Exam : 642-452

Title : Gateway Gatekeeper Exam

Ver : 10.05.07

QUESTION 1:

Which two functions are provided by a DSP farm? Select two.

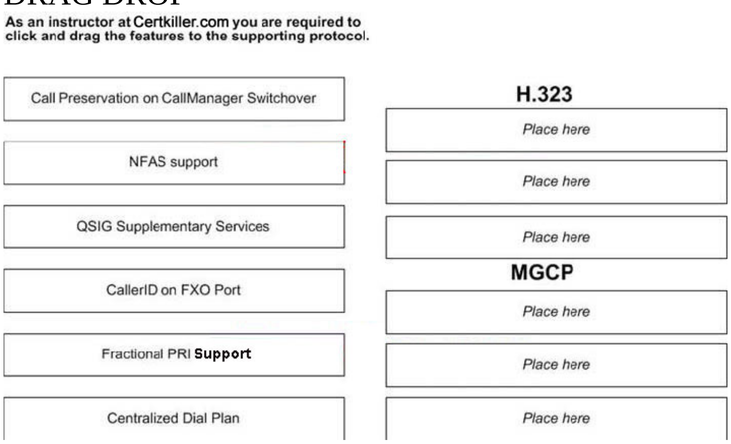
A. called ID
B. transcoding
C. E911
D. directory lookup
E. conference bridging
F. music on hold

Answer: B, E

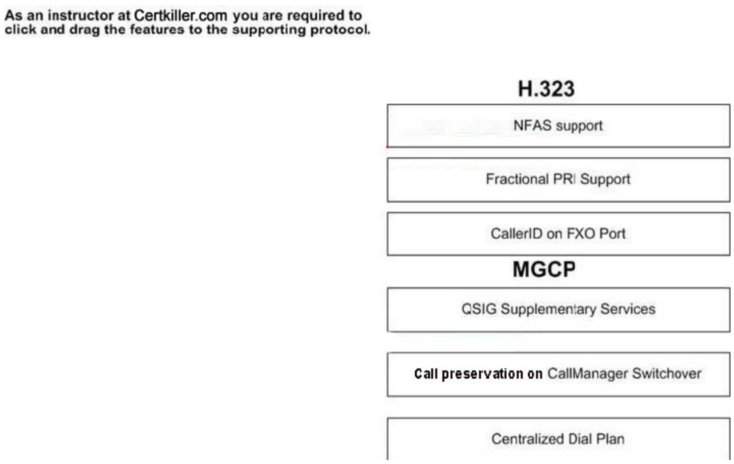
Explanation:
The DSP farm uses the DSP resources in network modules on Cisco routers to provide voice-conferencing, transcoding, and hardware MTP services.
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 4-56 Implementing Advanced Gateway Features

QUESTION 2:

DRAG DROP



Answer:



Explanation:
Why Choose H.323
* Integrated access
* Caller ID support on analog FXO
* Many more TDM interface types and signaling
* Dropping DSPs on hairpinned calls
* Gateway-resident applications like TCL and VXML
* CAC network design with H.323 gatekeepers
* No release dependencies between gateways and Cisco CallManager
* Much easier migration architecture to SIP
* Call preservation for Cisco SRST
* NFAS support
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 1-66 Function of Gateways and Gatekeepers
Using MGCP as the call control protocol to a gateway has the following advantages:
* Centralized configuration, control and download from Cisco CallManager
* Better feature interaction with capabilities like caller ID and name display
* Easy, centralized dial-plan management
* Gateway voice security features (voice encryption) as of Cisco IOS Software Release 12.3.(5th)T
* Q Signaling (QSIG) supplementary services as supported by Cisco CallManager:
- Cisco CallManager interconnects to a QSIG network using an MGCP gateway and T1 or E1 PRI connections to a private integrated services network (PISN). The MGCP gateway establishes the call connections. Using the PRI backhaul mechanism, the gateway passes the QSIG messages to the Cisco CallManager to set up QSIG calls and send QSIG messages to control features.
- When a PBX is connected to a gateway that is using QSIG via H.323, calls that are made between phones on the PBX and IP phones attached to the Cisco CallManager can have only basic PRI functionality. The gateway that terminates the QSIG protocol provides only the calling line ID (CLID) and DID number, instead of Cisco CallManager providing that information.
* Enhanced call survivability:
- Calls from IP phones through an MGCP gateway are preserved on a CallManager

failover. This feature avoids dropped calls when applying the monthly operation system service release on the Cisco CallManagers
- In SRST mode, calls from IP phones through an MGCP gateway are preserved on MGCP fallback for calls on analog or CAS circuits. Calls on ISDN circuits are dropped on fallback.
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
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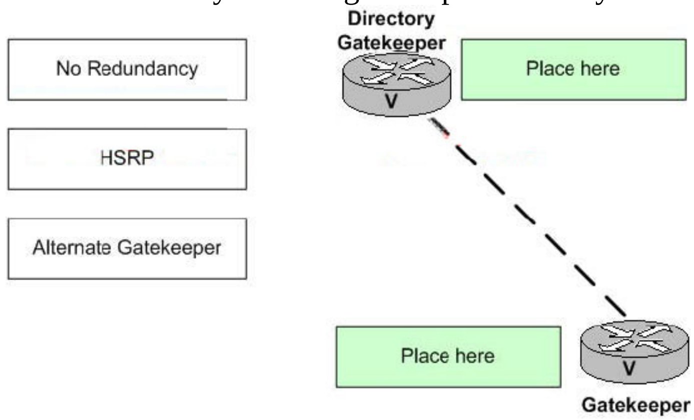
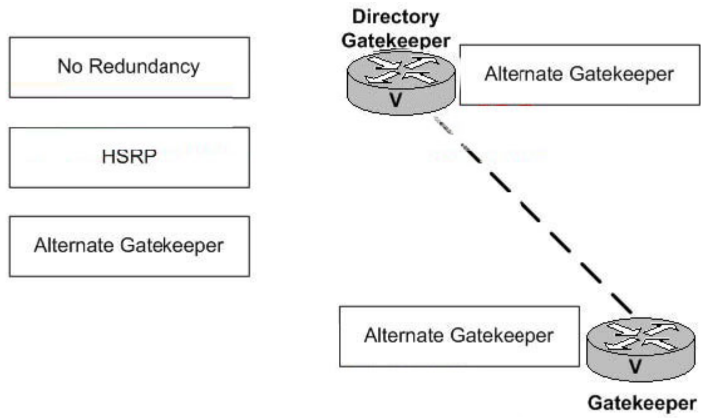
QUESTION 3:

DRAG DROP
As an instructor at Certkiller .com you are required to click and drag the recommended redundancy method to each layer of the gatekeeper hierarchy. An item may be used more than once.



Answer:



Explanation: Cisco recommends that you use gatekeeper clustering(that is Alternate Gatekeeper) to provide gatekeeper redundancy whenever possible. Use HSRP for redundancy only if gatekeeper clustering is not available in your software feature set.
In any layer of gatekeeper hierarchy, Alternate Gatekeeper method are recommended..

QUESTION 4:

An NM-HDV2 is being configured for transcoding. Which Cisco IOS command marks the beginning of the transcoding parameters?

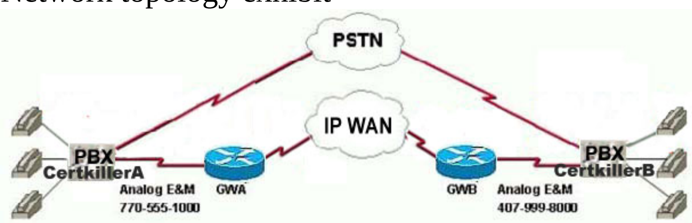
A. dsp services transcoding
B. associate application transcoding
C. dspfarm profile 10 transcoding
D. voice-card 2 transcoding

Answer: C

Explanation:
Configuring a DSP Farm on the NM-HDV2 or NM-HD-1V/2V/2VE
Step 1:router#
Step 2:router#conf t
Step 3:router(config)#voice-card 2
Step 4:router(config-voicecard)#dsp services dspfarm
Step 5:router(config-voicecard)#exit
Step 6:router(config)#dspfarm profile 10 transcode
Step 7:router(config-dspfarm-profile)#description SAMPLE TRANSCODE
Step 8:router(config-dspfarm-profile)#code g729r8
Step 9:router(config-dspfarm-profile)#maximum sessions 6
Step 10:router(config-dspfarm-profile)#associate application sccp
Step 11:router(config-dspfarm-profile)#no shutdown
Step 12:router(config-dspfarm-profile)#exit
Step 13:router(config)#gateway
Step 14:router(config-gateway)#timer receive-rtp 1200
Step 15: end or return to step 6 to continue configuring DSP farm profiles
IOS syntax for step 6 is: dspfarm profile profile-identifier {conference | mtp | transcode}
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 4-66 Implementing Advanced Gateway Features

QUESTION 5:

Network topology exhibit



You are working with Certkiller .com in the lab to test toll bypass. There are two small PBXs designated PBX Certkiller A and PBX Certkiller B that have been connected back-to-back to simulate a tie-line from the PSTN. The client would like to remove the tie-line and carry his voice traffic over the IP WAN. Which four main parameters define how the connection will be configured between the gateway and the PBX? Choose four.

A. WAN link speeds

B. E&M type and wiring scheme
C. support for option 81 on the PBX
D. start dial supervision signaling
E. address signaling type
F. audio implementation

Answer: B, D, E, F

Explanation:
PBX configuration support: The following presents the key
information required to ensure that a Cisco voice gateway can be configured to
support calls from a legacy PBX:
- E&M signaling type (I, II, III, IV or V)
- Audio implementation (2-wire or 4-wire)
- Start dial supervision (wink-start, immediate or delay-dial)
- Dial method (dual tone multifrequency [DTMF] or pulse)
- Call progress tones (standardized within geographic regions)
- PBX port impedance
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 2-9 Integrating a VoIP Network to the PSTN and PBXs

QUESTION 6:

Exhibit

```
1d20h: ISDN Se3/0:15: Outgoing call id = 0x85F4, dsl 0
1d20h: ISDN Se3/0:15: process_pri_call(): call id 0x85F4, number 35293315, speed 0,
call type VOICE, redialed? f, csm call? f, pdata? t
1d20h: calLED type/plan overridden by call_decode
1d20h: did't copy oct3a reason: not CALLER_NUMBER_IE
1d20h:  building outgoing channel id for call nfas_int is 0 len is 0
1d20h: ISDN Se3/0:15: TX <-  INFOc sapi = 0  tei = 0  ns = 19  nr = 19  i =
0x08020089050403809OA31802008905040380 90A3102008905040380 90A315
1d20h:      SETUP pd = 8  callref = 0x8089
1d20h:          Bearer capability i : 0x8089
1d20h:          Channel ID i = 0xA98381
1d20h:          Progress Ind i = 0x8183 - Origination address is non-ISDN
1d20h:          Called Party Number i = 0x80, '35293315', Plan:Unknown, Type:Unknown
1d20h: ISDN Se3/0:15: RX <-  RRr sapi = 0  tei = 0  nr = 20
1d20h: ISDN Se3/0:15: RX <-  INFOc sapi = 0  tei = 0  ns = 19  nr = 20  i =
0x080280895A08028286
1d20h:      RELEASE_COMP pd = 8  callref = 0x8089
1d20h:          Cause i = 0x8286 - Channel unacceptable
1d20h: ISDN Se3/0:15: TX ->  RRr sapi = 0  tei = 0  nr = 20
1d20h: ISDN Se3/0:15: CCPRI_ReleaseCall(): bchan 1, call id 0x85F4, call type VOICE
1d20h: CCPRI_ReleaseChan released b_dsl 0 B_Chan 1
1d20h: ISDN Se3/0:15: LIF_EVENT: ces/callid 1/0x85F4 CALL_REJECTION
1d20h: ISDN Se3/0:15: LIF_EVENT: ces/callid 1/0x85F4 CALL_CLEARED
1d20h: ISDN Se3/0:15: received CALL_CLEARED  call_id 0x85F4
```

Certkiller .com is integrating a Cisco CallManager system with the existing PBX via an E1 QSIG trunk.
After the initial configuration, no calls can be placed from IP phones to PBX phones.
How can this problem be resolved?

A. Add the command isdn contiguous-bchan to the serial interface.
B. change the channel selection order from descending to ascending.
C. Add the command isdn negotiate-bchan to the serial interface.
D. Increase the ISDN T302 timer to allow more time for call setup.

Answer: C

Explanation: By default, Cisco router will not accept a different B-channel. To enable the router to accept a

B-channel that is different from the B-channel requested in the outgoing call setup message, use the isdn negotiate-bchan interface configuration command.

QUESTION 7:

Which three functions can be performed by a gatekeeper? Select three.

A. voice QoS
B. admission control
C. zone management
D. address translation
E. voice media transport
F. conferencing for H.323 terminals

Answer: B, C, D

Explanation:
Mandatory Gatekeeper Functions
- Address Translation: A gatekeeper translates H.323 IDs and standard E.164 telephone numbers to endpoint IP addresses.
- Admission Control: Controls endpoint admission into the H.323 network. To achieve this, the gatekeeper uses H.225 Registration, Admission, and Status (RAS) messages and Admission Request (ARQ), Admission Confirmation (ACF), and Admission Rejection (ARJ) messages.
- Bandwidth Control: Gatekeepers use H.225 Bandwidth Request (BRQ), Bandwidth Confirmation (BCF), and Bandwidth Rejection (BRJ) messages to manage endpoint bandwidth requirements.
- Zone Management: The gatekeeper manages all registered endpoints in the zone.
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 1-7 Function of Gateways and Gatekeepers

QUESTION 8:

Certkiller .com is having trouble managing its fully meshed gatekeepers. What can be done to ease this administrative problem?

A. install an H.323 proxy server
B. implement a directory gatekeeper
C. group the gatekeepers into clusters
D. separate the gatekeepers into zones

Answer: B

Explanation: By using a directory gatekeeper, it is no longer necessary to have a full mesh between gatekeepers, which is a major advantage. Directory gatekeepers centralize the dial plan and also serve as a potential interface to other centralized

applications. Without a directory gatekeeper, you would have to add an entry in every gatekeeper on the network every time you add a new zone on one of the gatekeepers. However, with a directory gatekeeper, you can add the new zone in the local gatekeeper and the directory gatekeeper only. If the local gatekeeper cannot resolve a call request locally, it forwards that request to the directory gatekeeper with a matching zone prefix.
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pg 5-7, 88 Deploying Gatekeepers

QUESTION 9:

The phone company is delivering circuits for a new VoIP deployment to connect to the PSTN. The circuits will support four simultaneous calls and provide the name and number of the calling party. Which interface type should be installed?

A. E&M
B. FXO
C. FXS
D. T1 CAS

Answer: B

Explanation:
The two circuit options that are available are analog and digital. The advantages and shortcomings of individual analog and digital circuit options that can be deployed in Cisco voice gateways are summarized in the following list:
* Subscriber loop: Subscriber loop is usually a low-cost solution and is used when traditional phones connect directly to a voice gateway with a Foreign Exchange Station (FXS) interface. The two types of interfaces that make up subscriber loop trunks are the following:
- Foreign Exchange Office (FXO): Use FXO ports to connect to a central office (CO), PBX, or key telephone system. You can configure loop-start or ground-start signaling interfaces, depending on the model of voice interface card or network module selected.
- FXS: Use FXS ports to connect to any plain old telephone service (POTS) device such as analog phones, fax machines, and legacy voice-mail systems.
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 2-4 Integrating a VoIP Network to the PSTN and PBXs

QUESTION 10:

Certkiller .com is using a NH-HDV2 for transcoding services within a Cisco CallManager environment. Which command will instruct the voice gateway to register with a Cisco CallManager with IP address 10.172.15.4.

A. sccp ip-address 10.172.15.4 priority 1 version 4.0
B. sccp ccm 10.172.15.4 priority 1 version 4.0

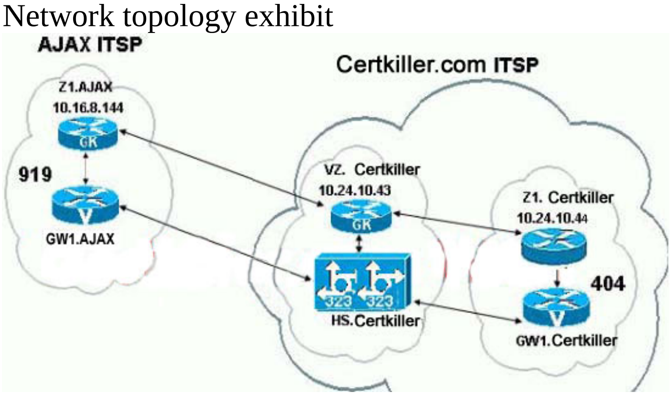
C. sccp register 10.172.15.4 priority 1 version 4.0
D. sccp server 10.172.15.4 priority 1 version 4.0

Answer: B

Explanation:
Perform this task to enable SCCP on the local interface that a DSP
farm uses to register with Cisco CallManager. This step is the same for either DSP
type.
Configuring a DSP Farm - Common Steps
Step 1: enable
Step 2: configure terminal
Step 3: sccp ccm {ip-address | dns} identifier identifier-number [port port-number]
[version version-number] or sccp ccm {ip-address | dns} priority priority priority [port
port-number] [version version-number]
Step 4: sccp local interface-type interface-number
Step 5: sccp
Step 6: sccp ip predecence value
Step 7: exit
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 4-65 Implementing Advance Gateway Features

QUESTION 11:

Network topology exhibit



Certkiller .com ITSP would like to be able to test an IP-to-IP gateway. Certkiller .com has set up two ITSPs
in the lab to simulate a real-world situation.
Which set of commands will configure the VZ. Certkiller gatekeeper correctly?

A. gatekeeper
!
zone local VZ Certkiller Certkiller 10.24.10.43
zone remote VZ Certkiller Certkiller 10.24.10.44 in Z1 Certkiller
zone remote Z1AJAX ajax 10.24.10.144 1079 in Z1 Certkiller
zone prefix Z1AJAX 919*
B. gatekeeper
!

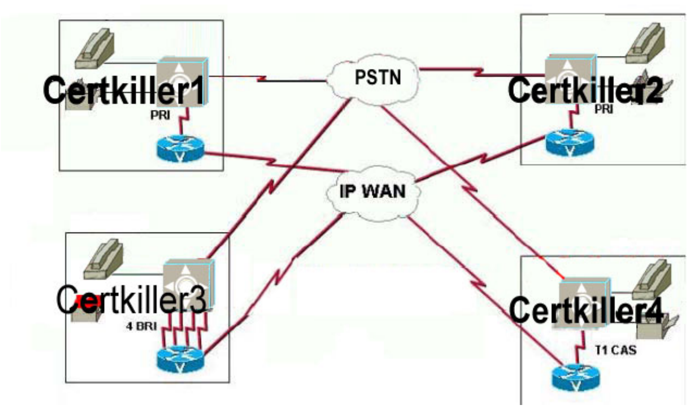
zone local Z1AJAX ajax 10.24.10.44
zone remote VZ Certkiller Certkiller 10.24.10.43 invia Z1AJAX
zone remote Z1AJAX ajax 10.16.8.144 1719 outvia Z1AJAX
zone prefix Z1AJAX 919*
C. gatekeeper
!
zone local Z1AJAX ajax 10.24.10.44
zone remote VZ Certkiller Certkiller 10.24.10.43 in Z1AJAX
zone remote Z1AJAX ajax 10.16.8.144 1719 out Z1AJAX
zone prefix Z1AJAX 919*
D. gatekeeper
!
zone local VZ Certkiller 10.24.10.43
zone remote Z1 Certkiller Certkiller 10.24.10.44 invia VZ Certkiller
zone remote Z1AJAX ajax 10.16.8.144 1719 outvia Z1 Certkiller
zone prefix Z1AJAX 919*

Answer: D

Explanation:
Cisco Multiservice IP-to-IP Gateway and Gatekeeper Design
- Via-zones are a new concept designed to assist in configuring an IP-to-IP gateway.
- IP-to-IP gateways register with the gatekeeper as an IP-to-IP gateway via-zones
The important thing to notice in the figure is that the gatekeeper called GK is configured to point the IP-to-IP gateway to process voice calls. The gatekeeper points to the IP-to-IP gateway via the commands invia and outvia. Also, note that the zone local specifies a zone controlled by a gatekeeper, thus obviously the zone local is VZ Certkiller .
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 6-25 Introducing Service Provider Offerings

QUESTION 12:

Exhibit



You are working with the customer Certkiller .com who is using QSIG to create a PINX between the four PBX locations. Certkiller 1, Certkiller 2, and Certkiller 4 have PRI connections to the PSTN, and Certkiller 3 as four BRI connections to the PSTN. Certkiller .com would like to start using the existing IP WAN to do toll bypass. The Certkiller 1 and Certkiller 4 locations use the same PBX manufacturer and Certkiller 2 uses a different manufacturer. In testing this solution, the Certkiller 2 location can communicate with the PSTN, but is not communicating with the local IP WAN gateway. The PBX in Certkiller 2 requires the PRI to be DMS-100 user-side signaling and is using 16 channels.

```
Certkiller 2#show running-configuration
!
controller t1 1/0:23
pri-group timeslots 1-16, 24
!
interface serial 1/0:23
isdn switch-type basic-5ess
isdn contiguous-bchan
isdn protocol-emulate network
isdn incoming-voice voice
!
end
```

From the above router configuration, which three commands need to be edited to resolve this issue? Select three.

A. The idsn contiguous-bchan command should be removed because it is only relevant to E1 interfaces.
B. The interface serial 1/0:23 command should actually be 1/0:16 because only 16 voice channels are being used.
C. The isdn protocol-emulate network command needs to be changed to isdn protocol-emulate user.
D. The configuration is missing the line code and framing statements.
E. The switch-type command needs to be changed to basic-dms100.

Answer: A, D, E

Explanation:
test-router(config-controller)#pri-group timeslots 1-16
The IOS will automatically append the ,24 so the controller t1 will be displayed in the start and running configuration as: pri-group timeslots 1-16,24
Note that after configuring the pri-group command, the D channel (interface serial 1/0:23) and the voice port (voice-port 1/0:23) are created automatically by the router.
Complete instructions for the configuration of the ISDN PRI voice-interface support can be found at
http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vvfax_c/isdnv_c/isdn01.htm#wp1038
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QUESTION 13:

Which two features are benefits of using HSRP for gatekeeper redundancy? Select two.

A. The gatekeepers can load balance

B. The HSRP priority can be adjusted.
C. End devices do not have to re-register after a primary gatekeeper failure.
D. The gatekeepers may be located across WAN links for spatial redundancy.
E. It uses a virtual IP address that is shared between the devices.

Answer: B, E

Explanation: Select one interface on each gatekeeper to serve as the HSRP interface and configure these two interfaces so that they belong to the same HSRP group but have different priorities. The one with the higher priority will be the active gatekeeper; the other assumes the standby role. Make a note of the virtual HSRP IP address shared by both of these interfaces. (For details on HSRP and HSRP configuration, refer to the Cisco IOS IP Configuration Guide.) If the primary gatekeeper fails in an HSRP redundancy model, the failure is transparent to the endpoint because the endpoints are pointing to the virtual HSRP router.
http://www.cisco.com/en/US/customer/products/sw/iosswrel/ps1835/products_configuration_guide_chapter0918
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pg 5-108 Deploying Gatekeepers

QUESTION 14:

Which configuration would be used to support a company whose DID ranges come out of multiple area codes served by the same PSAP?

A. voice-port 0/0
ani mapping 0 408
ani mapping 1 510
ani mapping 2 650
ani mapping 3 415
signal cama KP-2-ST
B. voice-port 0/0
ani mapping 0 408
ani mapping 1 510
ani mapping 2 650
ani mapping 3 415
signal cama KP-NPD-NXX.-XXXX-ST
C. translation-rule 1
rule 1^408 0
rule 2^510 0
rule 3^650 0
rule 4^415 0
voice-port 0/0
signal cama KP-NPD-NXX.-XXXX-ST
translate calling 1
D. translation-rule 1

rule 1 0 408 0
rule 2 1 510 0
rule 3 2 650 0
rule 4 3 415 0
voice-port 0/0
signal cama KP-2-ST
translate calling 1

Answer: B

Explanation:
Router(config-voiceport)# ani mapping NPD-value NPA-number
Use this command to build the table that translates the Numbering Plan Area (NPA), or area code, into a single MF digit. The number of Numbering Plan Digits (NPDs) that are programmed is determined by local policy and by the number of NPAs or area codes that the PSAP servers. The NPD value range is 0 to 3. The NPA number range is 100 to 999.
To disable ANI mapping, use the no form of this command.
Of the four CAMA signaling options for transmitting the calling number, the KP-NPD-NXX-XXXX-ST is the option used for an 8-digit ANI transmission in which the NPD is a single MF digit that is expanded into the NP
A. The NPD table is
preprogrammed by configuring ANI mapping in the sending and receiving equipment (on each end of the MF trunk);for example 0=408, 1=510, 2=650, 3=415
- NPD values range from 0 to 3. Examples of telephone numbers in this signaling option are:
* 05550123 = (408)555-0123
* 25550199 = (650)555-0199
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 2-43 to 46 Integrating a VoIP Network to the PSTN and PBXs

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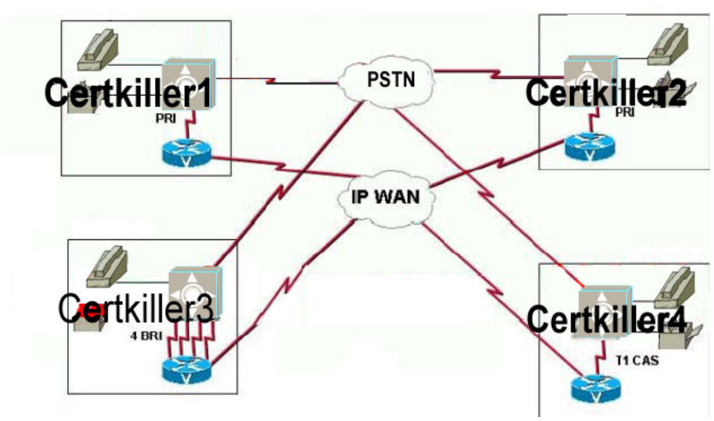
QUESTION 15:

Exhibit



You are working with the customer Certkiller .com who is using QSIG to create a PINX between the four PBX locations. Certkiller 1, Certkiller 2, and Certkiller 4 have PRI connections to the PSTN, and Certkiller 3

a 256-kbps connection to the PSTN. Certkiller .com would like to start using the existing IP WAN to do toll bypass. The Certkiller 1 and Certkiller locations use the same PBX manufacturer and Certkiller 2 uses a different manufacturer. The PBX in Certkiller 2 is configured for DMS-100 user-side signaling and is using 16 channels. The PRI is using ESF and B8ZS. Certkiller .com needs some help with configuring this gateway.
Which set of commands will make this operate correctly?

A. controller t1 1/0
pri-group timeslots 1-16, 24
!
interface serial 1/0:23
isdn switch-type basic-5ess
isdn continuous-bchan
isdn protocol-emulate network
no shut
B. controller t1 1/0
pri-group timeslots 1-16, 24
framing esf
linecode b8zs
!
interface serial 1/0:16
isdn switch-type basic-dms100
isdn protocol-emulate user
no shut
C. controller t1 1/0
pri-group timeslots 1-23, 24
framing esf
linecode hdb3
!
interface serial 1/0:23
isdn switch-type basic-net3
isdn protocol-emulate network
D. controller t1 1/0
pri-group timeslots 1-16, 24
framing esf
linecode b8zs
!
interface serial 1/0:23
isdn switch-type basic-qsig
isdn protocol.-emulate network
no shut
E. controller t1 1/0
pri-group timeslots 1-16, 24
framing esf
linecode b8zs
!

interface serial 1/0:23
isdn switch-type basic-dms100
isdn protocol.-emulate network
no shut

Answer: E

Explanation:
While in controller configuration mode, configure framing, linecode
and timeslots.
test-router(config-controller)#pri-group timeslots 1-16
This command shown above will automatically append the ,24 so the controller t1 will be
displayed in the start and running configuration as: pri-group timeslots 1-16,24
After configuring the pri-group command, the D channel (interface serial 1/0:23) and the
voice port (voice-port 1/0:23) are created automatically by the router.
Complete instructions for the configuration of the ISDN PRI voice-interface support can
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http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vvfax_c/isdnv_c/isdn01.htm#wp1038
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
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QUESTION 16:

Exhibit, Network Topology

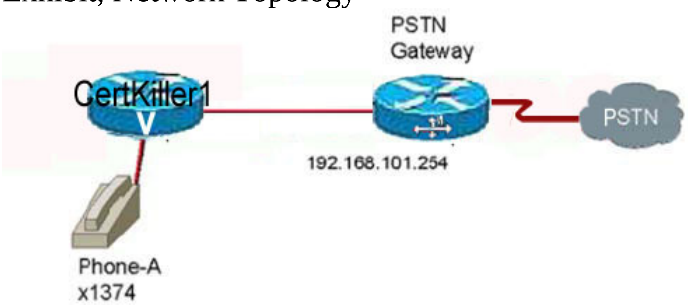

Exhibit #2

```
dial-peer cor custom
  name Emergency
  name Local
  name LD
  member Intl

dial-peer cor list LocalLst
  member Emergency

dial-peer cor list Local01
  member Local

dial-peer cor list LD01
  member LD

dial-peer cor list Intl01
  member Intl

dial-peer cor list LocalLst
  member Emergency
  member Local
dial-peer cor list IntlLst
  member Emergency
  member Local
  member LD

dial-peer cor list IntlLst
  member Emergency
  member Local
  member LD
  member Intl

dial-peer voice 1374 pots
  corlist incoming LocalLst
  destination-pattern 1374
  port 1/0/0

dial-peer voice 911 voip
  corlist outgoing Em01
  destination-pattern 911
  session target ipv4:192.168.101.254

dial-peer voice 7 voip
  corlist outgoing Local01
  destination-pattern 9[2-9]......
  session target ipv4:192.168.101.254
```

Certkiller .com has assigned extensions based on the dialing restrictions. All users in the range of 1000 to 1999 are to be set up so that they can dial only emergency and local calls via the PSTN. Given the configuration of Certkiller 1, which types of calls can Phone-A actually make via the PSTN?

A. None
B. emergency calls only
C. emergency calls and local calls only
D. emergency calls, local calls, and long-distance calls only
E. any calls

Answer: C

Explanation:
The Class of Restrictions (COR) feature provides the ability to deny certain call attempts
based on the incoming and outgoing CORs provisioned on the dial-peers.
COR is used to specify which incoming dial-peer can use which outgoing dial-peer to
make a call. Each dial-peer can be provisioned with an incoming and an outgoing COR list.
In the question, the corlist command sets the dial-peer COR parameter for dial-peers(VoIP)
and the directory number (1374) that is created for analog phone connected to Cisco VoIP gateway.

COR functionality provides the ability to deny certain call attempts on the basis of the incoming and outgoing class of restrictions that are provisioned on the dial-peers.
If the COR applied on an incoming dial-peer (for incoming calls) is a super set or equal to the COR applied to the outgoing dial-peer (for outgoing calls), the call goes through.
Based on the above rule , Phone A only can make emergency calls and local calls successfully..

QUESTION 17:

Which configuration will provision an E1 for ITU Q421 digital line signaling and compelled register signaling?

A. controller e1 1/0
cas-group 1 timeslots 1-31 type r2-digitial r2-compelled ani
B. controller e1 1/0
cas-group 1 timeslots 1-31 r2-compelled ani
C. controller e1 1/0
cas-group 1 timeslots 1-31 r2-digital ani
cas-custom 1
signaling r2-compelled
D. controller e1 1/0
cas-group 1 timeslots 1-31 type r2-compelled
cas-custom 1
signaling r2-digital

Answer: A

Explanation:
R2 inter-register signaling: These signaling types are configured using the cas-group (controller e1) command.
The three signaling types are described as follows:
Line signaling includes the following types:
- R2 digital:R2 line-signaling type ITU-U Q.421 is typically used for PCM systems (where A- and B-bits are used).
- R2 analog:R2 line-signaling type ITU-U Q.421 is typically used for carrier systems (where a tone A-bit is used).
- R2 pulse:R2 line-signaling type ITU-U Supplement 7 is typically used for systems that employ satellite links (where a tone A-bit is pulsed).
R2 compelled: When a tone pair is sent from the switch (forward signal), the tones stay on until the remote end responds (by sending an acknowledgment [ACK]). The remote responds with a pair of tones that signals the switch to turn off the tones. The tones are compelled to stay on until they are turned off.. Additional information on E1R2 can be

found in E1 R2 Signaling Theory at
http://www.cisco.com/en/US/tech/ CK6 52/ CK6 53/technologies_tech_note09186a00800943c2.shtml
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
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QUESTION 18:

What is a benefit of implementing an IP-to-IP gateway?

A. provides IP network privacy and trust boundary for security
B. offers equivalent quality and cost savings when compared to back.-to-back gateways
C. works in conjunctions with gateway proxies to provide scalable video solutions
D. enhances policy-routing capability by assigning carrier IDs to partner gateways

Answer: A

QUESTION 19:

A gateway is connected to the PSTN via eight analog circuits. Which two methods can be used to route calls across all eight circuits? Select two.

A. For each destination pattern, configure eight dial peers pointing to each port and use the preference command to select which port is used.
B. Assign each port to a trunk group. For each destination pattern, configure a single dial peer pointing to the trunk group.
C. Assign each port to a hunt list. For each destination pattern, configure a single dial peer pointing to the hunt list.
D. For each destination pattern, configure a single dial peer. Use the port range command to select which port is used.
E. For each destination pattern, configure eight dial peers pointing

Answer: A, B

QUESTION 20:

A service provider wants to add SIP devices to the existing H.323 voice network. Which Cisco device will allow the SIP devices to use the existing routing structure on the H.323 gatekeeper?

A. SIP voice gateway
B. Cisco SIP Proxy Server
C. Cisco SIP Redirect Server
D. Cisco SIP Registrar Server

Answer: B

QUESTION 21:

Exhibit
voice translation-rule 1
rule 1/^\(617\)\(...\)\(...\)//3\2\3/
What will be the result if the number 617-269-1212 is put through the Cisco IOS software voice translation rule displayed in the exhibit?

A. 36172691212
B. 1212269
C. 32691212
D. 6171212

Answer: C

Explanation: In general form , rule precedence /match pattern/ /replacement pattern/ In the question, 1/^\(617\)\(...\)\(...\)/ /3\2\3/ is the given translation rule. Hence, Rule number : 1 match pattern : ^\(617\)\(...\)\(...\) replacement pattern: 3\2\3
The match pattern says that the rule will apply to any pattern begins with 617. The replacement pattern says add digit 3 and insert set 2 and insert set 3 digits in the match pattern. In the question, the number given is 617-269-1212 . set 1 : 617 set 2 : 269 set 3 : 1212
Based on the replacement rule , the answer is C : 32691212.

QUESTION 22:

Which is a use of GKTMP?

A. updating information in directory gatekeepers
B. providing SIP to H.323 protocol translation for RAI in SIP networks
C. querying third-party applications to implement specific policy controls
D. transmitting state information to alternate gatekeepers for load balancing

Answer: C

Explanation: Gatekeeper Transaction Message Protocol (GKTMP) can extend the call control intelligence of a gatekeeper by providing an interface to a route application server where advanced routing decisions can be made. It converts incoming RAS messages to text messages and sends them to an off-board server. The server can override default gatekeeper behavior.
GKTMP is an independent platform and can run on Solaris, Linux, Microsoft Windows NT. It allows third parties to develop sophisticated applications to control RAS communication. An example of the use of GKTMP is where a service provider wants to control the call routing behavior of certain calls during a certain time of the day. The gatekeeper in this case will offload the routing instructions to the route application server and process the request from the server for altered call routing behavior.

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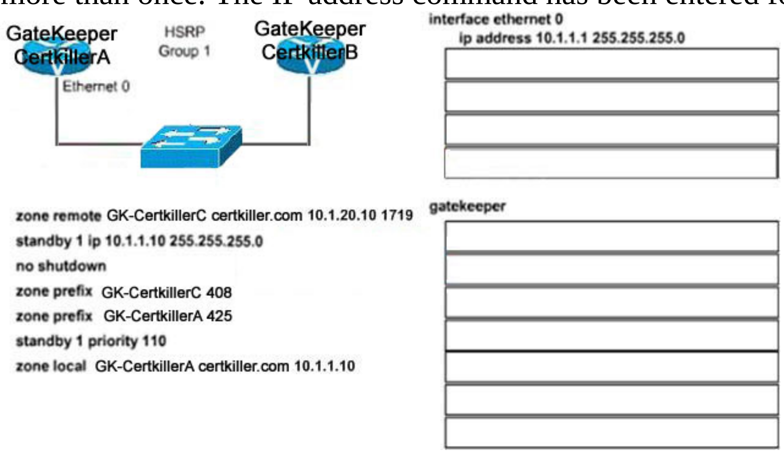
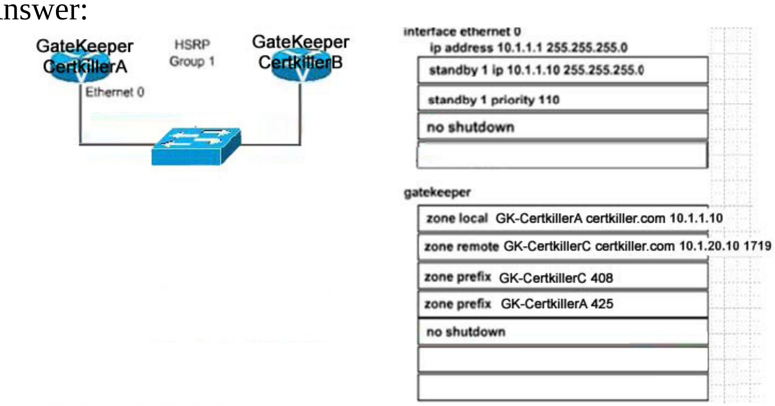
pg 5-4 Deploying Gatekeepers

QUESTION 23:

Drag and drag the commands to the proper location. Each command may be used either in the interface or the gatekeeper configurations, or in both. All boxes may not be filled and commands may be used more than once. The IP address command has been entered for you.



Answer:



Explanation: The primary gatekeeper will contain the following configuration on the ethernet interface.
interface ethernet 0
ip address 10.1.1.1 255.255.255.0
stanby 1 ip 10.1.1.10
standby 1 preempt
standby 1 timers 5 15
standby 1 priority 110
no shutdown
Though not shown in the list of commands to choose, the timers are very important, if the values are not sent to the same number in both routers, HSRP will not function properly. This router will be the priority gatekeeper for HSRP because of the standby 1 priority 110 statement. The default value is 100 in a range from 1 to 255. The standby router (alternate) will not need to have the standby priority statement. The standby router

(alternate) assumes the default priority of 100, meaning the primary gatekeeper is the one set with the higher priority at 110.
The gatekeeper mode will contain the following configuration:
gatekeeper
zone local GK- Certkiller A Certkiller .com 10.1.1.10
zone remote GK- Certkiller C Certkiller .com 10.1.20.10 1719
zone prefix GK- Certkiller A 408
zone prefix GK- Certkiller C 425
no shutdown
gatekeeper is the command to enter gatekeeper configuration. The zone local command specifies a zone controlled by a gatekeeper, include the gatekeeper name or zone name. Next, specify the domain name. Optionally specify the ras-ip-address, which is an ip address of one of the interfaces on the gatekeeper. Configure the zone remote. Then the zone prefix is the part of the called number that identifies the zone to which the call hops off.
http://www.cisco.com/en/US/customer/products/sw/iosswrel/ps1835/products_configuration_guide_chapter0918

QUESTION 24:

You have a client that is a national organization that has deployed an IP telephony network across all of the offices. The organization is divided into geographic regions. These regions include the east, the Midwest, and the west. The organization would like to deploy a directory gatekeeper to provide dial-plan resolution for all of the regions.
Which three statements correctly describe a DGK solution? (Choose three.)

A. Provides fault tolerance through a full mesh of regional gatekeepers
B. Allows up to a four-tier gatekeeper hierarchy to be deployed
C. Simplifies regional gatekeeper provisioning
D. Does not limit the number of hops in an LRQ
E. Allows local zones and LRQ forwarding zones to be mixed
F. The directory-gatekeeper maintains states about the forwarded-LRQ calls.

Answer: B, C, E

Explanation: There is a limit of five hops for an LRQ message, which allows up to a four-tier gatekeeper hierarchy.
Without a directory gatekeeper, you would have to add an entry in every gatekeeper on the network every time you add a new zone on one of the gatekeepers. However, with a directory gatekeeper, you can add the new zone in the local gatekeeper and the directory gatekeeper only.
A directory gatekeeper can be used to manage multiple gatekeepers in the network. LRQ forwarding allows a gatekeeper to be appointed as the directory gatekeeper or super gatekeeper. With this feature, it is only necessary to configure each gatekeeper with its own local zones and zone prefixes, and a single match-all wildcard prefix for the zone of

the directory gatekeeper. Only the directory gatekeeper has to be configured with the full set of all zones and zone prefixes within the network.

QUESTION 25:

Certkiller .com t is integrating a Cisco CallManager system with the existing PBX via an E1 QSIG trunk. During testing, the first 15 calls work normally. After 15 simultaneous calls, new calls have no audio path when they are established.
How can this problem be resolved?

A. Add the command isdn contiguous-bchan to the serial interface.
B. Change the channel selection order from descending to ascending.
C. Add the command isdn negotiate-bchan to the serial interface.
D. Increase the ISDN T302 timer to allow more time for call setup.

Answer: A

Explanation:
Router(config-if)#isdn contiguous-bchan
(E1 only) Specifies contiguous B channel handling so that B channels 1 to 30 map to timeslots 1 to 31 and skip timeslot 16. This command was added to allow interoperability with Siemens PBXs, which number the B channels consecutively from 1 to 30 instead of from 1 to 15 and 17 to 31.
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QUESTION 26:

Exhibit

```
set status [infotag get evt_status]
if { $status == "ls_000"} {
set creditTimeLeft [infotag get
leg_settlement_time leg_outgoing]
if { ($creditTimeLeft == "unlimited")
||
($creditTimeLeft == "uninitialized") }
{
puts "\n Unlimited Time"
} else {
# start the timer for ...
if { $creditTimeLeft < 10 } {
set beep 1
set delay $creditTimeLeft
} else {
set delay [expr $creditTimeLeft - 10]
}
timer start leg_timer $delay
leg_incoming
}
} else {
puts "Call [infotag get con_all] got
event $status while placing an outgoing
call"
call close
}
}
```

Refer to the exhibit. What is the purpose of the TCL script snippet?

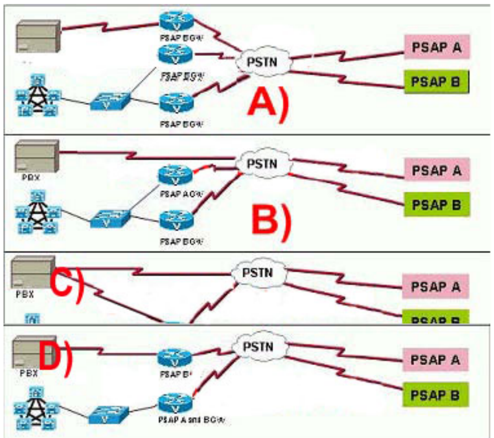
A. process a script exit
B. play an audio prompt
C. terminate a call
D. gather initial digits

E. interrupt a call in progress

Answer: C

QUESTION 27:

Exhibit



Refer to the exhibit. You have a client that has a campus that straddles two different PSAP areas for E911 calls. The current single PBX has been servicing all of the telephones on the campus. The client is now interested in looking at a design for an IP telephony solution.
Which solution will provide seamless connectivity between the PBX and an IP telephony processor and both PSAPs?

A. A
B. B
C. C
D. D

Answer: C

QUESTION 28:

Which dial peer will send calls to the PSTN via the CAS T1 using this controller configuration?
controllert1 3/0
framingesf
linecodeb8zs
ds0-group1 timeslots 1-24 type e&m-wink-start

A. dial-peer voice 1 pots
destination-pattern 9.@
port 3/0:1
B. dial-peer voice 1 pots
destination-pattern 9.@
port 3/0:24
C. dial-peer voice 1 pots

destination-pattern 91
port 3/0:1
D. dial-peer voice 1 pots
destination-pattern 91
port 3/0:24

Answer: C

Explanation:
The syntax that begins: ds0-group 1 timeslots uses the group number 1. This group number will appear in the port number used for the POTs dial-peer. The pots dial-peer will reference the port and the group number. For example:
dial-peer voice 9 pots
destination-pattern 9T
port 3/0:1 (the 1 is the group number from the syntax that begins ds0-group 1 timeslots)
[output from router]
test-router(config-dial-peer)#destination-pattern 9.@
Incorrect format for E.164 Number
regular expression must be of the form ^[][^0-9,A-F#*.?+%()-]*T?(\$)?$
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
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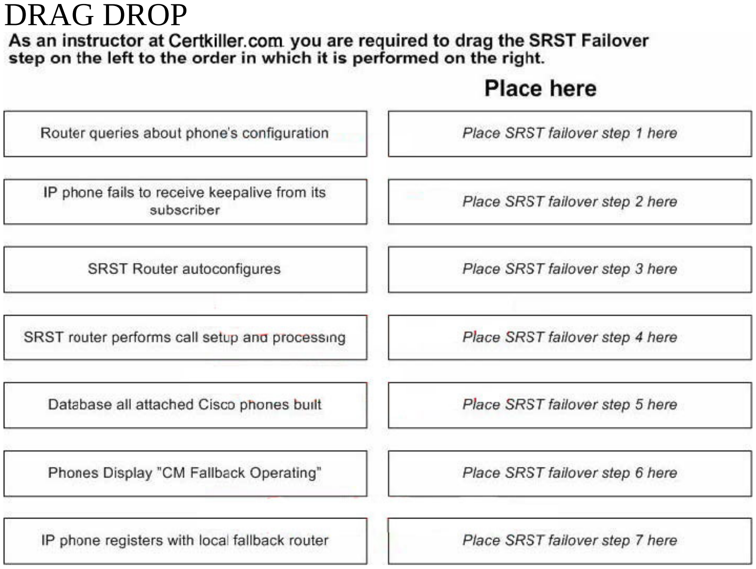
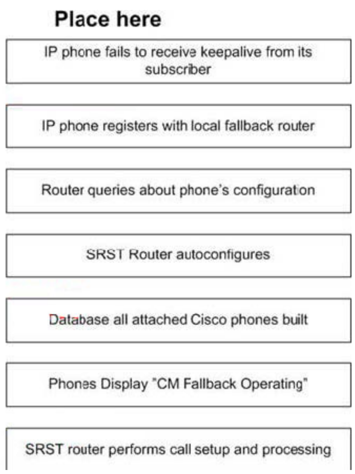
QUESTION 29:

DRAG DROP



Answer:



Explanation: When the WAN link fails, the Cisco (1) IP phones detect that they are no longer receiving keepalive packets from the Cisco CallManager. The (2) IP phones then register with the router, which (3) queries the phone about its configuration and (4) then autoconfigures itself. In this instance, the SRST is automatically activated and (5) builds a local database of all IP phones attached to up to its stated maximum. The IP phones are configured to query the router as a backup call-processing source when the central CallManager does not acknowledge keepalive packets. The (6) IP phones indicate on their display that they are in "CM Fallback Operating" mode for the duration of the failure. The (7) SRST router now perform call setup, call processing, call maintenance, and call termination.

QUESTION 30:

You have a client that is planning to deploy IP telephony in its European organization. The multiple locations will require gateway support. The organization has a list of features that need to be supported. Given the following list of features, which gateway protocol would you choose for this implementation?
BRI
E1 CAS
E1 QSIG
Fax relay
Modem realay

A. H.323
B. MGCP
C. SCCP
D. SIP
E. H.245

Answer: A

QUESTION 31:

Which three features are available during SRST failover? (Choose three.)

A. music on hold
B. IP phone speed dial
C. Distinctive ring
D. Call forwarding

Answer: A, B, C

QUESTION 32:

Exhibit

```
proc init { } {
global param
set param(interruptPrompt) true
set param(abortKey) *
set param(teterruptPrompt)#
}

proc init { } { } {
global dest
global beep
set beep 0
leg setupack leg_incoming
if { [infotag get leg_isdid] } {
set dest [infotag get leg_dnis]
leg proceeding leg_incoming
leg setup $dest callInfo leg_incoming
fsm setstate PLACECALL
} else {
playtone leg_incoming tn_dial
set param(dialPlan) true
leg collectdigits leg_incoming param
}
}
```

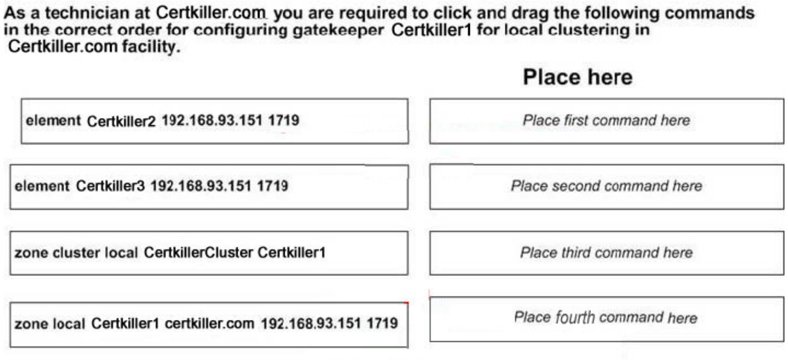
Refer to the exhibit. What is the purpose of the TCL script snippet?

A. process a script exit
B. play an audio prompt
C. terminate call
D. gather initial digits
E. interrupt a call in progress

Answer: D

QUESTION 33:

DRAG DROP,
Exhibit


As a technician at Certkiller.com you are required to click and drag the following commands in the correct order for configuring gatekeeper Certkiller1 for local clustering in Certkiller.com facility.

Place here

element Certkiller2 192.168.93.151 1719	Place first command here
element Certkiller3 192.168.93.151 1719	Place second command here
zone cluster local CertkillerCluster Certkiller1	Place third command here
zone local Certkiller1 certkiller.com 192.168.93.151 1719	Place fourth command here

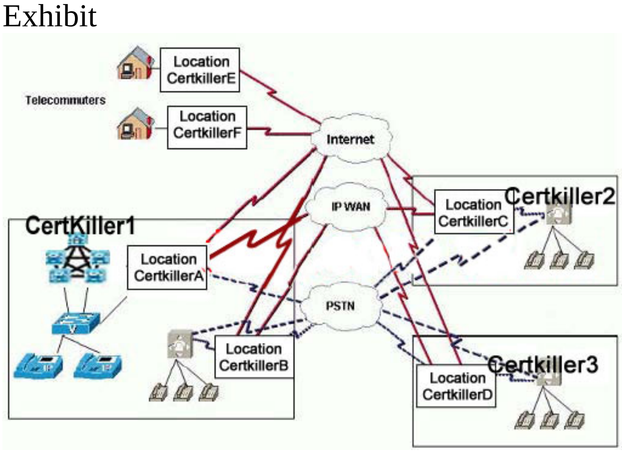
Answer:


Place here

zone local Certkiller1 certkiller.com 192.168.93.150 1719
zone cluster local CertkillerCluster Certkiller1
element Certkiller2 192.168.93.151 1719
element Certkiller3 192.168.93.151 1719

QUESTION 34:

Exhibit



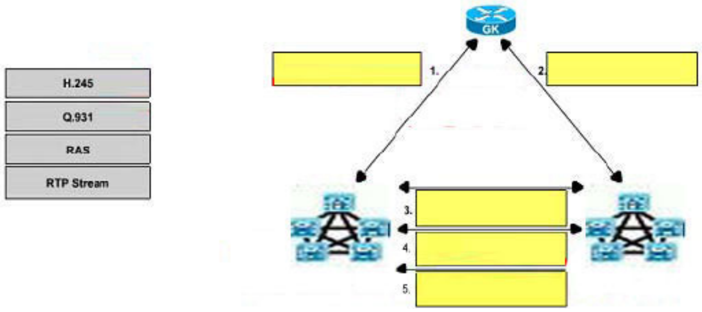
Refer to the exhibit. Certkiller .com is in the process of migrating from a traditional PBX telephony system to an IP telephony system at the Certkiller 1 headquarters. Certkiller .com would like to start migrating the regional offices in Certkiller 2 and Certkiller 3 off the existing tie-line and onto the IP WAN. In witch locations would voice-enabled gateways need to be deployed? (Choose four)

A. location Certkiller A
B. location Certkiller B
C. location Certkiller C
D. location Certkiller D
E. location Certkiller E
F. location Certkiller F

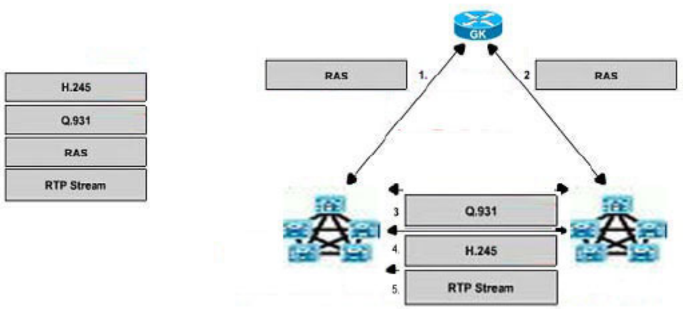
Answer: A, B, C, D

QUESTION 35:

DRAG DROP
As a network technician at Certkiller .com you are required to click and drag the call-setup messages and voice stream to the correct location on the diagram. The diagram is labeled in the orderthat the messages are sent, and some items may be used more than once.



Answer:



Explanation:
The initial signaling from a gateway to a gatekeeper is done through H.225 RAS.
Gateways can discover their gatekeepers through one of two processes. Unicast (UDP port 1718) or mulicast discovery (UDP multicast address 224.0.1.41).
Gateway-to-gateway signaling is H.225 call control or setup signaling. H.225 call control signaling is used to set up connections between H.323 endpoints. The ITU H.225 recommendation specifies the use and support of Q.931 signaling messages.
A reliable TCP call control channel is created accross an IP network on TCP port 1720.
This port initiates the Q.931 call control messages for the purpose of connecting, maintaining, and disconnecting calls.
Once call signaling is set up between the gateways, H.245 is negotiated. H.245, a control signaling protocol in the H.323 multimedia communication architecture is for the exchange of end-to-end H.245 messages between communicating H.323 endpoints or terminals.
http://www.cisco.com/en/US/tech/ CK1 077/technologies_tech_note09186a00800c5e0d.shtml

QUESTION 36:

Certkiller .com has configured two T1 trunks on a gateway. The first trunk connects to a call center and sends ANI to differentiate how each call is handled. The second trunk connects to the PSTN for all outgoing calls, but the local PSAP is unable to receive ANI from this trunk.
What can be done to resolve this issue?

A. configure an H.323 gateway so that the T1 to the call center has two DS-0 groups, one to send ANI and one to receive ANI
B. configure an MGCP gateway so that both T1s have two DS-0 groups each, one to send ANI and one to receive ANI
C. configure an H.323 gateway so that there are two DS-0 groups on the T1 from the PSTN, one to send ANI and one to receive ANI
D. configure an H.323 gateway so that the T1 from the PSTN receives ANI and the T1 to the call center sends ANI
E. configure an H.323 gateway so that there are four DS-0 groups; each T1 will have a DS-0 group that will send ANI and one that will receive ANI

Answer: C

Explanation:
If a location needs to send and receive ANI on a single T1, two DS-0 groups must be configured. One ds-0 group is configured with E&M-FGD to receive ANI and the second is configured as FGD-EANA to send ANI. The two separate ds-0 groups are configured on the same T1 controller.
Example:
ds-0 group (1-12) is configured with type e&m-fgd to receive ANI
ds-0 group (13-24) is configured as fgd-eana to send ANI
(config)#controller T1 0
(config-controller)#ds0-group 1 timeslots 1-12 type e&m-fgd
(config-controller)#ds0-group 2 timeslots 13-24 type fgd-eana
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QUESTION 37:

Certkiller .com is integrating the existing PBX to Cisco CallManager and is planning to use QSIG to support MWI. Which gateway protocol should be deployed?

A. H.225.0
B. H.345
C. H.323
D. MGCP
E. SCCP
F. SIP

Answer: D

Explanation: To implement the QSIG protocol with Cisco CallManager, the
gateway must run in Media Gateway Control Protocol (MGCP) mode. Once you
configure MGCP, you can setup the supplementary services. The most common
supplementary services used with CallManager are identification services, Message
Waiting Indicator (MWI) services, call diversion, and call transfer.
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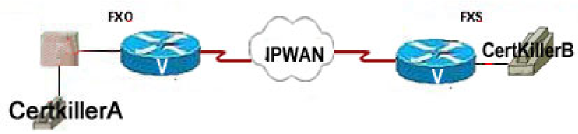
pg 2-168 Integrating a VoIP Network to the PSTN and PBXs

QUESTION 38:

Exhibit



Refer to the exhibit and the following steps for a call placed between Phone Certkiller A and Phone Certkiller B.
Phone Certkiller A calls Phone Certkiller B.
If Phone Certkiller B does not answer, Phone Certkiller B continues to ring even if Phone Certkiller A hangs up.
If the call is answered, it stays active until Phone Certkiller B hangs up, regardless of the actions of Phone Certkiller A.
How can this problem be resolved?

A. the amount of time that the PBX provides power denial is too long to be recognized by the FXO port
B. if the PBX is capable of supporting ground-start signaling, have the FXO port use this feature to receive signal disconnect from the PBX
C. in the configuration of the FXO port turn off tone based supervisory disconnect, this works only with FXS ports.
D. Configure battery reversal on the FXO port so the PBX is aware when Phone Certkiller B hangs up

Answer: B

Explanation:
The following is an example to configure ground-start signaling as the
signaling type for
a voice port such as FXO , which means that both sides of a connection can
place a call
and hang up:
Router(config)# voice-port 1/1/1
Router(config-voiceport)# signal ground-start

QUESTION 39:

Certkiller .com determines that all long-distance calls to area code 603 will route across the WAN. The destination gateway is 10.172.163.5 connected through serial interface 1/0. Which set of Cisco IOS commands will accomplish this?

A. dial-peer voice 100 pots
destination-pattern 1603.......
port ipv4:10.172.163.5
B. dial-peer voice 101 voip
destination-pattern 1603.......
port 1/0
C. dial-peer voice 102 voip
destination-pattern 1603.......
session-target ipv4:10.172.163.5
D. dial-peer voice 103 pots
destination-pattern 1603.......
session-target 1/0

Answer: C

QUESTION 40:

You are working with Certkiller .com who is interested in deploying a distributed IP telephony call-processing solution among the three corporate campuses. Each campus will have a Cisco CallManager cluster and a gateway to the PSTN. Which three design considerations need to be determined or selected for a successful gatekeeper deployment? (Choose three.)

A. determine if the total WAN bandwidth for voice and data will exceed 75% of total link capacity
B. calling patterns to the PSTN
C. intersite modem and fax traffic patterns
D. a common codec for all WAN connections
E. the correct WAN topology
F. if gatekeeper redundancy or high availability is required

Answer: D, E, F

Explanation: Gatekeepers are one of the key elements in the multisite WAN model with distributed call processing. Each gatekeeper provides dial-plan resolution and call access control. The following best practices apply to the use of a gatekeeper.
* Use a logical hub-and-spoke topology for the gatekeeper. A gatekeeper can manage the bandwidth into and out of a site, or between zones within a site, but it is not aware of the topology (you will need to consider the proper WAN topology to support the gatekeeper(s)).
* Size the platforms appropriately to ensure that performance and capacity requirements can be met.
* When deploying voice in a WAN environment, Cisco recommends that you use the lower bandwidth g.729 codec for any voice calls that will traverse WAN links because this pratice will provide bandwidth savings on these lower-speed links (need to consider a common codec for all WAN connections).
* Gatekeeper networks can scale to hunderds of sites, and the design is limited only by

the hub-and-spoke topology.
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 1-14 Function of Gateways and Gatekeepers

QUESTION 41:

Certkiller .com field offices route calls to headquarters out IP gateways to the PSTN. The numbers are all of the form 1-202-454-XXXX. When dialing, the filed offices wish to dial only the last four digits. Which of the following Cisco IOS command must be a part of the PSTN dial peer on the field office gateways?

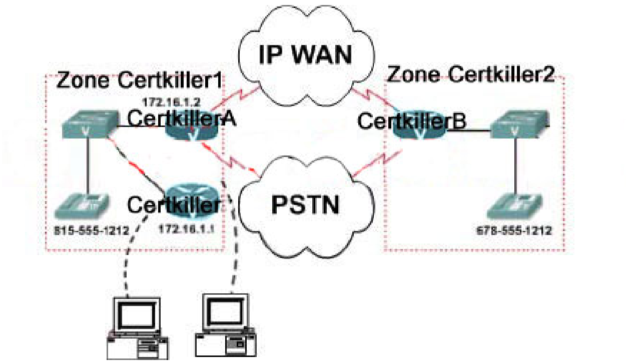
A. no digit-strip
B. prefix 1
C. num-exp .... 1202454....
D. Rule 1 ^202454 1

Answer: C

Explanation: Number expansion is a globally applied rule that enables you to define a set of digits for the gateway to prepend to the beginning of a dialed string before you pass it to the remote telephony device. This procedure reduces the number of digits that a user must dial to reach a remote location. Number expansion is similar to using a prefix, except that number expansion is applied globally to all dial-peers and the expansion is applied before the outbound dial-peer is matched.
[router output]
dial-peer voice 2003 voip
destination-pattern 1202.......
session target ipv4:10.10.10.81
!
num-exp .... 1202454....
If you dial any four digit number 0000 to 9999, the num-exp will prepend the 1202454 and it will match the dial-peer voice 2003 voip. So for example, you dial 0007,the number is converted to 12024540007 and will be matched to the dial-peer 2003 voip and will be sent to CallManager at the ip address 10.10.10.81

QUESTION 42:

SIMULATION
Exhibit, Network topology



Certkiller .com is looking to do a demonstration of a gateway registering with a gatekeeper. They have a small VoIP network in their lab to test their configurations. There are two routers that will be configured in this simulation. They are designated Certkiller A and Certkiller . The Certkiller A router has a POTS connection to the PSTN. It also has an Ethernet connection that is shared with Certkiller . Certkiller will need to be configured for two zones designated Certkiller 1 and Certkiller 2. Certkiller will need to provide Call Admission Control (CAC) for inter-zone calls. Configure Certkiller to allow 64kbps of bandwidth per call and permit enough bandwidth for thee calls in the zone.
The domain will be Certkiller .com. This test will also include a technology prefix. Use technology prefix 1# as the default. Modify Certkiller A VoIP dial peer to use Certkiller for registration authentication and status.
1. Configure Certkiller to accept registrations
2. Configure Certkiller A to register with Certkiller using an ID of Certkiller A and a domain of Certkiller .com. Certkiller A should be in the Certkiller 1 zone.
3. Configure Certkiller A to resolve calls to the 678 area code via Certkiller .
4. Configure Certkiller for two local zones. Certkiller 1 and Certkiller 2.

Answer:
<Gateway Router Certkiller A commands>
interface Ethernet0/0
ip address 172.16.1.2 255.255.0.0
half-duplex
h323-gateway voip interface
h323-gateway voip id Certkiller ip address 172.16.1.1
h323-gateway voip h323-id Certkiller A@ Certkiller .com
h323-gateway voip tech-prefix 1#
no shutdown
!
gateway
!
dial-peer voice 1 pots
destination-pattern 815T
port 2/0/0
!
dial-peer voice 2 voip
destination pattern 678.......
session target ras

!
<Gatekeeper Router Certkiller commands>

gatekeeper
zone local Certkiller 1 Certkiller .com
zone local Certkiller 2 Certkiller .com
zone prefix Certkiller 1 815.......
zone prefix Certkiller 2 678.......
gw-type-prefix 1#* default-technology
bandwidth total zone Certkiller 1 192
bandwidth total zone Certkiller 2 192
no shutdown

QUESTION 43:

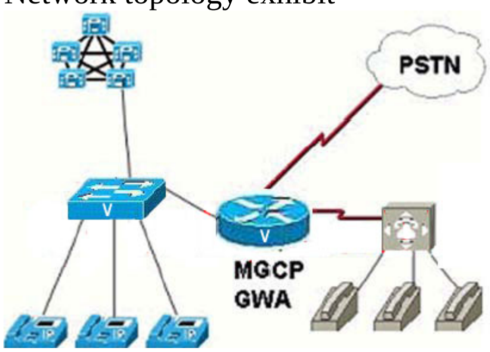
When COR is used in a gateway, under what circumstance will a call be completed between a specific pair of dial peers?

A. only when COR lists in the inbound and outbound dial peers are an exact match
B. when the COR list in the outbound dial peer is a subset of the COR list in the inbound dial peer
C. when the COR list in the inbound dial peer is a subset of the COR list in the outbound dial peer
D. when the COR list in the inbound and outbound dial peers have no matching members

Answer: B

QUESTION 44:

Network topology exhibit



Refer to the exhibit. Callers are complaining that the frequently get a busy signal when calling from IP phones to the PBX. Analysis of the problem shows that conflicting route patterns in the Cisco CallManager and the PBX are causing some calls to trombone across the T1 CAS connection.
Which action will prevent future problems with tromboning?

A. Implement COR in the gateway to prevent calls originating on the PBX from being routed back to the PBX across the T1.
B. Make sure the gateway's CSS does not include any partions that have route patterns whose target is the gateway.

C. Make sure the gateway's partition does not include any route patterns whose target is the gateway.
D. Set the Forward Maximum Hop Count service parameter in Cisco CallManager to 2.

Answer: B

QUESTION 45:

Certkiller .com has two routes to send calls from its headquarters to its subsidiary.
Headquarters: Manchester, NH--603-643-XXXX
Subsidiary: Seattle, WA--206-532-XXXX
The connections are made as follows:
First choice: WAN over router 10.172.16.111
Second choice: PSTN on port 1/0:1
Which two sets of Cisco IOS commands will provide this routing from the headquarters to the subsidiary? (Choose two.)

A. dial-peer voice 10 pots
destination-address 1206532....
session-target port 1/0:1 preference 0
B. dial-peer voice 11 voip
destination-address 1206532....
session-target ipv4:10.172.16.111
preference 0
C. dial-peer voice 12 pots
destination-address 1206532....
session-target port 1/0:1
preference 1
D. dial-peer voice 13 voip
destination-address 1206532....
session-target ipv4:10.172.16.111
preference 1

Answer: B, C

QUESTION 46:

When deployed in an enterprise network, which three features does an IP-to-IP gateway use support for video? (Choose three.)

A. endpoint functions
B. proxy functions
C. RSVP with call signaling
D. near-end camera control
E. simultaneous data, audio, and video conferencing
F. Microsoft NetMeeting data collaboration

Answer: B, C, E

Case Study # 1Certkiller com, Scenario

Certkiller .com is expanding their facility. As part of the expansion Certkiller .com plans to implement a CallManager system to provide phone support. The expansion will be phased over a 6 month period requiring the CallManager to be integrated with the existing PBX system. In order to minimize the impact on office staff, the office manager wants to only train employees on new phone procedures when they have been migrated to the CallManager system.
The existing voicemail will be used until all employees are migrated to the CallManager. Once the migration is complete, Cisco Unity will be used to provide Unified Messaging.
Certkiller .com currently has two T1s to the phone company for local and inbound calls and a T1 dedicated for Long Distance calling. There are 200 DIDs and Certkiller employees can call each other via four digit expressions. The migration schedule calls for moving the PSTN connections to the CallManager after all employees are migrated.
To facilitate the migration, a 3825 router will be used to provide voice gateway services. Two T1's will be used to connect the 3825 and the PBX. These T1s will be used to support calls between IP phones and PBX phones. The PSTN ** MISSING ***

Case study #1, (3 Questions)

QUESTION 47:

While testing calls between the CallManager and PBX, it is determined that IP phones can place and receive calls from the PBX phones but cannot receive calls from the PSTN on their DID.
What is the most likely cause of this problem?

A. The PBX is restricting trunk to trunk transfers
B. The PBX is not sending the correct digits to the CallManager.
C. The gateway's CSS does not contain the partition assigned to the IP phones.
D. The IP phones' CSS does not contain the partition assigned to the gateway.

Answer: A

QUESTION 48:

Which three statements describe how the T1s should be provisioned to provide maximum capacity while maintaining features such as voicemail support and callerid? Select three.

A. The T1s should be CAS circuits. One circuit should be configured for e&m-fgd and one circuit should be configured for fgd-eana.
B. The T1s should be CAS circuits configured for fxo-loop-start.
C. The T1s should be CCS circuits. The PBX should be configured for network side and the gateway for the user side.
D. The gateway controller's clock source should be configured as line.
E. The T1s should be configured as NFAS to provide maximum capacity.
F. The T1s should be configured for QSIG.

Answer: C, D, F

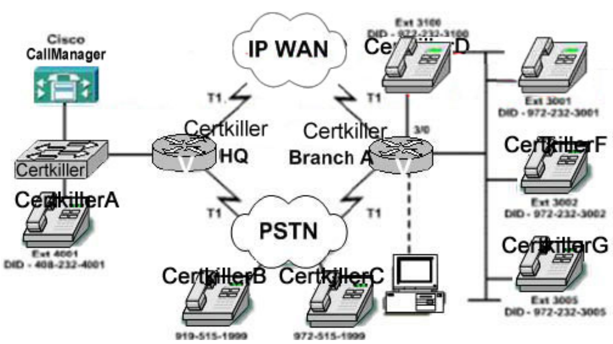
QUESTION 49:

While talking to customers, Certkiller sales staff frequently conference in technical support.
Certkiller .com would like to support 12 simultanous conference bridges with 4 participants each.
What is the minimum number of DSPs required in the gateway?

A. 2
B. 3
C. 5
D. 12
E. 15

Answer: C

Case Study #2, Scenario
Network Topology Exhibit:



Simulation output exhibit #1:

```
CertkillerBranchA#sho run
Building configuration...

Current configuration : 2532 bytes
!
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname CertkillerBranchA
!
enable password igwgk
!
tdm clock T1 1/0 both export line
voice-card 1
!
voice-card 2
!
voice-card 3
!
no aaa new-model
ip subnet-zero

no ip domain lookup
ip dhcp excluded-address 192.168.121.1 192.168.121.99
ip dhcp excluded-address 192.168.122.1 192.168.122.99
!
ip dhcp pool voice
   network 192.168.121.0 255.255.255.0
   default-router 192.168.121.1
   domain-name igwgk.com
   dns-server 10.1.1.200
   option 150 ip 10.1.1.100
!
ip dhcp pool data
   network 192.168.122.0 255.255.255.0
   default-router 192.168.122.1
   domain-name igwgk.com
   dns-server 10.1.1.200
!
!
ip cef
ip audit notify log
ip audit po max-events 100
ip multicast-routing
ip ssh break-string
no ftp-server write-enable
isdn switch-type primary-ni
!
!
!
ccm-manager fallback-mgcp
ccm-manager redundant-host 10.1.1.100
ccm-manager mgcp
```

Simulation output exhibit #2:

```
class-map match-all VOICE-CTRL-OUT
 match ip dscp af31
class-map match-all VOICE-OUT
 match ip dscp ef
class-map match-all VOICE
 match access-group name VOICE
class-map match-all VOICE-CTRL
 match access-group name VOICE_CTRL
!
!
policy-map INBOUNDCTRL
 class VOICE-OUT
  priority percent 50
 class VOICE-CTRL-OUT
  bandwidth 16
 class class-default
  fair-queue
policy-map INBOUND
 class VOICE
  set ip dscp ef
 class VOICE-CTRL
  set ip dscp af31
!
!
translation-rule 1
 Rule 1 3100 9722323100
!
!
!
!
interface Loopback254
 ip address 10.5.1.3 255.255.255.255
!
interface FastEthernet0/0
 description Trunk to 3550
 no ip address
 speed auto
!
interface FastEthernet0/0.1
 description Native VLAN
 encapsulation dot1Q 1 native
 ip address 192.168.120.1 255.255.255.0
 ip helper-address 10.1.1.100
 ip pim sparse-mode
 service-policy input INBOUND
 no cdp enable
!
interface FastEthernet0/0.121
 description Voice Interface
 encapsulation dot1Q 121
 ip address 192.168.121.1 255.255.255.0
 ip helper-address 10.1.1.100
 ip pim sparse-mode
 service-policy input INBOUND
!
interface FastEthernet0/0.122
 description Data Interface
 encapsulation dot1Q 122
 ip address 192.168.122.1 255.255.255.0
 ip helper-address 10.1.1.100
 ip pim sparse-mode
 service-policy input INBOUND
!
interface Serial0/0
 description F/R from Main
 no ip address
```

Simulation output exhibit #3:

```
interface Serial0/0.150 point-to-point
 ip address 10.12.1.2 255.255.255.0
 ip ospf network point-to-point
 frame-relay interface-dlci 150 ppp Virtual-Template150
!
interface Serial1/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-ni
 isdn incoming-voice voice
 isdn bind-l3 ccm-manager
 no cdp enable
!
interface Virtual-Template150
 ip address 10.3.1.2 255.255.255.0
 ip pim sparse-mode
 service-policy output MainToSite1
 ppp multilink
 ppp multilink fragment delay 10
 ppp multilink interleave
!
router ospf 1
 log-adjacency-changes
passive-interface FastEthernet0/0
network 10.0.0.0 0.255.255.255 area 0
network 192.168.120.0 0.0.0.255 area 120
network 192.168.121.0 0.0.0.255 area 120
network 192.168.122.0 0.0.0.255 area 120
!
ip classless
no ip http server
no ip http server server
!
!
!
ip access-list extended voice
 permit udp any any range 16384 32767
ip access-list extended VOICE-CTRL
 permit tcp any any range 2000 2002
 permit tcp any any eq 1720
 permit tcp any any range 11000 11999
 permit udp any any eq 2427
!
map-class frame-relay frts
 frame-relay cir 384000
 frame-relay bc 3840
 frame-relay be 0
 frame-relay cir    384000
!
interface
!
!
call application alternate default
!
voice-port 2/0
!
voice-port 2/1
!
voice-port 3/0
 station-id number 3100
 caller-id enable
!
voice-port 3/1
!
voice-port 3/2
!
```

Simulation output exhibit #4:

```
voice-port 1/0:23
!
mgcp
mgcp call-agent 10.1.1.101 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp modem passthrough voip mode nse
!
mgcp profile default
!
dial-peer cor custom
 name everyone
 name manager
 name admin
 name emergency
!
!
dial-peer cor list emergency
 member everyone
!
dial-peer cor list admin
 member admin
!
dial-peer cor list emergency
 member emergency
!
dial-peer cor list manager
 member manager
!
dial-peer cor list internal
 member emergency
!
dial-peer cor list local
 member everyone
 member emergency
!
dial-peer cor list ld
 member everyone
 member manager
 member admin
 member emergency
!
dial-peer cor list inbound
 member manager
 member admin
 member emergency
!
```

Case Study #2, 5 Questions

QUESTION 50:

Which configuration change will provide caller-name display for Phone Certkiller D?

A. voice-port 3/0 followed by caller-name Lobby
B. voice-port 3/0 followed by station-id name Lobby
C. dial-peer voice 3100 pots followed by caller-name Lobby
D. dial-peer voice 3100 pots followed by station-id name Lobby

Answer: B

QUESTION 51:

Phone Certkiller D has placed a call to 911. Which number will appear as the caller ID?

A. 3000
B. 3100
C. 9722323000
D. 9722323100

Answer: C

QUESTION 52:

Which two phones at Certkiller Branch A can be reached from the PSTN while operating in SRST mode? Select two.

A. Certkiller D
B. Certkiller E
C. Certkiller F
D. Certkiller G

Answer: B, C
Note: The exhibits shown are not enough to determine answer for this question.

QUESTION 53:

Certkiller Branch A experienced an IP WAN outage and is operating in SRST mode. Which three phones can Certkiller F call while in SRST mode? Select three

A. Certkiller A
B. Certkiller B
C. Certkiller C
D. Certkiller D
E. Certkiller G

Answer: A, C, E
Note: The exhibits shown are not enough to determine answer for this question.

QUESTION 54:

Which configuration will support fax pass-through while in MGCP mode? Select one

A. Enable fax pass-through using the mgcp command.
B. Enable fax pass-through using the ccm-manager command.
C. Disable fax relay using the ccm-manager command.
D. Disable fax relay using the mgcp command.

Answer: A
The command mgcp modem passthrough voip mode nse found in exihibit 4 supports fax pass-through in MGCP mode.