Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

https://www.2passeasy.com/dumps/350-801/

NEW QUESTION 1
A Cisco Unity Connection Administrator must set a voice mailbox so that it can be accessed from a secondary device. Which configuration on the voice mailbox makes this change?

A. Attempt Forward routing rule
B. Alternate Extensions
C. Alternate Names
D. Mobile User

Answer: A

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
An engineer encounters third-party devices that do not support Cisco Discovery Protocol. What must be configured on the network to allow device discovery?

A. LACP
B. TFTP
C. LLDP
D. SNMP

Answer: C

NEW QUESTION 4
Multiple route patterns match a number. How does Cisco Unified Communications Managers determine which pattern to use?

A. the one that comes first in numerical order
B. the one with the longest match
C. the one with the closest match
D. the one that discards everything PreDot

Answer: C

Explanation:
Reference: https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-callmanager/13920-call-routing.html#bcr

NEW QUESTION 5
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 6
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 7

Which command is used in Cisco IOS XE TDM gateway to configure the voice T1/E1 controller to provide clocking?

A. clock source line
B. Cisco IOS XE TDM gateway T1/E1 controller cannot provide clocking.
C. clocking source internal
D. clocking source network

Answer: C

Explanation:
Reference: https://www.cisco.com/c/en/us/td/docs/routers/access/interfaces/NIM/software/configuration/guide/4gen-t1-e1-nim-guide.html

NEW QUESTION 8
Refer to the exhibit.

```
INVITE sip:4000@172.16.1.1:5061 SIP/2.0
Via: SIP/2.0/TLS 172.16.2.143:5061;branch=z9hG4bK8FD315E7
Remote-Party-ID: <sip:+14088335000@172.16.2.143>;party=calling;screen=no; privacy=off
From: <sip:+14088335000@172.27.2.143>;tag=7B42E5F6-9B8
To: <sip:4000@172.16.1.1>
Date: Tue, 06 Aug 2019 15:03:05 GMT
Call-ID: 4EA4363-B77111E9-8A4AFFCF-10B6D71B@172.16.2.143
Supported: 100rel,timer,resource-priority, replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 0082391505-3077640681-2319777743-0280418075
User-Agent: Cisco-SIPGateway/IOS-15.5.3.S4b
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1565089565
Contact: <sip:+ 14088335000@172.16.2.143:5061;transport=tls>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 68
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 416
v=0
o=CiscoSystemsSIP-GW-UserAgent 8486 8298 IN IP4 172.16.2.143
s=SIP Call
c=IN IP4 172.16.2.143
t=0 0
m=audio 44612 RTP/SAVP 0 101
c=IN IP4 172.16.2.143
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
```

This INVITE is sent to an endpoint that only supports G729. What must be done for this call to succeed?

A. Nothing: both sides support G.729.
B. Add a transcoder that supports G711ulaw and G.729.
C. Add a media termination point that supports G.711ulaw and G.729.
D. Nothing: both sides support payload type 101.

Answer: D

NEW QUESTION 9
Calls are being delivered to the end user in the globalized format. Where does an engineer configure the calling number into a localized format?

A. route pattern
B. service parameters
C. IP phone
D. gateway

Answer: C

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-maxcapturerate=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 10
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 11
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 13
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 16
How can an engineer determine location-based CAC bandwidth requirements for Cisco Unified Communications Manager?

A. Set the requirements in the service parameters.
B. Add the requirements for each audio and video codec and how many calls must be supported.
C. Execute the Resource Reservation Protocol to return location-based requirements.
D. Calculate the number of calls against the license for Cisco Unified Border Element to determine calls per location.

Answer: A

NEW QUESTION 18
When a user dials a number with a phone that is registered to the Cisco Unified Communications Manager, what is the default timeout before the number is sent?

A. 15 seconds
B. 5 seconds
C. 10 seconds
D. 3 seconds

Answer: C

Explanation:
Reference: https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-callmanager/13920-call-routing.html

NEW QUESTION 19
Which action is required if an engineer wants to have Cisco Unified Communications Manager control the configuration for an MGCP gateway?

A. Apply the ccm-manager configuration commands to the gateway.
B. Upload the custom configuration in the TFTP server in Cisco Unified CM.
C. From Cisco Unified CM > Device > Gateway > Add gateway, check the auto-configuration check box.
D. Configure the Cisco Unified CM's IP in voice service VoIP.

Answer: C

NEW QUESTION 22
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 26
Which two conditions must a user meet to provision a new device using the Self-Provisioning feature? (Choose two.)

A. The user must have a primary extension.
B. At least two DNs must be assigned to the user device.
C. The user must be part of “Standard CCM Super User”.
D. The user must have the appropriate universal device template linked to the user profile.
E. The user must have at least user device profile assigned.

Answer: AD

NEW QUESTION 31
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 32
What is a valid class included in the 8-Class QoS Strategy in a VoIP network?

A. Assured Forwarding
B. Broadcast Video
C. Multimedia Conferencing
D. Real-Time Interactive

Answer: C

Explanation:
Reference: https://www.ciscopress.com/articles/article.asp?p=2756478&seqNum=8

NEW QUESTION 33
A present redundancy group is deployed, and an engineer with ID012345678 initiates a manual fallback. Which statement about Cisco Server Recovery Manager is true?

A. disconnects all users that had been failed over, and the users must log in again.
B. disconnects all users that had been failed over
C. restarts critical on the secondary node
D. restarts the Cisco Presence Engine

Answer: B

NEW QUESTION 38
An engineer with ID012345678 must build an international dial plan in Cisco Unified Communications Manager. Which action should be taken when building a variable-length route pattern?

A. reduce the T302 timer to less than 4 seconds
B. configure single route pattern for international calls
C. create a second route pattern followed by the # wildcard
D. set up all international route patterns to 0.!

Answer: A

NEW QUESTION 43
Refer to the exhibit

```
INVITE sip:2002@10.10.10.10:5060 SIP/2.0
[..truncated..]
v=0
o=UAC 6107 7816 IN IP4 10.10.10.11
s=SIP Call
c=IN IP4 10.10.10.11
t=0 0
m=audio 8190 RTP/AVP 18 110
c=IN IP4 10.10.10.11
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:110 telephone-event/8000
a=fmtp:110 0-16
a=ptime:20

SIP/2.0 200 OK
[..truncated..]
v=0
o=UAS 4692 9609 IN IP4 10.10.10.10
s=SIP Call
c=IN IP4 10.10.10.10
t=0 0
m=audio 8056 RTP/AVP 18
c=IN IP4 10.10.10.10
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
```

The SDP offer/answer has been completed successfully but there is no DTMF when users press keys. What is the cause of the issue?

A. DTMF was negotiated properly in these messages.
B. G.729 rather than G.711ulaw was negotiated.
C. Payload type 110 was negotiated rather than type 101.
D. DTMF was not negotiated on the call.

Answer: D

NEW QUESTION 45
A remote office has a less-than-optimal WAN connection and experiences packet loss, delay, and jitter. Which VoIP codec should be used in this situation?

A. G.711ulaw
B. iLBC
C. G.722.1
D. G.729A

Answer: D

Explanation:
Reference: https://community.cisco.com/t5/collaboration-voice-and-video/summary-of-cucm-supported-codecs/ta-p/3162905

NEW QUESTION 49
What is the major difference between the two possible Cisco IM and Presence high-availability modes?

A. Balanced mode provides user load balancing and user failover in the event of an outag
B. Active/standby mode provides an always on standby node in the event of an outage, and it also provides load balancing.
C. Balanced mode provides user load balancing and user failover only for manually generated failover
D. Active/standby mode provides an unconfigured standby node in the event of an outage, but it does not provide load balancing.
E. Balanced mode provides user load balancing and user failover in the event of an outag
F. Active/standby mode provides an always on standby node in the event of an outage, but it does not provide load balancing.
G. Balanced mode does not provide user load balancing, but it provides in the event of an outag
H. Active/standby mode provides an always on standby node in the event of an outage, but it does not provide load balancing.

Answer: C

NEW QUESTION 52
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 55
A collaboration engineer must configure Cisco Unified Border Element to support up to five concurrent outbound calls across an Ethernet link with a bandwidth of 160 kb to the Internet Telephony Service Provider. Which set of commands allows the engineer to complete the task without compromising voice quality?

A. dial-peer voice 1 voip translation- profile outgoing Strip9 max-conn 5 destination-pattern 91[2-9]..[2-9]……$session protocol sipv2 session target ipv4:142.45.10.1 dtmf-relay rtp-nte sip- notify sip-kpml
B. dial-peer voice 1 voip translation-profile outgoing Strip9 max- conn 5destination-pattern 91[2-9]..[2-9]……$ session protocol sipv2 session target ipv4:142.45.10.1 dtmf-relay rtp- nte sip-notify sip-kpml codec ilbc mode 20
C. dial-peer voice 1 voip translation- profile outgoing Strip9 max-conn 5 destination-pattern 91[2-9]..[2-9]……$ session protocol sipv2 session target ipv4:142.45.10.1 dtmf-relay rtp- nte sip-notify sip-kpml codec aacld
D. dial-peer voice 1 voip translation- profile outgoing Strip9 max-conn 5 destination-pattern 91[2-9]..[2-9]……$ session protocol sipv2 session target ipv4:142.45.10.1 dtmf-relay rtp- nte sip-notify sip-kpml codec mp4a- latm

Answer: B

NEW QUESTION 59
How does Cisco Unified Communications Manager perform a digit analysis on-hook versus off-hook for an outbound call from a Cisco IP phone that is registered to Cisco Unified CM?

A. On-hook, Unified CM performs a digit-by-digit analysis, off-hook, Unified GM considers all digits were dialed and does not wait for additional digits.
B. On-hook, Unified CM considers all digits were dialed and does not wait for additional digits, off-hook, Unified CM performs a digit-by-digit analysis.
C. On-hook, by pressing the digits and entering "#" to process the call, Unified CM performs a digit-by-digit analysis; off-hoo
D. Unified CM analyzes all digits as a string.
E. On-hook, no digit analysis is performed, off-hoo
F. Unified CM requires the "#* to start the digit analysis.

Answer: C

NEW QUESTION 62
A Cisco TelePresence SX80 suddenly has issues displaying main video to a display over HDMI. Which command can you use from the SX80 admin CLI to check the video output status to the monitor?

A. xStatus Video Output
B. xCommand Video Status
C. xConfiguration Video Output

D. xStatus HDMI Output

Answer: C

NEW QUESTION 63
Which two functions are provided by Cisco Expressway Series? (Choose two.)

A. interworking of SIP and H.323
B. endpoint registration
C. intercluster extension mobility
D. voice and video transcoding
E. voice and video conference

Answer: AD

Explanation:
Reference: https://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/expressway/config_guide/X8-11/Cisco-Meeting-Server-2-4-with-Cisco-Expressway-Deployment-Guide_X8-11-4.pdf

NEW QUESTION 64
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 69
Which statement about Cisco Unified Communications Manager and Cisco IM and Presence backups is true?

A. Backups should be scheduled during off-peak hours to avoid system performance issues.
B. Backups are saved as .tar files and encrypted using the web administrator account.
C. Backups are saved as unencrypted .tar files.
D. Backups are not needed for subscriber Cisco Unified Communications Manager and Cisco IM and Presence servers.

Answer: A

Explanation:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1_SU1/Administration/cucm_b_administration-guide-1151su1/cucm_b_administration-guide-1151su1_chapter_01010.html#CUCM_TK_S7FC26D5_00

NEW QUESTION 72
Refer to the exhibit.

```
ISDN Serial1:23 interface
         dsl 1, interface ISDN Switchtype =
primary-5ess
     Layer 1 Status:
         ACTIVE
     Layer 2 Status:
         TEI = 0, Ces = 1, SAPI = 0, State =
TEI_ASSIGNED
     Layer 3 Status:
         0 Active Layer 3 Call(s)
     Activated dsl 1 CCBs = 0
     The Free Channel Mask: 0x807FFFFF
     Total Allocated ISDN CCBs = 5
```

What is a possible cause of the PRI issue?

A. The cable is unplugged.
B. The clock source is incorrect.
C. The controller shut down.
D. The framing is configured incorrectly.

Answer: D

NEW QUESTION 74

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 79
Which two DNS records must be created to configure Service Discovery for or premises Jabber? (Choose two.)

A. _cisco-uds._tls.cisco.com pointing to the IP address of Cisco Unified Communications Manager
B. _cuplogin._tcp.cisco.com pointing to a record of IM&P
C. _cuplogin._tls.cisco com pointing to the IP address of IM&P
D. _cisco-uds._tcp.cisco com painting to a record of Cisco Unified CM
E. _xmpp._tls.cisco.com pointing to a record of IM&P

Answer: AB

Explanation:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/jabber/Windows/9_7/CJAB_BK_C606D8A9_00_cisco-jabber-dns-configuration-guide/CJAB_BK_C606D8A9_00_cisco-jabber-dns-configuration-guide_chapter_010.html

NEW QUESTION 84
Which endpoint feature is supported using Mobile and Remote Access through Cisco Expressway?

A. SRST
B. SSO
C. H.323 registration proxy to Cisco Unified Communications Manager
D. MGCP gateway registration

Answer: C

NEW QUESTION 85
A DTMF mismatch is occurring between an MGCP gateway registered FXS port and a Cisco Unified Communications Manager SIP trunk. Which media resource can be leveraged to interwork this mismatch?

A. Conference Bridge
B. Trusted Relay Point
C. Media Termination Point
D. Annunciator

Answer: C

NEW QUESTION 86
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 87
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 88
......