Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

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

NEW QUESTION 1
An engineer must extend the corporate phone system to mobile users connecting through the internet with their own devices. One requirement is to keep that as simple as possible for end users. Which infrastructure element achieves these goals?

A. Cisco Express Mobility
B. Cisco Expressway-C and Expressway-E
C. Cisco Unified Border Element
D. Cisco Unified Instant Messaging and Presence

Answer: C

NEW QUESTION 2
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 3
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 4
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 5
Refer to the exhibit.

```
C:\Users\CISCO>nslookup
Default Server: dns.example.com
Address: 192.168.100.1

>set type=SRV
> _collab-edge._tcp.example.com
Server: dns.example.com
Address: 192.168.100.1

Non-authoritative answer:
_collab-edge._tcp.example.com          SRV service location:
          priority       = 10
          weight         = 10
          port           = 8443
          srv hostname   = expe.example.com
```

You deploy Mobile and Remote Access for Jabber and discover that Jabber for Windows does no register to Cisco Unified Communication Manager while outside of the office. What is a cause of this issue?

A. Server 4.2.2.2 is not a valid DNS server.
B. The DNS record should be created for _cisco-uds._tcp.example.com.
C. The DNS record should be changed from _collab-edge._tcp.example.com to _collab-edge _tls.example.com.
D. The DNS record type should be changed from SRV to A.

Answer: C

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 6
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 7
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 8
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 9
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 10
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 10
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 15
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 18
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 21
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 22
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 25
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 27
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 30
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 33
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 38
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 40
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 43
How does an administrator make a Cisco IP phone display the last 10 digits of the calling number when the call is in the connected state, and also display the calling number in the E.164 format within call history on the phone?

A. Configure a translation pattern that has a Calling Party Transform Mask of XXXXXXXXXX.
B. On the inbound SIP trunk, change Significant Digits to 10.
C. Change the service parameter Apply Transformations On Remote Number to True.
D. Configure a calling party transformation pattern that keeps only the last 10 digits.

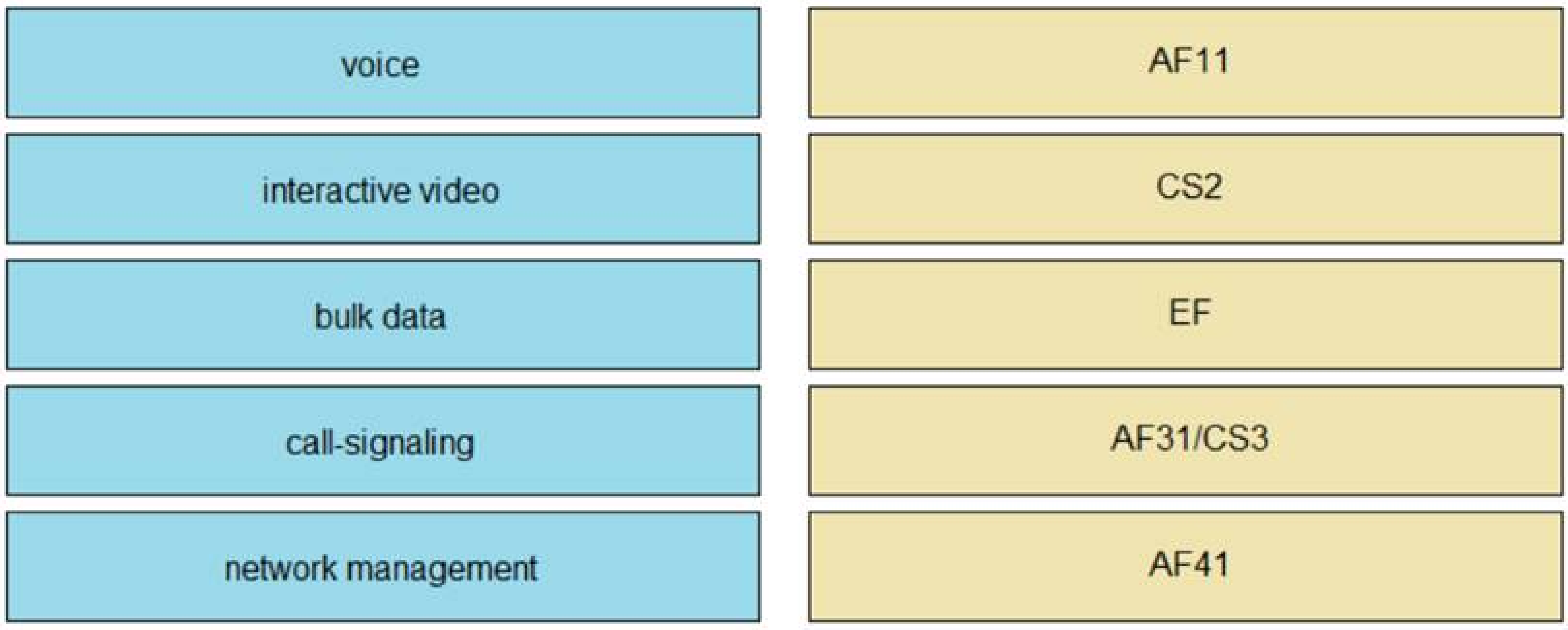
Answer: D

NEW QUESTION 47
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

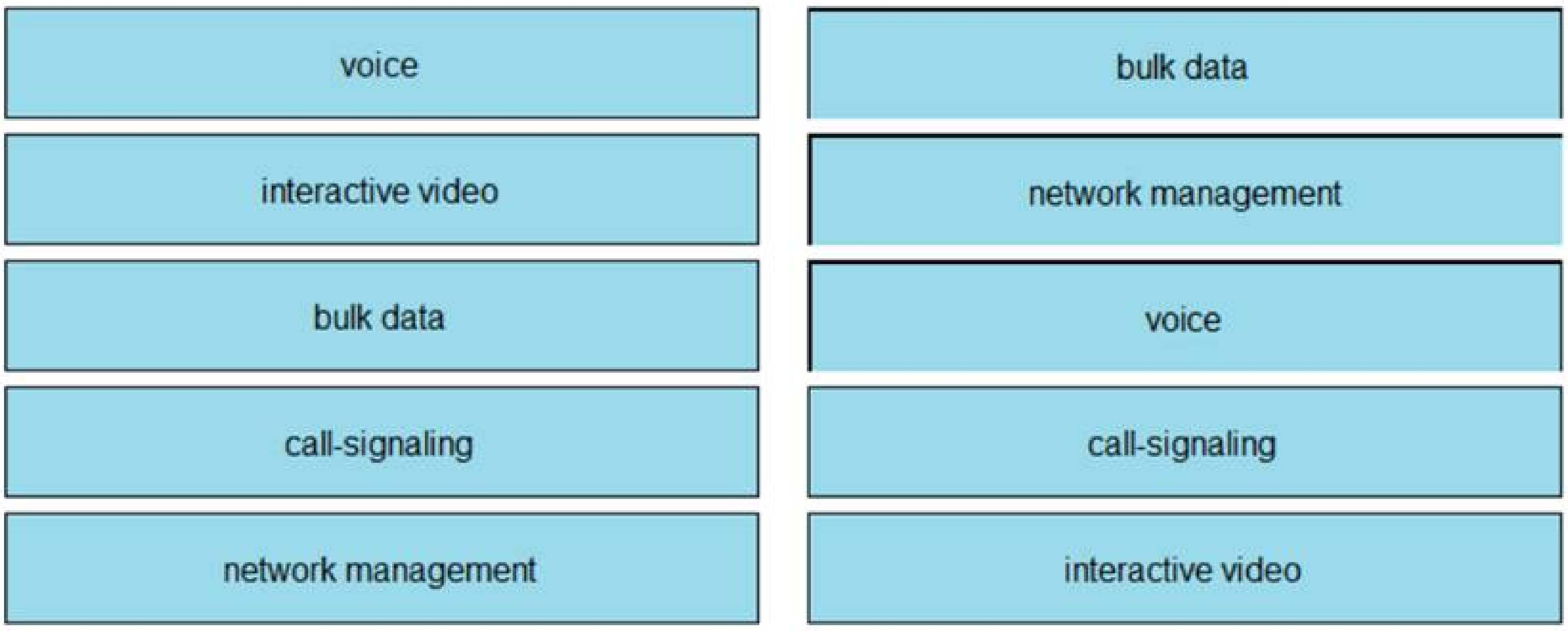
NEW QUESTION 52
DRAG DROP
According to the QoS Baseline Model, drag and drop the applications from the left onto the correct Per-Hop Behavior values on the right.



A. Mastered
B. Not Mastered

Answer: A

Explanation:



NEW QUESTION 57
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 62
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

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

Answer: A

NEW QUESTION 65
When a phone is registered over Mobile and Remote Access, where does it register?

A. Cisco Unified Presence Server
B. Expressway-E
C. Cisco Unified Communications Manager
D. Expressway-C

Answer: B

Explanation:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_b_system-configuration-guide-1201/cucm_b_system-configuration-guide-1201_chapter_01011010.html QUESTION

NEW QUESTION 69
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 72
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 76
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 77
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