[February-2023100% Exam Pass-350-801 PDF and 350-801 VCE Free from Braindump2go[Q110-Q144

February/2023 Latest Braindump2go 350-801 Exam Dumps with PDF and VCE Free Updated Today! Following are some new Braindump2go 350-801 Real Exam Questions!QUESTION 110A remote office has a less-than-optimal WAN connection end experiences packed loss, delay, and jitter.Which VoIP codec should be used in this situation?A. G.722.1B. G.711alawC. iLBC D. G.729AAnswer: CExplanation:iLBC: iLBC provides audio quality between that of G.711 and G.729 at bit rates of 15.2 kbps (38-bytes or 20msec) and 13.3 kbps (50 bytes or 30 msec). iLBC handles lossy networks in better way than G729 because it treats each packet independently. G729 depends on the previous packet to handle packet loss, jitter and delay which doesn't tolerate well in lossy networks.QUESTION 111A user reports transfer failures from an IP phone for calls received from a PSTN to another PSTN number.What is a reason for these failures?A. The incoming calling search space of the SIP trunk does not include the partition of the line in the IP phone.B. The service parameter related to Offnet to Offnet call Transfer is set to TRUE.C. The IP phone is configured with the Wrong region.D. The gateway is configured with the wrong device pool.Answer: BExplanation: https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_1_3/ccmfeat/fsxfer.html#wp1048448QUESTION 112A customer is deploying a SIP IOS gateway for a customer who requires that in-bond DTMF relay is first priority and out-of-band DTMF relay is second priority. Which IOS entry sets the required priority?A. dtmf-relay cisco-rtpB. sip-notify dtmf-relay rtp-nte C. dtmf-relay rtp-nte sip-notifyD. dtmf-relay sip-kmpl cisco-rtpAnswer: CQUESTION 113Refer to the exhibit. This INVITE is sent to an endpoint that only supports G.729. What must be done for this call to succeed?

```
INVITE sip:4000@172.16.1.1:5061 SIP/2.0
Via: SIP/2.0/TLS 172.16.2.143:5061;branch=z9hG4bK8FD315E7
Remote-Party-ID: <sip:+14088335000@172.16.2.143>;party=calling;screen=no; pr
From: <sip:+14088335000@172.27.2.143>;tag=7B42E5F6-9B8
To: <sip:4000@172.16.1.1>
Date: Tue, 06 Aug 2019 15:03:05 GMT
Call-ID: 4EA4363-B77111E9-8A4AFFCF-10B6D71B@172.16.2.143
Supported: 100rel,timer,resource-priority, replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 0082391505-3077640681-2319777743-0280418075
User-Agent: Cisco-SIPGateway/IOS-15.5.3.S4b
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, N
CSeq: 101 INVITE
Timestamp: 1565089565
Contact: <sip:+ 14088335000@172.16.2.143:5061;transport=tls>
Expires: 180
Al[illegible]
Ma[illegible]
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 416
v=0
o=CiscoSystemsSIP-GW-UserAgent 8486 8298 IN IP4 172.16.2.143
s=SIP Call
c=IN IP4 172.16.2.143
t=0 0
m=audio 44612 RTP/SAVP 0 101
c=IN IP4 172.16.2.143
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
```


A. Noting: both sides support G.729.B. Add a transcoder that supports G.711ualw and G.729.C. ADD a media termination point that supports G.711ulaw and G.729.D. Nothing: both sides support payload type 101.Answer: BExplanation:A transcoder takes the media stream of one codec and transcodes (converts) it from one compression type to another compression type. For example, it could take a stream from a G.711 codec and transcode (convert) it in real time to a G.729 stream. In addition to codec conversion, a transcoder resource can also provide MTP/TRP functionality to a call.QUESTION 114Refer to the exhibit. You deploy Mobile and Remote Access for Jabber and discover that Jabber for Windows does not register to cisco Unified Communications Manager while outside of the office. What is a cause of this issue?

```
C:\Users\CISCO>nslookup
Default Server: dns.example.com
Address: 192.168.100.1

>set type=SRV
> _collab-edge._tcp.example.com
Server: dns.example.com
A[illegible]

Non-authoritative answer:
_collab-edge._tcp.example.com          SRV service l
          priority       = 10
          weight         = 10
          port           = 8443
          srv hostname   = expe.example.com
```


A. The DNS record should be created for _ cisco-uds _tcp example.com.B. The DNS record should be changed from _collab-edge._tcp.example.com to __collab-edge _tls.example.com.C. The DNS record type should be changed from SRV to A.D. Server 4.2.2.2 is not a valid DNS server.Answer: BExplanation:

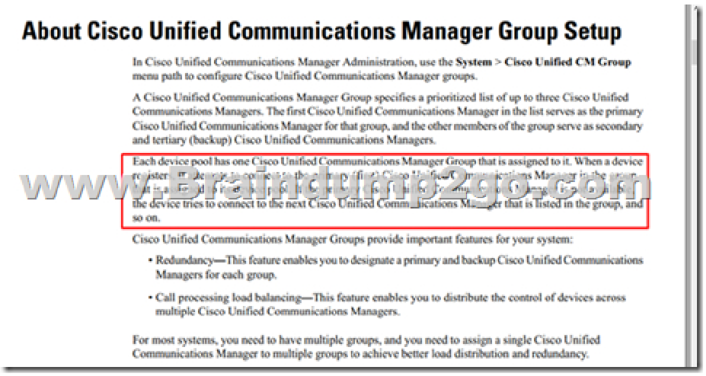
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/jabber/Windows/9_7/CJAB_BK_C606D8A9_00_cisco-jabber-dns-configuration-guide/CJAB_BK_C606D8A9_00_cisco-jabber-dns-configuration-guide_chapter_010.htmlQUESTION 115What is the major difference between the two possible Cisco IM and presence high-availability modes?A. Balanced mode provides user load balancing and user failover only for manually generated failovers.Active/standby made provides an unconfigured standby node in the event of an outage, but it does not provide load balancing.B. Balanced mode provides user load balancing and user failover in the event of an outage.Active/standby mode provides an always on standby node in the event of an outage, but it does not provide load balancing.C. Balanced mode does not provide user load balancing, but it provides user failover in the event of an outage. Active/standby mode provides an always on standby node in the event of an outage, but is does not provide load balancing.D. Balanced mode provides user load failover in the event of an outage. Active/standby mode provides an always on standby node in the event of an outage, and is also provides load balancing.Answer: BExplanation:Balanced mode: This mode provides redundant high availability with automatic user load balancing and user failover in the event that one nodes fails because of component failure or power outage.Active/standby mode: The standby node automatically takes over for the active node if the active node fails. It does not provide automatic load balancing.QUESTION 116Which two DNS records must be created to configure Service Discovery for on-premises Jabber? (Choose two.)A. _cuplogin._tcp.cisco com pointing to a record of IM&PB. _cisco-uds._tcp.cisco.com pointing to a record of Cisco Unified CMC. _cuplogin._tls.cisco.com pointing to the IP address of IM&PD. _cisco-uds._tls.cisco.com pointing to the IP address of Cisco Unified Communications managerE. _xmpp._tls.cisco.com pointing to a record of IM&PAnswer: ABExplanation: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/jabber/Windows/9_7/CJAB_BK_C606D8A9_00_cisco-jabber-dns-configuration-guide/CJAB_BK_C606D8A9_00_cisco-jabber-dns-configuration-guide_chapter_010.htmlQUESTION 117An engineer implements QoS in the enterprise network. Which command can to verify the correct classification and marking on a cisco IOS switch?A. show policy-mapB. show class-map interface GigabitEthernet 1/0/1C. show access-listsD. show policy-map interface GigabitEthernet 1/0/1Answer: DExplanation: https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/qos_classn/configuration/xe-16/qos-classn-xe-16-book/qos-classn-mrkg-ntwk-trfc-xe.htmlQUESTION 118Which field is configured to change the caller ID information on a SIP route pattern?A. Calling party Transformation MaskB. Route partitionC. Called party Transformation MaskD. Connected Line ID PresentationAnswer: A Explanation:Cisco Unified Communications Manager uses connected line ID presentation (COLP/COLR) as a supplementary service to allow or restrict the called party phone number on a call-by-call basis.Calling Party Transform Mask, Enter a transformation mask value. Valid entries include the digits 0 through 9; the wildcard characters X, asterisk (*), and octothorpe (#); and the international escape character +.QUESTION 119Which protocol does prime collaboration Assurance use to poll the health status of different systems in the collaboration environment?A. SIPB. SMTPC. SCCPD. SNMPAnswer: DExplanation: https://www.cisco.com/c/en/us/products/collateral/cloud-systems-management/prime-collaboration/guide-c07-736946.html#_Toc446633083QUESTION 120Refer to the exhibit. When a UC Administrator is troubleshooting DTMF negotiated by this SIP INVITE, which two messages should be examined next to further troubleshoot the issue? (Choose two.)

```
INVITE sip:1@10.10.10.219;user=phone SIP/2.0
Via: SIP/2.0/TCP 10.10.10.84:50083;branch=z9hG4bK471df613
From: "1234 - My Phone" <sip:1234@10.10.10.219>;tag=381c1aba7a78
To: <sip:1@10.10.10.219>
Call-ID: 381c1aba-7a78000d-4ca6894a-41dd3e0f@10.10.10.84
Max-Forwards: 70
CSeq: 101 INVITE
Contact: <sip:1234@10.10.10.84:50083;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDAT
Allow-Events: kpml,dialog
Content-Type: application/sdp
Content-Length: 658

v=0
o=Cisco-SIPUA 26529 0 IN IP4 10.10.10.84
s=SIP Call
b=AS:4064
t=0 0
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114
maxplaybackrate=16000;sprop-maxcapturerate=16000;maxaveragebitra
stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0,mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1,mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=sendrecv
```

A. PRACKB. UPDATEC. SUBSCRIBED. NOTIFYE. REGISTERAnswer: CDExplanation:The DTMF Events through SIP Signaling feature allows telephone event notifications to be sent through SIP NOTIFY messages, using the SIP

SUBSCRIBE/NOTIFY method as defined in the Internet Engineering Task Force (IETF) draft, SIP-Specific Event Notification.QUESTION 121An engineer wants to manually deploy a Cisco Webex DX80 video endpoint to a remote user. Which type of provisioning is configured on the endpoint?A. CUBEB. CMSC. CUCMD. EdgeAnswer: DExplanation:The provisioning modes/types are Off/Auto/CUCM/Edge/Spark/TMS/VCS.
https://www.cisco.com/c/dam/en/us/td/docs/telepresence/endpoint/ce95/dx70-dx80-administrator-guide-ce95.pdf#D1536209_DX70-DX80_Administrator_Guide_CE95.indd%3A.859768%3A307505QUESTION 122Which DiffServ marking is the most likely to drop packets?A. AF32B. AF12C. AF11D. AF13Answer: DExplanation:In the following example, packets in the AF13 class will be dropped before packets in the AF12 class, which in turn will be dropped before packets in the AF11 class:dP(AF13) >= dP(AF12) >= dP(AF11)
https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/qos_dfsrv/configuration/15-mt/qos-dfsrv-15-mt-book/qos-dfsrv.html#GUID-42E30ADD-2C7B-4273-B20D-ACE5182A9314QUESTION 123Endpoint A is attempting to call endpoint B. Endpoint A only supports G.711 ulaw with a packetization rate of 20 ms, and endpoint B supports packetization rate of 30 ms for G.711ulaw. Which two media are resources are allocated to normalize packetization rates through transrating?A. Hardware MTP on Cisco IOS SoftwareB. Software MTP on Cisco Unified Communication ManagerC. Software MTP on Cisco IOS SoftwareD. Software transcoder on Cisco unified Communications managerE. Hardware transcoder on Cisco IOS SoftwareAnswer: AEExplanation:Software MTP - Software-only implementation that does not use a DSP resource for endpoints using the same codec and the same packetization time. Hardware MTP - Hardware-only implementation that uses a DSP resource for endpoints using the same G.711 codec but a different packetization time. The repacketization requires a DSP resource so it cannot be done by software only. Cisco Unified Communications Manager also uses the term software MTP when referring to a hardware MTP.
https://www.cisco.com/c/en/us/td/docs/routers/access/vgd1t3/rel1_0/software/configuration/guide/VGD_transcoding.html
QUESTION 124When configuring Cisco Unified Communications Manager, which configuration enables phones to automatically register to a Cisco Unified Communications publisher when the connection to the subscriber is lost?A. Cisco Unified CM GroupB. Device PoolC. SRSTD. Route GroupAnswer: AExplanation:



About Cisco Unified Communications Manager Group Setup

In Cisco Unified Communications Manager Administration, use the **System > Cisco Unified CM Group** menu path to configure Cisco Unified Communications Manager groups.

A Cisco Unified Communications Manager Group specifies a prioritized list of up to three Cisco Unified Communications Managers. The first Cisco Unified Communications Manager in the list serves as the primary Cisco Unified Communications Manager for that group, and the other members of the group serve as secondary and tertiary (backup) Cisco Unified Communications Managers.

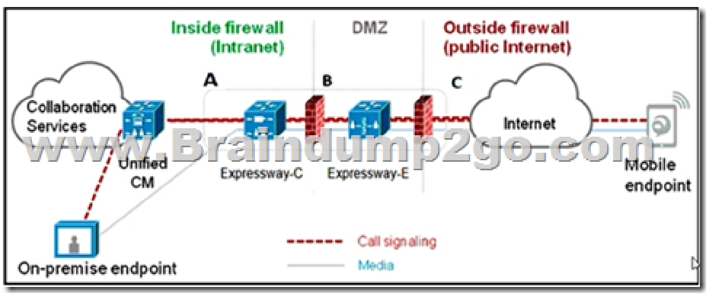
Each device pool has one Cisco Unified Communications Manager Group that is assigned to it. When a device registers, [illegible] the device tries to connect to the next Cisco Unified Communications Manager that is listed in the group, and so on.

Cisco Unified Communications Manager Groups provide important features for your system:

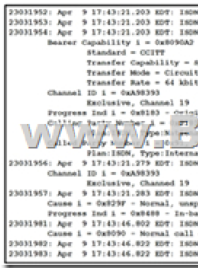
- Redundancy—This feature enables you to designate a primary and backup Cisco Unified Communications Managers for each group.
- Call processing load balancing—This feature enables you to distribute the control of devices across multiple Cisco Unified Communications Managers.

For most systems, you need to have multiple groups, and you need to assign a single Cisco Unified Communications Manager to multiple groups to achieve better load distribution and redundancy.

QUESTION 125Which two elements of a dial plan define the domains that are accessible and are assigned to an endpoint? (Choose two.)A. Call Admissions ControlB. Route patternsC. Calling Search SpacesD. Translation patternsE. partitionsAnswer: CEQUESTION 126An engineer with ID378163512 is designing a new dial plan for a customer that has offices in several countries on four continents around the world. This client also want to integrate with a Microsoft Lync backend, Which dial plan type does the engineer recommend?A. SIP URIB. TEHQC. H.323D. E.164Answer: DExplanation:E.164 is the international numbering plan meant to ensure each number is globally unique.QUESTION 127Which version is used to provide encryption for SNMP management traffic in collaboration deployments?A. SNMPv1B. SNMPv3C. SNMPv2D. SNMPv2cAnswer: BExplanation:
https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/snmp/configuration/xe-16/snmp-xe-16-book/nm-snmp-encrypt-snmp-support.htmlQUESTION 128Which field of a Real-Time Transport Protocol packet allows receiving devices to detect lost packets?A. CSRC (Contributing Source ID)B. TimestampC. Sequence numberD. SSRC (Synchronization identifier)Answer: CQUESTION 129
Refer to the exhibit. When making a call to a MRA client, what are the combinations of protocol on each of the different sections A-B-C?



A. IP TCP/TLS(A) +SIP TCP/TLS (B) +TLS (C)B. SIP TLS (A) +SIP TLS (B) +SIP TLS (C)C. SIP TCP/TLS (A) + SIP TLS (B) + SIP TLS (C)D. SIP TCP/TLS +SIP TCP/TLS (B) + SIP TCP/TLS (C)Answer: CExplanation: https://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/expressway/config_guide/X12-5/Cisco-Expressway-IP-Port-Usage-for-Firewall-Traversal-Deployment-Guide-X12-5.pdfQUESTION 130An engineer is designing a load balancing solution for two Cisco Unified Border routers. The first router (cube1.abc.com) takes 60% of the calls and the second router (cute2.abc.com) takes 40% of the calls, Assume all DNS A records have been created. Which two SRV records are needed for a load balanced solution? (Choose two.)A. sip_udp.abc.com 60 IN SRV 1 40 5060 cube2.abc.comB. sip_udp.abc.com 60 IN SRV 3 60 5060 cube2.abc.comC. sip_udp.abc.com 60 IN SRV 60 1 5060 cube1.abc.comD. sip_udp.abc.com 60 IN SRV 2 60 5060 cube1.abc.comE. sip_udp.abc.com 60 IN SRV 1 60 5060 cube1.abc.comAnswer: AEExplanation:- Service: The symbolic name of the desired service - Proto: The transport protocol of the desired service; this is usually either Transmission Control Protocol (TCP) or User Datagram Protocol (UDP)- Name: The domain name for which this record is valid, it ends in a dot- TTL: Standard DNS time to live field- Class: Standard DNS class field (this is always IN)- Priority: The priority of the target host, lower value means more preferred- Weight: A relative weight for records with the same priority, higher value means more preferred- Port: The TCP or UDP port on which the service is to be found- Target: The canonical hostname of the machine that provides the service, it ends in a dotQUESTION 131Refer to the exhibit. A call to an international number has failed. Which action corrects this problem?



A. Assign a transcoder to the MRGL of the gateway.B. Strip the leading 011 from the called party numberC. Add the bearer-cap speech command to the voice port.D. Add the isdn switch-type primart-dms100 command to the serial interface.Answer: BExplanation:Bearer-cap 0x8090A2 is already speech. Dialed nr starting 011 seems to me not an international number.https://www.cisco.com/c/en/us/support/docs/voice/h323/14006-h323-isdn-callfailure.htmlQUESTION 132Which user group is targeted by MRA services?A. Call center agents who dial out to remote customersB. Mobile workers in a host desk environment at HQ who log in every morning at possibility a different phoneC. Production floor users who need wireless mobility in remote areas of the factoryD. On-the-go mobile workforce who connect to corporate phone services using their own mobile deviceAnswer: DQUESTION 133Which Cisco Unified Communications Manager feature is to determine the maximum bit rate for a call between two video-endpoints?A. PartitionsB. LocationsC. RegionsD. transformationsAnswer: CExplanation:Regions provide capacity controls for Unified Communications Manager multi-site deployments where you may need to limit the bandwidth for certain calls. For example, you can use regions to limit the bandwidth for calls that are sent across a WAN link, while maintaining a higher bandwidth for internal calls. You can use regions to limit the bandwidth for audio and video calls by setting the maximum bitrate for intraregional or interregional calls to whatever the region(s) can provide.Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/11_5_1_SU1/cucm_b_system-configuration-guide-1151su1/cucm_b_system-configuration-guide-1151su1_chapter_0111.htmlQUESTION 134When a 8800 series phone is registered over MRA, where does it register?A. Cisco Unified Communications ManagerB. Expressway-CC. Cisco Unified Presence ServerD. Expressway-EAnswer: AQUESTION 135An engineer must manually provision a Cisco IP Phone 8845 using SIP. Which two fields must be configured for a successful provision? (Choose two.)A. Media resources group listB. SIP profile C. CSSD. LocationE. Device security profileAnswer: BEQUESTION 136What is the correct statement about CUCM and Cisco IM&P backups?A. Backups should the scheduled during hours to avoid system performance issues.B. Backups are saved as .tar files and encrypted using the web administrator account.C. Backups are saved as unencrypted tar files.D. Backups are not needed for subscriber Cisco Unified Communication Manager and Cisco IM and Presence servers.Answer: AQUESTION 137Due to service provider restriction, Cisco Unified Communications Manager cannot send video in the SDP. Which two options on Cisco Unified CM are configured to suppress video in the SDP is outgoing invites? (Choose two.)A. Check the Send send-receive SDP in mid-call INVITE check box on the SIP trunk SIP profile.B. Add the audio forced command to voice service voip on the Unified Border Elements.C. Check the Retry Call as Audio on the SIP trunkD. Set Video bandwidth in the Region settings to 0.E. Change the Video Capabilities dropdown on the endpoint to Disabled.Answer: CDExplanation:The reason is it has to be only in

CUCM per the question. You would create a specific Region relationship for the Device Pool applied to the trunk that cannot handle video and thus change the Maximum Video Bandwidth to 0 while setting retry a video call as audio on the Phone Configuration (default setting is this is already checked).QUESTION 138There is a saturated link that has traffic shaping configured. How is incoming traffic processed?A. Excess traffic is queued for later transmission.B. Excess traffic is dropped.C. Traffic is compressed so that the traffic fits within the bandwidth of the link.D. Excess traffic is queued, and then dropped after the timer expires.Answer: AExplanation: 

Policing Versus Shaping

shaping is a smoothed packet output rate.

https://www.cisco.com/c/en/us/support/docs/quality-of-service-qos/qos-policing/19645-policevsshape.html#trafficshaping
QUESTION 139Refer to the exhibit. A customer submits this output, captured on a Cisco IOS router. Assuming that an MGCP gateway is configured with a ISDN BRI interface, which BRI changes resolve the issue? 

```
05:50:14.102: ISDN BR0/1/1 Q921: User TX -> IDREQ ri
05:50:14.165: ISDN BR0/1/1 Q921 User RX <- SABMEp sa
```

A. interface BRI0/1/1no ip addressisdn switch-type basic-net3isdn incoming-voice voiceisdn send-alertingisdn static-tei 0B. interface BRI0/1/0no ip addressisdn switch-type basic-net3isdn point-to-multipoint-setupisdn incoming-voice voiceisdn send-alertingisdn static-tei 0C. interface BRI0/1/1no ip addressisdn switch-type basic-net3isdn point-to-point-setupisdn incoming-voice voiceisdn send-alertingisdn static-tei 0D. interface BRI0/1/1no ip addressisdn switch-type basic-net3isdn point-to-multipoint-setupisdn incoming-voice voiceisdn send-alertingisdn static-tei 0Answer: CQUESTION 140Which SNMP service must be activated manually on the Cisco Unified Communications Manager after installation?A. Cisco CallManager SNMPB. SNMP Master AgentC. Connection SNMP AgentD. Host Resources AgentAnswer: AExplanation:SNMP Master Agent serves as the primary service for the MIB interface. You must manually activate Cisco CallManager SNMP service; all other SNMP services should be running after installation.Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/service/9_0/admin/CUCM_BK_C136FE37_00_cisco-unified-serviceability-administration-90/CUCM_BK_C136FE37_00_cisco-unified-serviceability-administration-guide_chapter_0101.html
QUESTION 141Refer to the exhibit. An engineer verifies the configured of an MGCP gateway. The commands are already configured.Which command is necessary to enable MGCP? 

```
hostname GATEWAY
ccm-manager config

mgcp call-agent CCMSub1.domain.com 2427 service-type mgcp version 0.1
```

A. Device(config)# mgcp enableB. Device(config)# ccm-manager enableC. Device (config)# com-manager activeD. Device (config)# mgcpAnswer: DExplanation:
https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-callmanager/42105-vg200-cfg.htmlQUESTION 142What are two key features of the Expressway series? (Choose two.)A. VPN connect toward the internal UC resourcesB. B2B callsC. IP to PSTN call connectivityD. Device registration over the internetE. SIP header modificationAnswer: BDExplanation:
https://www.cisco.com/c/en/us/products/collateral/unified-communications/expressway-series/datasheet-c78-737605.html
QUESTION 143An engineer deploys a Cisco Expressway edge server for a customer who wants to utilize all feature on the server. Which feature does the engineer configure on the Expressway edge?A. VTC bridgeB. H.323 endpoint registrationC. SIP gateway for PSTN providersD. MRAAnswer: DQUESTION 144An engineer is configuring a Cisco Unified Border Element to allow the video endpoints to negotiate without the Cisco Unified Border Element interfering in the process. What should the engineer configure on the Cisco Unified Border Element to support this process?A. Configure path-thru content sdp on the voice service.B. Configure a hardcoded codec on the dial peers.C. Configure a transcoder for video protocols.D. Configure codec transparent on the dial peers.Answer: AExplanation:"You can use 'pass-thru content sdp', if you do not want to involve CUBE in the codec negotiation."https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/cube-codec-basic.html
Resources From:1.2023 Latest Braindump2go 350-801 Exam Dumps (PDF & VCE) Free Share:
https://www.braindump2go.com/350-801.html2.2023 Latest Braindump2go 350-801 PDF and 350-801 VCE Dumps Free Share:
https://drive.google.com/drive/folders/1pALdeSmYT0960h3st7rEH4WExsy2Mjgq?usp=sharingFree Resources from