

Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

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NEW QUESTION 1

What is the element of Cisco Collaboration infrastructure that allows Jabber clients outside of the network to register in Cisco Unified Communications Manager and use its resources?

- A. Cisco IM and Presence node
- B. Cisco Unified Border Element
- C. Cisco Expressway
- D. Cisco Prime Collaboration Provisioning server

Answer: C

NEW QUESTION 2

An engineer with troubleshoots poor voice quality on multiple calls. After looking at packet captures, the engineer notices high levels of jitter. Which two areas does the engineer check to prevent jitter? (Choose two.)

- A. The network meets bandwidth requirements.
- B. MTP is enabled on the SIP trunk to Cisco Unified Border Element.
- C. Cisco UBE manages voice traffic, not data traffic.
- D. All devices use wired connections instead of wireless connections.
- E. Voice packets are classified and marked.

Answer: AE

Explanation:

Reference: <https://www.cisco.com/c/en/us/support/docs/voice/voice-quality/20371-troubleshoot-qos-voice.html>

NEW QUESTION 3

A customer has Cisco Unity Connections that is integrated with LDAP. As a Unity Connection administrator, you have received a request to change the first name for VM user. Where must the change be performed?

- A. Cisco Unity Connection
- B. Cisco Unified Communications Manager end user
- C. Active Directory
- D. Cisco IM and Presence

Answer: C

NEW QUESTION 4

On which Cisco Unified Communications Manager nodes can the TFTP service be enabled?

- A. any node
- B. any two nodes
- C. only nodes that have Cisco Unified CM service enabled
- D. any subscriber nodes

Answer: C

Explanation:

You can configure the TFTP service on the first node or a subsequent node, but usually you should configure it on the first node. For small systems, the TFTP server can coexist with a Cisco Unified Communications Manager on the same server.

NEW QUESTION 5

Which issue causes slips on a PRI?

- A. incorrect clock source
- B. incorrect encapsulation
- C. incorrectly configured time zone
- D. change in the line code

Answer: A

NEW QUESTION 6

Refer to the exhibit.

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/net_mgmt/prime/collaboration/12-1/assurance/advanced/guide/cpc_b_cisco-prime-collaboration-assurance-guide-advanced-12-1/cpc_b_cisco-prime-collaboration-assurance-guideadvanced-12-1_chapter_01111.html

NEW QUESTION 10

Which command in the MGCP gateway configuration defines the secondary Cisco Unified Communications Manager server?

- A. mgcapp
- B. ccm-manager fallback-mgcp
- C. mgcp call-agent
- D. ccm-manager redundant-host

Answer: B

NEW QUESTION 10

Refer to the exhibit.

```
INVITE sip:1@10.10.10.219;user=phone SIP/2.0
Via: SIP/2.0/TCP 10.10.10.84:50083;branch=z9hG4bK471df613
From: "1234 - My Phone" <sip:1234@10.10.10.219>;tag=381c1aba7a78002c558eda31-12b8af63
To: <sip:1@10.10.10.219>
Call-ID: 381c1aba-7a78000d-4ca6894a-41dd3e0f@10.10.10.84
Max-Forwards: 70
CSeq: 101 INVITE
Contact: <sip:1234@10.10.10.84:50083;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Allow-Events: kpml,dialog
Content-Type: application/sdp
Content-Length: 658

v=0
o=Cisco-SIPUA 26529 0 IN IP4 10.10.10.84
s=SIP Call
b=AS:4064
t=0 0
m=audio 32136 RTP/AVP 114 9 124 113 115 0 8 116 18
c=IN IP4 10.10.10.84
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114
maxplaybackrate=16000;sprop-maxcapture=16000;maxaveragebitrate=64000;stereo=0;sprop-
stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0,mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1,mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=sendrecv
```

When a UC Administrator is troubleshooting DTMF negotiated by this SIP INVITE, which two messages should be examined next to further troubleshoot the issue? (Choose two.)

- A. REGISTER
- B. UPDATE
- C. PRACK
- D. NOTIFY
- E. SUBSCRIBE

Answer: DE

NEW QUESTION 11

An engineer wants to manually deploy a Cisco Webex DX80 video endpoint to an end user. Which type of provisioning can be configured on the endpoint?

- A. CUBE
- B. CMS
- C. CUCM
- D. Edge

Answer: C

Explanation:

Reference: <https://www.cisco.com/c/en/us/products/collateral/collaboration-endpoints/desktop-collaboration-experience-dx600-series/datasheet-c78-731879.html>

NEW QUESTION 13

Refer to the exhibit.

```
rule 1 /^\(0[25]..\)\- \(...\)\- \(...\$\)/ /\1\2\3/
```

The translation rule is configured on the voice gateway to translate DNIS. What is the outcome if the gateway receives 0255-343-1234 as DNIS?

- A. The translation is not matched because DNIS contains “-”.
- B. The translation is not matched because DNIS does not end with a “\$”.
- C. The translation is matched and the translated number is 02553431234.
- D. The translation is matched and the translated number is 025553431234.

Answer: A

NEW QUESTION 14

What field is configured to change the caller ID information on a SIP route pattern?

- A. Route Partition
- B. Called Party Transformation Mask
- C. Calling Party Transformation Mask
- D. Connected Line ID Presentation

Answer: D

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0100111.pdf

NEW QUESTION 18

An engineer configures Cisco Unified Communications Manager to prevent toll fraud. At which two points does the engineer block the pattern in Cisco Unified CM to complete this task? (Choose two.)

- A. route pattern
- B. route group
- C. translation pattern
- D. partition
- E. CSS

Answer: CE

NEW QUESTION 19

Where is the default for Maximum Session Bit Rate for a region configured?

- A. Service Parameter Configuration
- B. Enterprise Phone Configuration
- C. Enterprise Parameters Configuration
- D. Region Configuration

Answer: A

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_1_1/ccmcfg/CUCM_BK_A34970C5_00_admin-guide-91/CUCM_BK_A34970C5_00_admin-guide-91_chapter_0111.html

NEW QUESTION 20

What causes poor voice quality and video pixelization in a video call?

- A. The QoS is configured incorrectly.
- B. A firewall is blocking the RTP ports.
- C. Cisco Unified Communications Manager is configured to use G.711 instead of G.729.
- D. 1 Gbps network ports are used instead of 100 Mbps network ports.

Answer: A

NEW QUESTION 24

What is a software-based media resource that is provided by the Cisco IP Voice Media Streaming Application?

- A. video conference bridge
- B. auto-attendant
- C. transcoder
- D. annunciator

Answer: D

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab09/clb09/media.html

NEW QUESTION 28

Which protocols does Cisco IM and Presence use to authenticate Jabber?

- A. XMPP
- B. SOAP
- C. TCP
- D. LDAP
- E. QBE

Answer: AB

NEW QUESTION 32

Which call flow matches traffic from a Mobile and Remote Access registered endpoint to central call control?

- A. Endpoint > Expressway-E > Expressway-C > Cisco Unified CM
- B. Endpoint > Expressway-E > Cisco Unified CM
- C. Endpoint > Expressway-C > Cisco Unified CM
- D. Endpoint > Expressway-C > Expressway-E > Cisco Unified CM

Answer: A

NEW QUESTION 36

Which recommendation is the best practice for marking video and voice media in a Cisco Unified Communications network?

- A. Voice Cos 5 (IP Precedence 6, PHB AF41, or DSCP 16) Video Cos 4 (IP Precedence 5, PHB EF, or DSCP 32)
- B. Voice Cos 6 (IP Precedence 4, PHB AF41, or DSCP 24) Video Cos 5 (IP Precedence 4, PHB EF, or DSCP 34)
- C. Voice Cos 5 (IP Precedence 2, PHB EF, or DSCP 48) Video Cos 4 (IP Precedence 4, PHB AF41, or DSCP 46)
- D. Voice Cos 5 (IP Precedence 5, PHB EF, or DSCP 46) Video Cos 4 (IP Precedence 4, PHB AF41, or DSCP 34)

Answer: D

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/netstruc.html

NEW QUESTION 40

Refer to the exhibit.

```
v=0
o=Cisco-SIPUA 13439 0 IN IP4 10.10.10.10
s=SIP Call
b=AS:4064
t=0 0
m=audio 0 RTP/AVP 114 9 124 113 115 0 8 116 18 101
c=IN IP4 10.10.10.10
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114 maxplaybackrate=16000;sprop-maxcapture=16000;maxaveragebitrate=64000;stereo=0;sprop-stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0,mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1,mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

A call is failing to establish between two SIP Devices. The called device answers with this SDP. Which SDP parameter causes this issue?

- A. The payload for G.711ulaw must be 18.
- B. The calling device did not offer aptime value.
- C. The media stream is set to sendonly.
- D. The RTP port is set to 0.

Answer: D

NEW QUESTION 42

Regarding SIP integrations with Cisco Unified Communications Manager, if the Cisco Unity Connection is configured to listen for incoming IPv4 and IPv6 traffic, how should the addressing mode be set up in the Cisco Unity Connection?

- A. Set up is not required.

- B. Set up for each group to use IPv4 and IPv6.
- C. Set up media ports for each port group to use IPv4.
- D. Set up IPv4 and IPv6 in Cisco Unified CM.

Answer: B

NEW QUESTION 45

An engineer is notified that the Cisco TelePresence MX800 that is registered in Cisco Unified Communications Manager shows an empty panel, and the Touch 10 shows a corresponding icon with no action when pressed. Where does the engineer go to remove the inactive custom panel?

- A. The Software Upgrades page in CUCM OS Administration
- B. The In-Room Control Editor on the webpage of the MX800
- C. The phone configuration page in CUCM Administration
- D. The SIP Trunk Security Profile page in CUCM Administration

Answer: A

NEW QUESTION 49

Which DHCP option must be set up for new phones to obtain the TFTP server IP address?

- A. option 15
- B. option 6
- C. option 66
- D. option 120

Answer: C

Explanation:

Reference: <https://blog.router-switch.com/2013/03/dhcp-option-150-dhcp-option-66/>

NEW QUESTION 53

Given the H.323 gateway configuration and using Cisco best practices, how must the called party transformation pattern be configured to ensure that a proper ISDN type of number is set?

```

voice translation-rule 40
  rule 1 /3...$/ /408555&/
  !
voice translation-profile INT
  translate calling 40
  !
dial-peer voice 9011 pots
  translation-profile outgoing INT
  destination-pattern 9011T
  port 0/1/0:23
  
```

A. **Pattern Definition**

Pattern*	+!
Partition	PT_US_VG_CD_Out_xForm
Description	US International calling
Numbering Plan	< None >
Route Filter	< None >
<input checked="" type="checkbox"/> Urgent Priority	
<input type="checkbox"/> MLPP Preemption Disabled	

Called Party Transformations

Discard Digits	PreDot
Called Party Transformation Mask	
Prefix Digits	9011
Called Party Number Type*	International
Called Party Numbering Plan*	ISDN

B.

Pattern Definition

Pattern*

Partition

Description

Numbering Plan

Route Filter

Urgent Priority

MLPP Preemption Disabled

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type*

Called Party Numbering Plan*

Pattern Definition

Pattern*

Partition

Description

Numbering Plan

Route Filter

Urgent Priority

MLPP Preemption Disabled

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type*

Called Party Numbering Plan*

Pattern Definition

Pattern*

Partition

Description

Numbering Plan

Route Filter

Urgent Priority

MLPP Preemption Disabled

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type*

Called Party Numbering Plan*

Answer: C

NEW QUESTION 58

A user reports transfer failure from an IP phone for calls received from a PSTN to another PSTN number. What is a reason for these failures?

- A. The IP phone is configured with the wrong region.
- B. The incoming calling search space of the SIP trunk does not include the partition of the line in the IP phone.
- C. The service parameter related to Offnet to Offnet Call Transfer is set to TRUE.
- D. The gateway is configured with the wrong device pool.

Answer: D

NEW QUESTION 63

How are E.164 called-party numbers normalized on a globalized call-routing environment in Cisco Unified Communications Manager?

- A. Normalization is achieved by stripping or translating the called numbers to internally used directory numbers.
- B. Normalization is achieved by setting up calling search spaces and partitions at the SIP trunks for PSTN connection.
- C. Call ingress must be normalized before the call being routed.
- D. Normalization is not required.

Answer: A

NEW QUESTION 65

An engineer is configuring a BOT device for a Jabber user in Cisco Unified Communication Manager Which phone type must be selected?

- A. third-party SIP device
- B. Cisco Dual Mode for iPhone
- C. Cisco Dual Mode for Android
- D. Cisco Unified Client Services Framework

Answer: C

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/jabber/11_5/CJAB_BK_D00D8CBD_00_deployment-installation-guide-cisco-jabber115/CJAB_BK_D00D8CBD_00_deployment-installation-guide-ciscojabber115_chapter_01000.html

NEW QUESTION 69

Refer to the exhibit.

```
000193: Dec 5 14:35:31.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:32.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:33.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:34.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
```

Given this "debug isdn q921" output, what is the problem with the PRI?

- A. Layer 1 is down on the controller.
- B. PRI does not have an IP address configured on the interface.
- C. Nothing, the PRI is sending keepalives.
- D. Layer 2 is down on the controller.

Answer: D

NEW QUESTION 74

Which method is used to avoid toll fraud with Cisco Unified Communications Manager calls?

- A. call policy service
- B. TOLLFRAUD_APP
- C. default zone access rules
- D. class of service

Answer: D

NEW QUESTION 79

A customer wants a video conference with five Cisco TelePresence IX5000 Series systems. Which media resource is necessary in the design to fully utilize the immersive functions?

- A. Cisco PVDM4-128
- B. software conference bridge on Cisco Unified Communications Manager
- C. Cisco Webex Meetings Server
- D. Cisco Meeting Server

Answer: C

NEW QUESTION 80

Which settings are needed to configure the SIP route pattern in Cisco Unified Communications Manager?

- A. pattern usage, IPv6 pattern, and SIP trunk/Route list
- B. pattern usage, IPv4 pattern, IPv6 pattern, and description
- C. pattern usage, IPv4 pattern, and SIP trunk/Route list
- D. SIP trunk/Route list, description, and IPv4 pattern

Answer: C

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0100111.pdf

NEW QUESTION 85

Which issue can occur if QoS is not deployed on a Cisco Collaboration architecture across the WAN?

- A. 403 Forbidden errors on SIP calls
- B. excessive jitter
- C. unexpected shut-down on Cisco Unified Communications Manager
- D. packet fragmentation

Answer: B

NEW QUESTION 86

Which Cisco Unified Communications Manager service parameter should be enabled disconnect a multiparty call when the call initiator hangs up?

- A. Drop Ad Hoc Conference
- B. H.225 Black Setup Destination
- C. Block OffNet To OffNet Transfer
- D. Enterprise Feature Access Code for Conference

Answer: A

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmsys/CUCM_BK_SE5FCFB6_00_cucm-system-guide-100/CUCM_BK_SE5FCFB6_00_cucm-system-guide-100_chapter_011000.html#CUCM_TK_DFC66444_00

NEW QUESTION 90

When a remote office location is set up with limited bandwidth resources, which codec carries the most voice calls?

- A. G.711
- B. G.722
- C. G.723
- D. G.729

Answer: D

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/media.html

NEW QUESTION 91

Which call routing pattern is used for phone numbers that are in the E.164 format?

- A. \+! Route Pattern
- B. \+! Translation Pattern
- C. /+! Route Pattern
- D. \+1.[2-9]XX[2-9]XXXXXX Called Party Transformation Pattern

Answer: D

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12/dialplan.html#pgfId-1591747

NEW QUESTION 93

An engineer is designing a high availability and failover solution for two Cisco Unified Border Element routers. The first router (cube1.ab?.com) takes 60% of the calls and the second router (cube2.abc.com) takes 40% of the calls. Assume all DNS A records have been created. Which two SRV records are needed for a load balanced solution? (Choose two.)

- A. _sip._udp.abc.com 60 IN SRV 2 60 cube1.abc.com
- B. _sip._udp.abc.com 60 IN SRV 60 1 cube1.abc.com
- C. _sip._udp.abc.com 60 IN SRV 1 40 cube2.abc.com
- D. _sip._udp.abc.com 60 IN SRV 3 60 cube2.abc.com
- E. _sip._udp.abc.com 60 IN SRV 1 60 cube1.abc.com

Answer: CE

NEW QUESTION 97

Which two configuration elements are part of the Cisco Unified Communications Manager toll-fraud prevention? (Choose two.)

- A. SIP trunk security profile
- B. Calling Search Space
- C. SUBSCRIBE Calling Search Space
- D. feature control policy
- E. partition

Answer: BE

NEW QUESTION 102

Which configuration step is necessary for a Cisco SIP phone to synchronize its time with a specific source?

- A. Add a Phone NTP Reference to the Date/Time Group.
- B. Assign the device to the correct region.
- C. Change the Time Format from 24-hour to 12-hour.
- D. Change the Time Zone from "America/Los_Angeles" to "Etc/GMT+8".

Answer: A

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0110.html

NEW QUESTION 103

As a voice engineer, which two recommendations do you to make to your company to optimize Cisco Unified Communications Manager configuration to reduce the number of toll fraud incidents? (Choose two.)

- A. Classify all route patterns as on-net and prohibit on-net to on-net call transfers in Cisco Unified CM service parameters.
- B. Classify all route patterns as off-net and prohibit off-net to off-net call transfers in Cisco Unified CM service parameters.
- C. Classify all route patterns as on-net or off-net and prohibit off-net to off-net call transfers in Cisco Unified CM service parameters.
- D. Inbound CSS on any gateway typically should have access to internal destinations and PSTN destinations.
- E. Inbound CSS on any gateway typically should have access to internal destinations only and not PSTN destinations.

Answer: BE

NEW QUESTION 108

A user dials 9011841234567 to reach Vietnam. Which steps send the call to the PSTN provider as 011841234567?

- A. in the Called Party Transformation Pattern Configuration section, configure the Pattern as 9.011841234567
configure the Discard Digits as Predot
- B. in the Calling Party Transformation Patterns section, configure the Pattern as 9.011841234567
configure the Discard Digits as Predot 10-10-Dialing
- C. in the Called Party Transformation Pattern Configuration section, configure the Pattern as 9.011841234587
configure the Discard Digits as Predot 10-10-Dialing
- D. in the Calling Party Transformation Patterns section, configure the Pattern as a 011841234557
configure the Discard Digits as Predot

Answer: A

NEW QUESTION 113

Refer to the exhibit.

```
dial-peer voice 10 voip
  destination-pattern 1...
  session target ipv4:10.1.1.1
  no vad
```

An engineer configures a VoIP dial peer on a Cisco gateway. Which codec is used?

- A. G.711ulaw
- B. No codec is used (missing codec command).
- C. G.711alaw
- D. G.729r8

Answer: A

Explanation:

Reference: <https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/dialpeer/configuration/15-mt/vd-15-mt-book/vd-dp-cfg-examp.pdf>

NEW QUESTION 117

Due to provider requirements, outgoing calls from the Enterprise to the PSTN must start with channel 1. Which ISDN command changes the channel selection an IOS to meet this requirement?

- A. isdn bchan-number-order decending
- B. isdn bchan-number-order ascending
- C. isdn protocol-emulate network
- D. isdn incoming-voice voice

Answer: B

NEW QUESTION 122

Which description of the function of call handlers in Cisco Unity Connection is true?

- A. They answer calls, take messages, and provide menus of options.
- B. They provide access to a corporate directory by playing an audio list that users and outside callers use to reach users and leave messages.
- C. They collect information from callers by playing a series of questions and recording the answers.
- D. They control outgoing calls by allowing you to specify the numbers that Cisco Unity Connection can dial to transfer calls, notify users of messages, and deliver faxes.

Answer: A

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/10x/administration/guide/10xcucsagx/10xcucsag080.html

NEW QUESTION 127

An engineer must configure an MGCP gateway and register it to Cisco Unified Communications Manager. Which prerequisite must be met before applying the gateway commands to enable MGCP?

- A. The MGCP gateway and the Cisco Unified CM must be able to communicate over ports 5060 and 5061.
- B. Cisco Unified CM and the MGCP gateway must utilize the SIP OPTIONS ping feature to monitor status.
- C. The MGCP gateway must have voice service VoIP configured.
- D. The MGCP gateway and the Cisco Unified CM must be able to communicate over ports 2427, 2428, and 2727.

Answer: D

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/ios/voice/cminterop/configuration/guide/12_4t/vc_12_4t_book/vc_ucm_mgcp_gw.html

NEW QUESTION 131

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