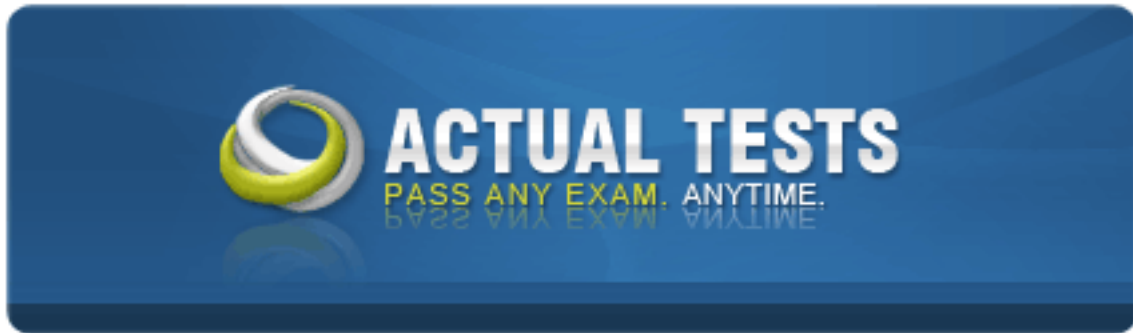


Cisco 642-457



**Implementing Cisco Unified Communications
Manager, Part 2 v8.0 (CIPT2 v8.0)**

Version: 14.0

QUESTION NO: 1

Which method can be used to address variable-length dial plans?

- A. Overlap sending and receiving.
- B. Add a prefix for all calls that are longer than 10-digits long
- C. Use nested translation patterns to eliminate inter-digit timeout
- D. Use the @macro on the route pattern
- E. Use MGCP gateways, which support variable-length dial plans

Answer: A

Explanation: Incorrect answer: BCDE

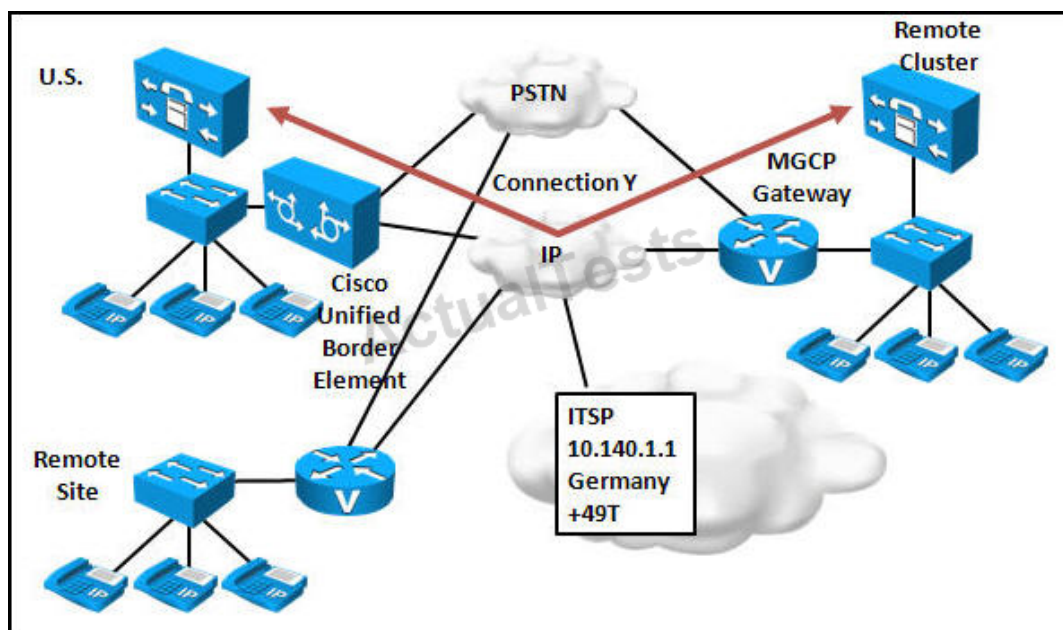
If the dial plan contains overlapping patterns, Cisco Unified Communications Manager does not route the call until the interdigit timer expires (even if it is possible to dial a sequence of digits to choose a current match). Check this check box to interrupt interdigit timing when Cisco Unified Communications Manager must route a call immediately.

By default, the Urgent Priority check box displays as checked. Unless your dial plan contains overlapping patterns or variable length patterns that contain!, Cisco recommends that you do not uncheck the check box.

Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fsintrcm.html

QUESTION NO: 2

Refer to the exhibit.



Which trunks would be most suitable for Connection Y?

- A. an H.225 trunk (gatekeeper-controlled)
- B. intercluster trunk (gatekeeper-controlled)
- C. a SIP trunk on the U.S. cluster and an intercluster trunk on the remote cluster
- D. intercluster trunk (nongatekeeper-controlled)
- E. No extra connections are required. As long as IP connectivity exists, you need only configure a route pattern for each site. The Cisco Unified Communications Manager will automatically forward the calls over the WAN if the destination directory number is not registered locally.

Answer: D

Explanation:

QUESTION NO: 3

Which two features require or may require configuring a SIP trunk? (Choose two.)

- A. SIP gateway
- B. Call Control Discovery between a Cisco Unified Communications Manager and Cisco Unified Communications Manager Express
- C. Cisco Device Mobility
- D. Cisco Unified Mobility
- E. registering a SIP phone

Answer: A,B

Explanation: Incorrect answer: CDE

All protocols require that either a signaling interface (trunk) or a gateway be created to accept and originate calls. Device mobility allows Cisco Unified Communications Manager to determine whether the phone is at its home location or at a roaming location. Cisco Unified Mobility gives users the ability to redirect incoming IP calls from Cisco Unified Communications Manager to different designated phones, such as cellular phones.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a08sip.html#wpixref77849

QUESTION NO: 4

A Cisco 3825 needs to be added in Cisco Unified Communications Manager as an H.323 gateway. What should the gateway type be?

- A. H.323 gateway
- B. Cisco 3825
- C. Cisco 3800 series router. The specific model will be selected after the Cisco 3800 is selected.
- D. The gateway can be added either as an H.323 gateway or Cisco 3800 series router.
- E. The gateway can be added either as an H.323 gateway or Cisco 3825 series router.

Answer: A

Explanation:

QUESTION NO: 5

Which statement best describes globalized call routing in Cisco Unified Communications Manager?

- A. All incoming calling numbers on the phones are displayed as an E.164 with the + prefix.
- B. Call routing is based on numbers represented as an E.164 with the + prefix format.
- C. All called numbers sent out to the PSTN are in E.164 with the + prefix format.
- D. The CSS of all phones contain partitions assigned to route patterns that are in global format.
- E. All phone directory numbers are configured as an E.164 with the + prefix.

Answer: B

Explanation: Incorrect answer: ACDE

For the destination to be represented in a global form common to all cases, we must adopt a global form of the destination number from which all local forms can be derived. The + sign is the mechanism used by the ITU's E.164 recommendation to represent any PSTN number in a global, unique way. This form is sometimes referred to as a fully qualified PSTN number.

Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/7x/dialplan.html#wp1153205

QUESTION NO: 6

When an incoming PSTN call arrives at an MGCP gateway, how does the calling number get normalized to a global E.164 number with the + prefix in Cisco Unified Communications Manager?

- A. Normalization is done using translation patterns.
- B. Normalization is done using route patterns.
- C. Normalization is done using the gateway incoming called party prefixes based on number type.
- D. Normalization is done using the gateway incoming calling party prefixes based on number type.
- E. Normalization is achieved by local route group that is assigned to the MGCP gateway.

Answer: D

Explanation: Incorrect answer: ABCE

Configuring calling party normalization alleviates issues with toll bypass where the call is routed to multiple locations over the IP WAN. In addition, it allows Cisco Unified Communications Manager to distinguish the origin of the call to globalize or localize the calling party number for the phone user.

Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fscallpn.html

QUESTION NO: 7

When an incoming PSTN call arrives at an MGCP gateway, how does the called number get normalized to an internal directory number in Cisco Unified Communications Manager?

- A. Normalization is done by configuring the significant digits for inbound calls on the MGCP gateway.
- B. Normalization is done using route patterns.
- C. Normalization is done using the gateway incoming called party prefixes based on number type.
- D. Normalization is done using the gateway incoming calling party prefixes based on number type.
- E. Normalization is achieved by local route group that is assigned to the MGCP gateway.

Answer: A

Explanation:

QUESTION NO: 8

Which process can localize a global E.164 with + prefix calling numbers for inbound calls to an IP phone so that users see the calling number in a local format?

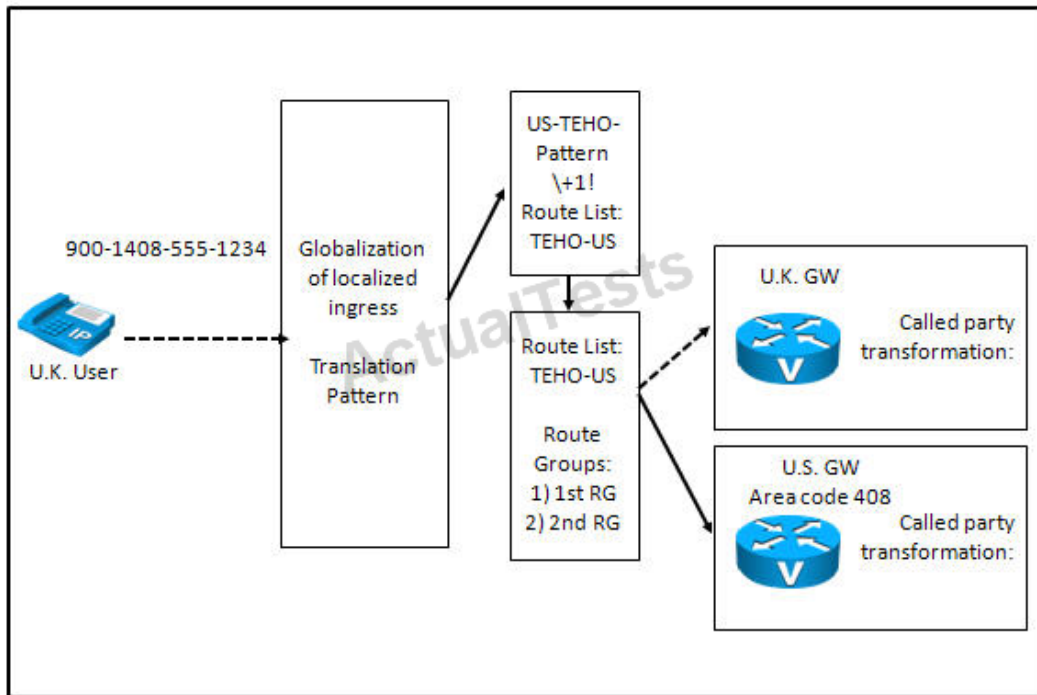
- A. Calling number localization is done using translation patterns.
- B. Calling number localization is done using route patterns.
- C. Calling number localization is done by configuring a calling party transformation CSS at the phone.
- D. Calling number localization is done by configuring a calling party transformation CSS at the gateway.
- E. Calling number localization is done by configuring the phone directory number in a localized format.

Answer: C

Explanation:

QUESTION NO: 9

Refer to the exhibit.



The exhibit shows centralized Cisco Unified Communications Manager configuration components for TEHO calls to U.S. area code 408 from the U.K. The PSTN access code for the U.K. is 9 and 001 for international calls to the U.S. To match the US-TEHO pattern \+!, how should the translation pattern be configured?

- A.** 9001.4085551234 with the Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: +
- B.** 9.0014085551234 with the Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: +1
- C.** 900.14085551234 with the Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: +1
- D.** 900.14085551234 with the Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: +
- E.** 001.4085551234 with the Called Party Transformation:
Prefix Digits Outgoing Calls: +

Answer: D

Explanation: Incorrect answer: ABC

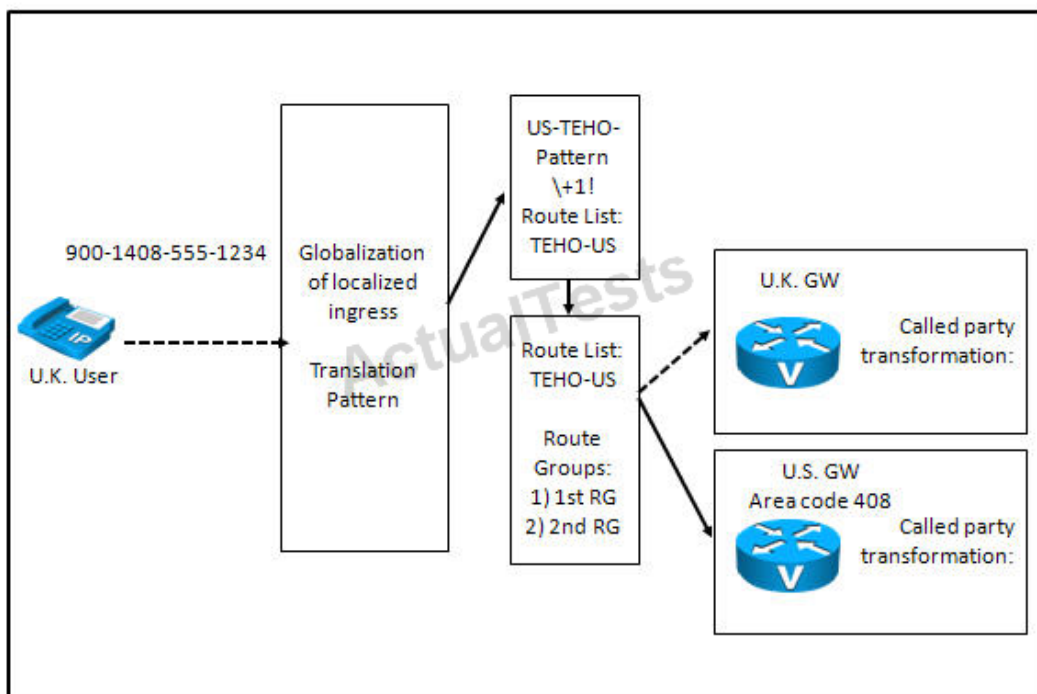
The PSTN access code for the UK is 9, International call code is 001, The international escape character, +, signifies the international access code in a complete E.164 number format

Link:

http://www.ciscosystems.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a03rp.html

QUESTION NO: 10

Refer to the exhibit.



The exhibit shows centralized Cisco Unified Communications Manager configuration components for TEHO calls to U.S. area code 408 from the U.K. The PSTN access code for the U.K. is 9 and 001 for international calls to the U.S. What should the TEHO-US route list configuration consist of?

- A.** First route group should point only to the U.K. gateway. The second route group should point to the U.S. gateway.
- B.** First route group should be only the local route group. The second route group should point to the U.S. gateway.
- C.** First route group should point only to the U.S. gateway. The second route group should be the local route group.
- D.** The TEHO-US route list should contain only the local route group. The globalized configuration

means that the appropriate gateway will be selected automatically.

E. The \+! route pattern should point directly to the U.S. gateway.

Answer: C

Explanation: Incorrect answer: ABD

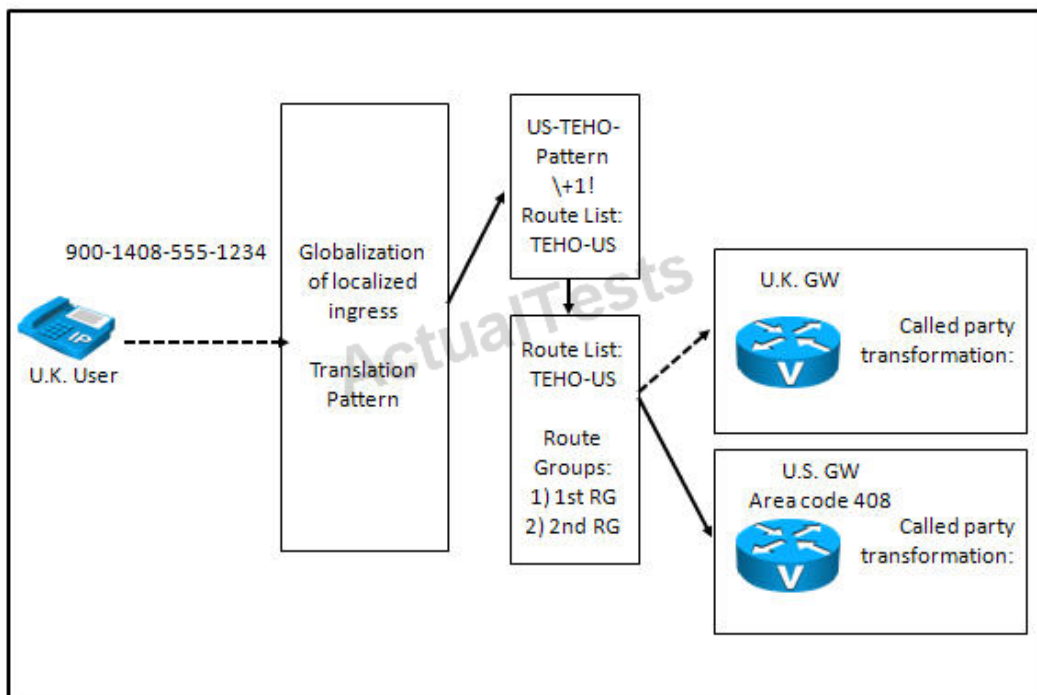
The route group points to one or more gateways and can choose the gateways for call routing based on preference. The route group can serve as a trunk group by directing all calls to the primary device and then using the secondary devices when the primary is unavailable. One or more route lists can point to the same route group.

Link:

http://www.ciscosystems.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a08gw.html#wp1167274

QUESTION NO: 11

Refer to the exhibit.



The exhibit shows centralized Cisco Unified Communications Manager configuration components for TEHO calls to U.S. area code 408 from the U.K. The PSTN access code for the U.K. is 9 and 001 for international calls to the U.S. Assuming the PSTN does not accept globalized numbers with + prefix. What should the Called Party Transformation Pattern at the U.S. gateway be configured as?

A. \+.! with the following Called Party Transformation:

Discard Digits PreDot

Prefix Digits Outgoing Calls: +

B. \+1.! with the following Called Party Transformation:

Discard Digits PreDot

Prefix Digits Outgoing Calls: None

C. \+408.! with the following Called Party Transformation:

Discard Digits PreDot

Prefix Digits Outgoing Calls: 1

D. \+1408.! with the following Called Party Transformation:

Discard Digits PreDot

Prefix Digits Outgoing Calls: None

E. \+1.408! with the following Called Party Transformation:

Discard Digits PreDot

Prefix Digits Outgoing Calls: None

Answer: D

Explanation:

QUESTION NO: 12

Refer to the exhibit.

```
router eigrp SAF
!
service-family ipv4 autonomous-system 1
!
sf-interface FastEthernet0/0
topology base
exit-sf-topology
exit-service-family
```

Which configuration elements must match for adjacent neighbors to establish a SAF neighbor relationship?

A. the label name specified in the router eigrp command

B. the autonomous-system number specified in the service-family ipv4 autonomous-system command

C. the sf-interface configuration

D. the topology base configurations

E. the label name specified in the router eigrp command and the autonomous-system number

Answer: B

Explanation: Incorrect answer: ACDE

service-family ipv4 autonomous-system 1 enables a Cisco SAF service family for the specified autonomous system on the router

Link:

http://www.cisco.com/en/US/docs/ios/saf/configuration/guide/saf_cg_ps10591_TSD_Products_Configuration_Guide_Chapter.html#wp1056363

QUESTION NO: 13

Which statement about Service Advertisement Framework is true?

- A. SAF requires that the EIGRP be configured on all routers, including non-SAF routers.
- B. SAF requires that the EIGRP be configured only on SAF routers. Non-SAF routers act as an IP cloud.
- C. SAF has no dependency on the underlying routing protocol, as long as it is a dynamic routing protocol. Static routes are not supported.
- D. SAF operates on any dynamic or static IP routing configuration. SAF is totally independent of the underlying routing protocol.

Answer: D

Explanation:

Because Cisco SAF is independent of IP routing and uses underlying Cisco routing technology to distribute service advertisements in a reliable and efficient manner, Cisco SAF will run in networks over any routing protocol they may have in place such as Enhanced Interior Gateway Routing Protocol (EIGRP), Open Shortest Path First (OSPF), Exterior Border Gateway Protocol (EBGP) over an MPLS service, or static routing (Figure 2).

http://www.cisco.com/en/US/prod/collateral/iosswrel/ps6537/ps6554/ps6599/ps10822/whitepaper_c11-636604.html

QUESTION NO: 14

Assume that the Cisco IOS SAF Forwarder is configured correctly. Which minimum configurations on Cisco Unified Communications Manager are needed for the SAF registration to take place?

- A. SAF Trunk, SAF Security Profile, SAF Forwarder, and CCD Advertising Service
- B. SAF Trunk, SAF Security Profile, SAF Forwarder, and CCD Requesting Service

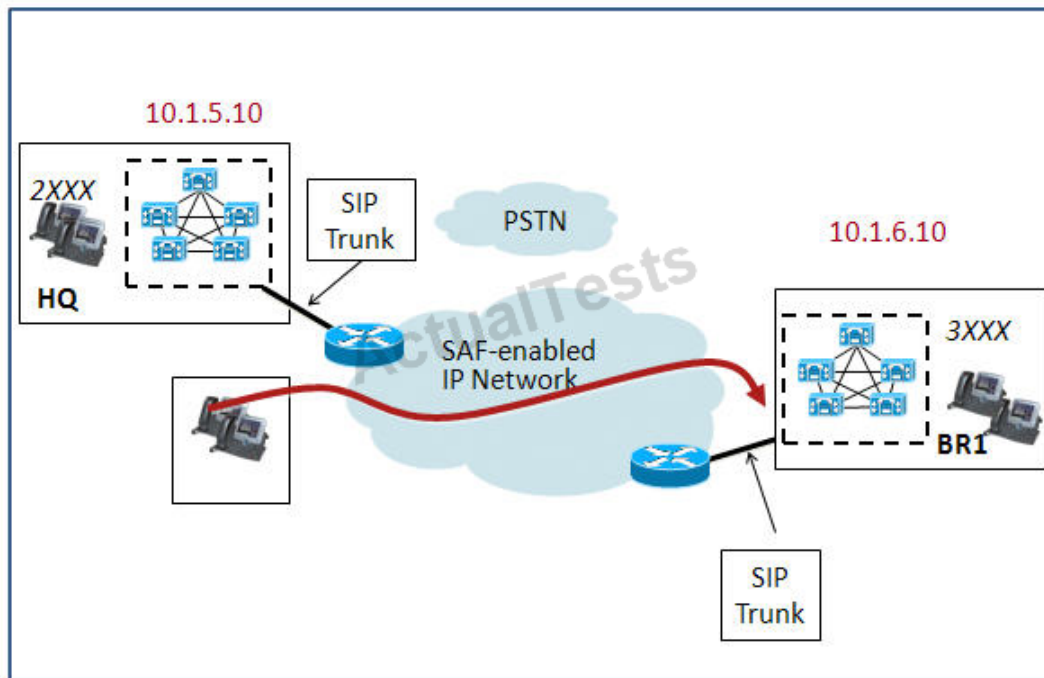
- C. SAF Trunk, SAF Security Profile, SAF Forwarder, CCD Requesting Service, and CCD Advertising Service
- D. SAF Trunk, SAF Security Profile, and SAF Forwarder
- E. SAF Trunk, CCD Requesting Service, and CCD Advertising Service

Answer: B

Explanation:

QUESTION NO: 15

Refer to the exhibit.



What should the destination IP address be configured as on the HQ and BR1 SIP trunks?

- A. The HQ SIP trunk destination IP address should be 10.1.6.10. The BR1 SIP trunk destination IP address should be 10.1.5.10.
- B. The destination IP address is not configurable. Each cluster will learn about the remote trunk IP address through SAF learned routes.
- C. The destination IP address will be learned automatically and configured on the SIP trunks after the Cisco Unified Communications Managers discover themselves.
- D. The HQ SIP trunk destination IP address should be the HQ SAF Forwarder IP address. The BR1 SIP trunk destination IP address should be the BR1 SAF Forwarder IP address.

Answer: B

Explanation: Incorrect answer: ACD

The gatekeeper changes the IP address of this remote device dynamically to reflect the IP address of the remote device.

Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a08trnk.html

QUESTION NO: 16

When a SIP trunk is added for Call Control Discovery, which statement is true?

- A.** The SIP trunk is added by selecting SIP Trunk and SIP Protocol. The Enable SAF check box should be selected.
- B.** The SIP trunk is added by selecting SIP Trunk and SIP Protocol. The Trunk Service Type should be Call Control Discovery.
- C.** The SIP trunk is added by selecting Call Control Discovery Trunk and then selecting SIP as the protocol to be used.
- D.** The SIP trunk is added by selecting SIP Trunk and SIP Protocol. The destination IP address field is configured as 'SAF' to indicate that this trunk is used for SAF.

Answer: B

Explanation:

QUESTION NO: 17

When an H.323 trunk is added for Call Control Discovery, which statement is true?

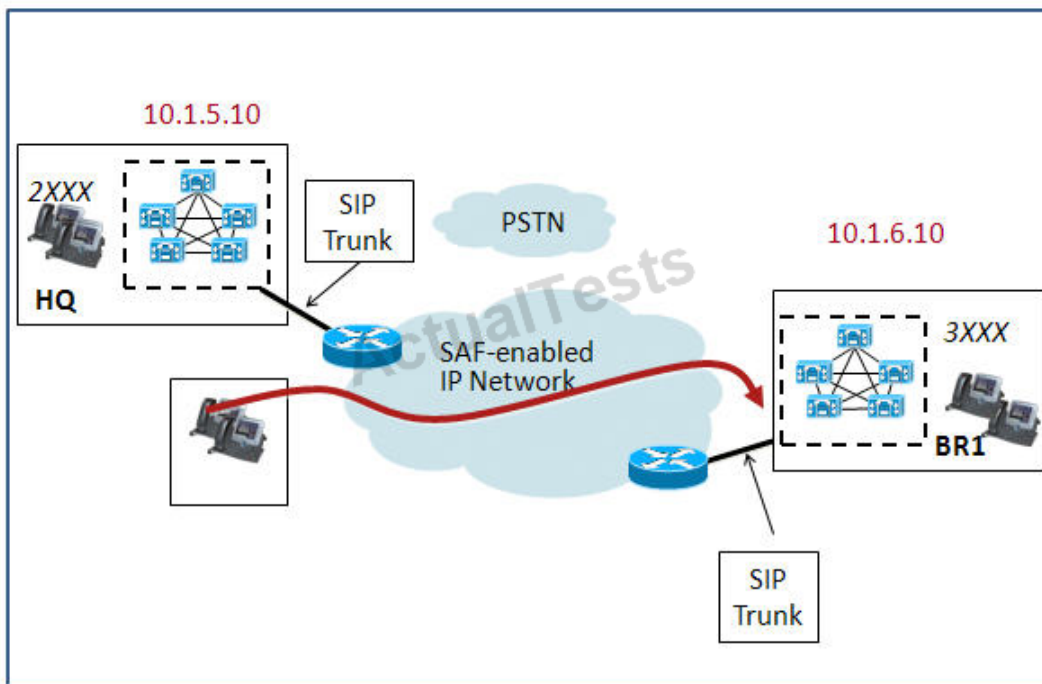
- A.** The H.323 trunk is added by selecting Inter-Cluster Trunk (Non-Gatekeeper Controlled) and Device Protocol Inter-Cluster Trunk. The Enable SAF check box should be selected in the trunk configuration.
- B.** The H.323 trunk is added by selecting Inter-Cluster Trunk (Non-Gatekeeper Controlled) and Device Protocol Inter-Cluster Trunk. The Trunk Service Type should be Call Control Discovery.
- C.** The H.323 trunk is added by selecting Call Control Discovery Trunk and then selecting H.323 as the protocol to be used.
- D.** The H.323 trunk is added by selecting H.323 Trunk, and selecting Inter-Cluster Trunk as the Device Protocol. The destination IP address field is configured as 'SAF' to indicate that this trunk is used for SAF.

Answer: A

Explanation: Reference: Implementing Cisco Unified Communications Manager Part 2 (CIPT2), Chapter 3: Implementing Multisite Connections, pg 70-71, Fig 3-14 and Fig 3-15

QUESTION NO: 18

Refer to the exhibit.



What must be configured on the HQ Cisco Unified Communications Manager to allow HQ users to dial the SAF learned directory number pattern 3XXX?

- A. Route pattern 3XXX should be configured and made available to HQ users through the phone CSS.
- B. Route pattern 3XXX should be configured and made available to HQ phone users through the phone AAR CSS.
- C. The SAF partition assigned to the SAF learned patterns must be available to the HQ phone users through the phone CSS.
- D. The SAF partition assigned to the SAF learned patterns must be available to the HQ phone users through the phone AAR CSS.
- E. The SAF directory number pattern 3XXX will be made available to all users automatically as soon as the SAF partition is selected.

Answer: C

Explanation: Incorrect answer: ABD

By adopting the SAF network service, the call control discovery feature allows Cisco Unified Communications Manager to advertise itself along with other key attributes.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fscallcontroldiscovery.html

QUESTION NO: 19

Refer to the exhibit.

```
dial-peer hunt 2
voice service saf
  profile trunk-route 1
    session protocol sip interface Loopback1 transport tcp port 5060
  !
  profile dn-block 1 alias-prefix 1972555
    pattern 1 type extension 4XXX
  !
  profile dn-block 2
    pattern 1 type global 14087071222
  !
  profile callcontrol 1
    dn-service
      trunk-route 1
      dn-block 1
      dn-block 2
    !
  !
channel 1 vrouter SAF asystem 1
  subscribe callcontrol wildcarded
  publish callcontrol 1
  !
```

Which CSS is used at the HQ Cisco Unified Communications Manager to reroute calls via the PSTN when the SAF network is unavailable?

- A. the phone device CSS
- B. the phone line CSS
- C. the phone line/device combined CSS
- D. the SAF CSS configured on the CCD requesting service
- E. the phone AAR CSS configured at the phone device
- F. No special CSS is required. If SAF patterns are accessible, the PSTN reroute is automatic.

Answer: E

Explanation:

QUESTION NO: 20

Refer to the exhibit.

```
dial-peer hunt 2
voice service saf
  profile trunk-route 1
    session protocol sip interface Loopback1 transport tcp port 5060
  !
  profile dn-block 1 alias-prefix 1972555
    pattern 1 type extension 4XXX
  !
  profile dn-block 2
    pattern 1 type global 14087071222
  !
  profile callcontrol 1
    dn-service
      trunk-route 1
      dn-block 1
      dn-block 2
    !
  !
  !
channel 1 vrouter SAF asystem 1
  subscribe callcontrol wilddcarded
  publish callcontrol 1
  !
```

How does the Cisco Unified Communications Manager advertise dn-block 1?

- A. 4XXX and the ToDID will 0:
- B. 4XXX and the ToDID will 0:1972555
- C. 4XXX
- D. 4XXX and the ToDID will 0:+ 1972555
- E. 19725554XXX

Answer: B

Explanation:

QUESTION NO: 21

Refer to the exhibit.

```

dial-peer hunt 2
voice service saf
  profile trunk-route 1
    session protocol sip interface Loopback1 transport tcp port 5060
  !
  profile dn-block 1 alias-prefix 1972555
    pattern 1 type extension 4XXX
  !
  profile dn-block 2
    pattern 1 type global 14087071222
  !
  profile callcontrol 1
    dn-service
      trunk-route 1
      dn-block 1
      dn-block 2
  !
  !
channel 1 vrouter SAF asystem 1
  subscribe callcontrol wildcarded
  publish callcontrol 1
  !

```

How does the Cisco Unified Communications Manager advertise dn-block 2?

- A. 14087071222 with number type international
- B. +14087071222 with number type international
- C. +14087071222
- D. 14087071222

Answer: C

Explanation:

QUESTION NO: 22

Refer to the exhibit.

Region Information			
Name* Default			
Region Relationships			
Region	Max Audio Bit Rate	Max Video Call Bit Rate (Includes Audio)	Link Loss Type
BR	8 kbps (G.729)	None	Lossy
Default	64 kbps (G.722, G.711)	None	Use System Default
SAF	8 kbps (G.729)	None	Use System Default
NOTE: Region(s) not displayed Use System Default Use System Default Use System Default			
Modify Relationship to other Regions			
Regions	Max Audio Bit Rate	Max Video Call Bit Rate (Includes Audio)	Link Loss Type
BR Default SAF	Keep Current Setting	☐ Keep Current Setting ☐ Use System Default ☒ None ☐ kbps	Lossy

Which statement about the configuration between the Default and BR regions is true?

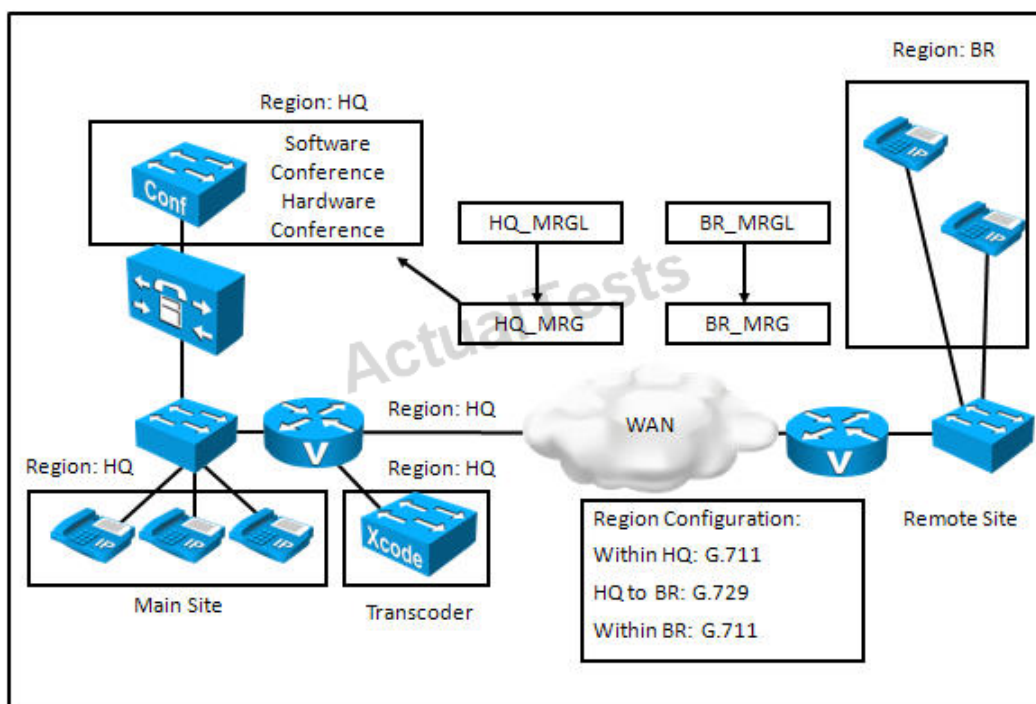
- A. Calls between the two regions can use either 64 kbps or 8 kbps.
- B. Calls between the two regions can use only the G.729 codec.
- C. Only 64 kbps will be used between the two regions because the link is "lossy".
- D. Both codecs can be used depending on the loss statistics of the link. When lossy conditions are high, the G.711 codec will be used.

Answer: B

Explanation:

QUESTION NO: 23

Refer to the exhibit.



When a call between two HQ users is being conferenced with a remote user at BR, which configuration is needed?

- A. The BR_MRG must contain the transcoder device. The BR_MRGL must be assigned to the BR phones.
- B. The HQ_MRG must contain the transcoder device. The HQ_MRGL must be assigned to the HQ phones.
- C. A transcoder should be configured at the remote site and assigned to all remote phones

through the BR_MRGL.

D. The HQ_MRGL must contain the transcoder device. The HQ_MRGL must be assigned to the software conference bridge.

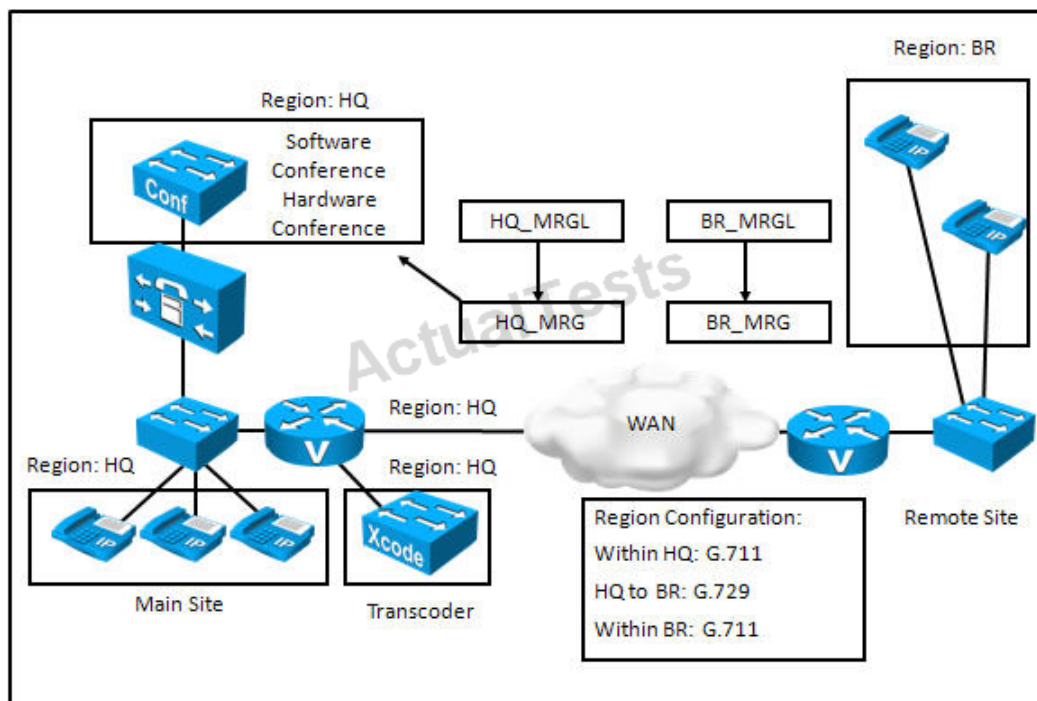
E. Enable the software conference bridge to support the G.711 and G.729 codecs in Cisco Unified Communications Manager Service Parameters.

Answer: D

Explanation:

QUESTION NO: 24

Refer to the exhibit.



All HQ phones are configured to use HQ_MRGL and all BR phones are configured to use BR_MRGL. For the HQ phones always to use the hardware conference bridge as a first choice, which configuration should be implemented?

- A.** Ensure that both the hardware and software conference bridges are listed in the HQ_MRGL. Ensure that the instance ID for the hardware conference bridge is 0.
- B.** Ensure that both the hardware and software conference bridges are listed in the HQ_MRGL. The hardware conference bridge must be configured first.
- C.** Assign the hardware conference bridge to HQ_MRGL. Configure a second HQ_MRGL_2 and assign the software conference bridge to it. Add both the HQ_MRGL and HQ_MRGL_2 to the HQ_MRGL and list the HQ_MRGL first.
- D.** Assign the hardware conference bridge to HQ_MRGL. Configure a second HQ_MRGL_2 and

assign the software conference bridge to it. Configure an additional HQ_MRGL_2. Add the HQ_MRG to HQ_MRGL. Add HQ_MRG_2 to HQ_MRGL_2. The HQ_MRGL should be assigned to the HQ phones. The HQ_MRGL_2 should be assigned to the HQ device pool.

Answer: C

Explanation: Expiation:

To ensure that the hardware bridge is utilized first with all its resources BEFORE the software bridge is used ... you need to have two separate MRG's and list the hardware MRG 1st in the MRGL ...

QUESTION NO: 25

Refer to the exhibit.

Location Information	
Name*	BR
Audio Calls Information	
Audio Bandwidth*	☐ Unlimited ☒ 96 kbps
If the audio quality is poor or choppy, lower the bandwidth setting. For ISDN, use multiples of 56 kbps or 64 kbps.	
Video Calls Information	
Video Bandwidth*	☒ None ☐ Unlimited ☐ kbps

Assuming the regions configuration to BR only permits G.729 codec, how many calls are allowed for the BR location?

- A. Total of four calls; two incoming and two outgoing.
- B. Total of two calls in either direction.
- C. Total of four calls to the BR location. Outgoing calls are not impacted by the location configuration.
- D. Total of four calls in either direction.
- E. Two outgoing calls. Incoming calls are unlimited.

Answer: D

Explanation: Incorrect answer: ABCE

In performing location bandwidth calculations for purposes of call admission control, Cisco Unified Communications Manager assumes that each G.729 call stream consumes 24 kb/s amount of

bandwidth

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a02cac.html#wpxref28640

QUESTION NO: 26

Refer to the exhibit.

```
!  
sccp local FastEthernet0/0  
sccp ccm 10.1.1.1 identifier 1 version 8.0  
sccp  
!  
sccp ccm group 1  
  associate ccm 1 priority 1  
  associate profile 1 register HQ-1_MTP  
!  
dspfarm profile 1 mtp  
  codec pass-through  
  rsvp  
  maximum sessions software 20  
  associate application SCCP  
!  
interface Serial0/1  
  description IP-WAN  
  ip address 10.1.4.101 255.255.255.0  
  duplex auto  
  speed auto  
  ip rsvp bandwidth 64  
!
```

How many calls are permitted by the RSVP configuration?

- A. one G.711 call
- B. two G.729 calls
- C. one G.729 call and one G.711 call
- D. eight G.729 calls
- E. four G.729 calls

Answer: B

Explanation: Incorrect answer: ACDE

In performing location bandwidth calculations for purposes of call admission control, Cisco Unified Communications Manager assumes that each call stream consumes the following amount of bandwidth:

- G.711 call uses 80 kb/s.
- G.722 call uses 80 kb/s.
- G.723 call uses 24 kb/s.
- G.728 call uses 26.66 kb/s.
- G.729 call uses 24 kb/s.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a02cac.html#wpxref28640

QUESTION NO: 27

Refer to the exhibit.

```
!  
sccp local FastEthernet0/0  
sccp ccm 10.1.1.1 identifier 1 version 8.0  
sccp  
!  
sccp ccm group 1  
  associate ccm 1 priority 1  
  associate profile 1 register HQ-1_MTP  
!  
dspfarm profile 1 mtp  
  codec pass-through  
  rsvp  
  maximum sessions software 20  
  associate application SCCP  
!  
interface Serial0/1  
  description IP-WAN  
  ip address 10.1.4.101 255.255.255.0  
  duplex auto  
  speed auto  
  ip rsvp bandwidth <value>  
!
```

To permit three G.729 calls, what should the bandwidth value be for the ip rsvp bandwidth command?

- A. 32
- B. 48
- C. 64
- D. 88

E. 128

Answer: D

Explanation:

QUESTION NO: 28

How is a SIP trunk in Cisco Unified Communications Manager configured for SIP precondition?

- A. The configuration is done by selecting a SIP precondition trunk for trunk type.
- B. The configuration is automatically selected when RSVP is enabled for the location assigned to the trunk.
- C. SIP precondition is configured by selecting E2E for RSVP over SIP on the default SIP profile assigned to the SIP trunk.
- D. SIP precondition is configured by configuring a new SIP profile and selecting E2E for RSVP over SIP. The new SIP profile must then be assigned to the SIP trunk.

Answer: D

Explanation:

QUESTION NO: 29

Which statement about SIP precondition is most correct?

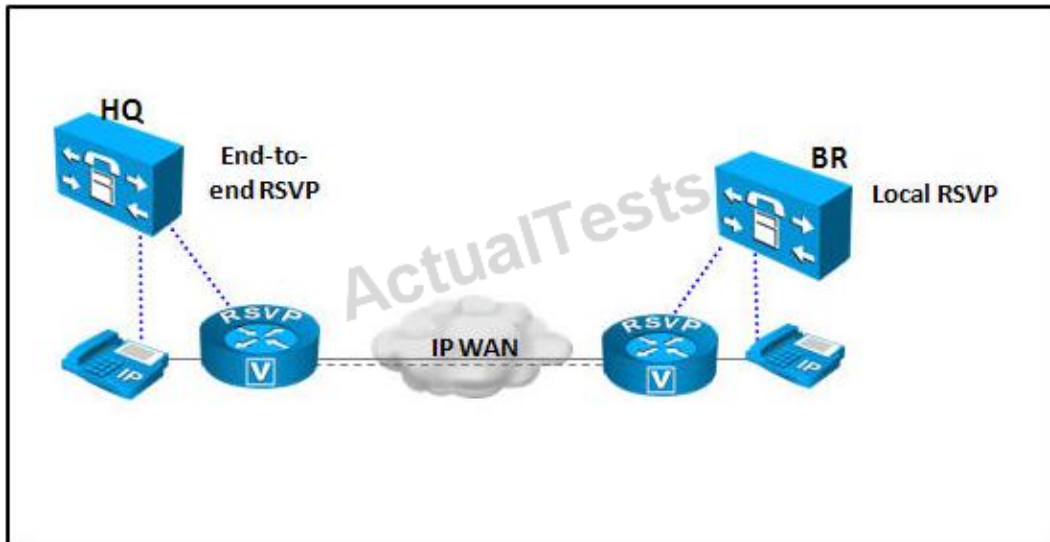
- A. When configuring SIP precondition, the SIP trunk must have access to an RSVP agent.
- B. When configuring SIP precondition, the IP phones must have access to an RSVP agent.
- C. When configuring SIP precondition, the IP phones and SIP trunk must have access to an RSVP agent.
- D. RSVP agents are only required for the IP phones. SIP trunks require RSVP agents only when fall back to local RSVP is configured.
- E. SIP trunk will always require RSVP agents regardless of what RSVP type is configured.

Answer: D

Explanation:

QUESTION NO: 30

Refer to the exhibit.



The HQ Cisco Unified Communications Manager has been configured for end-to-end RSVP. The BR Cisco Unified Communications Manager has been configured for local RSVP.

RSVP between the locations assigned to the IP phones and SIP trunks at each site are configured with mandatory RSVP. When a call is placed from the IP phone at HQ to the BR phone at the BR site, which statement is true?

- A. The Cisco Unified Communications Manager at HQ will fall back to local RSVP and place the call. No RSVP end-to-end will occur.
- B. RSVP end-to-end will occur.
- C. The Cisco Unified Communications Manager at HQ will use end-to-end RSVP. The BR Cisco Unified Communications Manager will use local RSVP.
- D. The call will fail.
- E. The call will proceed as a normal call with no RSVP reservation.

Answer: D

Explanation: Incorrect answer: ABC

A possible cause is that the same router is being used as the calling and called RSVP agents, and that router is not running the latest IOS version, which supports loopback on RSVP reservation. Make sure that the router is running the latest IOS version.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a02rsvp.html#wp1155102

QUESTION NO: 31

Which statement about H.323 Gatekeeper Call Admission Control is true?

- A. Gatekeeper Call Admission Control applies to centralized Cisco Unified Communications deployments that use locations based Call Admission Control.
- B. Gatekeeper Call Admission Control applies to distributed Cisco Unified Communications deployments.
- C. Gatekeeper Call Admission Control applies only to distributed Cisco Unified Communications Express deployments.
- D. Gatekeeper Call Admission Control setting is configured in Cisco Unified Communications Manager.

Answer: B

Explanation: Incorrect answer: ACD

in distributed call processing deployments on a simple hub-and-spoke topology, you can implement call admission control with a Cisco IOS gatekeeper. In this design, the call processing agent (which could be a Unified CM cluster, Cisco Unified Communications Manager Express (Unified CME), or an H.323 gateway) registers with the Cisco IOS gatekeeper and queries it each time the agent wants to place an IP WAN call.

Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/7x/cac.html#wp1044743

QUESTION NO: 32

Refer to the exhibit.

```
gatekeeper
zone local ClusterA lab.com 192.168.3.254
zone local ClusterB lab.com
zone prefix ClusterA 511*
zone prefix ClusterA 521*
zone prefix ClusterB 512*
zone prefix ClusterB 522*
bandwidth interzone default 64
bandwidth interzone zone ClusterB 48
gw-type-prefix 1#* default-technology
no shutdown
```

How many calls can be placed to Cluster B?

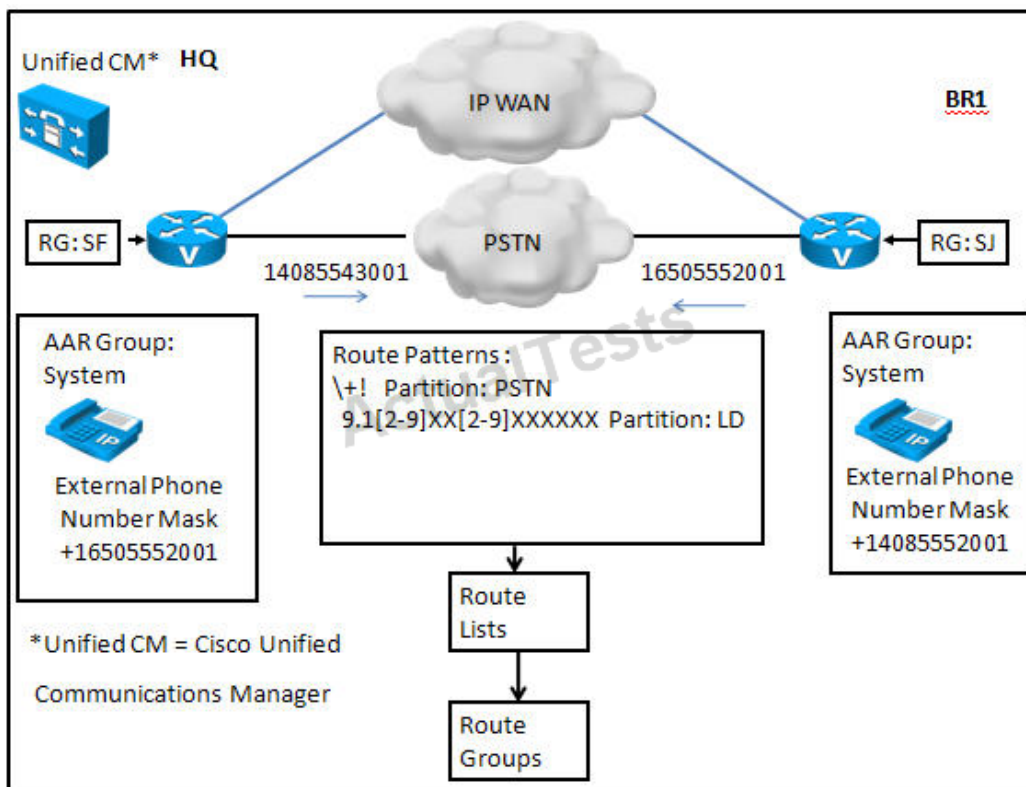
- A. three G.729 calls
- B. one G.711 call
- C. one G.711 and three G.729 calls
- D. There is no limit for incoming calls to Cluster B. Outgoing calls are limited to one G.711 and three G.729 calls.

Answer: A

Explanation:

QUESTION NO: 33

Refer to the exhibit.



The HQ site uses area code 650. The BR1 site uses area code 408. The long distance national code for PSTN dialing is 1. To make a long distance national call, an HQ or BR1 user dials access code 9, followed by 1, and then the 10-digit number. Both sites use MGCP gateways. AAR must use globalized call routing using a single route pattern. Assume that all outgoing PSTN numbers are localized at the egress gateway as shown in the exhibit.

What should the AAR group prefix be?

- A. 9

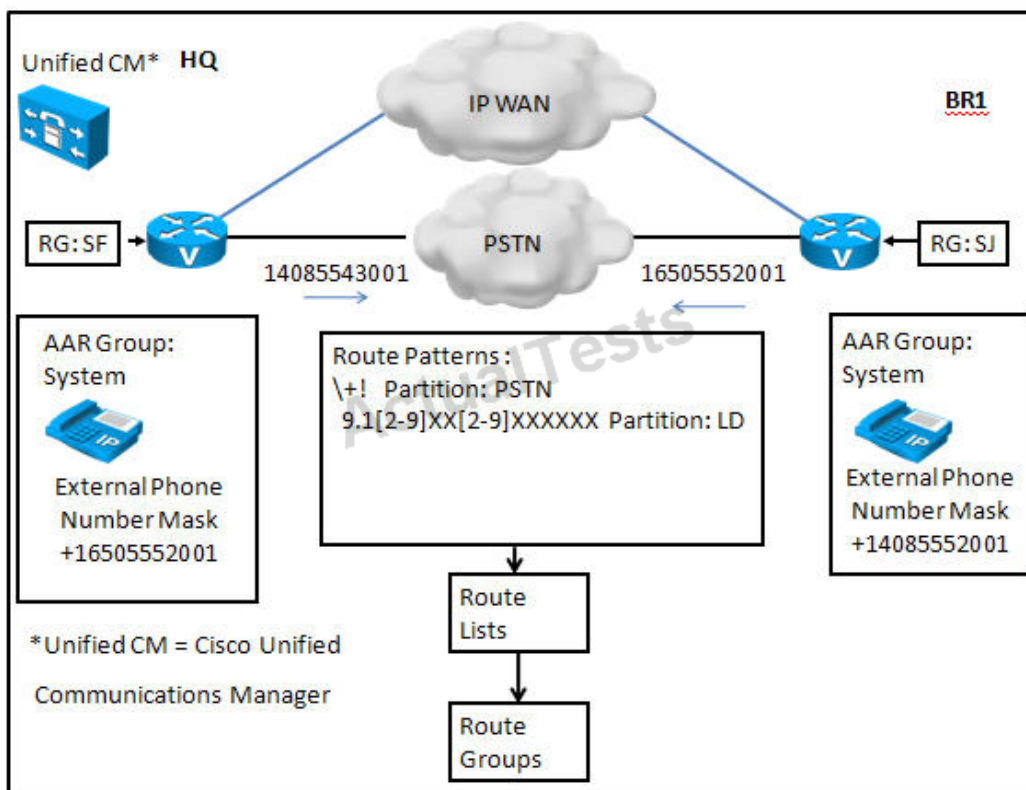
- B. 91
- C. none
- D. +
- E. +1

Answer: C

Explanation:

QUESTION NO: 34

Refer to the exhibit.



The HQ site uses area code 650. The BR1 site uses area code 408. The long distance national code for PSTN dialing is 1. To make a long distance national call, an HQ or BR1 user dials access code 9, followed by 1, and then the 10-digit number.

Both sites use MGCP gateways. AAR must use globalized call routing using a single route pattern. Assume that all outgoing PSTN numbers are localized at the egress gateway as shown in the exhibit.

Which partition should be configured in the AAR CSS applied at the phones?

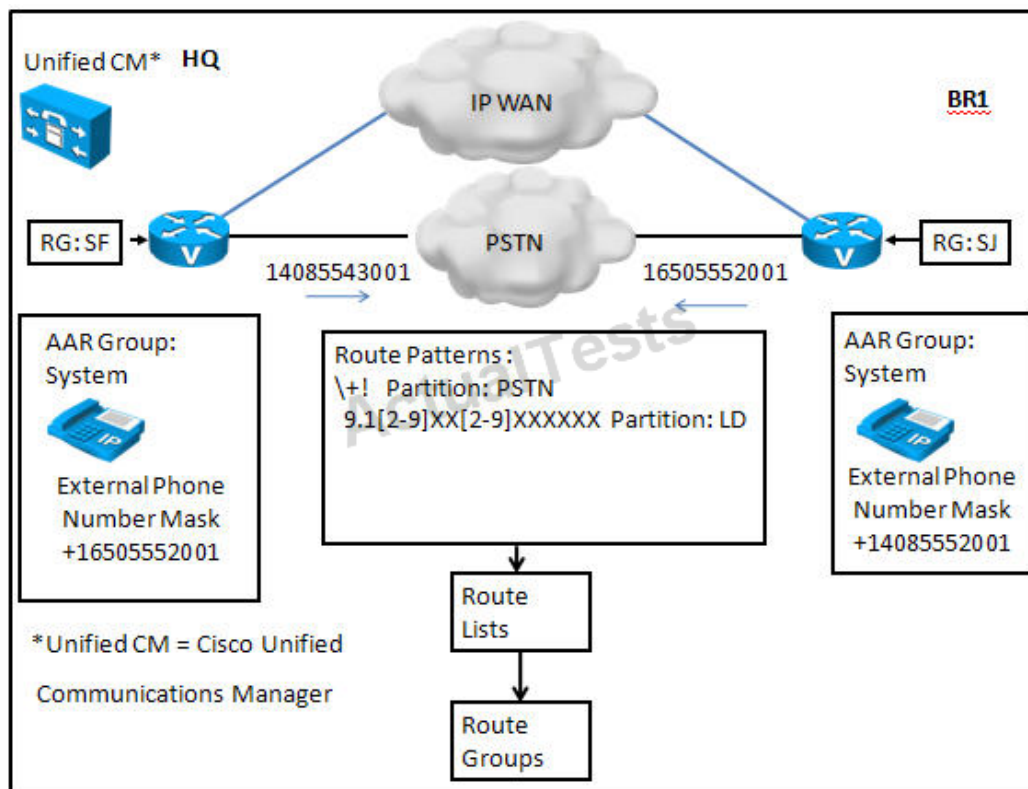
- A. PSTN partition
- B. LD partition
- C. The HQ AAR CSS must include a partition assigned to route pattern 91408XXXXXXX. The BR1 AAR CSS must include a partition assigned to route pattern 91650XXXXXXX.
- D. AAR CSS must contain translation pattern 9.1[2-9]XX[2-9]XXXXXX for each site that must be globalized. Otherwise the called numbers will not be localized at the egress gateway.

Answer: A

Explanation:

QUESTION NO: 35

Refer to the exhibit.



The HQ site uses area code 650. The BR1 site uses area code 408. The long distance national code for PSTN dialing is 1. To make a long distance national call, an HQ or BR1 user dials access code 9, followed by 1, and then the 10-digit number.

Both sites use MGCP gateways. AAR must use globalized call routing using a single route pattern. Assume that all outgoing PSTN numbers are localized at the egress gateway as shown in the exhibit.

How many route lists and route groups should be configured for AAR at a minimum?

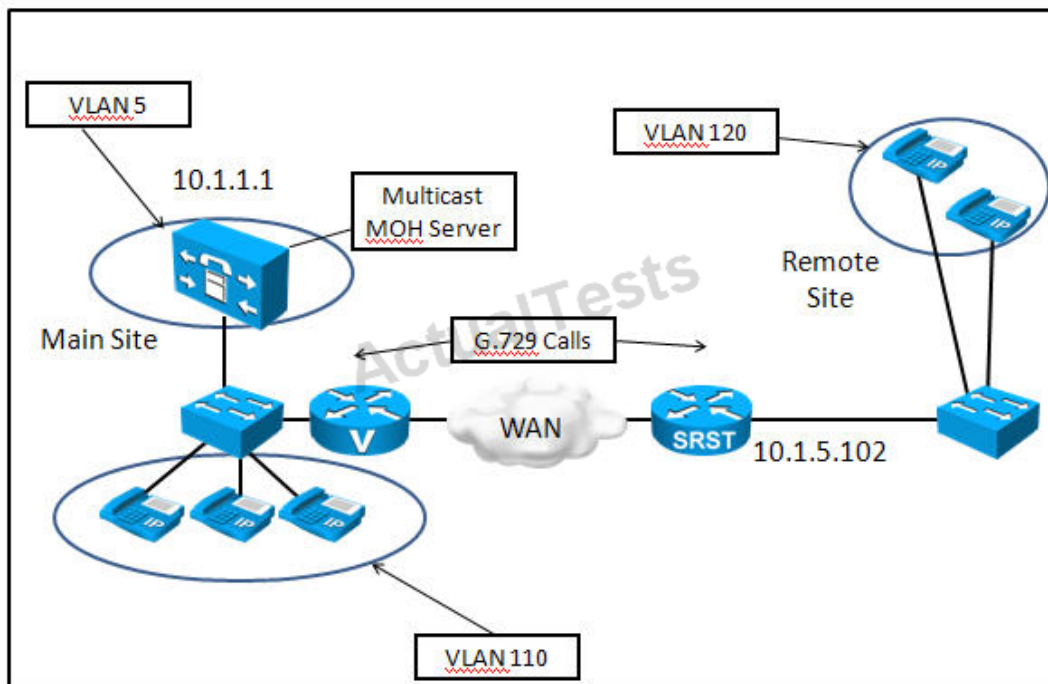
- A. a single route list with a local route group for each site
- B. two route lists and two route groups for each site
- C. a single route list and four route groups for each site
- D. None. The AAR CSS can point directly to the route pattern.

Answer: A

Explanation:

QUESTION NO: 36

Refer to the exhibit.



The remote site needs to run multicast MOH from flash. Which statement about the MOH server configuration in Cisco Unified Communications Manager is true?

- A. The MOH server must be enabled for G.729 in the Cisco IP Voice Media Streaming Application service parameters.
- B. Multicast MOH can use only G.711. So you must configure the command codec G711ulaw under the call-manager-fallback configuration at the remote site router.
- C. The MOH for the remote site is a standalone configuration. No extra configuration is required on the Cisco Unified Communications Manager MOH server.

D. Configure a separate region for the MOH server. The codec between the MOH region and all other regions should be specified as G.711. Apply the MOH region through a device pool at the MOH server configuration page.

E. Configure the location setting for the MOH server to 80 Kbps. This configuration forces the MOH server to use G.711 for the remote site.

Answer: D

Explanation:

QUESTION NO: 37

Assume that local route groups are configured. When an IP phone moves from one device mobility group to another, which two configuration components are not changed? (Choose two.)

- A.** IP subnet
- B.** user settings
- C.** SRST reference
- D.** region
- E.** phone button settings

Answer: B,E

Explanation: Incorrect answer: ACD

Although the phone may have moved from one subnet to another, the physical location and associated services have not changed.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/7_1_2/ccmfeat/fsdevmob.html#wp1137460

QUESTION NO: 38

Where do you specify the device mobility group and physical location after they have been configured?

- A.** phones
- B.** DMI
- C.** device mobility CSS
- D.** device pool
- E.** MRGL
- F.** locale

Answer: D

Explanation: Incorrect answer: ABCE

Before you configure a device pool, you must configure the following items if you want to choose them for the device pool, Cisco Unified Communications Manager group (required), Date/time group (required). Region (required) , SRST reference (optional). Media resource group list (optional), Calling search space for auto-registration (optional). Reverted call focus priority (optional), Device mobility group (optional), Device mobility calling search space, Physical location (optional). Location, AAR group. AAR calling search space.

Link:

https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmcfg/b02devpl.html

QUESTION NO: 39

Which two options for a Device Mobility-enabled IP phone are true? (Choose two.)

- A. The phone configuration is not modified.
- B. The roaming-sensitive parameters of the current (that is, the roaming) device pool are applied.
- C. The user-specific settings determine which location-specific settings are downloaded from the Cisco Unified Communications Manager device pool.
- D. If the DMGs are the same, the Device Mobility-related settings are also applied.

Answer: B,D

Explanation:

QUESTION NO: 40

What impact do roaming-sensitive settings and Device Mobility settings have on call routing?

- A. Device Mobility settings have no impact on call routing, but roaming-sensitive settings modify the AAR group, AAR CSS, and device CSS.
- B. Device Mobility settings modify the device CSS and the roaming-sensitive settings modify the AAR group and AAR CSS.
- C. Device Mobility settings modify the AAR group and the AAR CSS, the roaming-sensitive settings modify the device CSS.
- D. Roaming-sensitive settings are settings that do not have an impact on call routing. Device Mobility settings, on the other hand, may have an impact on call routing because they modify the device CSS, AAR group, and AAR CSS.

Answer: D

Explanation:

QUESTION NO: 41

In what Cisco solution is Simple Network-Enabled Auto Provision technology used?

- A. Cisco Unified Gateway Duplication
- B. Cisco Unified CallManager Redundancy
- C. Cisco Unified SRST
- D. Cisco Unified Call Survivability

Answer: C

Explanation: Incorrect answer: ABD

When the system automatically detects a failure, Cisco Unified SRST uses Simple Network Auto Provisioning (SNAP) technology to auto-configure a branch office router to provide call processing for the Cisco Unified IP phones that are registered with the router

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmestrst.html

QUESTION NO: 42

While configuring Call Survivability in Cisco Unified Communications Manager, what step is mandatory to reach remote sites while in SRST mode?

- A. Enable Cisco Remote Site Reachability.
- B. Configure CFUR.
- C. Enable the SRST checkbox in the MGCP gateway.
- D. Configure the H.323 gateway for SRST in Cisco Unified Communications Manager.
- E. Enable the Failover Service parameter.

Answer: B

Explanation: Incorrect answer: ACDE

Call Forward Unregistered (CFUR) functionality provides the automated rerouting of calls through the PSTN when an endpoint is considered unregistered due to a remote WAN link failure

Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/8x/models.html

QUESTION NO: 43

While operating in SRST, what is needed to route calls outside of the remote site location to the PSTN?

- A. SIP trunk
- B. CallManager route patterns
- C. translation patterns
- D. POTS dial peers
- E. VOIP dial peers

Answer: D

Explanation: Incorrect answer: ABCE

in time of srst configuration on router, please configure a dial-peer so that call flow in SRST mode.
Link:

QUESTION NO: 44

When using Cisco Unified Communications Manager Express in SRST mode, how many multicast music on hold streams can be utilized by the system at any given time?

- A. 3
- B. 6
- C. 2
- D. 4
- E. 1
- F. 5

Answer: B

Explanation:

QUESTION NO: 45

Refer to the exhibit.

```
voice moh-group 1
  moh flash:moh1.au
  description MOH: customer services
  multicast moh 239.1.1.1 port 16384
  extension-range 1000 to 1099
  extension-range 1300 to 1399
!
voice moh-group 2
  moh flash:moh2.au
  description MOH: marketing
  multicast moh 239.1.1.2 port 16384
  extension-range 3000 to 3099
!
call-manager-fallback
  moh-file-buffer 5000
  moh flash:default.wav
  multicast moh 239.1.1.3 port 16384
```

What music on hold audio source will be heard if a user at extension 1372 places the user at extension 3041 on hold?

- A. moh2.au
- B. moh1.wav
- C. default.wav
- D. moh2.wav
- E. moh1.au

Answer: E

Explanation:

Extension 1372 is in voice moh-group 1 and its moh that it uses to place people on hold is customer services moh moh1.au

QUESTION NO: 46

To preserve analog calls in an MGCP switchback event, which three commands must be configured in the MGCP fallback router? (Choose three.)

- A. h323
- B. mgcp-switchback-graceful
- C. voice service voip
- D. mgcp-graceful
- E. preserve-h323
- F. no h225 timeout keepalive

Answer: A,C,F

Explanation: Incorrect answer: BDE

these additional command for call preservation when using MGCP fallback:

voice service voip

h323

no h225 timeout keepalive

Reference: CCNP Voice CIPT2 642-457 Quick Reference, 2nd Edition

QUESTION NO: 47

Which two locations are the best locations that an end user can use to determine if an IP phone is working in SRST mode? (Choose two.)

A. Cisco Unified Communications Manager Administration

B. IP phone display

C. Cisco Unified SRST Router

D. Cisco Unified MGCP Fallback Router

E. physical IP phone settings

Answer: B,E

Explanation: Incorrect answer: ACD

IP Phone display and Physical phone IP settings are two locations where an end user can determine if an IP phone is working in SRST mode.

Link: <http://my.safaribooksonline.com/book/telephony/1587050757/survivable-remote-site-telephony-srst/529>

QUESTION NO: 48

What is the fastest way for an engineer to test the implementation of SRST in a production environment?

A. Shut down the Cisco Unified Communications Manager Servers.

B. Shut down the switch ports connected to the Cisco Unified Communications Manager Servers.

C. Add a null route to the publisher Cisco Unified Communications Manager at the remote router. Remove the null route when the operation is verified.

D. Unplug the IP phones from their switch ports.

E. Verification is not needed.

Answer: C

Explanation:

QUESTION NO: 49

When Cisco Extension Mobility is implemented, which CSS is used for calling privileges?

- A.** The user device profile line CSS combined with the device CSS of the physical phone used to log in the extension mobility user.
- B.** The user device profile device CSS combined with the line CSS of the physical phone used to log in the extension mobility user.
- C.** Only the user device profile device CSS is used.
- D.** The combined line/device CSS of the physical phone is used to log in the extension mobility user.
- E.** The combined line/device CSS of the user device profile.

Answer: A

Explanation:

QUESTION NO: 50

When multiple Cisco Extension Mobility profiles exist, which actions take place when a user tries to log in to Cisco Extension Mobility?

- A.** The login will fail because only a single Cisco Extension Mobility profile is allowed.
- B.** The user must select the desired profile.
- C.** The user must login to both profiles in the order they are presented.
- D.** The user may login to both profiles in any order.
- E.** Login will only be allowed to multiple profiles if the service parameter Allow Multiple Logins is enabled.

Answer: B

Explanation: Incorrect answer: ACDE

Users access Cisco Extension Mobility by pressing the Services or Applications button on a Cisco Unified IP Phone and then entering login information in the form of a Cisco Unified Communications Manager UserID and a Personal Identification Number (PIN). If a user has more than one user device profile, a prompt displays on the phone and asks the user to choose a device profile for use with Cisco Extension Mobility.

Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fsem.html

QUESTION NO: 51

When Cisco Extension Mobility is implemented, how is the audio source for the MOH selected?

- A.** The audio source that is configured at the user device profile is selected.
- B.** The audio source that is configured at the home phone of the user is selected.
- C.** The audio source that is configured at the physical phone used for the Cisco Extension Mobility login is selected.
- D.** The audio source that is configured in the IP Voice Media Streaming parameters is selected.

Answer: A

Explanation: Incorrect answer: BCD

To specify the audio source that plays when a user initiates a hold action, choose an audio source from the User Hold MOH Audio Source drop-down list box from device profile configuration settings.

Link: http://cisco.biz/en/US/docs/voice_ip_comm/cucmbe/admin/8_6_1/ccmcfg/b06dvprf.html

QUESTION NO: 52

Refer to the exhibit.

RTP Phone Device Configuration	Partitions	RTP Phone DN Configuration	Partitions
Device CSS	RTP_Emergency ALL_Phones	Line CSS	RTP_Local RTP_LongDistance RTP_International
AAR CSS	RTP_LongDistance	AAR Group	AAR

U.K. User Device Profile	Partitions	Partition	Route Pattern
Line CSS	U.K_Emergency ALL_Phones	RTP_Emergency	9.911
AAR Group	AAR	RTP_Local	9.[2-9]XXXXXX
		RTP_LongDistance	9.1[2-9]XX[2-9]XXXXXX
		RTP_International	9.011!#
		U.K_Emergency	0.000
		U.K_PSTN	9.!

Assume a centralized Cisco Unified Communications deployment with the headquarters in the

U.K, and remote site in RTP. All route patterns are assigned a route list that points to the local route group. Local route groups have been configured on the U.K and RTP device pools. A U.K. user logs onto an RTP phone using the Cisco Extension Mobility feature and places an emergency call to 0000. Which statement about the emergency call is true?

- A. The call will match the U.K_Emergency route pattern partition and will egress at the RTP gateway.
- B. The call will match the U.K_Emergency route pattern partition and will egress at the U.K. gateway.
- C. The call will match the RTP_Emergency route pattern partition and will egress at the U.K. gateway.
- D. The call will match the RTP_Emergency route pattern partition and will egress at the RTP gateway.
- E. The call will fail.

Answer: A

Explanation:

QUESTION NO: 53

Which Cisco IOS command is used to verify that a SAF Forwarder that is registered with Cisco Unified Communications Manager has established neighbor relations with an adjacent SAF Forwarder?

- A. show eigrp service-family ipv4 neighbors
- B. show eigrp address-family ipv4 neighbors
- C. show voice saf dndball
- D. show saf neighbors
- E. show ip saf neighbors

Answer: A

Explanation: Incorrect answer: BCDE

Router# show eigrp service-family ipv4 4453 neighbors

EIGRP SFv4 VR(test) Service-Family Neighbors for AS(4453)

Address	Interfaces	Hold Uptime	SRRT (sec)	RTO (sec)	Q Seq	(msec)
10.1.1.1	Ethernet0/0	13 00:00:41	30	1014	0	16
10.1.2.1	Ethernet0/0	14 00:02:02	20	200	0	10

Link: http://www.cisco.com/en/US/docs/ios/saf/command/reference/saf_s1.html

QUESTION NO: 54

Which Cisco IOS command is used to verify that the Cisco Unified Communications Manager Express has registered with the SAF Forwarder?

- A. show eigrp service-family ipv4 clients
- B. show eigrp address-family ipv4 clients
- C. show voice saf dn timer all
- D. show saf registration
- E. show ip saf registration

Answer: A

Explanation: Incorrect answer: B, C, D, E

show eigrp service-family ipv4 clients Displays information from the EIGRP IPv4 service-family results.

QUESTION NO: 55 DRAG DROP

Click and drag the minimum Cisco Unified SRST configuration steps on the left to the spaces on the right. Not all spaces on the right are used.

ccm-manager fallback-mgcp	
call-manager-fallback	
application	
ip source-address 10.5.1.1 port 2000	
service alternate default	
max-ephones 3	
max-dn 6	
limit-dn 7965 2	
keepalive 20	

Answer:

Click and drag the minimum Cisco Unified SRST configuration steps on the left to the spaces on the right. Not all spaces on the right are used.

ccm-manager fallback-mgcp	call-manager-fallback
call-manager-fallback	ip source-address 10.5.1.1 port 2000
application	max-ephones 3
ip source-address 10.5.1.1 port 2000	max-dn 6
service alternate default	
max-ephones 3	
max-dn 6	
limit-dn 7965 2	
keepalive 20	

Explanation:

Call-manager-fallback

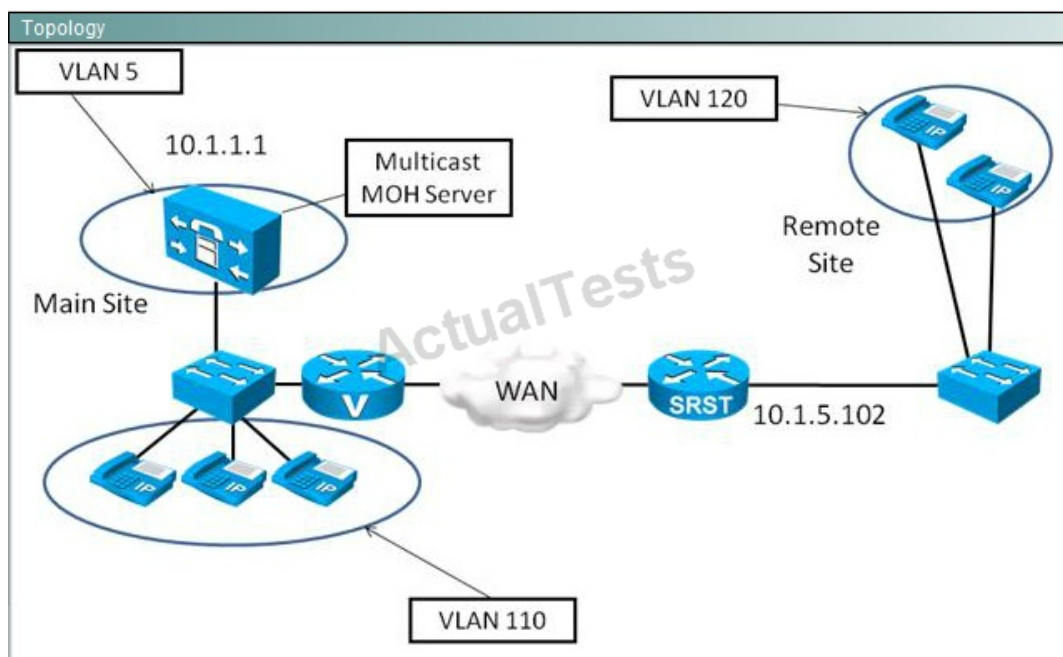
Ip source-address 10.5.1.1 port 2000

Max-ephones 3

Max-dn 6

QUESTION NO: 56

Refer to the exhibit.



MOH Server Config

Device Information

Registration: Registered with Cisco Unified Communications Manager 10.1.5.10

IP Address: 10.1.5.10

Host Server*: 10.1.5.10

Music On Hold Server Name*: MOH_2

Description: MOH_CUCM801Pub1

Device Pool*: Default

Location*: Hub_None

Maximum Half Duplex Streams*: 250

Maximum Multi-cast Connections*: 250000

Fixed Audio Source Device:

Use Trusted Relay Point*: Off

Run Flag*: Yes

Multi-cast Audio Source Information

☒ Enable Multi-cast Audio Sources on this MOH Server

Base Multi-cast IP Address*: 239.1.1.1

Base Multi-cast Port Number*: 16384 (Even numbers only)

Increment Multi-cast on*: ☐ Port Number ☒ IP Address

Selected Multi-cast Audio Sources

No.	Audio Source Name
1	SampleAudioSource

Save Reset Apply Config

To stream multicast MOH to the remote site across the WAN, what should the minimum value for the Max Hops be configured as?

- A. 1
- B. 2
- C. 3
- D. 4

Answer: C

Explanation:

The Max Hops field in the Music On Hold (MOH) Server Configuration window indicates the maximum number of routers that an audio source is allowed to cross. If max hops is set to zero, the audio source must remain in its own subnet. If max hops is set to one, the audio source can cross up to one router to the next subnet. Cisco recommends setting max hops to two.

Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fsmoh.html

QUESTION NO: 57

Refer to the exhibit.

CCD Requesting Service

CCD Requesting Service Info

Name* HQ_REQ_SVC

Description

Route Partition SAF_Pt

Learned Pattern Prefix +

PSTN Prefix 000

Available SAF Trunks SAF_Trunk_HQ_ICT

Selected SAF Trunks SAF_Trunk_HQ_SIP

☒ Activated Feature

RTMT

Learned Pattern

Select a Node CUCM801Pub1

Pattern	TimeStamp	Status	Protocol	AgentId	IP Address	ToDID	CUCMNodeId
3XXX	2010/05/07 14:52:06	Reachable	SIP	CID10.1.5.11	10.1.5.11(5060)	0:44228822	1

When the user of a phone registered to the Cisco Unified Communications Manager places a call to 3001 when the SAF network is down, what happens?

- A. The call fails.
- B. The call is rerouted to the PSTN with the constructed PSTN number as +442288223001
- C. The call is rerouted to the PSTN with the constructed PSTN number as 442288223001
- D. The call is rerouted to the PSTN with the constructed PSTN number as 0002288223001
- E. The call is rerouted to the PSTN with the constructed PSTN number as +0002288223001

Answer: A

Explanation: Incorrect answer: BCD

When the SAF forwarder loses network connection with its call-control entity, the SAF forwarder withdraws those learned patterns that were published by the call control entity. In this case, CCD requesting service marks those learned patterns as unreachable via IP, and the calls gets routed through the PSTN gateway.

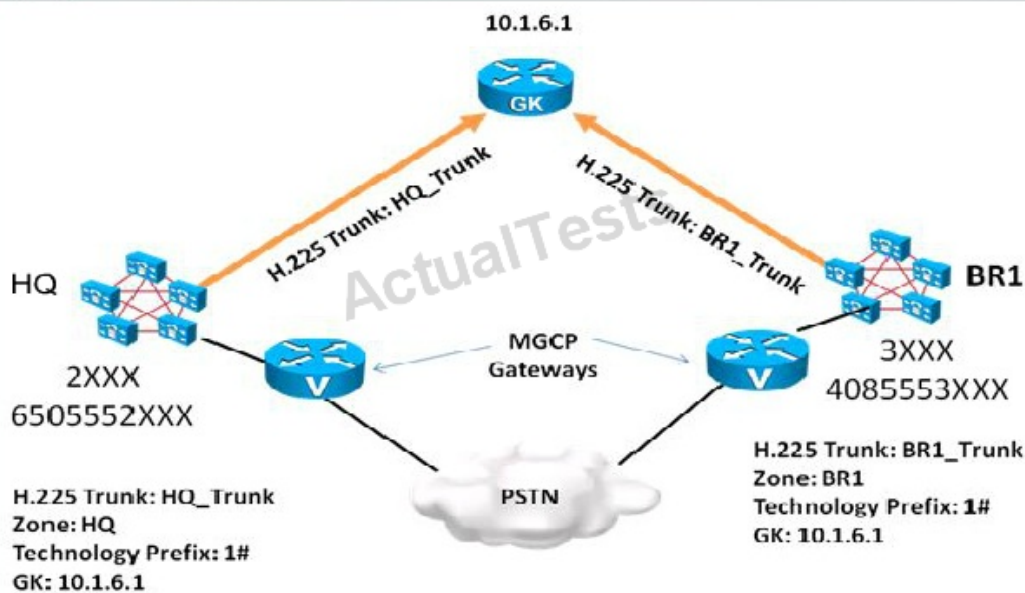
Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fscallcontroldiscovery.html

QUESTION NO: 58

Refer to the exhibit.

Topology



Gatekeeper Config

```

gatekeeper
zone local HQ cisco.com 10.1.6.1
zone local BR1 cisco.com
zone prefix HQ 2...
zone prefix BR1 3...
gw-type-prefix 1#* default-technology
no shutdown

```

Assume that NANP is being used and 9 is used for PSTN access code Long distance national calls are preceded with 1.

How should the HQ Cisco Unified Communications Manager be configured for calls to 3XXX to be sent to the gatekeeper at 1 0 1 6 1 with PSTN backups?

- A.** Configure a route pattern for 3XXX Assign this route pattern to a route list that points to two route groups The first route group contains the H 225 trunk The second route group contains the MGCP gateway with prefix digits 1 408555 for the outgoing called number.
- B.** Configure a route pattern for 1#3XXX Assign this route pattern to a route list that points to a route group that lists the H 225 trunk as first choice and the MGCP gateway as a second choice.
- C.** Configure a route pattern for 4085543XXX. Assign this route pattern to a route list that points to two route groups. The first route group contains the H 226 trunk The second route group contains MGCP gateway.
- D.** Configure a route pattern for 3XXX Assign this route pattern to a route list that points to two

route groups The first route group contains the H 225 trunk The second route group contains MGCP gateway with prefix digits 91 408554 for the called number.

Answer: A

Explanation:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmcfg/b03rtgrp.html#wpxr ef46617

QUESTION NO: 59

Refer to the following exhibit.

SAF Forwarder info

SAF Forwarder Info

Name* HQ_SAF_FWDER

Description

Client Label* HQ_SAF

SAF Security Profile* SAF_SEC_Prof.

SAF Forwarder Address* 10.1.111.1

SAF Forwarder Port* 5050

☒ Enable TCP Keep Alive

[+ Show Advanced](#)

Which Cisco IOS SAF Forwarder configuration is correct?

A

```
router eigrp SAF
|
| service-family ipv4 autonomous-system 1
| |
| | topology base
| |   external-client HQ_SAF
| |   exit-sf-topology
| |   exit-service-family
| |
| |
| service-family external-client listen ipv4 5050
| external-client HQ_SAF
| username SAFUSER
| password SAFPASSWORD
| keepalive 3600000
```

B

```
router eigrp SAF
|
| service-family ipv4 autonomous-system 1
| |
| | topology base
| |   external-client HQ_SAF_FWDER
| |   exit-sf-topology
| |   exit-service-family
| |
| |
| service-family external-client listen ipv4 5050
| external-client HQ_SAF
| username SAFUSER
| password SAFPASSWORD
| keepalive 3600000
```

C

```
router eigrp SAF
|
| service-family ipv4 autonomous-system 1
| |
| | topology base
| |   external-client HQ_SAF
| |   exit-sf-topology
| |   exit-service-family
| |
| |
| service-family external-client listen ipv4 5050
| external-client HQ_SAF_FWDER
| username SAFUSER
| password SAFPASSWORD
| keepalive 3600000
```

D

```
router eigrp SAF
|
| service-family ipv4 autonomous-system 1
| |
| | topology base
| |   external-client HQ_SAF_FWDER
| |   exit-sf-topology
| |   exit-service-family
| |
| |
| service-family external-client listen ipv4 5050
| external-client HQ_SAF_FWDER
| username SAFUSER
| password SAFPASSWORD
| keepalive 3600000
```

A. Exhibit A**B. Exhibit B**

C. Exhibit C

D. Exhibit D

Answer: A

Explanation: Incorrect answer: BCD

Summary steps to configure IOS SAF forwarder is given below

1.

enable
2.

configure terminal
3.

router eigrp *virtual-instance-name*
4.

service-family {ipv4 | ipv6} [vrf *vrf-name*] autonomous-system *autonomous-system-number*
5.

topology base
6.

external-client *client_label*
7.

exit-sf-topology
8.

exit-service-family
9.

exit
10.

service-family external-client listen {ipv4 | ipv6} *tcp_port_number*
11.

external-client *client-label*
12.

username *user-name*
13.

password *password-name*
14.

keepalive *number*
15.

exit

Hosted DN Group	
Hosted DN Group Info	
Name*	BR1_DN
Description	
PSTN Failover Strip Digits	0
PSTN Failover Prepend Digits	44228822
☐ Use HostedDN as PSTN Failover	

Hosted DN Pattern	
Hosted DN Pattern Info	
Hosted Pattern*	3XXX
Description	
Hosted DN Group*	BR1_DN
PSTN Failover Strip Digits	0
PSTN Failover Prepend Digits	
☒ Use HostedDN as PSTN Failover	

Which pattern will be advertised by the Cisco Unified Communications Manager?

- A. 3XXX and the ToDID will be 0:.
- B. 3XXX and the TnOID will be 0:44228822.
- C. 3XXX and the ToDID will be 44228822.
- D. 3XXX and the ToDID will be 0:+44228822.
- E. 3XXX and the ToDID will be 0:+

Answer: A

Explanation: Incorrect answer: BCD

the number of digits that want stripped from the hosted DN if the call fails over to the PSTN. Value can enter 0 and 16. Here 0 is defined.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fscallcontroldiscovery.html

QUESTION NO: 61

Refer to the exhibit.

Location Config

Location Information

Name* BR

Audio Calls Information

Audio Bandwidth* ☐ Unlimited ☒ 96 kbps

If the audio quality is poor or choppy, lower the bandwidth setting. For ISDN, use multiples of 56 kbps or 64 kbps.

Video Calls Information

Video Bandwidth* ☒ None ☐ Unlimited ☐ kbps

LLQ Config

```
class-map match-all ctraffic
match ip dscp cs3
class-map match-all ytraffic
match ip dscp ef
!
policy-map voice2br1
class ytraffic
priority 40
compress header ip rtp
class ctraffic
bandwidth 8
class class-default
fair-queue
policy-map shape2br1
class class-default
shape average 486400 4864 0
service-policy voice2br1
```

Locations-based CAC has been configured between HQ and the BR site. Assume that the priority queue has been provisioned correctly for three G.729 calls. What happens when the fourth call is placed from HQ to BR?

- A. The call will get through via the WAN, but it will experience poor audio quality.
- B. The call will fail.
- C. The call will be queued until one of the existing calls drop.
- D. The call will get through without any issues.

Answer: B

Explanation:

QUESTION NO: 62

Refer to the exhibit.

```
IOS Config
!
sccp local FastEthernet0/0
sccp ccm 10.1.1.1 identifier 1 version 8.0
sccp
!
sccp ccm group 1
  associate ccm 1 priority 1
  associate profile 1 register HQ-1_MTP
!
dspfarm profile 1 mtp
  codec pass-through
  rsvp
  maximum sessions software 20
  associate application SCCP
!
interface Serial0/1
  description IP-WAN
  ip address 10.1.4.101 255.255.255.0
  duplex auto
  speed auto
  ip rsvp bandwidth 40
!
```

What media resource should be configured in Cisco Unified Communications Manager?

- A. Cisco Media Termination Point Hardware
- B. Cisco IOS Enhanced Media Termination Point Cisco
- C. Cisco IOS Media Termination Point
- D. Cisco Media Termination Point Hardware (WS-SVC-CMM)

Answer: B

Explanation: Incorrect answer: ACD

Each media termination point receives a list of Cisco Unified Communications Managers, in priority order, to which it should attempt to register. Each media termination point can register with only one Cisco Unified Communications Manager at a time.

Link:

http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_configuration_example09186a0

080b4f916.shtml

QUESTION NO: 63

Refer to the exhibit

```
IOS SAF Forwarder Config
router eigrp SAF
!
service-family ipv4 autonomous-system 1
!
  topology base
    external-client HQ_SAF
  exit-sf-topology
exit-service-family
!
!
service-family external-client listen ipv4 5050
external-client HQ_SAF
  username SAFUSER
  password SAFPASSWORD
  keepalive 3600000
```

The exhibit shows a SAF Forwarder configuration attached to a Cisco Unified Communications Manager.

Which minimum configuration for a Cisco Unified Communications Manager Express SAF Forwarder is needed to establish a SAF neighbor relationship with this SAF Forwarder?

A.

```
router eigrp SAF
!
service-family ipv4 autonomous-system 1
!
topology base
exit-sf-topology
exit-service-family
voice service saf
profile trunkroute 1
session protocol sip interface Loopback1 transport tcp port 5060
!
```

B.

```
router eigrp SAF
```

```
!  
service-family ipv4 autonomous-system 1  
!  
topology base  
exit-sf-topology  
exit- service-family  
!  
voice service saf  
profile trunk-route 1  
session protocol sip interface Loopback1 transport tcp port 5060  
!  
profile dn-block 1 alias-prefix 1972555  
pattern 1 type extension 4xxx  
!  
profile callcontrol 1  
dn-service  
trunk-route 1  
dn-block 1  
dn-block 2  
!  
channel 1 vrouter SAF asystem 1  
subscribe callcontrol wildcarded  
publish callcontrol 1  
!
```

C.

```
router eigrp SAF  
!  
service-family ipv4 autonomous-system 1  
!  
topology base  
exit-sf-topology  
exit-service-family  
!
```

D.

None of above configurations contain sufficient information.

Answer: C

Explanation: Incorrect answer: ABD

only following configuration is enough

```
router eigrp SAF
```

```
service-family ipv4 autonomous-system 1
```

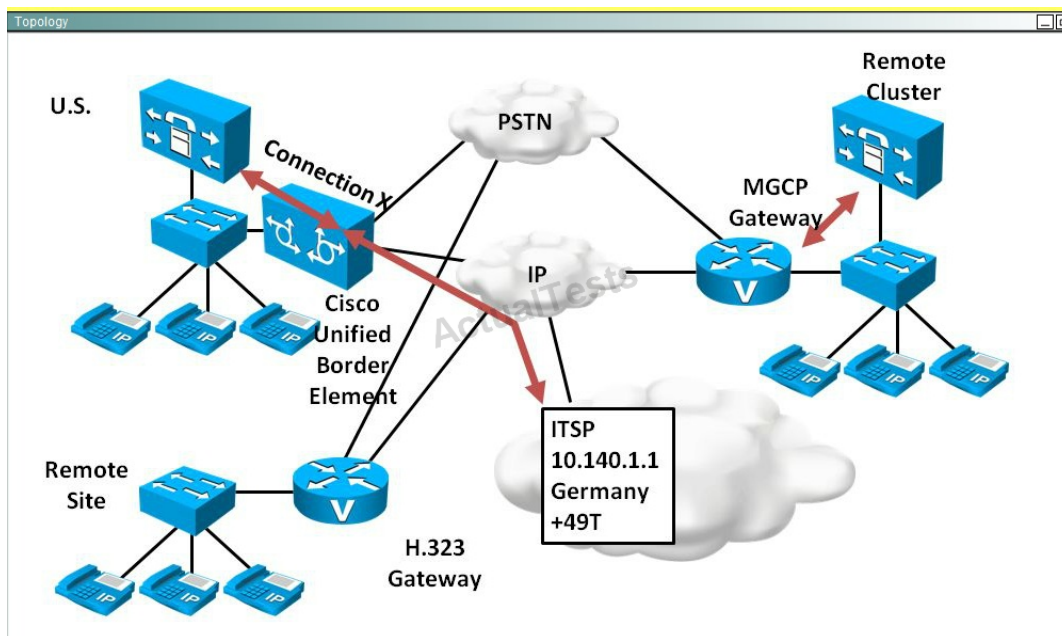
```
exit-service-family
```

link:

http://www.cisco.com/en/US/prod/collateral/iosswrel/ps6537/ps6554/ps6599/ps10822/whitepaper_c11-636604.html

QUESTION NO: 64

Refer to the following exhibits.



```
CUBE Config
!
hostname HQ_gateway
!
card type e1 0 0
enable password cisco123
!
no aaa new-model
network-clock-participate wic 0
!
ip source-route
ip cef
!
isdn switch-type primary-net5
!
voice-card 0
!
voice service voip
allow-connections sip to sip
!
controller E1 0/0/0
pri-group timeslots 1-12,16 service mgcp
!
interface Loopback0
ip address 10.1.111.1 255.255.255.0
!
interface GigabitEthernet0/0
no ip address
ip pim sparse-dense-mode
duplex auto
speed auto
media-type rj45
!
interface GigabitEthernet0/0.5
encapsulation dot1Q 5
ip address 10.1.5.1 255.255.255.0
ip pim sparse-dense-mode
!
interface GigabitEthernet0/0.10
encapsulation dot1Q 10
ip address 10.1.10.1 255.255.255.0
ip pim sparse-dense-mode
!
interface GigabitEthernet0/0.110
encapsulation dot1Q 110
ip address 10.1.110.1 255.255.255.0
ip helper-address 10.1.5.10
ip pim sparse-dense-mode
!
interface GigabitEthernet0/1
ip address 10.140.1.2 255.255.255.0
duplex auto
speed auto
media-type rj45
!
interface Serial0/0/0:15
no ip address
encapsulation hdlc
isdn switch-type primary-net5
isdn incoming-voice voice
no cdp enable
!
interface Serial0/1/0
no ip address
ip pim sparse-dense-mode
encapsulation frame-relay IETF
!
interface Serial0/1/0.101 point-to-point
ip address 10.12.1.1 255.255.255.0
ip pim sparse-dense-mode
snmp trap link-status
frame-relay interface-dlci 101
!
interface Serial0/1/0.102 point-to-point
ip address 10.13.1.1 255.255.255.0
snmp trap link-status
frame-relay interface-dlci 102
!
router eigrp 10
network 10.0.0.0
!
ip forward-protocol nd
!
voice-port 0/0/0:15
!
ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
ccm-manager config server 10.1.5.10
!
mgcp
mgcp call-agent 10.1.5.10 service-type mgcp version 0.1
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp package-capability prt-package
o mgcp package-capability rsa-package
o mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 ecm
mgcp rtp payload-type g726r16 static
mgcp behavior g729-variants static-pt

mgcp profile default

!
Hal-peer voice 1111 voip
session protocol sipv2
incoming called-number .

Hal-peer voice 222 voip
session protocol sipv2
destination-pattern +49T
session target ipv4:10.140.1.1

!
!
!
line con 0
line aux 0
line vty 0 4
!
end
```

Users in the U.S dial Germany by calling 9011 49 followed by the remaining digits. What would be

the most suitable configuration for Connection X?

- A.** Configure a SIP trunk to 10.140.1.1 and a SIP route pattern +49T in Cisco Unified Communications Manager.
- B.** Configure a SIP trunk to the Cisco Unified Border Element and route pattern +49T in Cisco Unified Communications Manager.
- C.** configure a SIP trunk to the Cisco Unified Border Element. Configure a translation pattern for 9011.49T using DDI Predot prefix + and CSS to point to a route pattern partition \+! which uses the SIP trunk.
- D.** Configure a SIP trunk to the ITSP. Configure a translation pattern for 9011.49T using DDI predot prefix + and CSS to point to a route pattern partition \+! which uses the SIP trunk.

Answer: C

Explanation: Incorrect answer: ABD

SIP trunks for public switched telephone network (PSTN) access are an important new access method for business collaboration. Service providers throughout the world offer SIP trunking as an alternative to traditional TDM (T1/E1) connections.

A discard digits instruction (DDI) removes a portion of the dialed digit string before passing the number on to the adjacent system. A DDI must remove portions of the digit string, for example, when an external access code is needed to route the call to the PSTN, but the PSTN switch does not expect that access code

Link: https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a03rp.html

QUESTION NO: 65

Refer to the exhibit.

MGCP Gateway

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). If field is empty in which case there is no prefix assigned.

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	+49	0	< None >
International Number	+	2	HQ_dng_pty_CSS
Unknown Number	Default	0	< None >
Subscriber Number	+4989	0	< None >

Pattern Definition

Pattern*

Partition

Description

Numbering Plan

Route Filter

☒ Urgent Priority

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Discard Digit Instructions

Calling Party Transformation Mask

Prefix Digits

Calling Line ID Presentation*

Calling Party Number Type*

A PSTN call arrived at the MGCP gateway. The calling number was received as 14087071222 with number set to type international. The HQ_clng__pty_CSS contains the HQ_clng_pty__Pt partition. Which caller ID is displayed on the IP phone?

- A. +087071222
- B. 14087071222
- C. 087071222
- D. 4087071222
- E. 14087071222

Answer: C

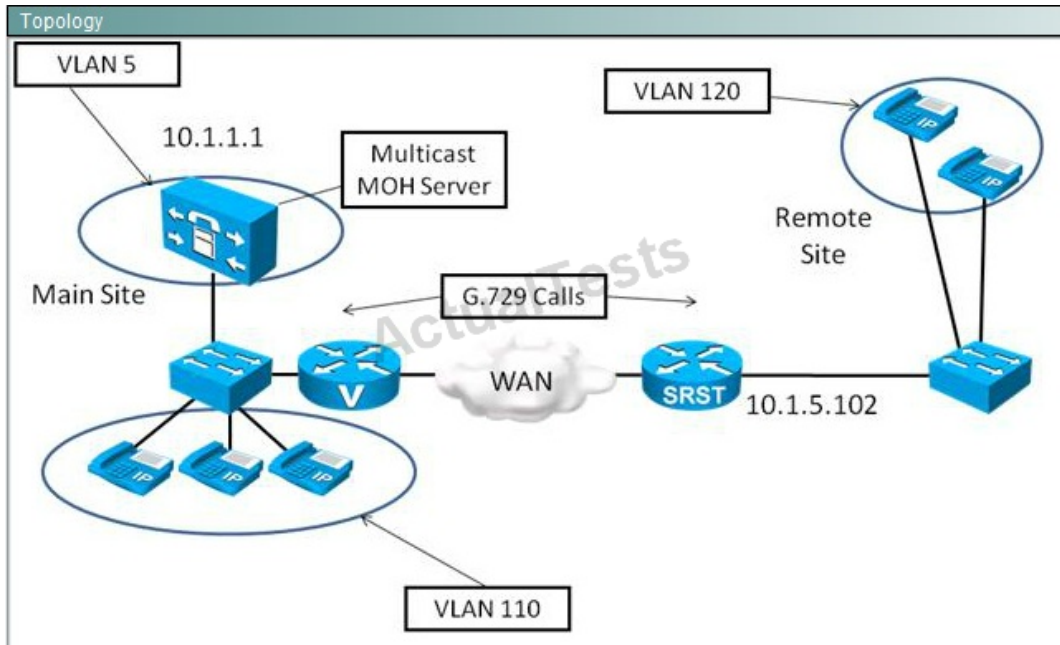
Explanation: Incorrect answer: ABDE

SIP trunks and MGCP gateways can support sending the international escape character, +, for calls. H.323 gateways do not support the +. QSIG trunks do not attempt to send the +. For outgoing calls through a gateway that supports +, Cisco Unified Communications Manager can send the + with the dialed digits to the gateway. For outgoing calls through a gateway that does not support +, the gateway strips the + when Cisco Unified Communications Manager sends the call information to the gateway.

Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/7_0_1/ccmfeat/fscallpn.html

QUESTION NO: 66

Refer to the exhibit.



MOH Server Config

Device Information

Registration: Registered with Cisco Unified Communications Manager 10.1.5.10

IP Address: 10.1.5.10

Host Server*: 10.1.5.10

Music On Hold Server Name*: MOH_2

Description: MOH_CUCM801Pub1

Device Pool*: Default

Location*: Hub_None

Maximum Half Duplex Streams*: 250

Maximum Multi-cast Connections*: 250000

Fixed Audio Source Device:

Use Trusted Relay Point*: Off

Run Flag*: Yes

Multi-cast Audio Source Information

☒ Enable Multi-cast Audio Sources on this MOH Server

Base Multi-cast IP Address*: 239.1.1.1

Base Multi-cast Port Number*: 16384 (Even numbers only)

Increment Multi-cast on*: ☐ Port Number ☒ IP Address

Selected Multi-cast Audio Sources

No.	Audio Source Name
1	SampleAudioSource

Save Reset Apply Config

Multicast MOH needs to be run from flash at the remote site. Which Cisco IOS configuration at the SRST router is correct?

A.

!

call-manager-fallback

ip source-address 10.1.5.102 port 2000

max-ephones 2

max-dn 8 dual-line

moh music-on-hold au

multicast moh 239.1.1.1 port 16384

B.

!

call-manager-fallback

ip source-address 10.1.5.102 port 2000

max-ephones 2

max-dn 8 dual-line

moh music-on-hold au

multicast moh 239.1.1.1 port 163t

C.

call-manager-fallback

ip source-address 10.1.120.1 port 2000 max-ephones 2

max-dn 8

moh music-on-hold.au

multicast moh 239.1.1.1 port 32767

D.

!

call-manager-fallback

ip source-address 10.1.5.102 port 2000

max-ephones 2

max-dn 8 dual-line

moh music-on-hold an

multicast moh 239.1.1.1 port 32767

Answer: A

Explanation: Incorrect answer: BCD

following configuration will required to Enable Multicast MOH on the Cisco SRST Gateway

1.

ccm-manager music-on-hold

2.

interface loopback *number*

3.

ip address *ip-address mask*

4.

exit

5.

interface fastethernet *slot/port*

6.

call-manager-fallback

9.

ip source-address *ip-address* [port *port*]

10.

max-ephones *max-phones*

11.

max-dn *max-directory-number*

12.

moh *filename*

13.

multicast moh *multicast-address* port *port* [route *ip-address-list*]

14.

exit

Link:

http://www.cisco.com/en/US/products/sw/iosswrel/ps5207/products_feature_guide09186a00802d1c31.html#wp1046789

Gateway Status				
	Device Name ^	Description	Device Pool	Device Type
H.323	10.1.120.1		Default	H.323 Gateway
HQ		HQ		Cisco 3825

QUESTION NO: 67

Refer to the exhibit.

ActualTests

```
interface Vlan1
  no ip address
  shutdown
!
interface Vlan20
  ip address 10.1.20.1 255.255.255.0
!
interface Vlan120
  ip address 10.1.120.1 255.255.255.0
  ip helper-address 10.1.5.2
  ip pim sparse-dense-mode
  h323-gateway voip bind srcaddr 10.1.120.1
!
router eigrp 10
  network 10.0.0.0
  no auto-summary
!
ip forward-protocol nd
!
!
no ip http server
!
control-plane
!
!
!
voice-port 0/0/0:23
  translation-profile incoming pstn-in
  translation-profile outgoing srst
!
```

```
ccm-manager fax protocol cisco
!
mgcp fax t38 ecm
!
!
dial-peer voice 911 pots
  destination-pattern 911
  port 0/0/0:23
  forward-digits all
!
dial-peer voice 9911 pots
  destination-pattern 9911
  port 0/0/0:23
  forward-digits all
!
```

```
dial-peer voice 123 pots
```

```
prefix 877
!
dial-peer voice 888 pots
 destination-pattern 91888.....
 port 0/0/0:23
 prefix 888
!
dial-peer voice 11 pots
 corlist outgoing ldPt
 destination-pattern 91[2-9]..[2-9].....
 port 0/0/0:23
!
dial-peer voice 3900 voip
 destination-pattern 3500
 session target ipv4:10.1.5.3
 dtmf-relay h245-alphanumeric
 no vad
!
!
gateway
 timer receive-rtp 1200
!
!
gatekeeper
 shutdown
!
!
call-manager-fallback
 max-conferences 8 gain -6
 transfer-system full-consult
 ip source-address 10.1.120.1 port 2000
!
max-ephones 4
max-dn 8 dual-line
after-hours block pattern 1 91900 7-24
voicemail 916506032000
call-forward busy 916506032000
call-forward noan 916506032000 timeout 7
moh music-on-hold.au
multicast moh 239.1.1.1 port 16384 route 10.1.20.1 10.1.120.1
!
!
line con 0
 location CIEV2-GA-LAB04, SJ
 exec-timeout 0 0
 logging synchronous
line aux 0
line vty 0 4
 exec-timeout 0 0
 password cisco123
 login
!
scheduler allocate 20000 1000
end
```

The H.323 Gateway is showing status "unknown". Which statement is true?

- A. The gateway must be reset in Cisco Unified Communications Manager.
- B. The no gateway command followed by the gateway command must be issued in Cisco IOS.
- C. The mgcp commands must be removed.
- D. H.23 gateways do not register with Cisco Unified Communications Manager H.323 gateways always show status "Unknown".
- E. VUAN 1 20 may be down and so the H.323 gateway appears offline to the Cisco Unified Communications Manager

Answer: D

Explanation: Incorrect answer: ABCE

After a gateway is registered with Cisco Unified Communications Manager, gateway registration status may display in Cisco Unified Communications Manager Administration as unknown.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucmbe/admin/8_6_1/ccmcfg/b06gtway.html

QUESTION NO: 68

Which Cisco IOS command is used for internal SAF Clients to check SAF learned routes?

- A. show eigrp address-family ipv4 saf
- B. show voice saf routes
- C. show voice saf detail
- D. show eigrp service-family ipv4 saf
- E. show voice saf dnDb all

Answer: E

Explanation: Incorrect answer: ABCD

Router# show voice saf dnDb all

Total no. of patterns in db/max allowed : 1/6000

Patterns classified under dialplans (private/global) : 0/1

Informational/Error stats -

Patterns w/ invalid expr detected while add : 0

Patterns duplicated under the same instance : 0

Patterns rejected overall due to max capacity : 0

Attempts to delete a pattern which is invalid : 0

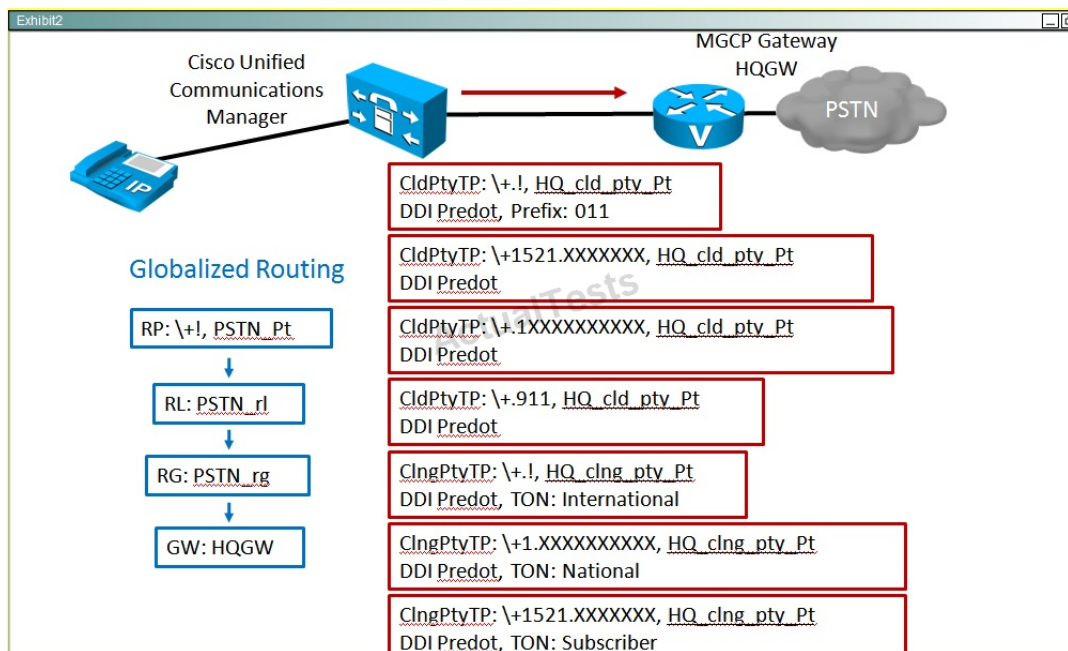
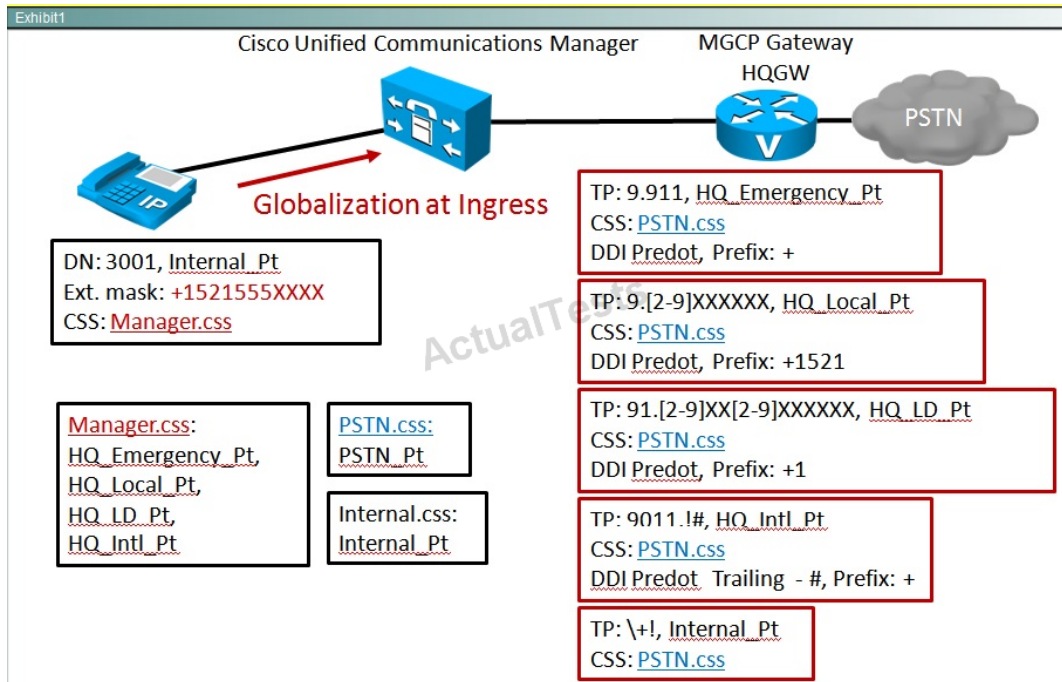
Last successful DB update @ 2009:12:14 15:42:45:967

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/feature/guide/SAF_FeatureModule.html#wp1201815

QUESTION NO: 69

Refer to the following exhibit.



The MGCP gateway has the following configurations:

called party transformation CSS HQ_cld_pty CSS (partition=HQ_cld_pty.Pt) call.ng party transformation CSS HQ_clng_pty CSS (partition=HQ_clng_pty Pt)

All translation patterns have the check box "Use Calling Party's External Phone Number Mask" enabled.

When the IP phone at extension 3001 places a call to 9011 49403021 56001# what is the resulting called and calling number that is sent to the PSTN?

- A.** The called number is 01 1 49403021 56001. The calling number will be 5553001 and number type set to subscriber.
- B.** The called number is 011 49403021 56001. The calling number will be 5215553001 and number type set to national.
- C.** The called number is 4940302156001 with number type set to international. The calling number will be 5215553001 and number type set to national.
- D.** The called number is +49403021 56001 with number type set to international. The calling number will be 5215553001 and number type set to subscriber.

Answer: A

Explanation: Incorrect answer: BCD

Check the check box "Use Calling Party's External Phone Number Mask" if you want the full, external phone number to be used for calling line identification (CLID) on outgoing calls. You may also configure an External Phone Number Mask on all phone devices.

Link:

http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_tech_note09186a00805b6f33.shtml

QUESTION NO: 70

Refer to the exhibit

Hosted DN Pattern	
Hosted DN Pattern Info	
Hosted Pattern*	2XXX
Description	
Hosted DN Group*	HQ_DN
PSTN Failover Strip Digits	0
PSTN Failover Prepend Digits	+498950555
☐ Use HostedDN as PSTN Failover	

Hosted DN Group	
Hosted DN Group Info	
Name*	HQ_DN
Description	
PSTN Failover Strip Digits	0
PSTN Failover Prepend Digits	+498953121
☐ Use HostedDN as PSTN Failover	

When the Cisco Unified Communications Manager advertises the Hosted DN Pattern, which pattern would be advertised?

- A. 2XXX and the T0D1D will be 0:+498950555
- B. 2XXX and the ToDID will be 0:+4989531 21
- C. 4989S05552XXX and the ToDiD will be 0:
- D. + 4989631 21 2XXX and the ToDiD will be 0:
- E. Both +4989505552XXXand +4989531 21 2XXX will be advertised with ToDID of 0:

Answer: A

Explanation: Incorrect answer: BCDE

PSTN failover prepend digit is +498950555

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_0_2/ccmfeat/fscallcontroldiscovery.html

QUESTION NO: 71

If your IP telephony administrator asks you to configure a local zone for your dial plan to control the volume of calls between two end points in a centralized multisite environment, which two types of Call Admission Control can be implemented? (Choose two.)

- A. locations based
- B. automated alternate routing
- C. gatekeeper based
- D. SRST
- E. Cisco Unified Communications Manager based

Answer: A,B

Explanation: Incorrect answer: CDE

Location-based call admission control (CAC) manages WAN link bandwidth in Cisco Unified Communications Manager. Automated alternate routing (AAR) provides a mechanism to reroute calls through the PSTN or other network by using an alternate number.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a02cac.html#wp1067747

QUESTION NO: 72

Which device is needed to integrate H.320 into the Cisco video solution? (Source: Introducing IP Video Telephony)

- A. video gateway
- B. MGCP gateway
- C. H.323 gatekeeper
- D. MCU

Answer: C

Explanation: Incorrect answer: ABD

As with H.323 MCUs, H.320 gateways are provisioned in Cisco Unified CallManager as H.323 gateways, and then route patterns are configured to extend calls to these devices.

Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/4x/42video.html#wp1046523

QUESTION NO: 73

Which statement about enrollment in the IP telephony PKI is true? (Source: Understanding Cisco IP Telephony Authentication and Encryption Fundamentals)

- A. CAPF enrollment supports the use of authentication strings.
- B. The CAPF itself has to enroll with the Cisco CTL client.
- C. LSCs are issued by the Cisco CTL client or by the CAPF.
- D. MICs are issued by the CAPF itself or by an external CA.

Answer: A

Explanation: Incorrect answer: BCD

The CAPF enrollment process is as follows:

1. The IP phone generates its public and private key pairs.

2. The IP phone downloads the certificate of the CAPF and uses it to establish a TLS session with the CAPF.
3. The IP phone enrolls with the CAPF, sending its identity, its public key, and an optional authentication string.
4. The CAPF issues a certificate for the IP phone signed with its private key.
5. The CAPF sends the signed certificate to the IP phone.

Link: <http://my.safaribooksonline.com/book/certification/cipt/9781587052613/understanding-cisco-ip-telephony-authentication-and-encryption-fundamentals/584>.

QUESTION NO: 74

Refer to the exhibit.

```
controller T1 1/0
framing esf
linecode b8zs
pri-group timeslots 1-6,24

interface Serial1/0:23
no ip address
encapsulation hdlc
isdn switch-type primary-ni
isdn incoming-voice voice
no cdp enable

interface Vlan120
ip address 10.2.120.1 255.255.255.0
h323-gateway voip bind srcaddr 10.2.120.1

voice-port 1/0:23

dial-peer voice 123 pots
incoming called-number .
direct-inward-dial

dial-peer voice 23000 voip
destination-pattern 2...
session target ipv4:10.1.5.11
dtmf-relay h245-alphanumeric
codec g711ulaw
no vad
```

IT shows an H.323 gateway configuration in a Cisco Unified Communications Manager environment. An inbound PSTN call to this H.323 gateway fails to connect to IP phone extension 2001. The PSTN user hears a reorder tone. Debug isdn q931 on the H.323 gateway shows the correct called-party number as 5015552001. Which two configuration changes can correct this issue? (Choose two.)

- A. Add port 1/0:23 to dial-peer voice 123 pots.
- B. Ensure that the Significant Digits for inbound calls on the H.323 gateway configuration is 4.
- C. Add a voice translation profile to truncate the number from 10 digits to 4 digits. Apply the voice translation profile to the Voice-port. The configuration field "Significant Digits for inbound calls" is left at default (All).

D. Add the command h323-gateway voip id on interface vlan120.

E. Change the destination-pattern on the dial-peer voice 23000 VoIP to 501501? and change the Significant Digits for inbound calls to 4.

Answer: B,E

Explanation: Incorrect answer: ACD

Choose the number of significant digits to collect, from 0 to 32. Cisco Unified Communications Manager counts significant digits from the right (last digit) of the number that is called.

Link: http://cisco.biz/en/US/docs/voice_ip_comm/cucmbe/admin/8_6_1/ccmcfg/b06trunk.html

QUESTION NO: 75

Which two have to be defined in the Forward All field? (Choose two.) (Source: Preventing Toll Fraud)

A. calling search space

B. destination

C. partition

D. hunt list

Answer: A,B

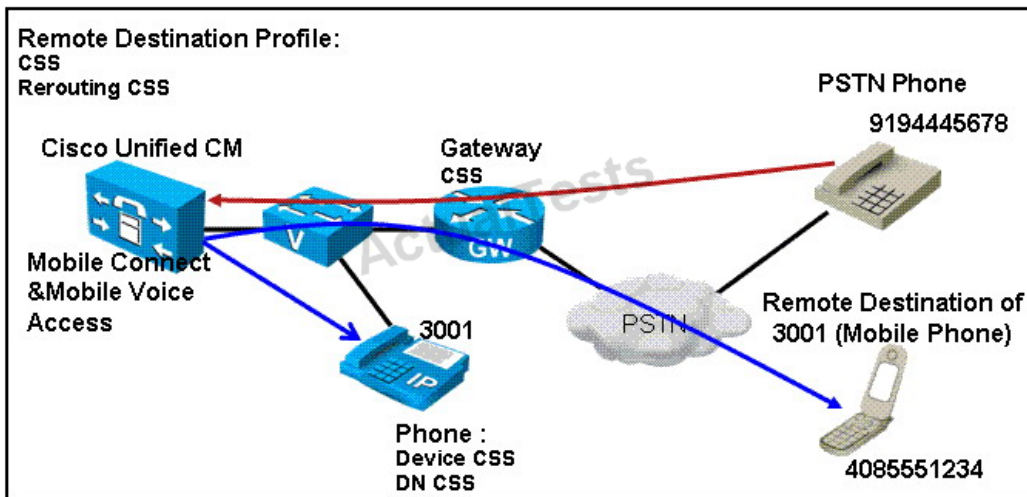
Explanation: Incorrect answer: CD

Destination—This setting indicates the directory number to which all calls are forwarded. Use any dialable phone number, including an outside destination.

Calling Search Space—This setting applies to all devices that are using this directory number.

QUESTION NO: 76

Refer to the exhibit. With the Mobile Connect feature configured, when the PSTN phone calls the Enterprise user at extension 3001, the Enterprise user's mobile phone does not ring. Which CSS is responsible for ensuring that the correct partitions are accessed when calls are sent to the Enterprise user's mobile phone?



- A. the gateway CSS
- B. the Phone Device CSS
- C. the Remote Destination Profile CSS
- D. the Remote Destination Profile Rerouting CSS
- E. the Phone Line (DN)CSS

Answer: D

Explanation: Incorrect answer: ABCE

Ensure that the gateway that is configured for routing mobile calls is assigned to the partition that belongs to the Rerouting Calling Search Space. Cisco Unified Communications Manager determines how to route calls based on the remote destination number and the Rerouting Calling Search Space.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucmbe/admin/8_6_1/ccmfeat/fsmobmgr.html

QUESTION NO: 77

Refer to the exhibit. You have configured transcoder resources in both an IOS router and a Cisco Unified Communications Manager. When you review the configurations in both devices the IP addresses and transcoder names are correct, but the transcoder is failing to register with the Cisco Unified Communications Manager. Which command needs to be edited to allow the transcoder to register properly?

```
voice-card 0
  dspfarm
  dsp services dspfarm

sccp local FastEthernet0/0
sccp ccm 10.1.1.1 identifier 1 version 6.0
sccp

sccp ccm group 1
  associate ccm 2 priority 1
  associate profile 1 register HQ_XCODER

dsp farm profile 1 transcode
  codec g711ulaw
  codec g711alaw
  codec g729ar8
  codec g729abr8
  maximum sessions 2
  associate application SCCP
  no shutdown
```

- A. The associate profile and dsp farm profile numbers need to match associate ccm 2 command.
- B. The associate ccm 2 priority 1 command needs to be changed so the ccm value matches identifier 1 in the sccp ccm 10.1.1.1 command.
- C. The sccp ccm group number needs to match the associate ccm 2 command.
- D. The maximum sessions command must match the number of codecs configured under the dsp farm profile.
- E. The sccp ccm group number must match the voice-card number.

Answer: B

Explanation: Incorrect answer: ACDE

The value of the IP address should match the IP address in the ip source-address command

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmevtrns.html

QUESTION NO: 78

Which statement is correct about AAR?

- A.** The end users sees, "Network Congestion Rerouting?" but AAR is otherwise transparent to the end user and works without user intervention.
- B.** AAR will display "not enough bandwidth" on the IP phone while it reroutes the call.
- C.** AAR allows calls to be rerouted because of insufficient Cisco Unified Border Element controlled bandwidth to an ITSP.
- D.** AAR allows calls to be rerouted due to insufficient gatekeeper controlled IP WAN bandwidth.

Answer: A

Explanation: Incorrect answer: BCD

Automated alternate routing (AAR) provides a mechanism to reroute calls through the PSTN or other network by using an alternate number when Cisco Unified Communications Manager blocks a call due to insufficient location bandwidth. With automated alternate routing, the caller does not need to hang up and redial the called party.

Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucmbe/admin/8_6_1/ccmcfg/b03aar.html

QUESTION NO: 79

The relationship between a Region and a Location is that the Region codec parameter is used between a Region and its configured Locations.

- A.** TRUE
- B.** FALSE

Answer: B

Explanation: Locations work in conjunction with regions to define the characteristics of a network link. Regions define the type of compression (G.711, G.722, G.723, G.729, GSM, or wideband) that is used on the link, and locations define the amount of available bandwidth for the link

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a02cac.html#wp1033331

QUESTION NO: 80

An update of the configuration using the Cisco CTL client not needed when _____.(Source. Configuring Cisco IP Telephony Authentication and Encryption)

- A.** a Cisco Unified CallManager has been removed

- B. an LSC of the IP phone is upgraded
- C. a security token is added to the system
- D. an IP address of the Cisco TFTP server has been changed

Answer: B

Explanation: Incorrect answer: ACD

The CTL file contains entries for the following servers or security tokens:

-

System Administrator Security Token (SAST)

-

Cisco CallManager and Cisco TFTP services that are running on the same server

-

Certificate Authority Proxy Function (CAPF)

-

TFTP server(s)

-

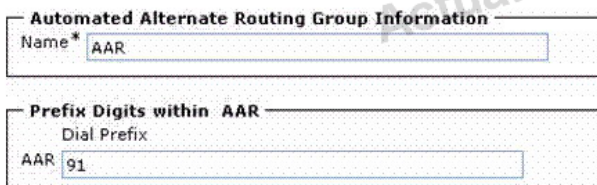
ASA firewall

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/security/8_6_1/secugd/secuauth.html#wp1028878

QUESTION NO: 81

Refer to the exhibit. A user in RTP calls a phone in San Jose during congestion with Call Forward No-Bandwidth (CFNB) configured to reach cell phone 4085550150. The user in RTP sees the message "Not Enough Bandwidth" on their phone and hears a fast busy tone. Which two conditions can correct this issue? (Choose two.)



Automated Alternate Routing Group Information

Name* AAR

Prefix Digits within AAR

Dial Prefix

AAR 91

- A. The calling phone (RTP) needs to have AAR Group value of AAR under the AAR Settings.
- B. The called phone (San Jose) needs to have AAR Group value of AAR under the AAR Settings.

- D. The calling phone (RTP) needs to have the AAR destination mask of 4085550150 configured under the AAR Settings.
- E. The called phone (San Jose) needs to have the AAR destination mask of 914085550150 configured under the AAR Settings.
- F. The called phone (San Jose) needs to have the AAR destination mask of 4085550150 configured under the AAR Settings.

Answer: B,F

Explanation: Incorrect answer: ACDE

Explanation: Automated alternate routing (AAR) provides a mechanism to reroute calls through the PSTN or other network by using an alternate number when Cisco Unified Communications Manager blocks a call due to insufficient location bandwidth. With automated alternate routing, the caller does not need to hang up and redial the called party.

Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucmbe/admin/8_6_1/ccmcfg/b03aar.html

QUESTION NO: 82

Which statement is not true about GARP? (Source: Hardening the IP Phone)

- A. GARP attacks require access to the target LAN or VLAN.
- B. GARP can be used for a man-in-the-middle attack.
- C. GARP is normally used for HSRP.
- D. GARP can be disabled at Cisco IP phones.

Answer: C

Explanation: Incorrect answer: ABD

GARP (Gratuitous ARP) announce the presence of IP Phone on the network.

Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/security/4_0_1/secuphne.html

QUESTION NO: 83

Which two statements describe RSVP-enabled locations-based CAC? (Choose two.)

- A. RSVP can be enabled selectively between pairs of locations.
- B. Using RSVP for CAC simply allows admitting or denying calls based on a logical configuration that is ignoring the physical topology.
- C. RSVP is topology aware, but only works with full mesh networks.
- D. An RSVP agent is a Media Termination Point that the call has to flow through.

E. RSVP and RTP are used between the two endpoints.

Answer: A,D

Explanation: Incorrect Answer: BC

The RSVP policy that is configured for a location pair overrides the default interlocation RSVP policy that configure in the Service Parameter Configuration window. RSVP supports audio, video, and data pass-through. Video data pass-through allows video and data packets to flow through RSVP agent and media termination point devices

Link:

https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a02rsvp.html#wp1070214

QUESTION NO: 84

You are the Cisco Unified Communications Manager in Certpaper.com. You use a remote site MGCP gateway to provide redundancy when connectivity to the central Cisco Unified Communications Manager cluster is lost. How to enable IP phones to establish calls to the PSTN when they have registered with the gateway? (Choose three.)

- A. POTS dial peers must be added to the gateway to route calls from the IP phones to the PSTN.
- B. The default service must be enabled globally.
- C. The command ccm-manager mgcp-fallback must be configured.
- D. COR needs to be configured to disallow outbound calls.

Answer: A,B,C

Explanation: Incorrect answer: D

Class of restriction: Cisco Unified Communications Manager Business Edition 3000 supports class of service (CoS) with respect to geographic reach as follows:

- Campus
- Local
- National
- International
- Emergency services
- Call waiting
- Default ringtones
- MoH
- Speed dials: Single-button, not BLF

Link:

http://www.cisco.com/en/US/prod/collateral/voicesw/ps6788/vcallcon/ps11370/data_sheet_c78-651909.html

QUESTION NO: 85

Which two configurations provide the best SIP trunk redundancy with Cisco Unified Communications Manager? (Choose two.)

- A. Configure all SIP trunks with DNS SRV
- B. Configure all SIP trunks with Cisco Unified Border Element
- C. Configure all SIP trunks to point to a SIP gateway
- D. Configure SIP trunks to be members of route groups and route lists
- E. Configure all SIP trunks to allow TCP ports 5060
- F. Configure all SIP trunks to point to a gatekeeper through SIP to H.323 gateway

Answer: A,D

Explanation: Incorrect answer: BCEF

For SIP trunks, Cisco Unified Communications Manager supports up to 16 IP addresses for each DNS SRV and up to 10 IP addresses for each DNS host name. The order of the IP addresses depends on the DNS response and may be identical in each DNS query. The OPTIONS request may go to a different set of remote destinations each time if a DNS SRV record (configured on the SIP trunk) resolves to more than 16 IP addresses, or if a host name (configured on the SIP trunk) resolves to more than 10 IP addresses. Thus, the status of a SIP trunk may change because of a change in the way a DNS query gets resolved, not because of any change in the status of any of the remote destinations.

Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a08sip.html

QUESTION NO: 86

When an external call is placed from Ajax, they would like the ANI that is sent to the PSTN to be the main number, not the extension. For domestic calls, they would like 10 digits sent; for international calls, they would like to send the country code 1 and the 10 digits. How can this be accomplished?

- A. Add a translation pattern to the dial peers in the gateway that adds the appropriate digits to the outgoing ANI.
- B. In the external call route patterns, set the external phone number mask to the main number. Use 10 digits in the domestic route pattern and 1 followed by the main number digits in the international route patterns.
- C. Use a calling party transform mask for each route group in the corresponding route list

configuration. Set the explicit 10-digit main number for domestic calls and 1 followed by the main number for the international route patterns.

D. In the directory number configurations, set the prefix digits field to the country code and the 10 digits of the main number. This will be truncated to the 10-digit number for domestic calls and sent out in its entirety for international calls.

Answer: C

Explanation: Incorrect answer: ABD

calling party transformation mask value is Valid entries for the NANP include the digits 0 through 9; the wildcard characters X, asterisk (*), and octothorpe (#); and the international escape character +.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucmbe/admin/8_6_1/ccmcfg/b03trpat.html

QUESTION NO: 87

The relationship between a Region and a Location is that the Region codec parameter is combined with Location bandwidth when communicating with other Regions.

A. FALSE

B. TRUE

Answer: A

Explanation: Locations work in conjunction with regions to define the characteristics of a network link. Regions define the type of compression (G.711, G.722, G.723, G.729, GSM, or wideband) that is used on the link, and locations define the amount of available bandwidth for the link

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a02cac.html#wp1033331

QUESTION NO: 88

Video calls using 384 kbps need to be supported across a gatekeeper-controlled trunk. What value should be entered into the gatekeeper to support this bandwidth? (Source: Configuring Cisco Unified Video Advantage)

A. 768 kbps

- B. 384 kbps
- C. 512 kbps
- D. 192 kbps

Answer: B

Explanation: Incorrect answer : ACD

Explanation : A 384-kb/s video call may comprise G.711 at 64 kb/s (for audio) plus 320 kb/s (for video). This sum does not include overhead. If the audio codec for a video call is G.729 (at 24 kb/s), the video rate increases to maintain a total bandwidth of 384 kb/s.

Link:

http://www.ciscosystems.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a08video.html#wp1059726

QUESTION NO: 89

Enabling authentication and encryption for CTI, JTAPI, and TAPI applications requires which two tasks? (Choose two.) (Source: Configuring Cisco IP Telephony Authentication and Encryption)

- A. Enter the encryption key into the application.
- B. Set up an IPsec association between the application and Cisco Unified CallManager.
- C. Configure related security parameters in the CTI, JTAPI, and TAPI application.
- D. Add the application user or end users to the Standard CTI Secure Connection user group, Standard CTI Allow Reception of SRTP Key Material user group, and Standard CTI Enabled user group.

Answer: C,D

Explanation: Incorrect answer: AB

You must also add the application users or the end users to the Standard CTI Secure Connection user group in Cisco Unified Communications Manager Administration to enable TLS for the application. After you add the user to this group and install the certificate, the application ensures that the user connects via the TLS port.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/security/8_6_1/secugd/secucti.html#wp1166397

QUESTION NO: 90

Which two statements about symmetric encryption are true? (Choose two.) (Source:

Understanding Cryptographic Fundamentals)

- A. With symmetric encryption, the encryption key equals the decryption key.
- B. Symmetric encryption is commonly used to sign asymmetric keys.
- C. Symmetric encryption is a good choice for real-time encryption of bulk data.
- D. Symmetric encryption uses asymmetric keys.

Answer: A,C

Explanation: Incorrect answer: BD

There are two basic techniques for encrypting information: symmetric encryption (also called secret key encryption) and asymmetric encryption (also called public key encryption.) in symmetric key As long as both sender and recipient know the secret key, they can encrypt and decrypt all messages that use this key. A public key is made freely available to anyone who might want to send you a message. A second, private key is kept secret, so that only you know it.

Link: <http://support.microsoft.com/kb/246071>

QUESTION NO: 91

You are the Cisco Unified Communications Manager in Certpaper.com. After you add the Tcl paramspace command, the application can_____.

- A. set aside memory for application variables
- B. access the data on an internal server
- C. access the data on an external server
- D. share parameters between different call applications

Answer: D

Explanation: Incorrect answer: ABC

application

service debitcard tftp://15.5.27.11/app_debitcard.2.0.2.8.tcl

paramspace english index 1

paramspace english language en

paramspace english location tftp://15.5.27.11/prompts/en/

param pid-len 4

paramspace english prefix en

param uid-len 6

Link:

http://www.cisco.com/en/US/products/sw/voicesw/ps5640/products_configuration_example09186a

QUESTION NO: 92

The following exhibit shows configs for H.323 gateway. You have been asked to implement TEHO from a remote branch office with area code 301 to the HQ office with area code 201 using Cisco Unified Communications Manager. The remote office has an MGCP gateway and the HQ office has an H.323 gateway. Once the call arrives at the HQ, it should break out to the PSTN as a seven-digit local call. Which statement about the route pattern is true?

```
dial-peer voice 7 pots
 destination-pattern 9[2-9].....
 port 1/0:23
!
dial-peer voice 11 pots
 destination-pattern 91[2-9]..[2-9].....
 prefix 1
 port 1/0:23
```

- A. route pattern should be 91201.[2-9]XXXXXX with Discard Digit as Predot and Prefix 9
- B. route pattern should be 91201.[2-9]XXXXXX with Discard Digit as Predot
- C. route pattern should be 91201.[2-9]XXXXXX
- D. route pattern should be 9.1201[2-9]XXXXXX with Discard Digit as Predot
- E. route pattern should be 9.1201[2-9]XXXXXX with Discard Digit as Predot and Prefix 9

Answer: A

Explanation: Incorrect answer: BCDE

Destination pattern is 91, HQ office area code is 201 .

QUESTION NO: 93

What is the default value for the Drop Ad Hoc Conference service parameter? (Source:Preventing Toll Fraud)

- A. Never
- B. When No On-Net Parties Remain in the Conference
- C. When No Off-Net Parties Remain in the Conference
- D. Drop Ad Hoc Conference When Creator Leaves

Answer: A

Explanation: Incorrect answer: BCD

To use the additional functionality that advanced ad hoc conferencing provides, Cisco recommends that you set this service parameter to Never. Any other setting can result in unintentional termination of a conference.

Link:

https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a05confb.html

QUESTION NO: 94

When you configure Cisco Unified Communications Manager, you need to configure the router for Survivable Remote Site Telephony in case the Cisco Unified Communications Manager stops working. On which two factors would the number of IP phones and Directory Numbers that can register to the SRST router depend? (Choose two.)

- A. The protocol that is used in Cisco Unified Communications Manager
- B. Cisco Unified Communications Manager version
- C. Cisco IOS Software version
- D. WAN link bandwidth
- E. capacity of the Cisco Media Convergence Server
- F. router platform

Answer: C,F

Explanation:

QUESTION NO: 95

With Media Gateway Control Protocol configuration on the voice gateway, which three types of messages are involved in the call flow between the call agent and the voice gateway? (Choose three.)

- A. audit endpoint

- B. modify endpoint
- C. create connection
- D. delete notification
- E. restart in progress
- F. end connections

Answer: A,C,E

Explanation:

QUESTION NO: 96

Which trunk should you use in an H.323 gatekeeper-controlled network?

- A. H.323
- B. H.225
- C. SIP
- D. Intercluster
- E. MGCP FXO trunk
- F. MGCP T1/E1 trunk

Answer: B

Explanation:

QUESTION NO: 97

Which sign is prefixed to the number in global call routing?

- A. -
- B. +
- C. #
- D. @
- E. &
- F. *

Answer: B

Explanation:

QUESTION NO: 98

What are the two tasks that you must perform to configure the Service Advertisement Framework forwarder in Cisco Unified Communications Manager? (Choose two.)

- A. create VPN groups
- B. create VPN profiles
- C. create a new Service Advertisement Framework security profile
- D. set feature configuration parameters of Call Control Discovery
- E. configure Service Advertisement Framework forwarder information
- F. enable enterprise parameter for Service Advertisement Framework forwarder

Answer: C,E

Explanation:

QUESTION NO: 99

You are entering the description for the Service Advertisement Framework forwarder. Which three characters should you avoid entering in the description? (Choose three.)

- A. < >
- B. &
- C. %
- D. #
- E. \$
- F. @

Answer: A,B,C

Explanation:

QUESTION NO: 100

You are deploying a remote office setup that connects with Cisco Unity Communications Manager at a hub location. You have an available dedicated bandwidth of 20% from the 2-Mb/s WAN circuit for VoIP that supports a maximum of 17 calls. Which codec do you configure in Cisco Unity Communications Manager to achieve this?

- A. G.722
- B. G.711
- C. G.729
- D. iSAC
- E. GSM-FR

F. iLBC

Answer: C

Explanation:

QUESTION NO: 101

Which two formats are supported by Cisco Unified Communications Manager to stream music on hold? (Choose two.)

- A. G.711
- B. G.729
- C. G.723
- D. G.722
- E. G.728
- F. Advanced Audio Codec

Answer: A,B

Explanation:

QUESTION NO: 102

You have been asked to deploy Cisco Extension Mobility Cross Cluster for a distributed call processing environment. During the initial extension mobility login request, how does the visiting cluster determine if the user is a local user or a remote user?

- A. by using a third-party automatic provisioning tool to verify user ID
- B. by broadcasting a request to all clusters to verify the user type
- C. from user IDs that are created by default when the user logs in
- D. by using Extension Mobility Cross Cluster Session Initiation Protocol (SIP) trunks
- E. by verifying against the local database
- F. by verifying the visiting Trivial File Transfer Protocol

Answer: E

Explanation:

QUESTION NO: 103

Which action is performed by the Media Gateway Control Protocol gateway when it loses

connectivity with the primary and backup Cisco Unified Communications Manager servers?

- A. The gateway continues to make an attempt to connect to the backup Cisco Unified Communications Manager server.
- B. The gateway falls back to the H.323 protocol for further call processing.
- C. The gateway continues with the MGCP call processing without any interruption.
- D. The gateway waits for the primary Cisco Unified Communications Manager server to come alive.
- E. All MGCP call processing is interrupted until the Cisco Unified Communications Manager servers are online.
- F. The MGCP calls are queued up until the Cisco Unified Communications Manager servers are online.

Answer: B

Explanation:

QUESTION NO: 104

You need to verify if the Media Control Gateway Protocol gateway is enabled and active. Which command should you use for this purpose?

- A. show running-config
- B. show fallback-mgcp
- C. show gateway
- D. show ccm-manager fallback-mgcp
- E. show running-config gateway
- F. show fallback-mgcp ccm-manager

Answer: D

Explanation:

QUESTION NO: 105

You want to perform Media Control Gateway Protocol gateway maintenance. For this purpose, you disable Media Control Gateway Protocol gateway using the no mgcp command.

After you perform the maintenance, you want to enable the Media Control Gateway Protocol gateway. Which command should you use?

- A. enable mgcp
- B. mgcp

- C. mgcp enable
- D. mgcp yes
- E. activate mgcp
- F. mgcp active

Answer: B

Explanation:

QUESTION NO: 106

In a cluster-wide deployment, what is the maximum number of Service Advertisement Framework forwarders to which the Cisco Unified Communications Manager can connect?

- A. 1
- B. 2
- C. 3
- D. 4
- E. 6
- F. as many as are configured

Answer: B

Explanation:

QUESTION NO: 107

In a node-specific Service Advertisement Framework forwarder deployment model, what is the maximum number of Service Advertisement Framework forwarders that you can assign to a specific node?

- A. 1
- B. 2
- C. 3
- D. 4
- E. 5
- F. 6

Answer: B

Explanation:

QUESTION NO: 108

Which component of Cisco Unified Communications Manager is responsible for sending keepalive messages to the Service Advertisement Framework forwarder?

- A. Call Control Discovery requesting service
- B. Hosted DNS service
- C. Service Advertisement Framework client control
- D. Cisco Unified Communications Manager database
- E. Service Advertisement Framework-enabled trunk
- F. gatekeeper

Answer: C

Explanation:

QUESTION NO: 109

How many hosted DN patterns per cluster does Cisco Unified Communications Manager allow you to configure?

- A. 500
- B. 1000
- C. 1500
- D. 2000
- E. 2500
- F. 5000

Answer: D

Explanation:

QUESTION NO: 110

Which two actions are performed by the Call Control Discovery service after the local Cisco Unified Communications Manager loses its TCP connection with the primary and secondary Service Advertisement Framework? (Choose two.)

- A. Calls are routed to the PSTN gateway after the Call Control Discovery Learned Pattern IP Reachable Duration parameter expires.
- B. All learned patterns are purged from the local cache after the Call Control Discovery PSTN Failover Duration parameter expires.
- C. The Service Advertisement Framework forwarder contacts all the remaining Service

Advertisement Framework forwarders in the cluster.

D. All the remaining Service Advertisement Framework forwarders are notified for their learned patterns.

E. The Cisco Unified Communications Manager establishes a connection with the primary and secondary Service Advertisement Framework after the Learned Pattern IP Reachable Duration parameter expires.

F. Call Control Discovery immediately redirects all the calls to the PSTN gateway based on the learned patterns.

Answer: A,B

Explanation:

QUESTION NO: 111

Which task must you perform before deleting a transcoder?

A. Delete the dependency records.

B. Unassign it from a media resource group.

C. Use the Reset option.

D. Remove the device pool.

E. Remove the subunit.

F. Delete the common device configuration.

Answer: B

Explanation:

QUESTION NO: 112

How many active gatekeepers can you can define in a local zone?

A. 1

B. 2

C. 5

D. 10

E. 15

F. unlimited

Answer: A

Explanation:

QUESTION NO: 113

Which codec is recommended by Cisco for multisite music on hold deployment that spans through the WAN links?

- A. G.711
- B. G.723
- C. G.729
- D. G.722
- E. G.728
- F. G.725

Answer: C

Explanation:

QUESTION NO: 114

As a voice operations engineer, you need to standardize the firmware loads on a Cisco Unified Communications Manager cluster. You need to ensure that all the IP phones are loaded with device defaults load information. Which two locations can you verify to find the IP phones that have their firmware mismatched? (Choose two.)

- A. Cisco Unified OS Administration > Software Upgrades > TFTP File Management
- B. Cisco Unified CM Administration > Firmware Load Information
- C. Cisco Unified CM Administration > Device > Phone
- D. Cisco Unified OS Administration > Settings > Version
- E. Cisco Unified CM Administration > Device Defaults Configuration
- F. Cisco Unified Reporting > System Reports

Answer: B,F

Explanation:

QUESTION NO: 115

Which interface can you use to configure alarms and traces in Cisco Unified Communications Manager?

- A. Cisco Unified Reporting
- B. Cisco Unified Serviceability
- C. Cisco Unified Communications Manager Administration
- D. disaster recovery system

- E. Cisco Unified OS Administration
- F. real-time monitoring

Answer: B

Explanation:

QUESTION NO: 116

Which solution is needed to enable presence and extension mobility to branch office phones during a WAN failure?

- A. SRST with MGCP fallback
- B. SRST without MGCP fallback
- C. Cisco Unified Communications Manager Express in SRST mode
- D. SRST with VoIP dial peers to Cisco Unified Communications Manager Express

Answer: C

Explanation:

QUESTION NO: 117

Which command displays the detailed configuration of all the Cisco Unified IP phones, voice ports, and dial peers of the Cisco Unified SRST router?

- A. show call-manager-fallback all
- B. show dial-peer voice summary
- C. show ephone summary
- D. show voice port summary

Answer: A

Explanation:

QUESTION NO: 118

Which option configures the secondary dial tone option for SRST mode to let the users hear the dial tone for PSTN calls?

- A. voice service voip

secondary dialtone 0

B. call-manager-fallback

secondary dialtone 0

C. dial-peer voice 1 pots

secondary dialtone 0

D. ccm-manager secondary dialtone 0

Answer: B

Explanation:

QUESTION NO: 119

During device failover to the secondary Cisco Unified Communications Manager server, how does the phone recognize that the primary server is back?

A. The secondary server keeps sending keepalive message to the primary server and when it succeeds, it unregisters the phones to force them to register to the primary.

B. When the primary server goes online, it sends out an "ALIVE" message via broadcast so that the phones re-register.

C. The phones never re-register with the primary server until the active (secondary) one goes down.

D. The phone sends keepalive messages to the primary server frequently and when it succeeds, the phone re-registers with it.

Answer: D

Explanation:

QUESTION NO: 120

Which statement about the function of the "+" symbol in the E.164 format is true?

A. The "+" symbol represents the international country code.

B. The "+" symbol represents the international call prefix.

C. The "+" symbol matches the preceding element one or more times.

D. The "+" symbol matches the preceding element zero or one time.

Answer: B

Explanation:

QUESTION NO: 121

What is an advantage of TEHO?

- A. TEHO implemented with ISRs eliminates PSTN toll charges.
- B. TEHO implemented with ISRs can reduce PSTN toll charges.
- C. TEHO implemented with AAR reduces toll charges.
- D. TEHO implemented with CFUR reroutes calls.

Answer: B

Explanation:

QUESTION NO: 122

Which two statements about international multisite dial plans are true? (Choose two.)

- A. TEHO is legal in all countries.
- B. TEHO is legal in most countries, but illegal in others.
- C. TEHO reduces WAN utilization.
- D. TEHO utilizes WAN links.

Answer: B,D

Explanation:

QUESTION NO: 123

Which gateway does the Cisco Unified Communications Manager control all call activity?

- A. SIP
- B. MGCP
- C. H.323
- D. Media

Answer: B

Explanation:

QUESTION NO: 124

Which device is used to connect to the H.323 gatekeeper?

- A. H.323 gateway
- B. SIP trunk
- C. H.323 trunk
- D. MGCP gateway

Answer: C

Explanation:

QUESTION NO: 125

The corporate WAN has been extended to two newly acquired sites and it includes gatekeeper support. Each site has a Cisco CallManager and an H.323 gateway that allows connection to the PSTN. Which connection method is best for these two new customers?

- A. H.225 trunk (gatekeeper-controlled)
- B. intercluster trunk (non-gatekeeper controlled)
- C. SIP trunk
- D. intercluster trunk (gatekeeper-controlled)

Answer: D

Explanation:

QUESTION NO: 126

Which two are gatekeeper-controlled trunk options that support gatekeeper call administration control? (Choose two.)

- A. H.323
- B. H.245
- C. H.225
- D. intercluster
- E. intracluster

Answer: C,D

Explanation:

QUESTION NO: 127

What is the purpose of the local route group?

- A. minimize PSTN costs
- B. help in the selection of the PSTN egress gateway
- C. eliminate the need for a route list
- D. allow manipulation of digits at the cost point to egress

Answer: B

Explanation:

QUESTION NO: 128

Which action configures PSTN backup for calls that are rejected by the gatekeeper CAC?

- A. Configure AAR in Cisco Unified Communications Manager.
- B. Configure CFUR in Cisco Unified Communications Manager.
- C. Configure a route pattern, a route list, and route groups to a trunk and a gateway in Cisco Unified Communications Manager.
- D. Configure a route pattern to a gateway in Cisco Unified Communications Manager.

Answer: C

Explanation:

QUESTION NO: 129

Cisco Unified Communications Manager is configured with CAC for a maximum of 10 voice calls.

Which action routes the 11th call through the PSTN?

- A. Configure an SIP trunk to the ISR.
- B. Configure Cisco Unified Communications Manager AAR.
- C. Configure Cisco Unified Communications Manager RSVP-enabled locations.
- D. Configure Cisco Unified Communications Manager locations.

Answer: B

Explanation:

QUESTION NO: 130

How do RSVP-enabled locations differ from Cisco Unified Communications Manager locations?

- A. RSVP is configured in the ISR independent of Cisco Unified Communications Manager.
- B. RSVP enables AAR within Cisco Unified Communications Manager.
- C. RSVP is topology aware.
- D. RSVP is configured in Cisco Unified Communications Manager independent of the ISR.

Answer: C

Explanation:

QUESTION NO: 131

A voice-mail product that supports only the G.711 codec is installed in headquarters.

Which action allows branch Cisco IP phones to function with voice mail while using only the G.729 codec over the WAN link to headquarters?

- A. Configure Cisco Unified Communications Manager regions.
- B. Configure transcoding within Cisco Unified Communications Manager.
- C. Configure transcoding resources in Cisco IOS and assign to the MRGL of Cisco IP phones.
- D. Configure transcoder resources in the branch Cisco IP phones.

Answer: C

Explanation:

QUESTION NO: 132

Which two statements about the functionality of a gatekeeper are true? (Choose two.)

- A. Cisco Unified Communications Manager has gatekeeper functionality built in.
- B. Cisco Unified Communications Manager registers with a gatekeeper via SIP.
- C. Cisco Unified Communications Manager registers with a gatekeeper via H.323.
- D. A gatekeeper can enable CAC and AAR.
- E. A gatekeeper can enable CAC, but not AAR.

Answer: C,E

Explanation:

QUESTION NO: 133

Which option describes a function of SIP preconditions?

- A. SIP preconditions enable end-to-end RSVP over an SIP trunk.
- B. SIP preconditions enable RSVP between Cisco IP Phones.
- C. SIP preconditions can be enabled in a gatekeeper.
- D. SIP preconditions enable end-to-end RSVP for calls through the PSTN.

Answer: A

Explanation:

QUESTION NO: 134

Which statement about AAR functionality is true?

- A. AAR reroutes calls through the PSTN upon WAN failure.
- B. AAR reroutes calls through the PSTN when Cisco Unified Communications Manager CAC blocks a call.
- C. AAR reroutes calls through the WAN when the PSTN connection fails.
- D. AAR reroutes calls to another phone number when a Cisco IP Phone unregisters.

Answer: B

Explanation:

QUESTION NO: 135

Which statement about RSVP is true?

- A. MTP is typically used for RSVP agent configuration.
- B. If the RSVP agent cannot reserve the required bandwidth on first effort, it enables low latency queuing and tries to connect the call again.
- C. Cisco Unified Communications Manager uses the RSVP-configured bandwidth between sites as a method of determining if there is sufficient bandwidth for the call.
- D. Cisco Unified Communications Manager uses an RSVP-enabled infrastructure and an RSVP-

controlled agent to request a bandwidth reservation from the network in order to place a call.

Answer: D

Explanation:

QUESTION NO: 136

Which system configuration is used to set a restriction on audio bandwidth?

- A. region
- B. location
- C. physical location
- D. licensing

Answer: B

Explanation:

QUESTION NO: 137

An MOH server is configured to stream a local .wav file that is located on its own hard disk drive. An external caller reports that they can only hear the TOH when they are placed on hold. Which two could be the problem? (Choose two.)

- A. The sound card is missing from the server.
- B. The music CD is corrupted.
- C. Insufficient bandwidth is configured.
- D. Unicast streams exceeded the maximum that was configured.

Answer: C,D

Explanation:

QUESTION NO: 138

What happens if location-based CAC is used and there is no bandwidth available when a remote caller is placed on hold?

- A. Cisco Unified Communications Manager sends TOH rather than MOH.
- B. Cisco Unified Communications Manager terminates the call.

- C. Cisco Unified Communications Manager plays default MOH.
- D. Cisco Unified Communications Manager attempts to reconnect the call immediately.

Answer: A

Explanation:

QUESTION NO: 139

On which Cisco Unified Communications Manager configuration parameter does the CODEC that a Cisco IP Phone uses for a call depend?

- A. enterprise parameters
- B. media resources
- C. physical location
- D. region
- E. location

Answer: D

Explanation:

QUESTION NO: 140

In a Cisco Unified Communications Manager centralized call processing model, what is the best CAC method recommended for this type of deployment?

- A. QoS-based
- B. location-based
- C. RSVP-based
- D. region-based
- E. gateway-based
- F. gatekeeper-based

Answer: B

Explanation:

QUESTION NO: 141

When device mobility mode is enabled or disabled for a cluster, to which does the cluster setting apply?

- A. all phones in the cluster that support device mobility
- B. all phones in the cluster that subscribed to device mobility
- C. mobile phones in the cluster that support device mobility
- D. mobile phones in the cluster that are in default mode

Answer: A

Explanation:

QUESTION NO: 142

Which statement is true when device mobility mode is enabled or disabled in the Phone Configuration window?

- A. The device mobility mode phone settings take precedence over the service parameter settings.
- B. The service parameter settings take precedence over the device mobility mode phone settings.
- C. The combined service parameter settings and the device mobility mode phone settings will be used.
- D. The default settings will be used due to the conflicts.

Answer: A

Explanation:

QUESTION NO: 143

Which two entities could be represented by device mobility groups? (Choose two.)

- A. countries
- B. regions
- C. directory numbers
- D. transcoders

Answer: A,B

Explanation:

QUESTION NO: 144

With Cisco Extension Mobility, when a user logs in to a phone type which has no user device profile, what will happen to the phone?

- A. The phone takes on the default clusterwide device profile.
- B. The phone creates a new device profile automatically.
- C. The phone immediately logs the user off.
- D. The phone crashes and reboots.

Answer: A

Explanation:

QUESTION NO: 145

Which technologies provide remote-site redundancy for Cisco IP Phones during a WAN failure?

- A. SRST and MGCP fallback
- B. SRST and TEHO
- C. TEHO and MGCP fallback
- D. SRST and AAR

Answer: A

Explanation:

QUESTION NO: 146

Which remote-site redundancy technology fails over to POTS dial peers from the Cisco Unified Communications Manager dial plan during a WAN failure?

- A. MGCP fallback
- B. H.323 fallback
- C. SCCP fallback
- D. SIP fallback

Answer: A

Explanation:

QUESTION NO: 147

Which statement about technology implementation strategy is true?

- A. Cisco Unified Communications Manager Express can be configured to function with no Cisco

Unified Communications Manager cluster in the enterprise.

B. Cisco Unified Communications Manager Express in SRST mode can be configured to function with no Cisco Unified Communications Manager cluster in the enterprise.

C. SRST can be configured to function with no Cisco Unified Communications Manager cluster in the enterprise.

D. SRST and MGCP fallback can be configured to function with no Cisco Unified Communications Manager cluster in the enterprise.

Answer: A

Explanation:

QUESTION NO: 148

How does the system intelligently shift call processing upon restoration of WAN connectivity?

A. automatically back to the primary Cisco Unified Communications Manager cluster

B. manually back to the primary Cisco Unified Communications Manager cluster

C. automatically back to the secondary Cisco Unified Communications Manager cluster

D. manually back to the secondary Cisco Unified Communications Manager cluster

Answer: A

Explanation:

QUESTION NO: 149

Which two options enable routers to provide basic call handling support for Cisco Unified IP Phones if they lose connection to all Cisco Unified Communications Manager systems? (Choose two.)

A. SCCP fallback

B. Cisco Unified Survivable Remote Site Telephony

C. Cisco Unified Communications Manager Express

D. MGCP fallback

E. Cisco Unified Communications Manager Express in SRST mode

Answer: B,E

Explanation:

QUESTION NO: 150

Which option configures call preservation for H.323-based SRST mode?

- A. voice service voip h323 call preserve
- B. call preservation not possible with H.323
- C. call-manager-fallback preserve-call
- D. dial-peer voice 1 voip call preserve

Answer: A

Explanation:

QUESTION NO: 151

Which configuration command disables the secondary dial tone on the branch office ISR for users calling from the PSTN into the branch office during a WAN failure?

- A. direct-inward-dial
- B. voice translation-rule
- C. incoming called-number
- D. application

Answer: A

Explanation:

QUESTION NO: 152

Which two statements about remote survivability are true? (Choose two.)

- A. SRST supports more Cisco IP Phones than Cisco Unified Communications Manager Express in SRST mode.
- B. Cisco Unified Communications Manager Express in SRST mode supports more Cisco IP Phones than SRST.
- C. MGCP fallback is required for ISDN call preservation.
- D. MGCP fallback functions with SRST.

Answer: A,D

Explanation:

QUESTION NO: 153

Refer to the exhibit.

```
dial-peer voice 901 pots
destination-pattern 9011T
port 1/0:23
```

Which configuration change is needed to enable NANP international dialing during MGCP fallback?

- A. Change the dial peer to dial-peer voice 901 voip.
- B. Change the dial peer to dial-peer voice 9011 pots.
- C. Add the command prefix 011 to the dial peer.
- D. Add the command prefix 9011 to the dial peer.

Answer: C

Explanation:

QUESTION NO: 154

Which three commands are mandatory to implement SRST for five Cisco IP Phones? (Choose three.)

- A. call-manager-fallback
- B. max-ephones
- C. keepalive
- D. limit-dn
- E. ip source-address

Answer: A,B,E

Explanation:

QUESTION NO: 155

A Cisco Unified Communications Manager cluster is installed in headquarters only.

How can international calls be blocked while national calls are allowed for branch office Cisco IP Phones during a WAN failure?

- A. Configure CSS and partitions in Cisco Unified Communications Manager and apply the CSS and partitions to the SRST ISR.
- B. Configure CSS and partitions in the SRST ISR.
- C. Configure COR in the SRST ISR.
- D. Configure voice translations in the SRST ISR.

Answer: C

Explanation:

QUESTION NO: 156

Which commands are needed to configure Cisco Unified Communications Manager Express in SRST mode?

- A. telephony-service and srst mode
- B. telephony-service and moh
- C. call-manager-fallback and srst mode
- D. call-manager-fallback and voice-translation

Answer: A

Explanation:

QUESTION NO: 157

Which command can be used to manually send the MGCP gateway to register with the secondary Cisco Unified Communications Manager server?

- A. ccm-manager switchover-to-backup
- B. mgcp use backup
- C. ccm-manager register backup
- D. not supported

Answer: A

Explanation:

QUESTION NO: 158

Which command is needed to utilize local dial peers on an MGCP-controlled ISR during an SRST failover?

- A. ccm-manager fallback-mgcp
- B. telephony-service
- C. dialplan-pattern
- D. isdn overlap-receiving
- E. voice-translation-rule

Answer: A

Explanation:

QUESTION NO: 159

Which option is a benefit of implementing CFUR?

- A. CFUR is designed to initiate TEHO to reduce toll charges.
- B. CFUR can prevent phones from unregistering.
- C. CFUR can reroute calls placed to a temporarily unregistered destination phone.
- D. CFUR eliminates the need for COR on an ISR.

Answer: C

Explanation:

QUESTION NO: 160

This is the configuration on the voice gateway:

telephony-service

max-ephones 30

max-dn 60 preference 0

srst mode auto-provision all

srst dn line-mode dual

srst dn template 3

srst ephone description srst fallback auto-provision phone

srst ephone template 5

Which ephone-dn would be expected upon activation of SRST?

A. ephone-dn 1 dual-line

number 7001

description 7001

name 7001

ephone-dn-template 5

This DN is learned from srst fallback ephones

B. ephone-dn 1 dual-line

number 7001

description 7001

name 7001

ephone-dn-template 3

This DN is learned from srst fallback ephones

C. ephone-dn 1

number 7001

description 7001

name 7001

ephone-dn-template 5

This DN is learned from srst fallback ephones

D. ephone-dn 1

number 7001

description 7001

name 7001

ephone-dn-template 3

This DN is learned from srst fallback ephones

Answer: A

Explanation:

QUESTION NO: 161

What are two important considerations when implementing TEHO to reduce long-distance cost?
(Choose two.)

- A. on-net calling patterns
- B. E911 calling
- C. number of route patterns
- D. caller ID

Answer: B,D

Explanation:

QUESTION NO: 162

Which ability does the Survivable Remote Site Telephony feature provide?

- A. a means to allow the local site to continue to send and receive calls in the event of a WAN failure
- B. a means to route calls on-net through other sites during high utilization periods
- C. a method that allows for backup calls in the event that your gateway fails
- D. the ability to force a call out of a certain trunk when the Cisco Unified Communications Manager is being upgraded

Answer: A

Explanation:

QUESTION NO: 163

Which E.164 transformation pattern represents phone numbers in Germany?

- A. \+49.!
- B. 49.!
- C. \49.!
- D. \+49.X

Answer: A

Explanation:

QUESTION NO: 164

Which two statements are true regarding the implementation of globalized call-routing in terms of localized call egress? (Choose two.)

- A. Calling-party numbers are routed from the gateway or trunks to phones.
- B. Called-party numbers are routed from the gateway or trunks to phones.
- C. Calling-party numbers of internal calls are routed from the gateway or trunks.
- D. Calling-party calls are routed to the gateway and trunks.

Answer: A,D

Explanation:

QUESTION NO: 165

When implementing a dial plan for multisite deployments, what must be present for SRST to work successfully?

- A. dial peers that address all sites in the multisite cluster
- B. translation patterns that apply to the local PSTN for each gateway
- C. incoming and outgoing COR lists
- D. configuration of the gateway as an MGCP gateway

Answer: B

Explanation:

QUESTION NO: 166

Which statement about TEHO is true?

- A. The dial plan is simplified with local route groups.
- B. Local route groups add complexity to the dial plan.
- C. Toll charges can be reduced when TEHO is implemented with CAC.
- D. Toll charges can be reduced when TEHO is implemented with MGCP fallback.

Answer: A

Explanation:

QUESTION NO: 167

How are Cisco IP Phones directly configured to utilize local route groups?

- A. with Cisco Unified Communications Manager device pools

- B. with Cisco Unified Communications Manager CSS and partitions
- C. with Cisco Unified Communications Manager regions
- D. with Cisco Unified Communications Manager locations
- E. with Cisco Unified Communications Manager AAR

Answer: A

Explanation:

QUESTION NO: 168

When you configure TEHO for long-distance calls and use the local PSTN gateways as fallback, how many route patterns do you require for a cluster with five sites that are located in different area codes?

- A. 15 when not using a local route group
- B. 6 when using a local route group
- C. 5 when using a local route group
- D. 10 when not using a local route group

Answer: B

Explanation:

QUESTION NO: 169

Which statement is true regarding the configuration of SAF Forwarder?

- A. In a multisite dial plan, SAF Forwarders may exist in multiple autonomous systems.
- B. The client label that is configured in Cisco Unified Communications Manager must match the configuration on the SAF Forwarder router.
- C. There should not be multiple nodes of Cisco Unified Communications Manager clusters acting as SAF clients.
- D. The destination IP address must match the loopback address of the SAF router.

Answer: A

Explanation:

QUESTION NO: 170

Which three devices support the SAF Call Control Discovery protocol? (Choose three.)

- A. Cisco Unified Border Element
- B. Cisco Unity Connection
- C. Cisco IOS Gatekeeper
- D. Cisco Catalyst Switch
- E. Cisco IOS Gateway
- F. Cisco Unified Communications Manager

Answer: A,E,F

Explanation:

QUESTION NO: 171

Which action configures phones in site A to use G.711 to site B but to use G.729 to site C?

- A. Configure Cisco Unified Communications Manager regions.
- B. Configure Cisco Unified Communications Manager locations.
- C. Configure transcoder resources in Cisco Unified Communications Manager.
- D. Configure a gatekeeper.

Answer: A

Explanation:

QUESTION NO: 172

Which statement is true when considering a Cisco VoIP environment for regional configuration?

- A. G.711 requires 128K of bandwidth per call.
- B. G.729 requires 24K of bandwidth per call.
- C. The default codec does not matter if you have defined a hardware MTP in your Cisco Unified Communications Manager environment.
- D. To deploy a Cisco H.323 gatekeeper, you must configure MTP resources on the gatekeeper and only use G.711 between regions.

Answer: C

Explanation:

QUESTION NO: 173

Which system configuration is used to set audio codecs?

- A. region
- B. location
- C. physical location
- D. licensing

Answer: A

Explanation:

QUESTION NO: 174

When you configure regions in a Cisco Unified Communications Manager multisite cluster, which two are ways to prevent G.722 from being advertised in the cluster? (Choose two.)

- A. modify the service parameter
- B. modify the enterprise parameter
- C. modify the device pool
- D. modify the line settings

Answer: A,B

Explanation:

QUESTION NO: 175

Which action configures the registration of transcoder resources?

- A. Cisco IOS software registers transcoder resources with SIP.
- B. Cisco IOS software registers transcoder resources with SCCP.
- C. Cisco IOS software registers transcoder resources with H.323.
- D. Cisco IOS software does not register transcoder resources.

Answer: B

Explanation:

QUESTION NO: 176

Which option describes the reason that transcoding resources are added in Cisco Unified Communications Manager?

- A. to enable transcoding resources in a Cisco Unified Communications Manager server
- B. to enable Cisco Unified Communications Manager to select the optimal single codec for end-to-end calls
- C. to enable transcoding resources in Cisco IP Phones
- D. to provide transcoding resources in Cisco IOS gateways to Cisco Unified Communications Manager

Answer: D

Explanation:

QUESTION NO: 177

Which action configures transcoding resources in Cisco Unified Communications Manager to function with branch office Cisco IP Phones?

- A. Configure the branch office IP phones with CSS and partitions.
- B. Configure the branch office IP phones with MRGs and MRGLs.
- C. Configure the branch office IP phones with regions.
- D. Configure the branch office IP phones with locations.

Answer: B

Explanation:

QUESTION NO: 178

What is the purpose of configuring a hardware-based MTP when deploying Cisco Unified Communications Manager?

- A. to allow for supplementary services such as hold, transfer, and conferencing
- B. when you need support for up to 24 MTP sessions on the same server and 48 on a separate server
- C. when you need the ability to grow support by using DSPs
- D. when you want to only use Cisco Unified Communications Manager resources

Answer: C

Explanation:

QUESTION NO: 179

Which two options are effective mechanisms to restrict the maximum number of voice calls on a WAN link? (Choose two.)

- A. Configure a gatekeeper with an SIP trunk.
- B. Configure a gatekeeper and a gatekeeper-controlled trunk in Cisco Unified Communications Manager with bandwidth control.
- C. Configure Cisco Unified Communications Manager regions.
- D. Configure Cisco Unified Communications Manager locations.

Answer: B,D

Explanation:

QUESTION NO: 180

Which action configures CAC utilizing only Cisco Unified Communications Manager software?

- A. Configure Cisco Unified Communications Manager regions.
- B. Configure Cisco Unified Communications Manager locations.
- C. Configure Cisco Unified Communications Manager RSVP-enabled locations.
- D. Configure Cisco Unified Communications Manager MTPs.

Answer: B

Explanation:

QUESTION NO: 181

Which bandwidth amounts are correct for configuring locations?

- A. 8 kb/s for G.729, 64 kb/s for G.711, and 64 kb/s for G.722
- B. 8 kb/s for G.729, 64 kb/s for G.711, and 16 kb/s for G.722
- C. 64 kb/s for G.729, 64 kb/s for G.711, and 64 kb/s for G.722
- D. 8 kb/s for G.729, 8 kb/s for G.711, and 8 kb/s for G.722

Answer: A

Explanation:

QUESTION NO: 182

Which statement about the function of a gatekeeper is true?

- A. A gatekeeper improves call routing between servers within a single Cisco Unified Communications Manager cluster.
- B. A gatekeeper can replace the dial plan of a Cisco Unified Communications Manager cluster.
- C. A gatekeeper can simplify the dial plan between many different Cisco Unified Communications Manager clusters.
- D. Gatekeepers can be implemented to deploy RSVP-based CAC.

Answer: C

Explanation:

QUESTION NO: 183

For which VoIP protocol does a gatekeeper provide address translation and control access?

- A. H.323
- B. SIP
- C. Skinny
- D. H.248

Answer: A

Explanation:

QUESTION NO: 184

Which CAC configuration on a gatekeeper restricts to 10 G.711 audio calls?

- A. Use the command bandwidth 10.
- B. Use the command bandwidth 1280.
- C. Use the command bandwidth 160.
- D. Use the command bandwidth session 10.

Answer: B

Explanation:

QUESTION NO: 185

Which action configures AAR to route the calls that have been rejected by the gatekeeper CAC through the PSTN?

- A. Configure Cisco IP Phones for AAR.
- B. Configure AAR to work with SRST.
- C. Configure AAR to work with CTI route points.
- D. This configuration is not possible using AAR.

Answer: D

Explanation:

QUESTION NO: 186

What is a prerequisite of AAR deployment?

- A. You must have a single distributed call processing deployment.
- B. Calls must be manually rerouted through the PSTN or other networks when Cisco Unified Communications Manager blocks a call due to insufficient location bandwidth.
- C. Calls must be automatically rerouted through the PSTN or other networks when Cisco Unified Communications Manager blocks a call due to insufficient location bandwidth.
- D. Clustering must be implemented over the WAN.
- E. You must have a centralized call processing deployment.

Answer: E

Explanation:

QUESTION NO: 187

If the device pool in the phone record does not match the device pools in the matching subnet, what will the system consider the phone to be?

- A. roaming
- B. unregistered
- C. unknown
- D. new device

Answer: A

Explanation:

QUESTION NO: 188

Which two Cisco Extension Mobility attributes are available in the user device profile? (Choose two.)

- A. regions
- B. description
- C. phone button template
- D. NTP information

Answer: B,C