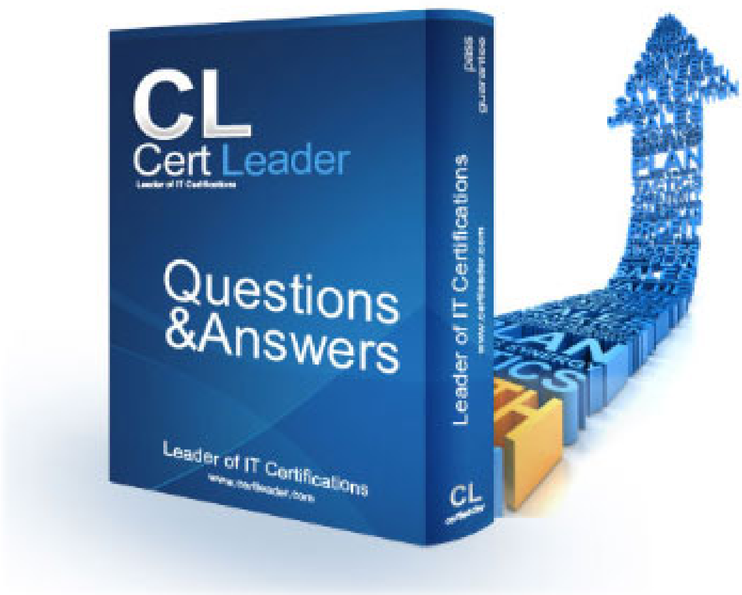
300-075 - CIPTV2 Implementing Cisco IP Telephony and Video, Part 2

https://www.certleader.com/300-075-dumps.html



1. The network administrator at Enterprise X is creating the guidelines for a new IPT deployment consisting of a large number of remote offices. Every user within Enterprise X is assigned a directory number of 5 digits. Which option might cause an issue in a multisite deployment?

A. Overlapping DID ranges are allocated to each site.

B. The maximum number of IP phones are in use at each remote site.

C. MoH cannot be provided for the remote sites.

D. All media streams are necessarily routed through the central office for calls to establish correctly.

Answer: A

2. 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)

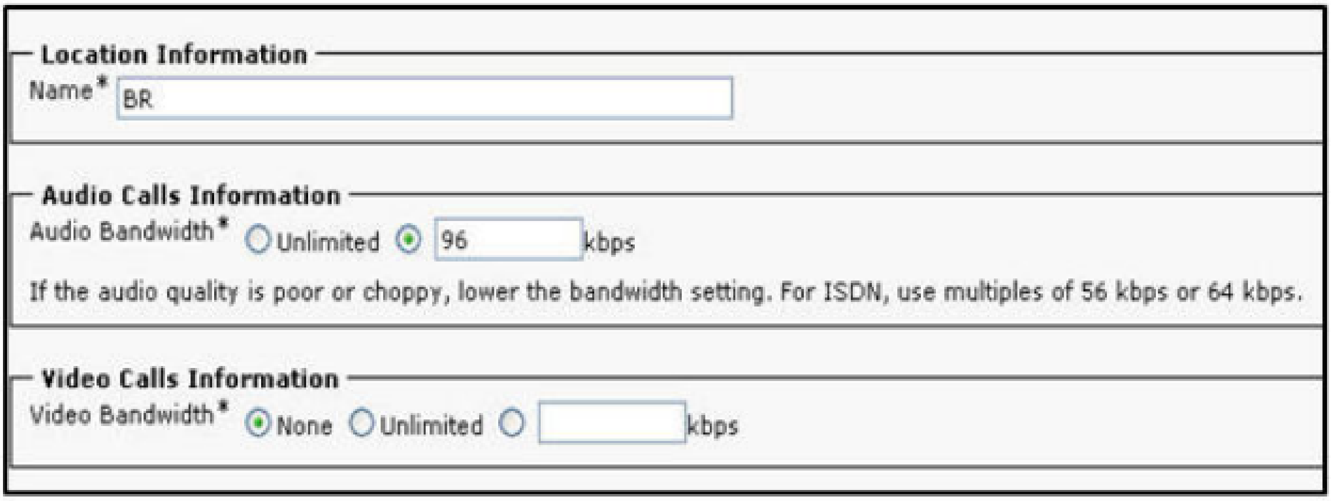
B. intercluster trunk (non-gatekeeper controlled)

C. SIP trunk

D. intercluster trunk (gatekeeper-controlled)

Answer: D

3. Refer to the exhibit.



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:** A, B, C, E 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

4. Which two options are requirements for deploying an H.323 gateway with Cisco Unified Communications Manager? (Choose two.)

A. Cisco Unified Communications Manager and the H.323 gateway must be configured use the same TCP port for H.323 calls.

B. The H.245TCSTimeout timer must be set to at least 25.

C. Cisco voicemail ports must be active.

D. The Media Exchange Interface Capability Timer must be set to less than 20.

E. The Media Exchange Timer must be set to less than 20.

Answer: A,B

5. 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

6. 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

7. What is the purpose of a SAF Client?

A. To decode address information and route calls to and from the end points

B. To pass IP information from the CUCM to the endpoint

C. To learn about and advertise or subscribe information about SAF network services

D. To reside in the Cisco IOS software, and to communicate with the SAF forwarder

Answer: C

8. On which two call legs is the media encryption enforced in a Collaboration Edge design? (Choose two.)

A. Expressway-C to Cisco Unified Communications Manager

B. Expressway-C to Expressway-E

C. Expressway-E to outside-located endpoint

D. Expressway-E to Cisco Unified Communications Manager

E. Expressway-C to internal endpoint

Answer: B,C

9. When an incoming PSTN call arrives at an H.323 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 H.323 gateway configuration in Cisco Unified Communications Manager.

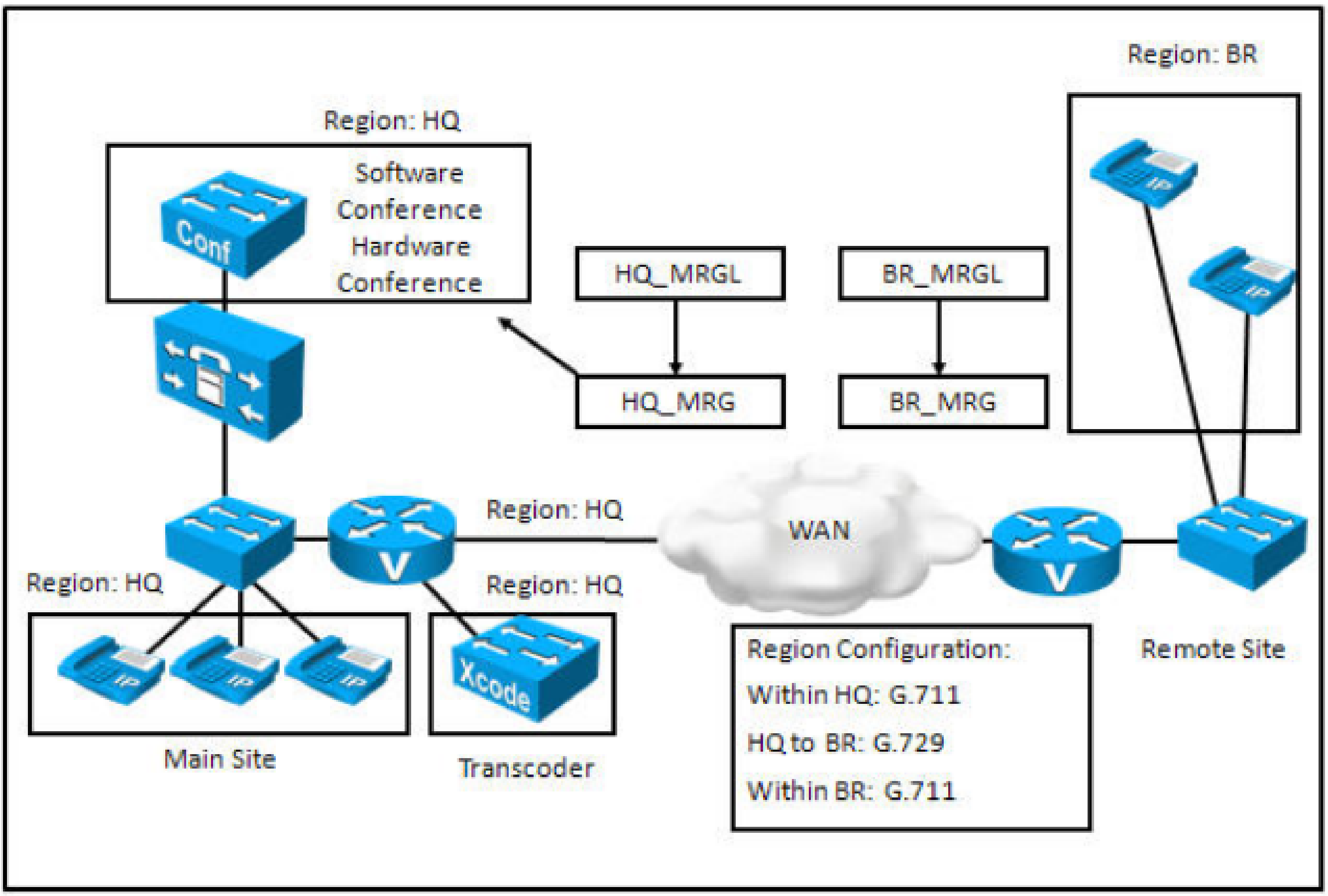
B. Normalization is done using route patterns.

C. Normalization is done using the gateway incoming calling party prefixes based on number type.

D. Normalization is achieved by local route group that is assigned to the H.323 gateway.

Answer: A

10. 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_MRG 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

11. Which minimum configuration is needed for the SAF Internal Client to register with this SAF Forwarder?

A. router eigrp SAF!service-family ipv4 autonomous-system 1!topology baseexit-sf-topologyexit-service-family!voice service safprofile trunk-route 1session protocol sip interface Loopbackl transport tcp port 5060i

B. router eigrp SAF!service-family ipv4 autonomous-system 1!topology baseexit-sf-topologyexit-service-family!voice service safprofile trunk-route 1session protocol sip interface Loopbackl transport tcp port 5060!profile dn-block 1 alias-prefix 1972555pattern 1 type extension 4xxx!profile callcontrol 1dn-servicetrunk-route 1dn-block 1dn-block 2i

C. router eigrp SAF!service-family ipv4 autonomous-system 1!topology baseexit-sf-topologyexit-service-family!voice service safprofile trunk-route 1session protocol sip interface Loopbackl transport tcp port 5060!profile dn-block 1 alias-prefix 1972555pattern 1 type extension 4xxx!profile callcontrol 1dn-servicetrunk-route 1dn-block 1dn-block 2!channel 1 vrouter SAF asystem 1subscribe callcontrol wildcardedpublish callcontrol 1i

D. router eigrp SAF!service-family ipv4 autonomous-system 1!topology baseexit-sf-topologyexit-service-family!voice service saf!channel 1 vrouter SAF asystem 1

E. router eigrp SAF!service-family ipv4 autonomous-system 1!topology base exit-sf-topologyexit-service-family i

Answer: A

12. 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

13. 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:** A, C, D IP Phone display and Physical phone IP settings are two locations were 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

14. 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

15. 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

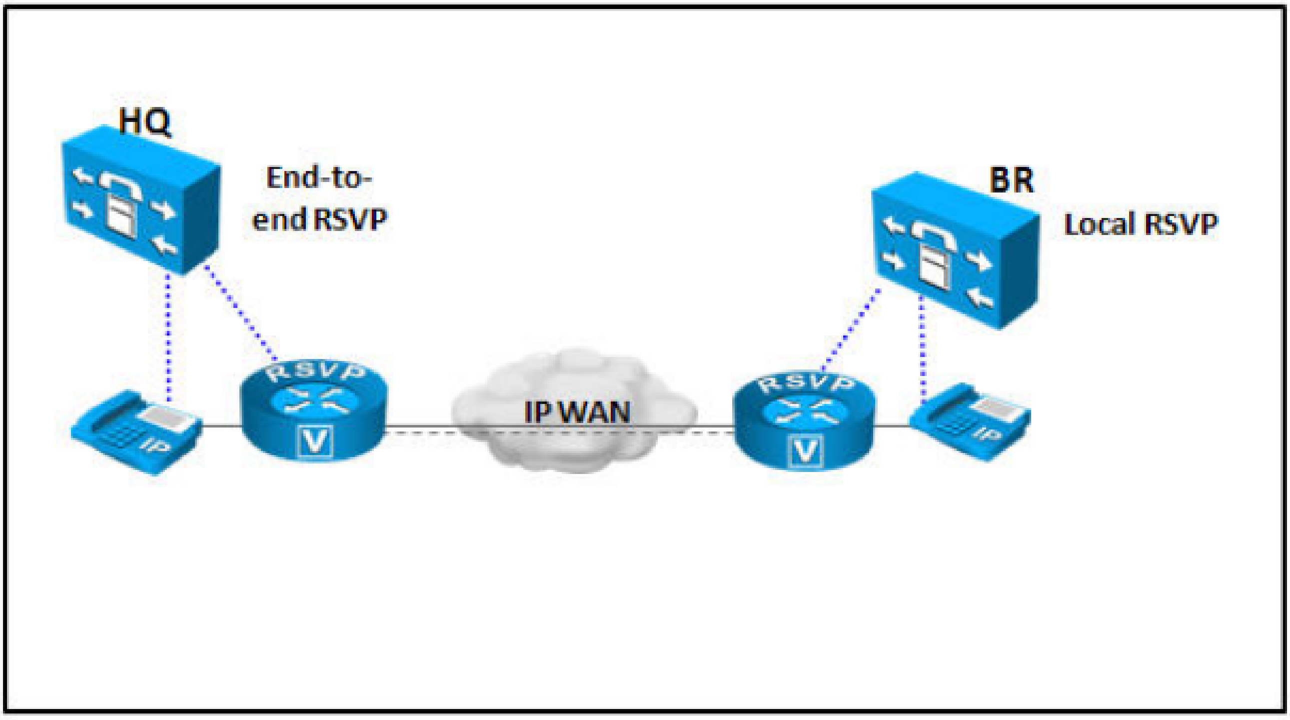
B. +14087071222 with number type international

C. +14087071222

D. 14087071222

Answer: C

16. Refer to the exhibit.



The Cisco Unified Communications Manager at HQ has been configured for end-to-end RSVP. The Cisco Unified Communications Manager at BR 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 the BR site to the IP phone at the HQ site, which statement is true?

A. The Cisco Unified Communications Manager at BR 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 BR will use local RSVP. The HQ Cisco Unified Communications Manager will use end-to-end RSVP.

D. The call will fail.

E. The call will proceed as a normal call with no RSVP reservation.

Answer: A

17. 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

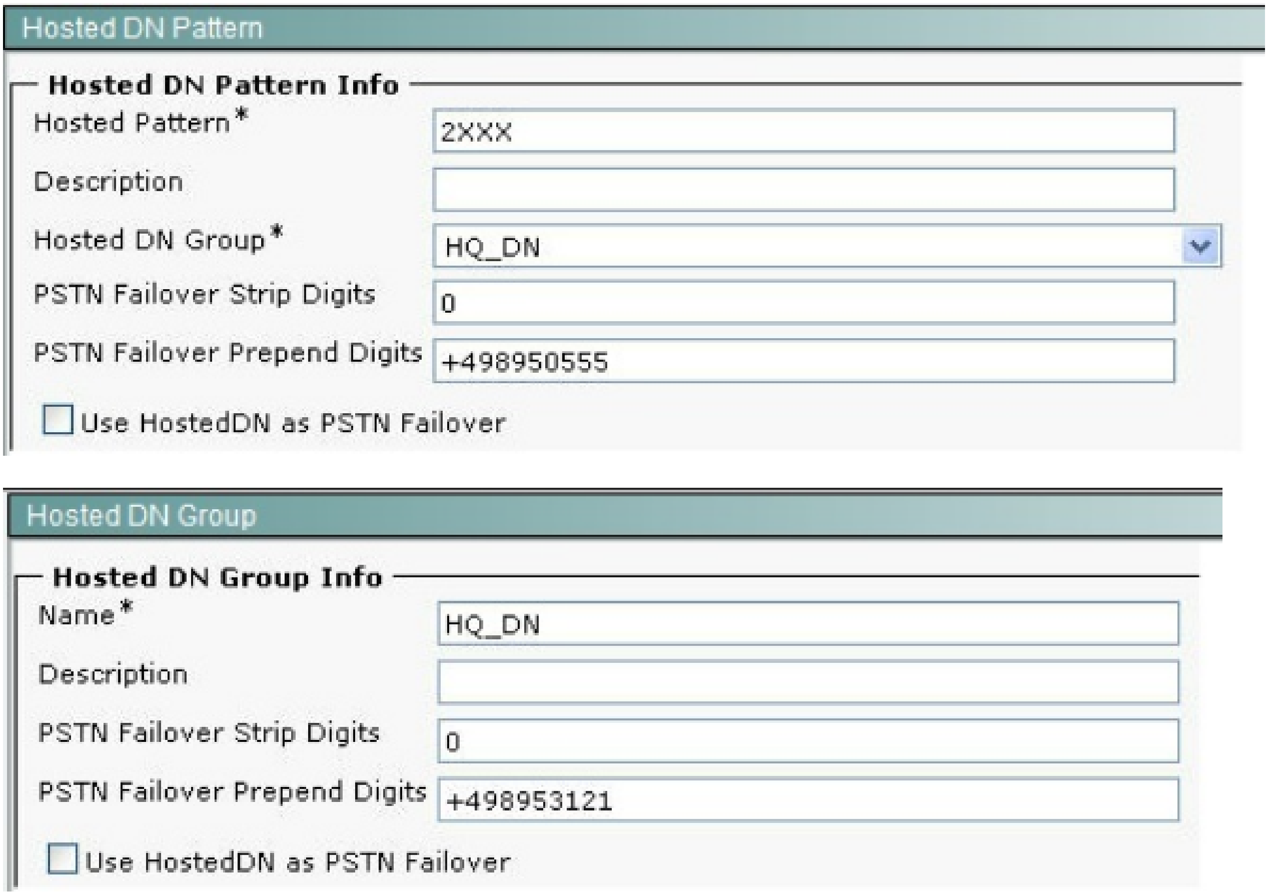
B. unregistered

C. unknown

D. new device

Answer: A

18. Refer to the exhibit



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:** B, C, D, E PSTN failover prepend digit is +498950555 Link:

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

19. 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

20. 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

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