

Exam : CISCO 642-453

Title : Gateway Gatekeeper(GWGK)

Version : Demo

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1. 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. 192 Kbps
- B. 240 Kbps
- C. 384 Kbps
- D. 480 Kbps

Answer: C

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

- A. The bandwidth commands for each zone that is registered with the directory gatekeeper are included in the directory gatekeeper configuration.
- 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 directory gatekeeper controls call bandwidth usage between the zones.

Answer: B

3. Refer to the exhibit. Which message ID can be used to track this call from the requesting endpoint?

```
4d16h: RecvUDP_IPSockData successfully rcvd message of length 134 from 10.100.100.100:49579
4d16h: ARQ (seq# 8204) rcvdpars_e_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) rcvdpars_e_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. 8204
- B. 10.100.100.99
- C. 10.200.99.99
- D. 49579

E. 10.100.100.100

F. 1640003356

Answer: A

4. 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 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.

C. 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.

D. 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.

Answer: B

5. Refer to the exhibit. 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?

```
Gateway#show dial-peer voice summary
dial-peer hunt 0
```

TAG	AD	TYPE	MIN	OPER	PREFIX	DEST-PATTERN	PRE	PASS	PER	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. Disable status checking for POTS dial peers so outbound call setup will activate the BRI.

B. Configure SPIDs on the BRI so that outbound calls are sent to the correct number.

C. Change the dial peer hunt logic so the appropriate dial peer is always used.

D. Configure preferences to always select dial peer 3.

Answer: A

6. 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. determining the intrasite modem and fax traffic patterns
- C. ensuring the correct location of voice gateways in the network
- D. provisioning a common codec for all WAN connections
- E. implementing the correct WAN topology
- F. determining if gatekeeper redundancy or high availability is required

Answer: DEF

7. 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 WAN link is bouncing.
- B. The keepalive timer in the SRST gateway is set too long.
- C. 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.
- D. 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.

Answer: C

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

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

Answer: D

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

Answer: C

10. Which statement is true regarding Cisco modem relay?

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

Answer: C

11. Refer to the exhibit. 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?

```
2811(config-app)#service aa flash:its-CISCO.2.0.1.0.tcl
2811(config-app-param)#
*Sep 20 17:24:11.545: //-1//HIFS:/hifs_ifs_cb: hifs_ifs file read succeeded. size=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
2811(config-app-param)#param operator 5000
Warning: parameter operator has not been registered under aa namespace
2811(config-app-param)#
```

- A. nothing, the warning may be ignored
- B. register the application with the gatekeeper
- 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 parameters with the Cisco Unified CallManager

Answer: A

12. 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

zone prefix hardware 408.....

zone prefix appliances 415.....

no shut

B. 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

C. 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

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 local VIA acme.com

zone prefix hardware 408*

zone prefix appliances 415*

no shut

Answer: D

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

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

Answer: D

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

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

Answer: BCE

15. 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. 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

16. Refer to the exhibit. Horst's Helicopters 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?

```
SRST-Router#show ccm-manager
MGCP Domain Name: BR1
Priority      Status      Host
-----
Primary      Down        10.1.5.10
First Backup  None
Second 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
SRST-Router#

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

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

Answer: A

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

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

Answer: D

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

- A. internal numbering plan
- B. path selection
- C. Class of Service
- D. digit manipulation
- E. call coverage

Answer: BD

19. Refer to the exhibit. 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 eam-wink-start
```

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

Answer: A

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