Exam : 642-452
Gateway Gatekeeper Exam

Demo Version

To Access Full Version, Please go to
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QUESTION 1:

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

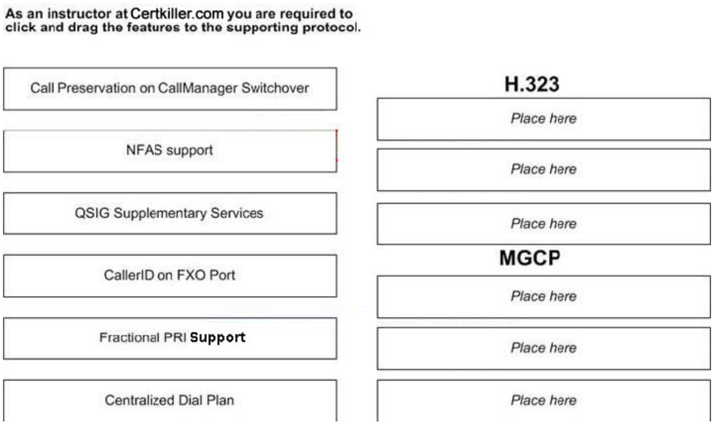
A. called ID
B. transcoding
C. E911
D. directory lookup
E. conference bridging
F. music on hold

Answer: B, E

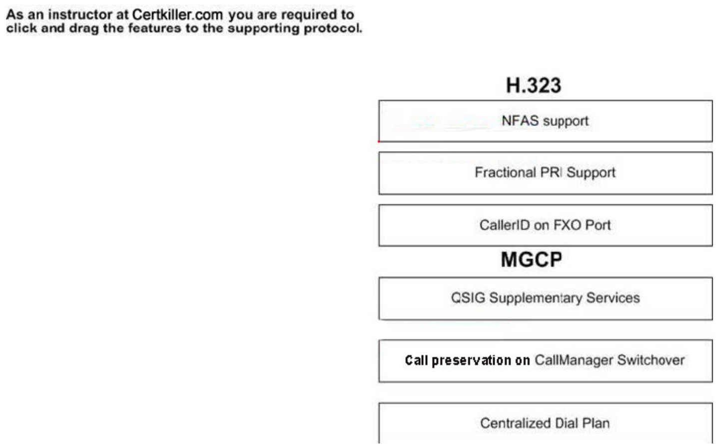
Explanation:
The DSP farm uses the DSP resources in network modules on Cisco
routers to provide voice-conferencing, transcoding, and hardware MTP services.
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 4-56 Implementing Advanced Gateway Features

QUESTION 2:

DRAG DROP



Answer:

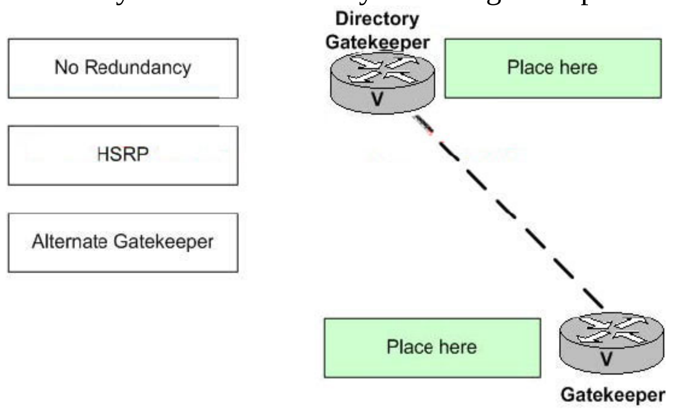


Explanation:
Why Choose H.323
* Integrated access
* Caller ID support on analog FXO
* Many more TDM interface types and signaling
* Dropping DSPs on hairpinned calls
* Gateway-resident applications like TCL and VXML
* CAC network design with H.323 gatekeepers
* No release dependencies between gateways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

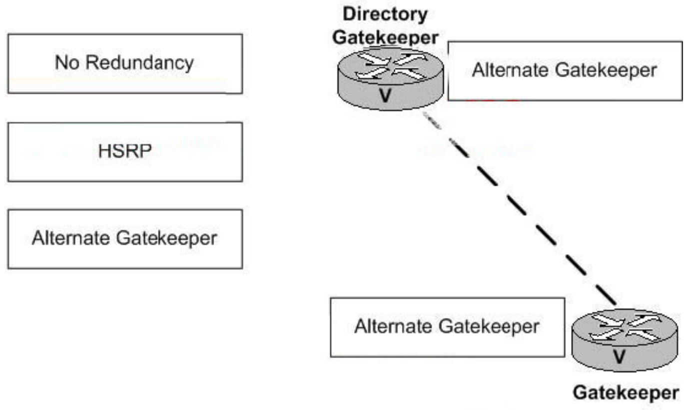
failover. This feature avoids dropped calls when applying the monthly operation system service release on the Cisco CallManagers
- In SRST mode, calls from IP phones through an MGCP gateway are preserved on MGCP fallback for calls on analog or CAS circuits. Calls on ISDN circuits are dropped on fallback.
Implementing Cisco Voice Gateways and Gatekeepers (GWGK) v1.0
pg 1-67 Function of Gateways and Gatekeepers

QUESTION 3:

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



Answer:



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

QUESTION 4:

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

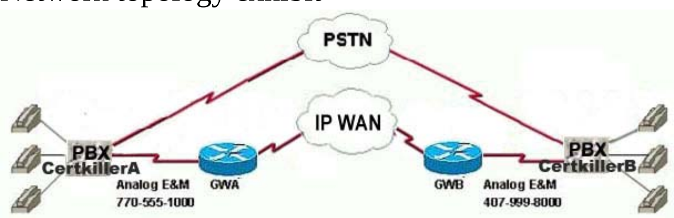
A. dsp services transcoding
B. associate application transcoding
C. dspfarm profile 10 transcoding
D. voice-card 2 transcoding

Answer: C

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

QUESTION 5:

Network topology exhibit



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

A. WAN link speeds

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

Answer: B, D, E, F

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

QUESTION 6:

Exhibit

```
1d20h: ISDN Se3/0:15: Outgoing call id = 0x85F4, dsl 0
1d20h: ISDN Se3/0:15: process_pri_call(): call id 0x85F4, number 35293315, speed 0,
call type VOICE, redialed? f, csm call? f, pdata? t
1d20h: calLED type/plan overridden by call_decode
1d20h: did't copy oct3a reason: not CALLER_NUMBER_IE
1d20h:  building outgoing channel id for call nfas_int is 0 len is 0
1d20h: ISDN Se3/0:15: TX <-  INFOc sapi = 0  tei = 0  ns = 19  nr = 19  i =
0x08020089050403809 0A31802008905040380 90A310200890504038090A315
1d20h:      SETUP pd = 8  callref = 0x8089
1d20h:          Bearer capability i : 0x8089
1d20h:          Channel ID i = 0xA98381
1d20h:          Progress Ind i = 0x8183 - Origination address is non-ISDN
1d20h:          Called Party Number i = 0x80, '35293315', Plan:Unknown, Type:Unknown
1d20h: ISDN Se3/0:15: RX <-  RRr sapi = 0  tei = 0  nr = 20
1d20h: ISDN Se3/0:15: RX <-  INFOc sapi = 0  tei = 0  ns = 19  nr = 20  i =
0x080280895A08028286
1d20h:      RELEASE_COMP pd = 8  callref = 0x8089
1d20h:          Cause i = 0x8286 - Channel unacceptable
1d20h: ISDN Se3/0:15: TX ->  RRr sapi = 0  tei = 0  nr = 20
1d20h: ISDN Se3/0:15: CCPRI_ReleaseCall(): bchan 1, call id 0x85F4, call type VOICE
1d20h: CCPRI_ReleaseChan released b_dsl 0 B_Chan 1
1d20h: ISDN Se3/0:15: LIF_EVENT: ces/callid 1/0x85F4 CALL_REJECTION
1d20h: ISDN Se3/0:15: LIF_EVENT: ces/callid 1/0x85F4 CALL_CLEARED
1d20h: ISDN Se3/0:15: received CALL_CLEARED  call_id 0x85F4
```

Itexamworld .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.
