

Cisco

Exam 300-080

Troubleshooting Cisco IP Telephony & Video v1.0

Version: 9.0

[Total Questions: 157]

Question No : 1

You are integrating a new video endpoint with Cisco VCS, but you find that the unit is failing to register. You assign extension 4000 to the device in the "vc.cisco.com" SIP domain, and you set its registration proxy to the IP address of 10.1.1.10 as the Cisco VCS. In order for the device to register via SIP, which format must you use when you set the SIP address of the device?

- A. 4000@vc.cisco.com
- B. 4000
- C. 4000@10.1.110
- D. 4000@cisco

Answer: A

Question No : 2

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

Question No : 3

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

Refer to the exhibit.

```

SW3-POE#sh int gi 0/12
GigabitEthernet0/12 is up, line protocol is up (connected)
  Hardware is Gigabit Ethernet, address is 000c.854d.a78c (bia 000c.854d.a78c)
  MTU 1500 bytes, BW 100000 Kbit, DLY 1000 usec,
    reliability 255/255, txload 34/255, rxload 23/255
  Encapsulation ARPA, loopback not set
  Keepalive set (10 sec)
  Half-duplex, 1000Mb/s, media type is 10/100/1000BaseTX
  input flow-control is off; output flow-control is unsupported
  ARP type: ARPA, ARP Timeout 04:00:00
  Last input never, output never, output hang never
  Last clearing of "show interface" counters never
  Input queue: 0/75/19872/0 (size/max/drops/flushes); Total output drops: 17526
  Queueing strategy: fifo
  Output queue: 0/40 (size/max)
  5 minute input rate 121221168 bits/sec, 1201221 packets/sec
  5 minute output rate 9879879 bits/sec, 76876 packets/sec
  
```

- A. There is a mismatch in the port configuration.
- B. The input flow control is off.
- C. The QoS that is configured on the port is set to FIFO
- D. There are packet drops in the ingress queues

Answer: D

Question No : 4

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 and ephone-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: A,B

Question No : 5

Refer to the exhibit.

Conference 1
Refresh
Collapse all
Expand all

ActiveControl Mode	Auto	
CallProtocolIPStack	IPv4	
Encryption Mode	BestEffort	
IncomingMultisiteCall Mode	Allow	
MaxReceiveCallRate	6000	(64 to 6000)
MaxTotalReceiveCallRate	10000	(64 to 10000)
MaxTotalTransmitCallRate	10000	(64 to 10000)
MaxTransmitCallRate	6000	(64 to 6000)
MicUnmuteOnDisconnect Mode	On	
Multipoint Mode	MultiSite	

AutoAnswer

Delay	1	(0 to 50)
Mode	On	
Mute	On	

DefaultCall

Protocol	Sip	
Rate	6000	(64 to 6000)

You are trying to establish a multipoint call via a Cisco TelePresence SX20 that is registered to Cisco Unified Communications Manager. When you attempt to bring a third party into the call, you receive an error that the call cannot be completed. You confirm that your MCU resources are configured correctly on Cisco Unified Communications Manager and that other devices are able to establish multipoint calls. What is the cause of this issue?

- A. The multipoint mode should be set to Multiway.
- B. The call protocol should be H.323.
- C. The MCU is at capacity.
- D. The multipoint mode should be set to Cisco Unified Communications Manager Resource Group.
- E. The multipoint mode should be set to None.

Answer: D

Question No : 6

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: A,B,G

Question No : 7

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

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

Answer: A,D

Question No : 8

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

Question No : 9

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: B,C,E

Question No : 10

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

Question No : 11

Where do you configure the phone book provisioning synchronization for an endpoint that is reentered to Cisco VCS?

- A.** in Cisco Unified Communications Manager, under the SIP trunk to Cisco VCS
- B.** on the Cisco Express C
- C.** in the Cisco TelePresence Management suite in the Cisco VCS configuration
- D.** on the Cisco VCS Expressway
- E.** on the Cisco VCS Control
- F.** in the Cisco TelePresence Management Suite under Administration > Provisioning

Answer: F

Question No : 12

Refer to the exhibit.

Pattern Definition	
Translation Pattern	2XXX
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

Question No : 13

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

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

Answer: D

Question No : 14

Refer to the exhibit.


```

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

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

- A.** Change the significant digits for inbound calls to 4 on the SIP trunk configuration in Cisco Unified Communications Manager.
- B.** Configure the digit strip 4 on the SIP trunk under Incoming Called Party Settings in Cisco Unified Communications Manager.
- C.** Configure a translation pattern in Cisco Unified Communications Manager that can be accessed by the trunk CSS to truncate the called number to four digits.
- D.** Configure a called-party transformation CSS on the gateway in Cisco Unified Communications Manager that includes a pattern that transforms the number from ten digits to four digits.
- E.** Configure a voice translation profile in the SIP Cisco IOS gateway with a voice translation rule that truncates the number from ten digits to four digits.
- F.** Configure the Cisco IOS command num-exp 2288223001 3001 on the gateway ISDN interface.

Answer: A,C,E

Question No : 15

Which two types of Cisco Unified Communications Manager trace files contain Call Processing information that is helpful for troubleshooting outbound and inbound calling issues? (Choose two.)

- A. Cisco Unified Communications Manager syslog trace
- B. Cisco Unified Communications Manager Dialed Number Analyzer trace
- C. Real Time Monitoring Tool Processes trace
- D. Cisco Unified Communications Manager SDL trace
- E. Cisco Unified Communications Manager Log4Jtrace
- F. Cisco Unified Communications Manager SDI trace

Answer: D,F

Question No : 16

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

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

Answer: A,B

Question No : 17

Which of these reasons can cause intrasite calls within a Cisco Unified Communications Manager cluster to fail?

- A. The route partition that is configured in the CCD requesting service is not listed in the calling phone CSS.
- B. The trunk CSS does not include the partition for the called directory number.
- C. The MGCP gateway is not registered.
- D. The calling phone does not have the correct CSS configured.
- E. The calling phone does not have the correct partition configured.

Answer: D