350-801 Dumps

Implementing and Operating Cisco Collaboration Core Technologies

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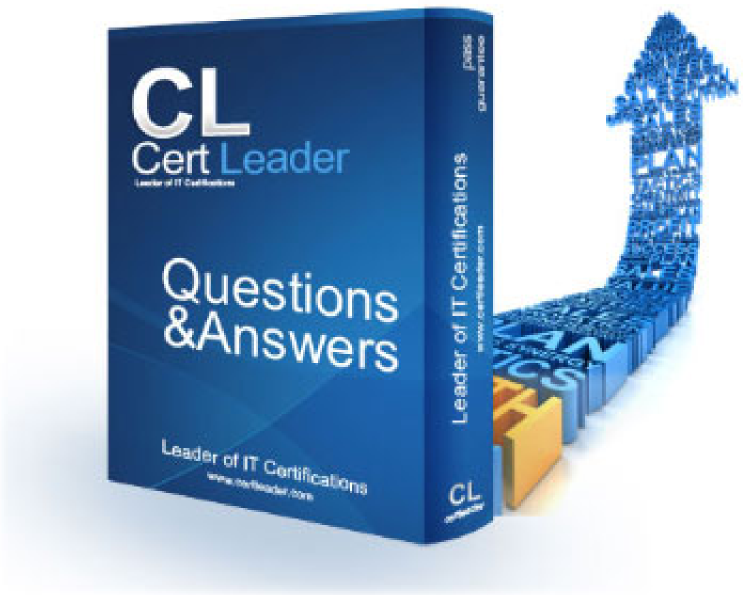

NEW QUESTION 1
Which Cisco Unified Communications Manager configuration is required for SIP MWI integrations?

A. Select “Redirecting Diversion Header Delivery - Inbound” on the SIP trunk.
B. Enable “Accept presence subscription” on the SIP Trunk Security Profile.
C. Enable “Accept unsolicited notification” on the SIP Trunk Security Profile.
D. Select “Redirecting Diversion Header Delivery - Outbound” on the SIP trunk.

Answer: A

NEW QUESTION 2
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 3
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:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
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 4
Which action prevents toll fraud in Cisco Unified Communications Manager?

A. Implement toll fraud restriction in the Cisco IOS router.
B. Implement route patterns in Cisco Unified CM.
C. Allow off-net to off-net transfers.
D. Configure ad hoc conference restriction.

Answer: B

Explanation:
Reference: https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-express/107626-cme-toll-fraud.html

NEW QUESTION 5
A user forwards a corporate number to an international number. What are two methods to prevent this forwarded call? (Choose two.)

A. Set the Call Classification to OnNet for the international route pattern.
B. Block international dial patterns in the SIP trunk CSS.
C. Configure a Forced Authorization Code on the international route pattern.
D. Set Call Forward All CSS to restrict international dial patterns.
E. Check Route Next Hop By Calling Party Number on the international route pattern.

Answer: BC

Explanation:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmetrans.html

NEW QUESTION 6
On which protocol and port combination does Cisco Prime Collaboration receive notifications (Traps and InformRequests) from several network devices in the Collaboration infrastructure for which it has requested notifications?

A. UDP 162
B. TCP 80
C. UDP 161
D. TCP 161

Answer: A

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

NEW QUESTION 7
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 8
Refer to the exhibit.

```
Endpoint A:
m=audio 21796 RTP/AVP 108 9 104 105 101
  b=TIAS:64000
  a=extmap:14 http://protocols.cisco.com/timestamp#100us
  a=rtpmap:108 MP4A-LATM/90000
  a=fmtp:108 bitrate=64000;profile-level-id=24;object=23
  a=rtpmap:9 G722/8000
  a=rtpmap:104 G7221/16000
  a=fmtp:104 bitrate=32000
  a=rtpmap:105 G7221/16000
  a=fmtp:105 bitrate=24000
  a=rtpmap:101 telephone-event/8000
  a=fmtp:101 0-15
  a=trafficclass:conversational.audio.immersive.aq:admitted

Endpoint B:
m=audio 21796 RTP/AVP 105 0 8 18 101
  b=TIAS:64000
  a=extmap:14 http://protocols.cisco.com/timestamp#100us
  a=rtpmap:105 G7221/16000
  a=fmtp:105 bitrate=24000
  a=rtpmap:0 PCMU/8000
  a=rtpmap:8 PCMA/8000
  a=rtpmap:18 G729/8000
  a=fmtp:18 annexb=no
  a=rtpmap:101 telephone-event/8000
  a=fmtp:101 0-15
  a=trafficclass:conversational.audio.immersive.aq:admitted
```

Endpoint A calls endpoint B. What is the only audio codec that can be used for the call?

A. Telephone-event/8000
B. G7221/16000
C. PCMA/8000

D. G722/8000

Answer: B

NEW QUESTION 9
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 10
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 13
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 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 22
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 23
How many DNS SRV entries can be defined in the SIP trunk destination address field in Cisco Unified Communications Manager?

A. 1
B. 8
C. 16
D. 4

Answer: C

Explanation:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/CUCM_BK_SE5DAF88_00_cucm-system-configuration-guide-1151/CUCM_BK_SE5DAF88_00_cucm-system-configuration- guide1151_chapter_01110.html

NEW QUESTION 25
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 30
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-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=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 SOP. Which SDP parameter causes this issue?

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

Answer: D

NEW QUESTION 33
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 37
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 39
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 40
An engineer implements QoS in the enterprise network. Which command can be used to verify the correct classification and marking on a Cisco IOS switch?

A. show class-map interface GigabitEthernet 1/0/1
B. show policy-map interface GigabitEthernet 1/0/1
C. show policy-map

D. show access-lists

Answer: C

Explanation:
Reference: https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/qos_classn/configuration/xe-16/qos-classn-xe-16-book/qos-classn-mrkg-ntwk-trfc-xe.html

NEW QUESTION 41
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 45
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 49
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 53
An engineer configures local route group to simplify a dial plan. Where does the engineer set the route groups according to the local route group names that are configured?

A. CSS
B. route pattern
C. device pool
D. route list

Answer: D

NEW QUESTION 55
Which two functionalities does Cisco Expressway provide in the Cisco Collaboration architecture? (Choose two.)

A. Survivable Remote Site Telephony functionality
B. customer interaction management services
C. secure firewall and NAT traversal for mobile or remote Cisco Jabber and TelePresence Video endpoints
D. MGCP gateway registration
E. Secure business-to-business communications

Answer: CE

NEW QUESTION 59
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 60
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 64
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 68
Which configuration tells a switch part to send Cisco Discovery Protocol packets that configure an attached Cisco IP phone to trust tagged traffic that is received from a device that is connected to the access port on the Cisco IP phone?

A. Router# configure terminalRouter(config)# interface gigabitethernet 5/1 Router(config-if)# platform qos trust extend
B. Router# configure terminalRouter(config)# interface gigabitethernet 5/1 Router(config-if)# platform qos trust extend cos 3
C. Router# configure terminalRouter(config)# interface gigabitethernet 5/1 Router(config-if)# platform qos trust extend cos 5
D. Router# configure terminalRouter(config)# interface gigabitethernet 5/1 Router(config-if)# platform qos extend trust

Answer: A

NEW QUESTION 69
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 74
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 75
Which access control group is required on an end user to allow Jabber to do deskphone mode?

A. Allow Control of Device from CTI
B. Standard CTI Enabled
C. Standard CTI Allow Reception of SRTP Key Material
D. Standard CTI Secure Connection

Answer: B

NEW QUESTION 80
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 82
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 87
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 92
Which protocol does Cisco Prime Collaboration Assurance use to poll the health status of different systems in the Collaboration environment?

A. SIP
B. SNMP
C. SCCP
D. SMTP

Answer: B

Explanation:
Reference: https://www.cisco.com/c/en/us/products/collateral/cloud-systems-management/prime-collaboration/guide-c07-736946.html#_Toc446633083

NEW QUESTION 95
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 96
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 97
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 100
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 103
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 106
Which statement describes the outcome when the trust boundary is defined at the Cisco IP phone?

A. Packets or Ethernet frames are remarked at the access layer switch.
B. Packets or Ethernet frames are not remarked by the IP phone.
C. Packets or Ethernet frames are not remarked at the layer switch.
D. Packets or Ethernet frames are remarked at the distribution layer switch.

Answer: C

Explanation:
Reference: https://networklessons.com/quality-of-service/how-to-configure-qos-trust-boundary-on-cisco-switches

NEW QUESTION 107
An administrator is trying to change the default LINECODE for a voice ISDN T1 PRI. Which command makes this change?

A. linecode esf
B. linecode ami
C. linecode hdb3
D. linecode b8zs

Answer: D

Explanation:
Reference: https://www.cisco.com/en/US/docs/ios/dial/configuration/guide/dia_cfg_isdn_pri_external_docbase_0900e4b1806c752c_4container_external_docbase_0900e4b18216dd1b.html

NEW QUESTION 110
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 114
After an engineer runs the utils ntp status command on the Cisco Unified Communications Manager publisher, the stratum value is 16. Which issue can the Cisco Unified CM cluster experience?

A. Unified CM sends an NTPV4 packet.
B. Database replication is not synchronized on the Unified CM nodes.
C. The cluster loses access to port 124 at the firewall.
D. The date/time group on all phones defaults to the time zone of the engineer.

Answer: B

NEW QUESTION 119
Which configuration on Cisco Unified Communications Manager is required for SIP MWI to work?

A. The line partition must be inside the inbound CSS assigned to the CUC SIP trunk.
B. The line partition must be inside the rerouting CSS assigned to the Cisco Unity Connection SIP trunk.
C. Set the “Enable message waiting indicator” on the part group.
D. Assign a MWI extension on the mailbox.

Answer: C

NEW QUESTION 124
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
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