Cisco.350-801.v2026-09-16.q298

Exam Code:	350-801
Exam Name:	Implementing and Operating Cisco Collaboration Core Technologies
Certification Provider:	Cisco
Free Question Number:	298
Version:	v2026-09-16
# of views:	131
# of Questions views:	2980
https://www.freepdfdumps.com/Cisco.350-801.v2026-09-16.q298.html	

NEW QUESTION: 1

Due to provider requirements, outgoing calls from the Enterprise to the PSTN must start with channel 1. Which ISDN command changes the channel selection on IOS to meet this requirement?

A. isdn bchan-number-order ascending

B. isdn incoming-voice

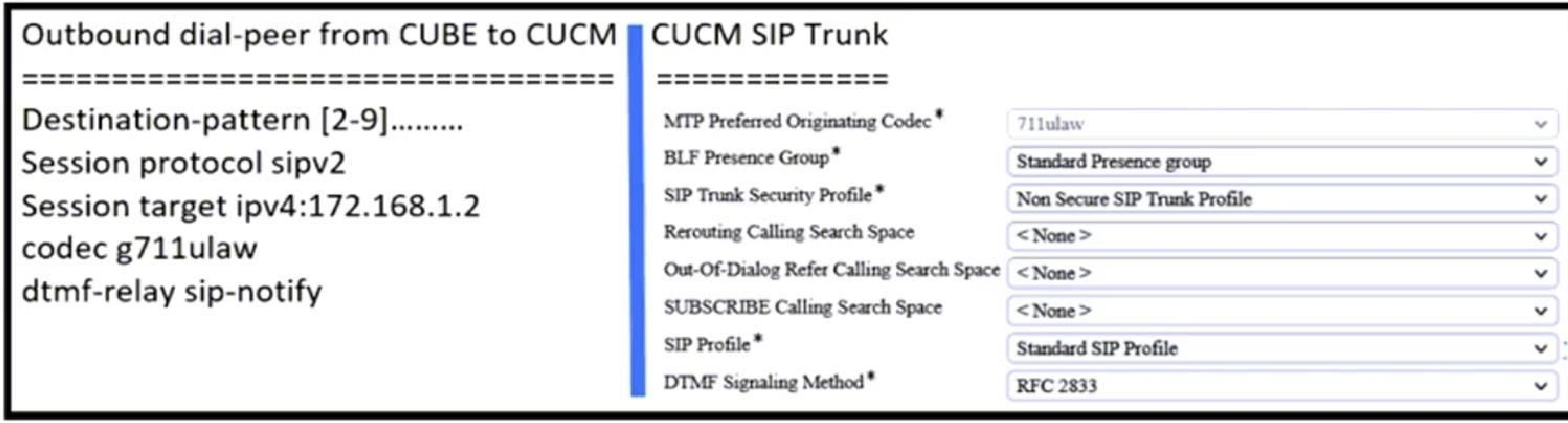
C. isdn protocol-emulate network

D. isdn bchan-number-order decending

Answer: A (LEAVE A REPLY)

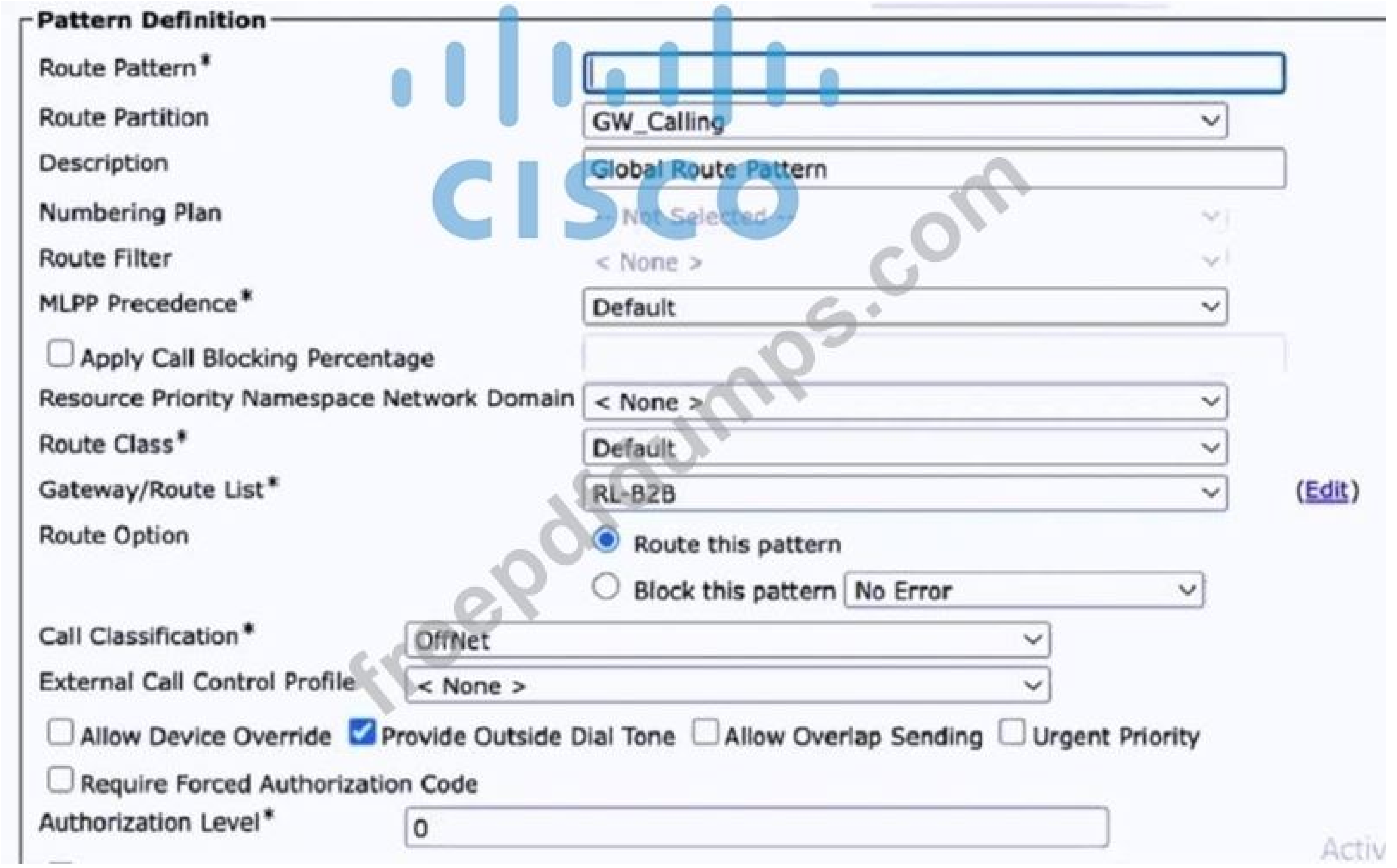
NEW QUESTION: 2

Refer to the exhibit. A customer reports that MTP resources are being used in the recently deployed Cisco UCM and wants no MTP resources to be used. Cisco UCM SIP trunk configuration does not have the MTP Required box checked. Which configuration change resolves the issue?



A. Change Cisco Unified Border Element configuration to "dtmf-relay rtp-nte".
B. Change Cisco Unified Border Element configuration to "dtmf-relay sip-notify sip-kpml".
C. Change Cisco UCM SIP trunk DTMF Signaling Method to "sip-notify".
D. Change Cisco UCM SIP trunk DTMF Signaling Method to "sip-kpml".
Answer: A (LEAVE A REPLY)
In this scenario, the Media Termination Point (MTP) is being used despite the "MTP Required" option not being checked on the SIP trunk. One common reason for this is a mismatch in the DTMF relay method between Cisco Unified Border Element (CUBE) and Cisco Unified Communications Manager (CUCM).
The current CUBE configuration uses dtmf-relay sip-notify, while CUCM is set to use RFC 2833, which is equivalent to rtp-nteon CUBE. These two methods are incompatible, leading to MTP invocation to bridge the mismatch.
To resolve this, you should configure CUBE to use dtmf-relay rtp-nte, ensuring that both sides use RFC 2833 for DTMF signaling, preventing unnecessary MTP usage.

NEW QUESTION: 3
Refer to the exhibit. Which route pattern can the Cisco UCM administrator apply to the configuration to allow calls to "1306, 1316, 1326, 13*6, 13#6" only?



A. 13[^3-9]6
B. 13XX
C. 13[25-8]6
D. 13!#
Answer: A (LEAVE A REPLY)

NEW QUESTION: 4
Which statement about Cisco Unified Communications Manager and Cisco IM and Presence backups is true?
A. Backups should be scheduled during off-peak 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 Communications Manager and Cisco IM and Presence servers.
Answer: (SHOW ANSWER)
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1_SU1/Administration/ cucm_b_administration-guide-1151su1/cucm_b_administration-guide-1151su1_chapter_01010.html#CUCM_TK_S7FC26D5_00

NEW QUESTION: 5
Which characteristic of distributed class-based weighted fair queueing addresses jitter prevention?
A. It provides additional granularity by allowing a user to create classes
B. It minimizes jitter by implementing a priority queue for voice traffic

C. It uses a priority queue for voice traffic to avoid jitter
D. It provides additional granularity by allowing a user to define custom classes
Answer: A (LEAVE A REPLY)
https://www.cisco.com/en/US/docs/ios/12_0t/12_0t5/feature/guide/cbwfq.html

NEW QUESTION: 6
A company deploys centralized cisco ucm architecture for a hub location and two remote sites.
- The company has only one ITSP connection at the hub connection, and
ITSP supports only G.711 calls
- Remote site A has a 1-Gbps fiber connection to the hub connection and calls to and from remote side A use G.711 codec
- Remote site B has a 1 T1 connection to the hub location and calls to
and from remote site B use G.729 codec
Based on the provided guidance, a Cisco voice engineer must design media resource management for the customer. What is the method that needs to be followed?
A. configure the hardware transcoder on the site B router.
B. configure the hardware transcoder on the site A router.
C. configure the hardware transcoder on the hub location router.
D. configure the software transcoder on Cisco UCM to support voice calls to and from both remote sites.
Answer: (SHOW ANSWER)
Site B has a T1 connection and we don't want to force them to limit the number of calls. As an administrator, i would take the incoming calls from site B in G729 and do a transcode at the hub location, and receive an Incoming call to Site B in G 711 and transcode it to G729 and traverse over T1.

NEW QUESTION: 7
How is bandwidth allocated to traffic flows in a flow-based WFQ solution?
A. All the bandwidth is divided based on the QoS marking of the packets.
B. Bandwidth is divided among traffic flows. Voice has priority.
C. Voice has priority and the other types of traffic share the remaining bandwidth
D. Each type of traffic flow has equal bandwidth.
Answer: D (LEAVE A REPLY)

NEW QUESTION: 8
An end used at a remote site is trying initiate an ad hoc conference call to an end user at the main site the conference bridge is configured to support G.711 remote user phone only support
G.729 the remote end user receives an error message on the phone "cannot complete conference call, what is the cause of the issue?
A. A software conference bridge is not assigned
B. The remote phone does not have the conference feature assigned.
C. The transcoder resource is missing.
D. A Media Termination Point is missing
Answer: (SHOW ANSWER)

NEW QUESTION: 9
What is the maximum one-way packet loss and jitter for voice before quality issues arise?
A. < 10 percent packet loss and < 10 ms jitter

B. < 1 percent packet loss and < 30 ms jitter
C. < 3 percent packet loss and < 60 ms jitter
D. < 9 percent packet loss and < 30 ms jitter
Answer: B (LEAVE A REPLY)
Voice calls, either one-to-one or on a conference connection capability, require the following: ≤
150 ms of one-way latency from mouth to ear (per the ITU G.114 standard) / ≤ 30 ms jitter / ≤ 1 percent packet loss.

NEW QUESTION: 10
What must be configured on a Cisco Unity Connection voice mailbox to access the mailbox from a secondary device?
A. mobile user
B. alternate names
C. alternate extensions
D. Attempt Forward routing rule
Answer: C (LEAVE A REPLY)
To access a Cisco Unity Connection voice mailbox from a secondary device, you must configure an alternate extension for the mailbox. This is a phone number that is different from the mailbox's primary extension. When you call the alternate extension, you will be prompted to enter the mailbox's PIN. Once you have entered the PIN, you will be able to access the mailbox just as you would if you were calling from the primary device.

NEW QUESTION: 11
An engineer is configuring a Cisco IOS XE gateway to join three callers together in the same conversation. Which IOS XE media resource must the engineer configure to accomplish this goal?
A. IOS XE Hardware Transcoder
B. IOS XE Hardware Conference Bridge
C. IOS XE Hardware MTP
D. IOS XE Software MTP
Answer: B (LEAVE A REPLY)
To join three or more callers in the same conversation, a conference bridge is required. A conference bridge is a media resource that mixes multiple audio streams into a single stream, allowing multiple participants to communicate simultaneously.
A hardware-based conference bridge uses DSP (Digital Signal Processor) resources to mix audio streams. It is highly scalable and provides better performance compared to software-based solutions, making it ideal for handling multiple participants or high-quality calls.
Hardware conference bridges are typically required for real-time audio mixing in larger environments, as they offload the processing to dedicated DSPs on the gateway.
Reference: https://community.cisco.com/t5/collaboration-knowledge-base/basic-hardware-base- ios-media-resources-configuration-and/ta-p/3164231

NEW QUESTION: 12
Which Cisco Unified communications manager configuration is required for SIP MWI integration?
A. Select "Redirecting Diversion Header Delivery - Inbound" on the SIP trunk.
B. Enable "Accept presence subscription" on the SIP trunk security profile.
C. Select "Redirecting Diversion Header Delivery - outbound" on the SIP trunk.
D. Enable "Accept unsolicited notification" on the SIP Trunk security profile.
Answer: D (LEAVE A REPLY)
https://community.cisco.com/t5/collaboration-voice-and-video/understanding-troubleshooting- mwi-on-unity-connection/ta-p/3162948

NEW QUESTION: 13
Which two steps are required for bulk configuration transactions on the Cisco UCM database utilizing BAT? (Choose two)
A. A data file in comma-separated values format must be uploaded to Cisco UCM
B. A server template must be created in Cisco UCM.
C. A device template must be created in Cisco UCM.
D. A data file in Abstract Syntax Notation One format must be uploaded to Cisco UCM
E. A data file in Extensible Markup Language format must be uploaded to Cisco UCM
Answer: A,C (LEAVE A REPLY)
https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/bat/9_0_1/CUCM_BK_C22BD805_00_ cucm-bulk-administration-guide_chapter_01.pdf

NEW QUESTION: 14
Which type of dial plan implements destinations by using a format that is similar to email addresses?
A. URI
B. E.164
C. H.323
D. DNS
Answer: A (LEAVE A REPLY)

NEW QUESTION: 15
Endpoint 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 Software
B. Software MTP on Cisco Unified Communication Manager
C. Software MTP on Cisco IOS Software
D. Software transcoder on Cisco unified Communications manager
E. Hardware transcoder on Cisco IOS Software
Answer: A,E (LEAVE A REPLY)
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

NEW QUESTION: 16
Which two approaches are taken to implement CAC within Cisco UCM in full-mesh topology?
(Choose two.)
A. device pool
B. RSVP-enabled locations
C. location-based CAC
D. Audio Codec Preference List
E. physical location
Answer: B,C (LEAVE A REPLY)

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmsys/CUCM_BK_ SE5FCFB6_00_cucm-system-guide-100/CUCM_BK_SE5FCFB6_00_cucm-system-guide-100_chapter_01000.html

NEW QUESTION: 17

A company wants to block all users from forwarding calls to international destinations. An engineer must configure a calling search space named NATIONAL across all phones for call forwarding by using the Bulk Administration Tool. Which action must the engineer take from Bulk Administration in Cisco UCM Administration?

A. From Phones, select Update Phones, and then click Update Lines.

B. From Lines, click Update Lines.

C. From Phones, select Update Phones, and then click Query.

D. From Phones, select Add/Update Lines, and then click Update Lines.

Answer: D (LEAVE A REPLY)

The Bulk Administration Tool allows bulk editing of phone line settings.

Call forwarding calling search space is a line-level setting, so you update lines, not just phones.

The "Add/Update Lines" option lets you modify existing line configurations, including the calling search space for call forwarding.

After selecting "Add/Update Lines," you use "Update Lines" to apply the changes.

NEW QUESTION: 18

Refer to the exhibit. A collaboration engineer needs to replace the original, single NTP server that was configured during the initial install of a Cisco UCM server.

What is the first step to accomplish this task?

```
admin:utils ntp status
ntpd (pid 17428) is running...

     remote           refid        st t when poll reach   delay   offset  jitter
==============================================================================
*192.168.1.1     17.253.14.125     2 u  39   64     3    0.456   -0.236   0.116
*192.168.1.2     17.253.14.125     2 u  38   64     3    0.817   -0.695   0.395
```

A. Stop the NTP service on Cisco UCM.

B. Restart the NTP service on Cisco UCM.

C. Delete the original NTP server from Cisco UCM

D. Enable NTP authentication for the new NTP server on Cisco UCM.

Answer: C (LEAVE A REPLY)

NEW QUESTION: 19

An engineer must configure a dial plan tor a new Cisco UCM cluster. Standard local route groups must be used for all local calls. After the route pattern, route list and route groups are configured, which settings must the engineer configure for the standard local route groups?

A. device pool

B. voice gateway

C. trunk group
D. SIP trunk
Answer: A (LEAVE A REPLY)

NEW QUESTION: 20
An engineer is designing a high availability and failover solution for two Cisco Unified Border Element routers. The first router (cube 1.abc.com) takes 60% of the calls and the second router (cube2 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 2 60 cube1.abc.com
B. _sip_udp.abc.com 60 IN SRV 3 60 cube2.abc.com
C. _sip_udp.abc.com 60 IN SRV 60 1 cube1abc.com
D. _sip_udp.abc.com 60 IN SRV 1 40 cube2.abc.com
E. _sip_udp.abc.com 60 IN SRV 1 60 cube1.abc.com
Answer: D,E (LEAVE A REPLY)

NEW QUESTION: 21
An 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-map
B. show class-map interface GigabitEthernet 1/0/1
C. show access-lists
D. show policy-map interface GigabitEthernet 1/0/1
Answer: D (LEAVE A REPLY)
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.html

NEW QUESTION: 22
What are two common attributes of XMPP XML stanzas? (Choose two.)
A. destination
B. Source
C. to
D. version
E. from
Answer: C,E (LEAVE A REPLY)

NEW QUESTION: 23
Which QoS marking is used when an administrator configures voice call signaling?
A. CS4
B. CS3
C. AF41
D. EF
Answer: B (LEAVE A REPLY)

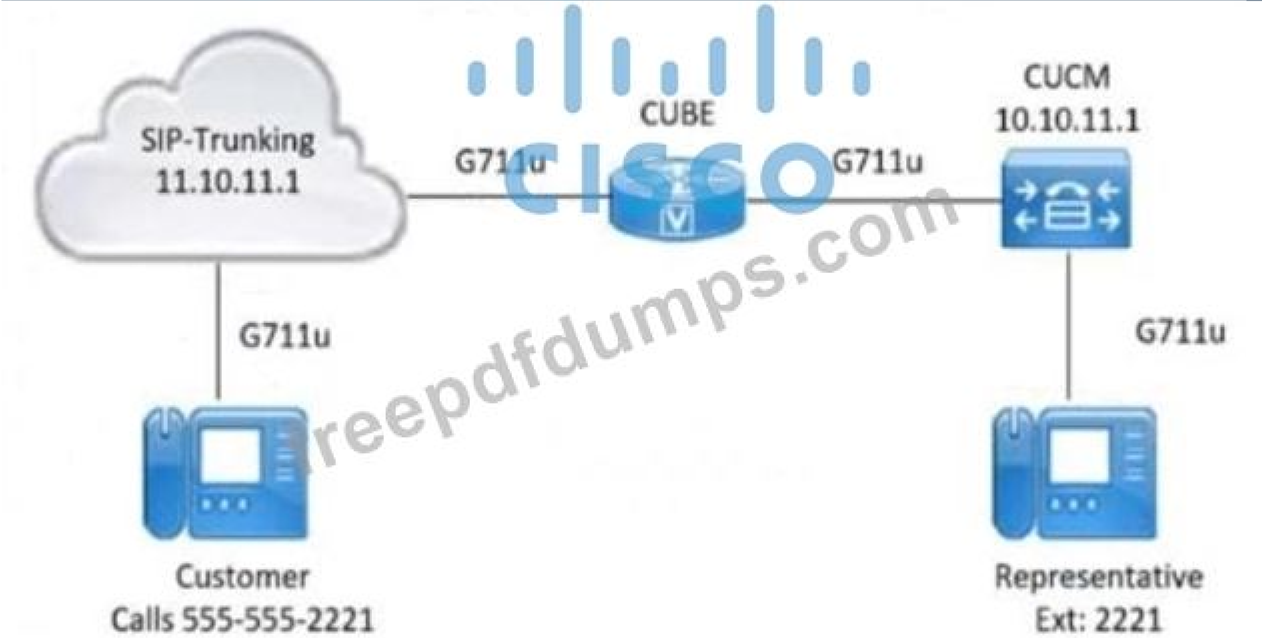
NEW QUESTION: 24

Refer to the exhibit. An engineer deploys this Cisco Unified Border Element configuration, but the phone at extension 2221 fails to ring. These configurations were performed already in Cisco UCM:

- phone configuration and registration
- SIP trunk configuration
- translation pattern configuration

Which action resolves the issue?

```
CUBE(config)# <output omitted>
CUBE(config)# voice translation-rule 3
CUBE(config-translation-rule)# rule 1 /^555555\(....\)/ /\1/
CUBE(config)# voice translation-profile profile1
CUBE(cfg-translation-profile)# translate called 3
CUBE(cfg-translation-profile)# exit
CUBE(config)# dial-peer voice 10 voip
CUBE(config-dial-peer)# destination-pattern ...$
CUBE(config-dial-peer)# session target ipv4:10.10.11.1
CUBE(config-dial-peer)# session protocol sipv2
CUBE(config-dial-peer)# dtmf-relay rtp-nte
CUBE(config-dial-peer)# codec G711u
CUBE(config-dial-peer)# no vad
CUBE(config-dial-peer)# end
CUBE(config)# dial-peer voice 11 voip
CUBE(config-dial-peer)# incoming called-number .
CUBE(config-dial-peer)# session target ipv4:11.10.11.1
CUBE(config-dial-peer)# session protocol sipv2
CUBE(config-dial-peer)# dtmf-relay rtp-nte
CUBE(config-dial-peer)# codec G711u
CUBE(config-dial-peer)# no vad
CUBE(config-dial-peer)# end
CUBE(config)# <output omitted>
```



A. Change the destination pattern for dial-peer voice 10 to 2... $.

B. Replace voice translation-rule 3 with voice translation-rule 1.

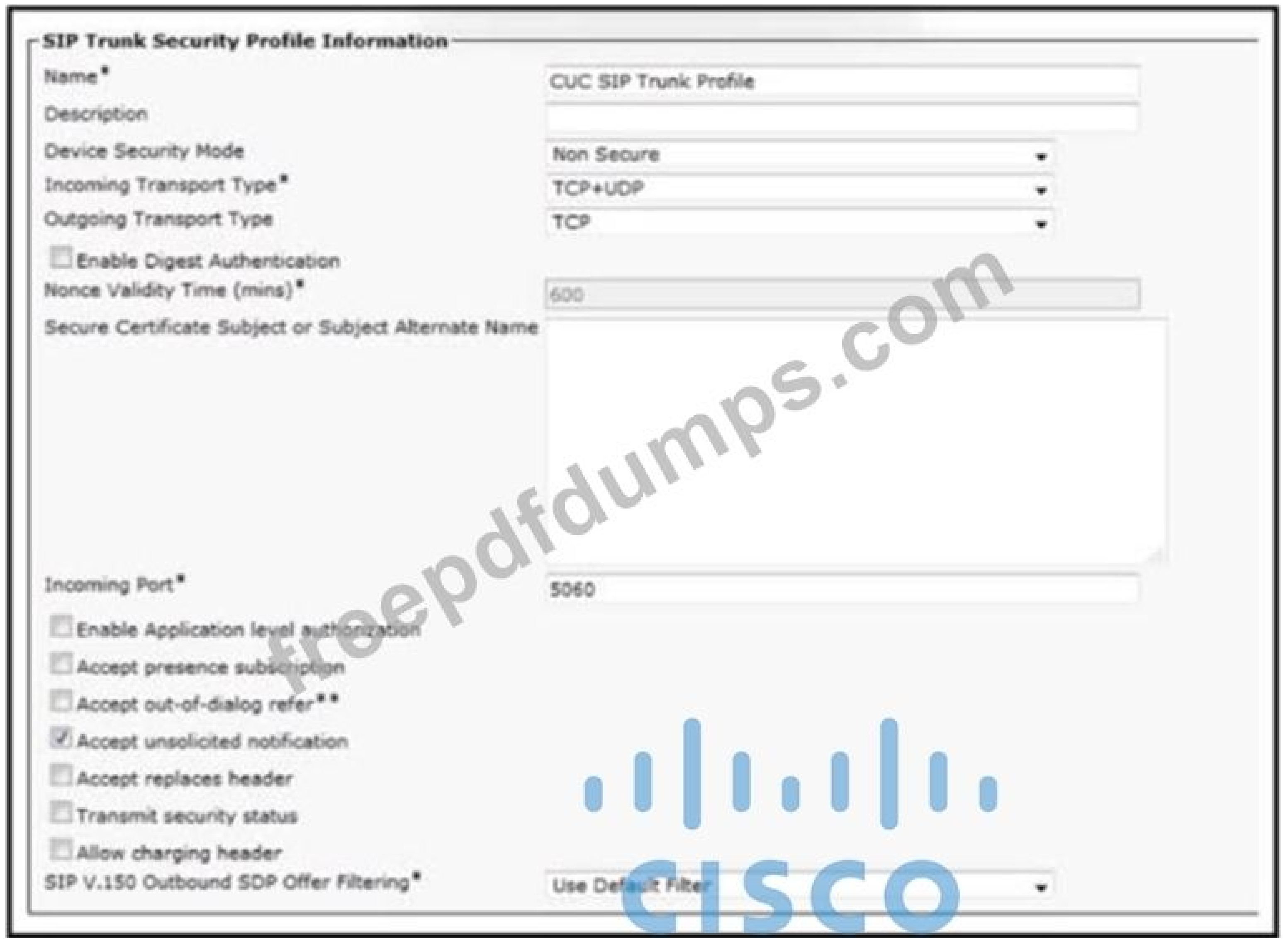
C. Apply voice translation-rule 3 to dial-peer 10 voip.
D. Apply voice translation-rule 3 to dial-peer 11 voip.
Answer: C (LEAVE A REPLY)

NEW QUESTION: 25
When designing the capacity for a Cisco UCM 12.x cluster, an engineer must decide which VMware template will be used for each node. What is the lowest number of users supported in a template and the highest number of users in a template?
A. 500 and 10.000 users
B. 1000 and 10.000 users
C. 750 and 15.000 users
D. 750 and 10.000 users
Answer: (SHOW ANSWER)

NEW QUESTION: 26
Which wildcard must an engineer configure to match a whole domain in SIP route patterns?
A. *
B. @
C. .
D. !
Answer: (SHOW ANSWER)
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_ C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0100111.html

NEW QUESTION: 27
A network administrator with ID0123456789 has determined that a WAN link between two Cisco UCM clusters supports only 1 Mbps of bandwidth for voice traffic.
How many calls does this link support if G.711 as the audio codec is used?
A. 15
B. 13
C. 12
D. 16
Answer: C (LEAVE A REPLY)

NEW QUESTION: 28
Refer to the exhibit. An engineer is integrating Cisco UCM with Cisco Unity Connection via SIP.
Which two boxes should be checked to complete the configuration of the SIP trunk security profile? (Choose two.)



A. to accept presence subscription

B. to accept replaces header

C. to enable digest authentication

D. to enable application-level authorization

E. to accept out-of-dialog refer

Answer: C,D (LEAVE A REPLY)

NEW QUESTION: 29

Why does Cisco UCM use DNS?

A. It provides certificate-based security for media.

B. It provides SRV resolution to the endpoints registered.

C. It resolves FQDN to IP address resolution for trunks.

D. It connects endpoints to single sign-on services

Answer: C (LEAVE A REPLY)

NEW QUESTION: 30

Refer to the exhibit. Which Codec is negotiated?

```
Received:
SIP/2.0 200 OK
Via: SIP/2.0/TCP
10.10.10.2:5060;branch=a8bH5bK7954A198F
From:
<sip:012345678@10.10.10.2>;tag=8D79AF62-DB2
To: <sip:90123456@10.10.4.14>;
tag=811681~ffa80926-5fac-4cc5-b802-
2dbde74ae7w2-
v=0
o=CiscoSystemsCCM-SIP 811681 1 IN IP4
10.10.4.14
s=SIP Call
c=IN IP4 10.5.4.3
t=0 0
m=audio 27839 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```


A. iLBC
B. G.729
C. G.728
D. G.711ulaw
Answer: D (LEAVE A REPLY)

NEW QUESTION: 31
An engineer must configure an MGCP gateway and register it to Cisco Unified Communications Manager. Which prerequisite must be met before applying the gateway commands to enable MGCP?
A. The MGCP gateway and the Cisco Unified CM must be able to communicate over 2427,2428, and 2727.
B. The MGCP gateway must have voice service VoIP configured.
C. Cisco Unified CM and the MGCP gateway must utilize the SIP OPTIONS ping feature to monitor status.

D. The MGCP gateway and the Cisco Unified CM must be able to communicate over ports 5060 and 5061.
Answer: (SHOW ANSWER)
https://www.cisco.com/c/en/us/td/docs/ios/voice/cminterop/configuration/guide/12_4t/vc_12_4t_b ook/vc_ucm_mgcp_gw.html

NEW QUESTION: 32
Refer the exhibit. Given this "debug isdn q921" output, what is the problem with the PRI?

```
000193: Dec 5 14:35:31.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:32.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:33.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:34.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
```

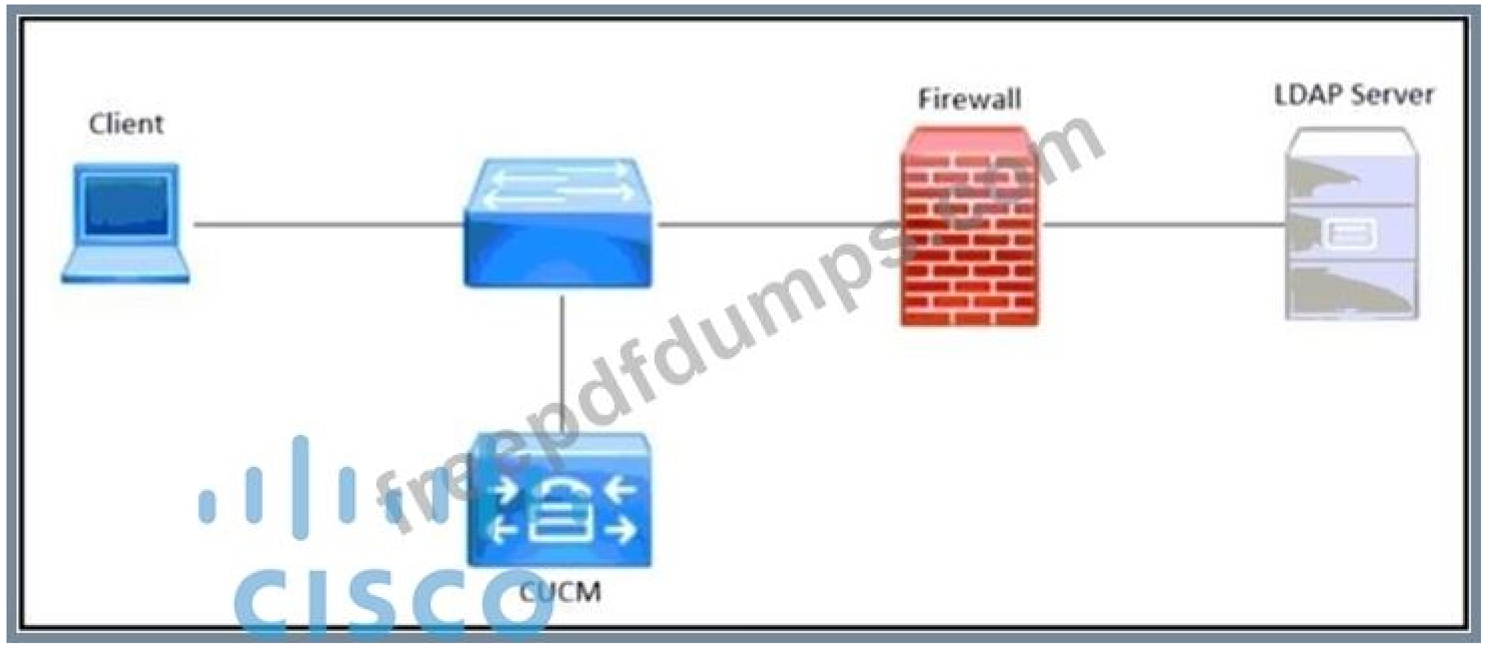
A. Layer 1 is down on the controller.
B. PRI does not have an IP address configured on the interface.
C. Noting, the PRI is sending keepalives.
D. Layer 2 is down on the controller.
Answer: (SHOW ANSWER)
If Layer 2 is stable, the router and switch must begin to synchronize with each other. The Set Asynchronous Balanced Mode Extended (SABME) message appears on the display. This message indicates that Layer 2 tries to initialize with the other side. Either side can send the message and try to initialize with the other side. If the router receives the SABME message, it must send back an Unnumbered Acknowledge frame (UAf).
https://www.cisco.com/c/en/us/support/docs/wan/t1-e1-t3-e3/8131-T1-pri.html

NEW QUESTION: 33
Which description of the function of call handlers in Cisco Unity Connection is true?
A. They control outgoing calls by allowing you to specify the numbers that Cisco Unity Connection can dial to transfer calls, notify users of massages, and delivery faxes.
B. They collect information from callers by playing a series of questions and recording the answers.
C. They answer calls, take message, and provide menus of options.
D. They provide access to a corporate directory by playing an audio list that users and outside callers users and leave messages.
Answer: (SHOW ANSWER)
Call handlers are used to answer calls, greet callers with recorded prompts and provide them with information and options, route calls, and take messages. You can use the predefined Unity Connection call handlers, or can create unlimited new call handlers.
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/10x/administration/guide/10xcucsagx/10xcucsag080.html

NEW QUESTION: 34
What 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: (SHOW ANSWER)
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.

NEW QUESTION: 35
Refer to the exhibit. An engineer must configure Cisco UCM to sync with an LDAP server. After the LDAP configuration is complete, the authentication times out before Cisco UCM logs in.
Which port must be open on the firewall to resolve the issue?



A. 389
B. 161
C. 3389
D. 123
Answer: A (LEAVE A REPLY)

NEW QUESTION: 36
Which action is required for a firewall configuration on a Mobile and Remote Access through Cisco Expressway deployment?
A. The traversal zone on Expressway-C points to Expressway-E through the peer address field on the traversal zone, which specifies the Expressway-E server address. For dual NIC deployments, set the Expressway-E address using an FQDN that resolves the IP address of the internal interface
B. The internal firewall must allow these inbound and outbound connections between Expressway-C and Expressway-E. SIP: HTTPS (tunnelled overSSH between C and E): TCP 2222; TCP 7001; Traversal Media: UDP 2776 to 2777 (or 36000 to 36011 for large VM/appliance): XMPP: TCP
7400.

C. The external firewall must allow these inbound connections to Expressway: SIP: TCP 5061; HTTPS: TCP 8443: XMPP: TCP 5222; Media: UDP 36002 to 59999.
D. Do not use a shared address for Expressway-E and Expressway-C, as the firewall cannot distinguish between them If static NAT for IP addressing on Expressway-E is used, ensure that any NAT operation on Expressway-C does not resolve the same traffic IP address Shared NAT is not supported.
Answer: C (LEAVE A REPLY)
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/expressway/config_guide/X12-5/exwy_b_mra-expressway-deployment-guide/exwy_b_mra-expressway-deployment-guide_chapter_0100.html#reference_E0C70BF89ECFBCED9E86107C6DB013D4

NEW QUESTION: 37
When designing location-based CAC, how much overprovisioning is calculated for video bandwidth?
A. 15%
B. 25%
C. 10%
D. 20%
Answer: D (LEAVE A REPLY)

NEW QUESTION: 38
Refer to the exhibit. An engineer runs the utils diagnose test CLI command on a Cisco UCM subscriber and receives an error message during the validate_network test. What are two causes of the error? (Choose two.)

```
admin:utils diagnose test

Log file: platform/log/diag3.log

Starting diagnostic test(s)
===========================
test - disk_space          : Passed (available: 15614 MB, used: 10126 MB)
skip - disk_files          : This module must be run directly and off hours
test - service_manager     : Passed
test - tomcat              : Passed
test - tomcat_deadlocks    : Passed
test - tomcat_keystore     : Passed
test - tomcat_connectors   : Passed
test - tomcat_threads      : Passed
test - tomcat_memory       : Passed
test - tomcat_sessions     : Passed
skip - tomcat_heapdump     : This module must be run directly and off hours
test - validate_network    : Reverse DNS lookup failed
test - raid                : Passed
test - system_info         : Passed (Collected system information in diagnostic log)
test - ntp_reachability    : Passed
test - ntp_clock_drift     : Passed
test - ntp_stratum         : Passed
skip - sdl_fragmentation   : This module must be run directly and off hours
skip - sdi_fragmentation   : This module must be run directly and off hours
test - ipv6_networking     : Passed

Diagnostics Completed
```

A. The DNS server is missing PTR records for the Cisco UCM servers.

B. The firewall is blocking UDP port 53 traffic from Cisco UCM to the DNS servers.

C. The DNS servers are unreachable from the Cisco UCM server.

D. A failure is expected when the utils diagnose test command is run on a subscriber.

E. The DNS server has multiple PTR records for the Cisco UCM servers.

Answer: A,E (LEAVE A REPLY)

NEW QUESTION: 39

Which two actions must be taken to provision a new device using self-provisioning?

A. Enable the self-provisioning IVR in the Cisco UCM

B. Import the user profile to the corporate LDAP directory.

C. Link the appropriate service profile to the provisioning template.

D. Link the appropriate universal device template to the user profile.

E. Ensure the user has a directory URI and a primary extension.

Answer: A,D (LEAVE A REPLY)

Self-Provisioning of a phone to a user needs:

* an End user with a primary extension and a Feature Group Template
* the Feature Group Template needs a User Profile
* the User Profile needs a Universal Line and Universal Device Template Finally to activate it you need a CTI route point and a configured IVR service to actually register it.

HOWEVER, it looks like you dont actually NEED the IVR to technically fullfil the goal of self- registration, as users can use the URL based registration. but i am not 100% sure if you wont need the IVR anyways, even if you arent using it - the guide is not 100% clear on that

https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications- manager-callmanager/214228-configure-self-provisioning-feature-on-c.html

NEW QUESTION: 40

What is required when deploying co-resident VMs by using Cisco UCM?

A. Ensure that applications will perform QoS.

B. Avoid hardware oversubscription.

C. Provide a guaranteed bandwidth of 10 Mbps.

D. Deploy the VMs to a server running Cisco UCM.

Answer: B (LEAVE A REPLY)

NEW QUESTION: 41

Refer to the exhibit. A call to an international number has failed.

Which action corrects this problem?

```
23031952: Apr  9 17:43:21.203 EDT: ISDN Se0/1/0:23 Q931: Applying typeplan for sw-type 0xD is 0x2 0x1, Calling num 4085556100
23031953: Apr  9 17:43:21.203 EDT: ISDN Se0/1/0:23 Q931: Sending SETUP callref = 0x12BE callID = 0xA3F5 switch = primary-ni interface = User
23031954: Apr  9 17:43:21.203 EDT: ISDN Se0/1/0:23 Q931: TX -> SETUP pd = 8 callref = 0x12BE
        Bearer Capability i = 0x8090A2
                Standard = CCITT
                Transfer Capability = Speech
                Transfer Mode = Circuit
                Transfer Rate = 64 kbit/s
        Channel ID i = 0xA98393
                Exclusive, Channel 19
        Progress Ind i = 0x8183 - Origination address is non-ISDN
        Calling Party Number i = 0x2181, '4085556100'
                Plan:ISDN, Type:National
        Called Party Number i = 0x91, '011443075552222'
                Plan:ISDN, Type:International
23031956: Apr  9 17:43:21.279 EDT: ISDN Se0/1/0:23 Q931: RX <- CALL_PROC pd = 8 callref = 0x92BE
        Channel ID i = 0xA98393
                Exclusive, Channed 19
23031957: Apr  9 17:43:21.283 EDT: ISDN Se0/1/0:23 Q931: RX <- PROGRESS pd = 8 callref = 0x92BE
        Cause i = 0x829F - Normal, unspecified
        Progress Ind i = 0x8488 - In-band info or appropriate now available
23031981: Apr  9 17:43:46.802 EDT: ISDN Se0/1/0:23 Q931: TX -> DISCONNECT pd = 8 callref = 0x12BE
        Cause i = 0x8090 - Normal call clearing
23031982: Apr  9 17:43:46.822 EDT: ISDN Se0/1/0:23 Q931: RX <- RELEASE pd = 8 callref = 0x92BE
23031983: Apr  9 17:43:46.822 EDT: ISDN Se0/1/0:23 Q931: TX -> RELEASE_COMP pd = 8 callref = 0x12BE
```


A. Assign a transcoder to the MRGL of the gateway.

B. Strip the leading 011 from the called party number

C. Add the bearer-cap speech command to the voice port.

D. Add the isdn switch-type primart-dms100 command to the serial interface.

Answer: B (LEAVE A REPLY)

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.html

NEW QUESTION: 42

An engineer must deploy the Webex app to a Windows Virtual Desktop Infrastructure environment by using these settings:

- Thin client plugin: Installed
- Call service-enabled user: Media is optimized
- Calls on Webex app: Media is optimized

Which two command-line arguments are valid when running the installer? (Choose two.)

A. ALLUSERS=1

B. ALLUSERS=0

C. ENABLEVDI=1

D. ENABLEVDI=0

E. ENABLEVDI=2

Answer: A,E (LEAVE A REPLY)

NEW QUESTION: 43

Which command in the MGCP gateway configuration defines the secondary Cisco Unified Communications Manager server?

A. ccm-manager redundant-host

B. Mgcp call-agent

C. Mgcpapp

D. ccm-manager fallback-mgcp

Answer: (SHOW ANSWER)

https://www.cisco.com/c/en/us/td/docs/routers/access/vgd1t3/rel1_0/software/configuration/guide/ VGD_mgcp.html ccm-manager fallback-mgcp - Enables the MGCP fallback feature ccm-manager redundant-host {ip-address | DNS-name} [ip-address | DNS-name] - Identifies up to two backup Cisco Unified Communications Manager servers.

upvoted 2 times

NEW QUESTION: 44

By default, IP phones upgrade their images using the HTTP Service from the TFTP Server integrated into one or more of the call processing platforms. Which Port / Transport Protocol does this service use?

A. 6970 / UDP

B. 6970 / TCP

C. 443 / TCP

D. 8080 / TCP

Answer: A (LEAVE A REPLY)

NEW QUESTION: 45

Refer to the exhibit. A collaboration engineer adds a redundant NTP server to an existing Cisco Collaboration solution.

On the Cisco UCM OS Administration page the new NTP server shows as "Not Accessible".

Which action resolves this issue?

```
admin:utils ntp status
ntpd (pid 17428) is running...

     remote           refid      st t when poll reach  delay   offset   jitter
==============================================================================
*192.168.1.1     17.253.14.125    2 u   36   64   377   0.435   0.039    0.047
 192.168.1.2     .INIT.          16 u    -   64     0   0.000   0.000    0.000
```

A. Start the NTP service on the new NTP server
B. Delete and re-add the new NTP server via the Cisco UCM command-line interface
C. Restart NTPD on the Cisco UCM server
D. Configure the "reach" value as "377" for the new NTP server
Answer: B (LEAVE A REPLY)

NEW QUESTION: 46
What is a valid characteristic of Application Dial Rules configured on Cisco UCM?
A. They can be used to transform caller IDs for lookup in the directory.
B. They can be used to perform digit analysis and routing.
C. They can be used for dialing with Cisco Jabber CSF devices.
D. They can apply only to dialing from Cisco hard phones.
Answer: C (LEAVE A REPLY)

NEW QUESTION: 47
A company has security policies that prevent direct internet access from the publisher node of the Cisco UCM cluster. How must the licenses be activated for the Cisco UCM cluster?
A. configuration of an IPsec policy to secure the communication between Cisco UCM Publisher and Cisco Smart Software Manager
B. setup of a Cisco ASA to provide a Cisco Smart Software Manager satellite system for license reservation
C. use of a Cisco UCM subscriber node as an HTTPS proxy to communicate with the Cisco Smart Software Manager
D. manual exchange of information between Cisco UCM and Cisco Smart Software Manager to associate and reserve the licenses
Answer: C (LEAVE A REPLY)

NEW QUESTION: 48
Where does an administrator establish the trust boundary on a LAN?
A. access switch
B. distribution switch

C. voice VLAN
D. core router
Answer: A (LEAVE A REPLY)
On a LAN, the trust boundary is configured on the access switch.

NEW QUESTION: 49
SNMP notifications send traps immediately if the corresponding trap flags are enabled. Which two messages are Cisco UCM SNMP trap/inform messages? (Choose two.)
A. TLS Connection Failure
B. Media Resource Exhaustion
C. Route List Availability
D. Quality Report
E. Gateway Layer 3 Change
Answer: B,E (LEAVE A REPLY)

NEW QUESTION: 50
Which type of message must an administrator configure in the SIP Trunk Security Profile for a Message Waiting Indicator light to work with a SIP integration between Cisco UCM and Cisco Unity Connection?
A. TCP port 5060
B. SIP Register
C. Unsolicited NOTIFY
D. 200 OK
Answer: C (LEAVE A REPLY)
https://community.cisco.com/t5/collaboration-voice-and-video/understanding-troubleshooting- mwi-on-unity-connection/ta-p/3162948

NEW QUESTION: 51
An engineer must configure a dial plan for a new Cisco UCM cluster. Standard local route groups must be used for all local calls. After the route pattern, route list, and route groups are configured, which settings must the engineer configure for the standard local route groups?
A. device pool
B. trunk group
C. voice gateway
D. SIP trunk
Answer: A (LEAVE A REPLY)
In Cisco Unified Communications Manager (UCM), Standard Local Route Groups allow dynamic routing decisions based on the device pool assigned to the originating device. The device pool is the critical configuration that specifies the local route group to be used for routing calls. This eliminates the need for static route patterns for each site, simplifying the dial plan and increasing scalability.
Key Steps:
1. Configure Standard Local Route Group in Device Pool:
- Go to System > Device Pool in the UCM Administration interface.
- Select the appropriate device pool for the devices that will use the standard local route group.
- Assign the Standard Local Route Group in the "Local Route Group" field.
2. Dynamic Routing via Device Pools:
- When a call matches a route pattern configured to use Standard Local Route Group, the system dynamically selects the appropriate route group based on the calling device's assigned device pool.

NEW QUESTION: 52
What is used to create a QoS solution that delivers priority to voice and video traffic across a network?
A. traffic shaping
B. DiffServ
C. traffic policing
D. CBWFQ
Answer: D (LEAVE A REPLY)

NEW QUESTION: 53
A customer wants to deploy Cisco Jabber Phone Mode with contacts.
Which two configurations are needed to accomplish this goal? (Choose two)
A. Add the Application Client Users role to the Cisco UCM end users.
B. Remove the IM and Presence servers from the Cisco UCM service profile.
C. Enable Cisco UCM end-users for IM and Presence.
D. Import Contacts.xml into Jabber.
E. Disable Instant Messaging on IM and Presence.
Answer: C,E (LEAVE A REPLY)

NEW QUESTION: 54
When multiple potential patterns are present, which two things are considered when Cisco UCM selects a destination pattern? (Choose two.)
A. The pattern matches the shortest explicit prefix.
B. The pattern does not match the dialed string.
C. The pattern represents the smallest number of endpoints.
D. The pattern matches the dialed string.
E. The pattern represents the largest number of endpoints.
Answer: (SHOW ANSWER)
C is right since for example if we dial 3593 and we have 2 patterns 359X & 35XX CUCM will match 359X since this is the closest match and by chance this also represent the least number of endpoint (3590 to 3590) unlike 35XX (3500 to 3599).

NEW QUESTION: 55
What are two functions of a dial-plan? (Choose two.)
A. calling privileges
B. call recording
C. digit manipulation
D. ring cadence
E. endpoint features
Answer: A,C (LEAVE A REPLY)

NEW QUESTION: 56
Which configuration tells a switch port to send Cisco Discovery protocol packets that configure an attached Cisco IP phone to trust tagged traffic that is received from a device that is connected to the access port on the Cisco IP phone?

Router# configure terminal
A. Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos extend trust
B. Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos trust extend cos 5
Router# configure terminal
C. Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos trust extend
Router# configure terminal
D. Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos trust extend cos 3
Router# configure terminal
Answer: C (LEAVE A REPLY)

NEW QUESTION: 57
An administrator configures Cisco UCM to use UDP for SIP signaling and finds that an endpoint cannot make calls.
Which action resolves this issue?
A. Change the phone security profile.
B. Change the SIP dial rules.
C. Change the SIP profile
D. Change the common phone profile.
Answer: (SHOW ANSWER)
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/security/11_5_1/secugd/CUCM_BK
_SEE2CFE1_00_cucm-security-guide-1151/CUCM_BK_SEE2CFE1_00_cucm-security-guide-
1151_chapter_0111.html

NEW QUESTION: 58
Which DSCP value and PHB equivalent are the default for audio calls?
A. 46 and EF
B. 34 and AF41
C. 32 and AF41
D. 32 and CS4
Answer: (SHOW ANSWER)
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

NEW QUESTION: 59
Which DiffServ marking is the most likely to drop packets?
A. AF11
B. AF32
C. AF12
D. AF13

Answer: (SHOW ANSWER)

NEW QUESTION: 60

A collaboration engineer is troubleshooting issues with a T1 PRI. The engineer wants to confirm if the router received an ISDN Set Asynchronous Balanced Mode Extended message and responded with an Unnumbered Acknowledge frame. Which Cisco IOS command is used to debug ISDN Layer 2 issues?

A. debug isdn q931

B. debug isdn q921

C. debug voip ccapi inout

D. debug voice q921

Answer: (SHOW ANSWER)

NEW QUESTION: 61

Which type of traffic is marked as DSCP 46?

A. voice

B. call signaling

C. mission-critical data

D. IP routing

Answer: (SHOW ANSWER)

NEW QUESTION: 62

Which two are functions of a local gateway in Webex Calling? (Choose two.)

A. providing PSTN access to users in the local location

B. providing trunk access for the cloud connected PSTN services

C. connecting Webex Calling with an on-premises Cisco UCM

D. connecting Webex Calling endpoints to the Webex Calling cloud

E. connecting Webex Calling to the PSTN for the Cisco Calling Plan

Answer: A,C (LEAVE A REPLY)

NEW QUESTION: 63

What is a capability of a DNS server?

A. to assign a hostname to a host

B. to convert a hostname to an IP address

C. to convert an IP address into a MAC address

D. to assign an IP address to a host

Answer: B (LEAVE A REPLY)

NEW QUESTION: 64

User A calls user B. The call gets connected, but the quality is bad. What are two reasons for this issue? (Choose two.)

A. Incorrect QoS

B. Network congestion

C. Incompatible codec

D. Incorrect partition

E. No region relationship

Answer: (SHOW ANSWER)

NEW QUESTION: 65

Which action is needed to configure ReadOnly SNMPv3 on a Cisco UCM cluster?

A. Define an SNMP user with ReadOnly access privilege on the publisher node only.

B. Define an SNMP community string with ReadOnly access privilege on the publisher node only.

C. Define an SNMP community string with ReadOnly access privilege on all nodes in the cluster.

D. Define an SNMP user with ReadOnly access privilege and apply to all nodes in cluster.

Answer: (SHOW ANSWER)

NEW QUESTION: 66

What is a QoS requirement for VoIP calls?

A. 150 ms of one-way latency from mouth to ear

B. 15% packet loss

C. 150 bps per phone of guaranteed bandwidth

D. 150 ms of jitter

Answer: A (LEAVE A REPLY)

The following list summarizes the key QoS requirements and recommendations for voice (bearer traffic):

- Voice traffic should be marked to DSCP EF per the QoS Baseline and RFC 3246.
- Loss should be no more than 1 percent.
- One-way latency (mouth to ear) should be no more than 150 ms.
- Average one-way jitter should be targeted at less than 30 ms.
- A range of 21 to 320 kbps of guaranteed priority bandwidth is required per call (depending on the sampling rate, the VoIP codec, and Layer 2 media overhead).

Voice quality directly is affected by all three QoS quality factors: loss, latency, and jitter.

Loss

https://www.ciscopress.com/articles/article.asp?p=357102

NEW QUESTION: 67
Which action prevent toll fraud in Cisco Unified Communications Manager?
A. Configure ad hoc conference restriction.
B. Implement toll fraud restriction in the Cisco IOS router.
C. Allow off-net to off-net transfers.
D. Implement route patterns in Cisco Unified CM.
Answer: (SHOW ANSWER)
https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified- communications-manager-express/107626-cme-toll-fraud.html

NEW QUESTION: 68
An administrator is trying to change the default LINECODE for a voice ISDN T1 PRI.
Which command makes the change?
A. linecode ami
B. linecode hdb3
C. linecode b8zs
D. linecode esf
Answer: C (LEAVE A REPLY)
AMI is the default so to change the default you would need to set to something else and the most common linecode is b8zs.

Step 7	**linecode {ami | b8zs | hdb3 }** **Example:** `Router(config-controller)# linecode ami`	Sets the line-encoding method to match that of your telephone-company service provider. Keywords are as follows: • **ami** --Alternate mark inversion (AMI), valid for T1 or E1 controllers. Default for T1 lines. • **b8zs** --B8ZS, valid for T1 controllers only. • **hdb3** --High-density bipolar 3 (hdb3), valid for E1 controllers only. Default for E1 lines.

https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/isdn/configuration/15-mt/vi-15-mt-book/vi- isdn-vi-cfg.html

NEW QUESTION: 69
An administrator must configure several audio codecs for optimal bandwidth utilization in a large Cisco UCM solution that includes a gateway to ensure audio quality without call failures.
Where must a list of codecs be configured?
A. in the Cisco IOS router

B. in the Cisco ISDN gateway
C. under regions in the Cisco UCM
D. under service parameters in the Cisco UCM
Answer: C (LEAVE A REPLY)

NEW QUESTION: 70
An administrator must manipulate a number when a specific criterion is met. If this happens, the administrator wants to route the call again based on the new information to send the call to the correct destination. How must the administrator archive this function in the Cisco UCM?
A. Use a translation pattern with a CSS inside.
B. Set up a route pattern to manipulate numbers without sending the call directly to a gateway.
C. Set up a calling search space together with the route pattern to manipulate the call before the gateway.
D. Use a route pattern with a partition inside.
Answer: A (LEAVE A REPLY)

NEW QUESTION: 71
How does Cisco Unified Communications Manager perform a digit analysis on-hook versus off-hook for an outbound call from a Cisco IP phone that is registered to Cisco Unified CM?
A. On-hook, by pressing the digits and entering "#" to process the call, Unified CM performs a digit-by-digit analysis; off-hook, Unified CM analyses all digits as a string.
B. On-hook, no digit analysis is performed; off-hook, Unified CM requires the "#" to start the digit analysis.
C. On-hook, unified CM performs a digit-by-digit analysis; off-hook, Unified CM considers all digit were dialed and dos not wait for additional digits.
D. On-hook, Unified CM considers all digits were dialed and does not wait for additional digits; off-hook, Unified CM performs a digit-by-digit analysis.
Answer: (SHOW ANSWER)

NEW QUESTION: 72
In a Cisco Unified Communications Manager design where +E.164 destinations are populated in directory entries, which call routing practice is critical to prevent unnecessary toll charges caused by internal calls routed through the PSTN?
A. client matter codes
B. automated alternate routing
C. forced on-net routing
D. forced authorization codes
E. tail-end hop-off
Answer: C (LEAVE A REPLY)

NEW QUESTION: 73
Which command enables SIP OAuth mode on a Cisco UCM publisher node?
A. utils sip-OAuth-mode enable
B. utils sipOAuth-mode enable
C. utils sip-OAuth-mode activate
D. utils sipOAuth-mode activate
Answer: (SHOW ANSWER)
Use the Publisher's Command Line Interface to enable SIP OAuth mode globally. Run the command: utils sipOAuth-mode enable.
https://www.cisco.com/c/en/us/support/docs/unified-communications/jabber/215572-cisco-jabber-and-sip-oauth-mode.html

NEW QUESTION: 74

What is a capability of the call forwarding feature in a Cisco Webex dial plan?

A. device pool selection

B. Call Admission Control

C. business continuity

D. ringtone selection

Answer: (SHOW ANSWER)

Call forwarding is a feature that allows users to forward incoming calls to another number. This can be useful in a number of situations, such as when a user is not available to take a call, or when a user wants to forward calls to a different number during certain times of the day.

Call forwarding can be used to improve business continuity by ensuring that calls are always answered, even if the user is not available. For example, if a user is out of the office, they can forward their calls to their voicemail or to another employee. This ensures that customers and clients can always reach someone, even if the user is not available.

NEW QUESTION: 75

Refer to the exhibit. The translation rule is configured on the voice gateway to translate DNIS.

What is the outcome if the gateway receives 0255-343-1234 as DNIS?

```
rule 1 /^\(0[25]..\)\-\(...\)\-\(....$\)/ /\1\2\3/
```

A. The translation rule is not matched because DNIS does not end with a "$".

B. The translation rule is matched and the translated number is 02553431234.

C. The translation rule is not matched because DNIS contains "-".

D. The translation rule is matched and the translated number is 025553431234.

Answer: B (LEAVE A REPLY)

ISR4331(config)#voice translation-rule 10

ISR4331(cfg-translation-rule)#rule 1 /^\(0[25]..\)\-\(...\)\-\(....$\)/ /\1\2\3/ ISR4331#test voice translation-rule 10 0255-343-1234 Matched with rule 1 Original number: 0255-343-1234 Translated number: 02553431234 Original number type: none Translated number type: none Original number plan: none Translated number plan: none

NEW QUESTION: 76

An administrator installs a new Cisco Telepresence video endpoint and receives this error: "AOR is not permitted by Allow/Deny list." Which action should be taken to resolve this problem?

A. Correct the restriction policy settings.

B. Change the SIP trunk configuration.

C. Upload a new policy In VCS.

D. Reboot the VCS server and attempt reregistration.

Answer: A (LEAVE A REPLY)

https://www.cisco.com/c/en/us/td/docs/telepresence/infrastructure/articles/vcs_endpoint_registrati on_problems_kb_460.html

NEW QUESTION: 77
A Cisco UCM administrator wants to enable the Self-Provisioning feature for end users.
Which two prerequisites must be met first? (Choose two.)
A. End users must have a secondary extension.
B. Cisco Extended Functions service must be running
C. End users must belong to Standard CCM Admin Users group, the Standard CCM End Users group, and the Standard CCM Self-Provisioning group.
D. End users must have a primary extension.
E. End users must be associated to a user profile or feature group template that includes a universal line template and universal device template.
Answer: (SHOW ANSWER)
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_ C95ABA82_00_admin-guide-100/ CUCM_BK_C95ABA82_00_admin-guide-100_chapter_01101001.html

NEW QUESTION: 78
Refer to the exhibit. An administrator must fix the SRV records to ensure that server1.sample.com is always contacted first from the three servers. Which solution should the engineer apply to resolve this issue?

NAME	TTL	CLASS	TYPE	Priority	Weight	Port	Target Address
_sip._tcp.sample.com	86400	IN	SRV	10	60	5060	server1.sample.com
_sip._tcp.sample.com	86400	IN	SRV	10	30	5060	server2.sample.com
_sip._tcp.sample.com	86400	IN	SRV	5	20	5060	server3.sample.com

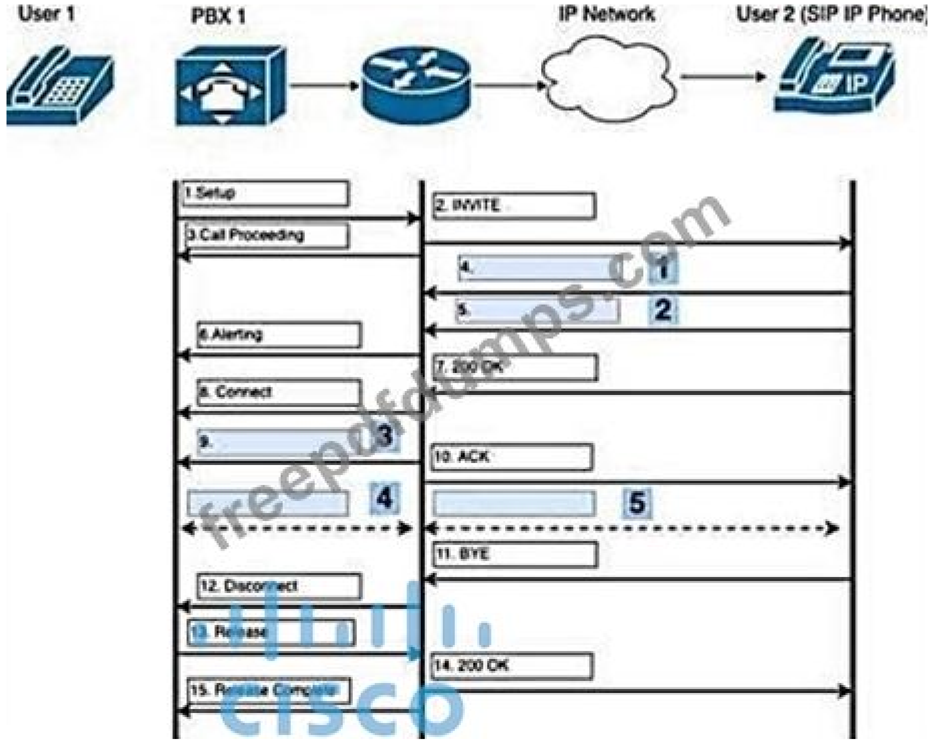
A. Priority = 100, Weight = 90
B. Priority = 10, Weight = 5
C. Priority = 5, Weight = 70
D. Priority = 10, Weight = 10
Answer: (SHOW ANSWER)

NEW QUESTION: 79
Which two features of Cisco Prime Collaboration Assurance require advanced licensing?
(Choose two.)
A. multicluster support
B. real time alarm browse
C. customizable events
D. email notifications
E. call quality monitoring
Answer: A,E (LEAVE A REPLY)

NEW QUESTION: 80

Drag and Drop Question

Refer to the exhibit.



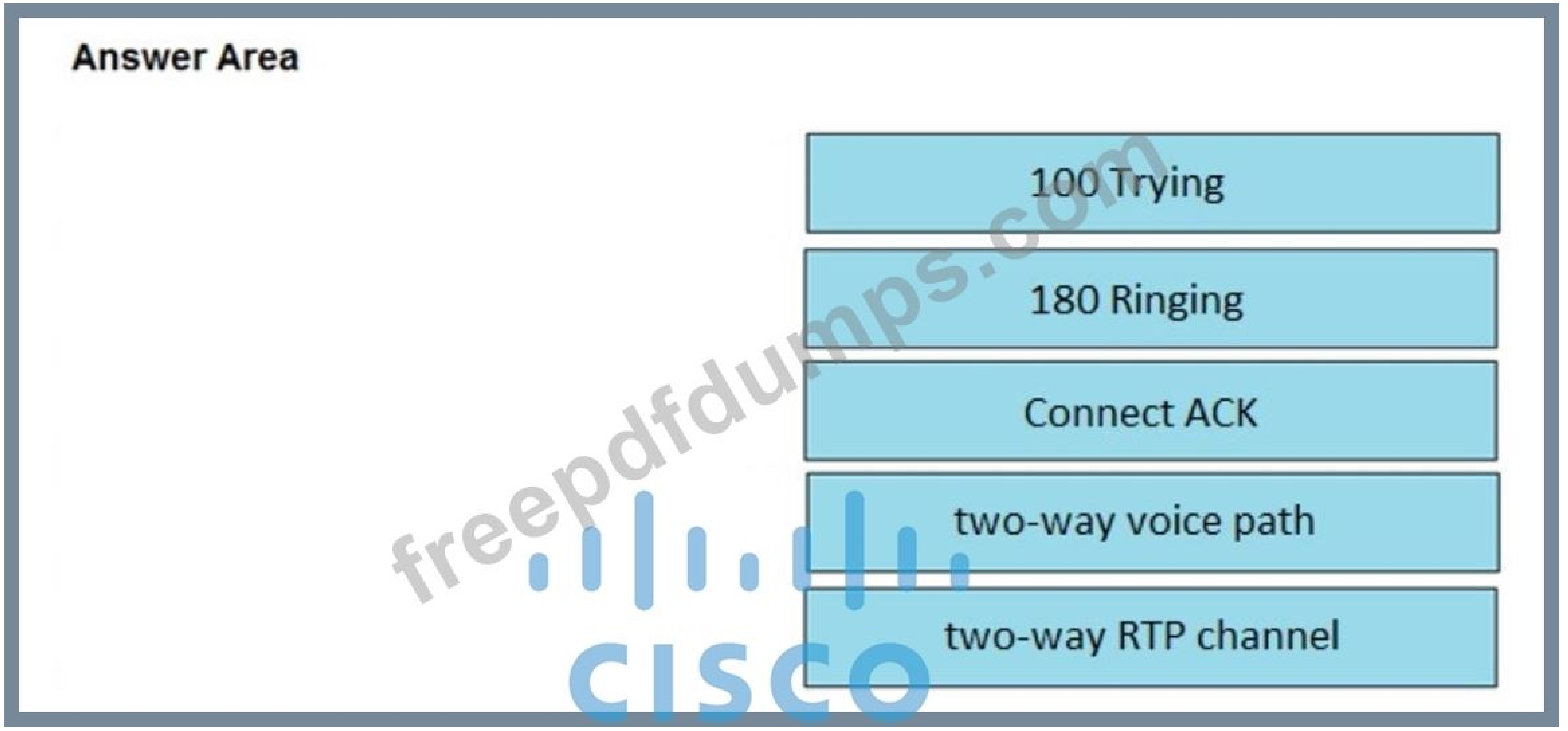
Drag and drop the flow step labels from the left into the correct order on the right to establish this call flow:

- User 1 calls user 2
- User 2 answers the call
- user 2 disconnects the call

Answer Area

two-way voice path	step 1
100 Trying	step 2
180 Ringing	step 3
two-way RTP channel	step 4
Connect ACK	step 5

Answer:



Explanation:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cuipph/7960g_7940g/sip/8_0/english/admi nistration/guide/8_0/sipaxb80.html

NEW QUESTION: 81

What is the purpose of Mobile and Remote Access (MRA) in the Cisco UCM architecture?

A. MRA is used to access Webex cloud services only if authenticated with on-premises LDAP service.

B. MRA is used to make B2B calls through Expressway registration.

C. MRA is used to access the collaboration services offered by Cisco UCM from off-premises network connections

D. MRA is used to make secure PSTN calls by Cisco UCM only while on-premises authentication.

Answer: (SHOW ANSWER)

NEW QUESTION: 82

What is the purpose of Mobile and Remote Access with Cisco Jabber in Cisco UCM?

A. to allow Cisco Jabber to register with Cisco UCM via a site-to-site VPN from a branch office

B. to allow Cisco Jabber to register with Cisco UCM via a remote access VPN from a branch office

C. to allow Cisco Jabber to register with Cisco UCM without the use of a VPN from outside the corporate network

D. to allow Cisco Jabber to register with Cisco UCM via a remote access VPN from outside the corporate network

Answer: (SHOW ANSWER)

Mobile and Remote Access (MRA) in Cisco Unified Communications Manager (CUCM) enables Cisco Jabber and other collaboration endpoints to securely register with CUCM from outside the corporate network without requiring a VPN.
MRA leverages Cisco Expressway-E and Expressway-C to provide a seamless and secure communication path, allowing users to access voice, video, messaging, and presence services remotely. This improves user experience and security by eliminating the need for complex VPN configurations.

NEW QUESTION: 83
A company has a Cisco collaboration infrastructure that includes Cisco UCM and a Cisco Unified Border Element. The company CTO asks an engineer to improve security and implement toll fraud prevention. The engineer configures partitions and a calling search space to provide segmentation and access control. Which two additional configurations are needed to block all OffNet to OffNet transfers? (Choose two.)
A. Classify the SIP trunk to Cisco Unified Border Element as OffNet.
B. Classify all SIP devices as OffNet.
C. Set the Call Classification Cisco CallManager service parameter to OffNet.
D. Set the Allow OffNet to OffNet Transfer Cisco CallManager service parameter to false.
E. Set the Block OffNet to OffNet Transfer CallManager service parameter to true.
Answer: (SHOW ANSWER)
To prevent OffNet-to-OffNet call transfers (which can be exploited for toll fraud), you must properly classify calls and configure restrictions in Cisco Unified Communications Manager (CUCM):
Classify the SIP trunk to Cisco Unified Border Element as OffNet
This ensures that calls passing through the trunk are considered external (OffNet). This classification helps in applying call restrictions effectively.
Set the Allow OffNet to OffNet Transfer Cisco CallManager service parameter to false This setting explicitly prevents OffNet-to-OffNet transfers, stopping users from transferring external calls to another external destination, which is a common toll fraud scenario.

NEW QUESTION: 84
Refer to the exhibit. An administrator is attempting to resolve a DTMF issue to a specific IVR and has supplied logs from the most recent issue. What is the cause and solution of the issue?

```
08315647.002 |04:43:03.673 |AppInfo  |SIP DTMF Info: mLocalDtmfCaps...UNSOL=1, KPML=1,
Inband=0(0) mEndppointsDtmfCaps...UNSOL=0, KPML=0, Inband=1(101)
mDefaultTelephonyEvent=101, mDtmfPreference=3, mMtpAllocated=0
```

A. The local endpoint supports RFC2833, and the IVR supports only KPML. Configure a transcoder to resolve the DTMF mismatch.
B. The local endpoint supports RFC2833, and the IVR supports only KPML. Configure an MTP to resolve the DTMF mismatch.
C. The local endpoint does not support RFC2833, and the IVR supports only RFC2833. Configure a transcoder to resolve the DTMF mismatch.
D. The local endpoint does not support RFC2833, and the IVR supports only RFC 2833. Configure an MTP to resolve the DTMF mismatch.
Answer: (SHOW ANSWER)

NEW QUESTION: 85
A company's employees have been complaining that they have been unable to select options on the internal IVR of the help desk IT support has been given Cisco UCM traces and below is the snippet of the SDP of the INVITE packet.
m=audio 25268 RTP/AVP 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=ptime:20
a=fmtp:18 annexb=no

a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
How is this issue resolved?
A. Configure CODEC for G 722
B. Configure CODEC for G.729
C. Configure DTMF for RFC 2833.
D. Configure DTMF for KPML
Answer: (SHOW ANSWER)

NEW QUESTION: 86
Refer to the exhibit. An engineer is troubleshooting an E1 interface module that fails to come up.
The E1 module must have these configurations:
- The linecode type must be Alternate Mark Inversion.
- The frame type must be Extended Super Frame.
- All available channels must be used.
An engineer already deployed the configurations shown in the exhibit to the Cisco voice gateway.
Which additional configurations must be performed to resolve the issue?

```
<output omitted>
voice-card 0
!
isdn switch-type primary-net5
!
network-clock-participate wic 2
network-clock-select 1 E1 0/2/0
!
card type e1 0 2
!
controller E1 0/2/0
 framing esf
 linecode ami
!
<output omitted>
```


A. controller E1 0/2/0
pri-group timeslots 1-31
B. controller E1 0/2/0
pri-group timeslots 1-23
C. interface E1 0/2/0:31
pri-group timeslots 1-31
D. interface E1 0/2/0:23
pri-group timeslots 1-23
Answer: A (LEAVE A REPLY)
The configuration provided in the exhibit partially sets up an E1 PRI (Primary Rate Interface) on the Cisco voice gateway. However, to enable the PRI interface fully, the pri-group configuration is required to define the timeslots being used for the E1 interface.
The configuration already sets the linecode to AMI (Alternate Mark Inversion) and the framing type to ESF (Extended Super Frame), which are correct for the requirement.
An E1 interface uses 31 timeslots (1-31), with timeslot 0 reserved for framing and timeslot 16 used for signaling. The command pri-group timeslots 1-31 is required to enable PRI and use all the available timeslots.

The pri-group command is applied under the controller configuration (e.g., controller E1 0/2/0) to define the E1 channel allocation.

NEW QUESTION: 87
Which type of input is required when configuring a third-party SIP phone?
A. digest user
B. serial number
C. manufacturer
D. authorization code
Answer: A (LEAVE A REPLY)
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_0_1/sysConfig/CUCM_B K_C733E983_00_cucm-system-configuration-guide-1101/Configure___Third_Party_SIP_Phones.pdf

NEW QUESTION: 88
Which device is a Cisco IP Phone 8800 series phone registered in, if it was registered using Mobile and Remote Access?
A. Cisco Unified Presence
B. Cisco Expressway-C.
C. Cisco Expressway-E
D. Cisco UCM
Answer: (SHOW ANSWER)
A Cisco IP Phone 8800 series phone registered using Mobile and Remote Access (MRA) ultimately registers to the Cisco Unified Communications Manager (UCM). MRA provides secure remote connectivity for endpoints, but it does not act as the endpoint registration platform itself.
Cisco UCM is the central call control system that manages endpoint registrations, call routing, and feature provisioning for Cisco IP Phones. Even with MRA, the actual phone registration happens with Cisco UCM.
Cisco Expressway-C and Expressway-E are used to enable secure MRA. However, they do not directly register the endpoints.
Cisco Unified Presence is used for presence and instant messaging services, not for IP phone registration.

NEW QUESTION: 89
An engineer builds this configuration on a Cisco IOS gateway for fax dial-peers:

```
dial-peer voice 2 voip
destination-pattern 5419213780
sessiontarget ipv4:10.5.6.7
fax protocol t38 Is-redundancy 0 hs-redundancy 0 fallback cisco
fax rate voice
```

Which command is required to complete the configuration?
A. Codec g726r32
B. Codec g729cr81
C. Codec g723ar63
D. Codec g711ulaw
Answer: D (LEAVE A REPLY)
https://www.cisco.com/c/dam/en/us/td/docs/ios/voice/dialpeer/configuration/guide/xe_3s/vd_dp_s e_3s_book.pdf

NEW QUESTION: 90
Which field of a Real-Time Transport Protocol packet allows receiving devices to detect lost packets?

A. Timestamp
B. CSRC (Contributing Source ID)
C. Sequence number
D. SSRC (Synchronization identifier)
Answer: C (LEAVE A REPLY)

NEW QUESTION: 91
Which command is used in a Cisco IOS XE TDM gateway to configure the voice T1/E1 controller to provide clocking?
A. Clock source network
B. Clock source line
C. Clock source internal
D. Cisco IOS XE TDM gateway T1/E1 controller cannot provide clocking.
Answer: C (LEAVE A REPLY)
XE IOS only have line and internal clock source.
https://www.cisco.com/c/en/us/td/docs/routers/ncs4200/configuration/guide/tdm/16-12-1/b-tdm-t1- e1-xe-16-12-1-ncs4200.html

NEW QUESTION: 92
What does average rate limiting allow?
A. transmits traffic bursts up to the Bc size
B. more traffic than the CIR to be sent when there is available bandwidth
C. bandwidth up to the Be size
D. traffic to burst to the Be size when there is available bandwidth
Answer: D (LEAVE A REPLY)
https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/qos_plcshp/configuration/15-mt/qos-plcshp-15- mt-book/qos-plcshp-oview.html

NEW QUESTION: 93
A user dials 9011841234567 to reach Vietnam. Which steps send the call to the PSTN provider as 011841234567?

A.
in the Called Party Transformation Pattern Configuration section,
configure the Pattern as 9.011841234567
configure the Discard Digits as Predot

B.
in the Calling Party Transformation Patterns section,
configure the Pattern as a 011841234557
configure the Discard Digits as Predot

C.
in the Calling Party Transformation Patterns section,
configure the Pattern as 9.011841234567
configure the Discard Digits as Predot 10-10-Dialing

in the Called Party Transformation Pattern Configuration section,
configure the Pattern as 9.011841234587
D. configure the Discard Digits as Predot 10-10-Dialing

Answer: (SHOW ANSWER)

NEW QUESTION: 94

Where in Cisco UCM are codec negotiations configured for endpoints?

A. in enterprise parameters

B. under device profiles in device settings

C. under regions using preference lists

D. in in-service parameters

Answer: C (LEAVE A REPLY)

You choose your Codec Preference List in the Region configuration when creating new relationships between Regions.

NEW QUESTION: 95

Which Cisco Unified Communications Manager feature is to determine the maximum bit rate for a call between two video-endpoints?

A. Partitions

B. Locations

C. Regions

D. transformations

Answer: (SHOW ANSWER)

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_S U1/cucm_b_system-configuration-guide-1151su1/cucm_b_system-configuration-guide-1151su1_chapter_0111.html

NEW QUESTION: 96

An engineer with ID012345678 must build an international dial plan in Cisco Unified Communications Manager.

Which action should be taken when building a variable-length route pattern?

A. Configure single route pattern for international calls

B. Reduce the T302 timer to less than 4 seconds

C. Create a second route pattern followed by the # wildcard

D. Set up all international route pattern to 0.1

Answer: (SHOW ANSWER)

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/dialplan.html

NEW QUESTION: 97

On which Cisco Unified Communications Manager nodes can the TFTP service be enabled?

A. Any node

B. Any subscriber nodes
C. Only nodes that have Cisco Unified CM service enabled
D. Any two nodes
Answer: A (LEAVE A REPLY)
A TFTP subscriber or server node performs two main functions as part of the Unified CM cluster:
The serving of files for services, including configuration files for devices such as phones and gateways, binary files for the upgrade of phones as well as some gateways, and various security files Generation of configuration and security files, which are usually signed and in some cases encrypted before being available for download The Cisco TFTP service that provides this functionality can be enabled on any server node in the cluster.

NEW QUESTION: 98
An engineer must configure a DHCP server to save automatic bindings on a remote host that has an IP address of 172.16.1 1. The IP address range of 172.16.1.103 to 172.16.1.106 must be blocked from the DHCP server. Which set of commands must the engineer run?

A.
```
enable
configure terminal
ip dhcp database ftp://user:password@172.16.1.1/
router-dhcp timeout 80
ip dhcp excluded-address 172.16.1.100 172.16.1.108
end
```

B.
```
enable
configure terminal
ip dhcp database ftp://user:password@172.16.1.1/
router-dhcp timeout 80
no ip dhcp conflict logging
ip dhcp excluded 172.16.1.108 172.16.1.100
end
```

C.
```
enable
configure terminal
no ip dhcp conflict logging ftp://
user:password@172.16.1.1/router-dhcp timeout 80
ip dhcp excluded-address 172.16.1.108 172.16.1.100
end
```

D.
```
enable
configure terminal
ip dhcp database timeout 80 ftp://
user:password@172.16.1.1/router-dhcp
no ip dhcp conflict logging
ip dhcp excluded-address 172.16.1.108 172.16.1.100
end
```

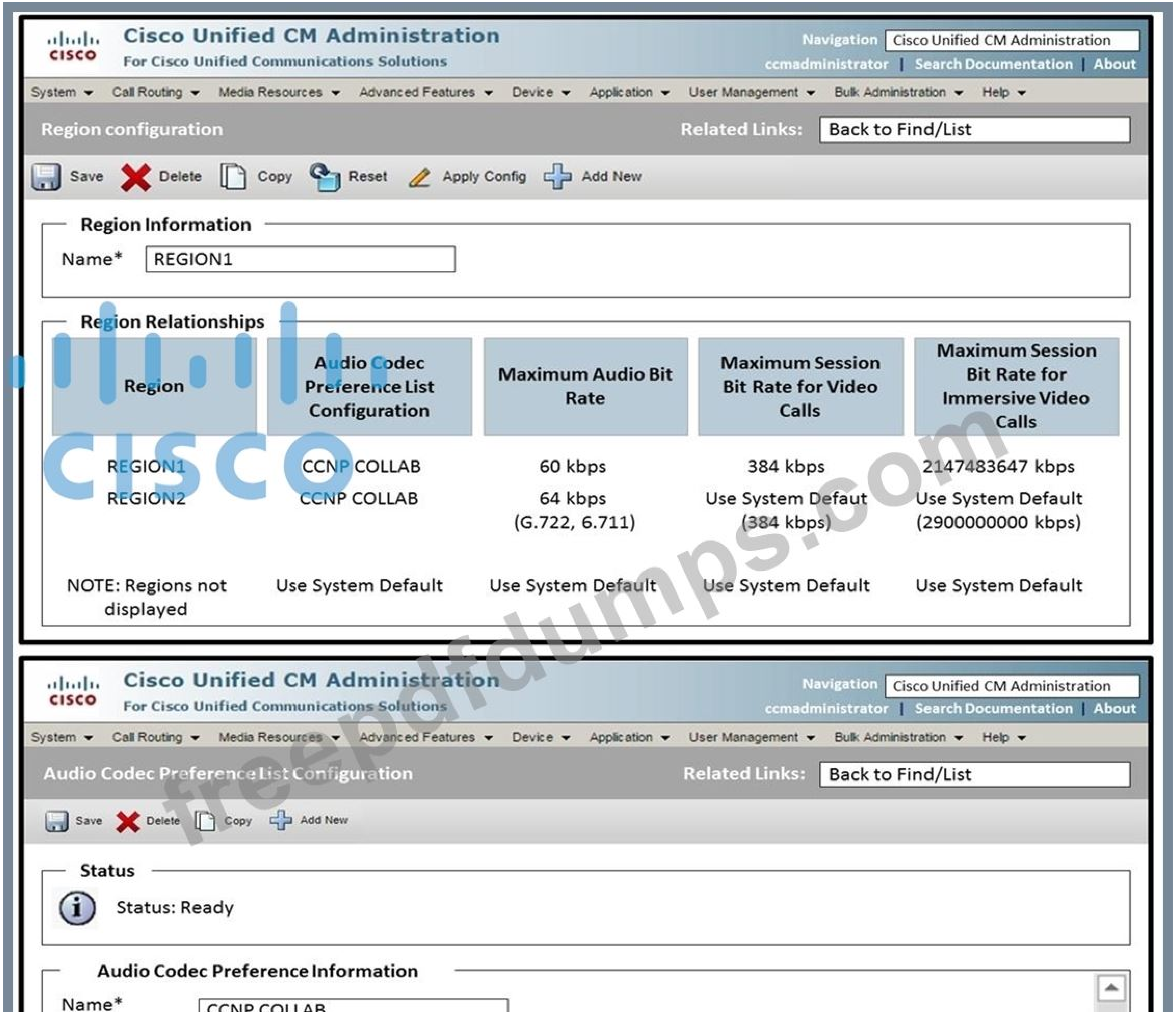
Answer: (SHOW ANSWER)

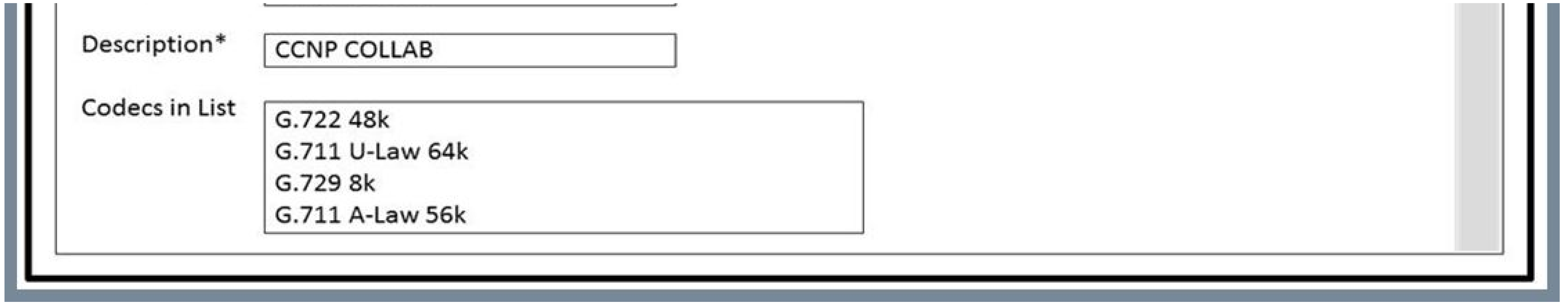
NEW QUESTION: 99
Refer to the exhibit. An engineer is troubleshooting this video conference issue:

A video call between a Cisco 9971 in Region1 and another Cisco 9971 in Region1 works.

As soon as the Cisco 9971 in Region1 conferences in a Cisco 8945 in Region2, the Region1 endpoint cannot see the Region2 endpoint video.

What is the cause of this issue?


Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unified CM Administration
ccmadministrator | Search Documentation | About
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Region configuration
Related Links: Back to Find/List
Save Delete Copy Reset Apply Config Add New
Region Information
Name* REGION1
Region Relationships
Region
Audio Codec Preference List Configuration
Maximum Audio Bit Rate
Maximum Session Bit Rate for Video Calls
Maximum Session Bit Rate for Immersive Video Calls
REGION1 CCNP COLLAB 60 kbps 384 kbps 2147483647 kbps
REGION2 CCNP COLLAB 64 kbps (G.722, 6.711) Use System Defaut (384 kbps) Use System Default (2900000000 kbps)
NOTE: Regions not displayed Use System Default Use System Default Use System Default Use System Default
Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unified CM Administration
ccmadministrator | Search Documentation | About
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Audio Codec Preference List Configuration
Related Links: Back to Find/List
Save Delete Copy Add New
Status
Status: Ready
Audio Codec Preference Information
Name* CCNP COLLAB



A. Cisco 8945 does not have a camera connected.

B. Maximum Audio Bit Rate must be increased.

C. Maximum Session Bit Rate for Video Calls is too low.

D. Maximum Session Bit Rate for Immersive Video Calls is too low.

Answer: C (LEAVE A REPLY)

The section in the question of "when the 9971 CONFERENCES IN the 8945..." indicating that the 8945 is being added to an already in-progress video call, totaling three or more video streams.

This would mean 384kbps is definitely way too low.

NEW QUESTION: 100

Which two functions are provided by Cisco Expressway series? (Choose two.)

A. endpoint registration

B. interworking of SIP and H.323

C. voice and video conferencing

D. voice and video transcoding

E. intercluster extension mobility

Answer: A,B (LEAVE A REPLY)

https://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/expressway/config_guide/X8-11/Cisco-Meeting-Server-2-4-with-Cisco-Expressway-Deployment-Guide_X8-11-4.pdf

NEW QUESTION: 101

An engineer configures Cisco Unified Communications Manager to prevent toll fraud.

At which two point does the engineer block the pattern in Cisco Unified CM to complete this task?

(Choose two.)

A. route pattern

B. translation pattern

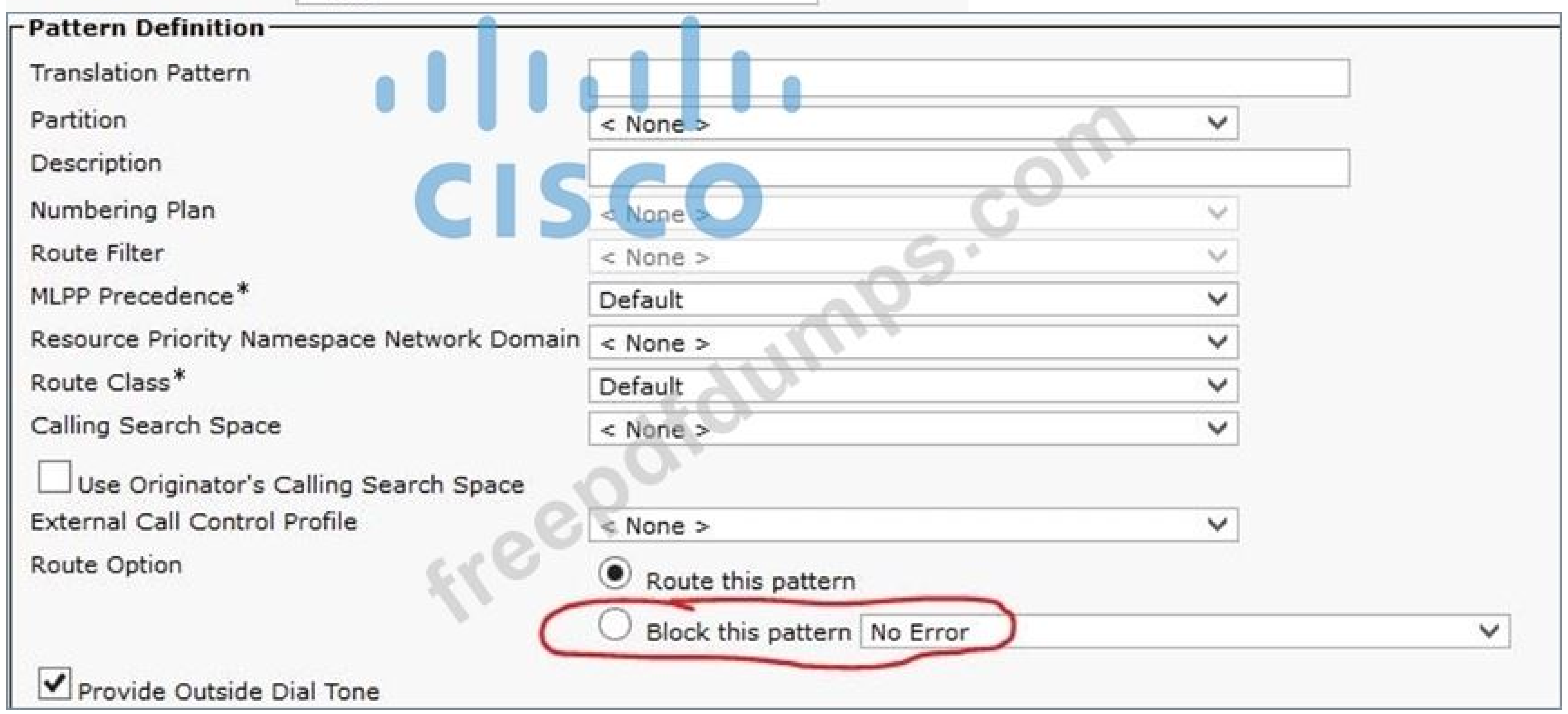
C. route group

D. partition

E. CSS

Answer: (SHOW ANSWER)

If we focus on patterns only, both translation pattern & route pattern have the ability to check box (block this pattern).



NEW QUESTION: 102

Which service must be enabled when LDAP on cisco UCM is used?

A. Cisco AXL Web Service

B. Cisco CallManager SNMP Service

C. Cisco DirSync

D. Cisco Bulk provisioning Service

Answer: (SHOW ANSWER)

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/10_
5_1/CUP0_BK_CE43108E_00_config-admin-guide- imp-105/CUP0_BK_CE43108E_00_config- admin-guide-imp-105_chapter_01000.html

NEW QUESTION: 103

Which protocol is used between Cisco Jabber clients for instant messaging and presence?
A. Jabber
B. P2P
C. SIP/SIMPLE
D. XMPP
Answer: (SHOW ANSWER)
An XMPP connection handles the presence information exchange and instant messaging operations for XMPP-based clients.
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/10_
5_1/CUP0_BK_CE43108E_00_config-admin-guide-imp-105/CUP0_BK_CE43108E_00_config- admin-guide-imp-105_chapter_00.html

NEW QUESTION: 104
A network administrator deleted a user from the LDAP directory of a company. The end user shows as Inactive LDAP Synchronize User in Cisco Unified Communications Manager. Which step is next to remove this user from Cisco Unified Communications Manager?
A. Restart the Dirsync service after the user is deleted from LDAP directory.
B. Delete the user directly from Cisco Unified Communications Manager.
C. Wait 24 hours for the garbage collector to remove the user.
D. Execute a manual sync to refresh the local databased and delete the end user.
Answer: C (LEAVE A REPLY)
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/directry.html
#pgfId-1045229

NEW QUESTION: 105
What should be used to detect common issues on a Cisco IOS XE-based Local Gateway and generate an email?
A. Real-Time Monitoring Tool
B. diagnostic signatures
C. syslog
D. SNMP
Answer: B (LEAVE A REPLY)
Diagnostic signatures are a feature that proactively detects commonly observed issues in the IOS XE-based Local Gateway and generates email, syslog, or terminal message notification of the event. You can also install the diagnostic signatures to automate diagnostics data collection and transfer collected data to the Cisco TAC case to accelerate resolution time.

NEW QUESTION: 106
What is the default TCP port for SIP OAuth mode in Cisco UCM?
A. 3174
B. 5090
C. 8443
D. 5011
Answer: (SHOW ANSWER)

NEW QUESTION: 107
An end user at a remote site is trying to initiate an Ad Hoc conference call to an end user at the main site. The conference receives an error message on the phone:
Cannot complete conference call.
What is the cause of the issue?
A. The remote phone does not have the conference feature assigned
B. The transcoder resource Is missing.
C. A software conference bridge is not assigned.
D. A Media Termination Point is missing.
Answer: B (LEAVE A REPLY)

NEW QUESTION: 108
Which traversal server must be deployed in the Demilitarized Zone when configuring Mobile and Remote Access?
A. Cisco IM and Presence
B. Cisco Expressway-E
C. Cisco Unified Border Element
D. Cisco Expressway-C
Answer: (SHOW ANSWER)
When configuring Mobile and Remote Access (MRA) for Cisco Collaboration, the Cisco Expressway-E server must be deployed in the Demilitarized Zone (DMZ). This deployment enables secure traversal of signaling and media traffic between the internal corporate network and external endpoints (remote users or devices).
Cisco Expressway-E:
- Acts as the external-facing component of the Expressway solution.
- Handles requests from external endpoints.
- Interfaces securely with the Cisco Expressway-C, which resides inside the corporate network.
- Ensures firewall traversal for SIP, H.323, and other collaboration protocols.

NEW QUESTION: 109
A collaboration engineer troubleshoots issues with a Cisco IP Phone 7800 Series. The IPv4 address of the phone is reachable via ICMP and HTTP and the phone is registered to Cisco UCM. However, the engineer cannot reach the CLI of the phone.
Which two actions in Cisco UCM resolve the issue? (Choose two)
A. Enable FIPS Mode under Product Specific Configuration Layout in Cisco UCM
B. Enable Settings Access under Product Specific Configuration Layout in Cisco UCM.
C. Enable SSH Access under Product Specific Configuration Layout in Cisco UCM
D. Disable Web Access under Product Specific Configuration Layout in Cisco UCM
E. Set a username and password under Secure Shell Information in Cisco UCM
Answer: (SHOW ANSWER)
Enable ssh for phone, and then setup username and password so that engineer can login to phone via CLI.
https://www.cisco.com/c/en/us/support/docs/collaboration-endpoints/ip-phone-7800- series/200850-Troubleshoot-Cisco-Phone-7800-8800-Serie.pdf

NEW QUESTION: 110

A customer asked to integrate Unity Connection with Cisco UCM using SIP protocol.

Which two features must be enabled on SIP security profiles? (Choose two)

A. Accept presence subscription

B. Enable application level authorization.

C. Allow charging header

D. Accept replaces header

E. Accept unsolicited notification

Answer: D,E (LEAVE A REPLY)

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/11x/integration/guide/cucm_si p/b_cucintcucmsip/b_cucintcucmsip_chapter_011.html

NEW QUESTION: 111

When a 8800 series phone is registered over MRA, where does it register?

A. Expressway-E

B. Expressway-C

C. Cisco Unified Communications Manager

D. Cisco Unified Presence Server

Answer: C (LEAVE A REPLY)

NEW QUESTION: 112

Which call routing pattern is used for phone numbers that are in the E.164 format?

A. /+.! Route Pattern

B. \+.! Route pattern

C. \+.! Translation pattern

D. \+1. [2-9]XX[2-9]XXXXXXX called Party Transformation pattern

Answer: B (LEAVE A REPLY)

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12/dialplan.html
#pgfId-1591747

NEW QUESTION: 113

Which IP precedence value maps to DSCP EF?

A. 5

B. 3

C. 1

D. 0

Answer: A (LEAVE A REPLY)

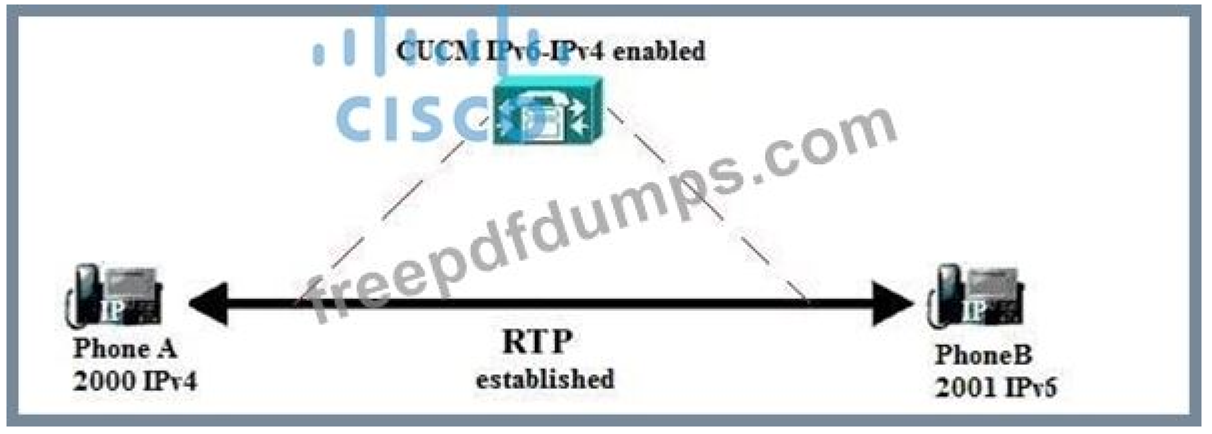
NEW QUESTION: 114

What are two Cisco UCM location bandwidths that are deducted when G.729 and G.711 codecs are used? (Choose two.)

A. If a call uses G.729, Cisco UCM subtracts 16k.

B. If a call uses G.711, Cisco UCM subtracts 64k
C. If a call uses G.711, Cisco UCM subtracts 80k
D. If a call uses G.729, Cisco UCM subtracts 24k.
E. If a call uses G.729, Cisco UCM subtracts 40k
Answer: (SHOW ANSWER)
After the call is established, the Cisco Unified Communications Manager will subtract bandwidth from the locations, depending on the codec that is used in that call.
- If the call is using G.711, Cisco Unified Communications Manager subtracts 80k.
- If the call is using G.723, Cisco Unified Communications Manager subtracts 24k.
- If the call is using G.729, Cisco Unified Communications Manager subtracts 24k.
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/trouble/11_0_1/CUCM_BK_T1F724
3A_00_troubleshooting-guide-cucm-1101/ device_issues.html

NEW QUESTION: 115
Refer to the exhibit. IP phone A makes a call to IP phone B and RTP voice stream is successfully established between the endpoints.



What option describes how to accomplish this call on phones with incompatible IP addressing versions?
A. Set IP addressing mode preference for signalling to IPv4
B. Set common device configuration to IPv4 and IPv6
C. Enable alternative network address types
D. Assign a media resource group with available MTPs
Answer: D (LEAVE A REPLY)

NEW QUESTION: 116
An administrator is asked to implement toll fraud prevention in Cisco UCM, specifically to restrict off- net to off-net call transfers How is this implemented?
A. Implement time-of-day routing.
B. Use the correct route filters.
C. Set the appropriate service parameter
D. Enforce ad-hoc conference restrictions
Answer: C (LEAVE A REPLY)
https://www.ciscopress.com/articles/article.asp?p=2218297&seqNum=10
NEW QUESTION: 117
After an engineer implements the FAC and CMC features together, users report that calls take almost one minute to complete and that they occasionally hear the reorder tone.

Which two actions address this issue? (Choose two.)

A. Change the code if the problem persists.

B. Adjust the T302 timer from the default of 15 seconds to 5 seconds to shorten the interdigit timer.

C. Do not wait for the tones immediately dial the FAC and CMC

D. Advise the user to hang up and try again.

E. Advise the user to press the "#" button after dialling the FAC and CMC codes.

Answer: B,E (LEAVE A REPLY)

NEW QUESTION: 118

Which access control group is required on an end user to allow Jabber to do deskphone mode?

A. Standard CTI Enabled

B. Standard CTI Allow Reception of SRTP key Material

C. Allow control of device from CTI

D. Standard CTI Secure Connection

Answer: (SHOW ANSWER)

Enable device for CTI: Allows Cisco Jabber desktop clients to control the desk phone of the user.

NEW QUESTION: 119

A collaboration engineer is deploying a cloud-registered Cisco Webex Desk series device. This device is being configured for a single user. Which step must the engineer complete first?

A. Configure the TFTP server setting on the device.

B. Generate an activation code via Webex Control Hub.

C. Configure DHCP Option 150.

D. Generate an activation code via the Cisco UCM Administration webpage.

Answer: B (LEAVE A REPLY)

When deploying a cloud-registered Cisco Webex Desk series device for a single user, the first step is to generate an activation code via Webex Control Hub. This code is required to register the device with the Webex cloud.

Cisco Webex devices that are cloud-registered do not use Cisco UCM or TFTP settings for provisioning. Instead, they rely on Webex Control Hub for management and activation.

The activation code is entered on the Webex device to complete registration, allowing the device to connect to Webex services.

NEW QUESTION: 120

Refer to the exhibit. When a call is received on Cisco Unified Border Element, from which IP does it allow a connection?

```
!
voice service voip
 ip address trusted list
  ipv4 192.168.100.101
  ipv4 192.168.101.0 255.255.255.128
!
dial-peer voice 1 voip
 destination-pattern +T
 session protocol sipv2
 session target ipv4:192.168.102.102
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
```
A. 192.168.101.201
B. 192.168.100.102
C. 192.168.100.103
D. 192.168.102.102
Answer: A (LEAVE A REPLY)
The ip address trusted list configuration under the voice service voip section in Cisco Unified Border Element (CUBE) defines which IP addresses or subnets are allowed to send SIP requests to the CUBE. If the source IP address of an incoming SIP message is not in the trusted list, the CUBE rejects the connection. The trusted list allows SIP connections only from 192.168.101.201, as it matches the subnet 192.168.101.0/25 in the configuration.

NEW QUESTION: 121
After an administrator installs the Cisco jabber Client, it fails to register with the cisco UCM server. Which two actions should be taken to troubleshoot this problem? (Choose two.)
A. Verify the installation of the LMHOST file on the PC
B. Verify the configuration of the DNS SRV record.
C. Verify corporate firewall settings for client connections.
D. Verify Layer 3 connectivity on the gateway.
E. Reboot the Cisco Unified Border Element Gateway.
Answer: B,C (LEAVE A REPLY)
https://www.cisco.com/c/en/us/support/docs/unified-communications/jabber/212471-jabber- login.html

NEW QUESTION: 122

What is the maximum DNS SRV entries that should be defined in the SIP Trunk destination address field in Cisco UCM?

A. 4

B. 8

C. 1

D. 16

Answer: C ([LEAVE A REPLY]())

"You can assign up to 16 different destination addresses for a SIP trunk, using IPv4 or IPv6 addressing, fully qualified domain names, or you can use a single DNS SRV record."
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/CUCM_B K_SE5DAF88_00_cucm-system-configuration-guide-1151/CUCM_BK_SE5DAF88_00_cucm- system-configuration-guide-1151_chapter_01110.html

NEW QUESTION: 123

An engineer roust configure DTMF relay on a Cisco Unified Border Element by using RFC2833 as the preferred relay mechanism and KPML as the next preferred relay mechanism. The engineer logs in to the CUBE and enters the dial-peer configuration level. Which command should be run at dial-peer configuration level?

A. dtmf-relay sip-kvmi rtp-nte

B. dtmf-relay rtp-nte sip-kpml

C. dtmf-relay sip-kgml rtp-inband

D. dtmf-relay rtp-inband sip-kvmi

Answer: ([SHOW ANSWER]())

The dtmf-relay command is used to configure DTMF relay on a Cisco Unified Border Element.
The rtp-nte option specifies that RFC2833 is the preferred relay mechanism, and the sip-kpml option specifies that KPML is the next preferred relay mechanism.

NEW QUESTION: 124

What is the purpose of configuring outgoing call permissions in a Cisco Webex dial plan?

A. to provide access to the dial plan section of the Webex admin portal

B. to configure the calling search space and the user partition

C. to allow or block access to operator assistance calls

D. to force users to call into meetings by disabling the call back feature

Answer: C ([LEAVE A REPLY]())

NEW QUESTION: 125

Refer to the exhibit. An administrator must replace the T1 card with an E1 card.
What is the correct configuration if the administrator was asked to configure 12 time slots?

```
controller t1 0/0/1
 pri-group timeslots 1-24
 clock source line
 linecode b8zs
 framing esf
```

A.
```
controller e1 0/0/1
 pri-group timeslots 1-11, 12
 clock source line
 linecode hdb3
 framing crc4
```
B.
```
controller e1 0/0/1
 pri-group timeslots 1-12
 clock source line
 linecode hdb3
 framing crc4
```
C.
```
controller e1 0/0/1
 pri-group timeslots 1-12
 clock source line
 linecode crc4
 framing hd3
```
D.
```
controller e1 0/0/1
 pri-group timeslots 1-12
 clock source network
 linecode hdb3
 framing crc4
```
Answer: (SHOW ANSWER)

NEW QUESTION: 126

An administrator is configuring a new Cisco UCM with PSTN capabilities. Due to bandwidth constraints, audio compression is used on the codec. DTMF must work as expected because the customer is calling many call centers where the users must select options in the call.

Where is DTMF out-of-band in a CCM 12.5 with SIP-based gateway configured?

A. in the DTMF setting under SIP profile on the Cisco Unified Border Element

B. in DTMF settings in the audio codec preference list under regions in the Cisco UCM

C. in regions on the Cisco UCM where you set the appropriate codec to use

D. in the dial peer on the Cisco IOS router

Answer: (SHOW ANSWER)

https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-border- element/200412-DTMF-Relay-and-Interworking-on-CUBE.html

NEW QUESTION: 127

According to QoS guidelines, what is the packet loss for streaming video?

A. Not more than 8%

B. Not more than 1%

C. Not more than 3%

D. Not more than 5%

Answer: D (LEAVE A REPLY)

When addressing the QoS needs of Streaming-Video traffic, the following guidelines are recommended:

- Streaming-Video (whether unicast or multicast) should be marked to DSCP CS4, as designated by the QoS Baseline.
- Loss should be no more than 5 percent.

https://www.ciscopress.com/articles/article.asp?p=357102&seqNum=2

NEW QUESTION: 128

An engineer is configuring a PRI card on a Cisco ISR 4300 Series Router. When the engineer configures the voice PRI on the router, this error is received: "T1 0/1/0: No DSP resources to configure voice feature". Which action must the engineer take to resolve the issue if a PVDM card was installed in the motherboard?

A. Install the NIM card on a different slot.

B. Replace the flashcard, restart the router, and try it again.

C. Install the PVDM directly to the NIM card.

D. Replace the memory slot in the motherboard.

Answer: C (LEAVE A REPLY)

NEW QUESTION: 129

An administrator troubleshoots a DNS issue with Cisco UCM and observes that the hostname resolution returns the wrong IP address. The DNS team informs the administrator that the A/AAA record was recently updated on the DNS server. Which action must be taken on Cisco UCM to reflect the correct entries?

A. Restart the DNS server to flush any stale entries

B. Flush the current cache using this CLI command: admin:utils network name-service hosts cache validate

C. Flush the current cache using this CLI command: admin:utils network name-service hosts cache invalidate

D. Delete the arp entry of the host utils network arp delete

Answer: C (LEAVE A REPLY)

NEW QUESTION: 130

What is the element of Cisco collaboration infrastructure that allows Jabber clients outside of the network to register in Cisco Unified communications Manager and use its resources?

A. Cisco Expressway

B. Cisco prime collaboration provisioning server

C. Cisco Unified Border Element

D. Cisco IM and Presence node

Answer: A (LEAVE A REPLY)

Cisco Unified Communications Mobile and Remote Access (MRA) is part of the Cisco Collaboration Edge Architecture. It allows endpoints such as Cisco Jabber to have their registration, call control, provisioning, messaging and presence services provided by Cisco Unified Communications Manager (Unified CM) when the endpoint is outside the enterprise network. The Expressway provides secure firewall traversal and line-side support for Unified CM registrations.

NEW QUESTION: 131

What is the function of DiffServ?

A. It categorizes traffic.

B. It provides reserve bandwidth by using RSVP

C. It automatically identifies traffic

D. It guarantees end-to-end bandwidth for traffic.

Answer: A (LEAVE A REPLY)

NEW QUESTION: 132

Which two conditions must a user meet to provision a new device using the self-provisioning feature? (Choose two.)

A. The user must have the appropriate universal device template linked to the linked to the user profile.

B. The user must have at least one user device profile assigned.

C. The user must have a primary extension.

D. At least DNs must be assigned to the user device.
E. The user must be part of "standard CCM Super User".
Answer: A,C (LEAVE A REPLY)
Before your end users can use self-provisioning, your end users be configured with the following items:
* Your end users must have a primary extension.
* Your end users must be associated to a user profile or feature group template that includes a universal line template, universal device template. The user profile must be enabled for self-provisioning.

NEW QUESTION: 133
An administrator is troubleshooting an issue where an 8845 is not joining the correct voice VLAN configuration from a switch and they believe that LLDP is the issue. Which command allows the administrator to see the packets sent from the phone to the switch?
A. show lldp
B. show lldp run
C. show lldp traffic
D. show lldp status <interface>
Answer: C (LEAVE A REPLY)

NEW QUESTION: 134
On which protocol and port combination does Cisco prime collaboration receive notifications (Traps and InformRequests) from several network devices in the collaboration infrastructure for which it has requested notifications?
A. UDP 161
B. TCP 161
C. TCP 80
D. UDP 162
Answer: (SHOW ANSWER)
https://www.cisco.com/c/en/us/td/docs/net_mgmt/prime/collaboration/12-1/assurance/advanced/guide/cpco_b_cisco-prime-collaboration-assurance-guide-advanced-12-1/cpco_b_cisco-prime-collaboration-assurance-guide-advanced-12-1_chapter_01111.html

NEW QUESTION: 135
A Cisco voice gateway is configured to use a sip-kpml DTMF relay in global settings. A new SIP dial peer is configured for a third-party application that only supports an in-band DTMF relay.
Which command must an engineer run on the dial peer?
A. dtmf-relay sip-info
B. no dtmf-relay sip-kpml
C. dtmf-relay rtp-nte
D. dtmf-relay sip-notify
Answer: (SHOW ANSWER)

NEW QUESTION: 136
Which value should be changed when each Cisco UCM node does not allow for more than 5000 phones to be registered?
A. Maximum Number of Registered and Unregistered Devices service parameter on each node
B. Minimum Number of Phones service parameter on each node
C. Maximum Number of Phones service parameter on the Publisher

D. Maximum Number of Registered Devices service parameter on each node
Answer: D (LEAVE A REPLY)

NEW QUESTION: 137
Refer to the exhibit. Endpoint A calls endpoint B.
What is the only audio codec that can be used for the call?

```
Endpoint A:
m=audio 21796 RTP/AVP 108 9 104 105 101
  b=TIAS:64000
  a=extmap:14 http://protocols.cisco.com/timestamp#100us
  a=rtpmap:108 MP4A-LATM/90000
  a=fmtp:108 bitrate=64000;profile-level-id=24;object=23
  a=rtpmap:9 G722/8000
  a=rtpmap:104 G7221/16000
  a=fmtp:104 bitrate=32000
  a=rtpmap:105 G7221/16000
  a=fmtp:105 bitrate=24000
  a=rtpmap:101 telephone-event/8000
  a=fmtp:101 0-15
  a=trafficclass:conversational.audio.immersive.aq:admitted

Endpoint B:
m=audio 21796 RTP/AVP 105 0 8 18 101
  b=TIAS:64000
  a=extmap:14 http://protocols.cisco.com/timestamp#100us
  a=rtpmap:105 G7221/16000
  a=fmtp:105 bitrate=24000
  a=rtpmap:0 PCMU/8000
  a=rtpmap:8 PCMA/8000
  a=rtpmap:18 G729/8000
  a=fmtp:18 annexb=no
  a=rtpmap:101 telephone-event/8000
  a=fmtp:101 0-15
  a=trafficclass:conversational.audio.immersive.aq:admitted
```

A. G722/8000
B. PCMA/8000
C. Telephone-event/8000
D. G7221/16000
Answer: D (LEAVE A REPLY)
Endpoint B Is not supported G722/8000, both endpoint only supported G721/16000.

NEW QUESTION: 138
What is the purpose of a hybrid Local Gateway?
A. to handle calls between Cisco UCM and Webex Calling
B. to handle calls between Webex Calling and Cisco Calling Plans
C. to handle calls between Webex Calling and Cloud Connected PSTN
D. to handle calls between the Public Switched Telephone Network and Webex Calling
Answer: (SHOW ANSWER)

NEW QUESTION: 139
SIP proxies have operations defined in RFC 3261 and supporting extensions. Though no IETF RFC completely defines how SBCs must function. SBCs evolved over the years.
Which two operations demonstrate the high-level differences between SBCs and SIP proxies?
(Choose two.)
A. SIP proxies are SDP-aware and can change the SDP bodies
B. SBCs are capable of interworking completely different protocols to set up, modify, and tear down communication sessions. It includes SIP, H.323,and MGCP protocols
C. Stateful proxies are context-aware and can terminate communication sessions by themselves
D. SBCs can modify headers such as To, From, Contact, and Call-D.It can introduce new headers into the SIP message
E. SIP proxies add a Via header and optionally a Record-Route header, and the rest of the headers are left untouched
Answer: B,E (LEAVE A REPLY)

NEW QUESTION: 140
An administrator must configure a dial peer on a Cisco Unified Border Element to redirect specific calls to an IP PBX that has an IP address of 10.18.92.2 when the number 9999 is dialed. Which command must be run on CUBE?
A. destination-pattern 9999
B. session target ipv4:10.18.92.2
C. session target 9999
D. destination-pattern ipv4:10.18.92.2
Answer: B (LEAVE A REPLY)
The session target command defines the remote IP address to which matching calls are sent, and specifying the IPv4 address of the IP PBX ensures that calls dialed with the configured pattern are routed to that system.

NEW QUESTION: 141
Where would an administrator update learned patterns from the DNA UI?
A. Service -> Control Center
B. Service -> Control Center -> DNA
C. System -> Enterprise parameters
D. System -> Service Parameters
Answer: A (LEAVE A REPLY)

NEW QUESTION: 142
(Choose two.)
Which two devices are supported by the Flexible DSCP Marking and Video Promotion feature?
A. pass-through MTPs

B. DX80
C. SCCP devices
D. MGCP devices
E. H.323 trunks
Answer: A,C (LEAVE A REPLY)

NEW QUESTION: 143
All calls that involve external parties must be based on a format to simplify the International multisite dial plans when globalized call routing is used. Which standard must be followed to implement normalized call routing?
A. Internal to External
Calling-party number: E.164
Called-party number: E.164
B. Internal to External (local number)
Calling-party number: E.164 (localized, subscriber)
Called-party number: E.164
C. External (local number) to Internal
Calling-party number: E.164
Called-party number: E.164 (localized, subscriber)
D. External to Internal
Calling-party number: E.164
Called-party number: E.164
Answer: (SHOW ANSWER)

NEW QUESTION: 144
Which program is required to deploy the Cisco Jabbar client on an on-premises Cisco collaboration solution?
A. Cisco Expressway-C
B. Cisco UCM IM and Presence
C. Cisco Unity Connection
D. Cisco UCM
Answer: D (LEAVE A REPLY)
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/jabber/14_0/cjab_b_deploy-jabber-on- premises-14_0/cjab_b_deploy-jabber-on-premises-129_chapter_010000.html

NEW QUESTION: 145
Which action enables Cisco MRA?
A. Clients such as Cisco Jabber can use call control on Cisco UCM.
B. Internal SIP clients registered to Cisco UCM can call external companies
C. Cisco UCC Express clients can obtain VPN connectivity to Cisco UCC Enterprise.
D. VPN connectivity can be established to Cisco UCM.
Answer: A (LEAVE A REPLY)

NEW QUESTION: 146

What is a capability provided by the Cisco Expressway Series?
A. voice and video conferencing
B. voice and video transcoding
C. intercluster extension mobility
D. endpoint registration
Answer: D (LEAVE A REPLY)
The Cisco Expressway Series is primarily used for secure communication and collaboration services across networks. One of its key capabilities is endpoint registration, which allows remote Cisco Jabber clients and SIP-based endpoints to register with Cisco Unified Communications Manager (CUCM) via Mobile and Remote Access (MRA).

NEW QUESTION: 147
What is the purpose of configuring locations in a Cisco Webex dial plan?
A. to restrict access to international calling capabilities
B. to identify the location of Webex data centers
C. to restrict access to long distance calling capabilities
D. to organize calling features within the same physical location
Answer: D (LEAVE A REPLY)

NEW QUESTION: 148
An employee of company ABC just quit. The IT administrator deleted the employee's user id from the active directory at 10 a m. on March 4th The nightly sync occurs at 10 p.m daily.
The IT administrator wants to troubleshoot and find a way to delete the user id as soon as possible How is this issue resolved?
A. Wait until 10 pm on March 4th when the user is automatically removed from Cisco UCM
B. Wait until 10 pm on March 5th when the user is automatically removed from Cisco UCM.
C. Wait until 3:15 a.m. on March 6th for garbage collection to remove the user from Cisco UCM.
D. Wait until 3:15 a.m. on March 5th for garbage collection to remove the user from Cisco UCM
Answer: (SHOW ANSWER)
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/directry.html
#pgfId-1045229

NEW QUESTION: 149
Which transport protocol does the application layer protocol SNMP use?
A. HTTP
B. XML
C. UDP
D. SIP
Answer: C (LEAVE A REPLY)
SNMP is an application layer protocol which uses UDP port number 161/162. SNMP is used to monitor the network, detect network faults and sometimes even used to configure remote devices. It is a software management software module installed on a managed device.

NEW QUESTION: 150
What is the purpose of the Claim users to your organization feature in Cisco Webex Control Hub?
A. to allow self-registration and sideboarding for new users
B. to add new users to an organization without sending automated emails

C. to manage users who have registered for a Webex account on their own
D. to assign users subscription licenses for an organization
Answer: (SHOW ANSWER)
In Cisco Webex Control Hub, the "Claim users to your organization" feature allows administrators to incorporate users who have independently registered for Webex accounts using the organization's email domain. This ensures that all users associated with the organization's domain are managed centrally, providing consistent access to services and adherence to organizational policies.
By claiming these users, administrators can:
- Consolidate User Management: Ensure all users with the organization's email domain are under a single administrative umbrella.
- Assign Appropriate Licenses: Provide users with the correct subscription licenses as per organizational entitlements.
- Enforce Security Policies: Apply consistent security and compliance policies across all users.
This process is particularly useful when users have created personal Webex accounts using their work email addresses, leading to potential discrepancies in service access and policy enforcement. By claiming these users, organizations can streamline user management and maintain control over their collaboration environment.
Reference: https://help.webex.com/en-us/article/nceb8tm/Claim-users-to-your-organization- ("convert"-users)

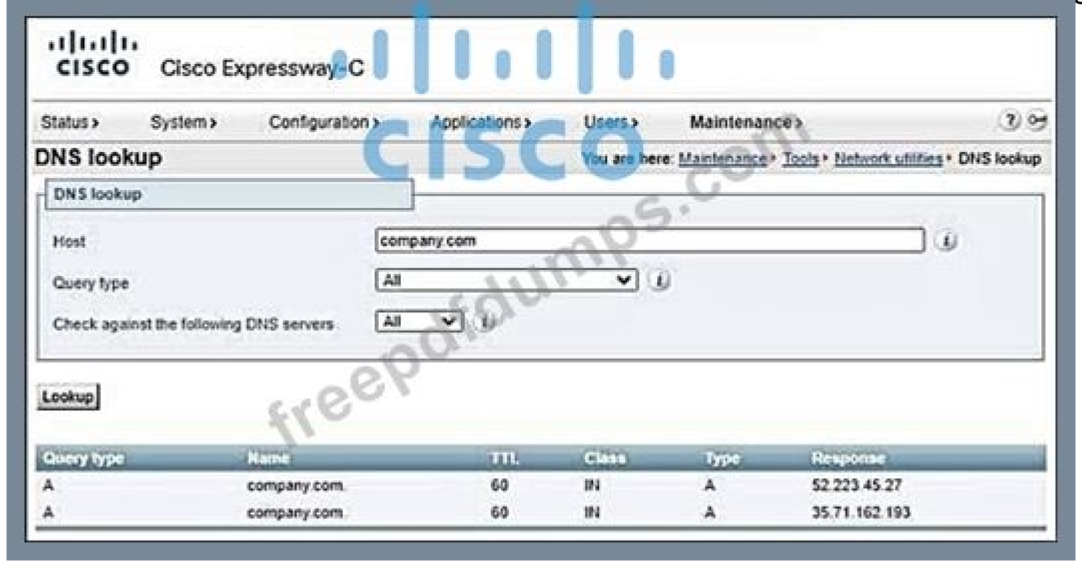
NEW QUESTION: 151
A Cisco voice gateway is configured to use a sip-kpml DTMF relay in global settings. A new SIP dial peer is configured for a third-party application that only supports an in-band DTMF relay.
Which command must an engineer run on the dial peer?
A. dtmf-relay sip-notify
B. dtmf-relay sip-info
C. dtmf-relay rtp-nte
D. no dtmf-relay sip-kpml
Answer: (SHOW ANSWER)

NEW QUESTION: 152
What is an advantage of using Cisco Webex Control HuB?
A. enables the provisioning, administration, and management of Webex services and Webex Hybrid Services
B. brings Video, audio, and web communication together to meet the collaboration needs of the modern workplace
C. provides streamlined communication and collaboration for a hybrid workforce
D. offers easy contact management, centralized administration, and centralized configuration management
Answer: (SHOW ANSWER)
Cisco Webex Control Hub is a cloud-based management platform that enables you to provision, administer, and manage Webex services and Webex Hybrid Services. It provides a single pane of glass for managing all of your Webex services, including Webex Meetings, Webex Teams, and Webex Calling.
Webex Control Hub offers a number of features and benefits, including:
A single pane of glass for managing all of your Webex services
Centralized user management
Simplified provisioning and administration

Real-time analytics and reporting
Enhanced security and compliance
Webex Control Hub is a powerful tool that can help you manage your Webex services more effectively. It is easy to use and provides a number of features and benefits that can help you improve your productivity and efficiency.

NEW QUESTION: 153
A customer wants to conduct B2B video calls with a partner using an on-premises conferencing solution. Which two devices are needed to facilitate this request? (Choose two.)
A. Expressway-C
B. Expressway-E
C. Cisco Unified Border Element
D. Cisco TelePresence Management Suite
E. MGCP gateway
Answer: (SHOW ANSWER)

NEW QUESTION: 154
An administrator must configure the local route group feature on cisco UCM.
Which step will enable this feature?
A. For each route list configure a route group to use as a Local Route Group.
B. For each route pattern, select the Local Route Group as the destination.
C. For each route group, check the box for the Local Route Group feature.
D. For each device pool, configure a route group to use as a Local Route Group for that device pool
Answer: (SHOW ANSWER)
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmfeat/CUCM_BK_ F3AC1C0F_00_cucm-features-services-guide-100/CUCM_BK_F3AC1C0F_00_cucm-features- services-guide-100_chapter_0100110.html

NEW QUESTION: 155
Refer to the exhibit. A company recently deployed Cisco Jabber. Users log in to Jabber by using their email address in a domain named company.com. The users report that they cannot register their Jabber telephony services when working from home unless they use a VPN. An engineer runs the DNS lookup tool in Cisco Expressway-C to troubleshoot the issue. What is the cause of the issue?


A. The TTL value for the company.com is too short.
B. There is a missing SRV record for the company.com domain

C. The company com domain must be resolved only in Expressway-E
D. There must be only response for the company.com domain
Answer: B (LEAVE A REPLY)

NEW QUESTION: 156
What is the purpose of the Disaster Recovery System in Cisco UCM?
A. to create and manage Cisco UCM backup clone servers
B. to provide data backup and restore by using an SFTP server
C. to transparently present failover process events to users
D. to speed up the rebuilding of a crashed Cisco UCM server
Answer: B (LEAVE A REPLY)

NEW QUESTION: 157
How are E.164 called-party numbers normalized on a globalized call-routing environment is Cisco Unified Communications Manger?
A. Normalization is achieved by stripping or translating the called numbers to internally used directory numbers.
B. Call ingress must be normalized before the call being routed.
C. Normalization is not required
D. Normalization is achieved by setting up calling search and partitions at the SIP trunk for PSTN connection.
Answer: A (LEAVE A REPLY)
Normalized called-party numbers: E.164 format with the +prefix is used for external destination.
Therefore, called-number normalization is the result of globalization. Internal directory numbers are used for internal destinations. Normalization is achieved by stripping or translating the called numbers to internally used directory numbers.

NEW QUESTION: 158
An engineer is setting up a new voicemail server, and the requirements state that the integration for call control must be secure. During the configuration the engineer wants to select next Generation Encryption? What is the prerequisite for using this option to ensure an encrypted SIP integration?
A. secure TLS connection and SIP certificates
B. TCP connection using port 5061 with explicit TLS1.2 and signed SIP certificates
C. TCP connection using port 5061 with explicit TLS1.2 and ACME-signed SIP certificates
D. secure TLS connection and Tomcat certificates
Answer: (SHOW ANSWER)

NEW QUESTION: 159
Why would we not include an end user's PC device in a QoS trust boundary?
A. The end user could incorrectly tag their traffic to bypass firewalls.
B. There is no reason not to include an end user's PC device m a QoS trust boundary
C. The end user may incorrectly tag their traffic to be prioritized over other network traffic
D. The end user could incorrectly tag their traffic to advertise their PC as a default gateway.
Answer: C (LEAVE A REPLY)
When it comes to QoS, it is best practice to mark packets as close to the source as possible.
Most devices, such as computers and servers cannot mark their own packets and should not be trusted even if they can. Cisco phones, however, can mark their own packets and can be trusted with the QoS

markings they provide.
Therefore, QoS trust boundaries should be set up so that the switch will trust the QoS markings phones place on their own packets

NEW QUESTION: 160
Which data security feature is included with the Pro Pack for Cisco Webex Control Hub?
A. Hybrid Data Security
B. end-to-end encryption of content
C. end-to-end encryption for meetings
D. encryption in transit
Answer: A (LEAVE A REPLY)

NEW QUESTION: 161
A customer enters no IP domain lookup on the Cisco IOS XE gateway to suppress the interpreting of invalid commands as hostnames Which two commands are needed to restore DNS SRV or A record resolutions? (Choose two.)
A. ip dhcp pool
B. transport preferred none
C. ip dhcp excluded-address
D. ip domain lookup
E. ip dhcp-sip
Answer: B,D (LEAVE A REPLY)

NEW QUESTION: 162
What are two characteristics of QoS Traffic shaping? (Choose two.)
A. Traffic shaping queues excess packets.
B. When the traffic rate equals the configured maximum rate, excess traffic is dropped.
C. Traffic shaping can be applied to inbound traffic on an interface.
D. Traffic shaping can be applied to outbound traffic on an interface.
E. When the traffic rate equals the configured maximum rate, excess traffic is remarked.
Answer: A,D (LEAVE A REPLY)

NEW QUESTION: 163
What is the purpose of configuring locations in a Cisco Webex dial plan?
A. to specify the maximum bandwidth to allocate per Webex call
B. to choose how many Webex data centers to use for redundancy
C. to conserve bandwidth by selecting the closest data center
D. to allow the use of an existing PSTN provider for local calls
Answer: D (LEAVE A REPLY)
In Webex Calling, locations group users or devices by geographical site and are used to manage PSTN access, whether on-premises or cloud connected PSTN. This allows calls to be routed appropriately based on the user's location and the associated PSTN trunk or gateway.
Locations enable linking to a specific PSTN provider or trunk (such as a Local Gateway or Cloud Connected PSTN), so calls from users in that location can be routed through the correct PSTN connection for local calling.

Locations also support inter-site dialing by using location codes and routing prefixes, helping maintain consistent dialing habits and call routing across sites.
While bandwidth management and data center selection are important, these are not the primary purposes of configuring locations in the dial plan for Webex Calling.
Therefore, configuring locations primarily facilitates local PSTN call routing by associating users with the correct PSTN provider or trunk based on their site.

NEW QUESTION: 164

Which statement describes the Maximum Serving Count service parameter of the Cisco TFTP service on Cisco Unified Communications Manager?

A. It specifies the maximum file support by the Cisco TFTP service.

B. It specifies the maximum file counts, in cache as well as in disk, that are supported by the Cisco TFTP service.

C. It specifies the maximum number of TFTP client requests to accept and to serve files at a given time.

D. It specifies the maximum number of files in the TFTP server disk storage.

E. It specifies the maximum number of TFTP client requests to accept and to serve files in a 120- minute window.

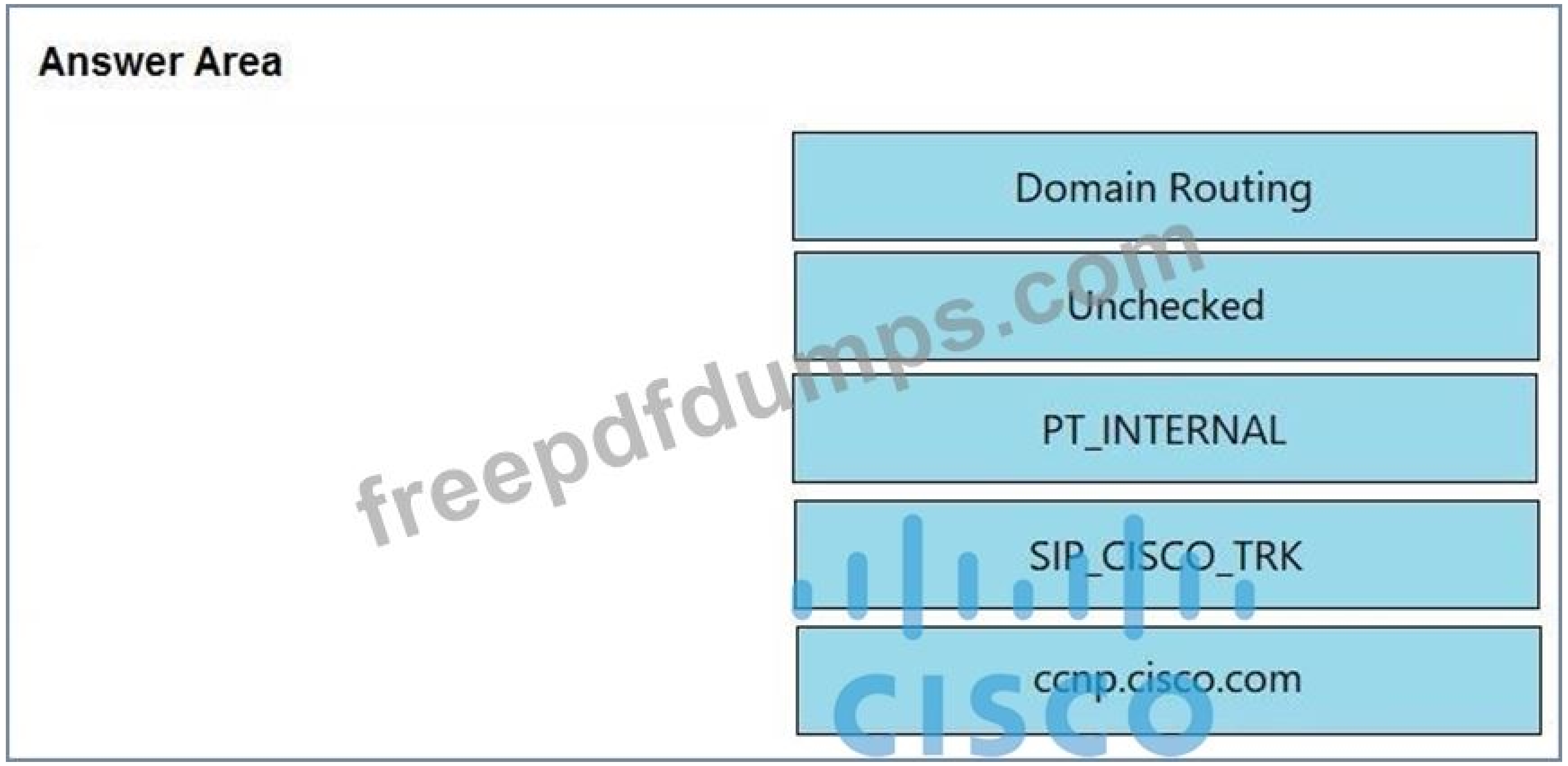
Answer: (SHOW ANSWER)

NEW QUESTION: 165

Drag and Drop Question

In Cisco UCM, an engineer must create a new SIP route pattern to allow ccnp.cisco.com for all internal callers. The internal callers have a calling search space named CSS_INTERNAL that includes the partition named PT_INTERNAL. If allowed, the calls must be routed to a SIP trunk named SIP CISCO_TRK. Drag and drop the values from the left onto the corresponding fields on the right.

Answer:



Explanation:

Domain Routing: Required for routing SIP URI / domain patterns.

Unchecked (IPv4Pattern): This is an FQDN, not an IPv4 address pattern.

PT_INTERNAL: Makes the route pattern reachable by callers whose CSS includes PT_INTERNAL (CSS_INTERNAL).

SIP_CISCO_TRK: Sends the call to the specified SIP trunk.

ccnp.cisco.com: The destination domain to be matched; in this matching set it is placed in the remaining field (Block Pattern) due to the "use each item once" constraint in the question UI.

NEW QUESTION: 166

What is NBAR?

A. It intelligently classifies traffic from different applications

B. It collects traffic flow statistics on routing devices

C. It controls the forwarding of routed packets

D. It allows asymmetric flows that have stateful protocols

Answer: A (LEAVE A REPLY)

NEW QUESTION: 167

Refer 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=21653 ai=127
05:50:14.134: ISDN BR0/1/1 Q921: User RX <- SABMEp sapi=0 tei=0
05:50:14.150: ISDN BR0/1/1 Q921: User TX -> IDREQ ri=19004 ai=127
05:50:14.165: ISDN BR0/1/1 Q921 User RX <- SABMEp sapi=0 tei=0
```

A. interface BRI0/1/1
no ip address
isdn switch-type basic-net3
isdn point-to-point-setup
isdn incoming-voice voice
isdn send-alerting
isdn static-tei 0

B. interface BRI0/1/0
no ip address
isdn switch-type basic-net3
isdn point-to-multipoint-setup
isdn incoming-voice voice
isdn send-alerting
isdn static-tei 0

C. interface BRI0/1/1
no ip address
isdn switch-type basic-net3
isdn incoming-voice voice
isdn send-alerting
isdn static-tei 0

D. interface BRI0/1/1
no ip address
isdn switch-type basic-net3
isdn point-to-multipoint-setup
isdn incoming-voice voice
isdn send-alerting
isdn static-tei 0

Answer: (SHOW ANSWER)

NEW QUESTION: 168

A customer recently deployed a Cisco UCM solution. When employees with numbers 1234567XXX dial 8005556666, they want the caller ID to be the company's toll-free number 8885550000, and for all other calls, it should be the line DN. What is the best way to configure this solution, assuming partitions and CSSs are configured correctly where needed?

A. Configure a translation pattern for 8885550000, calling party transform mask to 8005556666, and calling line ID presentation to "allowed".

B. For all 1234567XXX lines, set External Phone Number Mask and External Presentation Number to 8005556666.

C. Configure a calling party transformation pattern for 1234567XXX, calling party transform mask to 8885550000, and calling line ID presentation to "allowed".

D. Configure a translation pattern for 8005556666, calling party transform mask to 8885550000, and calling line ID presentation to "restricted".
Answer: C (LEAVE A REPLY)
To meet the requirements of displaying 8885550000 as the Caller ID for calls to 8005556666, while using the line DN as the Caller ID for all other calls, a Calling Party Transformation Pattern is the most appropriate solution. This approach ensures:
A calling party transformation pattern can be configured to match the specific calling numbers (e.g., 1234567XXX) and modify the outgoing caller ID to 8885550000 only for calls matching the specified pattern.
Setting the calling line ID presentation to "allowed" ensures the modified caller ID is sent to the called party.
Since the transformation pattern applies only to calls from 1234567XXX to 8005556666, other calls retain their default caller ID (the line DN).

NEW QUESTION: 169
A Cisco UCM location between a client's headquarters and branch office is configured for 512 kbps of audio bandwidth and 384 kbps of video bandwidth. What is the maximum number of G.711 calls allowed through this link?
A. 6
B. 7
C. 8
D. 11
Answer: A (LEAVE A REPLY)
To determine the maximum number of G.711 calls that can be supported over a link with a specific bandwidth allocation, we need to consider the bandwidth consumption of each G.711 call, including overhead.
1. G.711 Codec Bandwidth Requirements: A G.711 call requires 64 kbps for audio payload.
2. IP Packet Overhead: In addition to the audio payload, overhead from RTP (Real-Time Protocol), UDP, and IP headers must be considered. Including overhead, the bandwidth for a G).711 call increases to approximately 80 kbps per call.
3. Audio Bandwidth Available: The audio bandwidth for this link is 512 kbps.
4. Maximum Number of Calls: Divide the available audio bandwidth by the bandwidth required per G).711 call:

$$\text{Number of Calls} = \frac{\text{Audio Bandwidth}}{\text{Bandwidth per Call}} = \frac{512}{80} = 6.4$$

Since partial calls are not possible, the maximum is 6 calls.
By supporting 6 calls at 80 kbps each, the configuration ensures optimal use of the available 512 kbps audio bandwidth while adhering to the QoS limitations.

NEW QUESTION: 170
Which two types of devices are included in the conditionally trusted endpoints category of Cisco UCM? (Choose two.)
A. unsecure PCs
B. Cisco TelePresence endpoints
C. analog gateways
D. Cisco IP phones
E. handheld mobile devices
Answer: B,D (LEAVE A REPLY)

NEW QUESTION: 171
A network engineer must convert an inbound number 12345551111 when it reaches Cisco UCM and then converts it to an internal extension 3022, so it can ring to a specific desk. Which action meets this requirement?
A. Create a route pattern 9.1XXXXXXXXXX and assign it to the trunk.

B. Create translation pattern 12345551111, called party transform mask 3022.
C. Create a route group and route list and assign it to the translation pattern.
D. Change the partition and calling search space on the DID for the specific desk.
Answer: (SHOW ANSWER)
A Translation Pattern in Cisco UCM is used to modify the dialed number (called party number) before routing the call. By configuring a translation pattern that matches the incoming number 12345551111 and setting the called party transform mask to 3022, incoming calls to 12345551111 will be redirected to the internal extension 3022.
Reference:
https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_1_3/ccmcfg/b03txpat.html

NEW QUESTION: 172
An administrator set up Cisco UCM LDAP Directory and already synced the users to Cisco UCM from the Active Directory, but they forgot to set up feature group templates when. doing the import. Which action resolves the issue?
A. Add another LDAP Directory in Cisco UCM, configure feature group templates, and do a manual sync for that new LDAP Directory.
B. Use the Bulk Administration Tool to update those users with the feature group template.
C. Configure feature group templates in the Cisco UCM LDAP Directory and do a manual sync.
D. Configure feature group templates in the Cisco UCM LDAP Directory and configure the resync interval.
Answer: B (LEAVE A REPLY)
To assign feature group templates to users who were previously synchronized from Active Directory (AD) into Cisco Unified Communications Manager (UCM) without these templates, you can utilize the Bulk Administration Tool (BAT) to update the existing user records.
1. Export User Data:
- Navigate to Bulk Administration > Users > Export Users in the Cisco UCM Administration interface.
- Select the users you wish to update and export their data to a CSV file.
2. Modify the CSV File:
- Open the exported CSV file in a spreadsheet editor.
- Locate the column corresponding to the Feature Group Template.
- Assign the desired feature group template to each user by entering the appropriate template name in this column.
- Save the modified CSV file.
3. Upload the Modified CSV File:
- Return to the Cisco UCM Administration interface and go to Bulk Administration > Upload/Download Files.
- Upload the modified CSV file.
4. Update Users:
- Navigate to Bulk Administration > Users > Update Users > Custom File.
- Select the uploaded CSV file.
- Choose the appropriate update options to apply the feature group templates to the selected users.
- Execute the update process.
By following these steps, you can efficiently assign feature group templates to users who were initially imported without them, ensuring that all users have the necessary configurations.
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/bat/14SU1/cucm_b_bulk-administration-guide-14SU1/cucm_b_bulk-administration-guide-1251su2_chapter_010010.html

NEW QUESTION: 173

On a cisco catalyst switch, which command is required to send CDP packets on a switch port that configures a cisco IP phone to transmit voice traffic in 802.1q frames tagged with the voice VLAN ID 221?

A. Device(config-if)# switchport access vlan 221

B. Device(config-if)# switchport trunk allowed vlan 221

C. Device(config-if)# switchport vlan voice 221

D. Device(config-if)# switchport voice vlan 221

Answer: D (LEAVE A REPLY)

https://www.cisco.com/c/en/us/td/docs/switches/lan/catalyst2960/software/release/12-2_40_se/configuration/guide/scg/swvoip.pdf

NEW QUESTION: 174

An 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.com

B. sip_udp.abc.com 60 IN SRV 3 60 5060 cube2.abc.com

C. sip_udp.abc.com 60 IN SRV 60 1 5060 cube1.abc.com

D. sip_udp.abc.com 60 IN SRV 2 60 5060 cube1.abc.com

E. sip_udp.abc.com 60 IN SRV 1 60 5060 cube1.abc.com

Answer: A,E (LEAVE A REPLY)

- 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 dot

NEW QUESTION: 175

AN administrator needs to help remote employee make a free call to an international destination.

The administrator calls the employee, then conferences in the international party. The administrator drops the call, and the employee and the international party continue their conversation. Which action prevents this type of toll fraud in the Cisco USM?

A. Set service parameter "Advanced Ad Hoc Conference" to FALSE

B. Set service parameter "Drop Ad Hoc Conference" to "Do not allow outside parties "

C. Set service parameter "Drop Ad Hoc Conference" to "When Conference Controller leaves"

D. Set service parameter "Advanced Ad Hoc Conference" to 2.

Answer: C (LEAVE A REPLY)

NEW QUESTION: 176

Refer to the exhibit. The SDP offer/answer has been completed successfully but there is no DTMF when users press keys.

What is the cause of the issue?

```
INVITE sip:2002@10.10.10.10:5060 SIP/2.0
[..truncated..]
v=0
o=UAC 6107 7816 IN IP4 10.10.10.11
s=SIP Call
c=IN IP4 10.10.10.11
t=0 0
m=audio 8190 RTP/AVP 18 110
c=IN IP4 10.10.10.11
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:110 telephone-event/8000
a=fmtp:110 0-16
a=ptime:20

SIP/2.0 200 OK
[..truncated..]
v=0
o=UAS 4692 9609 IN IP4 10.10.10.10
s=SIP Call
c=IN IP4 10.10.10.10
t=0 0
m=audio 8056 RTP/AVP 18
c=IN IP4 10.10.10.10
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
```

A. Payload type 110 was negotiated rather than type 101.

B. DTMF was negotiated properly in these messages.

C. G.729 rather than G.711ulaw was negotiated.

D. DTMF was not negotiated on the call.

Answer: D (LEAVE A REPLY)
It is offered 18 (g729) and 110 (type-event), however the answer only contains 18. Therefore, DTMF was not negotiated.

NEW QUESTION: 177

Which layer is the optimal trust boundary for an untrusted Cisco endpoint?

A. access

B. core

C. WAN

D. distribution

Answer: A (LEAVE A REPLY)

The access layer is the optimal trust boundary for an untrusted Cisco endpoint because it is the first layer in the network where devices connect. This layer is responsible for marking and classifying traffic for Quality of Service (QoS) purposes. Establishing a trust boundary at the access layer ensures that untrusted endpoints cannot alter QoS markings further upstream.
The core and distribution layers are deeper into the network, where traffic aggregation occurs. By this point, traffic should already be marked and trusted. Moving the trust boundary further up the network increases the risk of untrusted devices introducing incorrect QoS markings.
The WAN is an external connection to another network. The trust boundary must already be defined before traffic reaches this point to ensure consistent QoS treatment across the network.

NEW QUESTION: 178

How are network devices monitored in a collaboration network?

A. The Cisco Discovery Protocol table is shared among devices.

B. Ping Sweep reports "unmanaged' state devices.

C. System logs are collected in a Cisco Prime Collaboration Server.
D. Simple Network Managed Protocol is enabled on each device to poll specific values periodically.
Answer: D (LEAVE A REPLY)
The device details (such as model number, version, and so on) are collected from Prime Collaboration inventory and also by polling the endpoints using SNMP or HTTP.

NEW QUESTION: 179
What are two requirements for deploying Directory Connector in Cisco Webex? (Choose two.)
A. Active Directory with a forest functional level of 1 or higher
B. TCP port 443 open from Directory Connector to the internet
C. single sign-on between Directory Connector and Active Directory
D. support for TLS 1.2 between Directory Connector and Webex Control Hub
E. .NET Framework 3.4 installed on the Directory Connector server
Answer: B,D (LEAVE A REPLY)

NEW QUESTION: 180
Refer 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
ccm-manager config server 192.168.1.100
ccm-manager mgcp

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


A. Device(config)# mgcp enable
B. Device(config)# ccm-manager enable
C. Device (config)# com-manager active
D. Device (config)# mgcp
Answer: D (LEAVE A REPLY)
https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified- communications-manager-callmanager/42105-vg200-cfg.html

NEW QUESTION: 181
An engineer with troubleshoots poor voice quality on multiple calls. After looking at packet captures, the engineer notices high levels of jitter.
Which two areas does the engineer check to prevent jitter? (Choose two.)
A. Voice packets are classified and marked.
B. All devices use wired connections instead of wireless connections.
C. The network meets bandwidth requirements.
D. Cisco UBE manages voice traffic, not data traffic.
E. MTP is enable on the sip trunk to cisco Unified Border Element.
Answer: A,C (LEAVE A REPLY)
https://www.cisco.com/c/en/us/support/docs/voice/voice-quality/20371-troubleshoot-qos- voice.html

NEW QUESTION: 182
What is required for Cisco UCM to accept SIP calls with a URI in the format of "sip 2001@cucmpub.cisco.com"?
A. Define Cluster Fully Qualified Domain Name under Servers in Cisco UCM.
B. Change the Destination Address to a Fully Qualified Domain Name on the SIP trunk.
C. Set the SIPS URI Handling to True in CallManager Service Parameters.
D. Define Cluster Fully Qualified Domain Name in Enterprise Parameters.
Answer: D (LEAVE A REPLY)
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_ b_system-configuration-guide-1201/cucm_b_system- configuration-guide-1201_chapter_0101.html

NEW QUESTION: 183
What happens when a Cisco IP phone loses connectivity to the duster during an active call?
A. The call continues to be active, but features like transfer or hold do not work.
B. The call continues and all features work.
C. The call drops immediately.
D. The call drops after missing two keepalives from Cisco UCM.
Answer: A (LEAVE A REPLY)

Scenario	Call Preservation Handling
Cisco Unified Communications Manager fails.	A Cisco Unified Communications Manager failure causes the call-processing function for all calls that were set up through the failed Cisco Unified Communications Manager to be lost. The affected devices recognize that their current Cisco Unified Communications Manager failed. Similarly, the other Cisco Unified Communications Managers in the cluster detect the Cisco Unified Communications Manager failure. Cisco Unified Communications Manager maintains affected active calls until the end user hangs up or until the devices can determine that the media connection has been released. Users cannot invoke any call-processing features for calls that are maintained as a result of this failure.
Communication failure occurs between Cisco Unified Communications Manager and device.	When communication fails between a device and the Cisco Unified Communications Manager that controls it, the device recognizes the failure and maintains active connections. The Cisco Unified Communications Manager recognizes the communication failure and clears call-processing entities that are associated with calls in the device where communication was lost. The Cisco Unified Communications Managers still maintain control of the surviving devices that are associated with the affected calls. Cisco Unified Communications Manager maintains affected active calls until the end user hangs up or until the devices can determine that the media connection has been released. Users cannot invoke any call-processing features for calls that are maintained as a result of this failure.

https://community.cisco.com/t5/unified-communications/call-in-progress-when-one-cucm-go- down/td-p/2453662

NEW QUESTION: 184

```
ROUTER-1(config)# policy-map LLQ_POLICY
ROUTER-1(config-pmap)# class VOICE
ROUTER-1(config-pmap-c)# bandwidth 170
ROUTER-1(config-pmap-c)# exit
ROUTER-1(config-pmap)# class VIDEO
ROUTER-1(config-pmap-c)# bandwidth remaining percent 30
ROUTER-1(config-pmap-c)# exit
ROUTER-1(config-pmap)# exit
```

ent to implement LLQ for voice traffic. Which change must the engineer make in the configuration?

A. bandwidth 170 to LLQ 170

B. bandwidth 170 to priority 170

C. bandwidth 170 to reserve 170

D. bandwidth 170 to percent 170

Answer: B (LEAVE A REPLY)

https://www.cisco.com/c/en/us/td/docs/ios/qos/configuration/guide/12_2sr/qos_12_2sr_book/llq_f or_ipsec.html

NEW QUESTION: 185

If a trust boundary is placed on a Cisco IP phone, what occurs with the packets that come to the phone from a PC?

A. The phone does not mark any packets, and the access switch sets up the CoS values.
B. The phone sets/resets the CoS field to 0; voice signaling is set to 3; and voice traffic is set to 5.
C. The phone sets/resets the CoS field to 3 for voice signaling and 5 for voice traffic.
D. QoS is performed at the distribution layer, and packets are marked at this boundary.
Answer: B (LEAVE A REPLY)

NEW QUESTION: 186
In which location does an administrator look to determine which subscriber the phone registers to if loses registration with the current Cisco UCM subscriber?
A. On Cisco UCM Administrator page system > Device Pool > Cisco UCM group
B. On Cisco UCM Administration Page Device > Phone > Phone Configuration page
C. On Cisco UCM Administrator Page server > Cisco UCM
D. On Cisco UCM Administrator page system > Enterprise Parameters
Answer: A (LEAVE A REPLY)

NEW QUESTION: 187
Refer to the exhibit. An administrator is configuring a new phone on a Cisco UCM cluster and the phone failed to register. The Cisco UCM cluster is in non-secure mode. Which action resolves the issue?

```
9:41:32a TFTP Timeout : SEP0CDEA7824C58.cnf.xml
9:45:15a No Trust List Installed
9:46:51a TFTP Timeout : SEP0CDEA7824C58.cnf.xml
9:50:34a No Trust List Installed
9:52:10a TFTP Timeout : SEP0CDEA7824C58.cnf.xml
9:52:30a File Not Found : SEP0CDEA7824C58.cnf.xml
9:52:31a XMLDefault.cnf.xml
```

A. Ensure that DHCP scope has option 150 configured.
B. Manually configure voice VLAN on the phone.
C. Delete phone ITL file.
D. Update phone configuration in Cisco UCM with the correct MAC address.
Answer: C (LEAVE A REPLY)
The ITL file is used for certificate-based trust between the phone and the Cisco UCM TFTP server. If the ITL file is invalid or missing, the phone cannot securely communicate with the TFTP server to download its configuration.
To resolve this, the administrator must manually delete the ITL file from the phone. This forces the phone to request a new ITL file from the TFTP server during its next boot.
Steps to Delete the ITL File:
1. On the phone, go to Settings > Security Configuration > Trust List.
2. Select ITL File and delete it.
3. Restart the phone to allow it to re-download the correct ITL file and configuration.

NEW QUESTION: 188
Which dial plan function restricts calls that are made by a lobby phone to internal extensions only?
A. path selection

B. endpoint addressing
C. calling privileges
D. manipulation of dialed destination
Answer: C (LEAVE A REPLY)

NEW QUESTION: 189
What are two features of Cisco Expressway that the customer gets if Expressway-C and Expressway-E are deployed? (Choose two.)
A. highly secure firewall-traversal technology to extend organizational reach
B. complete endpoint registration and monitoring capabilities for devices that are local and remote
C. session-based access to comprehensive collaboration for remote workers, without the need for a separate VPN client
D. additional visibility of the edge traffic in an organization
E. utilization and adoption metrics of all remotely connected devices
Answer: (SHOW ANSWER)
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/expressway/admin_guide/X14-0-2/exwy_b_cisco-expressway-administrator-guide-x1402/exwy_m_introduction.html

NEW QUESTION: 190
When Mobile and Remote Access is used, which feature is available to Cisco Jabber clients outside an enterprise?
A. Cisco TelePresence
B. visual voicemail
C. VPN links
D. many-to-one traversal connections
Answer: B (LEAVE A REPLY)
When Mobile and Remote Access (MRA) is used with Cisco Jabber, remote users can access Cisco Unified Communications services without a VPN. This includes features like voice and video calling, instant messaging, presence, and visual voicemail.
Visual voicemail allows users to see a list of their voicemails, listen to them, and manage them directly from the Cisco Jabber client.
MRA works through Cisco Expressway-E and Expressway-C, ensuring secure remote connectivity while maintaining enterprise-level security and compliance.

NEW QUESTION: 191
As a voice engineer, which two recommendations do you make to your company to optimize Cisco Unified Communications Manager configuration to reduce the number of toll fraud incidents? (Choose two.)
A. Inbound CSS on any gateway typically should have access to internal destinations and PSTN destinations.
B. Classify all route patterns as on-net or off-net and prohibit off-net call transfers in Cisco Unified CM Service parameters.
C. Classify all route patterns as on-net and prohibit on-net to on-net call transfers in Cisco Unified CM service parameters.
D. Classify all route pattern as off-net and prohibit off-net to off-net call transfers in Cisco Unified CM service parameters.
E. Inbounds CSS on any gateway typically should have access to internal destinations only and not PSTN destinations.
Answer: D,E (LEAVE A REPLY)

NEW QUESTION: 192
Where is it recommended to enforce the QoS trust boundary on a network?
A. LAN access ingress and the WAN access ingress
B. LAN access egress
C. LAN access ingress

D. LAN access egress and the WAN access egress
Answer: C (LEAVE A REPLY)
The QoS trust boundary should be enforced at the LAN access ingress so that traffic markings are trusted only from known and controlled devices, such as IP phones, before entering the network. This prevents untrusted endpoints from improperly marking traffic and ensures consistent QoS treatment throughout the network.

NEW QUESTION: 193
How does an administrator ensure compliance with ITU-T G.114 latency requirements when deploying voice and video applications over routes that include a geostationary satellite?
A. The satellite is 35,786 km (22,236 miles) above sea level (from the equator), so an engineer will not get below 150 ms latency in one direction and will need to manage the expectation of the user.
B. For voice and video applications over satellite, Cisco recommends using WAN optimization compression, which will save space or time of the transmission. The latency will then be below the recommended latency threshold for voice.
C. Using G.729 voice codec Instead of G.711 voice codec will reduce the packet size and the latency to stay within the recommended threshold.
D. The application will work because it will be below the recommenced latency requirement If an engineer uses compressed headers to speed up the voice packets.
Answer: A (LEAVE A REPLY)

NEW QUESTION: 194
For Webex Calling, how are outbound transfers and forwards restricted for a particular call type?
A. together on a site-by-site basis
B. independently on an organization-by-organization basis
C. together on an organization-by-organization basis
D. independently on a site-by-site basis
Answer: D (LEAVE A REPLY)
In Webex Calling, outbound call transfer and call forward permissions are configured independently at the site (location) level, allowing granular control of call behavior based on location-specific policies.

NEW QUESTION: 195
What are the predefined call handlers in Cisco Unity Connection?
A. opening greeting, operator, and goodbye
B. opening greeting, welcome, and default system
C. greetings, operator, and closed
D. caller input, greetings, and transfer
Answer: (SHOW ANSWER)
https://www.cisco.com/en/US/docs/voice_ip_comm/connection/1x/administration/guide/acm030.ht ml

NEW QUESTION: 196
A user forwards a corporate number to an international number.
What are two methods to prevent this forwarded call? (Choose two.)
A. Configure all forced code on all router.
B. Block international dial patterns in the SIP trunk CSS.
C. Configure a Forced Authorization Code on the international router pattern.
D. Set the call Classification to OnNet for the international route pattern.
E. Set call forward All CSS to restrict international dial patterns.
Answer: C,E (LEAVE A REPLY)

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmead m/cmetrans.html

NEW QUESTION: 197
An engineer must deploy the Cisco Webex app to a shared Windows Virtual Desktop Infrastructure environment in the C:\Program Files folder. Which command line argument must be run when running the installer?
A. ALLUSERS=1
B. ROAMINGENABLED=1
C. ENABLEVDI=2
D. INSTYALLWV2=1
Answer: (SHOW ANSWER)

NEW QUESTION: 198
Refer to the exhibit. In this cisco UCM setup configured for EARLY offer, what is the codec preference line in the initial SIP INVITE SDP?

A. m=audio 16444 RTP/AVP 9 18 101
B. m=audio 16444 RTP/AVP 0 8 18 101
C. m=audio 16444 RTP/AVP 18 101
D. m=audio 16444 RTP/AVP 9 0 8 18 101
Answer: C (LEAVE A REPLY)
40kbps will not allow G.711 & G.722 to be advertised as they have bandwidth of 64kbps, only G).729 will be allowed.
https://en.m.wikipedia.org/wiki/RTP_payload_formats

NEW QUESTION: 199

What is a function of a System Call Handler in Cisco Unity Connections?
A. Filter incoming calls to prevent robocalls.
B. Greet callers with recorded prompts.
C. Identify high-priority calls.
D. Place calls in a queue.
Answer: B (LEAVE A REPLY)

NEW QUESTION: 200
Which type of traffic must be marked with the CS4 DiffServ value according to Cisco design guidance?
A. Cisco TelePresence calls
B. streaming video calls
C. SIP signaling
D. video calls of Cisco Jabber
Answer: (SHOW ANSWER)
According to Cisco QoS design guidance, CS4 is used for call signaling traffic associated with video conferencing systems. SIP signaling falls into this category because it controls call setup, modification, and teardown and must be reliably delivered with higher priority than best-effort traffic but below real-time media.

NEW QUESTION: 201
An engineer is going to redesign a network, and while looking at the QoS configuration, the engineer sees that a portion of the network is marked with AF42.
Which type of traffic is marked with this tag?
A. voice
B. streaming video
C. signaling
D. video conference
Answer: D (LEAVE A REPLY)
https://www.cisco.com/c/en/us/td/docs/solutions/CVD/Collaboration/hybrid/12x/hybcvd/bwm.html

NEW QUESTION: 202
Which Cisco IM and Presence service handles failover and state changes in the cluster?
A. Cisco Server Recovery Manager
B. XCP router
C. Cisco XCP Connection Manager
D. XCP Sync Agent
Answer: A (LEAVE A REPLY)

NEW QUESTION: 203
A high-speed network is often configured with a five-class QoS model. Which classes are used in the model?
A. voice, video, signaling critical data, and best-effort
B. real-time, call-signaling, critical data, best-effort, and scavenger
C. real-time, signaling, critical data, best-effort, and drop-class
D. call-signaling, real-time, critical data, best-effort, and drop-class
Answer: (SHOW ANSWER)

https://www.cisco.com/en/US/technologies/tk543/tk759/technologies_white_paper0900aecd8029
5a9b.pdf

NEW QUESTION: 204

Which two statements are about location CAC on Cisco UCM? (Choose two.)

A. It stands for Call Allocation Control.

B. It allows for bandwidth restrictions between locations.

C. It allows for codec restrictions between locations.

D. It allows for devices making call restrictions between locations based on numbers.

E. It allows for QoS-based restrictions between locations.

Answer: (SHOW ANSWER)

Location Call Admission Control (CAC) in Cisco Unified Communications Manager (UCM) is a mechanism to manage and limit the bandwidth available for voice and video calls between locations. This helps ensure that the available bandwidth is not oversubscribed, preserving call quality.

Bandwidth Restrictions Between Locations:

CAC enforces bandwidth limitations by defining the maximum available bandwidth for voice and video calls between locations. Once the bandwidth is consumed, additional calls are either blocked or rerouted.

Codec Restrictions Between Locations:

By limiting bandwidth, CAC indirectly enforces codec selection, as certain codecs (like G.711 or G).722) consume more bandwidth than others (like G.729). This helps control which codecs are used based on the available bandwidth.

Reference: https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube- book/voi-cube-call-admission-control.pdf

NEW QUESTION: 205

What are two QoS requirements for VoIP traffic? (Choose two.)

A. Voice traffic must be marked to DSCP AF41.

B. Voice traffic must be marked to DSCP EF.

C. One-way latency must be no more than 200 ms.

D. Average one-way jitter is greater than 50 ms.

E. Loss must be no more than 1 percent.

Answer: B,E (LEAVE A REPLY)

NEW QUESTION: 206

Refer to the exhibit. An engineer is troubleshooting why PSTN phones are not receiving the caller's name when called from a remote Cisco UCM site.

An ISDN PRI connection is being used to reach the PSTN.

What must the administrator select to resolve the issue?

```
    Bearer Capability i = 0x8090A2
        Standard = CCITT
        Transfer Capability = Speech
        Transfer Mode = Circuit
        Transfer Rate = 64 kbit/s
    Channel ID i = 0xA98388
        Exclusive, Channel 8
    Calling Party Number i = 0x2181, '5125551212'
        Plan: ISDN, Type:National
    Called Party Number i = 0xA1, '2145551212'
        Plan:ISDN, Type:National
Mar 1 02:35:37: ISDN Se0/1/1:23 Q931: RX <- CALL_PROC pd = 8 callref = 0x809A
    Channel ID i = 0xA98388
        Exclusive, Channel 8

interface Serial0/1/1:23
 description PRI Circuit to R1
 no ip address
 encapsulation hdlc
 isdn switch-type primary-ni
 isdn protocol-emulate network
 isdn incoming-voice voice
 no cdp enable
```


A. isdn supp-service name calling
B. isdn outgoing display-ie
C. isdn enable did
D. isdn send display le
Answer: (SHOW ANSWER)
https://www.cisco.com/c/en/us/td/docs/ios/voice/command/reference/vr_book/vr_i2.html

NEW QUESTION: 207
What dialed number match this cisco UCM route pattern?
1[23]XX
A. 12300 through 12399 only
B. 1230 through 1239 only
C. 1200 through 1300 only
D. 1200 through 1399 only
Answer: (SHOW ANSWER)

NEW QUESTION: 208
What is a capability of the call forwarding feature in a Cisco Webex dial plan?

A. It forwards calls when a line is busy by enabling the call waiting feature.
B. It forwards calls to the voicemail of a mobile phone when there is no answer.
C. It provides the ability to forward a Webex meeting to another user.
D. It chooses a tone to be played when forwarding all calls.
Answer: C (LEAVE A REPLY)

NEW QUESTION: 209
Which capability is supported by Cisco Discovery Protocol but not by LLDP-MED?
A. LAN speed and duplex discovery
B. network policy discovery
C. location identification discovery
D. power discovery
E. trust extension
Answer: E (LEAVE A REPLY)
Cisco Discovery Protocol provides an additional capability not found in LLDP-MED that allows the switch to extend trust to the phone. In this case, the phone is now trusted to mark the packets received on the PC port accordingly. This feature can be used to off-load the switch because now it does not need to police the information being received from the phone.

NEW QUESTION: 210
An engineer must extend the corporate phone system to mobile users connecting through the internet with their own devices. One requirement is to keep this as simple as possible for end users. Which infrastructure element achieves these goals?
A. Cisco Expressway-C and Expressway-E
B. Cisco Unified Instant messaging and presence
C. Cisco Unified Border Element
D. Cisco Express Mobility
Answer: A (LEAVE A REPLY)
Cisco Unified Border Element (CUBE) provides connectivity between the enterprise network and service provider network.

NEW QUESTION: 211
An administrator configures international calling on a Cisco UCM cluster and wants to minimize the number of route patterns that are needed.
Which route pattern enables the administrator to match variable-length numbers?
A. 9.011@
B. 9.011#
C. 9.011*
D. 9.011!
Answer: D (LEAVE A REPLY)
https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/3_1_2/ccmcfg/b03spchr.html

!	The exclamation point (!) wildcard matches one or more digits in the range 0 through 9.	The route pattern 91! routes or blocks all numbers in the range 910 through 9199999999999999999999.
?	The question mark (?) wildcard matches zero or more occurrences of the preceding digit or wildcard value.	The route pattern 91X? routes or blocks all numbers in the range 91 through 9199999999999999999999.
+	The plus sign (+) wildcard matches one or more occurrences of the preceding digit or wildcard value.	The route pattern 91X+ routes or blocks all numbers in the range 9100 through 9199999999999999999999.
[ ]	The square bracket ([ ]) characters enclose a range of values.	The route pattern 813510[012345] routes or blocks all numbers in the range 8135100 through 8135105.
-	The hyphen (-) character, used with the square brackets, denotes a range of values.	The route pattern 813510[0-5] routes or blocks all numbers in the range 8135100 through 8135105.
^	The circumflex (^) character, used with the square brackets, negates a range of values. It must be the first first character following the opening bracket ([). Each route pattern can have only one ^ character.	The route pattern 813510[^0-5] routes or blocks all numbers in the range 8135106 through 8135109.
.	The dot (.) character is used as a delimiter to separate the Cisco CallManager access code from the directory number. Use this special character, with the discard digits instructions, to strip off the Cisco CallManager access code before sending the number to an adjacent system. Each route pattern can have only one . character.	The route pattern 9.@ identifies the initial 9 as the Cisco CallManager access code in an NANP call.
*	The asterisk (*) character can provide an extra digit for special dialed numbers.	You can configure the route pattern *411 to provide access to the internal operator for directory assistance.

NEW QUESTION: 212

In an eleven-class queueing model following realtime, best effort, and scavenger queuing rules, what recommended percentage of bandwidth must be allocated for interactive video?

A. 10%

B. 12%

C. 15%

D. 20%

Answer: C (LEAVE A REPLY)

https://www.ciscopress.com/articles/article.asp?p=357102&seqNum=7#:~:text=If%20Scavenger%20and%20Bulk%20traffic,illustrated%20in%20Figure%2012%2D10

NEW QUESTION: 213

Which configuration setting is used to configure PLAR on a SIP phone?
A. SIP profile
B. SIP dial rule
C. common phone profile
D. outbound rollover
Answer: B (LEAVE A REPLY)
A SIP Dial Rule in CUCM allows administrators to define specific dialing behaviors for SIP phones. By creating a SIP Dial Rule with a PLAR configuration, you can set a phone to automatically dial a predetermined number when the handset goes off-hook.
Configuration Steps:
1. Create a SIP Dial Rule:
- Navigate to Call Routing > Dial Rules > SIP Dial Rules in the CUCM Administration interface.
- Click Add New.
- Provide a description for the dial rule.
- Click Add Plar to define the PLAR behavior.
- Specify the line button (e.g., Button 1) and the destination number for the PLAR.
- Save the configuration.
2. Assign the SIP Dial Rule to the SIP Phone:
- Navigate to Device > Phone and select the desired SIP phone.
- In the phone configuration page, locate the SIP Dial Rules field.
- Select the SIP Dial Rule you created from the dropdown menu.
- Save and apply the configuration.
By following these steps, the SIP phone will automatically dial the specified number when the handset is lifted, implementing the PLAR functionality.

NEW QUESTION: 214
Which Cisco Unified Communications Manager service parameter should be enabled to disconnect a multiparty call when the call initiator hangs up?
A. Block OffNet to OffNet Transfer
B. Drop Ad Hoc Conference
C. H.225 Block Setup destination
D. Enterprise Feature Access code for conference
Answer: B (LEAVE A REPLY)
Cisco Unified Communications Manager Administration provides the clusterwide service parameter, Drop Ad Hoc Conference, to allow the prevention of toll fraud (where an internal conference controller disconnects from the conference while outside callers remain connected).
The service parameter settings specify conditions under which an ad hoc conference gets dropped.
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmsys/CUCM_BK_SE5FCFB6_00_cucm-system-guide-100/CUCM_BK_SE5FCFB6_00_cucm-system-guide-100_chapter_011000.html#CUCM_TK_DFC66444_00

NEW QUESTION: 215
Which congestion management mechanism must be used to reduce VoIP packet drops?
A. policy-based routing
B. route maps
C. digital signal processor transcoders

D. low latency queueing

Answer: D (LEAVE A REPLY)

Low Latency Queueing provides a strict priority queue for real-time voice traffic, ensuring VoIP packets are transmitted first during congestion, which significantly reduces packet drops and preserves voice quality.

NEW QUESTION: 216

In a Dual Video Queue approach, what is the purpose of the CS4 class for Cisco video devices?

A. Create a simpler provisioning with one CBWFQ for desktop and immersive video traffic

B. Immersive video traffic is marked with CS4 and placed in a CBWFQ separate from desktop video

C. Desktop video traffic needs strict priority queuing marked with CS4 class

D. Enforce a strict priority queueing for both desktop and immersive video traffic

Answer: B (LEAVE A REPLY)

NEW QUESTION: 217

What is a capability of configuring the Incoming call permissions in a Cisco Webex dial plan?

A. to allow users to receive collect calls

B. to allow chargeable directory assistance calls

C. to disable operator assistance calls

D. to stop users from making collect calls

Answer: A (LEAVE A REPLY)

NEW QUESTION: 218

An engineer implements a new Cisco UCM based telephony system per these requirements:

- The local Ethernet bandwidth is sized based on the total bandwidth
per call
- A G.736 codec is used.
- The bit rate is 64 kbps
- The codec sample interval is 10 ms.
- The voice payload size is 160 bytes per 20 ms.

What should the size of the Ethernet bandwidth be per call?

A. 31.2 kbps

B. 38.4 kbps

C. 55.2 kbps

D. 87.2 kbps

Answer: D (LEAVE A REPLY)

All details match with G711 protocol, for which the calculation is :

(160+(40+18))/160 x 64 =87.2 kbps

https://www.cisco.com/c/en/us/support/docs/voice/voice-quality/7934-bwidth-consume.html Codec Information Bandwidth Calculations Codec & Bit Rate (Kbps) Codec Sample Size (Bytes) Codec Sample Interval (ms) Mean Opinion Score (MOS) Voice Payload Size (Bytes) Voice Payload Size (ms) Packets Per Second (PPS) Bandwidth MP or FRF.12 (Kbps) Bandwidth w/cRTP MP or FRF.12 (Kbps) Bandwidth Ethernet (Kbps) G).711 (64 Kbps) 80 Bytes 10 ms 4.1 160 Bytes 20 ms 50 82.8 Kbps 67.6 Kbps 87.2 Kbp

NEW QUESTION: 219

Refer 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; privacy=off
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, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1565089565
Contact: <sip:+ 14088335000@172.16.2.143:5061;transport=tls>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 68
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:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
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: B (LEAVE A REPLY)

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.

NEW QUESTION: 220

Which two SNMP services are available in Cisco UCM? (Choose two.)
A. SYS-MIB
B. HOST-RESOURCES MIB
C. APPL-MIB
D. CISCO-CCM-MIB
E. RESOURCES-MIB
Answer: (SHOW ANSWER)
Cisco Unified Communications Manager supports the HOST-RESOURCES MIB to provide information about system resources such as CPU, memory, and storage usage, and the CISCO- CCM-MIB to expose Cisco Unified Communications Manager-specific operational and performance data for monitoring and management via SNMP.

NEW QUESTION: 221
An administrator recently upgraded a Cisco Webex DX80 through its web interface but discovered the next morning that the unit had reverted to its previous version. What must the administrator do to prevent this from happening again?
A. Set the prepare cluster for rollback to pre-8.0 enterprise parameter to true.
B. Assign a universal device template to the phone.
C. Confirm the phone load name in the phone configuration.
D. Assign a phone security profile with secure SIP.
Answer: C (LEAVE A REPLY)

NEW QUESTION: 222
An engineer is asked to implement on-net/off-net call classification in Cisco UCM.
Which two components are required to implement this configuration? (Choose two)
A. SIP trunk
B. CTI route point
C. route group
D. route pattern
E. SIP route patterns
Answer: A,D (LEAVE A REPLY)
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_5_2/ccmfeat/CUCM_BK_ C3A84B33_00_cucm-feature-configuration- guide_1052/CUCM_BK_C3A84B33_00_cucm- feature-configuration-guide_chapter_0100010.html

NEW QUESTION: 223
Drag and Drop Question
Drag and drop the steps from the left into order on the right to activate self-provisioning services in Cisco UCM.

Answer Area

Configure CTI route point.	Step 1
Configure a self-provisioning system.	Step 2
Assign a directory number to the CTI route point.	Step 3
Enable auto-registration.	Step 4
Configure an application user.	Step 5
Activate self-provisioning services.	Step 6

Answer:

Answer Area

Configure an application user.
Configure CTI route point.
Assign a directory number to the CTI route point.
Configure a self-provisioning system.
Activate self-provisioning services.
Enable auto-registration.

NEW QUESTION: 224

What is a capability of a Cisco Unity Connection system call handler?

A. control transferred outgoing calls that can be dialed from the Cisco Unity Connection system

B. collect information from callers by playing prerecorded questions and then recording the answers
C. answer calls, greet callers with Interactive Voice Response options to route calls, and take messages
D. provide access to a corporate directory that callers can use to reach Cisco Unity Connection users that have mailboxes
Answer: C (LEAVE A REPLY)
A Cisco Unity Connection system call handler provides the capability to answer incoming calls, interact with callers using prerecorded greetings or menu options, and route calls based on caller input. It can also take voicemail messages if necessary.
Key Functions of a System Call Handler:
- Answering Calls: The call handler answers calls based on predefined rules, such as specific schedules (e.g., business hours, after hours).
- Interactive Voice Response (IVR): The call handler can offer options like "Press 1 for Sales, Press 2 for Support," allowing the caller to route their own call using touch-tone input.
- Taking Messages: If the destination is unavailable or if routing fails, the call handler can take messages and store them in Unity Connection mailboxes.
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/14/administration/guide/b_14c ucsag/b_14cucsag_chapter_0111.html

NEW QUESTION: 225
A collaboration engineer configures Global Dial Plan Replication for multiple Cisco UCM clusters.
The local cluster acts as the hub cluster and the remaining clusters act as spoke clusters.
Which service must the engineer configure on the local cluster?
A. intra-Cluster Communication Signaling
B. Location Conveyance on intercluster SIP trunks
C. Intercluster Lookup Service
D. Mobility Cross Cluster
Answer: (SHOW ANSWER)

NEW QUESTION: 226
An engineer configures a Cisco Unified Border Element and must ensure that the codecs negotiated meet the ITSP requirements. The ITSP supports G.711ulaw and G.729 for audio and H.264 for video. The preferred voice codec is G.711. Which configuration meets this requirement?

A.
```
voice class codec 10
 codec preference 1 g711ulaw
 codec preference 2 g729r8
 video codec h264

dial-peer voice 101 voip
 session protocol sipv2
 destination e164-pattern-map 1
 voice-class codec 100
```

B.

```
voice class codec 10
 codec preference 1 g711ulaw
 codec preference 2 g729r8

dial-peer voice 101 voip
 session protocol sipv2
 destination e164-pattern-map 1
 voice-class codec 10
```

C.

```
voice class codec 10
 codec preference 1 g711ulaw
 codec preference 2 g729r8
 video codec h264

dial-peer voice 101 voip
 session protocol sipv2
 destination e164-pattern-map 1
 voice-class codec 10
```

```
voice class codec 10
 codec preference 1 g729r8
 codec preference 2 g711ulaw
 video codec h264

dial-peer voice 101 voip
 session protocol sipv2
 destination e164-pattern-map 1
 voice-class codec 10
```

D.

Answer: C (LEAVE A REPLY)

https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/cube- codec-basic.html

NEW QUESTION: 227

Which two parameters influence the total number of supported conference participants on a Cisco IOS XE router that has DSP modules? (Choose two.)

A. voice codec

B. session capacity of the PVDM module

C. number of protocol data units

D. software version Of the router

E. license types

Answer: A,B (LEAVE A REPLY)

These are two parameters that influence the total number of supported conference participants on a Cisco IOS XE router that has DSP modules. A voice codec is a software algorithm that compresses and decompresses voice signals for transmission over a network. Different voice codecs have different bandwidth requirements and quality levels. A PVDM module is a hardware component that provides digital signal processing (DSP) resources for voice applications such as conferencing and transcoding. A PVDM module has a fixed session capacity, which is the maximum number of voice channels that it can support simultaneously.

NEW QUESTION: 228

Refer 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=381c1aba7a78002c558eda31-12b8af63
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,UPDATE,SUBSCRIBE,INFO
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
m=audio 32136 RTP/AVP 114 9 124 113 115 0 8 116 18
c=IN IP4 10.10.10.84
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114
maxplaybackrate=16000;sprop-maxcapturerate=16000;maxaveragebitrate=64000;stereo=0;sprop-
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. PRACK
B. UPDATE
C. SUBSCRIBE
D. NOTIFY
E. REGISTER
Answer: (SHOW ANSWER)
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.

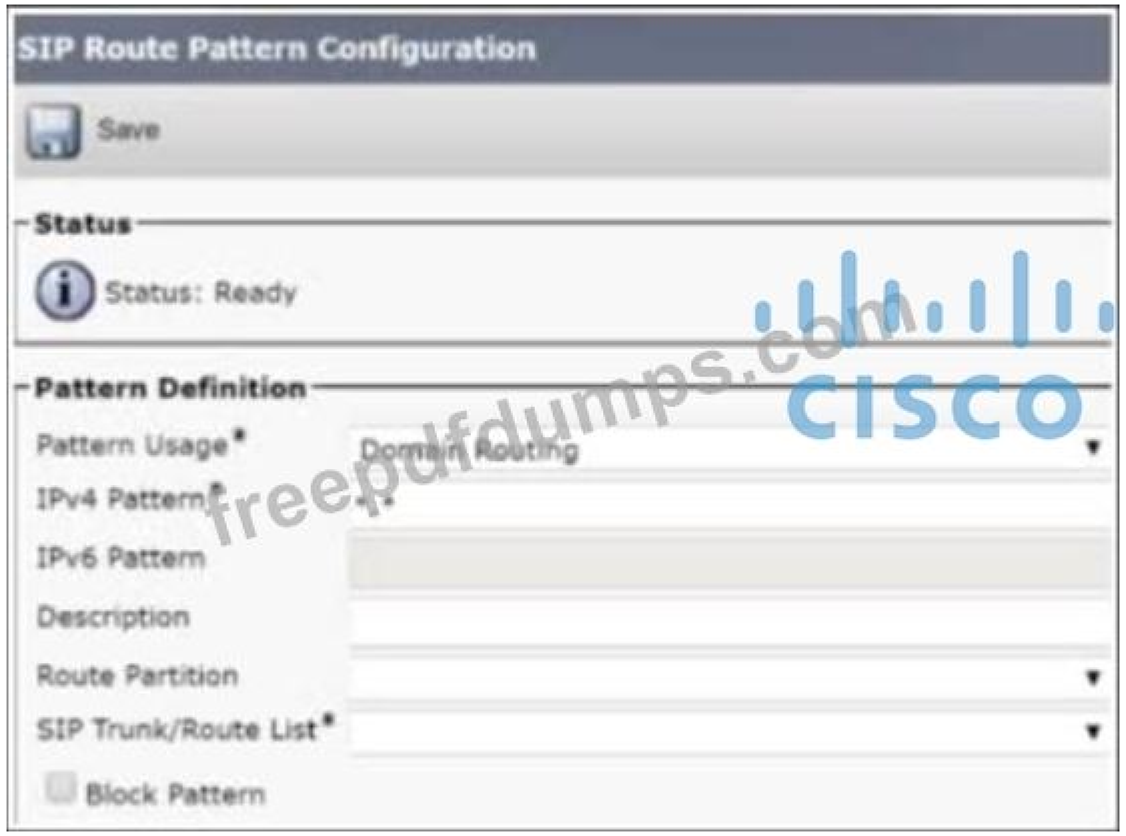
NEW QUESTION: 229
An engineer wants to manually deploy a Cisco Webex DX80 video endpoint to a remote user.
Which type of provisioning is configured on the endpoint?
A. CUBE
B. CMS

C. CUCM
D. Edge
Answer: (SHOW ANSWER)
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%3A307505

NEW QUESTION: 230
Which packet delay is the maximum supported between Cisco Unified Communications Manager nodes for clustering over WAN deployments?
A. 80 ms round trip
B. 40 ms round trip
C. 510 ms round trip
D. 150 ms round trip
Answer: A (LEAVE A REPLY)
Cisco Collaboration System 12.x Solution Reference Network Designs (SRND):
The call processing service of Unified CM does support the deployment of a single cluster's call processing nodes across an IP WAN as long as the total end-to-end round-trip time between the nodes does not exceed 80 ms and an appropriate quantity of QoS-enabled bandwidth is provisioned. The maximum one-way delay between any two Unified CM servers should not exceed 40 ms, or 80 ms round-trip time.
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12/models.html

NEW QUESTION: 231
Which SNMP service must be activated manually on the Cisco Unified Communications Manager after installation?
A. Cisco CallManager SNMP
B. SNMP Master Agent
C. Connection SNMP Agent
D. Host Resources Agent
Answer: A (LEAVE A REPLY)
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_C13
6FE37_00_cisco-unified-serviceability-administration-90/CUCM_BK_C136FE37_00_cisco- unified-serviceability-administration-guide_chapter_0101.html

NEW QUESTION: 232
Refer to the exhibit. In Cisco UCM, an engineer must create a new SIP route pattern that must be reachable by all Internal callers. The internal callers have a calling search space named CSS_INTERNAL that includes a partition named PT_INTERNAL. The SIP trunk has a calling search space named CSS_SIP_RP that includes the two partitions named PT_SIP_INTERNAL and PT_SIP_LOCAL. What must be used as the route partition for the SIP route pattern?



A. PT_SIP_INTERNAL
B. CSS_INTERNAL
C. PT_SIP_LOCAL
D. PT_INTERNAL
Answer: (SHOW ANSWER)
Routepattern or SIP Route pattern is something that is directing from CUCM to a different/outside destination - > Phone has to reach this pattern.
Sip-Trunk CSS is not required at all, because SIP-Trunk only use "incoming CSS" for routing decisions Phone CSS_INTERNAL PT_INTERNAL Trunk CSS_SIP_RP PT_SIP_INTERNAL PT_SIP_LOCAL

NEW QUESTION: 233
Refer to the exhibit. What is the reason for this call failure?

```
92195663.002 |12:41:19.104 |AppInfo  |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.1.1.1 on
port 32814 index 198 with 1554 bytes:
[887563054,NET]
INVITE sip:+12222222222@20.2.2.2:5095 SIP/2.0
Via: SIP/2.0/TCP 10.1.1.1:5095;branch=z9hG4bK3dce30168ab3a48
From: <sip:+13333333333@abc.com>;tag=119828044-fe171de1-7ea6-4a70-adc1-5c722bb7010f-266705388
To: <sip:+12222222222@20.2.2.2>
Date: Tue, 22 Dec 2020 18:41:19 GMT
Call-ID: 46877700-1ec16288-36f2729-f0607a2@10.1.1.1
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM12.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Supported: X-cisco-srtp-fallback,X-cisco-original-called
Call-Info:
<urn:x-cisco-remotecc:callinfo>;x-cisco-loc-id=e2cc7a6e-99f7-6f4b-a868-c639c0a4998f;x-cisco-loc-name=OffNet_
oc;x-cisco-fateshare-id=cucmsme:266705387;x-cisco-video-traffic-class=MIXED
Session-ID: e6cac8f0d533d35811ee2ab119828043;remote=00000000000000000000000000000000
Cisco-Guid: 1183282944-0000065536-0009379795-0252053410
Session-Expires:  1800
P-Asserted-Identity: <sip:+13333333333@abc.com>
Remote-Party-ID: <sip:+13333333333@abc.com>;party=calling;screen=yes;privacy=off
Contact: <sip:+13333333333@10.1.1.1:5095;transport=tcp>
Max-Forwards: 67
Content-Type: application/sdp
Content-Length: 275
[887563059,NET]
SIP/2.0 488 Not Acceptable Media
Via: SIP/2.0/TCP 10.1.1.1:5095;branch=z9hG4bK3dce30168ab3a48
From: <sip:+13333333333@abc.com>;tag=119828044-fe171de1-7ea6-4a70-adc1-5c722bb7010f-266705388
To: <sip:+12222222222@20.2.2.2>;tag=300491835-3e79dabf-55a9-4df7-b733-dd0511327a60-98363145
Date: Tue, 22 Dec 2020 18:41:19 GMT
Call-ID: 46877700-1ec16288-36f2729-f0607a2@10.1.1.1
CSeq: 101 INVITE
Allow-Events: presence
Server: Cisco-CUCM12.5
Session-ID: 00000000000000000000000000000000;remote=e6cac8f0d533d35811ee2ab119828043
Warning: 370 20.2.2.2 "Insufficient Bandwidth"
Reason: Q.850; cause=125
Remote-Party-ID: "Branch Line 1 - 20000"
<sip:+12222222222@20.2.2.2;user=phone>;party=x-cisco-original-called;privacy=off
Content-Length: 0
```

A. MRGL contains more media groups than the endpoint can support.

B. The available location bandwidth between sites is exceeded.

C. The hunt group contains more devices than the endpoint can support.

D. The device pool contains more call processing agents in the CMG group than the endpoint can support.

Answer: B (LEAVE A REPLY)

NEW QUESTION: 234

What is a capability of Cisco Expressway?

A. It provides access to on-premises Cisco Unified Communications infrastructure for remote endpoints

B. It functions as an analog telephony adapter

C. It has remote endpoint enrollment with Certificate Authority Proxy Function

D. It gives directory access for remote users via Cisco Directory Integration

Answer: A (LEAVE A REPLY)

NEW QUESTION: 235

A Cisco UCM administrator sets up new route patterns to support phones in four different locations, all with local gateways. The administrator wants to use the same route pattern for all four locations.

How must the system be configured to achieve this goal?

A. Use standard local route groups.

B. Use transforms in the route groups.

C. Add a CSS to each local gateway.

D. Use CSS alternate routing rules.

Answer: (SHOW ANSWER)

NEW QUESTION: 236

A recent audit discovered incidents of toll fraud. Restrictions should include Cisco UCM class of service and supplementary services features. Which two actions in Cisco UCM prevent further toll fraud from happening? (Choose two.)

A. Implement transformation pattern for all off-net calls.

B. Set the call forward all CSS to restrict international patterns.

C. Restrict international patterns on SIP trunk CSS to the gateway.

D. Set the service parameter to block off-net to off-net transfers.

E. Implement a class of restrictions on the gateway.

Answer: (SHOW ANSWER)

NEW QUESTION: 237

A network engineer at Cisco.com has 30 video endpoints registered to Cisco UCM 12.5.x. Which type of pattern must the engineer create to allow the video endpoints to dial meet.cisco.com?

A. Translation pattern with the pattern: meet.cisco.com, choose the correct partition and route list.

B. Route pattern with the pattern: meet.cisco.com, choose the correct partition and route list.

C. SIP route pattern with the pattern: meet.cisco.com, and choose the correct partition and route list.

D. Route group with the pattern: meet.cisco.com, choose the correct partition and route list.

Answer: (SHOW ANSWER)

To enable video endpoints registered to Cisco Unified Communications Manager (CUCM) to dial an external SIP address like meet.cisco.com, a SIP Route Pattern must be created. This pattern specifies how CUCM should handle SIP URI-based dialing.

SIP Route Pattern:

- SIP route patterns allow CUCM to match and route SIP URIs to specific destinations, such as external SIP domains or services.
- In this case, the pattern meet.cisco.com matches any call attempting to dial this URI and routes it appropriately.

Partition and Route List:

- The partition organizes route patterns and ensures proper call routing.
- The route list directs the call to the correct trunk or gateway (e.g., an Expressway-C/E or a SIP trunk to Cisco Webex).

Why meet.cisco.com?

- This pattern ensures that calls to this external domain are handled as SIP-based calls and routed via the configured route list.

NEW QUESTION: 238

An engineer is notified that the Cisco TelePresence MX800 that is registered in Cisco Unified communications Manager shows an empty panel, and the Touch 10 shows a corresponding icon with no action when pressed. Where does the engineer go to remove the inactive custom panel?

A. The phone configuration page in CUCM Administration

B. The SIP Trunk security profile page in CUCM Administration

C. The software Upgrades page in CUCM OS Administration
D. The In-Room control Editor on the webpage of the MX800
Answer: D (LEAVE A REPLY)
Perform the following steps to remove the in-room control panel and icon from Touch10:
a. Launch the in-room control editor from the video system's web interface.
b. Select the panel to be removed (Global, Homescreen or Incall)
c. Click Recycle bin ( )in the lower right corner.
https://www.cisco.com/c/dam/en/us/td/docs/telepresence/endpoint/ce82/sx-mx-in-room-control- guide-ce82.pdf

NEW QUESTION: 239
Cisco UCM users with Cisco 8800 series IP telephones require access to a DTMF-based IVR solution. The IVR is connected via SIP through a Cisco Unified Border Element. Which configuration uses in-band signaling for the users' keypresses?
A. dtmf-relay rtp-nte
B. dtmf-relay sip-inband
C. dtmf-relay sip-notify
D. dtmf-relay sip-kpml
Answer: A (LEAVE A REPLY)

NEW QUESTION: 240
What is the most efficient way to configure a speed dial to number 1111 on multiple phones using the Self-Provisioning feature in Cisco UCM?
A. Configure a universal device template that has speed dials in the phone button configuration field, then specify 1111 as a speed dial by editing the speed dial within the universal device template.
B. Configure the phones with a phone button template that has speed dials then specify 1111 as a speed dial via BAT>Phones>Add/Update phones web page to specify the speed dial number after the phone is in Cisco UCM.
C. Configure each phone with the same DN and then configure a speed dial of 1111 on one of the phones manually by navigating to Devices>Phones and modifying the phone. This will automatically replicate to all the other phones.
D. Configure a universal device template that has speed dials in the phone button configuration field, then allow the user to specify 1111 as a speed dial via the Cisco Unified Communications Self Care Portal.
Answer: A (LEAVE A REPLY)

NEW QUESTION: 241
When an 8800 phone is registered over Mobile and Remote Access, where does it register?
A. Cisco Unified Presence Server
B. Expressway-E
C. Cisco UCM
D. Expressway-C
Answer: (SHOW ANSWER)

NEW QUESTION: 242

What is the validity period of the ITL Recovery certificate in Cisco UCM?

A. 1 year

B. 20 years

C. 5 years

D. 10 years

Answer: B (LEAVE A REPLY)

The validity of ITLRecovery has been extended from 5 years to 20 years to ensure that the ITLRecovery certificate remains same for a longer period Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/security/12_0_1/secugd/cucm_b_cu cm-security-guide-1201/cucm_b_cucm-security- guide-1201_chapter_011.html

NEW QUESTION: 243

Regarding SIP integrations with Cisco UCM, if the Cisco Unity Connection is configured to listen for incoming IPv4 and IPv6 traffic, how is the addressing mode set up in the Cisco Unity Connection?

A. Set up for each group to use IPv4 and IPv6.

B. Set up media ports for each port group to use IPv4.

C. Set up IPv4 and IPv6 in Cisco UCM.

D. Set up is not required.

Answer: D (LEAVE A REPLY)

NEW QUESTION: 244

Which two protocols can be configured for the Cisco Unity Connection and Cisco UCM integration? (Choose two.)

A. H. 323

B. SIP

C. SCCP

D. MGCP

E. RTP

Answer: B,C (LEAVE A REPLY)

The two protocols that can be configured for the Cisco Unity Connection and Cisco UCM integration are SIP and SCCP.

SIP, or Session Initiation Protocol, is a signaling protocol used for initiating, maintaining, and terminating real-time sessions, including voice, video, and messaging applications.

SCCP, or Skinny Client Control Protocol, is a Cisco proprietary signaling protocol used for controlling Cisco IP phones.

H. 323 is an older signaling protocol that is no longer widely used. MGCP, or Media Gateway Control Protocol, is a protocol used for controlling media gateways. RTP, or Real-time Transport Protocol, is a protocol used for transporting real-time data, such as voice and video

NEW QUESTION: 245

Which set of QoS commands must be configured on an interface of a catalyst switch so the ingress traffic can be trusted for a desk phone?

A. Switch(config)# interface gigabitethernet2/0/1
Switch(config-if)# mis qos trust device cisco-phone

B. Switch(config)# interface gigabitethernet2/0/1
Switch(config-if)# qos mis trust device cisco-phone

C. Switch(config)# interface gigabitethernet2/0/1
Switch(config-if)# mis trust qos device desk-phone

D. Switch(config)# interface gigabitethernet2/0/1
Switch(config-if)# qos trust device desk-phone

Answer: (SHOW ANSWER)

NEW QUESTION: 246
Which two types of device are supported by the Bulk Administration Tool? (Choose two.)
A. Cisco Unified IP phones (all models)
B. H.323 clients
C. H.225 trunks
D. Music on hold servers
E. SIP trunks
Answer: A,B (LEAVE A REPLY)
You can use BAT to work with the following types of devices and records:
Add, update, and delete Cisco Unified IP Phones including voice gateway (VG) phones, computer telephony interface (CTI) ports, and H.323 clients, and migrate phones from Skinny Client Control Protocol (SCCP) to Session Initiation Protocol (SIP).
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/bat/12_5_1/cucm_b_bulk- administration-guide-1251/cucm_b_bulk-administration-guide-1251_chapter_01.html

NEW QUESTION: 247
Which traffic class is part of the 11 classes of the QoS baseline model?
A. realtime
B. critical data
C. network control
D. transactional
Answer: D (LEAVE A REPLY)
The 11 classes of the QoS baseline model, defined by Cisco, include the following:
1. Network Control (e.g., routing protocols, control traffic)
2. Signaling (e.g., call signaling protocols such as SIP, H.323)
3. Broadcast Video (e.g., one-way live video streams)
4. Realtime Interactive (e.g., two-way voice and video traffic)
5. Multimedia Streaming (e.g., video streams with less interactivity)
6. Mission-Critical Data (e.g., high-priority business data)
7. Transactional Data (e.g., database interactions, financial apps)
8. Bulk Data (e.g., backups, large file transfers)
9. Scavenger (e.g., traffic with the lowest priority)
10. Best Effort (e.g., typical non-critical traffic)
11. Network Management (e.g., SNMP, NTP traffic)
Reference:
https://www.cisco.com/en/US/technologies/tk543/tk759/technologies_white_paper0900aecd8029
5a9b.pdf

NEW QUESTION: 248
Refer to the exhibit. An engineer must configure DHCP service on the Cisco UCM. In which field must the engineer configure the server IP to provide the configuration files to the endpoints?


DHCP Subnet Information

DHCP Server*	-- Not Selected --
Subnet IPv4 Address*	
Primary Start IPv4 Address*	
Primary End IPv4 Address*	
Secondary Start IPv4 Address	
Secondary End IPv4 Address	
Primary Router IPv4 Address	
Secondary Router IPv4 Address	
IPv4 Subnet Mask*	
Domain Name	
Primary DNS IPv4 Address	
Secondary DNS IPv4 Address	
TFTP Server Name(Option 66)	
Primary TFTP Server IPv4 Address(Option 150)	
Secondary TFTP Server IPv4 Address(Option 150)	
Bootstrap Server IPv4 Address	
ARP Cache Timeout(sec)*	0
IP Address Lease Time(sec)*	0
Renewal(T1) Time(sec)*	0
Rebinding(T2) Time(sec)*	0

A. Primary Router IPv4 Address

B. Primary TFTP Server IPv4 Address (Option 150)

C. Primary Start IPv4 Address

D. Bootstrap Server IPv4 Address

Answer: B (LEAVE A REPLY)

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_ C95ABA82_00_admin-guide-100/ CUCM_BK_C95ABA82_00_admin-guide-100_chapter_01010.html

NEW QUESTION: 249

What is an optional component in a Mobile and Remote Access solution?

A. Cisco Expressway-C

B. Cisco Expressway-E

C. Cisco UCM

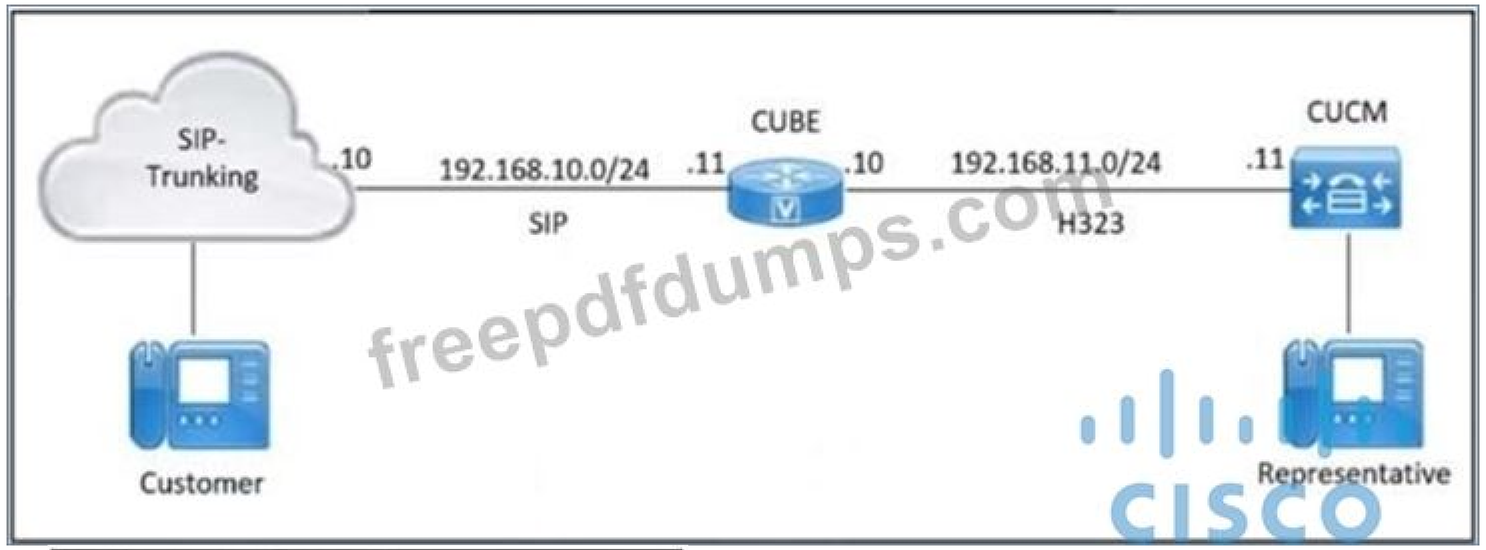
D. Cisco Unity Connection

Answer: (SHOW ANSWER)

Cisco Unity Connection is the voicemail system in a Cisco Collaboration environment. While it is commonly deployed alongside MRA solutions to provide voicemail services to remote users, it is not a mandatory component of the MRA architecture itself. MRA primarily focuses on enabling secure communication and remote access for endpoints.
In a Mobile and Remote Access (MRA) solution, the following components are essential:
- Cisco Expressway-C: The internal component of the MRA architecture. It connects to Cisco UCM and facilitates communication between the internal network and Cisco Expressway-E.
- Cisco Expressway-E: The external-facing component that resides in the DMZ. It provides firewall traversal and secures communication for remote endpoints.
- Cisco Unified Communications Manager (UCM): Acts as the call control system, managing SIP endpoints and providing call routing.
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/expressway/config_guide/X12-5/exwy_b_mra-expressway-deployment-guide/exwy_b_mra-expressway-deployment- guide_chapter_00.html

NEW QUESTION: 250
Refer to the exhibit. An engineer must implement toll fraud prevention on a Cisco UCM cluster by allowing only the indicated IP address and protocols through Cisco Unified Border Element. What must be configured?

```
!
voice service voip
 allow-connections h323 to sip
 supplementary-service media-renegotiate
 sip
 registrar server expires max 120 min 120
!
!
dial-peer voice 1 voip
 destination-pattern 4422...
 session protocol sipv2
 session target ipv4:192.168.11.11
 incoming called-number 4433...
 direct-inward-dial
 dtmf-relay sip-notify
 codec g711ulaw
!
dial-peer voice 2 voip
 destination-pattern 91...
 session protocol sipv2
 session target ipv4:192.168.10.10
 incoming called-number 2...
 forward-digits 4
!
```

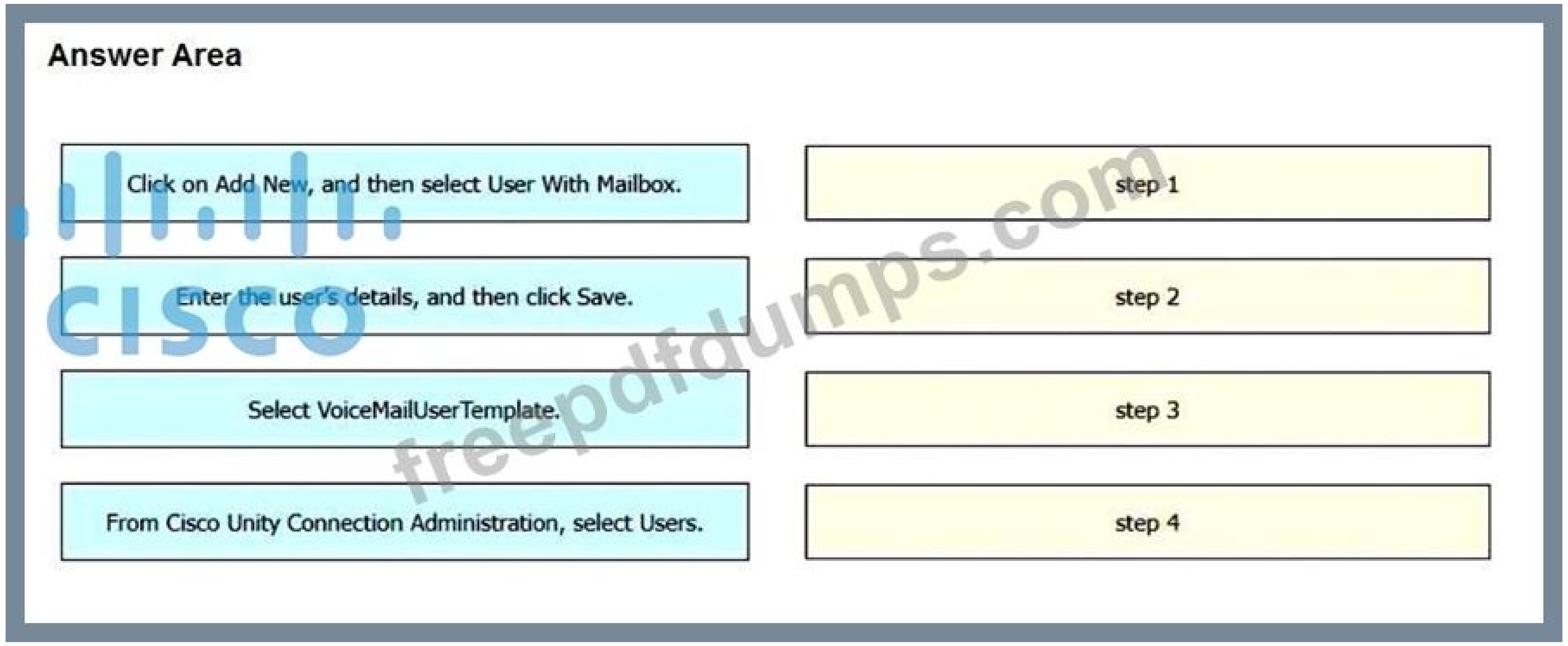


A.
```
voice service voip
 allow-connections sip to h323
 ip address trusted list
  ipv4 192.168.10.10
```
B.
```
voice service voip
 allow-connections sip to sip
 ip address trusted list
  ipv4 192.168.11.10
```
C.
```
voice service voip
 allow-connections h323 to sip
 ip address trusted list
  ipv4 192.168.11.11
```
D.
```
voice service voip
 allow-connections sip to h323
 ip address trusted list
  ipv4 192.168.10.11
```
Answer: C (LEAVE A REPLY)

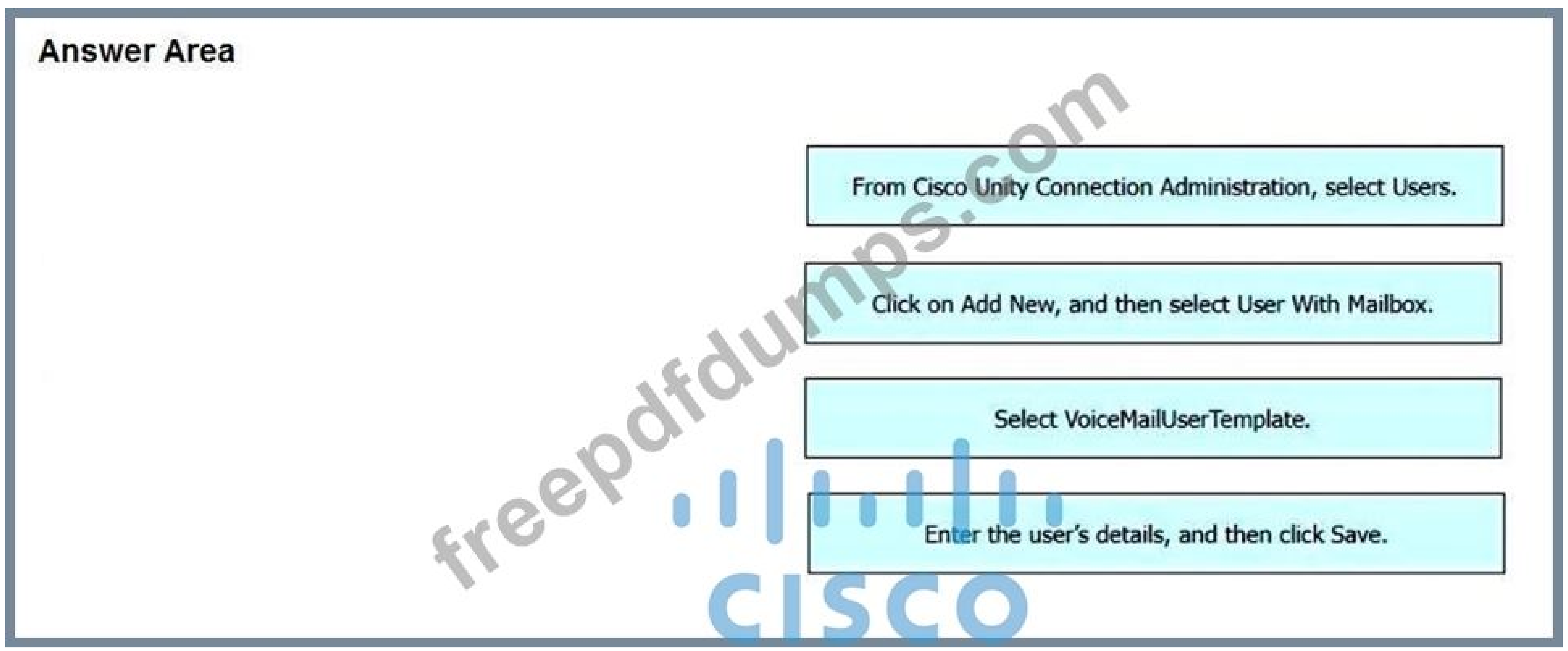
NEW QUESTION: 251

Drag and Drop Question

An engineer must add a new user account and mailbox in Cisco Unity Connection. Drag and drop the configuration steps from the left into the correct order on the right.



Answer:



NEW QUESTION: 252

During the Cisco IP Phone registration process the TFTP download fails. What are two reasons for this issue? (Choose two)

A. The Cisco IP Phone does not know the IP address of the TFTP server

B. Option 150 string was not specified, or an incorrect Option 150 string was specified

C. The Cisco IP Phone does not know the IP address of any of the Cisco UCM Subscriber nodes

D. Option 100 string was not specified, or an incorrect Option 100 string was specified

E. The DNS server was not specified, which is needed to resolve a hostname in an Option 150 string.

Answer: A,B (LEAVE A REPLY)

NEW QUESTION: 253

An engineer is configuring IP telephony. The network relies on DHCP to provide TFTP server addresses to the endpoints. Policy requires the endpoints to receive two server addresses. Which DHCP option must be configured?

A. 66

B. 143

C. 150

D. 166

Answer: C (LEAVE A REPLY)

Option 66 provides a single TFTP server address. Option 150 provides a list of TFTP server addresses. Also if both are configured, Cisco phones will target 150 by default before 66.

NEW QUESTION: 254

Which DTMF relay method configured on a SIP dial-peer will ensure that a media resource is not invoked by Unified CM for calls to UCCX IVRs?

A. dtmf-relay slp-kpml

B. dtmf-relay rtp-nte

C. sdtmf-relay h245-slgnal

D. dtmf-relay sip-notify

Answer: A (LEAVE A REPLY)

UCCX only supports out-of-band DTMF. This can be applied at the CUBE to avoid a media resource being needed to address as it can directly handle the DTMF relay from RTP-NTE to SIP-KPML
https://www.cisco.com/c/dam/en/us/td/docs/ios-xml/ios/voice/cube_uccx/cube_uccx_interop_bestprac_guide.pdf

NEW QUESTION: 255

A Cisco Unity Connection administrator receives a name change request from a voice-mail user, whose Cisco Unity Connection user account was imported from Cisco Unified Communications Manager. What should the administrator do to execute this change?

A. Change the user data in the Cisco Unity Connection administration page, then use the Synch User page in Cisco Unity Connection administration to push the change to Cisco Unified Communications Manager.

B. Change the user data in the Cisco Unified Communications Manager administration page, then use the Synch User page in Cisco Unity Connection administration to pull the changes from Cisco Unified CM.

C. Change the user data in the Cisco Unified Communications Manager administration page, then use the Synch User page in Cisco Unified CM administration to push the change to Cisco Unity Connection.

D. Change the user profile from Imported to Local on Cisco Unity Connection Administration, then edit the data locally on Cisco Unity Connection.

E. Change the user data in Cisco Unity Connection and Cisco Unified Communications Manager separately.

Answer: B (LEAVE A REPLY)

As we can see user are getting synch from call manager so we first have to change the details of user on call manager so that user will synch the changes from call manager.

NEW QUESTION: 256

What are two functions of Cisco Expressway in the Collaboration Edge? (Choose two)

A. Expressway-E provides a VPN entry point for Cisco IP phones with a Cisco AnyConnect client using authentication based on certificates.

B. Expressway-C provides encryption for Mobile and Remote Access but not for business-to- business communications.

C. The Expressway-C and Expressway-E pair can interconnect H.323-to-SIP calls for voice.

D. The Expressway-C and Expressway-E pair can enable connectivity from the corporate network to the PSTN via a T1/E1 trunk.

E. Expressway-E provides a perimeter network that separates the enterprise network from the Internet.

Answer: C,E (LEAVE A REPLY)

https://community.cisco.com/t5/unified-communications/cisco-expressway-secure-b2b-tls- encrypted/td-p/2957680

NEW QUESTION: 257
Which information is needed to restore the backup of a Cisco UCM publisher successfully?
A. the TFTP server details
B. the FTP server details
C. the application credentials for Cisco UCM
D. the security password for Cisco UCM
Answer: D (LEAVE A REPLY)

NEW QUESTION: 258
Refer to the exhibit. An engineer must provide a list of codecs based on the media attribute of an SDP body snippet. Which codecs are included in the media attribute ordered by priority?

```
m=audio 50024 RTP/AVP 0 8 9 18
```

A. G.711alaw, G.711ulaw, G.729, and G.722
B. G.711ulaw, G.711alaw, G.722, and G.729
C. G.711ulaw, G.722, G.711alaw, and G.729
D. G.722, G.711alaw, G.729, and G.711ulaw
Answer: (SHOW ANSWER)

NEW QUESTION: 259
A Cisco voice gateway is configured to use a sip-kpml DTMF relay in global settings. A new SIP dial peer is configured for a third-party application that only supports an in-band DTMF relay.
Which commands must an engineer run on the dial peer?
A. dtmf-relay sip-info
B. dtmf-relay sip-notify
C. dtmf-relay rtp-net
D. no dtmf-relay sip-kpml
Answer: C (LEAVE A REPLY)

NEW QUESTION: 260
What is a statement about shadow locations in Cisco UCM?
A. Shadow location enables a SIP intercluster trunk to pass enhanced location-based CAC information between Unified CM clusters.
B. Shadow location enforces low-bandwidth codec for calls between Unified CM clusters.

C. Only the intercluster trunk can be assigned to shadow location.

D. Shadow location enables a non-SIP intercluster trunk to pass enhanced location-based CAC information between Unified CM clusters.

Answer: A (LEAVE A REPLY)

NEW QUESTION: 261

Where is Native Call Queuing configured in Cisco UCM?

A. hunt list

B. hunt pilot

C. line group

D. Advanced Features menu

Answer: (SHOW ANSWER)

NEW QUESTION: 262

Refer to the exhibit. An engineer configures a VoIP dial peer on a Cisco gateway.

Which codec is used?

```
dial-peer voice 10 voip
         destination-pattern 1…
         session target ipv4:10.1.1.1
         no vad
```

A. G.711ulaw

B. No codec is used (missing codec command).

C. G.711alaw

D. G.729r8

Answer: D (LEAVE A REPLY)

Default codec for POTS is g.711

Default codec for VOIP is g.729

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmead m/cmebasic.html#concept_2A67E83FAC004CF5AC02FCFF49FCE6BB

NEW QUESTION: 263

Which DSCP class selector is necessary to mark scavenger traffic?

A. CS1

B. CS2

C. AF11

D. AF21

Answer: A (LEAVE A REPLY)

NEW QUESTION: 264

Which two media encryptions should be used when configuring SRTP between Cisco UCM and Cisco VCS Expressway? (Choose two.)

A. none

B. pass-through

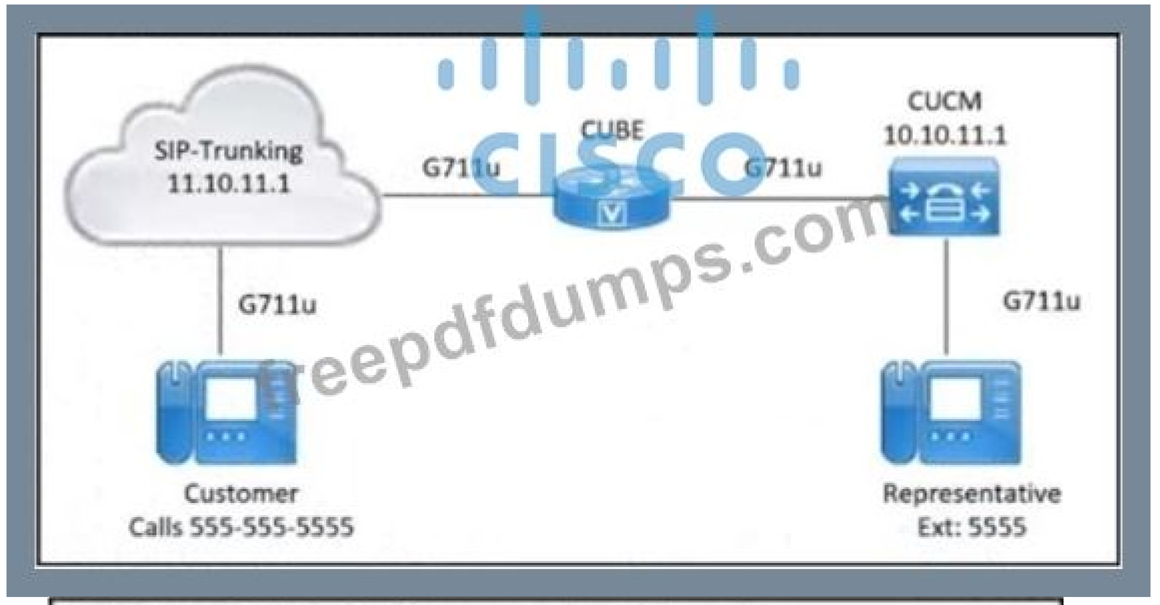
C. priority

D. mandatory
E. optional
Answer: A,D (LEAVE A REPLY)

NEW QUESTION: 265
An engineer must configure a voice translation rule on a Cisco voice gateway to truncate numbers down to the last four digits. Which voice translation regex rule must the engineer use?
A. rule 1 /^.*\(....\)/ /\1/
B. rule 1 /*1(2-9]..../ /\1/
C. rule 1 /^.*\(..\)/ /\4/
D. rule 1 /^[2-9]\(..\)/ /\.3/
Answer: A (LEAVE A REPLY)
A voice translation rule on a Cisco voice gateway uses regular expressions (regex) to match and modify numbers. To truncate a number down to its last four digits, the regex pattern must match the entire number and capture only the last four digits for replacement.
rule 1 /^.*\(...\)/ /\1/
- ^.*: Matches any leading digits before the last four digits.
- \(...\): Captures the last four digits in parentheses. The parentheses create a capture group.
- /\1/: Replaces the entire number with the content of the first capture group (the last four digits).

NEW QUESTION: 266
Refer to the exhibit. An engineer must deploy a Cisco Unified Border Element that will allow the last four digits of a dialed number to be sent from the Cisco Unified Border Element to Cisco UCM. The configuration in the exhibit already exists on die Cisco Unified Border Element. Which additional configuration must be performed on the Cisco Unified Border Element?

```
CUBE(config)# <output omitted>
CUBE(config)# voice translation-rule 2
CUBE(config-translation-rule)# rule 1 /^5555555555$/ /5555/
CUBE(config)# voice translation-profile profile1
CUBE(cfg-translation-profile)# translate called 2
CUBE(cfg-translation-profile)# exit
CUBE(config)# dial-peer voice 10 voip
CUBE(config-dial-peer)# session target ipv4:10.10.11.1
CUBE(config-dial-peer)# session protocol sipv2
CUBE(config-dial-peer)# dtmf-relay rtp-nte
CUBE(config-dial-peer)# codec G711u
CUBE(config-dial-peer)# no vad
CUBE(config-dial-peer)# end
CUBE(config)# dial-peer voice 11 voip
CUBE(config-dial-peer)# incoming called-number .
CUBE(config-dial-peer)# session target ipv4:11.10.11.1
CUBE(config-dial-peer)# session protocol sipv2
CUBE(config-dial-peer)# dtmf-relay rtp-nte
CUBE(config-dial-peer)# codec G711u
CUBE(config-dial-peer)# no vad
CUBE(config-dial-peer)# end
CUBE(config)# <output omitted>
```



A.
```
CUBE(config)# dial-peer voice 10 voip
CUBE(config-dial-peer)# session target 5555
```
B.
```
CUBE(config)# dial-peer voice 10 voip
CUBE(config-dial-peer)# destination-pattern 5555$
```
C.
```
CUBE(config)# dial-peer voice 11 voip
CUBE(config-dial-peer)# destination-pattern 5555
```
D.
```
CUBE(config)# dial-peer voice 11 voip
CUBE(config-dial-peer)# incoming called-number 5555
```
Answer: (SHOW ANSWER)

NEW QUESTION: 267
Cisco Prime Collaboration Manager is using Cisco Discovery Protocol to discover devices in an automated setup. What must be used to get alerts and faults via traps from the devices at regular polling intervals?
A. HTTPS
B. SNMP
C. CDP
D. XML
Answer: B (LEAVE A REPLY)

NEW QUESTION: 268
what causes poor voice quality and video pixelization in a video call?
A. The QoS is configured incorrectly.
B. Cisco Unified Communications Manager is configured to use G.711 instead G.729.
C. 1 Gbps network ports are used instead of 100 Mbps network ports.
D. A firewall is blocking the RTP ports.
Answer: (SHOW ANSWER)

NEW QUESTION: 269
An engineer encounters third-party devices that do not support Cisco Discovery Protocol. What must be configured on the network to allow device discovery?

A. LACP
B. TFTP
C. SNMP
D. LLDP
Answer: D (LEAVE A REPLY)

NEW QUESTION: 270
A company wants to provide remote user with access to its premises Cisco collaboration features. Which components are required to enable cisco mobile and remote access for the users?
A. Cisco Unified Border Element. Cisco UCM, and Cisco Video Communication Server
B. Cisco Expressway-E, Cisco IM and Presence Server, and Cisco Video Communication Server
C. Cisco Unified Border Element, Cisco IM and Presence Server, and Cisco Video Communication Server
D. Cisco Expressway-E Cisco Expressway-C and Cisco UCM
Answer: (SHOW ANSWER)

NEW QUESTION: 271
Which four requirements are mandatory to enable a mixed mode Cisco Unified CM cluster?
(Choose four.)
A. Cisco CTL Provider Service activated and enabled
B. a minimum of one soft e-token
C. Cisco Certificate Authority Proxy Function activated and enabled
D. a minimum of one USB e-token
E. Cisco Trust Verification activated and enabled
F. Cisco CTL client
G. a minimum of two USB e-token
Answer: A,C,F,G (LEAVE A REPLY)

NEW QUESTION: 272
Which DSCP class selector is necessary to mark scavenger traffic?
A. CS1
B. AF21
C. AF11
D. CS2
Answer: (SHOW ANSWER)
NEW QUESTION: 273
An administrator needs to create a partial PRI consisting of the first seven timeslots available.

Which configuration snippet configures the ISDN E1 PRI for this task?
config t
A. 2900(config)#isdn switch-type primary-ni
2900(config)#pri-group timeslots 1-7
config t
B. 2900(config)#isdn switch-type primary-ni
2900(config)#interface Serial0/0/0:15
2900(config-controller)#pri-group timeslots 1-7
config t
C. 2900(config)#isdn switch-type primary-ni
2900(config)#controller e1 0/0/0
2900(config-controller)#pri-timeslots 1-7
config t
D. 2900(config)#isdn switch-type primary-ni
2900(config)#controller e1 0/0/0
2900(config-controller)#pri-group timeslots 1-7
Answer: (SHOW ANSWER)
Enable the ISDN Switch type as Primary-ni globally:
config t
#isdn switch-type primary-ni
Then the controller E1 and Pri group timeslot configuration:
#controller e1 0/0/0
#pri-group timeslots 1-10

NEW QUESTION: 274
What is a capability of a Cisco IOS XE media resource?
A. It provides a hardware conferencing solution.
B. It provides call forwarding capabilities.
C. It provides redundancy for voice calls.
D. It provides a voice packet optimization solution.
Answer: A (LEAVE A REPLY)
A Cisco IOS XE media resource provides a hardware conferencing solution. It can be used to mix multiple media streams, such as audio and video, into a single stream that can be sent to all participants in a conference call. This is done using a digital signal processor (DSP), which is a specialized processor that is designed to handle the processing of digital signals, such as audio and video.

NEW QUESTION: 275
Which set of four functions are performed by Cisco Unified Border Element?
A. session control, meet-me conferencing, interworking. and demarcation
B. session control, security. SIP. and H.323
C. session control, security, interworking, and demarcation
D. PRI connections, security, faxing, and border enforcement
Answer: C (LEAVE A REPLY)

NEW QUESTION: 276

Which task is required when configuring self-provisioning for an end user in Cisco UCM?

A. Enable Auto-Registration.

B. Associate the end user to the Standard CCM Super Users group.

C. Associate the end user to a SIP Profile.

D. Disable Auto-Registration.

Answer: A (LEAVE A REPLY)

This auto registration must be enabled for the self-provisioning to work due to needing the phone able to register to CUCM first.

NEW QUESTION: 277

Which configuration concept allows for high-availability on IM and Presence services in a UC environment?

A. IM and Presence subclusters (configured on Cisco UCM)

B. Presence Redundancy Groups (configured on Cisco Unified IM and Presence)

C. IM and Presence subclusters (configured on Cisco Unified IM and Presence)

D. Presence Redundancy Groups (configured on Cisco UCM)

Answer: (SHOW ANSWER)

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_ b_system-configuration-guide-1201/cucm_b_system- configuration-guide-1201_chapter_0110010.html

NEW QUESTION: 278

Which call flow matches traffic from a Mobile and Remote Access registered endpoint to central call control?

A. Endpoint > Expressway-E > Expressway-C > Cisco Unified CM

B. Endpoint > Expressway-E > Cisco Unified CM

C. Endpoint > Expressway-C > Cisco unified CM

D. Endpoint > Expressway-C > Expressway-E > Cisco Unified CM

Answer: A (LEAVE A REPLY)

NEW QUESTION: 279

Which Layer 3 PHB marking is used for VoIP applications?

A. 0

B. EF

C. CS1

D. AF21

Answer: B (LEAVE A REPLY)

NEW QUESTION: 280

An administrator configures the voicemail feature in a Cisco collaboration deployment. The user mailboxes must be configured when the Cisco Unity Connection server is configured. Which action accomplishes this task?

A. Configure an SCCP integration with Cisco UCM.

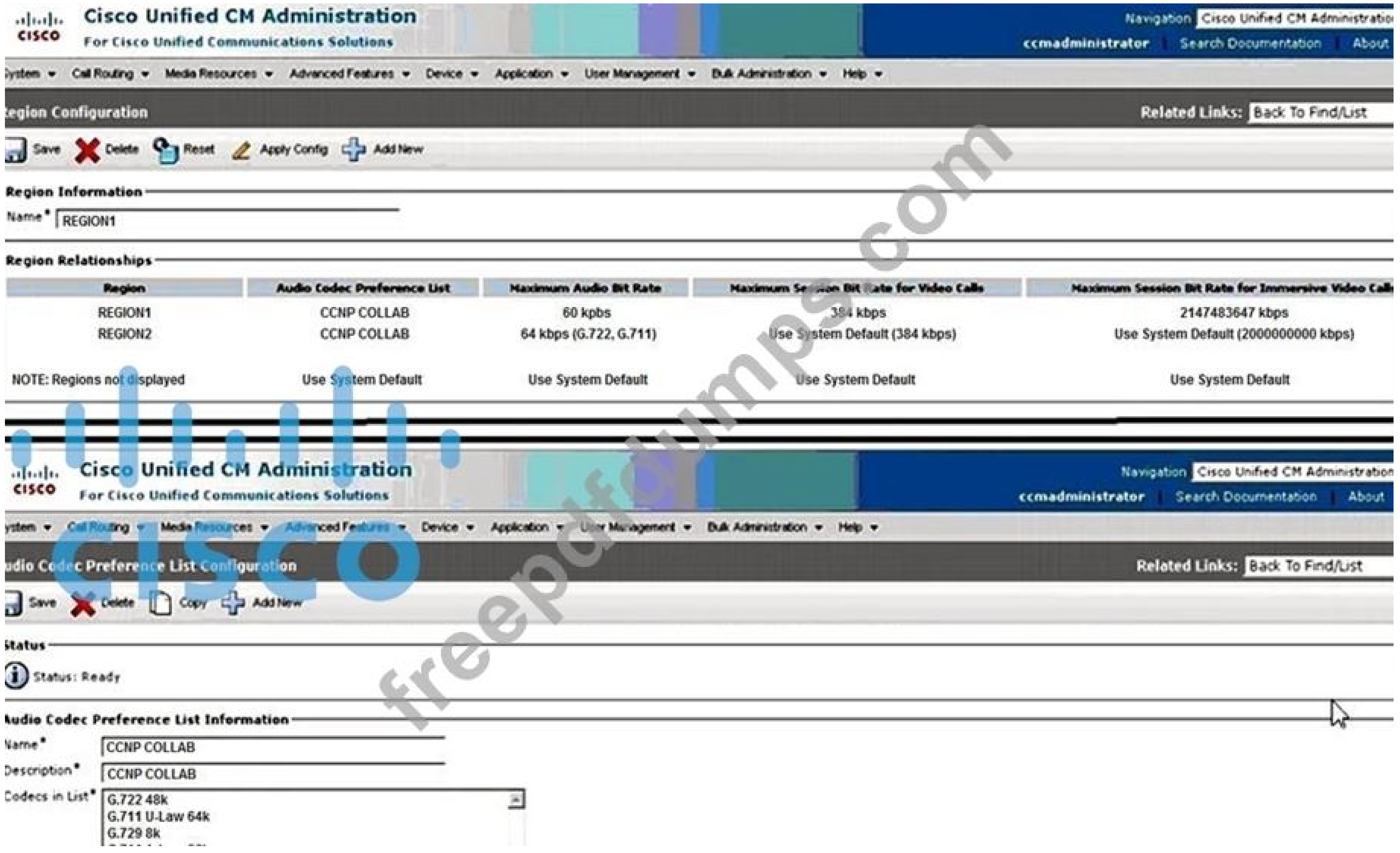
B. Configure a SIP integration with Cisco UCM to sync users.

C. Configure an AXL server to access the Cisco UCM users.

D. Configure an active directory to sync the users who will have a voicemail box.

Answer: (SHOW ANSWER)
SIP integration doesn't sync users automatically.
AXL server can be configured in CUC and then sync CUCM users to it.
Another way is LDAP / AD configuration.

NEW QUESTION: 281
Refer to the exhibit. An engineer is troubleshooting a codec negotiation issue where both endpoints that are involved in the call support the codecs listed in the exhibit.
Which audio codec is selected il a call between two endpoints in Region 1 is placed?



A. G.711u
B. G.722
C. G.729
D. G.711a
Answer: B (LEAVE A REPLY)
The codec used cannot exceed 60k. G.722 has the ability to use 48, 56, and 64k.
NEW QUESTION: 282
A user reports that when receiving an inbound call on their IP Phone from the PSTN they are unable to transfer this call to another PSTN number What is the reason for this failure?
A. The service parameter related to Offnet to Offnet Call Transfer is set to TRUE
B. The IP phone is configured with the wrong region

C. The incoming calling search space of the SIP trunk does not include the partition of the line in the IP phone
D. The gateway is configured with the wrong device pool
Answer: (SHOW ANSWER)

NEW QUESTION: 283
A collaboration engineer is configuring the QoS trust boundary for Cisco UCM voice and video conferencing. Which two trust boundary configurations are valid? (choose two)
A. QoS trust boundaries exclude Jabber softphone running on a PC
B. QoS trust boundaries include all the devices directly attached to the access switch ports
C. QoS trust boundaries can be extended to voice and video devices if the connected PCs are included
D. QoS trust boundaries can be extended to Jabber running on a PC
E. QoS trust boundaries can be extended to voice and video devices exclusively
Answer: A,C (LEAVE A REPLY)

NEW QUESTION: 284
Which service on the Presence Server is responsible for maintaining the point-to-point chat connections between Jabber clients?
A. Cisco SIP Proxy
B. Cisco XCP Text Conference Manager
C. Cisco XCP Router
D. Cisco XCP XMPP Federation Manager
Answer: (SHOW ANSWER)
XCP Router in both types of chat
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/10_
5_1/CUP0_BK_CE43108E_00_config-admin-guide-imp-105/CUP0_BK_CE43108E_00_config- admin-guide-imp-105_chapter_00.html XCP Text Conference Manager only in external DB which is for persistent chat which is only applicable for group chat
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/11_
5_1/cup0_b_config-and-admin-guide-1151su5/cup0_b_imp-system-configuration-
1151su5_chapter_010000.html#task_2BA3E8B8D67B3AA5E200B9703818BB39

NEW QUESTION: 285
Which configuration step is necessary for a Cisco SIP phone to synchronize its time with a specific source?
A. Change the Time Zone from "America/Los_Angeles" to "Etc/GMT+8".
B. Add a phone NTP Reference to the date/time Group.
C. Assign the device to the correct region.
D. Change the time format from 24-hour to 12-hour.
Answer: B (LEAVE A REPLY)
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_ C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0110.html
NEW QUESTION: 286
An engineer is trying to add a new subscriber to the cluster of Cisco UCM, and the synchronization is failing. The engineer obtained the output from the publisher and noticed the error. Based on the output, what is the issue?

```
admin:utils diagnose test

Log file: platform/log/diag1.log

Starting diagnostic test(s)
===========================
test - disk_space          : Passed (available: 6463 MB, used: 12681 MB)
skip - disk_files          : This module must be run directly and off hours
test - service_manager     : Passed
test - tomcat              : Passed
test - tomcat_deadlocks    : Passed
test - tomcat_keystore     : Passed
test - tomcat_connectors   : Passed
test - tomcat_threads      : Passed
test - tomcat_memory       : Passed
test - tomcat_sessions     : Passed
skip - tomcat_heapdump     : This module must be run directly and off hours
test - validate_network    : Passed
test - raid                : Passed
test - system_info         : Passed (Collected system information in diagnostic log)
test - ntp_reachability    : Passed
test - ntp_clock_drift     : Passed
test - ntp_stratum         : Failed
The reference NTP server is a stratum 5 clock.
```

A. The NTP server must be replaced by another server with NTP stratum 1, 2, or 3 running version NTPv4.
B. The NTP service is pointing to the wrong publisher.
C. The NTP server for Cisco UCM must be stratum 6 or higher.
D. The NTP server is running out of space in the hard drive, which prevents the cluster service from starting.
Answer: (SHOW ANSWER)

NEW QUESTION: 287
An administrator troubleshoots call flows and suspects that there are issues with the dial plan.

Which tool enables a quick analysis of the dial plan and provides call flows of dialed digits?

A. Digit Analysis Analyzer

B. Dialed Number Analyzer

C. Cisco Dial Plan Analyzer

D. Dial Plan Analyzer

Answer: (SHOW ANSWER)

https://aurus5.com/blog/cisco/configuring-dialedconfiguring-dialed-number-analyzer-in-cucm/

NEW QUESTION: 288

An administrator is designing a new Cisco UCM for a company with many departments and firm structure on their communications policies. The administrator must make sure that these communication policies are reflected in the phone system setup. Certain departments cannot be accessed directly, even if they have dedicated DID numbers. Some phones, especially public phones, must not be able to dial international numbers Which type of function is configured to control which device is allowed to call another device in Cisco UCM?

A. calling patterns and route patterns

B. links and pipes

C. partitions and calling search spaces

D. regions and device pools

Answer: C (LEAVE A REPLY)

NEW QUESTION: 289

When a new SIP phone is registered to Cisco Unified communications Manager, it keeps failing and showing an "unprovisioned" error message in the phone display. Which problem is a possible cause of this issue?

A. Auto-registration is disabled on the Cisco Unified Communications Manager nodes and the phone device does not have a DN configured.

B. The DHCP settings are incorrectly and the phone does not have an alternate TFTP defined.

C. The phone cannot download and install the latest firmware.

D. The DN assigned to the phone is already in use by another SIP phone.

E. The DN configuration for this phone is shared with SCCP phone, which is not supported.

Answer: A (LEAVE A REPLY)

https://community.cisco.com/t5/ip-telephony-and-phones/how-can-i-registration-a-sip-phone- without-dn/td-p/2765543

NEW QUESTION: 290

Which attribute contains an XMPP stanza?

A. type

B. presence

C. message

D. iq

Answer: A (LEAVE A REPLY)

NEW QUESTION: 291

The security department will audit an IT department to ensure that the proper guidelines are being followed. The reports of the call detail records show unauthorized access to PSTN. Which two actions should an administrator check to prevent the unauthorized use of the telephony system? (Choose two.)

A. Ensure that ad hoc conference calls are dropped if an external user is added.

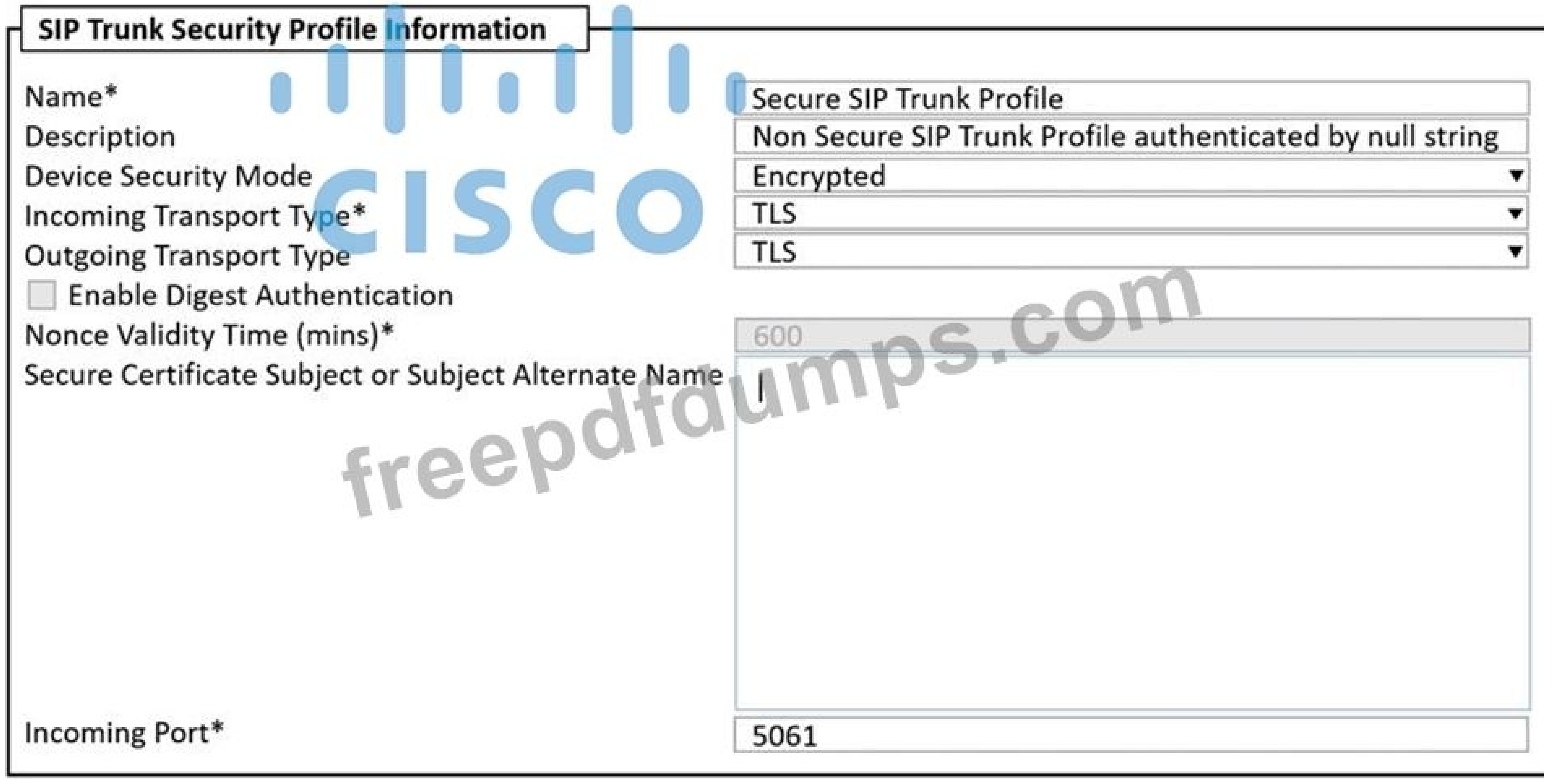
B. For extension mobility, logged-out CSS is restricted to internal extensions and emergencies.

C. Call forward settings (All/Busy/No Answer) are restricted to internal extensions in the network.
D. Forced authorization code is used to recognize a dialing extension and authorize an international call.
E. Add an additional firewall between the Cisco UCM server and the Expressway Core server.
Answer: (SHOW ANSWER)

NEW QUESTION: 292
Which two configuration elements are part of the Cisco Unified Communications Manager toll- fraud prevention? (Choose two.)
A. SIP trunk security profile.
B. Partition
C. Calling search space
D. SUBSCRIBE calling search space
E. Feature control policy
Answer: B,C (LEAVE A REPLY)
https://www.ciscopress.com/articles/article.asp?p=2218297&seqNum=10

NEW QUESTION: 293
Due 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 trunk
D. Set Video bandwidth in the Region settings to 0.
E. Change the Video Capabilities dropdown on the endpoint to Disabled.
Answer: C,D (LEAVE A REPLY)
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).

NEW QUESTION: 294
Refer to the exhibit. An administrator configures a secure SIP trunk on Cisco UCM.
Which value is needed in the Secure Certificate Subject or Subject Alternate Name field to accomplish this task?



A. the fully qualified domain name of the remote device that is configured on the SIP trunk
B. the common name of the Cisco UCM CallManager certificate
C. the common name of the remote device certificates
D. the fully qualified domain name of all Cisco UCM nodes that run the CallManager service
Answer: A (LEAVE A REPLY)
https://www.cisco.com/c/en/us/support/docs/conferencing/meeting-server/213820-configure-cisco-meeting-server-and-cucm.html

NEW QUESTION: 295
Which two describe the main components of the methodology for system sizing adopted by Cisco? (Choose two.)
A. Interaction between individual products may require the addition of Cisco UCM server nodes.
B. Cisco TelePresence conductor capacity and sizing can be considered on a region-by-region basis.
C. Performance testing measures resource usage (CPU, memory) accurately for each function.
D. System Busy Hour Call Attempts is calculated by multiplying the number of users by the maximum number of call attempts during the busiest hour of the year.
E. Busy hour call attempts, where the busy hour call attempts system is the user busy hour call attempts multiplied by the number of users.
Answer: (SHOW ANSWER)

NEW QUESTION: 296
What is the maximum number of servers that are in an IM and Presence presence redundancy group?
A. 10
B. 6
C. 2

D. 4

Answer: (SHOW ANSWER)

A presence redundancy group is comprised of two IM and Presence Service nodes from the same cluster. Each node in the presence redundancy group monitors the status, or heartbeat, of the peer node. You can configure a presence redundancy group to provide both redundancy and recovery for IM and Presence Service clients and applications.

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/11_
5_1/cup0_b_config-and-admin-guide-1151su5/ cup0_b_imp-system-configuration-
1151su5_chapter_0100.html

NEW QUESTION: 297

Where is the default for maximum session Bit Rate for a region configured?

A. Region configuration

B. Enterprise Phone configuration

C. Service parameter configuration

D. Enterprise parameters configuration

Answer: (SHOW ANSWER)

For enhanced scalability and to conserve resources, Cisco recommends that you properly set the default values in the Clusterwide Parameters (System - Location and Region) section of the Cisco Unified Communications Manager Administration Service Parameters Configuration window for the audio codec, video call bandwidth, and link loss type values and then choose the Use System Default entries in the Cisco Unified Communications Manager Administration Region Configuration window for these fields.

NEW QUESTION: 298

Refer to the exhibit. Calls to Cisco Unity Connection are failing across Cisco Unified Border Element when callers try to select a menu prompt. Why is this happening and how is it fixed?

```
Sent:
INVITE sip:2004@192.168.100.100:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.100.200:5060;branch=z9hG4bKF1FED
From: "7000" <sip:7000@192.168.100.200>;tag=43CDE-1A22
To: <sip:2004@192.168.100.100>
Call-ID: 12BCA00-3C3E11EA-01234567@192.168.100.200
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
User-Agent: Cisco-SIPGateway/IOS-16.9.5
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Contact: <sip:7000@192.168.100.200:5060>
Expires: 180
Max-Forwards: 68
P-Asserted-Identity: "7000" <sip:7000@192.168.100.200>
Session-Expires: 1800
Content-Type: application/sdp
Content-Length: 254

v=0
o=CiscoSystemsSIP-GW-UserAgent 5871 9974 IN IP4 192.168.100.200
s=SIP Call
c=IN IP4 192.168.100.200
t=0 0
m=audio 8002 RTP/SAVP 0
c=IN IP4 192.168.100.200
a=rtpmap:0 PCMU/8000
a=ptime:20
```

A. The Cisco Unity Connection Call Handler is configured for a -Release to Switch" transfer type.
Transfers must be disabled for the Cisco Unity Connection Call Handler.

B. Cisco Unity Connection is configured on G.729 only. Cisco Unity Connection must be reconfigured to support iLBC.

C. Cisco Unified Border Element is not sending any support for DTMF. DTMF configuration must be added to the appropriate dial peer.

D. Cisco Unified Border Element is sending the incorrect media IP address. The IP address of the loopback interface must be advertised in the SDP.

Answer: C (LEAVE A REPLY)