

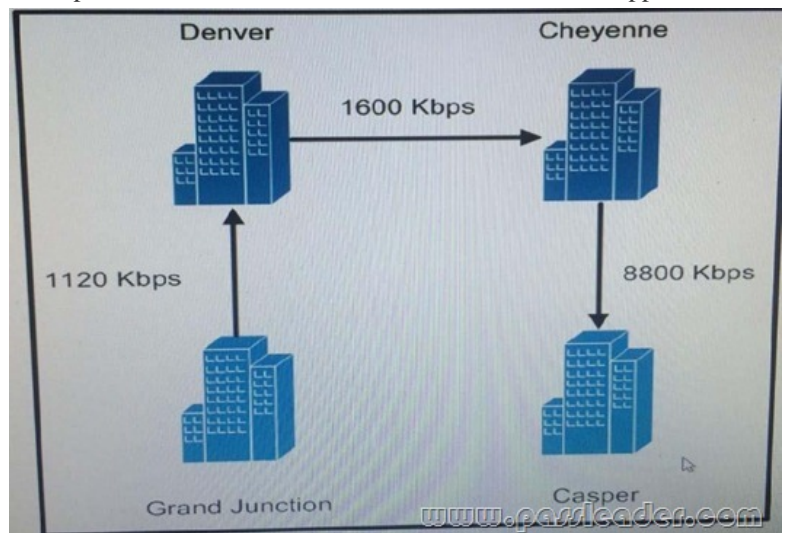
## [July-2016 Dumps PassLeader Free New Update 400-051 Exam Questions Collection

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```
voice service voip
 no ip address trusted authenticate
 allow-connections sip to sip
 sip
 no update-callerid
!
voice register global
 mode cme
 max-dn 10
 max-pool 10
!
voice register dn 1
 number 5000 name Phone A
!
voice register dn 2
 number 5001 name Phone B
!
voice register dn 3
 number 5002 name Phone C
!
voice register pool 1
 id mac 0000.1111.112A
 type 8841 number 1 dn 1
!
voice register pool 1
 id mac 0000.1111.112B
```

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A. The registrar server command must be added under the voice register global configuration  
B. The IP address trusted authenticate command must be added under voice service voip  
C. The source-address command must be added under the voice register global configuration  
D. The local SIP proxy address must be configuration under the sip-ua configuration  
E. The registrar server command must be added under the sip section of voice service voip **Answer: CE**  
**NEW QUESTION 397** A collaboration engineer has been asked to implement secure real-time protocol between a Cisco Unified CM and SIP gateway. Which option is a consideration for this implementation? A. Only T.38 and Cisco fax protocol are supported B. SIP require the all the time be sent in GMT C. Call hold RE-INVITE is not supported D. SRTP is supported only in cisco IOS 15.x and higher **Answer: B**  
**NEW QUESTION 398** Refer to the exhibit. A collaboration engineer configures Cisco Unified CM location using G.711 and iLBC for each site. The bandwidth for each link is shown. Which two options represent the maximum concurrent number of calls supported from Grand Junction to Casper for each Codec? (Choose two)



A. 20 G.711 calls    B. 18 G.711 calls    C. 36 iLBC calls  
D. 42 iLBC calls    E. 11 G.711 calls    F. 51 iLBC calls

**Answer: CE NEW QUESTION 399** A collaboration engineer is troubleshooting an MOH problem on a Cisco IOS SIP gateway. While searching through a debug ccsip message output, which three parameters in the SIP messages can be used to determine if the call was placed on hold? (Choose three) A. OPTIONS WITH 301 CALLHOLD B. INVITE WITH a=INACTIVE C. INVITE WITH a=SENDONLY D. OPTION WITH c=INACTIVE E. c=IN IP4 0.0.0.0 F. BYE WITH A = CALLHOLD

**Answer: BCE NEW QUESTION 400** Refer to the exhibit. A cisco collaboration engineer discovers that an instance of IOS media termination point (MTP) could not maintain stable registration with CUCM. Call manager traces is showing in the exhibit. What is the reason for the flapping registration?

```
NOC_CALLINAGER -> StatusConnectionRecv: [DeviceName=MFTFITEA][ReasonCode=0][ClusterID=StreamDoneCluster]
[MediaTerminalPointControl(176)] Status device is closing the connection
[AppInfo] New connection accepted. DeviceName= TCPv4 = (1.100.14.145),
[Header] 79, 88, 145, 1, Reason=0x0, Device Control=(0,0,0)
[Socket] [StatusCloseTwaking [MediaTerminalPointControl(1,100,137,136) [StreamIndex(1,100,42)],
1.100.14,145-2732,39,140-1]] [IRV-MIO,MIO,Liv,Vio,Dio] CloseStreamReason = 0 & StreamId =
[AppInfo] [MediaTerminalPointControl(176)] [App_ReasonCode DeviceName= MFTFITEA
[AppInfo] [MediaTerminalPointControl(176)] [SetTotalCounter Counter=150
[AppInfo] [MediaTerminalPointControl(176)] [MediaLiveCounter Counter=100
[AppInfo] [MediaTerminalPointControl(176)] [MediaLiveCounter Counter=0
[AppInfo] [MediaTerminalPointControl(176)] [MediaLiveCounter Counter=0
Delete Data Structure for HqoControl Successes - Device = MFTFITEA ,Index= 1
[HALD] [MediaTerminalPointControlConf] [Building Conf] [MediaTerminalPointControl(1,100,137,136)]
[AppInfo] [MediaTerminalPointControl(176)] [Building New_MediaTerminalPointConf - Device = MFTFITEA - Unregistered
```

A. The CCM version on IOS configuration does not match the CUCM version. B. The IOS MTP is experiencing high CPU and is missing its keep-alive. C. A Firewall is blocking port 2000 intermittently between IOS Device and CUCM. D. Another IOS Media device is attempting to register with the same name. **Answer: D**

**NEW QUESTION 401** A collaboration engineer is designing Cisco IM&P implementation to support instant messaging logging for compliance. Which two external databases can be used to support that functionality? (Choose two.) A. Oracle database B. MySQL database C. Microsoft SQL database D. PostgreSQL database E. Informix SQL database **Answer: AD**

**NEW QUESTION 402** Refer to the exhibit. A cisco collaboration engineer is troubleshooting a gateway and gatekeeper problem and sees this output from a debug command. Which two configuration can cause this problem? (Choose two)

```

RAS INCOMING PDU
value RasMessage
{
    requestSeqNum
    protocolIden
    rejectReason
    {
    }
    gatekeeperId
}

```

A. The same zone prefix is configured in two different gatekeepers  
 B. The same H323-ID is configured in two different gateways  
 C. The same gw-type-prefix is configured in two different zone subnets  
 D. The same zone subnet ID is configured in two different gatekeepers  
 E. The same E164-ID is configured in two different gateways

**Answer: BE**

**NEW QUESTION 403** The Cisco Unified Border Element is configured using high availability with the Hot Standby Routing Protocol. Which two pieces of information can be gathered about the calls traversing these border elements? (Choose two.)

```
CUBE_PSTN#show voice high-avail

===== Voice HA DB INFO =====
Number of calls in HA DB: 100
Number of calls in HA sync pend
Number of calls in HA preserved
```

A. The total number of calls is 150. B. The number of non-native calls is 70.  
C. The number of native calls is 50. D. The number of calls preserved is 220.  
E. The total number of active calls is 100. **Answer: AB NEW QUESTION 404** Refer to the exhibit.  
Which two SIP packet handling behaviour will result with this cisco Unified Border Element (CUBE) configuration? (Choose two)

```
voice service voip
no ip address trusted authenticate
rtcp keepalive
mode border-element
allow-connections sip to sip
redundancy
no supplementary-service sip refer
signaling forward unconditional
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
min-se 360 session-expires 360
header-passing
error-passthru
asserted-id pai
midcall-signaling passthru
privacy-policy passthru
pass-thru headers unsupp
pass-thru content unsupp
no call service stop
```

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A. Unsupported content/MIME pass-through B. SIP Refer is not support when received on this CUBE C. Privacy headers received on SIP message will be replaced with NON-privacy headers on this CUBE D. P-Preferred identities E. Mid-call codec changes

**Answer: AE** **NEW QUESTION 405** A CUCM engineer has deployed Type B SIP Phones on a remote site and no SIP dial rules were deployed for these phones. How Will CUCM receive the DTMF after the phone goes off-hook and the button are pressed?

A. Each digit will be received by CUCM in a SIP NOTIFY message as soon as they are pressed.

B. The first digit will be received in a sip invite and subsequent digits will be received using NOTIFY message as soon as they are pressed.

C. Each digit will be received by CUCM in a SIP INVITE as soon as the dial soft key has been pressed.

D. All digits will be received by CUCM in a SIP INVITE as soon as the dial soft key has been pressed.

**Answer: A** **NEW QUESTION 406** The Video engineer wants to enable the LATM codec to allow video endpoint to communicate over audio With other IP devices Which two Characteristic should the voice engineer be aware of before enabling LATM on the Cisco Unified border element router? (Choose two)

A. Dual tone

B. Multi-frequency interworking with LATM codec is not supported

C. Codec transcoding between LATM and other codecs is not supported

D. SIP UPDATE message outlined in RFC3311 is not supported

E. Box-to-Box High availability support feature is not supported

F. Configure LATM under a voice class or dial peer is not supported

**Answer: AB** **NEW QUESTION 407** ?? Download the newest PassLeader 400-051 dumps from

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