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## 問題集

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**Exam** : **642-452**

**Title** : Gateway Gatekeeper Exam  
(GWGK)

**Version** : DEMO

1. Refer to the exhibit. Highland Park Property Development 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?

```

1d20h: ISDN se3/0:15: outgoing call id = 0x85F4, ds1 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: didn'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 =
0x080200890504038090A31803A983811E0281837009803335323933333135
1d20h:     SETUP pd = 8 callref = 0x0089
1d20h:     Bearer Capability i = 0x8090A3
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_ds1 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

```

- 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**

2. Which dial peer will send calls to the PSTN via the CAS T1 using this controller configuration?  
`controller t1 3/0 framing eslinecode b8zsds0-group 1 timeslots 1-24 type e&m-wink-start`

- A. `dial-peer voice 1 potsdestination-pattern 9.@port 3/0:1`
- B. `dial-peer voice 1 potsdestination-pattern 9.@port 3/0:24`
- C. `dial-peer voice 1 potsdestination-pattern 91port 3/0:1`
- D. `dial-peer voice 1 potsdestination-pattern 91port 3/0:24`

**Answer: C**

3. 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:** ABC

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

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

proc act_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
  }
}
```

- 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:** D

5. In a particular company, 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 field offices wish to dial only the last four digits. Which of the following Cisco IOS commands 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**

6. A company 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 potsdestination-pattern 1603.....port ipv4:10.172.163.5
- B. dial-peer voice 101 voipdestination-pattern 1603.....port 1/0
- C. dial-peer voice 102 voipdestination-pattern 1603.....session-target ipv4:10.172.163.5
- D. dial-peer voice 103 potsdestination-pattern 1603.....session-target 1/0

**Answer: C**

7. 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: BCE**

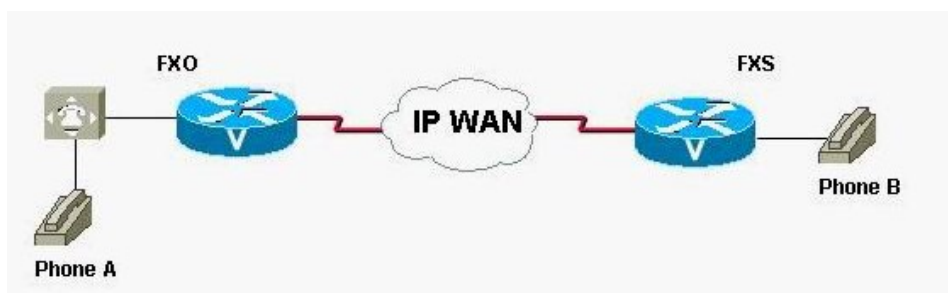
8. A service provider wants to add SIP devices to the existing H.323 voice network. Which Cisco device will allow the SIP devices to use the existing routing structure on the H.323 gatekeeper?

- A. SIP voice gateway
- B. Cisco SIP Proxy Server
- C. Cisco SIP Redirect Server

D. Cisco SIP Registrar Server

**Answer: B**

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

**Answer: B**

10. LAB

Question #1

Which two phones at Branch A can be reached from the PSTN while operating in SRST mode? (Choose two.)

- 1
- 2
- 3
- 4
- 5
- 6

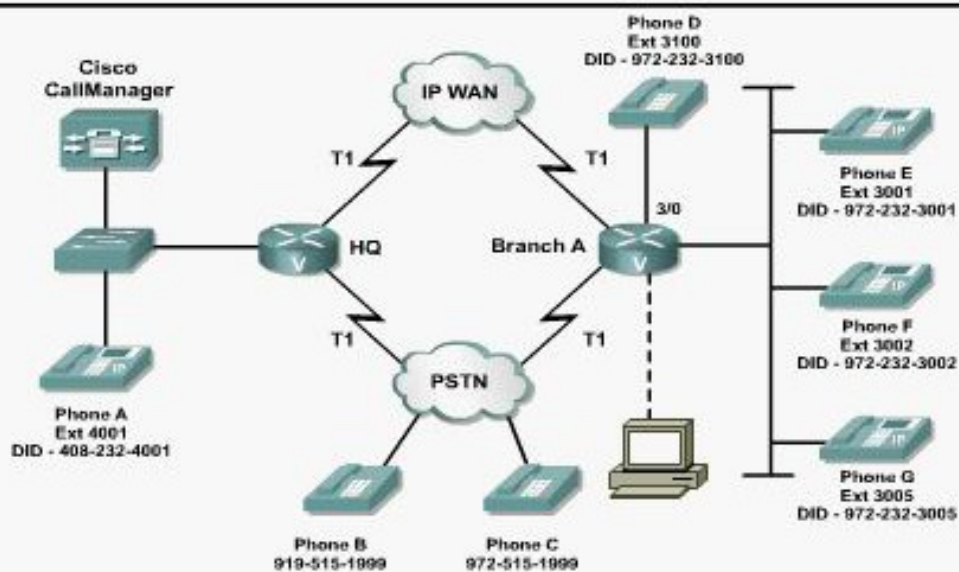
- Phone D
- Phone E
- Phone F
- Phone G

**eSIM™ Professional** 00:00:08  
Scenario 1 Version 1.0

This eSimlet consists of a router simulation and associated multiple choice items. It is necessary to use the output of the simulation to determine the answers to the multiple choice items.

To access the HyperTerminal screen to enter

Hide Topology




**Question #2**

- 1
- 2
- 3
- 4
- 5
- 6

Which two changes will allow all IP phones at Branch A to call each other while operating in SRST mode? (Choose two.)

- Remove all outgoing COR statements from call-manager-fallback.
- Assign all extensions to outgoing COR list admin in call-manager-fallback.
- Assign all extensions to outgoing COR list manager in call-manager-fallback.
- Assign all extensions to outgoing COR list emergency in call-manager-fallback.
- Assign all extensions to outgoing COR list everyone in call-manager-fallback.

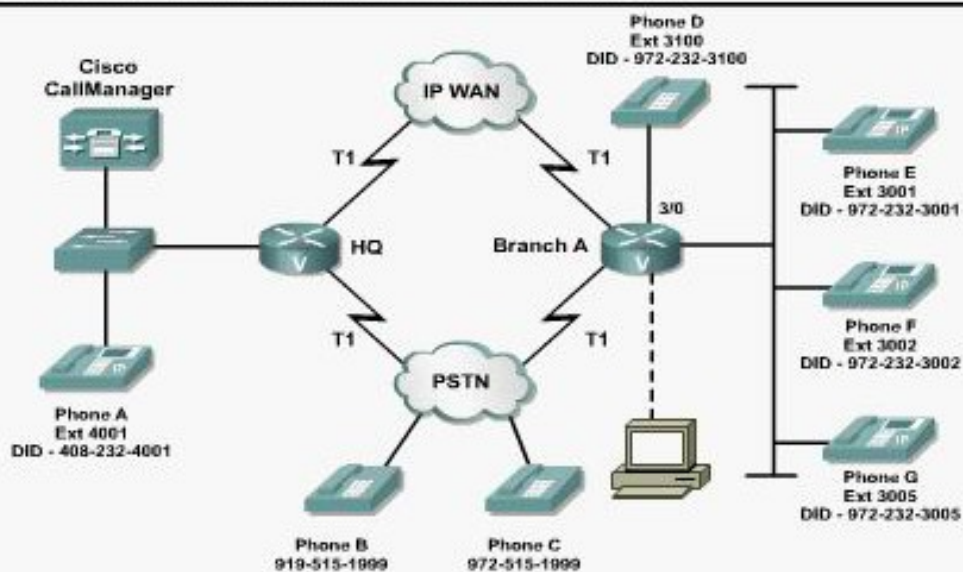


eSIM™ Professional 00:00:28  
Scenario 1 Version 1.0

This eSimlet consists of a router simulation and associated multiple choice items. It is necessary to use the output of the simulation to determine the answers to the multiple choice items.

To access the HyperTerminal screen to enter

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Question #3

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

- 1
- 2
- 3
- 4
- 5
- 6

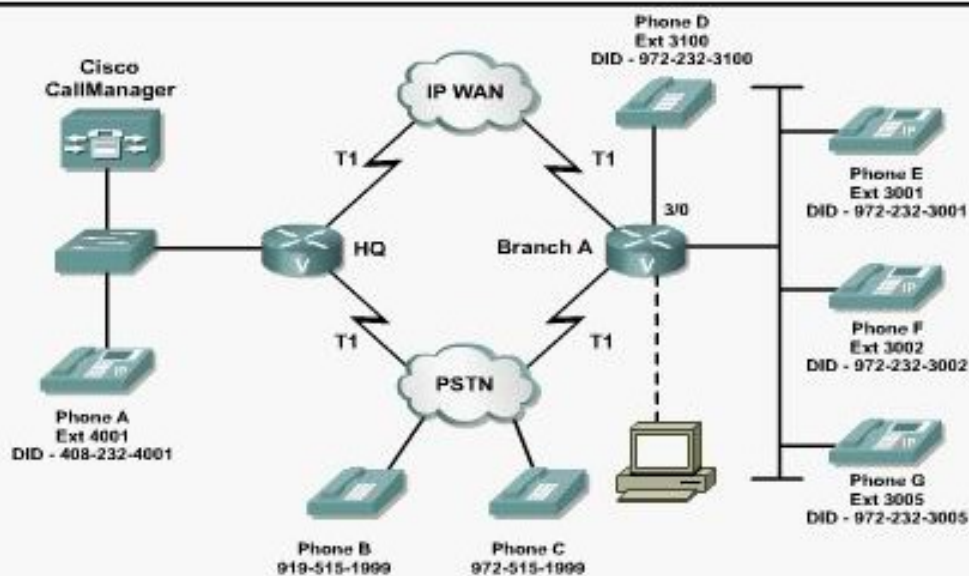
- voice-port 3/0 followed by caller-name Lobby
- voice-port 3/0 followed by station-id name Lobby
- dial-peer voice 3100 pots followed by caller-name Lobby
- dial-peer voice 3100 pots followed by station-id name Lobby

**eSIM™ Professional** 00:00:44  
Scenario 1 Version 1.0

This eSimlet consists of a router simulation and associated multiple choice items. It is necessary to use the output of the simulation to determine the answers to the multiple choice items.

To access the HyperTerminal screen to enter

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Question #4

Branch A has experienced an IP WAN outage and is operating in SRST mode. Which three phones can Phone F call while in SRST mode? (Choose three.)

- 1
- 2
- 3
- 4
- 5
- 6

- Phone A
- Phone B
- Phone C
- Phone E
- Phone G

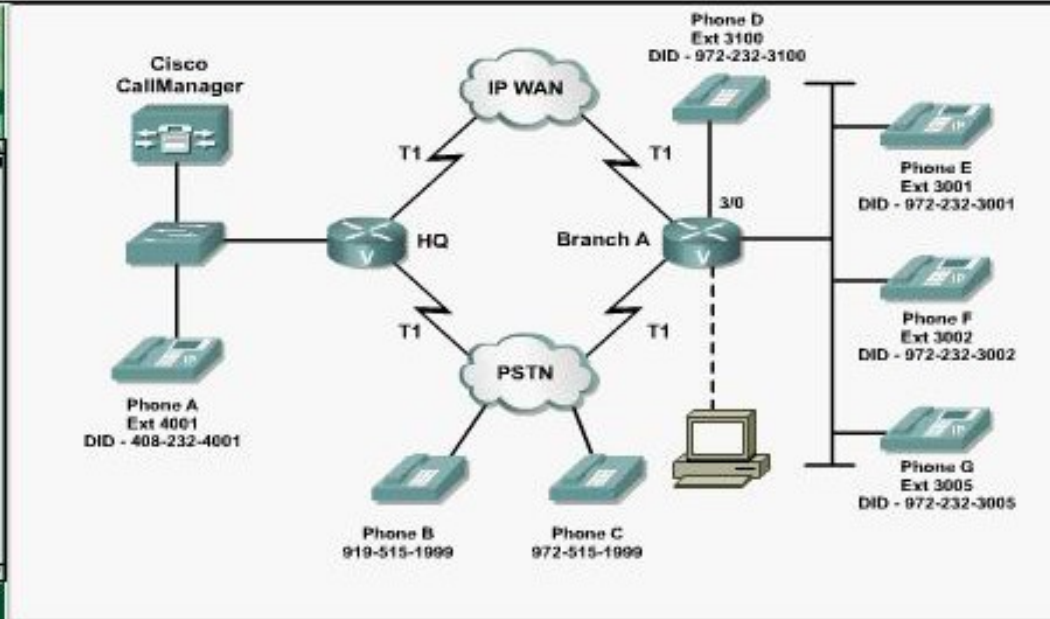


eSIM™ Professional 00:01:01  
Scenario 1 Version 1.0

This eSimlet consists of a router simulation and associated multiple choice items. It is necessary to use the output of the simulation to determine the answers to the multiple choice items.

To access the HyperTerminal screen to enter

Hide Topology




Question #5

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

- 1
- 2
- 3
- 4
- 5
- 6

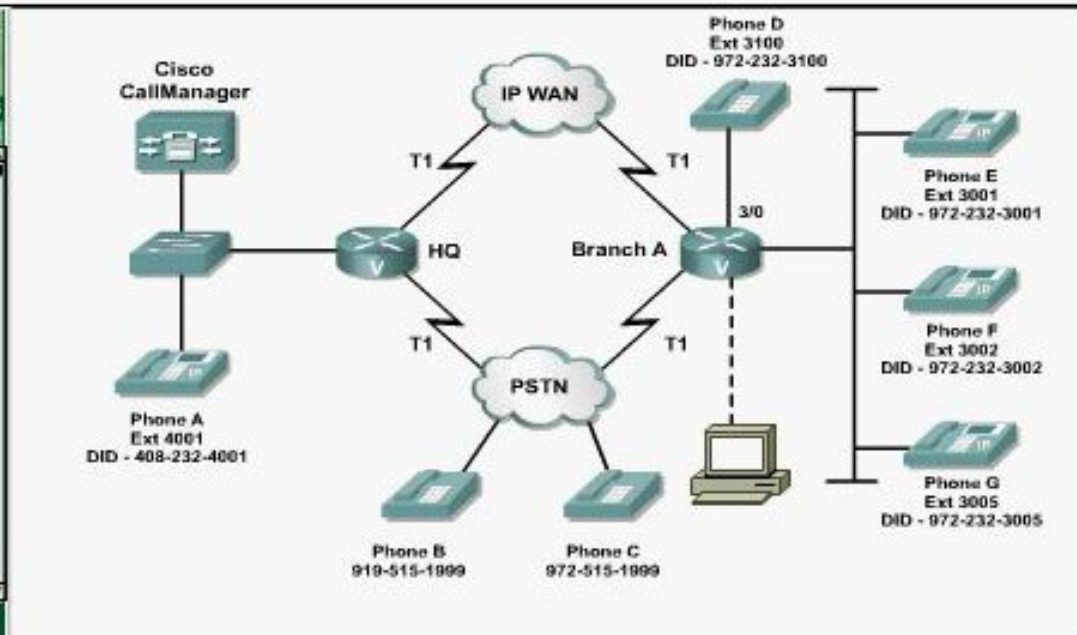
- 3000
- 3100
- 9722323000
- 9722323100

 eSIM™ Professional 00:01:26  
Scenario 1 Version 1.0

This eSimlet consists of a router simulation and associated multiple choice items. It is necessary to use the output of the simulation to determine the answers to the multiple choice items.

To access the HyperTerminal screen to enter

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**Question #6**

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

- Enable fax pass-through using the mgcp command.
- Enable fax pass-through using the ocm-manager command.
- Disable fax relay using the ocm-manager command.
- Disable fax relay using the mgcp command.

eSIM™ Professional 00:01:45  
Scenario 1 Version 1.0

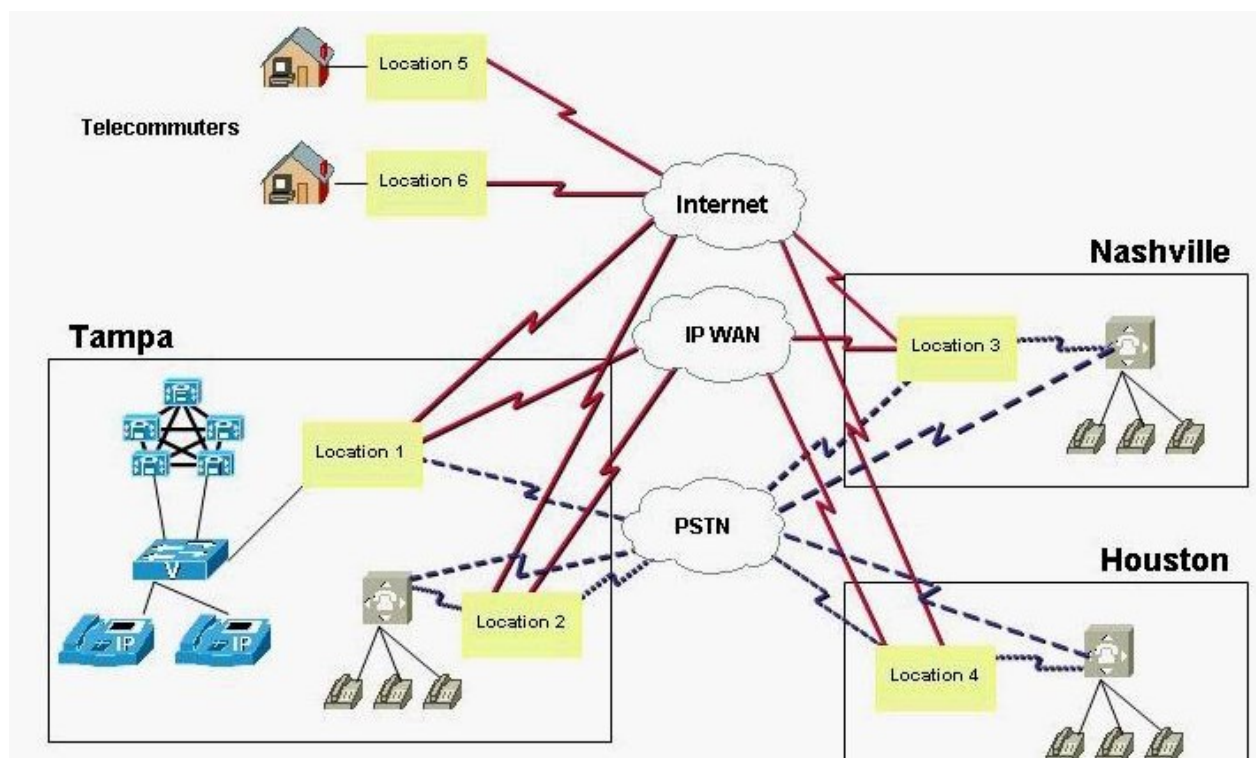
This eSimlet consists of a router simulation and associated multiple choice items. It is necessary to use the output of the simulation to determine the answers to the multiple choice items.

To access the HyperTerminal screen to enter

[Hide Topology](#)

The diagram illustrates a network topology for a migration project. It features two routers: HQ and Branch A. HQ is connected to Branch A via an IP WAN link. Both HQ and Branch A are connected to a central PSTN cloud via T1 lines. Phone A (Ext 4001, DID - 408-232-4001) is connected to HQ. Phone B (919-515-1999) and Phone C (972-515-1999) are connected to the PSTN. Phone D (Ext 3100, DID - 972-232-3100) is connected to Branch A via a 310 line. Phone E (Ext 3001, DID - 972-232-3001), Phone F (Ext 3002, DID - 972-232-3002), and Phone G (Ext 3005, DID - 972-232-3005) are connected to Branch A via a shared line. A Cisco CallManager server is connected to HQ. A computer is connected to Branch A via a dashed line.

11. Refer to the exhibit. A client is in the process of migrating from a traditional PBX telephony system to an IP telephony system at the Tampa headquarters. The client would like to start migrating the regional offices in Nashville and Houston off the existing tie-lines and onto the IP WAN. In which locations would voice-enabled gateways need to be deployed? (Choose four)



- A. location 1
- B. location 2
- C. location 3
- D. location 4
- E. location 5
- F. location 6

**Answer:** ABCD

12. 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 the 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 lists in the inbound and outbound dial peers have no matching members

**Answer:** B

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

signaling?

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

**Answer: A**

14. Highland Park Property Development 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**

15. Which two features are benefits of using HSRP for gatekeeper redundancy? (Choose 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: BE**

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

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

- 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**

17. Which two functions are provided by a DSP farm? (Choose two.)

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

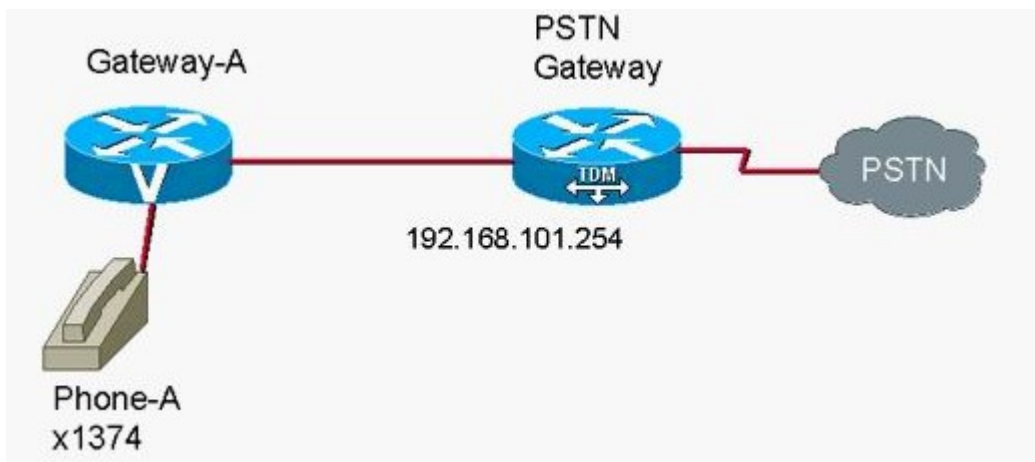
**Answer: BE**

18. Acme Widgets 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**

19. Refer to the exhibit. Acme Widgets 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 Gateway-A, which types of calls can Phone-A actually make via the PSTN?





Partial configuration on Gateway-A:

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

dial-peer cor list Em01
  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

dial-peer voice 10 voip
  corlist outgoing LD01
  destination-pattern 91[2-9]..[2-9].....
  session target ipv4:192.168.101.254
```

```
dial-peer voice 100 voip
  corlist outgoing Intl01
  destination-pattern 9011T
  session target ipv4:192.168.101.254
```

- 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**

20. 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 conjunction with gateway proxies to provide scalable video solutions
- D. enhances policy-routing capability by assigning carrier IDs to partner gateways

**Answer: A**