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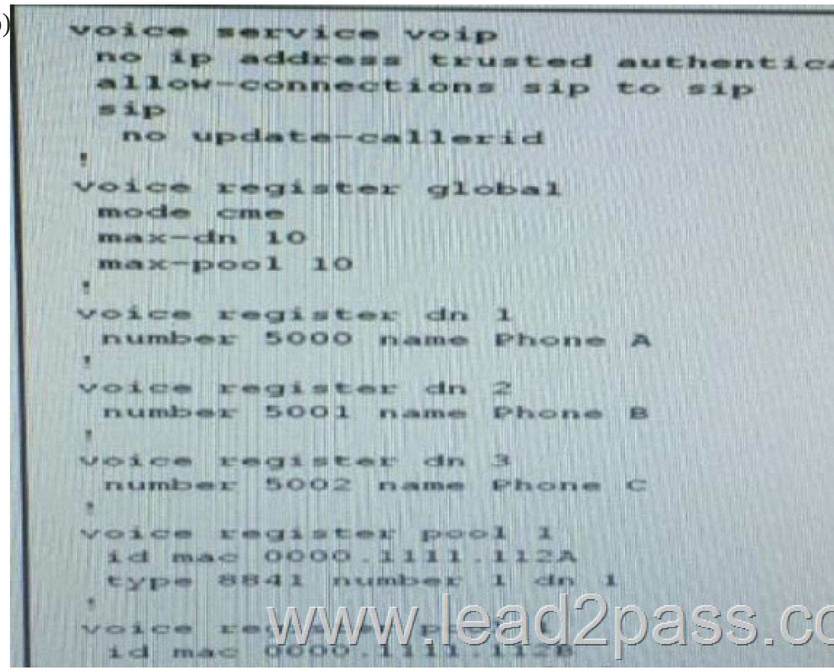
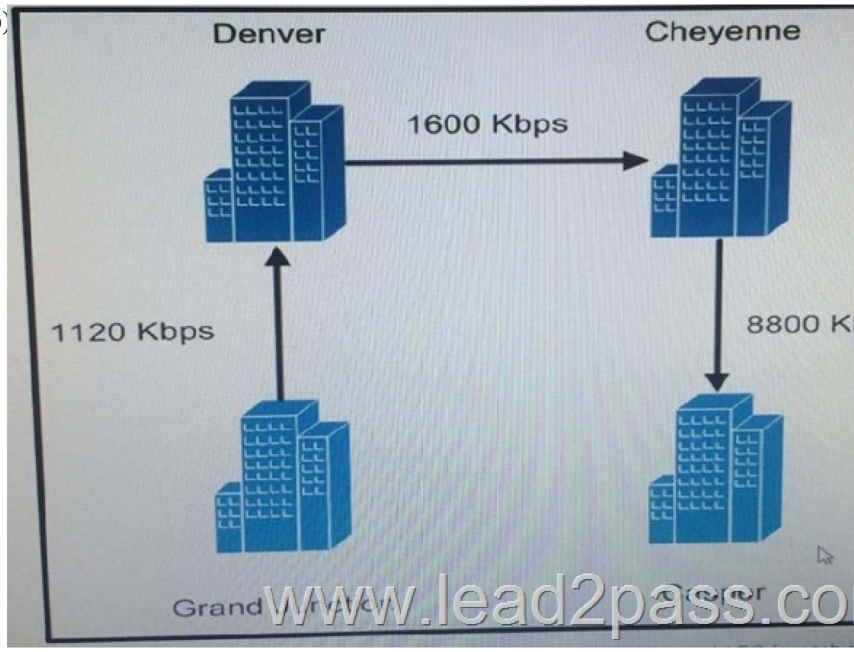
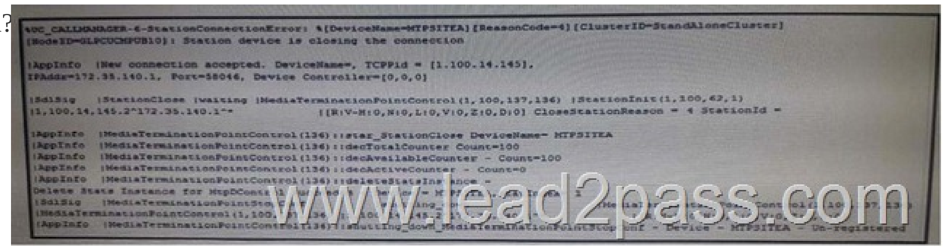

```
voice service voip
 no ip address trusted authentic
 allow-connections sip to sip
 sip
  no update-callerid
!
voice register global
 mode cme
 max-dn 10
 max-pool 10
!
voice register dn 1
 number 5000 name Phone A
!
voice register dn 2
 number 5001 name Phone B
!
voice register dn 3
 number 5002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


A. The registrar server command must be added under the voice register global configurationB. The IP address trusted authenticate command must be added under voice service voipC. The source-address command must be added under the voice register global configurationD. The local SIP proxy address must be configuration under the sip-ua configurationE. The registrar server command must be added under the sip section of voice service voip Answer: CE QUESTION 397A 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 supportedB. SIP require the all the time be sent in GMTC. Call hold RE-INVITE is not supportedD. SRTP is supported only in cisco IOS 15.x and higher Answer: B QUESTION 398Refer 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)



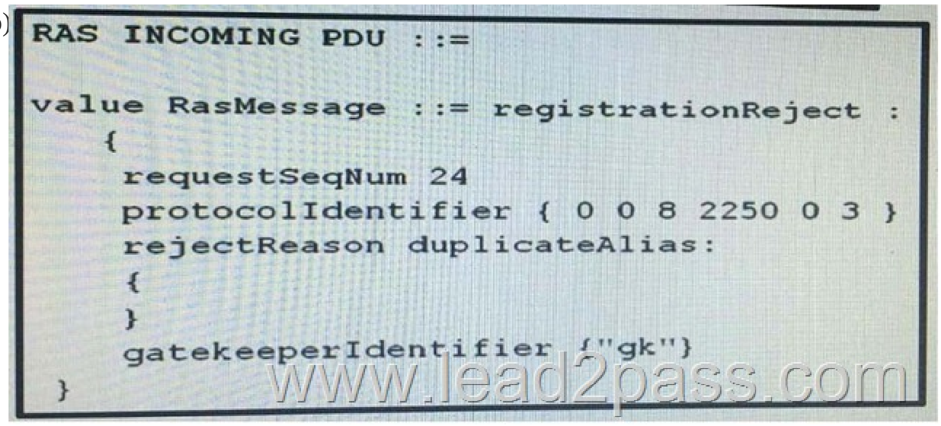
A. 20 G.711 callsB. 18 G.711 callsC. 36 iLBC callsD. 42 iLBC callsE. 11 G.711 callsF. 51 iLBC calls Answer: CE QUESTION 399A collaboration engineer is troubleshooting an MOH problem on a Cisco IOS SIP gateway. While searching through a debug ccsip message output, which three parameters in the SIP messages can be used to determine if the call was placed on hold? (Choose three) A. OPTIONS WITH 301 CALLHOLDB. INVITE WITH a=INACTIVEC. INVITE WITH a=SENDONLYD. OPTION WITH c=INACTIVEE. c=IN IP4 0.0.0.0F. BYE WITH A = CALLHOLD Answer: BCE QUESTION 400Refer 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?



```
|AppInfo  |New connection accepted. DeviceName=, TCPPid = [1.100.14.145],
IPAddr=172.35.140.1, Port=58046, Device Controller=[0,0,0]

|AppInfo  |MediaTerminationPointControl(136)::decTotalCounter Count=100
|AppInfo  |MediaTerminationPointControl(136)::decAvailableCounter - Count=100
|AppInfo  |MediaTerminationPointControl(136)::decActiveCounter - Count=0
```

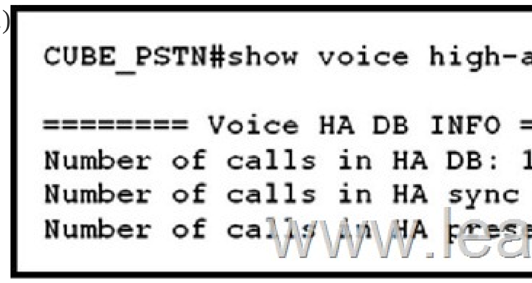

A. The CCM version on IOS configuration does not match the CUCM version.B. The IOS MTP is experiencing high CPU and is missing its keep-alive.C. A Firewall is blocking port 2000 intermittently between IOS Device and CUCM.D. Another IOS Media device is attempting to register with the same name. Answer: D QUESTION 401A 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 databaseB. MySQL databaseC. Microsoft SQL databaseD. PostgreSQL databaseE. Informix SQL database Answer: AD QUESTION 402Refer 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 ::= registrationReject :
  {
    requestSeqNum 24
    protocolIdentifier { 0 0 8 2250 0 3 }
    rejectReason duplicateAlias:
    {
    }
    gatekeeperIdentifier {"gk"}
  }
```

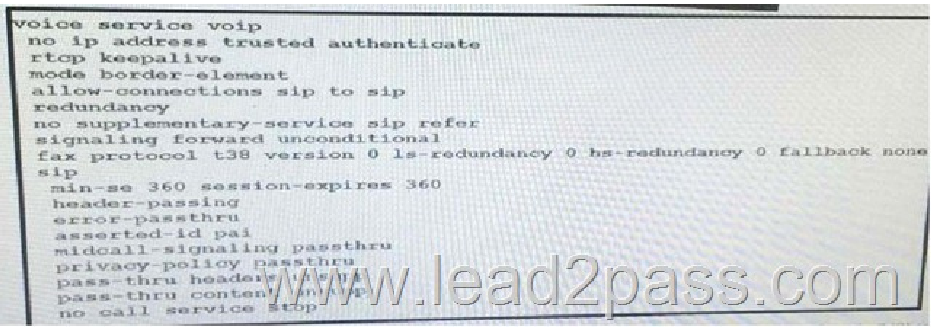

A. The same zone prefix is configured in two different gatekeepersB. The same H323-ID is configured in two different gatewaysC. The same gw-type-prefix is configured in two different zone subnets IDsD. The same zone subnet ID is configured in two different gatekeepersE. The same E164-ID is configured in two different gateways Answer: BE QUESTION 403The 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-a

======== Voice HA DB INFO =
Number of calls in HA DB: 1
Number of calls in HA sync
Number of calls in HA pres
```

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 QUESTION 404Refer to the exhibit. Which two SIP packet handing 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
```


A. Unsupported content/MIME pass-throughB. SIP Refer is not support when received on this CUBEC. Privacy headers received on SIP message will be replaced with NON-privacy headers on this CUBED. P-Preferred identitiesE. Mid-call codec changes Answer: AE QUESTION 405A 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 pressedB. 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 bill 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 QUESTION 406The 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 Multi-frequency interworking with LATM codec is not supportedB. Codec transcoding between LATM and other codecs is not supportedC. SIP UPDATE message outlined in RFC3311 is not supportedD. Box-to-Box High availability support feature is not supportedE. Configure LATM under a voice class or dial peer is not supportedF. Basic calls using flow-around or flow-through is not supported Answer: AB Lead2pass 400-051 pdf dumps is perfect! Totally! Thanks so much! 2016 Cisco 400-051 exam dumps (All 454 Q&As) from Lead2pass: http://www.lead2pass.com/400-051.html [100% Exam Pass Guaranteed!!!]