Exam : 642-453

Title : Gateway Gatekeeper(GWGK)

Ver : 11.23.07

QUESTION 1:

When configuring a gatekeeper-controlled H.255 trunk, what is the maximum number of Cisco Unified CallManager systems that can be assigned to the trunk?

A. 4
B. 1
C. 3
D. 6
E. 2
F. 8

Answer: C

QUESTION 2:

Which codec complexity should be configured to maximize the number of simultaneous voice calls that can be supported by a PVDM2-16?

A. Flex
B. High
C. Medium
D. Low

Answer: A

QUESTION 3:

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:RX -> INFO sapi   = 0  tei = 0  ns = 19  nr = 19  i =
0x08020089050403800x08020089050400x08020089050x08020089050403809
1d20h:      SETUP pd=8 called Rx 0089
1d20h:          Bearer capability 1 = 0x809013
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 <-INFO sapi = 0 tei 0  ps = 19  nr = 20  i =
0x080280895A08028286
1d20h:      RELEASE_COMP ps = 0 call ef = 078189
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
```

Please study the exhibit carefully. Certkiller .com is integrating a Cisco Unified 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 negotiate-bchan to the serial interface.
B. Increase the ISDN T302 timer to allow more time for call setup.

C. Change the channel selection order from descending to ascending.
D. Add the command isdn contiguous-bchan to the serial interface.

Answer: A

QUESTION 4:

Which two statements are correct descriptions of bandwidth management and bandwidth control? (Choose two.)

A. Both bandwidth management and bandwidth control are H.323 mandatory functions.
B. Bandwidth management is an H.323 optional function and bandwidth control is a mandatory function.
C. Bandwidth management manages endpoint bandwidth and provides intrazone call admission control, while bandwidth control provides interzone call admission control.
D. Bandwidth management manages endpoint bandwidth requirements and bandwidth control provides call admission control.
E. Bandwidth management provides call admission control and bandwidth control manages endpoint bandwidth requirements.
F. Bandwidth management is an H.323 mandatory function and bandwidth control is an optional function.

Answer: B,E

QUESTION 5:

Please study the exhibit carefully. Certkiller .com needs to have the receptionist at extension 5000 handle all callers who exit their auto attendant by pressing "0". After changing the Operator parameter from 5500 to 5000, callers are still sent to extension 5500 when they press "0" in the auto attendant. What needs to be done to correct this issue?

```
!
application
 service aa flash:its-CISCO.2.0.1.0.tcl
  paramspace english index 0
  paramspace english language en
  param operator 5000
  paramspace english location flash:
  paramspace english prefix en
  param an-pilot 5099
  !
  !output omitted
!
dial-peer voice 1 pots
 service aa
 incomming called number 5999
 !
```

A. The aa application needs to be reloaded in order to recognize the parameter changes.
B. The application needs to be edited to point to the correct operator.
C. Dial peer 1 needs to be restarted using the shutdown and no shutdown commands.
D. The Operator parameter should be configured under dial peer 1 using the param command.
E. The gateway needs to be reloaded for application changes to take effect.

Answer: A

QUESTION 6:

Which issue found in CAS T1s can be eliminated by converting to PRI?

A. the inability to send and receive ANI on all DS0s
B. long call setup due to few signaling bits per frame
C. voice quality issues involving robbed bits
D. PSTN switch type mismatches
E. outgoing call channel order mismatch (descending vs. ascending)

Answer: A

QUESTION 7:

You are working with a client who is interested in deploying a distributed IP telephony call-processing solution among the three corporate campuses. Each campus will have a Cisco Unified CallManager cluster and a gateway to the PSTN. Which three steps are required for a successful gatekeeper deployment? (Choose three.)

A. determining if each zone will be supported by more than one gatekeeper
B. ensuring the correct location of voice gateways in the network
C. implementing the correct WAN topology
D. determining the intrasite modem and fax traffic patterns
E. provisioning a common codec for all WAN connections
F. determining if gatekeeper redundancy or high availability is required

Answer: C,E,F

QUESTION 8:

How can you configure an MGCP gateway so that it is controlled by the Cisco Unified CallManager?

A. The Cisco Unified CallManager and the MGCP gateway are peers; the Cisco Unified CallManager does not control MGCP gateways.
B. Use the telephony-service ccm-compatible gateway command to enable the Cisco Unified CallManager to control the gateway.

C. Configure the MGCP gateway in the Cisco Unified CallManager; the gateway will automatically be sent a configuration.
D. Configure the gateway on the Cisco Unified CallManager, configure the gateway for MGCP, and define the Cisco Unified CallManager as the call agent on the gateway.
E. Use the mgcp-gateway voip srcadd gateway command to bind the gateway to the Cisco Unified CallManager.

Answer: D

QUESTION 9:

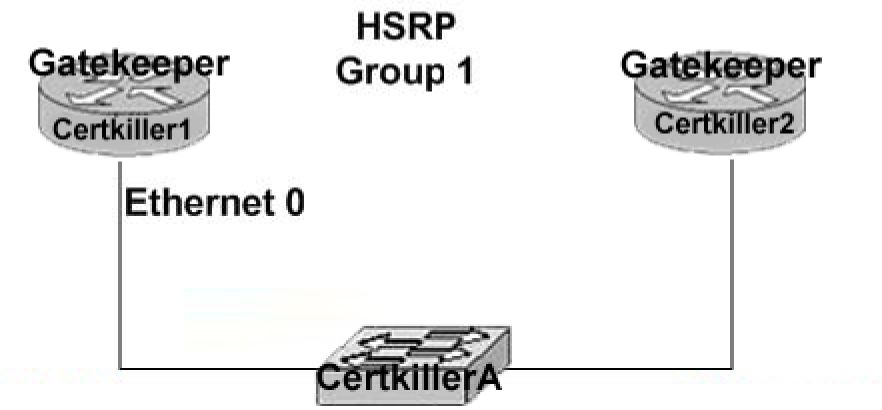
When an IP-to-IP gateway registers with a gatekeeper, what type of endpoint is the gateway listed as?

A. VOIP
B. H323-GW
C. gateway
D. IP-IP

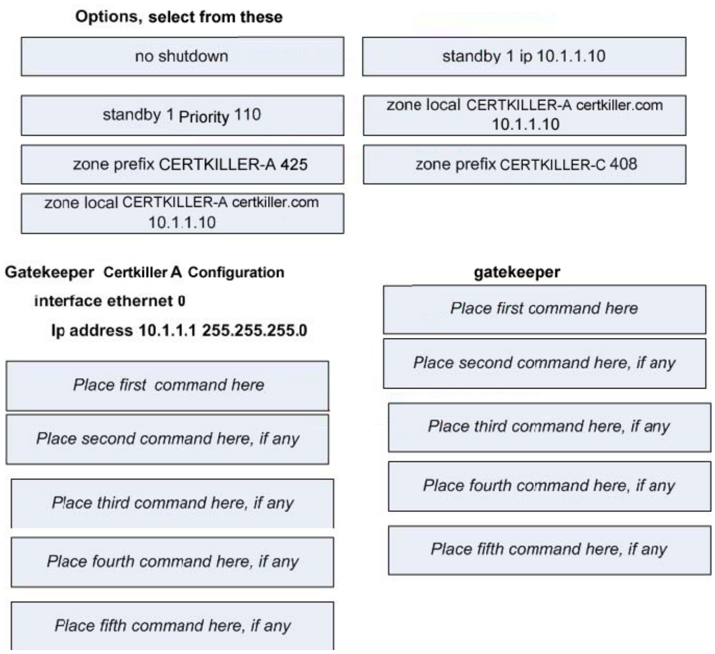
Answer: B

QUESTION 10:

DRAG DROP
Network topology exhibit



You work as a network technician at Certkiller .com. You study the network topology exhibit carefully. You are required to configure the Certkiller 1 Gatekeeper, Your boss, Miss Certkiller, has already entered the IP address command. A command can be entered both as a interface command and as a gatekeeper command. You are not required to use all options.



Answer:

QUESTION 11:

If an endpoint sends the digit string "2125551212" to a gatekeeper that contains the following configuration, what will happen next?
gatekeeper
zone local SJ acme.com 192.168.1.1
zone remote NY acme.com 172.16.1.100
zone prefix SJ 408*
zone prefix NY 212.......
no shut

A. An ARJ will be returned to the endpoint because the gatekeeper is unable to match all the digits.
B. The call will be routed to the gateway registered with the gatekeeper that controls the NY zone.
C. The call will be routed to the PSTN.
D. An ACF will be returned to the endpoint instructing it to set up a call with the endpoint that uses the IP address contained in the ACF.
E. The gatekeeper will send an LRQ to the gatekeeper that controls the NY zone and return a RIP RAS message to the endpoint.
F. The call will be routed to the gateway registered with the gatekeeper for the SJ zone.

Answer: E

QUESTION 12:

Certkiller .com field offices route calls to headquarters through gateways to the PSTN. The headquarters numbers are all in the form 1-202-454-5XXX. At present, calls to headquarters require 12 dialed digits: the access code "9" followed by the 11-digit number. Employees in the field offices would like to call people at headquarters by dialing only their four-digit extensions. Which Cisco IOS command will allow them to do this? (Assume that any associated commands needed in order to apply the one chosen will also be added)

A. Certkiller 3(config-dial-peer)#forward-digits 11
B. Certkiller 3 (cfg-translation-rule)#rule 1 /912024545/ /^5/
C. Certkiller 3 (config-dial-peer)#prefix 912024545
D. Certkiller 3 (config)#num-exp 5... 912024545...

Answer: D

QUESTION 13:

In which three situations would an IP-to-IP gateway be useful? (Choose three.)

A. an organization has deployed a SIP IP telephony solution and needs to connect to a SIP service provider
B. an organization needs to integrate larger PSTN gateways into their SIP network
C. two organizations have merged and both use SCCP signaling
D. an organization would like to migrate from H.323 to SIP
E. an organization needs to migrate from a Cisco Unified CallManager IP telephony solution based on SCCP to an H.323 solution.
F. an organization using H.323 needs to integrate an acquisition that also uses H.323

Answer: A,D,F

QUESTION 14:

Please study the exhibit carefully. Why is the gatekeeper sending a RIP RAS message to the endpoint?

```
4d16h:  RecvUDP_IPSockData  successfully rcvd message of length 134 from 10.100.100.100:49579
4d16h:  ARQ (seq# 8204) rcvdparse_arq_nonstd: ARQ Nonstd decode succeeded, remlen = 1640003356
4d16h:  IPSOCK_RAS_sendto:   msg length 102 from 10.100.100.99:1719 to 10.200.99.99: 1719
4d16h:  RASLib::RASSendLRQ: LRQ (seq# 2058) sent to 10.200.99.99
4d16h:  IPSOCK_RAS_sendto:   msg length 7 from  10.100.100.99:1719 to 10.100.100.100: 49579
4d16h:  RASLib::RASSendRIP: RIP (seq# 8204) sent to 10.100.10010
4d16h:  RecvUDP_IPSockData  successfully rcvd message of length 123 from 10.200.99:1719
4d16h:  LCF (seq# 2058) rcvdparse_lcf_nonstd: LCF Nonstd decode succeeded, remlen = 1640003356
4d16h:  IPSOCK_RAS_sendto:   msg length 69 from 10.100.100.99:1719 to 10.100.100.100: 49579
4d16h: ASLib::RASSendACE:ACF (seq#8204)sent to 10.100.100.100
```

A. because it has exceeded the timeout and needs to restart the call
B. because it cannot respond to a request within the specified timeout
C. because the current call needs to be restarted due to an RRJ
D. because it has to initiate a restart to avoid timing out

Answer: B

QUESTION 15:

When a gatekeeper cluster is deployed, how does a gatekeeper in the cluster indicate to a gateway that it should register with an alternate gatekeeper?

A. It sends an ARJ or RRJ message containing the IP address of the alternate gatekeeper that the gateway should attempt to register with to complete the call.
B. It sends an ARJ or RRJ message accompanied by an ordered list of alternate gatekeepers the gateway should attempt to register with.
C. It sends an ARJ or RRJ, which triggers the automatic re-registration of the gateway with the first gatekeeper in the alternate gatekeeper list.
D. It sends an ARJ or RRJ message containing instructions to use the first member of the alternate gatekeeper list provided by the gatekeeper.

Answer: A

QUESTION 16:

The following Cisco IOS voice translation rule is entered:
voice translation-rule 1
rule 1 /^91\(360\)\(.+\)/ /9\2/
What will be the result if the rule is tested with the number 9-1-360-269-1212 (without the hyphens)?

A. 991
B. The number will not match the rule
C. 91360
D. 9360
E. 92691212
F. 913601212

Answer: E

QUESTION 17:

A customer has two Cisco Unified CallManager clusters, each with an IP-to-IP gateway. When deploying RSVP Call Admission Control between the Cisco Unified CallManager clusters using the IP-to-IP gateways, which two are required? (Choose two.)

A. that the IP-to-IP gateways be configured as RSVP peers
B. media flow-through
C. that an IP-to-IP gateway and a gateway or gatekeeper be configured as RSVP peers
D. media flow-around
E. that the IP-to-IP gateways and the Cisco Unified CallManager clusters be configured with an RSVP agent

Answer: A,B

QUESTION 18:

When a C5510 DSP is configured for conferencing, what other services can it be configured to support?

A. transcoding
B. no other services
C. voice port termination
D. supplementary services

Answer: B

QUESTION 19:

Users are complaining that they are not seeing caller names on calls received from the PSTN. Which debug command can be used to troubleshoot this problem?

A. debug vpm signal
B. debug ccs signal
C. debug isdn q921
D. debug isdn setup
E. debug isdn q931

Answer: E

QUESTION 20:

The Acme Corporation network contains gateways, gatekeepers, and a directory gatekeeper. If a gatekeeper containing the following configuration receives the digit string "2125551212" from an endpoint, what will the gatekeeper do next?

```
gatekeeper
zone local SJ acme.com 192.168.1.1
zone remote DGK acme.com 172.16.1.100
zone prefix SJ 408*
zone prefix DGK *
no shut
```

A. send a call setup message to the gateway registered with the gatekeeper that controls the NY zone
B. send a call setup message to the gateway registered with DGK
C. send an LRQ message to DGK to resolve the call routing
D. send an ACF back to the endpoint instructing it to set up a call with the gateway registered with the SJ gatekeeper

E. send an ARJ message to the endpoint because the gatekeeper is unable to match all the digits

Answer: C

QUESTION 21:

Certkiller .com wants to route inbound faxes directly to the recipient's e-mail. Which gateway fax protocol will support this?

A. T.37
B. Fax Pass-Through
C. T.38
D. Fax Relay

Answer: A

QUESTION 22:

Please study the exhibit carefully. Which dial peer will send calls to the PSTN via the T1 configured as shown?

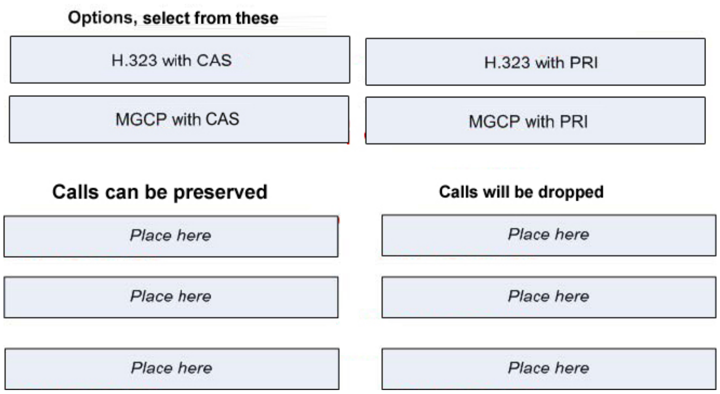
```
controller t1 3/0
 framing esf
 linecode b8zs
 ds0-group 0 timeslots 1-24 type e&m-wink-start
```

A. dial-peer voice 1 pots
destination-pattern 9T
port 3/0:0
B. dial-peer voice 1 pots
destination-pattern 9T
session target 3/0:1
C. dial-peer voice 1 pots
destination-pattern 9T
session target 3/0:0
D. dial-peer voice 1 pots
destination-pattern 9T
port 3/0:1
E. dial-peer voice 1 pots
destination-pattern 9T
port 3/0:24
F. dial-peer voice 1 pots
destination-pattern 9T
session target 3/0:24

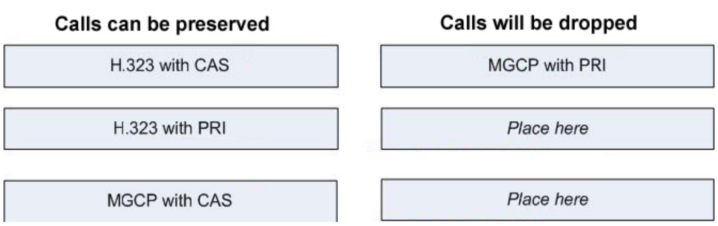
Answer: A

QUESTION 23:

DRAG DROP
You work as a network administrator at Certkiller .com.
Your Certkiller trainee Certkiller is curious about gateway signaling protocols and PSTN trunk types. Assuming all gateways are SRST gateways what will happen to active calls to or from the PSTN when the gateway fails over the SRST mode (either preserved or dropped)?



Answer:



QUESTION 24:

A single call is active through an IP-to-IP gateway. If you use the show gatekeeper calls command, how many active call legs will be indicated?

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

Answer: C

QUESTION 25:

Which statement accurately describes the amount of bandwidth required for a fax call using fax pass-through when packet redundancy is not enabled?

A. The fax call uses the same amount of bandwidth as a G.711 voice call.
B. The fax call uses the same amount of bandwidth as a G.729 voice call.
C. The fax call uses 20% more bandwidth than a G.711 voice call because the fax sample

size is smaller.
D. The fax call uses 20% more bandwidth than a G.729 voice call because the fax sample size is smaller.
E. The fax call uses 20% more bandwidth than a G.729 voice call because the fax sample size is larger.
F. The fax call uses 20% more bandwidth than a G.711 voice call because the fax sample size is larger.

Answer: C

QUESTION 26:

You need to configure a gatekeeper to limit the number of voice calls between any local zone and any other zone to five simultaneous G.729 calls. Which bandwidth command would you enter into the gatekeeper configuration?

A. bandwidth total interzone default 80
B. bandwidth total default 80
C. bandwidth interzone zone SJ 80
D. bandwidth interzone default 80

Answer: D

QUESTION 27:

A dial peer is configured with an outgoing COR list. Which two conditions must be met for calls matching this outbound dial peer to be blocked? (Choose two.)

A. The outgoing COR list must be a superset of the incoming COR list.
B. The incoming COR list and the outgoing COR list must contain the same members.
C. The outgoing COR list must be a subset of the incoming COR list.
D. An incoming COR list must be configured on the matched outbound dial peer.
E. An incoming COR list must be configured on the matched inbound dial peer.

Answer: A,E

QUESTION 28:

Which gatekeeper configuration will send all calls between zone hardware and zone appliances through the IP-to-IP gateway in the VIA zone?

A. gatekeeper
zone local hardware acme.com 192.168.1.1
zone local appliances acme.com
zone local via acme.com invia via outvia via
zone prefix hardware 408*

zone prefix appliances 415*
no shut
B. gatekeeper
zone local hardware acme.com 192.168.1.1 invia VIA outvia VIA
zone local appliances acme.com invia VIA outvia VIA
zone local VIA acme.com
zone prefix hardware 408*
zone prefix appliances 415*
no shut
C. gatekeeper
zone local hardware acme.com 192.168.1.1
zone local appliances acme.com
zone local via acme.com
zone prefix hardware 408.......
zone prefix appliances 415.......
no shut
D. gatekeeper
zone local hardware acme.com 192.168.1.1
invia VIA outvia VIA
zone local appliances acme.com invia VIA outvia VIA
zone prefix hardware 408.......
zone prefix appliances 415.......
no shut

Answer: B

QUESTION 29:

DRAG DROP
You work as a network administrator at Certkiller .com. You need to explain how a H.323 Gateway is operating for your boss Miss Certkiller. You tell Jack that when a call is placed by an H.323 gateway registered with a gatekeeper, a sequence of RAS messages is exchanged.
You are required to place the options in the correct order for an interzone call.

Steps, select from these	Steps,place here
The Zone A gatekeeper looks in its zone prefix table to determine if it has a match	Place first step here
The Zone A gatekeeper returns an ACF to the originating gateway	Place second step, if any, here
The Zone A gatekeeper sends an LRQ to the Zone B gatekwwer	Place third step, if any, here
The zone B gatekeeper consults the prefix table and returns an LOF to the ZoneA gatekeeper	Place fourth step, if any, here
The gateway sends an ARQ to the Zone A gatekeeper	Place 5th siep, if any, here
The originating gateway sends a call setup message to the terminating gateway	Place 6th siep, if any, here

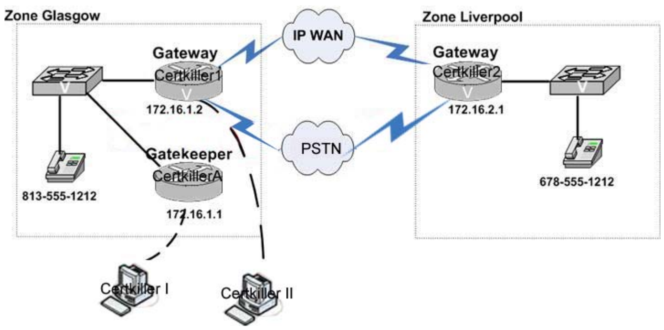
Answer:

Steps, place here
The originating gateway sends a call setup message to the terminating gateway
The gateway sends an ARQ to the Zone A gatekeeper
The Zone A gatekeeper returns an ACF to the originating gateway
The Zone A gatekeeper looks in its zone prefix table to determine if it has a match
The Zone A gatekeeper sends an LRQ to the Zone B gatekwwer
The zone B gatekeeper consults the prefix table and returns an LCF to the Zone A gatekeeper

---

QUESTION 30:

SIMULATION
Network topology exhibit:



You work as a network administrator at Certkiller .com. You are giving a lecture for a class of Certkiller trainees. The lecture is on how a gateway registers with a gatekepper. You have set up a lab nVoIP etwork which is displayed in the network topology exhibit. During the demonstration you will be required to configure a Gateway router named Certkiller 1 and a Gatekeeper Router named Certkiller A.

Further information:

* Certkiller 1 has a POTS connection to the PSTN
* Certkiller 1's Ethernet connection is shared with Certkiller A.

Answer:

QUESTION 31:

Please study the exhibit carefully. Your customer is using BRIs to the PSTN. Inbound calls can be made without error. However, outbound calls only succeed if there has been a recent inbound call. What can be done to correct this issue?

certkiller3#show dial-peer voice summary

dial-peer hunt 0

TAG	TYPE	AD MIN	OPER	PREFIX	DEST-PATTERN	PRE PER	PASS THRU	SESS-TARGET	OUT STAT	PORT
1	pots	up	up		0.+	0			down	1/0/0
2	pots	up	up		0.+	0			down	1/0/1
3	pots	up	up		0.+	0			up	1/1/0
4	pots	up	up		0.+	0			down	1/1/1

A. Configure preferences to always select dial peer 3.
B. Configure SPIDs on the BRI so that outbound calls are sent to the correct number.
C. Disable status checking for POTS dial peers so outbound call setup will activate the BRI.
D. Change the dial peer hunt logic so the appropriate dial peer is always used.

Answer: C

QUESTION 32:

Highland Park Property Development is integrating a Cisco Unified 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. Change the channel selection order from descending to ascending.
B. Add the command isdn negotiate-bchan to the serial interface.
C. Add the command isdn contiguous-bchan to the serial interface.
D. Increase the ISDN T302 timer to allow more time for call setup.

Answer: C

QUESTION 33:

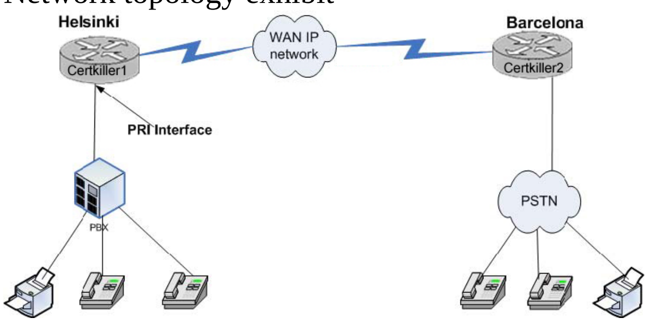
Which describes a proper CAC implementation in an H.323 network that uses directory gatekeepers?

A. The directory gatekeeper controls call bandwidth usage between the zones.
B. The zone gatekeepers maintain the bandwidth commands and the directory gatekeeper is only responsible for dial plan resolution.
C. The zone gatekeeper negotiates bandwidth requirements with the target zone gatekeeper by means of an LRQ message forwarded by the directory gatekeeper.
D. The bandwidth commands for each zone that is registered with the directory gatekeeper are included in the directory gatekeeper configuration.

Answer: B

QUESTION 34:

Network topology exhibit



Certkiller 1 partial show running config output exhibit.

```
interface serial0/0:23
 no ip address
 isdn switch-type primary-qsig
 isdn incoming-voice voice
 no cdp enable
!
dial-peer voice 1 voip
 destination-pattern 901149T
 translation-profile outgoing Barcelona
 session target ipv4:10.0.0.1
!
Dial-peer voice 1 pots
 incoming called-number .
 direct-inward-dial
!
```

Configuration exhibit:

Certkiller A

```
interface serial0/0:23
 no ip address
 isdn switch-type primary-qsig
 isdn overlap-receiving T302 2000
 isdn incoming-voice voice
 isdn send-alerting
 no cdp enable
!
```

Certkiller B

```
interface serial0/0:23
 no ip address
 isdn switch-type primary-qsig
 isdn protocol-emulate network
 isdn incoming-voice voice
 isdn send-alerting
 no cdp enable
!
```

Certkiller C

```
interface serial0/0:23
 no ip address
 isdn switch-type primary-qsig
 isdn t-activate 2000
 isdn incoming-voice voice
 isdn send-alerting
 no cdp enable
!
```

Certkiller D

```
interface serial0/0:23
 no ip address
 isdn switch-type primary-qsig
 isdn integrate calltype all
 isdn incoming-voice voice
 isdn send-alerting
 no cdp enable
!
```

You work as a network technician at Certkiller .com. Please study the exhibit carefully. Users calling to Spain from phones connected to the PBX in Helsinki dial the full telephone number. Calls should be re-routed over the WAN to Barcelona instead of using the PSTN connection in Helsinki. However, the calls are being dropped. Troubleshooting indicates that dialed digits are being sent from the PBX in multiple ISDN information messages. The exhibit contains the output from the show running-config command on Router.
Which interface configuration will most likely solve the problem?

A. Configuration Certkiller A
B. Configuration Certkiller B
C. Configuration Certkiller C
D. Configuration Certkiller D

Answer: A

QUESTION 35:

A service provider wants to add SIP devices to an 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. UA server
C. H.323 proxy server
D. SIP proxy server

Answer: D

QUESTION 36:

Exhibit: *** Missing***
Please study the exhibit carefully. The following issues are encountered when calls are placed from Phone A to Phone B:
- If the caller on Phone B does not answer, Phone B continues to ring even if the caller on

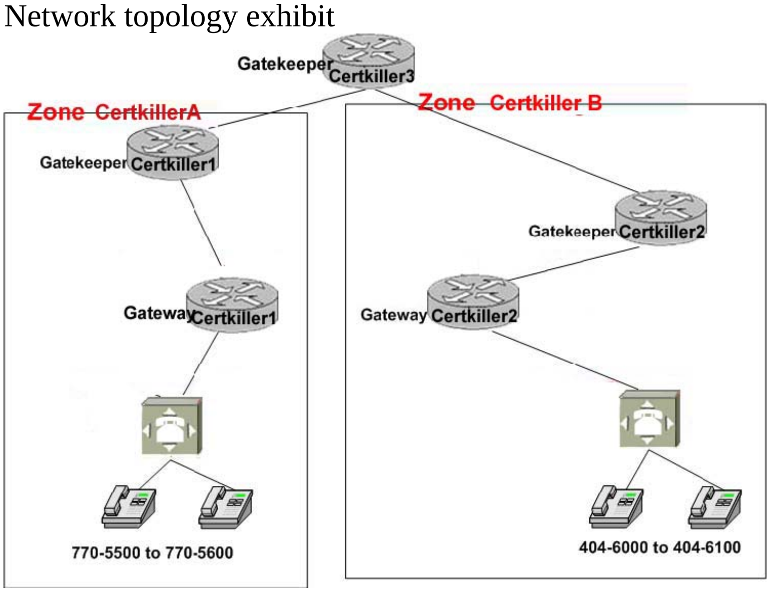
Phone A hangs up.
- If the call is answered, it stays active until the caller on Phone B hangs up, regardless of the actions of the caller on Phone A.
How can the underlying problem be resolved?

A. Shorten the amount of time that the PBX provides power denial so it will be recognized by the FXO port.
B. Turn off tone-based supervisory disconnect on the FXO port.
C. Configure battery reversal on the FXO port.
D. Use ground-start signaling on the connection between the PBX and the router's FXO port.

Answer: D

QUESTION 37:

Network topology exhibit



Gatekeeper Certkiller 3# show running-config exhibit

```
Gatekeeper Certkiller3#   show running-config
....

zone local Gatekeeper Certkiller3 certkiller.com
zone local GatekeeperCertkiller1 certkiller.com 172.16.14.44 1719
zone remote GatekeeperCertkiller2 certkiller.com 172.16.14.99 1719
zone prefix Gatekeeper Certkiller1 770.........
zone prefix Gatekeeper Certkiller1 404.........
no shutdown
!
....
```

Please study the exhibits carefully. You have a client that is testing a directory gatekeeper in the lab to provide address resolution between two different zones. Two of the commands in the running-config output are incorrect. Which two changes will correct the configuration? (Choose two.)

A. replace zone remote Gatekeeper Certkiller 2 Certkiller .com 172.16.14.99 1719 with zone local Gatekeeper Certkiller 2 Certkiller .com 172.16.14.99 1719
B. replace zone prefix Gatekeeper Certkiller 2 404 . . . . with zone prefix Gatekeeper Certkiller 2 404 . . . .
C. replace zone Gatekeeper Certkiller 1 Certkiller .com 172.16.14.99 1719 with zone remote Gatekeeper Certkiller 1 Certkiller .com 172.16.14.44 1719
D. replace zone local Gatekeeper Certkiller 3 Certkiller .com with zone remote Gatekeeper Certkiller 3 Certkiller .com
E. replace zone prefix Gatekeeper Certkiller 1 770 . . . . with zone prefix Gatekeeper Certkiller 1 770 . . . .

Answer: C,E

QUESTION 38:

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

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

Answer: C,D

QUESTION 39:

You are adding a gatekeeper to an H.323 network to provide Call Admission Control. You need to be able to support three concurrent G.711 calls from a specified zone to any other zone. How much interzone bandwidth should be configured in the gatekeeper?

A. 384 Kbps
B. 192 Kbps
C. 480 Kbps
D. 240 Kbps

Answer: A

QUESTION 40:

Which debug command can be used to troubleshoot an issue with a Tcl script?

A. debug custom script
B. debug call script
C. debug tcl script
D. debug voice application

Answer: D

QUESTION 41:

Please study the exhibit carefully. Certkiller .com wants to prohibit all callers whose numbers start with 425 from making international calls. Outgoing calls from numbers starting with 425 should match dial peer 1 inbound to the gateway. However, dial peer 1 is configured incorrectly and its operational status is "down."What can be done to change the dial-peer operational status to "up" and provide the desired functionality?

The following is a dial configuration on their voice gateway

```
dial-peer voice 1 voip
  answer-address 425....
  destination-pattern 425....
  corlist incoming LD

dial-peer voice 10 pots
  corlist outgoing LD
  destination-pattern 91[2-9]..[2-9]......
  port 1/0:23
```

The following is output from the show dial peer voice summary command:

show dial peer voice summary

```
              AD                                    PRE  PASS
TAG    TYPE   MIN   OPER   PREFIX   DEST-PATTERN    FER  THRU  SESS-TARGET  PORT
1      voip   up    down                             0
10     pots   up    up              91[2-9]..[2-9]..- 0  syst  ras          1/0:23
```

A. remove the destination-pattern 425.... command
B. remove the wildcard characters from the destination pattern and specify a complete number
C. add a session target ras command
D. change the answer-address 425.... command to incoming called-number 425.... .

Answer: A

QUESTION 42:

Which statement is true regarding Cisco modem relay?

A. It requires that codec complexity be set to "high" or "flex mode".
B. It is supported on H.323, MGCP, SIP, and SCCP gateways.
C. It supports the V.150.1 signaling standard.
D. It uses more bandwidth than modem pass-through.
E. It is supported by some third party vendors.

Answer: A

QUESTION 43:

Which gatekeeper will a gateway register with when using the following command syntax?
!
interface FastEthernet 0/1 ip address 192.168.2.9 255.255.255.0
h323-gateway voip interface
h323-gateway voip id GK-East1 multicast priority 1
h323-gateway voip h323-id BOS
!

A. The gateway will attempt to register with the GK-East1 zone, since it has the highest priority.
B. The gateway will attempt to register with the first gatekeeper that has the same priority number in the RCF message.
C. The gateway will attempt to register with the first gatekeeper in the East zone that replies, since the priority is set to 1.
D. The gateway will attempt to register with the BOS zone.
E. The gateway will attempt to register with the first gatekeeper in the East zone that replies, and then will attempt to register with GK-East1.

Answer: A

QUESTION 44:

Which three are features provided by an IP-to-IP gateway? (Choose three.)

A. security
B. address hiding using flow-around signaling
C. Call Admission Control
D. IP to PSTN gateway
E. video integration
F. protocol interworking for SCCP, SIP and H.323

Answer: A,C,E

QUESTION 45:

How can you configure an H.323 gateway so that it is controlled by the Cisco Unified CallManager?

A. Use the telephony-service ccm-compatible gateway command to enable the Cisco Unified CallManager to control the gateway.

B. Use the H.323-gateway voip srcadd gateway command to bind the gateway to the Cisco Unified CallManager.
C. Configure the gateway on the Cisco Unified CallManager, and it will automatically configure the gateway.
D. The Cisco Unified CallManager and the H.323 gateway are peers. The Cisco Unified CallManager does not control H.323 gateways.

Answer: D

QUESTION 46:

You have a client who is designing a gateway solution for an IP communications network. The T1 needs to support ANI for both incoming and outgoing calls.How should the gateway be configured?

A. Configure an MGCP gateway so that there are two DS-0 groups on the T1 from the PSTN, one to send ANI and one to receive ANI.
B. Configure an MGCP gateway so that there is a single DS-0 group on the T1 to the PSTN, to both send and receive ANI.
C. Configure an H.323 gateway so that there is a single DS-0 group on the T1 to the PSTN, to both send and receive ANI.
D. 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.

Answer: D

QUESTION 47:

An ISR 2811 has two PVDM-32s installed on its mainboard. What is the maximum number of simultaneous voice calls the router can support if voice card 0 is configured for high codec complexity?

A. 48
B. 64
C. 128
D. 24
E. 16
F. 32

Answer: D

QUESTION 48:

If a via-zone is created for a gatekeeper, enabling it to support an IP-IP gateway, are the functions of the gatekeeper dedicated to the via-zone?

A. No, the gatekeeper can support the via-zone and still function as a directory gatekeeper
B. No, the gatekeeper can either be dedicated to the via-zone or support both the via-zone and other zones simultaneously
C. Yes, but the via-zone gatekeeper can still communicate with other gatekeepers
D. Yes, once the gatekeeper is configured for the via-zone it is dedicated to the via-zone

Answer: B

QUESTION 49:

You have a client who is planning to implement a toll bypass service for the PBXs. The organization has four locations and the PBXs are a mixture of QSIG- and non-QSIG-capable devices. Which type of services will be available between a QSIG and a non-QSIG PBX?

A. The gateway will translate all the supplementary services for both PBXs.
B. Only basic calls that do not require supplementary services are supported.
C. The supplementary services on the non-QSIG PBX will be available to both devices.
D. The supplementary services supported by the QSIG PBX will be terminated by the gateway and translated to the non-QSIG PBX.

Answer: B

QUESTION 50:

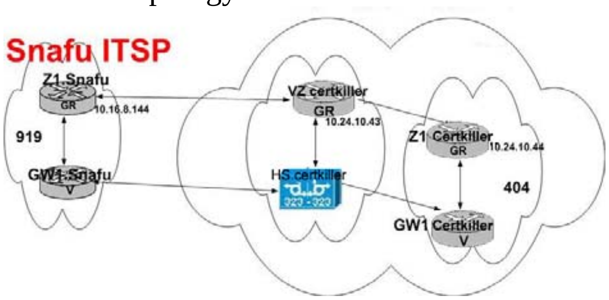
A Cisco voice gateway is connected to the PSTN via an analog line. How can the gateway be configured to support caller ID?

A. using MGCP and an FXO port to the PSTN
B. using MGCP and an FXS port to the PSTN
C. using H.323 and an FXS port to the PSTN
D. using H.323 and an FXO port to the PSTN

Answer: D

QUESTION 51:

Network topology exhibit



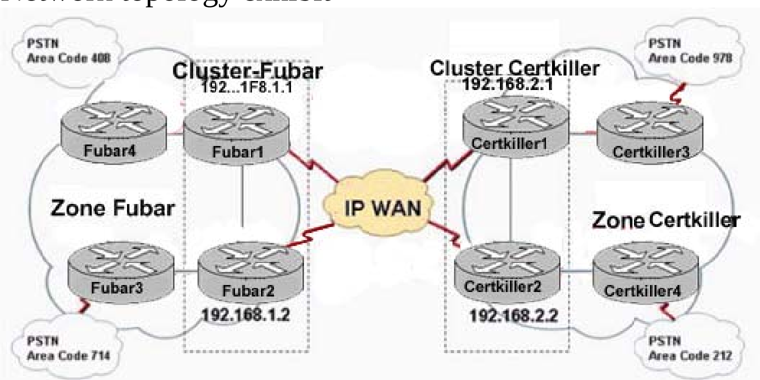
Please study the exhibit carefully. Certkiller ITSP would like to be able to test an IP-to-IP gateway. The company has set up two ITSPs in the lab to simulate a real-world situation. Which set of commands will configure the VZ. Certkiller gatekeeper correctly?

A. gatekeeper
!
zone local VZ Certkiller 10.24.10.43
zone remote Z1 Certkiller Certkiller 10.24.10.44 invia VZ Certkiller
zone remote Z1SNAFU Snafu 10.16.8.144 1719 outvia VZ Certkiller
zone prefix Z1SNAFU 919*
B. gatekeeper
Eeeee
/////
zone local VZ Certkiller Certkiller 10.24.10.43
zone remote VZ Certkiller Certkiller 10.24.10.44 in Z1 Certkiller
zone remote Z1SNAFU Snafu 10.16.8.144 1719 out Z1 Certkiller
zone prefix Z1SNAFU 919*
C. gatekeeper
!
zone local Z1 Certkiller 10.24.10.44
zone remote VZ Certkiller Certkiller 10.24.10.43 in Z1 Certkiller
zone remote Z1SNAFU Snafu 10.16.8.144 1719 out Z1 Certkiller
zone prefix Z1SNAFU 919*
D. gatekeeper
!
zone local Z1SNAFU Snafu
zone remote VZ Certkiller Certkiller 10.24.10.43 invia Z1SNAFU
zone remote Z1SNAFU Snafu 10.16.8.144 1719 outvia Z1SNAFU
zone prefix Z1SNAFU 919*

Answer: A

QUESTION 52:

Network topology exhibit



Please study the exhibit carefully. Certkiller .com has two zones: Certkiller and Fubar.

Each zone has two gatekeepers: Certkiller 1 and Certkiller 2, and Fubar1 and Fubar2. Acme would like to configure a gatekeeper cluster for each zone. Which two sets of configuration commands would need to be added to the Fubar1 and Fubar2 gatekeeper configurations to create the Fubar cluster? (Choose two.)

A. zone local Fubar Certkiller .com 192.168.1.1
zone cluster remote Cluster-Fubar
element Fubar1 192.168.1.1 1719
B. zone local LA Certkiller .com 192.168.1.2
zone cluster remote Cluster-Fubar
element Fubar1 192.168.1.1 1719
C. zone local Fubar Certkiller .com 192.168.1.2
zone cluster local Cluster-Fubar
element Fubar1 192.168.1.1 1719
D. zone local Fubar4 Certkiller .com 192.168.1.1
element Fubar2 192.168.1.2 1719
E. zone local Fubar Certkiller .com 192.168.1.2
element Fubar2 192.168.1.2 1719
F. zone local Fubar Certkiller .com 192.168.1.1
zone cluster local Cluster-Fubar
element Fubar2 192.168.1.2 1719

Answer: C,F

QUESTION 53:

Which IOS mechanism is used to restrict calling based on the device initiating the call?

A. NBAR
B. COR
C. RAI
D. RSVP

Answer: B

QUESTION 54:

A gateway is going to provide media resources useable by three different Cisco Unified CallManager clusters. Which Cisco IOS structure makes this possible?

A. applications
B. port numbers
C. service types
D. profiles

Answer: D

QUESTION 55:

Please study the exhibit carefully. On a router running Cisco IOS version 12.3(14)T, an auto-attendant Tcl script is loaded and a warning message is displayed, stating that the operator parameter has not been registered. See the exhibit for an example of the error. What must you do to continue?

```
Certkiller3 (config-app)#service aa flash:its-CISCO.2.0.1.0.tcl
Certkiller3 (config-app-param)#
    *Sep 20 17:24:11.545: //-1//HIFS:/hifs_its_cb: hifs ifs file read succeeded. siz
    e=6627, url=flash:its-CISCO.2.0.1.0.tcl
    *Sep 20 17:24:11.549: //-1//HIFS:/hifs_free_idata: hifs_free_idata: 0x48CA4354
    *Sep 20 17:24:11.549: //-1//HIFS:/hifs_hold_idata: hifs_hold_idata: 0x48CA4354
Certkiller3 (config-app-param)#param operator 5000
    Warning: parameter operator has not been registered under aa namespace
Certkiller3 (config-app-param)#
```

A. nothing, the warning may be ignored
B. register the application parameters with the Cisco Unified CallManager
C. register the application parameters with the gatekeeper
D. register the application parameters with the application
E. register the application with the Cisco Unified CallManager
F. register the application with the gatekeeper

Answer: A

QUESTION 56:

When a WAN link problem occurs, it takes over three minutes for IP phones to become registered with the SRST gateway. What is the most likely cause of this?

A. The keepalive timer in the SRST gateway is set too long.
B. Each phone has a list of two alternate Cisco Unified CallManager systems, and it tries to register with each before registering with the SRST gateway.
C. The SRST gateway is an MGCP gateway, and it must stop the MGCP process and switch over to the default H.323 process to initiate the SRST process.
D. The WAN link is bouncing.

Answer: B

QUESTION 57:

Configuration exhibit:

```
Certkiller2 #show ccm-manager
MGCP Domain Name: BR1
Priority        Status                   Host
============================================================
Primary         Down                     10.:.5.10
First Backup    None
Secret Backup   None

Current active call manager: None
Backhaul/Redundant link port:   2428
Failover Interval:              30 seconds
Keepalive Interval:             15 seconds
Last keepalive sent:            15:53:22 UTC Aug 4 2006 (elapsed time: 1h11m)
Last MGCP traffic time:         15:52:52 UTC Aug 4 2006 (elapsed time: 1h12m)
Last failover time:             None
Last switchback time:           None
Switchback mode:                Graceful
MGCP Fallback mode:             Not Selected
Last MGCP Fallback start time:  None
Last MGCP Fallback end time:    None
MGCP Download Tones:            Disabled

Backhaul/Redundant link is down
Configuration Error History
FAX mode: cisco
 Certkiller2

Certkiller2 #show dial-peer voice summary
dial-peer hunt 0
          AD                               PRE PASS                OUT
TAG   TYPE MIN  OPER PREFIX  DEST-PATTERN  FER THRU SESS-TARGET    STAT   PORT
1     pcts up   up           9T             0                      down   1/0:23
```

Please study the exhibit carefully. Certkiller .com has a centralized Cisco Unified CallManager deployment, with SRST implemented on MGCP gateways at the remotes sites. The WAN link to Seattle has failed. All phones in Seattle have registered to the SRST gateway and can call each other internally, but all outside calls fail. Based on the output shown in the exhibit, what is the most likely solution to the problem?

A. add the isdn outgoing-voice command to the configuration
B. call the phone provider and report that the PRI link is down
C. configure MCGP fallback on the gateway
D. enable the outgoing POTS dial peer with the no shutdown command

Answer: C

QUESTION 58:

ACME Widgets has several geographically dispersed call centers. Calls come into the central location on H.323 gateways, and are forwarded across WAN links to the correct call centers using G.729 codecs. IVR devices in the call centers are not correctly interpreting digits entered by callers. What can be done to solve this problem?

A. use a transcoder to convert the calls to G.711 at the call center
B. use a DTMF relay to send the digits out of band
C. adjust the DTMF inter-digit timer on the gateway
D. hairpin calls back through the PSTN to the appropriate call center

Answer: B

QUESTION 59:

Which two dial plan components are required to implement TEHO? (Choose two.)

A. Class of Service
B. internal numbering plan

C. digit manipulation
D. path selection
E. call coverage

Answer: C,D

QUESTION 60:

Please study the exhibit carefully. Which message ID can be used to track this call from the requesting endpoint?

```
4d16h: Recv UDP_IP SockData   successfully rcvd message of length 134 from 10.100.100.100:49579
4d16h: ARQ (seq# 8204) rcvaparse_arq_nonstd: ARQ Nonstd decode succeeded, remlen = 1640003356
4d16h: IPSOCK_RAS_sendto:   msg length 102 from 10.100.100.99:1719 to 10.200.99.99: 1719
4d16h: RASLib::RASSendLRQ: LRQ (seq# 2058) sent to 10.200.99.99
4d16h: IPSOCK_RAS_sendto:   msg length 7 from 10.100.100.99:1719 to 10.100.100.100: 49579
4d16h: RASLib::RASSendRIP: RIP (seq# 8204) sent to 10.100.100.100
4d16h: RecvUDP_IPSockData  successfully rcvd message of length 123 from 10.200.99.99:1719
4d16h: LCF (seq# 2058) rcvdparse_lcf_nonstd: LCF Nonstd decode succeeded, remlen = 1640003356
4d16h: IPSock_RAS_sendto:   msg length 69 from 10.100.100.99:1719 to 10.100.100.100: 49579
4d16h: RASLib::RASSendACF ACF (seq# 8204) sent to 10.100.100.100
```

A. 49579
B. 8204
C. 10.100.100.99
D. 1640003356
E. 10.200.99.99
F. 10.100.100.100

Answer: B

QUESTION 61:

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

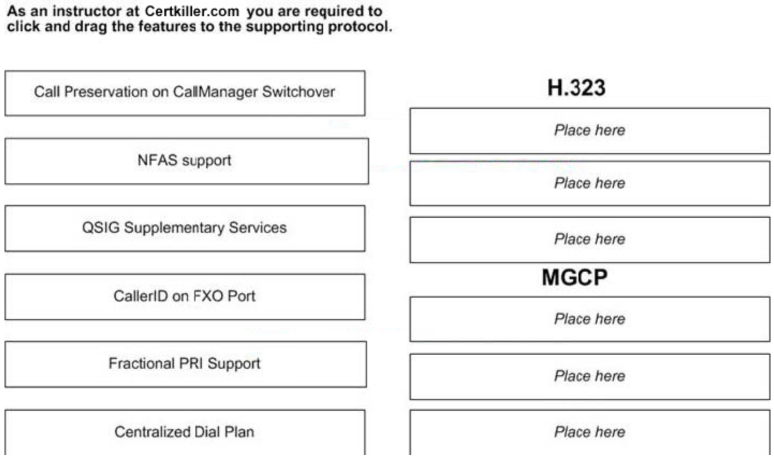
A. called ID
B. transcoding
C. E911
D. directory lookup
E. conference bridging
F. music on hold

Answer: B, E

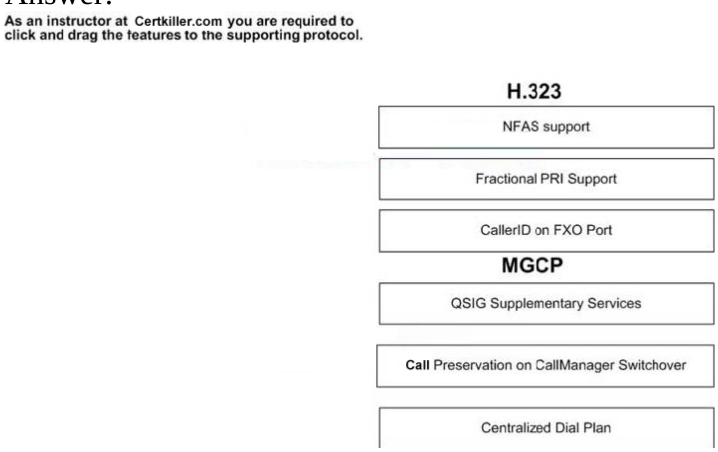
Explanation: The DSP farm uses the DSP resources in network modules on Cisco routers to provide voice-conferencing, transcoding, and hardware MTP services.
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 4-56 Implementing Advanced Gateway Features

QUESTION 62:

DRAG DROP



Answer:



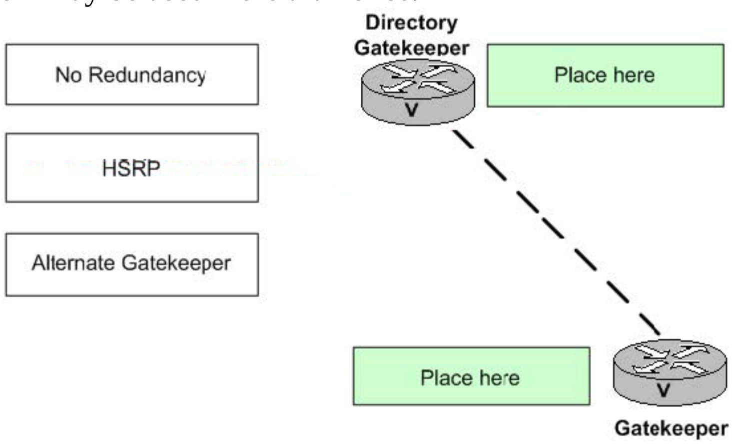
Explanation:
Why Choose H.323
* Integrated access
* Caller ID support on analog FXO
* Many more TDM interface types and signaling
* Dropping DSPs on hairpinned calls
* Gateway-resident applications like TCL and VXML
* CAC network design with H.323 gatekeepers
* No release dependencies between gateways 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.

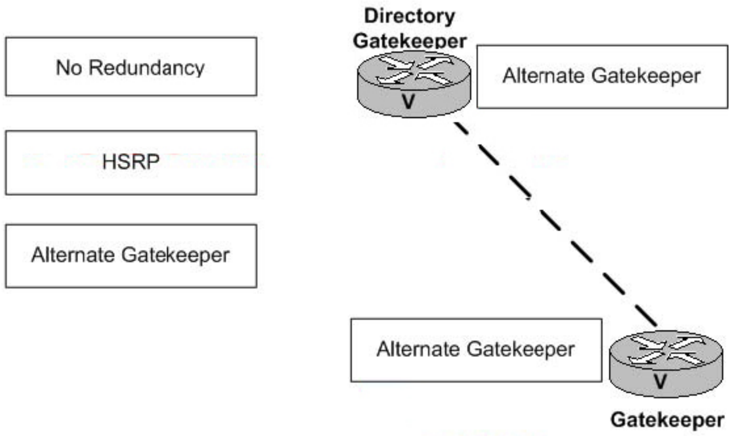
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 1-67 Function of Gateways and Gatekeepers

QUESTION 63:

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 64:

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

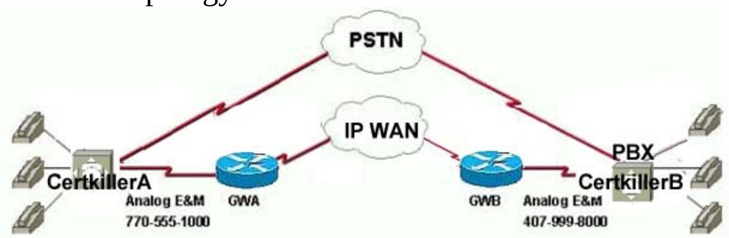
Answer: C

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 65:

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 66:

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 =
0x08020089050403809OA31803A98381180 90A31803A983^2008905040313590504 03
1d20h:      SETUP pd = 8  callref = 0x0089
1d20h:          Bearer Capability i = 0x008913
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 67:

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 68:

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.
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 5-7, 88 Deploying Gatekeepers

QUESTION 69:

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 70:

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.

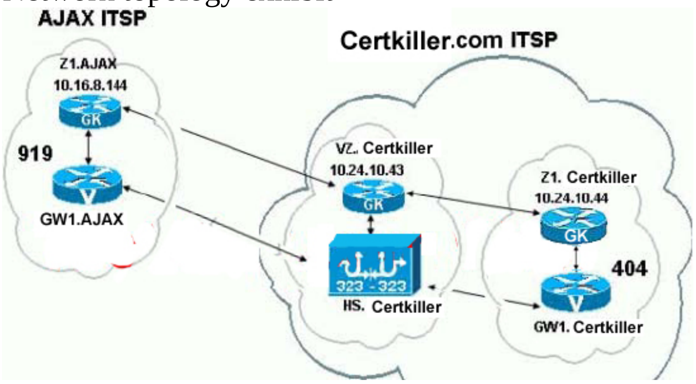
A. sccp ip-address 10.172.15.4 priority 1 version 4.0
B. sccp ccm 10.172.15.4 priority 1 version 4.0
C. sccp register 10.172.15.4 priority 1 version 4.0
D. sccp server 10.172.15.4 priority 1 version 4.0

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 71:

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

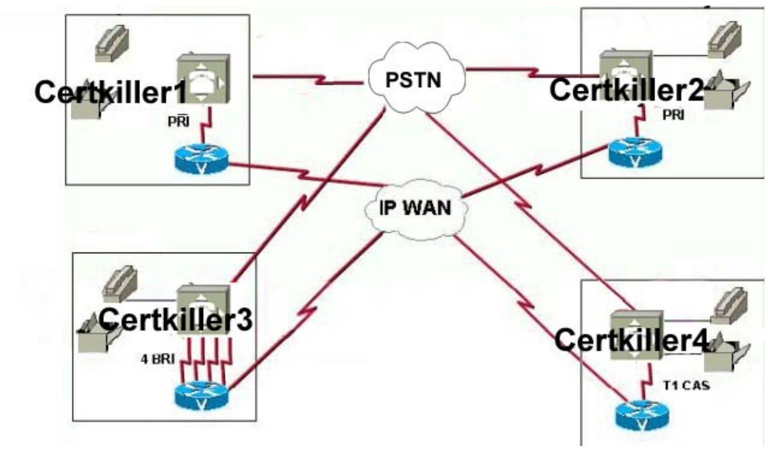
Explanation: Cisco Multiservice IP-to-IP Gateway and Gatekeeper Design

- 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 72:

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
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 2-121 to 130 Integrating a VoIP Network to the PSTN and PBXs

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QUESTION 73:

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
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 5-108 Deploying Gatekeepers

QUESTION 74:

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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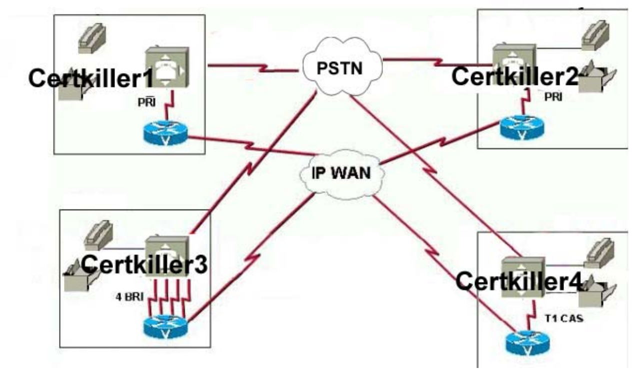
QUESTION 75:

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 be found at
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 76:

Exhibit, Network Topology

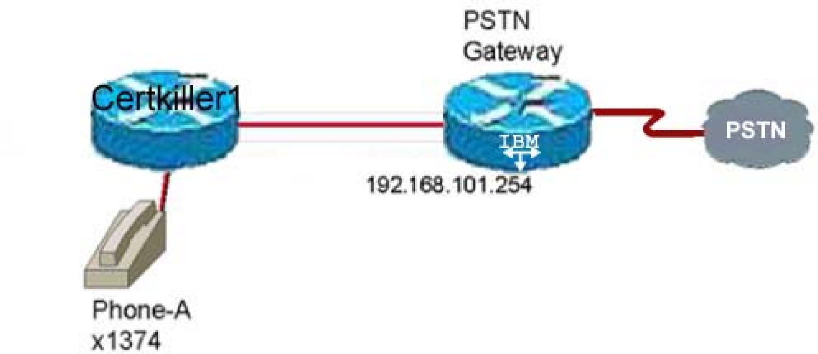

Exhibit #2

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

dial-peer cor list Intl01
  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 LDLst
  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 77:

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
pg 2-80 to 90 Integrating a VoIP Network to the PSTN and PBXs

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QUESTION 78:

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 79:

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 80:

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 81:

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 82:

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

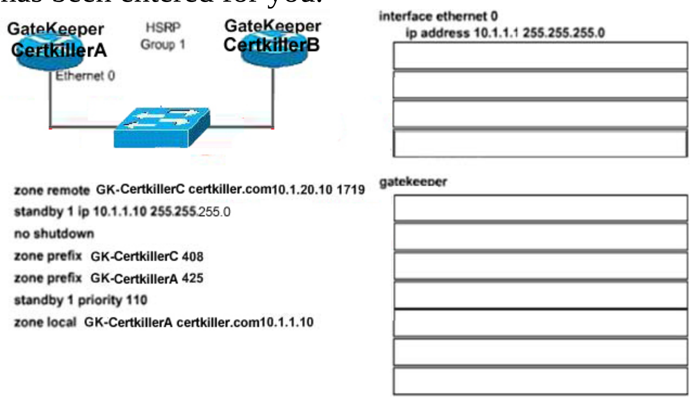
Answer: C

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.

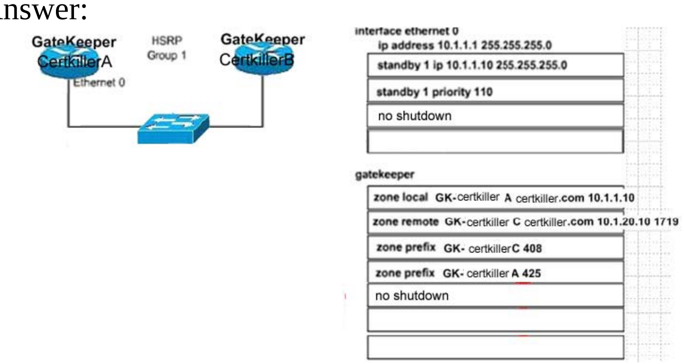
The server can override default gatekeeper behavior.
GKTMP is an independent platform and can run on Solaris, Linux, Microsoft Windows NT. It allows third parties to develop sophisticated applications to control RAS communication. An example of the use of GKTMP is where a service provider wants to control 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.
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 5-4 Deploying Gatekeepers

QUESTION 83:

DRAG DROP
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 84:

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 85:

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.
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 2-172 Integrating a VoIP Network to the PSTN and PBXs

QUESTION 86:

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_incomming
}
} else {
puts "Call [infotag get con_all] got
event $status while placing an outgoing
call"
call close
}
}
```

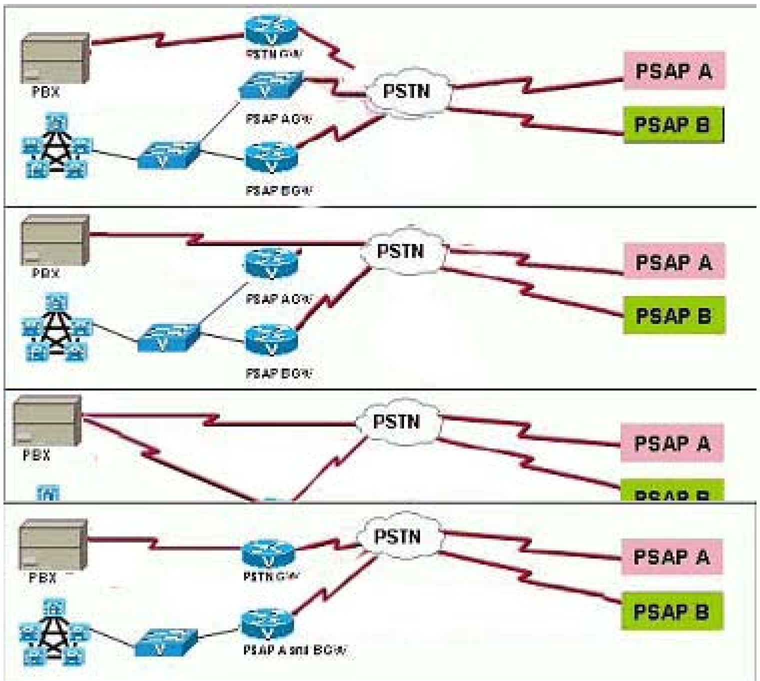
Refer to the exhibit. What is the purpose of the TCL script snippet?

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 87:

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 88:

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
pg 2-86 and 87 Integrating a VoIP Network to the PSTN and PBXs

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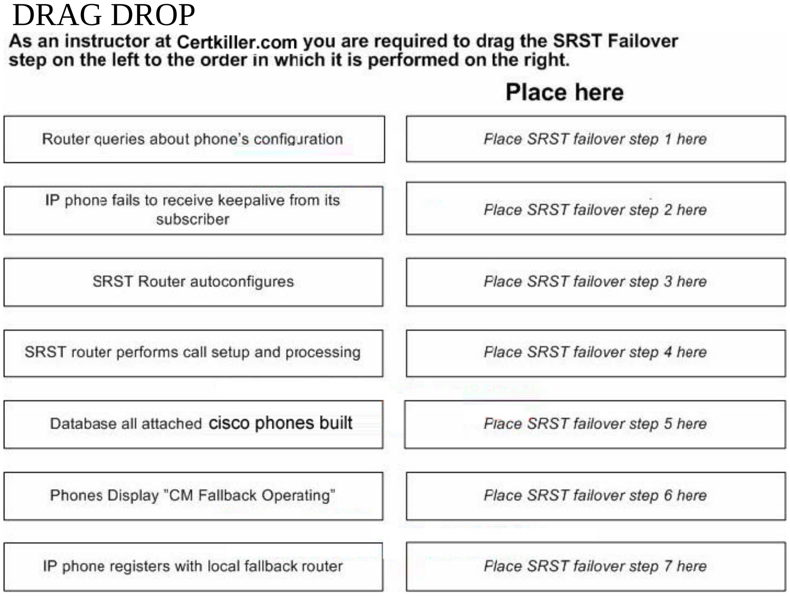
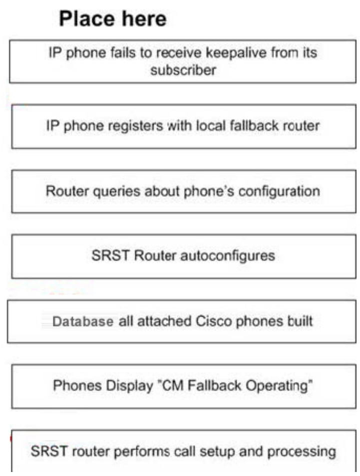
QUESTION 89:

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 90:

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 91:

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 92:

Exhibit

```
proc init { } {
global param
set param(interruptPrompt) true
set param(abortKey) *
set param (termin to key)

Prac out_setup () (
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
}
}
```

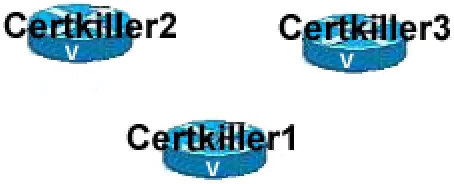
Refer to the exhibit. What is the purpose of the TCL script snippet?

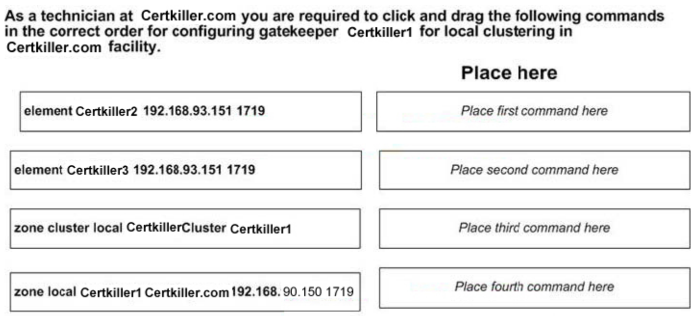
A. process a script exit
B. play an audio prompt
C. terminate call
D. gather initial digits
E. interrupt a call in progress

Answer: D

QUESTION 93:

DRAG DROP
Exhibit

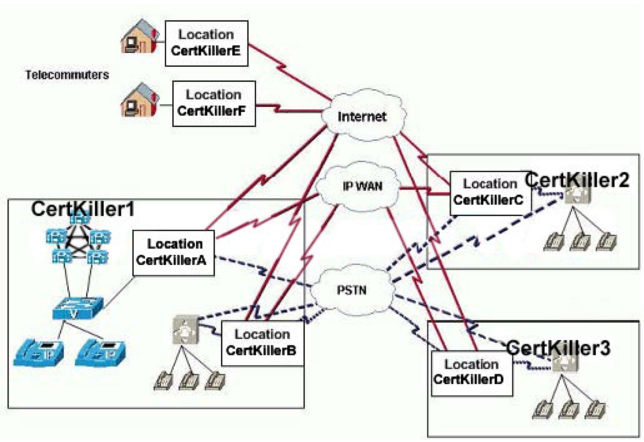




Answer:

QUESTION 94:

Exhibit



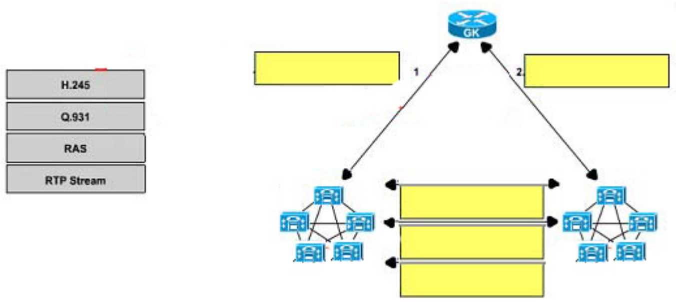
Refer to the exhibit. Certkiller .com is in the process of migrating from a traditional PBX telephony system to an IP telephony system at the Certkiller 1 headquarters. Certkiller .com would like to start migrating the regional offices in Certkiller 2 and Certkiller 3 off the existing tie-line and onto 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

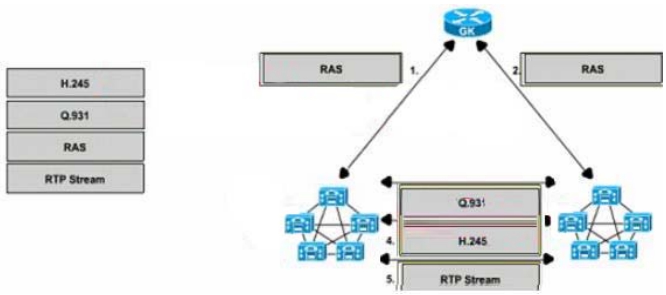
Answer: A, B, C, D

QUESTION 95:

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 order that 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 96:

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
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 2-78 and 86 Integrating a VoIP Network to the PSTN and PBXs

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QUESTION 97:

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.
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
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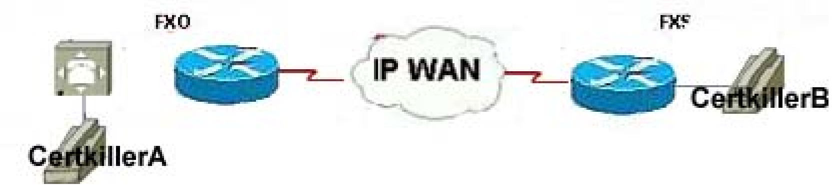
QUESTION 98:

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 99:

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 100:

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.
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pg 1-14 Function of Gateways and Gatekeepers

QUESTION 101:

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?

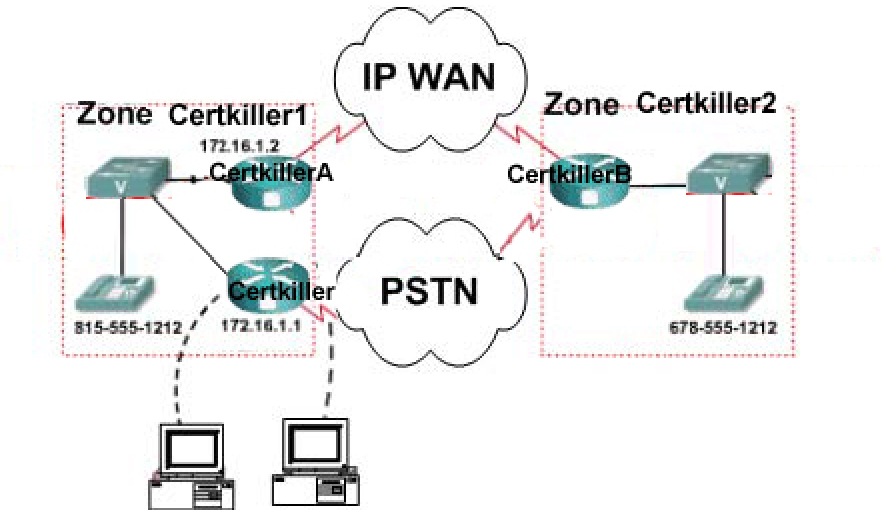
A. no digit-strip
B. prefix 1
C. num-exp .... 1202454....
D. Rule 1 ^202454 1

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 102:

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 103:

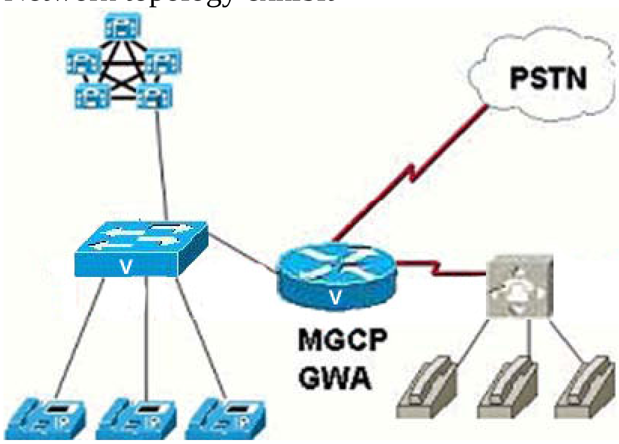
When COR is used in a gateway, under what circumstance will a call be completed between a specific pair of dial peers?

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 104:

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 105:

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 106:

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 Certkiller 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 Certkiller com (3 Questions)

QUESTION 107:

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 108:

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 109:

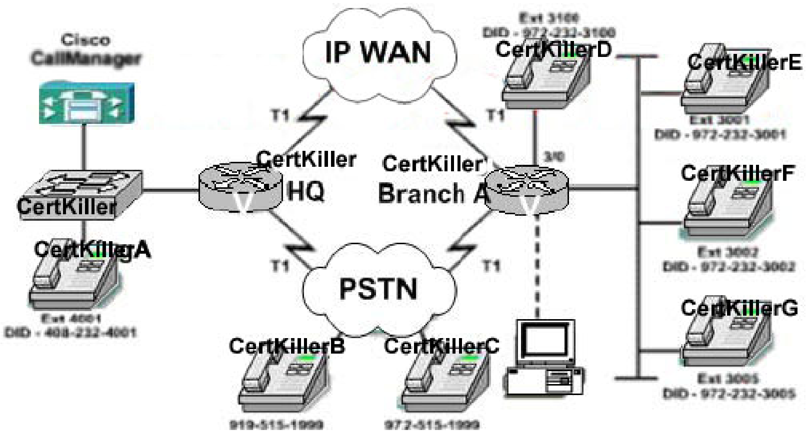
While talking to customers, Certkiller sales staff frequently conference in technical support. Certkiller .com would like to support 12 simultanous conference bridges with 4 participants each.
What is the minimum number of DSPs required in the gateway?

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

Answer: C

Case Study #2, Scenario

Network Topology Exhibit:



Simulation output exhibit #1:

```
CertkillerBranchA#show 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-CTR1-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-CTR1
 match 100aaa-same-name VOICE_CTR1


Policy-asp Tosatch
 class VOICE-OUY
  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 secure 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-relax ninear 384000
 !
control-plane
!
!
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
 Same Emergency
!
!
dial-peer cor list everyone
 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 110:

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 111:

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 112:

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 113:

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 114:

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.