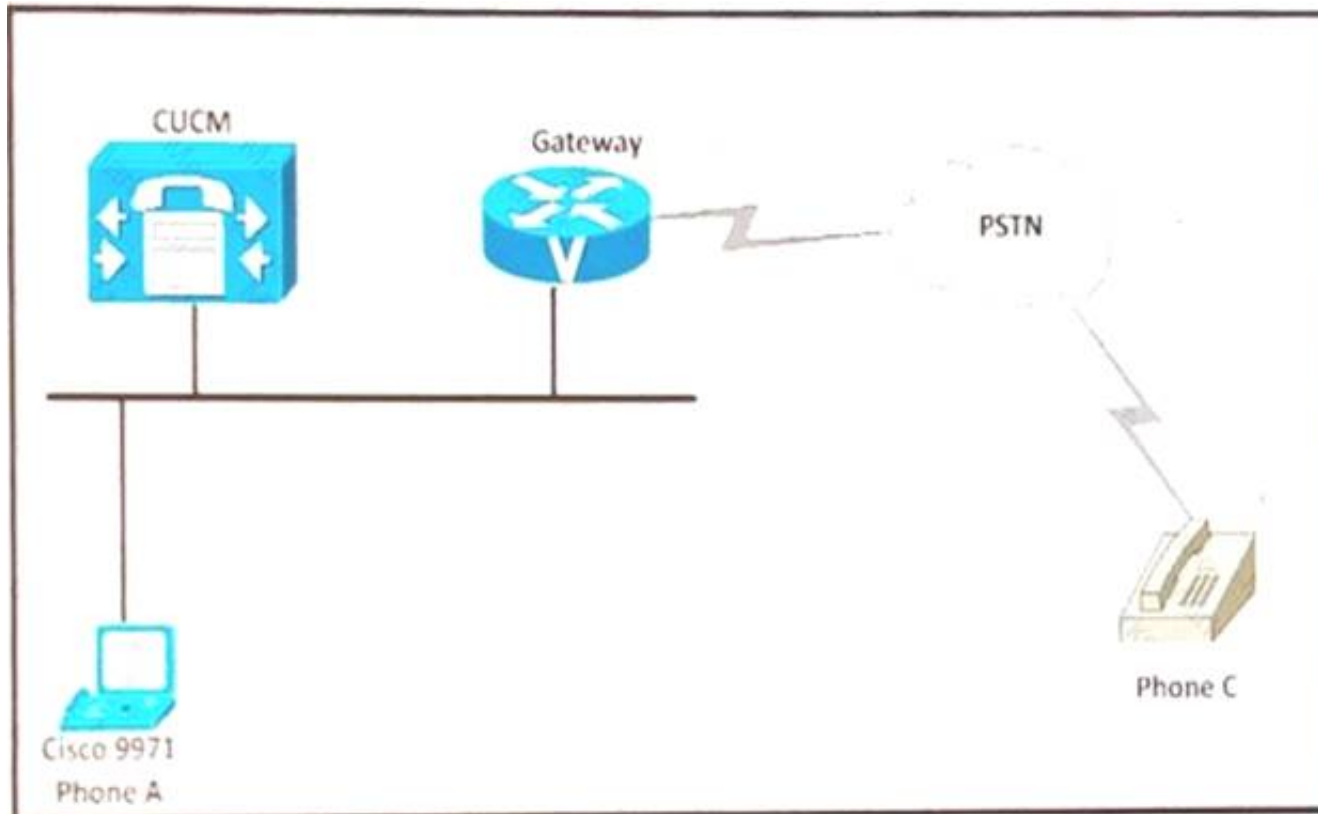
- A. interface loopback 0
- B. router eigrp saf
- C. service-family external client
- D. service-family ipv4

Answer: C

NEW QUESTION 304

- (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 308

- (Exam Topic 2)

After you deploy cisco Unified Commutations Manager Device Mobility in your environment, you note that phones in a remote site are failing to register. Which two actions correct the problem? (Choose two.)

- A. Verify that the remote site is assigned to the device mobility group that matches its dialing pattern.
- B. Verify that the phones are attempting to register with the correct megACluster in the device pool of the central office.
- C. Verify that extension mobility is configured correctly for the remote site.

- D. verify that the LDAP server is configured to support device Mobility at the remote site.
- E. verify that the subnet of the remote site is configured in device mobility info.

Answer: CE

NEW QUESTION 310

- (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?

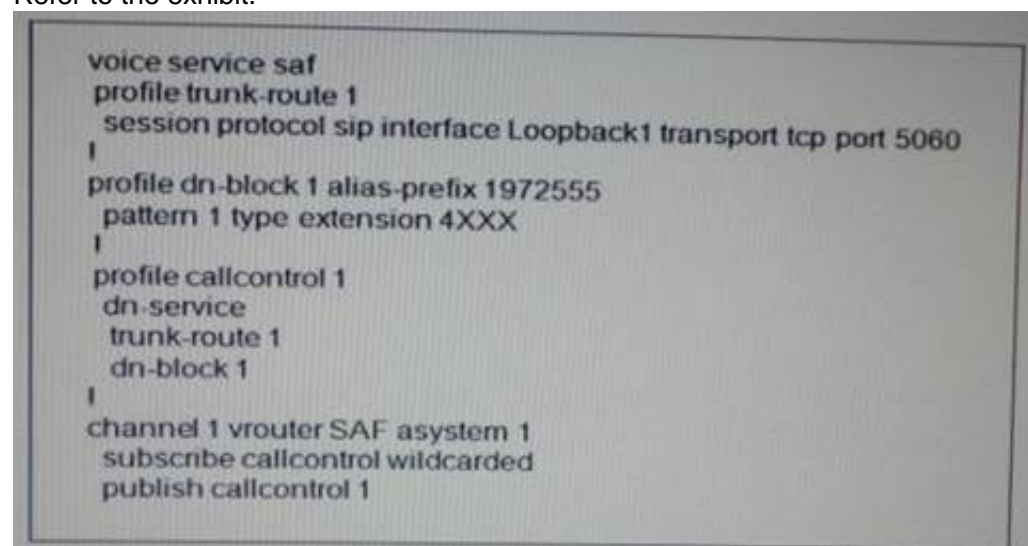
- A. Adjust the URI lookup policy to case desensitive.
- B. Adjust the URI lookup policy to case sensitive.
- C. Adjust the URI lookup policy to case insensitive.
- D. Adjust the URI lookup policy to case nonsensitive.

Answer: C

NEW QUESTION 314

- (Exam Topic 2)

Refer to the exhibit.



When a Cisco Unified Communications Manager Express advertises the directory number pattern in the exhibit, what would the learned pattern be in the RTMT tool on the Cisco Unified Communications Manager?

- A. 4XXX and the ToDID will be 0:+1972555
- B. 4XXX and the ToDID will be 0:+19725554XXX
- C. 4XXX and the ToDID will be 0:19725554XXX
- D. 4XXX and the ToDID will be 0:1972555
- E. 19725554XXX and the ToDID will be 0:+1972555

Answer: D

Explanation:

The answer is 4XXX and the ToDID will be 0:1972555.

Exhibit explain profile dn-block1 alias-prefix 1972555 and pattern 1 type extension 4xxx.

NEW QUESTION 317

- (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. Verity that the phone's network can access the option 150 server.
- C. Disable HSRP on the access layer switch.
- D. Verity 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 320

- (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 321

- (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 325

- (Exam Topic 2)

You are configuring Troubleshooting Perfmon Data-Logging parameters.

You want to specify the Cisco recommended File size that can be stored in a perfmon log file. What is the maximum file size value that you must specify?

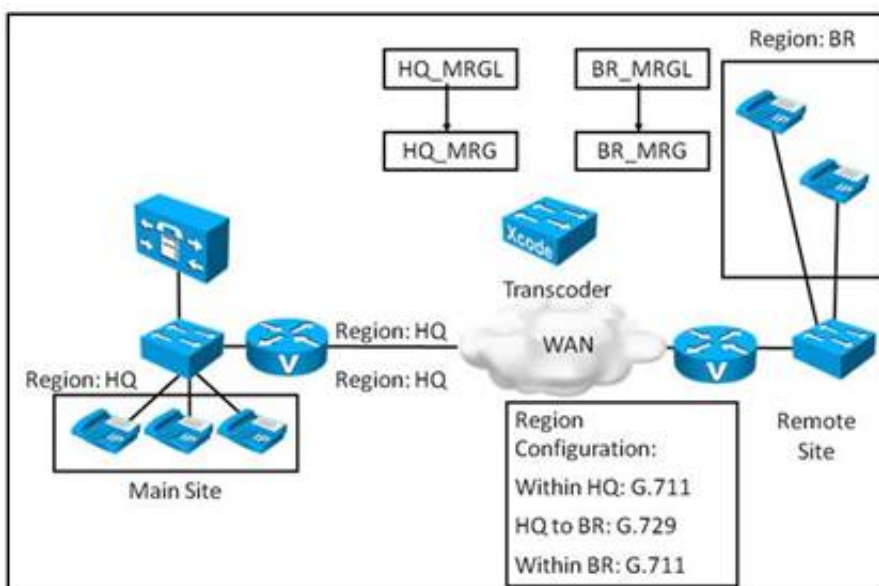
- A. 5 MB
- B. 50 MB
- C. 100 MB
- D. 150 MB
- E. 500 MB
- F. 600 MB

Answer: C

NEW QUESTION 330

- (Exam Topic 2)

Refer to the exhibit.



HQ_MRGL is assigned to the HQ IP phones BR_MRGL is assigned to the BR IP phones. The remote site BR IP phones only support G.711 codec. When a Call is placed From an HQ phone to ABR phone, the call fails. Which statement indicates how this issue is resolved?

- A. Configure the transcoder at the HQ site and assign it to HQ_MRG
- B. A transcoder is not needed
- C. The HQ phones will automatically change over to G.711 codec.
- D. The transcoder should be assigned to its own MRG, which should then be assigned to the default device pool at HQ.
- E. Configure the transcoder at the BR site and assign it to BR_MRG

Answer: D

NEW QUESTION 335

- (Exam Topic 2)

Refer to the exhibit.

```
Received
SIP/2.0 503 Service Unavailable
Via: SIP/2.0/UDP 192.168.1.1:5060;branch=z9hG4bK19A240C
From: "Joe Doe" <sip:1002@192.168.1.4>;tag=C865BC-1D1
To: <sip:4001@192.168.1.3>;tag=129582830
Date: Wed, 28 Oct 2015 15:34:25 GMT
Call-ID: E5195C1E-7CBF11E5-8376C4E8-E4AE0459@192.168.1.1
Cseq: 101 INVITE
Allow-Events: presence
Warning: 399 INT-CUCM"Unable to find a device handler for the request received on port 54394 from
192.168.1.1"
Content-Length: 0
```

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

- A. Cisco CallManager service is shut down on the CAL cluster.
- B. The SIP service is shut down on the CAL cluster.
- C. The SIP trunk on the CAL cluster is configured incorrectly.
- D. Cisco CTI Manager service is shut down on the CAL cluster.
- E. The remote Cisco IP phone is not registered.

Answer: E

NEW QUESTION 340

- (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 345

- (Exam Topic 2)

Which two tools can be used to collect log files for advanced troubleshooting of call setup failures? (Choose two.)

- A. Dialed Number Analyzer on Communications Manager servers
- B. CLI of Communications Manager servers
- C. Cisco Unified Realtime Monitoring Tool for Communications Manager servers
- D. Cisco Call Analyzer for Communications Manager Publisher servers
- E. Cisco Prime Collaboration Provisioning server

Answer: AC

NEW QUESTION 350

- (Exam Topic 2)

a Company has a headquarters and a remote site. Cisco Unified Communications Manager (CUCM) acts as a DHCP server. Both sites use their local voice gateways for PSTN calls. At the headquarters site, the PSTN prefix is 9, and the emergency number is 911. At the remote site, the PSTN prefix is 0, and the emergency number is 112. Here are the deployment policies for roaming devices:

- Softphones can roam between two sites.
- A roaming softphone uses the local gateway for all PSTN calls.
- The user keeps home dial habits on a roaming softphone, with the exception of the emergency number. When a headquarters user uses a softphone at the remote site, prefix 9 can be used to make PSTN calls via voice gateway in the remote site, but emergency number 112 does not work. Which setting on CUCM should be checked?

- A. Device Mobility Group
- B. Physical Location
- C. DHCP subnet
- D. Device Mobility Info

Answer: D

NEW QUESTION 355

- (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 359

- (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 364

- (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 368

- (Exam Topic 2)

Which three items are displayed on the endpoint registration verification page in Cisco VCS? (Choose three.)

- A. E.164 address
- B. endpoint name
- C. endpoint MAC address
- D. endpoint registration status
- E. device description
- F. device type
- G. device pool

Answer: ABF

NEW QUESTION 371

- (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 375

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

Answer: CD

NEW QUESTION 376

- (Exam Topic 2)

An engineer notices that some SCCP phones are not displaying the correct time, but the phones are registered and working property. Which three options should be performed in Cisco Unified Communications Manager to fix the phone time issue? (Choose three.)

- A. Verify that the device pool has the coned Date/Time Group configured.
- B. Ensure that the phone on time field on the phone configuration page is chosen.
- C. Check the CUCM OS Admin page to ensure that NTP servers are accessible.
- D. Verify that the Date/Time Group has the correct phone NTP reference configured.
- E. Check the phone NTP Reference configuration for configured server.
- F. Verify that the Date/Time group has the correct time zone configured.

Answer: ADF

NEW QUESTION 380

- (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 382

- (Exam Topic 2)

Refer to the exhibit.

```
RTR1# eigrp service-family ipv4 nei det
EIGRP-SFv4 VR(SAF) Service-Family Neighbors for AS(100)
H   Address   Interface   Hold Uptime   SRTT   RTO     Q       Seq
                        (sec)      (ms)
1  172.17.0.2 Lo0           12  00:00:07    24    300     0       20
Static neighbor
Version 6.0/3.0, Retrans: 1, Retries: 0
Topology-ids from peer - 0
0 1.1.1.2     Se0/1/0.103  14  00:00:10   1593   5000     0       21
Version 6.0/3.0, Retrans: 0, Retries: 0
Topology-ids from peer - 0
```

An engineer is troubleshooting a SAF forwarder adjacency in which two interfaces from the same router are being advertised to all other neighbors. Which set of IOS commands successfully stops the router from advertising the Loopback0 interface?

- A. route eigrp SAFservice-family ipv4 autonomous-system 100 sf-interface Loopback0no dampening-change
- B. router eigrp SAFservice-family ipv4 autonomous-system 100 sf-interface Loopback0exit-sf-interface
- C. router eigrp SAFservice-family ipv4 autonomous-system 100 sf-interface Loopback0no split-horizon
- D. router eigrp SAFservice-family ipv4 autonomous-system 100 sf-interface Loopback0shutdown

Answer: D

NEW QUESTION 385

- (Exam Topic 2)

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

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

- A. Revert the changes for hostname or IP address on the Publisher server.
- B. Verify and ensure that the replication is working on all nodes in the cluster.
- C. Restart the DBL service on the publisher.
- D. Modify the hostname for all Subscriber nodes in the cluster.
- E. Reboot all nodes in the cluster.
- F. Restart the DNS service in the cluster.

Answer: AB

NEW QUESTION 388

- (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 389

- (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 390

- (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
ccm-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 395

- (Exam Topic 2)

An engineer configured Cisco Extension Mobility for a user. The user can log in to the profile but notices that the extension is not correct. Which component should the engineer check first to verify the user extension?

- A. UC Service
- B. Device Profile
- C. User Profile
- D. Phone Service
- E. SIP Profile

Answer: A

NEW QUESTION 400

- (Exam Topic 2)

When you are dialing from an internally registered device on Cisco VCS, in which order are the options processed to 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 403

- (Exam Topic 2)

Cisco TelePresence system ex90#1 is trying to call EX90 #2. Both endpoints are registered to Cisco Unified Communications Manager. The call connects, but the quality is not good. The video image is breaking up on EX90 #2. The media path from A to B supports only 2 Mb. Which three reasons are possible causes? (choose three.)

- A. The camera on system B has a loose cable.
- B. The line is congested, and QoS is not configured correctly between the sites.
- C. Bandwidth Limitation between IP zones is set too low for video.
- D. Bandwidth Limitation between regions is set too low for video.
- E. The autonegotiation protocol H 296 is tuned off on one of the EX90s.
- F. The system is set to use a higher bandwidth than the media path allows.

Answer: BDE

NEW QUESTION 404

- (Exam Topic 2)

Which alarm string purges all learned patterns from the Cisco Unified Communications Manager?

- A. DuplicateLearnedPattern
- B. SAFForwarderError
- C. CCDIPReachableTimeOut
- D. CCDPSTNFailOverDurationTimeOut
- E. CCDLearnedPatternLimitReached

Answer: E

NEW QUESTION 406

- (Exam Topic 2)

Which call control model does MGCP use?

- A. Distributed
- B. Centralized
- C. Ad hoc
- D. Hybrid

Answer: B

NEW QUESTION 409

- (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 414

- (Exam Topic 2)

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

- A. The Cisco Unified Communications Manager FQDN was not set.
- B. The cluster FQDN was not set in Enterprise Parameters.
- C. The certificate is not valid.
- D. The cluster FQDN was not set in Service Parameters.
- E. The FQDN was not registered in DNS.

Answer: B

NEW QUESTION 418

- (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 422

- (Exam Topic 2)

Transcoder Information

Transcoder: MTP-Remote-BR1 (Branch 1 Transcoder)
Registration: Unknown
IPv4 Address: None

IOS Transcoder Info

Transcoder Type*

Cisco IOS Enhanced Media Termination Point

Description

Branch 1 Transcoder

Device Name*

MTP-Remote-BR1

Device Pool*

Default

View Details

Common Device Configuration

< None >

View Details

Special Load Information

Leave blank to use default

☐ Trusted Relay Point

```

sccp local FastEthernet0/0
sccp ccm 192.168.10.100 identifier 1 priority 1 version 7.0+
sccp

sccp ccm group 1
associate ccm 1 priority 1
associate profile 10 register MTP-Remote-BR_1
keepalive retries 5
switchover method immediate
switchback method immediate
switchback interval 15

dspfarm profile 10 transcode
codec g711ulaw
codec g711alaw
codec g729ar8
codec g729r8
maximum sessions 10
associate application SCCP

```

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 424

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