Website: https://vceplus.com
VCE to PDF Converter: https://vceplus.com/vce-to-pdf/
Facebook: https://www.facebook.com/VCE.For.All.VN/
Twitter : https://twitter.com/VCE_Plus

350-801

Implementing Cisco Collaboration Core Technologies

Version 1.0

Exam A

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

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 2
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

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 3 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.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 4 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.

Correct Answer: AE
Section: (none)

Explanation

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

QUESTION 5 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

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 6 What is a characteristic of video traffic that governs QoS
requirements for video?

A. Video is typically variable bit rate.
B. Voice and video traffic are different, but they have the same QoS requirements.
C. Video is typically constant bit rate.
D. Voice and video traffic are the same, so they have the same QoS requirements.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/cac.html

QUESTION 7
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

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 8
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
Correct Answer: C

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

QUESTION 9
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

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 10 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

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
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.

QUESTION 11 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

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 12
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**

Correct Answer: C
Section: (none)
Explanation

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

QUESTION 13

```
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:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
```

Refer to the exhibit. 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.

Correct Answer: D
Section: (none)
Explanation
Explanation/Reference:

QUESTION 14 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.

Correct Answer: B
Section: (none)
Explanation

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

QUESTION 15
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.

Correct Answer: BC
Section: (none)
Explanation

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

QUESTION 16
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

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 17 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

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
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

QUESTION 18
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**

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:

QUESTION 19

```
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
```

Refer to the exhibit. 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.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
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

QUESTION 20

```
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
```

Refer to the exhibit. 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

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:

QUESTION 21

```
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
```

Refer to the exhibit. 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

Correct Answer: DE
Section: (none)
Explanation

Explanation/Reference:

QUESTION 22 An administrator recently upgraded a Cisco Webex DX80 through its web interface but discovered the next morning that the unit has received to its previous version. What must the administrator do to prevent this from happening again?

A. Assign a phone security profile with secure SIP.
B. Set the prepare cluster for rollback to pre-8-0 enterprise parameter to true.
C. Confirm the phone load name in the phone configuration.

D. Assign a universal device template to the phone.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 23
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

Correct Answer: C
Section: (none)
Explanation

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

QUESTION 24 Which description of the Mobile and Remote Access
feature is true?

A. Collaboration Edge feature that enables remote individuals to perform international calls from Jabber with a VPN connection.
B. Collaboration Edge feature that enables remote individuals to access all enterprise collaboration services using a PC within the corporate environment.C. Collaboration Edge feature that enables remote individuals to access enterprise collaboration services via Jabber without the use of a VPN connection.
D. Collaboration Edge feature that enables remote individuals to access enterprise collaboration services via Jabber with the use of a VPN connection.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 25

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

Refer to the exhibit. 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.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 26
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

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
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

QUESTION 27 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

Correct Answer: CE
Section: (none)
Explanation

Explanation/Reference:

QUESTION 28 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

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
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

QUESTION 29 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.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
QUESTION 30 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

Correct Answer: D
Section: (none)
Explanation

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

QUESTION 31 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.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 32
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

Correct Answer: C
Section: (none)
Explanation

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

QUESTION 33
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.

Correct Answer: C
Section: (none)
Explanation
Explanation/Reference:

QUESTION 34 Which protocols does Cisco IM and Presence use to
authenticate Jabber?

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

Correct Answer: AB
Section: (none)
Explanation

Explanation/Reference:

QUESTION 35
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.

Correct Answer: AD
Section: (none)
Explanation

Explanation/Reference:

QUESTION 36
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

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
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

QUESTION 37
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

Correct Answer: A

Section: (none)
Explanation

Explanation/Reference:

QUESTION 38 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)

Correct Answer: D
Section: (none)
Explanation

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

QUESTION 39 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

Correct Answer: C
Section: (none)
Explanation

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

QUESTION 40

```
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
```

Refer to the exhibit. 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.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:

QUESTION 41 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.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:

QUESTION 42

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

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 43 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

Correct Answer: C
Section: (none)
Explanation

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

QUESTION 44
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

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:

QUESTION 45
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**

Correct Answer: C
Section: (none)
Explanation

QUESTION 46
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
```
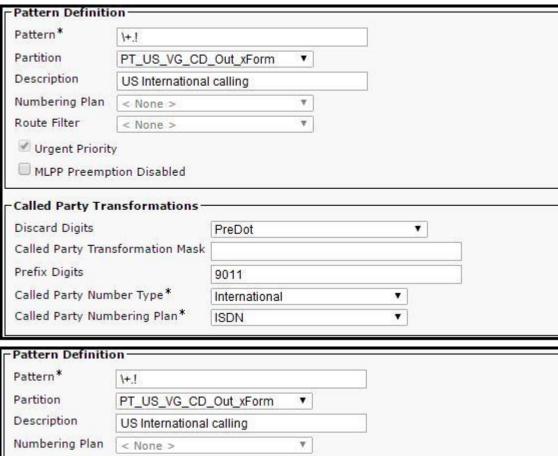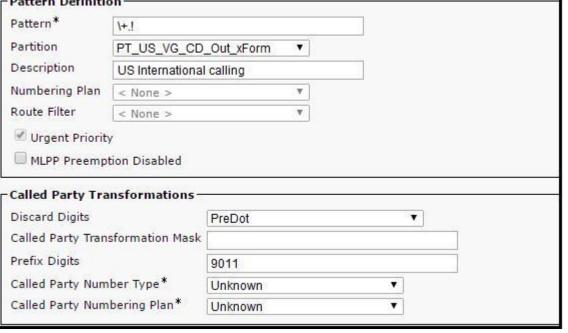
Pattern Definition

Pattern*	\+.!
Partition	PT_US_VG_CD_Out_xForm ▼
Description	US International calling
Numbering Plan	< None > ▼
Route Filter	< None > ▼

☑ Urgent Priority

☐ 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 ▼

Pattern Definition

Pattern*	\+.!
Partition	PT_US_VG_CD_Out_xForm ▼
Description	US International calling
Numbering Plan	< None > ▼
Route Filter	< None > ▼

☑ Urgent Priority

☐ MLPP Preemption Disabled

Called Party Transformations

Discard Digits	PreDot ▼
Called Party Transformation Mask	
Prefix Digits	9011
Called Party Number Type*	Unknown ▼
Called Party Numbering Plan*	Unknown ▼

A.

B.

Pattern Definition

Pattern*	\+.!
Partition	PT_US_VG_CD_Out_xForm ▼
Description	US International calling
Numbering Plan	< None > ▼
Route Filter	< None > ▼
☑ Urgent Priority	
☐ MLPP Preemption Disabled	

Called Party Transformations

Discard Digits	PreDot ▼
Called Party Transformation Mask	
Prefix Digits	9011
Called Party Number Type*	Cisco CallManager ▼
Called Party Numbering Plan*	Cisco CallManager ▼

Pattern Definition

Pattern*	\+.!
Partition	PT_US_VG_CD_Out_xForm ▼
Description	US International calling
Numbering Plan	< None > ▼
Route Filter	< None > ▼
☑ Urgent Priority	
☐ MLPP Preemption Disabled	

Called Party Transformations

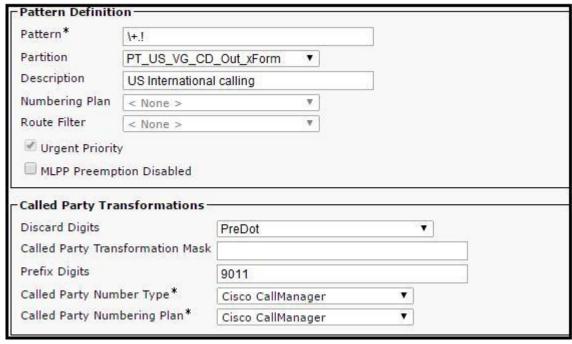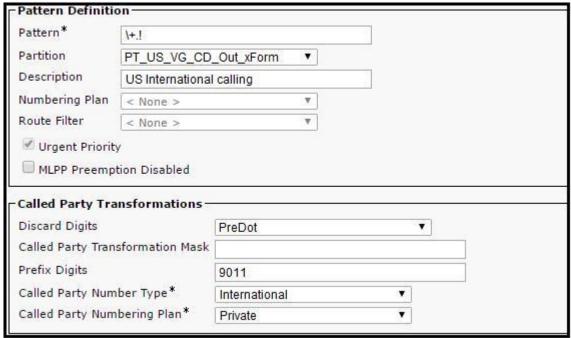|	
Discard Digits	PreDot ▼
Called Party Transformation Mask	
Prefix Digits	9011
Called Party Number Type*	International ▼
Called Party Numbering Plan*	Private ▼

C.

D.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 47 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.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:

QUESTION 48
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 callsC. create a second route pattern followed by the # wildcard
D. set up all international route patterns to 0.!

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 49

```
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
```

Refer to the exhibit 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.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:

QUESTION 50 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.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 51
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

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:

QUESTION 52 Which two types of device are supported by the Bulk Administration
Tool? (Choose two.)

A. H.322 clients
B. Cisco Unified IP phones (all models)
C. SIP trunks
D. H.225 trunks
E. music on hold servers

Correct Answer: AB
Section: (none)
Explanation

Explanation/Reference:

QUESTION 53
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 endpointsD. MGCP gateway registration
E. Secure business-to-business communications

Correct Answer: CE
Section: (none)
Explanation

Explanation/Reference:

QUESTION 54
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

Correct Answer: D

Section: (none)
Explanation

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

QUESTION 55
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 outage. Active/standby mode provides an always on standby node in the event of an outage, and it also provides load balancing.
B. Balanced mode provides user load balancing and user failover only for manually generated failovers. Active/standby mode provides an unconfigured standby node in the event of an outage, but it does not provide load balancing.
C. Balanced mode provides user load balancing and user failover in the event of an outage. Active/standby mode provides an always on standby node in the event of an outage, but it does not provide load balancing.
D. Balanced mode does not provide user load balancing, but it provides in the event of an outage. Active/standby mode provides an always on standby node in the event of an outage, but it does not provide load balancing.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 56
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

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
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

QUESTION 57

```
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
```

Refer to the exhibit. 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.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:

QUESTION 58

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 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 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**

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:

QUESTION 59 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 terminal**
   Router(config)# interface gigabitethernet 5/1
   Router(config-if)# platform qos trust extend
B. **Router# configure terminal**
   Router(config)# interface gigabitethernet 5/1
   Router(config-if)# platform qos trust extend cos 3
C. **Router# configure terminal**
   Router(config)# interface gigabitethernet 5/1
   Router(config-if)# platform qos trust extend cos 5
D. **Router# configure terminal**
   Router(config)# interface gigabitethernet 5/1
   Router(config-if)# platform qos extend trust

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 60
A customer is deploying a SIP IOS gateway for a customer who requires that in-band DTMF relay is first priority and out-of-band DTMF relay is second priority.

Which IOS entry sets the required priority?

A. **dtmf-relay rtp-nte sip-notify** B.

dtmf-relay cisco-rtp

C. **sip-notify dtmf-relay rtp-nte**

D. **dtmf-relay sip-kpml cisco-rtp**

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 61 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

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:

QUESTION 62
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-hook. Unified CM analyzes all digits as a string.
D. On-hook, no digit analysis is performed, off-hook. Unified CM requires the "#* to start the digit analysis.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 63 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

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:

QUESTION 64
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

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 65 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**
Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 66 You are adding regions in Cisco Unified Communications Manager. Which codec(s) are selected when a call is placed if you set up the max audio bit rate to use 8 kbps?

A. G.729
B. G 729 and G.711ulaw
C. G.711ulaw and G 711alaw
D. G.722

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 67
An incoming off-net call to a user fails. An engineer notices that the off-net call is G.711, but the phone accepts only G.729. Which media resource on a Cisco Unified Border Element and Cisco Unified Communications Manager must the engineer configure to manage the codec negotiation?

A. transcoder
B. CFB
C. MOH

D. MTP

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 68
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

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
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

QUESTION 69 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

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:

QUESTION 70 Which transport protocol does the application layer
protocol SNMP use?

A. XML
B. UDP
C. SIP
D. HTTP

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Reference: https://www.geeksforgeeks.org/simple-network-management-protocol-snmp/

QUESTION 71 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

Correct Answer: AD
Section: (none)
Explanation

Explanation/Reference:
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

QUESTION 72
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

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
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

QUESTION 73
A network administrator deleted a user from the LDAP directory of a company. The end user shows as Inactive LOAP Synchronized User in Cisco Unified Communications Manager. Which step is next to remove this user from Cisco Unified Communications Manager?

A. Delete the user directly from Cisco Unified Communications Manager
B. Restart the Dirsync service after the user is deleted from LDAP directory.
C. Execute a manual sync to refresh the local database and delete the end user.
D. Wait 24 hours for the garbage collector to remove the user.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:

QUESTION 74
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.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 75

```
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
```

Refer to the exhibit. 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.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:

QUESTION 76 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

Correct Answer: B
Section: (none)
Explanation

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

QUESTION 77 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

Correct Answer: D
Section: (none)
Explanation

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

QUESTION 78 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

Correct Answer: D
Section: (none)
Explanation

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

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

Correct Answer: AB
Section: (none)
Explanation

Explanation/Reference:
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

QUESTION 80
An engineer is designing a high availability and failover solution for two Cisco Unified Border Element routers. The first router (cube1.abc.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

Correct Answer: CE
Section: (none)
Explanation

Explanation/Reference:

QUESTION 81
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

Correct Answer: BE
Section: (none)
Explanation

Explanation/Reference:

QUESTION 82
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.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
QUESTION 83 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".

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
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

QUESTION 84 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.

Correct Answer: C
Section: (none)
Explanation

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

QUESTION 85 Why would we not include an end user's PC device in a QoS
trust boundary?

A. The end user could incorrectly tag their traffic to bypass firewalls.
B. The end user may incorrectly tag their traffic to be prioritized over other network traffic.
C. There is no reason not to include an end user's PC device in a QoS trust boundary.
D. The end user could incorrectly tag their traffic to advertise their PC as a default gateway.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:

QUESTION 86
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**

Correct Answer: D
Section: (none)
Explanation

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

QUESTION 87
How can an administrator stop Cisco Unified Communications Manager from advertising the OPUS codec for recording enabled devices?

A. Route recorded calls through Cisco Unified Border Element because it does not support OPUS.
B. Go to the phone's configuration page and set "Advertise OPUS Codec" to be "false".
C. Integrate the Cisco Unified CM with 3 recording solution that does not support OPUS.
D. In CUCM Service Parameters set "Opus Codec Enabled" to "Enabled for all Devices Except Recording-Enabled Devices."

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/211297-Configure-Opus-Support-on-Cisco-Unified.pdf

QUESTION 88 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

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 89 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

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 90
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.

Correct Answer: BE
Section: (none)
Explanation

Explanation/Reference:

QUESTION 91
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.

Select and Place:

Correct Answer:

Section: (none)
Explanation

Explanation/Reference:

QUESTION 92
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.

Correct Answer: B

Section: (none)
Explanation

Explanation/Reference:

QUESTION 93 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.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 94
Which packet delay is the maximum supported between Cisco Unified Communications Manager nodes for clustering over WAN deployments?

A. 150 ms round tip
B. 510 ms round trip
C. 40 ms round trip
D. 80 ms round trip

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/callpros.html

QUESTION 95
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

in the Calling Party Transformation Patterns section,
    configure the Pattern as 9.011841234567
    configure the Discard Digits as Predot 10-10-Dialing

in the Called Party Transformation Pattern Configuration section,
    configure the Pattern as 9.011841234587
    configure the Discard Digits as Predot 10-10-Dialing

in the Calling Party Transformation Patterns section,
    configure the Pattern as a 011841234557
    configure the Discard Digits as Predot

B.

C.

D.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

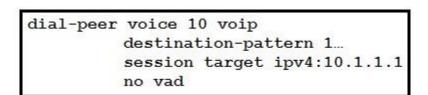**QUESTION 96** When a phone is registered over Mobile and Remote Access, where
does it register?

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

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
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**

97

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

Refer to the exhibit. 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

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
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

QUESTION 98
When a new SIP phone Is registered to Cisco Unified Communications Manager, it keeps failing and showing an "unprovisioned" error message in the phone display. Which problem is a possible cause of this issue?

A. Auto-registration is disabled on the Cisco Unified Communications Manager nodes and the phone device does not have a DN configured.
B. The DN assigned to the phone is already in use by another SIP phone.
C. The phone cannot download and install the latest firmware.
D. The DHCP settings are set incorrectly and the phone does not have an alternate TFTP defined.
E. The DN configuration for this phone is shared with an SCCP phone. which is not supported.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 99
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**

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:

QUESTION 100 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.

Correct Answer: A
Section: (none)
Explanation

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

QUESTION 101
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 aver 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.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
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

QUESTION 102 Which wildcard must an engineer configure to match a whole domain in
SIP route patterns?

A. .
B. !

C. @

D. *

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
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.html