Vendor: Cisco

Exam Code: 350-801

Exam Name: Implementing and Operating Cisco Collaboration Core Technologies (CLCOR)

Version: DEMO

QUESTION 1
What is the validity period of the ITL Recovery certificate in Cisco UCM?

A. 1 year
B. 20 years
C. 5 years
D. 10 years

Answer: B
Explanation:
The validity of ITLRecovery has been extended from 5 years to 20 years to ensure that the ITLRecovery certificate remains same for a longer period
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/security/12_0_1/secugd/cucm_b_cucm-security-guide-1201/cucm_b_cucm-security- guide-1201_chapter_011.html

QUESTION 2
A remote office has a less-than-optimal WAN connection and experiences packet loss, delay and jitter. Which VoIP codec is used in this situation?

A. G722.1
B. iLBC
C. G.711alaw
D. G.729A

Answer: B
Explanation:
iLBC: iLBC provides audio quality between that of G.711 and G.729 at bit rates of 15.2 kbps (38-bytes or 20msec) and 13.3 kbps (50 bytes or 30 msec). iLBC handles lossy networks in better way than G729 because it treats each packet independently. G729 depends on the previous packet to handle packet loss, jitter and delay which doesn't tolerate well in lossy networks.

QUESTION 3
Which Cisco Unified Communications Manager service parameter should be enabled to disconnect a multiparty call when the call initiator hangs up?

A. Block OffNet to OffNet Transfer
B. Drop Ad Hoc Conference
C. H.225 Block Setup destination
D. Enterprise Feature Access code for conference

Answer: B
Explanation:
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 4
Which Cisco UCM configuration is required for SIP MWI integrations?

A. Enable "Accept presence subscription" on the SIP Trunk Security Profile.

B. Select "Redirecting Diversion Header Delivery - Outbound" on the SIP trunk.
C. Enable "Accept unsolicited notification" on the SIP Trunk Security Profile.
D. Select "Redirecting Diversion Header Delivery - Inbound" on the SIP trunk.

Answer: C
Explanation:
In case of SIP integration, CUC sends SIP notify messages to the phone system to turn the MWI on or OFF so “Accept unsolicited notification” should be checked under SIP security profile assigned to the trunk.
https://community.cisco.com/t5/collaboration-voice-and-video/understanding-troubleshooting-mwi-on-unity-connection/ta-p/3162948

QUESTION 5
What is a description of the DiffServ model used for implementing QoS?

A. AF41 has higher drop precedence than AF42, which has higher drop precedence than AF43.
B. Voice and video calls are marked with different DSCP values and placed in different queues.
C. AF43 has higher drop precedence than AF42 but lower drop precedence than AF41.
D. RTP traffic from voice and video calls is marked EF and placed in the same queue.

Answer: B
Explanation:

Assured Forwarding

RFC 2597 icon_popup_short.gif defines the assured forwarding (AF) PHB and describes it as a means for a provider DS domain to offer different levels of forwarding assurances for IP packets received from a customer DS domain. The Assured Forwarding PHB guarantees a certain amount of bandwidth to an AF class and allows access to extra bandwidth, if available. There are four AF classes, AF1x through AF4x. Within each class, there are three drop probabilities. Depending on a given network's policy, packets can be selected for a PHB based on required throughput, delay, jitter, loss or according to priority of access to network services.

Classes 1 to 4 are referred to as AF classes. The following table illustrates the DSCP coding for specifying the AF class with the probability. Bits DS5, DS4 and DS3 define the class; bits DS2 and DS1 specify the drop probability; bit DS0 is always zero.

Drop	Class 1	Class 2	Class 3	Class 4
Low	001010 AF11 DSCP 10	010010 AF21 DSCP 18	011010 AF31 DSCP 26	100010 AF41 DSCP 34
Medium	001100 AF12 DSCP 12	010100 AF 22 DSCP 20	011100 AF32 DSCP 28	100100 AF42 DSCP 36
High	001110 AF13 DSCP 14	010110 AF23 DSCP 22	011110 AF33 DSCP 30	100110 AF43 DSCP 38

https://www.cisco.com/c/en/us/support/docs/quality-of-service-qos/qos-packet-marking/10103-dscpvalues.html

QUESTION 6
Which external DNS SRV record must be present for Mobile and Remote Access?

A. _cisco-uds._tcp.example.com
B. _collab-edge._tls.example.com
C. _collab-edge._tcp.example.com
D. _cisco-uds._tls example.com

Answer: B
Explanation:
Cisco Expressway supports Mobile and Remote Access with multiple external domains. With this deployment, you will have more than one external domain where your MRA clients may reside. Expressway-E must be able to connect to all of them. To configure this deployment, do the following:

For Expressway-E:
On Expressway-E, configure _collab-edge._tls.<domain> and _sips_tcp.<domain>DNS SRV records for each Edge domain.
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/expressway/config_guide/X14-0-1/mra/exwy_b_mra-deployment-guide-x1401/exwy_m_requirements-for-mra.html

QUESTION 7
Refer to the exhibit. A collaboration engineer adds an analog gateway to a Cisco UCM cluster. The engineer chooses MGCP over SCCP as the gateway protocol.
Which two actions ensure that the gateway registers? (Choose two.)

```
Gateway1#show sccp
SCCP Admin State: UP
Gateway Local Interface: Loopback0
        IPv4 Address: 192.168.12.1
        Port Number: 2000

Gateway1#
Gateway1#show ccm-manager
% Call Manager Application is not enabled
Gateway1#

Gateway1#show mgcp
MGCP Admin State DOWN, Oper State DOWN - Cause Code NONE
MGCP call-agent: none Initial protocol service is MGCP 0.1
MGCP validate call-agent source-ipaddr DISABLED
MGCP validate domain name DISABLED
MGCP block-newcalls DISABLED
MGCP send SGCP RSIP: forced/restart/graceful/disconnected DISABLED
```

A. Delete and re-add the gateway configuration in Cisco UCM.
B. Enter "mgcp" on the gateway in configuration mode.
C. Enter "no sccp" on the gateway in configuration mode.
D. Enter "ccm-manager config" on the gateway in configuration mode.
E. Enter "ccm-manager mgcp" on the gateway in configuration mode.

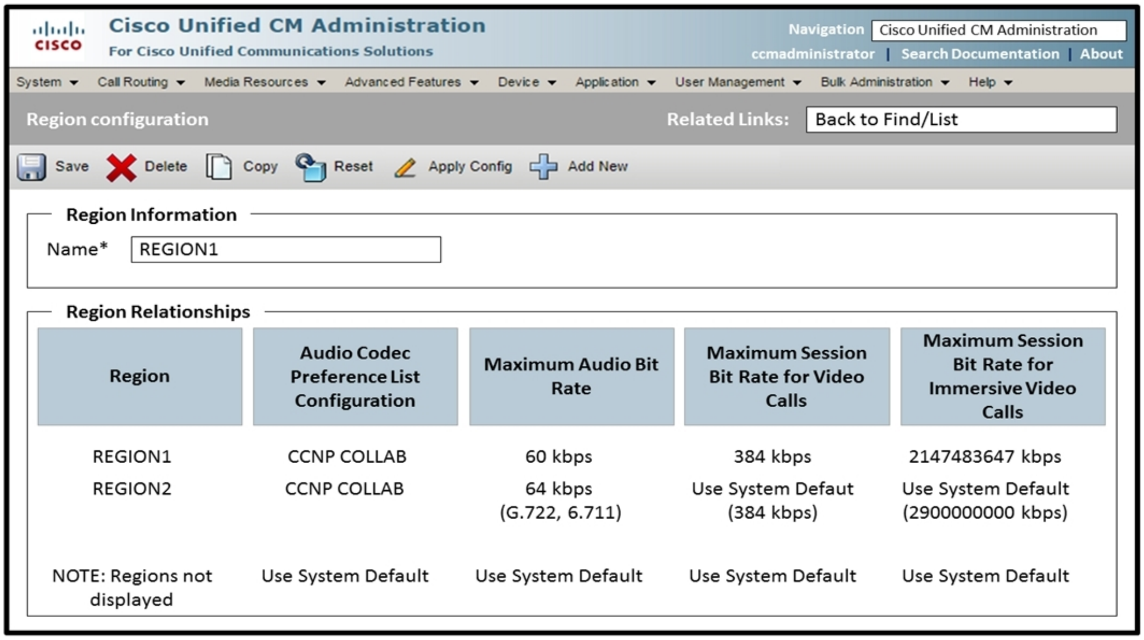
Answer: BE
Explanation:
To enable the gateway to communicate with Cisco CallManager through the Media Gateway Control Protocol (MGCP) and to supply redundant control agent services, use the "ccm-manager mgcp" command in global configuration mode.
http://www.cisco.com/en/US/docs/ios/12_3t/voice/command/reference/vrht_c4_ps5207_TSD_Products_Command_Reference_Chapter.html#wp1072910
Do a mgcp / no mgcp once its added. Make sure that the domain name on cucm is the same as it appears in the 'show ccm-manager' output on the gateway.

QUESTION 8
Refer to the exhibit. An engineer is troubleshooting this video conference issue:

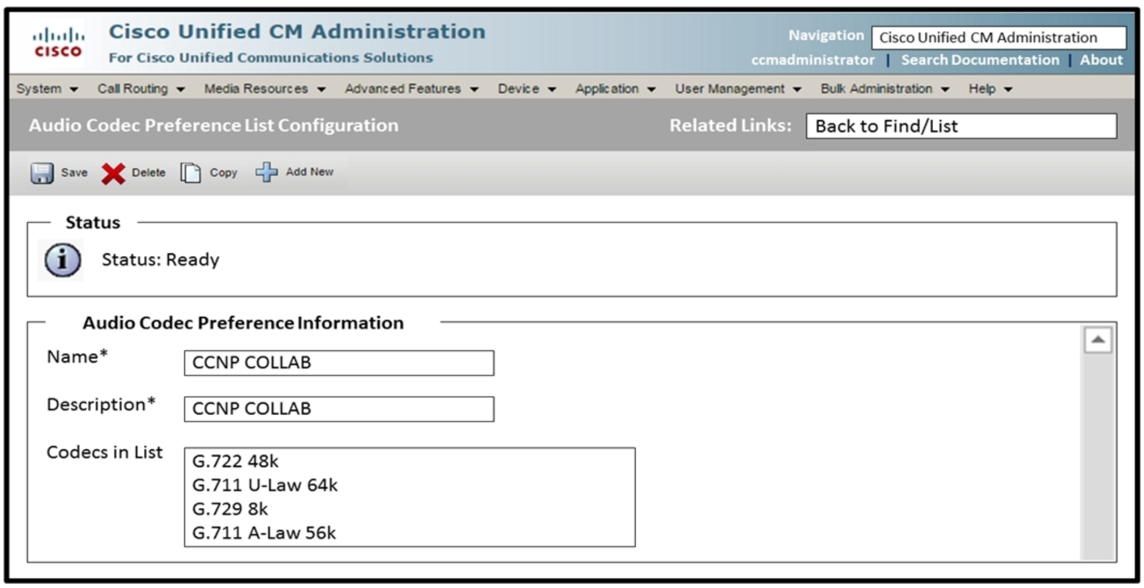
- A video call between a Cisco 9971 in Region1 and another Cisco 9971 in Region1 works.

- As soon as the Cisco 9971 in Region1 conferences in a Cisco 8945 in Region2, the Region1 endpoint cannot see the Region2 endpoint video.

What is the cause of this issue?



Region	Audio Codec Preference List Configuration	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
REGION1	CCNP COLLAB	60 kbps	384 kbps	2147483647 kbps
REGION2	CCNP COLLAB	64 kbps (G.722, 6.711)	Use System Defaut (384 kbps)	Use System Default (2900000000 kbps)
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

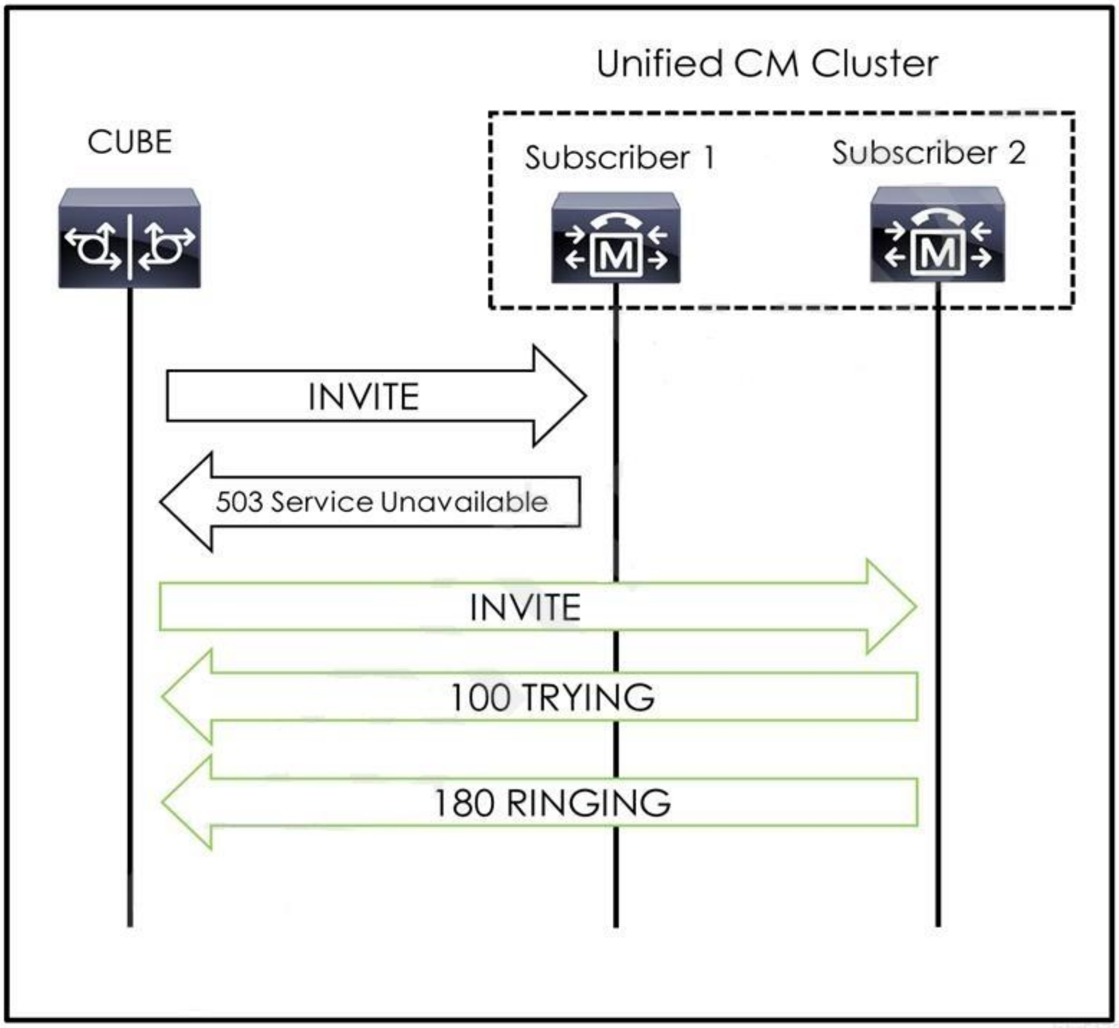


A. Cisco 8945 does not have a camera connected.
B. Maximum Audio Bit Rate must be increased.
C. Maximum Session Bit Rate for Video Calls is too low.
D. Maximum Session Bit Rate for Immersive Video Calls is too low.

Answer: C
Explanation:

The section in the question of "when the 9971 CONFERENCES IN the 8945..." indicating that the 8945 is being added to an already in-progress video call, totaling three or more video streams. This would mean 384kbps is definitely way too low.

QUESTION 9
Refer to the exhibit. Cisco Unified Border Element is attempting to establish a call with Subscriber 1, but the call fails. Cisco Unified Border Element then retries the same call with Subscriber 2, and the call proceeds normally. Which action resolves the issue?



A. Verify that the Run On All Active Unified CM Nodes checkbox is enabled.
B. Verify that the correct calling search space is selected for the Inbound Calls section.
C. Verify that the Significant Digits field for Inbound Calls is set to All.
D. Verify that the PSTN Access checkbox is enabled.

Answer: A
Explanation:
Run On All Unified CM Nodes allows Subscribers NOT configured within the Call Manager Group that is applied to Route Lists and/or SIP Trunks to process calls.

There is a whole rabbit whole with how this feature works when only applied to one or the other but per best practice, unless there is a VERY specific reason, you want this check box enabled on ALL Route Lists and SIP Trunks.

QUESTION 10
What is a characteristic of a SIP endpoint configured in Cisco UCM with "Use Trusted Relay Point" set to "On"?

A. It enables Cisco UCM to insert an MTP or transcoder designated as a TRP.
B. It enables the Use Trusted Relay Point setting from the associated common device configuration.
C. If TRP is allocated and MTP is also required for the endpoint, calls fail.
D. It creates a trust relationship with the called party.

Answer: A
Explanation:
A Trusted Relay Point (TRP) is an MTP or transcoder that Cisco Unified Communications Manager can insert into the media stream to act as a control point for call media. The TRP can provide further processing on the stream and can ensure that the stream follows a specific path.

QUESTION 11
An administrator works with an ISDN PRI that is connected to a third-party PBX.
The ISDN link does not come up; and the administrator finds that the third-party PBX uses the QSIG signalling method.
Which command enables the Cisco IOS Gateway to use QSIG signalling on the ISDN link?

A. isdn switch-type basic-qsig
B. isdn switch-type basic-ni
C. isdn switch-type primary-qsig
D. isdn incoming-voice voice

Answer: C
Explanation:
The switch type confiugred must be QSIG:
```
isdn switch-type primary-qsig
```

https://community.cisco.com/t5/collaboration-knowledge-base/qsig/ta-p/3126953

QUESTION 12
A collaboration engineer troubleshoots issues with a Cisco IP Phone 7800 Series. The IPv4 address of the phone is reachable via ICMP and HTTP and the phone is registered to Cisco UCM. However, the engineer cannot reach the CLI of the phone.
Which two actions in Cisco UCM resolve the issue? (Choose two)

A. Enable FIPS Mode under Product Specific Configuration Layout in Cisco UCM
B. Enable Settings Access under Product Specific Configuration Layout in Cisco UCM.
C. Enable SSH Access under Product Specific Configuration Layout in Cisco UCM
D. Disable Web Access under Product Specific Configuration Layout in Cisco UCM
E. Set a username and password under Secure Shell Information in Cisco UCM

Answer: CE

Explanation:
Enable ssh for phone, and then setup username and password so that engineer can login to phone via CLI.
https://www.cisco.com/c/en/us/support/docs/collaboration-endpoints/ip-phone-7800-series/200850-Troubleshoot-Cisco-Phone-7800-8800-Serie.pdf

QUESTION 13
How many minutes does it take for automatic fallback to occur in a Presence Redundancy Group if the primary node lost a critical service?

A. 5 min
B. 10 min
C. 30 min
D. 60 min

Answer: C
Explanation:
Enable Automatic Fallbackג€"This parameter specifies whether to do automatic fallback. In the event of a failover, the IM and Presence Service moves users automatically from the backup node to the primary node thirty minutes after the primary node returns to a healthy state.
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/11_5_1/cup0_b_config-and-admin-guide-1151su5/ cup0_b_imp-system-configuration-1151su5_chapter_0100.html

QUESTION 14
A company hosts a conference call with no local users.
How does the administrator stop the conference from continuing?

A. modifies the Block OffNet to OffNet Transfer service parameter
B. removes the transcoder
C. changes the codecs that are supported on the conference resource
D. modifies the Drop Ad Hoc Conference service parameter

Answer: D
Explanation:
Drop Ad Hoc Conference: This parameter determines how an ad hoc conference terminates. This is an important toll-fraud prevention setting, because inside facilitators can set up a conference call to expensive international numbers and then drop out of the call. Without the conference controller, international tariffs are billed back to the company in which the conference call was set up. Valid values are as follows:
- Never (default): The conference remains active after the conference controller and all on-net parties hang up. This default setting could result in potential toll fraud.
- When Conference Controller Leaves: Terminate the conference when the conference controller hangs up.
- When No On-Net Parties Remain in the Conference: Terminate the conference when there are no on-net parties remaining in the conference. This distinction is important because the conference controller might have to drop out of the call, but other business partners on the call should continue the conference. The When Conference Controller Leaves option would hang up the call when the conference controller left the conference.

QUESTION 15

Refer to the exhibit. What is the registration state of the analog port in this debug output?

```
000142: *Apr 23 19:41:49.050: MGCP Packet received from 192.168.100.100:2427--->
AUEP 4 AALN/S0/SU0/0@VG320.cisco.local MGCP 0.1
F: X, A, I
<---

000143: *Apr 23 19:41:49.050: MGCP Packet sent to 192.168.100.101:2427--->
200 4
I:
X: 2
L: p:10-20, a:PCMU;PCMA;G.nX64, b:64, e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
L: p:10-220, a:G.729;G.729a;G.729b, b:8. e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
L: p:10-110, a:G.726-16;G.728; b:16, e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
L: p:10-70, a:G.726-24; b:24, e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
L: p:10-50, a:G.726-32; b:32, e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
L: p:30-270, a:G.723.1-H;G.723;G.723.1a-H, b:6. e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
L: p:30-330, a:G.723.1-L;G.723.1a-L, b:5, e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
M: sendonly, recvonly, sendrecv, inactive, loopback, conttest, data, netwloop, netwtest
<---
```

A. The MGCP Gateway is not communicating with the Cisco UCM.
B. The analog port failed to register to Cisco UCM with an error code 200.
C. The analog port is currently shut down.
D. The analog port is registered to Cisco UCM.

Answer: D
Explanation:
CUCM sends an AUEP (Audit Endpoint) to the Gateway to determine the status of the given Endpoint. The response from the Gateway is an ACK with the endpoints capabilities. Once this is complete the Endpoint is registered with the CUCM.
https://www.cisco.com/c/en/us/support/docs/voice/media-gateway-control-protocol-mgcp/214635-configure-and-troubleshoot-mgcp-gateways.html

QUESTION 16

Which two actions must be taken to provision a new device using self-provisioning?

A. Enable the self-provisioning IVR in the Cisco UCM
B. Import the user profile to the corporate LDAP directory.
C. Link the appropriate service profile to the provisioning template.
D. Link the appropriate universal device template to the user profile.
E. Ensure the user has a directory URI and a primary extension.

Answer: AD
Explanation:
Self-Provisioning of a phone to a user needs:

* an End user with a primary extension and a Feature Group Template
* the Feature Group Template needs a User Profile
* the User Profile needs a Universal Line and Universal Device Template

Finally to activate it you need a CTI route point and a configured IVR service to actually register it. HOWEVER, it looks like you dont actually NEED the IVR to technically fullfil the goal of self-registration, as users can use the URL based registration. but i am not 100% sure if you wont need the IVR anyways, even if you arent using it - the guide is not 100% clear on that

https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/214228-configure-self-provisioning-feature-on-c.html

QUESTION 17
A collaboration engineer must optimize the dial plan within Cisco UCM. There are multiple remote sites. And each site has its own route patterns and local gateways.
What should the engineer do on the cisco UCM to optimize the dial plan?

A. Configure a Standard Local Route Group to use a single route pattern for all calls within the cluster.
B. Create a centralized dial plan with a Cisco UCM Session Management Edition cluster or a Cisco gatekeeper.
C. Implement ILS with GDPR so the dial plan can dynamically replicate across clusters.
D. Leverage Cisco UCM Express as SRST so the phones can have more features of each remote site

Answer: A
Explanation:
The Local Route Group feature helps reduce the complexity and maintenance efforts of provisioning in a centralized Cisco Unified Communications Manager deployment that uses a large number of locations.

QUESTION 18
In the cisco expressway solution, which two features does mobile and Remote access provide? (Choose two)

A. VPN-based enterprise access for a subset of Cisco Unified IP Phone models
B. Secure reverse proxy firewall travelsal connectivity
C. the ability to register third-party SIP or H.323 devices on Cisco UCM without requiring VPN
D. the ability of Cisco IP Phones to access the enterprise through VPN connection
E. the ability for remote users and their devices to access and consume enterprise collaboration applications and services

Answer: BE
Explanation:
Third-party SIP or H.323 devices can register to the Expressway-C and, if necessary, interoperate with Unified CM-registered devices over a SIP trunk. Not CUCM.
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/expressway/config_guide/X12-5/exwy_b_mra-expressway-deployment-guide/exwy_b_mra-expressway-deployment-guide_chapter_00.html

QUESTION 19
Which DTMF relay method configured on a SIP dial-peer will ensure that a media resource is not invoked by Unified CM for calls to UCCX IVRs?

A. dtmf-relay slp-kpml
B. dtmf-relay rtp-nte
C. sdtmf-relay h245-slgnal
D. dtmf-relay sip-notify

Answer: A
Explanation:

UCCX only supports out-of-band DTMF. This can be applied at the CUBE to avoid a media resource being needed to address as it can directly handle the DTMF relay from RTP-NTE to SIP-KPML
https://www.cisco.com/c/dam/en/us/td/docs/ios-xml/ios/voice/cube_uccx/cube_uccx_interop_bestprac_guide.pdf

QUESTION 20
Refer to the exhibit. A Cisco UCM user with directory number 4401 dials 5507 and the call is routed to a Cisco Unified Border Element. Which IP address will the call be sent to?

```
voice class dpg 2000
 dial-peer 2001 preference 1
 dial-peer 2002 preference 2
 dial-peer 2003 preference 3

dial-peer voice 1001 voip
 description INBOUND
 session protocol sipv2
 session target ipv4:10.0.0.1
 destination dpg 2000
 incoming called-number 5T

dial-peer voice 2001 voip
 destination-pattern 5506
 session protocol sipv2
 session target ipv4:10.0.0.2

dial-peer voice 2002 voip
 destination-pattern 55..
 session protocol sipv2
 session target ipv4:10.0.0.3

dial-peer voice 2003 voip
 destination-pattern 5507
 session protocol sipv2
 session target ipv4:10.0.0.4
```

A. 10.0.0.3
B. 10.0.0.1
C. 10.0.0.2
D. 10.0.0.4

Answer: A
Explanation:
Once an incoming call is matched by an inbound dial peer with an active destination dial-peer group, dial peers from this group are used to route the incoming call.
No other outbound dial-peer provisioning to select outbound dial peers is used.
A preference can be defined for each dial peer in a dial-peer group.

QUESTION 11
What is the maximum DNS SRV entries that should be defined in the SIP Trunk destination address field in Cisco UCM?

A. 4
B. 8

C. 1
D. 16

Answer: C
Explanation:
"You can assign up to 16 different destination addresses for a SIP trunk, using IPv4 or IPv6 addressing, fully qualified domain names, or you can use a single DNS SRV record."
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-guide-1151_chapter_01110.html

QUESTION 22
Which type of greeting in the Call Handler configuration in Cisco Unity Connection overrides all other greetings?

A. holidays
B. supervisory
C. alternate
D. priority

Answer: C
Explanation:
Alternate Can be used for a variety of special situations, such as vacations or a leave. An alternate greeting overrides all other greetings.
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/10x/administration/guide/10xcucsagx/10xcucsag080.html

QUESTION 23
A Cisco IP Phone 7841 that is registered to a Cisco Unified Communications Manager with default configuration receives a call setup message.
Which codec is negotiated when the SDP offer includes this line of text?

```
M=audio 498181 RTP/AVP 0 8 97
```

A. G.711ulaw
B. iLBC
C. G.711alaw
D. G.722

Answer: A
Explanation:
Those are the codecs.
They are RTP payload types, and in preference order.
Eg 8=g711alaw, 0=g711ulaw, 18=g729, 97=this one is special, it's actually what they want to Mark RTP-NTP (RFC2833) DTMF relay as.
Reference:
https://community.cisco.com/t5/ip-telephony-and-phones/sip-sdp-no-codec-specifed/td-p/3913990
https://en.m.wikipedia.org/wiki/RTP_payload_formats